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Noise Reduction Methods for Detecting Impulses in Seismic Data

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Prepared for U.S. Army Engineer Division, South Pacific

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Preface

This report describes work sponsored by the U.S. Army Engineer Division, South Pacific. The research was performed at the U.S. Army Engineer Waterways Experiment Station (WES) by Dr. Cary B. Cox, Instrumentation Systems Development Division (ISDD), Information Technology Laboratory (ITL), and Dr. Richard Lewis, Engineering and Geosciences Division (EEGD), Geotechnical Laboratory (GL). The work was performed under the general supervision of Mr. George P. Bonner, Chief, ISDD, Dr. N. Radhakrishnan, Director, ITL, Mr. Joseph R. Curro, Jr, Chief, Engineering Geophysics Branch, EEGD, and Drs. A. G. Franklin, Chief, EEGD, and William F. Marcuson III, Director, GL.

At the time of publication of this report, Director of WES was Dr. Robert W. Whalin. Commander was COL Bruce K. Howard, EN.

Conversion Factors, Non-SI to SI Units of Measurement

Non-SI units of measurement used in this report can be converted to SI units as follows:

Multiply	By	To Obtain
feet	0.3048	meters
pounds	0.45359	kilograms

1 Introduction

Description of Problem

People are often trapped inside collapsed buildings in the aftermath of a catastrophic event such as an earthquake, tornado, or explosion. A trapped person's ability to call out for help is greatly diminished due to dehydration and weakness after several days of being trapped. Many victims of such events try to signal to rescuers by hitting or tapping on the ground or parts of the structure. One of the techniques rescuers use to detect distress calls is seismic geophones or accelerometers. These detect the elastic waves generated by the trapped person(s), which travel through the building debris. The ability of rescuers to hear these "thumping" sounds (distress calls) is hindered by the presence of background noise. These background noises include mechanical devices such as bulldozers, cranes, or other traffic. Urban environments also produce a generally high level of coherent background "noise", generated by machinery, vehicles, etc. which are in continuous operation. The goal of this effort is to improve the quality of these elastic wave signals by digitally processing them to remove the background noise.

This report discusses several methods of separating these "thumping" signals from urban "noise". Both simulated and field data are presented to assess the potential strengths and limitations of each method. Several techniques of hardware implementation are presented and recommendations for further research are discussed.

Description of System

The two most important considerations that must be investigated for a prospective noise cancellation system are portability and speed. The supporting hardware must be easily moved and operated in the field. Generally, it must be powered with batteries and draw little current. Any noise reduction or cancellation signal processing algorithms must operate in real-time. A typical system with the above attributes would have the following characteristics:

a. It should be less than 1 cu ft in size and weigh less that 15 lb.

b. It should be battery operated with removeable rechargeable batteries that power the system for at least 6 hours.
c. It should digitize the outputs of the geophones and process the data in real-time. The result should be signal conditioned and channeled to earphones on which rescuers could listen.

Rescuers need to arrive at a disaster site and deploy the listening systems at locations where people are believed to be trapped. Quick determinations need to be made as to whether people were communicating distress signals by voice or tapping. Generally, there is not time for detailed digital signal processing, spectral plots, or other custom signal conditioning in order to evaluate results. Portability and real-time processing are imperative.

2 Methods of Addressing Noise Cancellation

Noise cancellation was addressed by several methods of digitally processing previously collected data. Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) filters may be designed to band-pass a limited bandwidth of data to reduce background noise. Noise Cancellation techniques are also applicable where the developed algorithm automatically designs a digital filter which cancels or reduces background noise. Beamforming has also been applied to specifically reduce noise coming from all but one direction.

To determine the best method of signal processing to use, the characteristics of both the signal and the noise need to be considered. The anticipated thumping signal is a set of narrow band signals (short impulses) between 0.5 and 5.0 seconds apart. These signals are generated by a person hitting a section of a collapsed building or ground and causing it to vibrate at a resonant frequency. The background noise will have a more varied spectrum. Some machinery may produce narrow band signals while operating at a constant RPM, but as the speeds and loads endured by these machines change, so will the received signal spectrum be modified. Other noise sources may produce signals with a much wider bandwidth. The combinations of these noise sources generally provide a very complicated noise spectrum.

Digital Filtering

Digital filtering provides a very simple method of processing the data. If the narrow bandwidth of the signal is known, this portion of the spectra can be bandpass filtered and a great portion of the background noise will be eliminated. This is the easiest of all methods considered to implement. However, for proper application the bandwidth and center frequency of the signal must be established. This information would not normally be known in advance. Additionally each site condition has its own unique characteristics which cause the bandwidth and center frequency to differ from locale to locale. However, one practical option is to divide the spectrum into multiple narrow bands and band pass each section. The output of each section could be monitored in order to detect the distress "taps." Certain sections of the audio spectrum have been clearly ruled out as containing survivor response taps. These may be always filtered out of the collected signals. For example signals under 20 Hz need not be processed since they cannot be heard by the human ear. It is unlikely that signals above 250 Hz would originate from a tapping signal source. Consequently a set of nine or more filters between 20 and 250 Hz with overlapping pass bands would provide excellent separation of the signal from background noises outside the passband. The user would however have the problem of monitoring all nine bands in order to determine if a distress call was present, or to determine which band contained the greatest signal to noise ratio.

A modified portable computer could be utilized in monitoring all nine channels and assisting the listening system operators. This would be accomplished by scanning each band as it is filtered and numerically searching it for the high peaks or impulses produced by a survivor's tapping. The determination of which signals constitute an impulse should be made by measuring the signal's peak value relative to its overall noise level. The algorithm in the computer could then select the frequency band that separates the data and noise most noticeably. By this process the survivor's signal will be robustly and optimally filtered for each varying site condition and environment.

Either IIR or FIR filters could be implemented for use in such systems. FIR filters offer the security of assured numerical stability, but IIR filters can be designed to operate more quickly. These trade-offs must be carefully considered and investigated when designing a final implementation for a trapped survivor detection and location system. Both speed and stability are of critical importance since real-time operation is required.

FIR filter with the transfer function

$$H(z) = \sum_{K=0}^{M} h(n)\ z^{-K} \tag{1}$$

can be implemented using the following difference equation:

$$y(n) = \sum_{k=0}^{M} h_k\ x(n-k) \tag{2}$$

Where y(n) is the filtered output, x(n) are the input samples, h(k) are the filter weights, and M is the number of filter points.

IIR filters with the transfer function

$$H(z) = \frac{\sum_{k=0}^{M} b_k z^{-k}}{1 - \sum_{k=1}^{N} a_k z^{-k}} \quad (3)$$

can be implemented using the following difference equation:

$$y(n) = \sum_{k=1}^{N} a_k\, y(n-k) + \sum_{k=0}^{M} b_k\, x(n-k) \quad (4)$$

Where "y(n)" are the output samples, "x(n)" are the input samples, and "a" and "b" are the filter coefficients.

Both IIR and FIR filters are discussed in detail in Oppenheim and Schafer [2].

Noise Cancellation

Noise cancellation can also be employed in the form of an adaptive filter. Conventional filtering accomplishes noise cancellation in an open-loop fashion with FIR or IIR filters. Adaptive filters operate in a closed-loop system where the output is monitored to determine the error of the system. This error is then used to modify or change the adaptive filter. Several different algorithms have been proposed to implement adaptive filters, however, most use a form of the Least Mean Square (LMS) algorithm presented by Widrow [1]. A closed-loop adaptive filter is shown in Figure 1. In this example an input signal, s, is supplied to a processor and an output, y, is obtained. This output is then compared with a desired output, d, and the performance or error, e, is measured. This error is then used by an adaptation algorithm to adjust the processor until a minimum amount of error is present. As the characteristics of the input and desired output change so will the learned transfer function.

If there is a linear relationship between s and d, the adaptive filter will learn it and the output e will show the signal with the linearly correlated signal removed.

Single-channel noise cancellation

A single-channel noise cancellation system can be implemented in the form shown in Figure 2. In this diagram a signal, s, is used as the desired signal input. A delayed version of this signal is then used as the main input to the adaptive processor. If this input signal contains periodic signals, both the initial signal and its delayed version will be well correlated. The adaptive processor will then learn a transfer

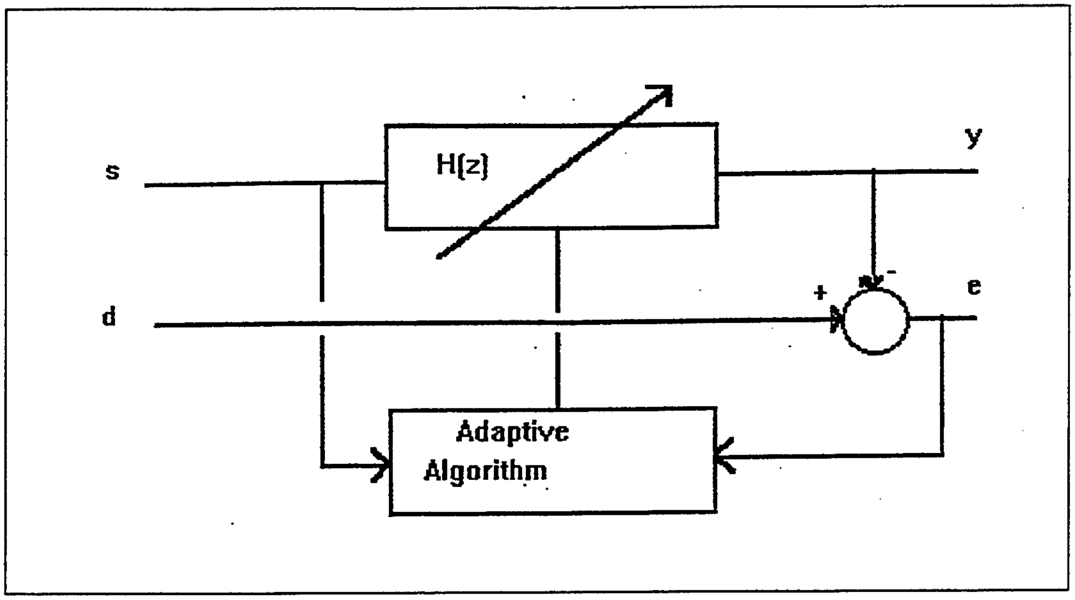

Figure 1. Closed-loop adaptive filter

function that predicts what future values of the signal will be. The adaptive processor's output and the delayed initial signal are subtracted to determine the error. If the initial signal is very periodic the error will eventually decrease to near zero. However, as noncorrelated data occurs in the signal it can be observed at the error output, e.

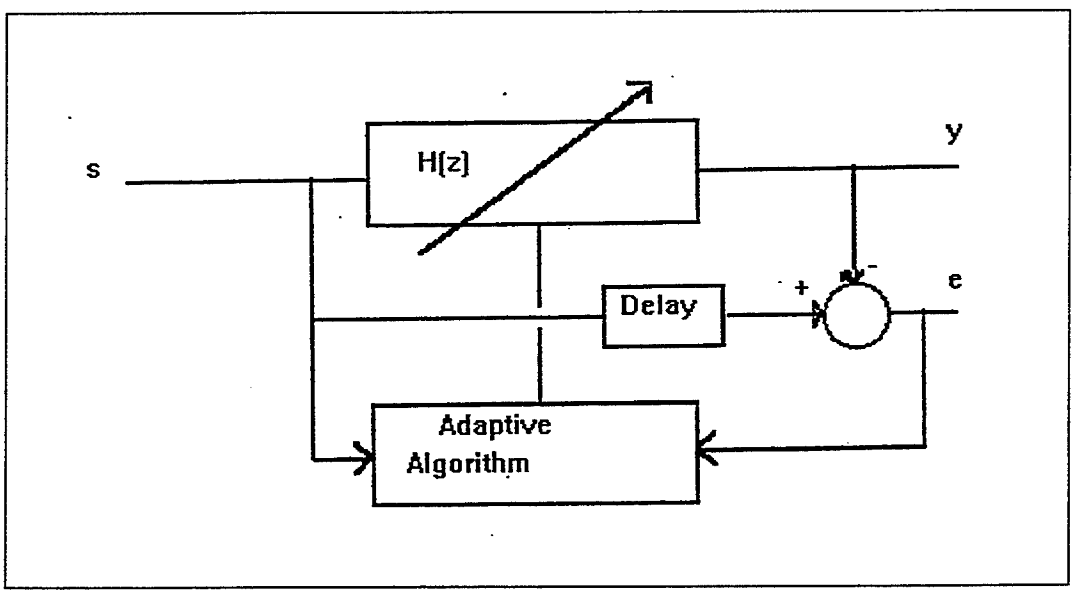

Figure 2. Single-channel noise cancellation

This form of noise cancellation can be used in applications in which the noise is periodic and the desired signal is not. One example of such a situation is when human voices are being detected in the presence of machine noise [8].

Two-channel noise cancellation

The basic form of a two-channel noise cancellation system is shown in Figure 3. In this diagram one input is a desired signal, s, combined with noise, n, to form s+n. The other input contains just noise that is correlated with the noise in the first channel. The signal and noise channel is used as our desired output and the correlated noise channel is used as our input to the adaptive process. The adaptive process will eventually learn the linear relationship between the noise in the first channel and the noise in the second channel. As this relationship is learned the output of the adaptive process will become equal in magnitude and phase with the noise in the first channel. Since they are subtracted to obtain the error, this difference will also be equal to the signal without the corrupting noise.

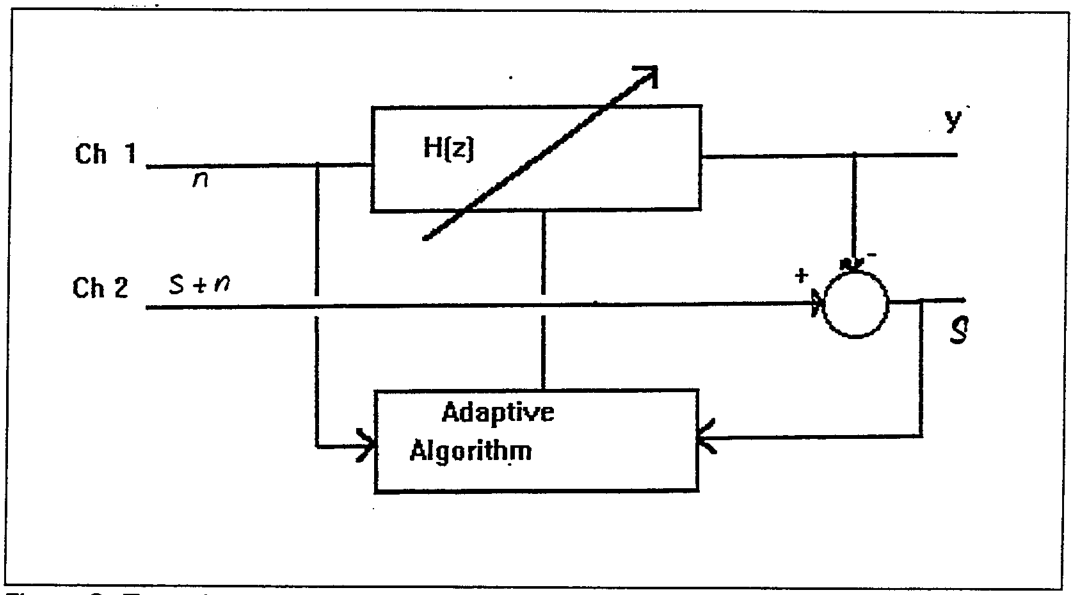

Figure 3. Two-channel noise cancellation

Two-channel noise cancellation systems have been used in applications such as reducing the noise in the cockpit of an airplane, or in detecting the ECG of a fetus when it is corrupted by noise from the mother's heart beat [9].

The LMS Algorithm

The Least Mean Squared (LMS) algorithm is the equation used by the one and two channel noise cancellation systems to modify the filter weights and thus continually redesign the filter's response. This enables it to adapt to a changing noise environment. The LMS algorithm is derived and discussed in detail in Adaptive Signal Processing, by Widrow and Stearns [1]. The adaptive filter or linear combiner can be configured as a transversal filter as shown in Figure 4. The output is then calculated

$$Y = X_k^T W_k \tag{5}$$

The error can then be calculated

$$e_k = d_k - X_k^T W_k \tag{6}$$

Then the filter's weights can be adjusted with the formula

$$W_{k+1} = W_k + 2 \mu e_k X_k \tag{7}$$

As can be seen from formula 7, the weights are continually being adjusted in proportion to the error e. It is important to note that if both the desired signal and the input signal are the same, no error will result and no change in filter weights will occur. Also, if there is not a linear relationship between the two signals, the weights will change at random and no noise reduction will occur.

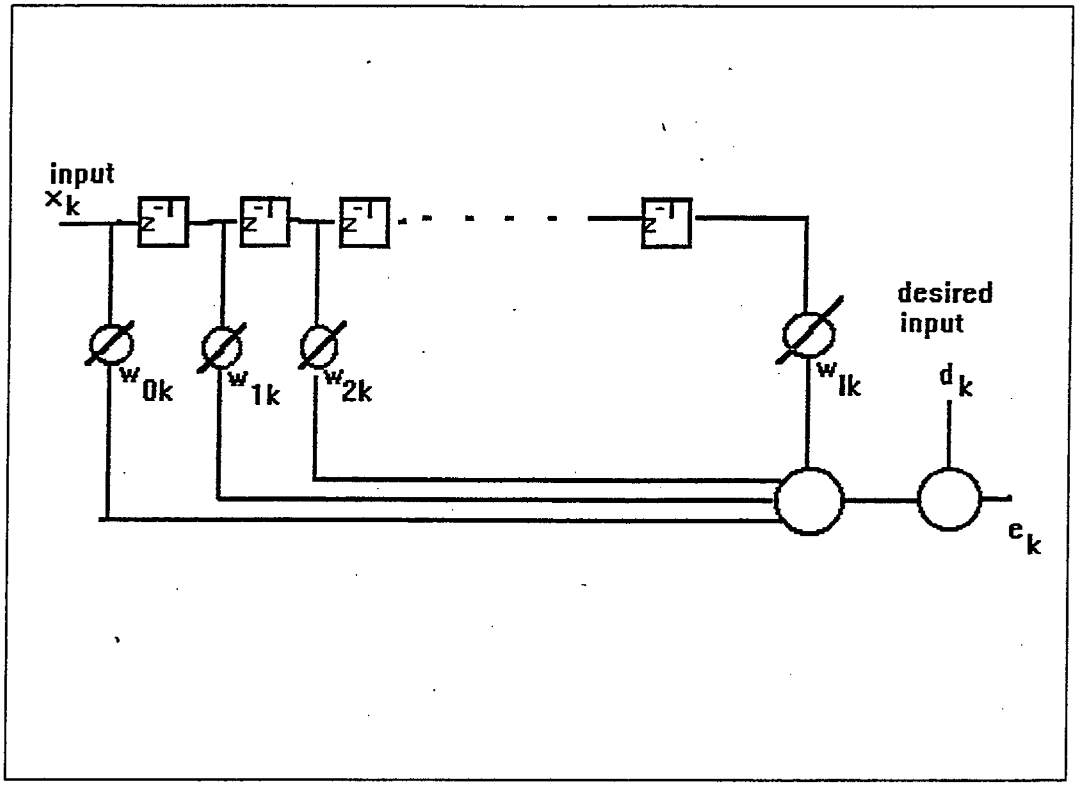

Figure 4. Linear combiner as transversal filter

3 Simulated Data

In order to test the proposed digital processing algorithms, simulated data was generated. It was used with the proposed algorithms to evaluate how well it worked under controlled conditions.

Description of Data

The simulated data consisted of eight tests. The simulations used four noise sources and one signal source. Two noise sources were correlated and two were uncorrelated noise. A diagram of the model for the simulation is shown in Figure 5.

This model contains a wideband noise source, N1, and a narrow band noise source, N2. Each of these noise sources travels through a different transfer function to channels 1 and 2. The signal also travels through different transfer functions to channels 1 and 2. However the gain of the transfer function between the signal source and channel 2 is much smaller than the gain between the signal source and channel 1. In addition to these signals random noise is added to both channels 1 and 2. The transfer functions, H1(z) through H6(z) are shown in Appendix A. For each of the eight tests the amplitudes of the noncorrelated random noise, the amplitudes of the correlated noise, and the frequency of the distress signal were varied. The actual parameters used are shown in Table 1.

All signals were normalized to a 1 Volt RMS amplitude over a 0.2 Sec duration. All wideband signals were from 50 to 150 Hz. All narrow-band signals were from 58.9 to 61.1 Hz.

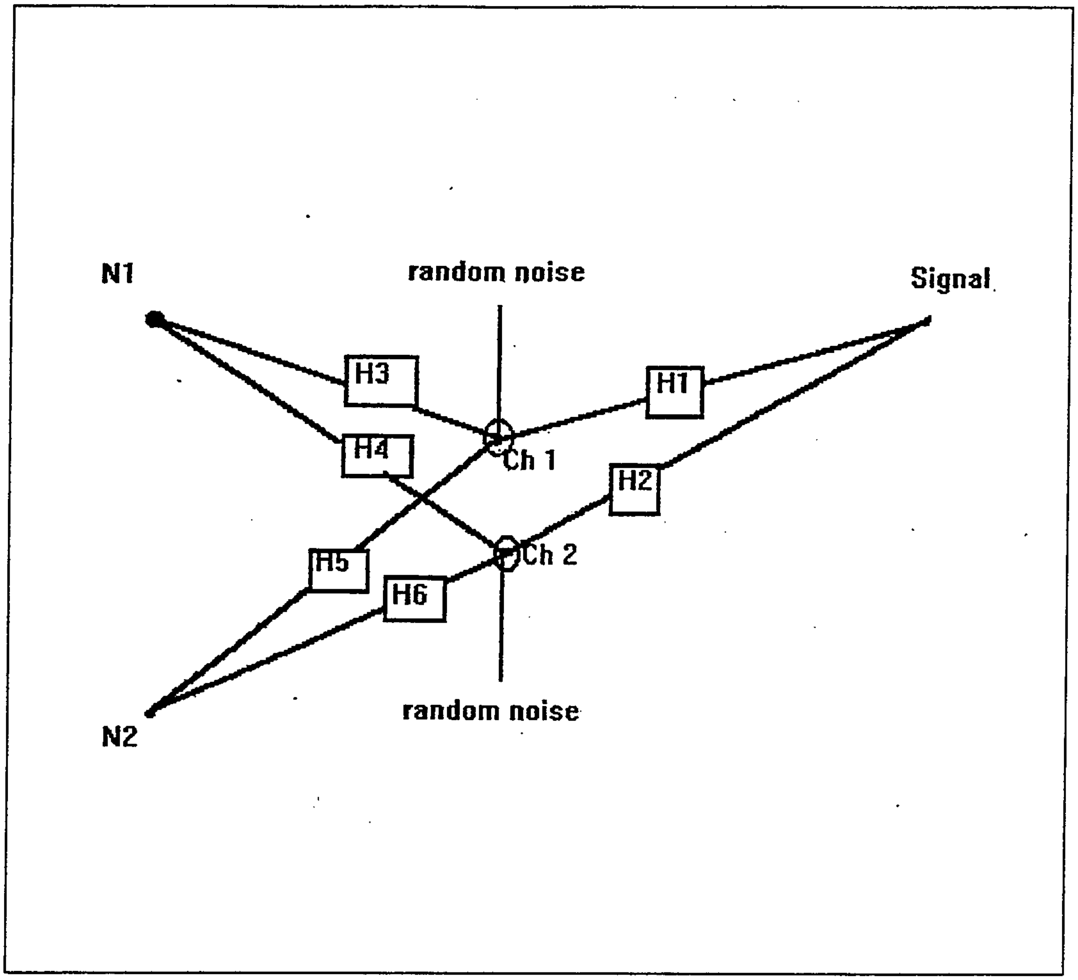

Figure 5. Simulation model

Table 1 Simulated Data Parameters of a 1 Volt RMS Signal				
Test	Signal Freq Hz	Wide-band Amplitude Volts	Narrow-band Amplitude 58.9-61.1 Hz Volts	Noise Amplitude Volts
Test S1	111	.1	.2	.01
Test S2	200	.1	.2	.01
Test S3	111	.1	.2	.50
Test S4	200	.1	.2	.50
Test S5	111	.5	.5	.01
Test S6	200	.5	.5	.01
Test S7	111	.5	.5	.50
Test S8	200	.5	.5	.50

A plot of each channel for each of these tests and its spectrum is shown in Appendix A. The most corrupted test, Test S8, and the least corrupted test, Test S1, are shown in Figure 6.

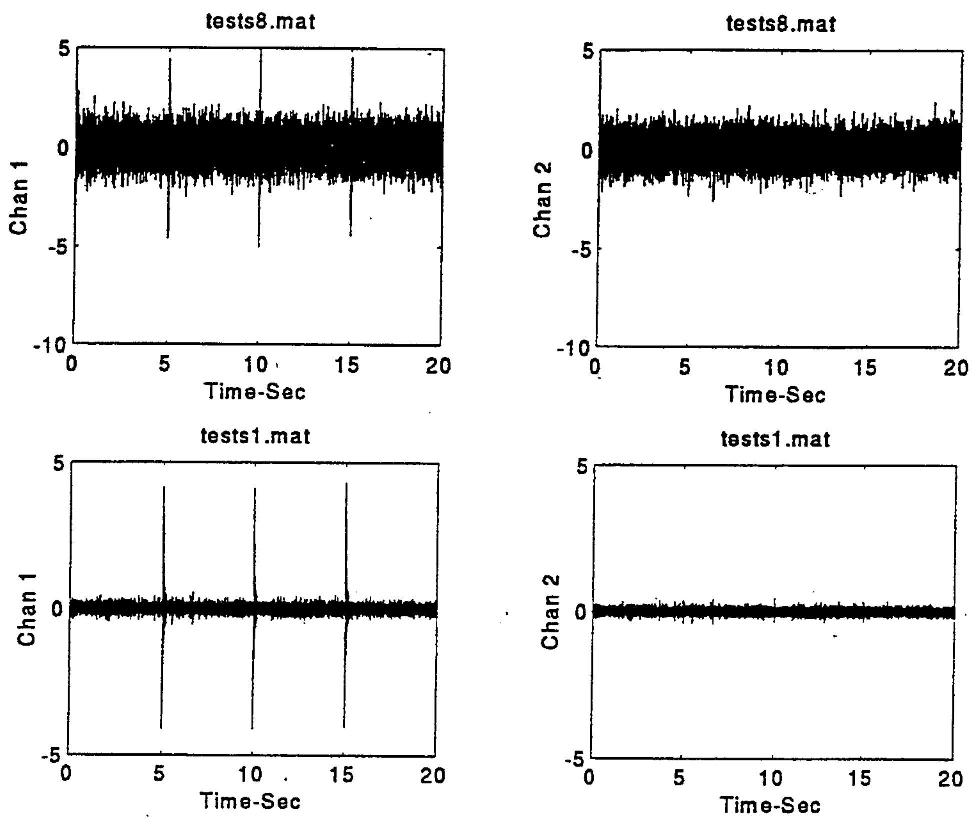

Figure 6. Most corrputed (S8) and least corrupted (S1) simulated test

Filtering Results

To test the ability of using classical FIR filtering to detect impulse signals a set of nine FIR band-pass filters was implemented using 61 points. The results of these nine bands for test S6 and test S7 are shown in Figures 7 and 8. As can be seen from these plots, the filter well separates the impulse signal from the noise so the bands containing the impulse can be easily identified. The nine bandpass FIR filters were designed with the following bandwidths: 20-66 Hz, 43-89 Hz, 66-112 Hz, 89-135 Hz, 112-158 Hz, 135-181 Hz, 158-204 Hz 181-227 Hz, and 204-250 Hz.

One problem with this method is that the user would need either advance knowledge of the bandwidth of the signal to be detected or to listen to all nine bands in order to determine which band has the best signal to noise ratio. A better method of accomplishing this is to allow a program or algorithm in a modified PC computer to determine which band has a signal that most resembles the distress calls. This can be achieved by calculating the ratio of the maximum value in each band to the standard deviation of the continuous noise level of each band. This will give a <u>near estimate</u> of the signal to noise ratio (SNR) in each of the bands. The filtered spectral sections with the highest SNR can then be chosen for continuous monitoring at that specific site.

In order to demonstrate how well this method works, each of the eight simulated tests were passed through a set of bandpass filters and the one with the best SNR was chosen for comparison with the original signal. The results of these tests are shown in Figures 9 through 12. It should be noted when viewing these plots that, it is not the signal's amplitude that is important but the signal's amplitude relative to the background noise level.

The results show that considerable noise reduction can be realized by this type of filtering. In the even numbered simulated test, the signal and wideband noise were at distinct different frequencies, but in the odd numbered tests the signal frequency was near the center frequency of the wideband noise. It can be seen from Figure 7 that when the signal and noise are separated in frequency excellent noise reduction is achieved. Filter 8, for example, shows a much greater SNR than the raw data in test S6. It should also be noted that when the signal and noise are not separated in frequency, less impressive noise reduction is realized. This can be observed in Figure 8 (Test S7) where some noise reduction occurs when comparing Filter 4 with the raw data. However, the noise reduction is much less than that achieved in Test S6.

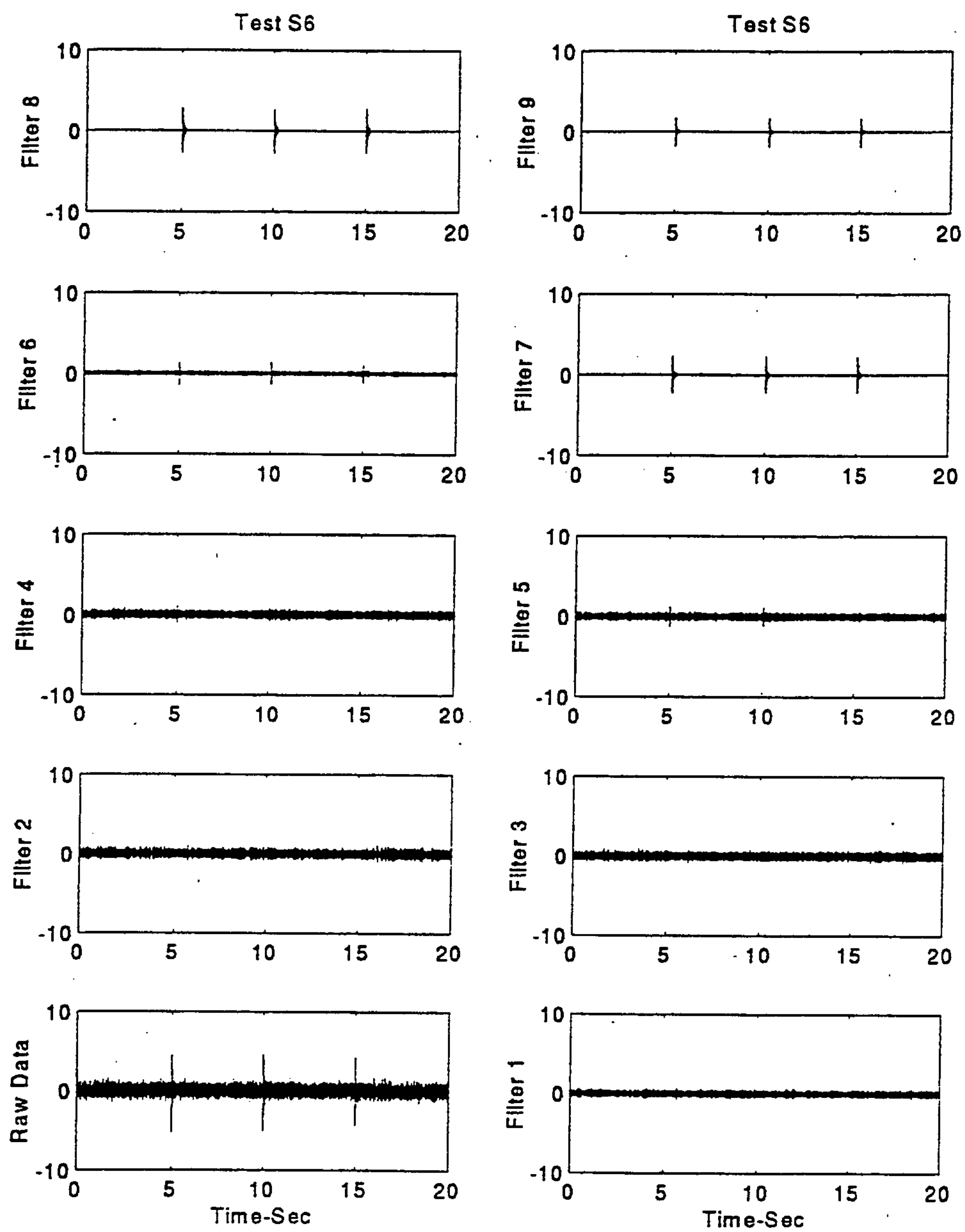

Figure 7. Bandpass Filtering Test S6

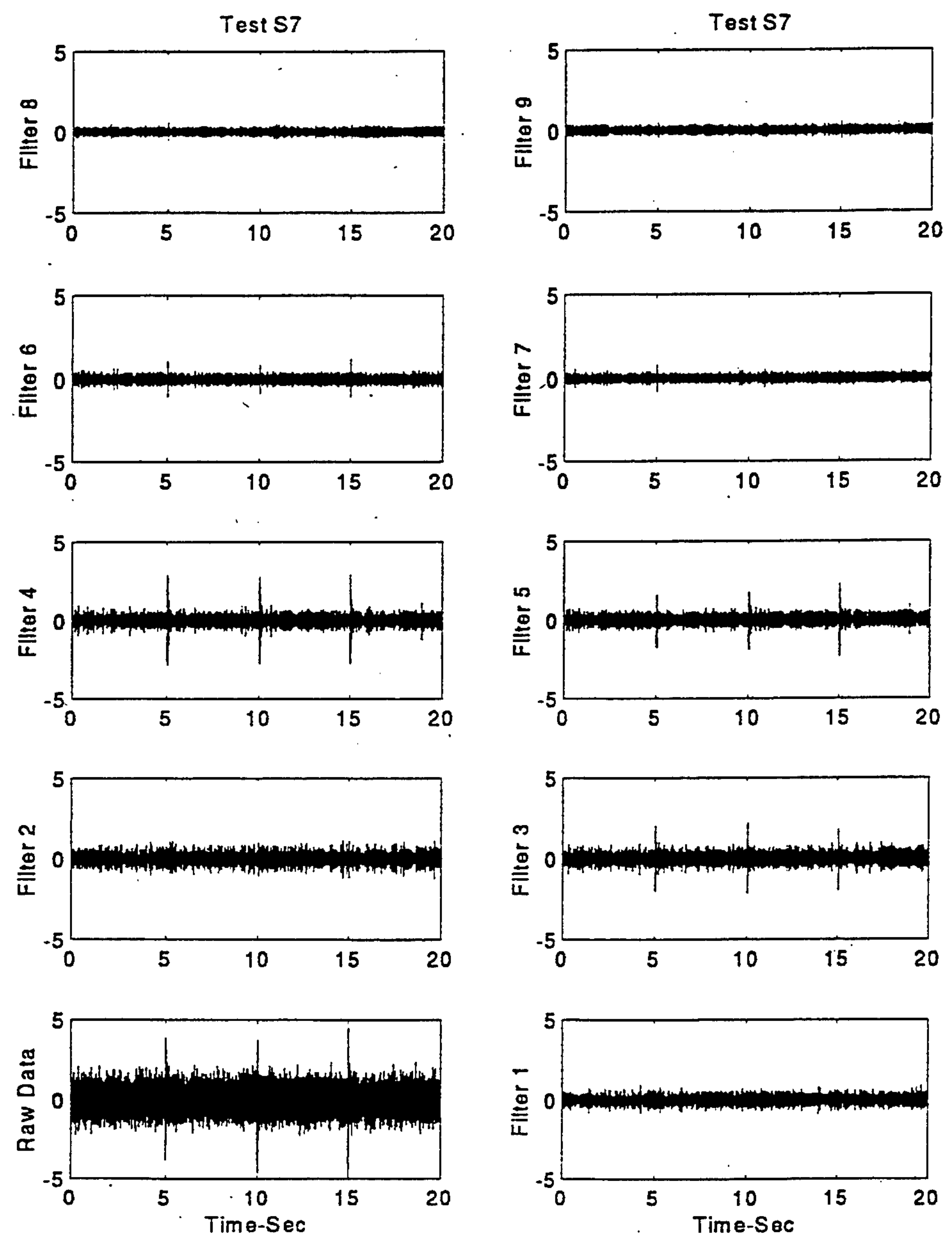

Figure 8. Bandpass Filtering Test S7

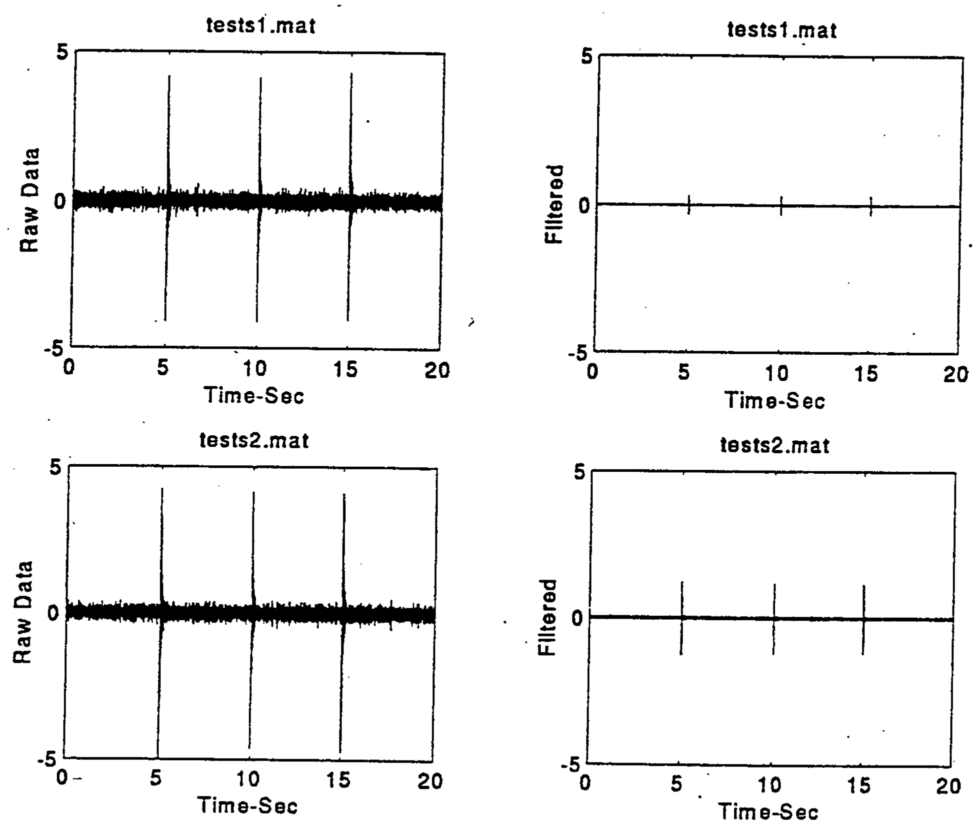

Figure 9. Filtered Output Test S1 and S2

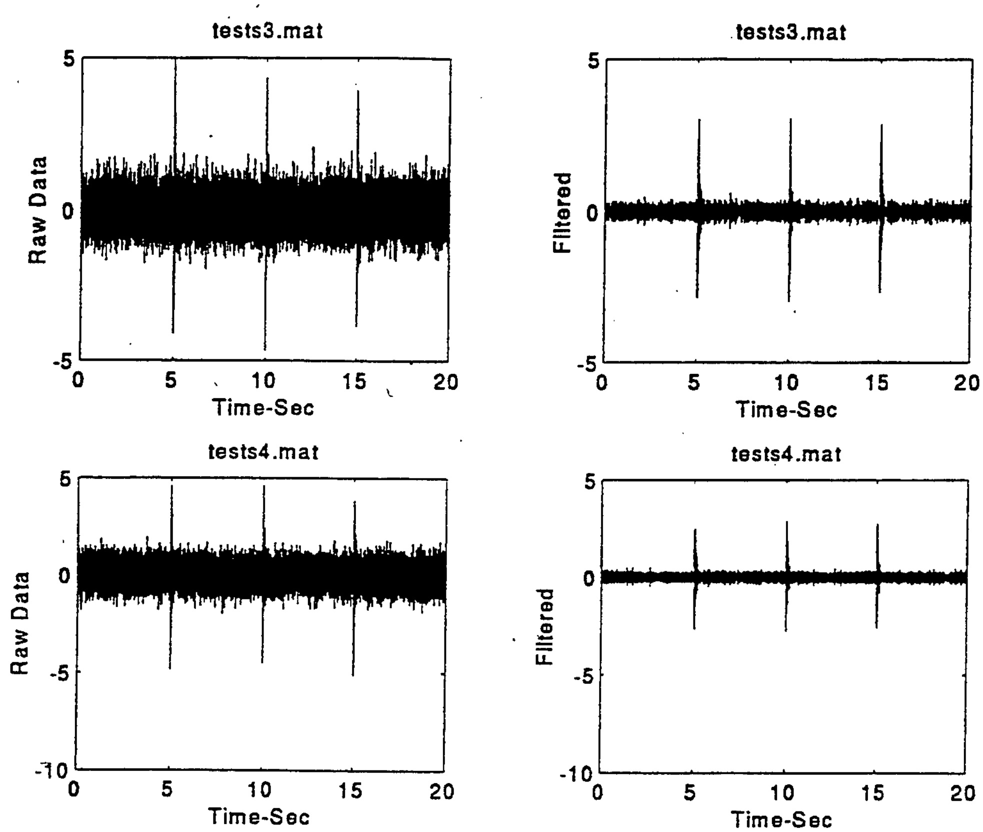

Figure 10. Filtered Output Test S3 & S4

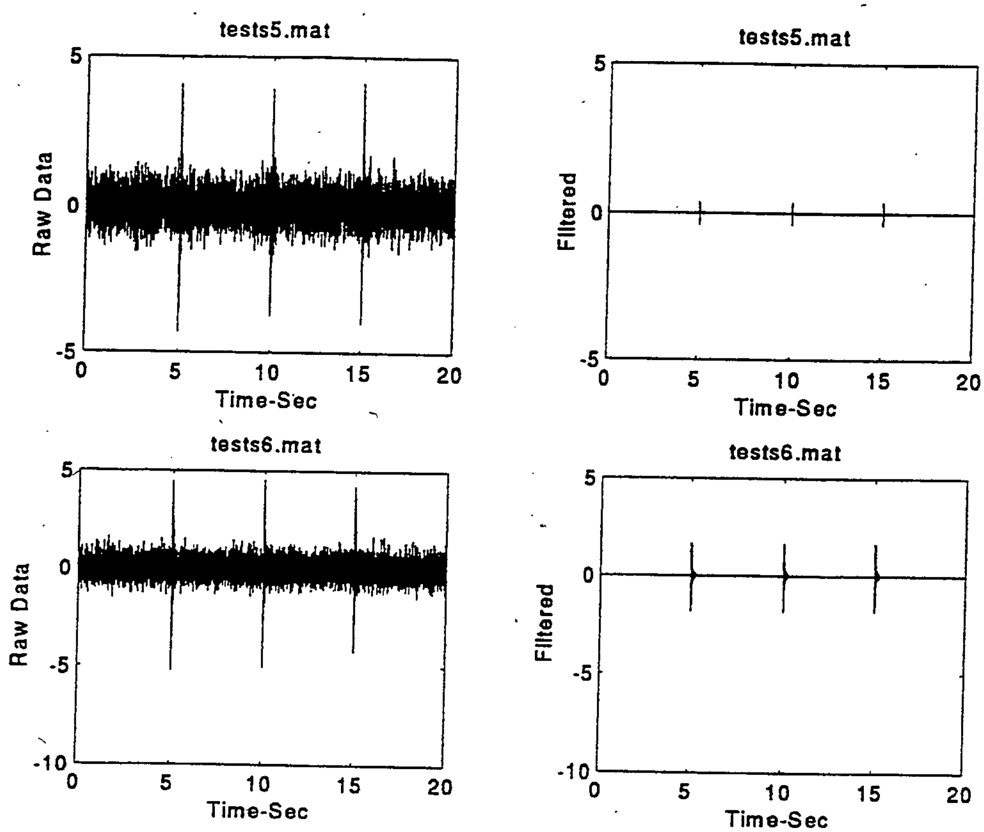

Figure 11. Filtered Output Test S5 & S6

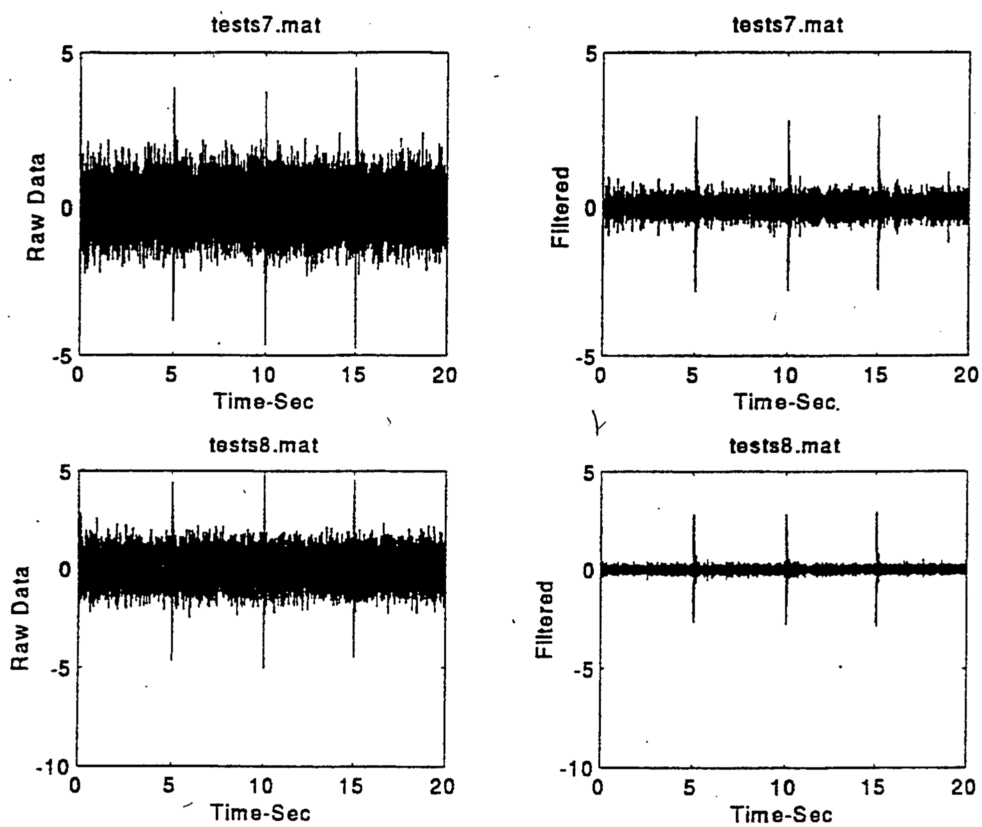

Figure 12. Filtered Output Test S7 & S8

This demonstrates that for the bandpass filter to work well, the signal and noise must be separated in frequency.

Noise Cancellation Results

Both the single-channel (predictor) and the two-channel (noise cancellation) routines were used to process the simulated data. For the single-channel noise cancellation routine to function, the input must be very periodic. Channel 1 was delayed by 50 msec and then used with itself to filter out periodic background noise. The results for the eight simulated tests are shown in Figures 13 through 16.

The two-channel noise cancellation routine was also implemented and used to process the eight simulated tests. For this method to work the noise in both channels must be correlated and the signal must not be correlated. The results are shown in Figures 17 thru 20.

The results show that both single-channel and two-channel noise cancellation techniques provide good noise reduction with the simulated data. For the two-channel noise cancellation to work effectively, the noise on the two channels must be correlated, or for one-channel noise cancellation to be effective, the noise must be periodic (correlated with itself). In simulated Tests S1, S2, S5, and S6, the noise signals were well correlated, but in Tests S3, S4, S7, and S8 large amounts of noncorrelated random noise was added. The results of the analysis show that little or no SNR improvement is realized in the tests with noncorrelated noise. However substantial improvement is realized in the others.

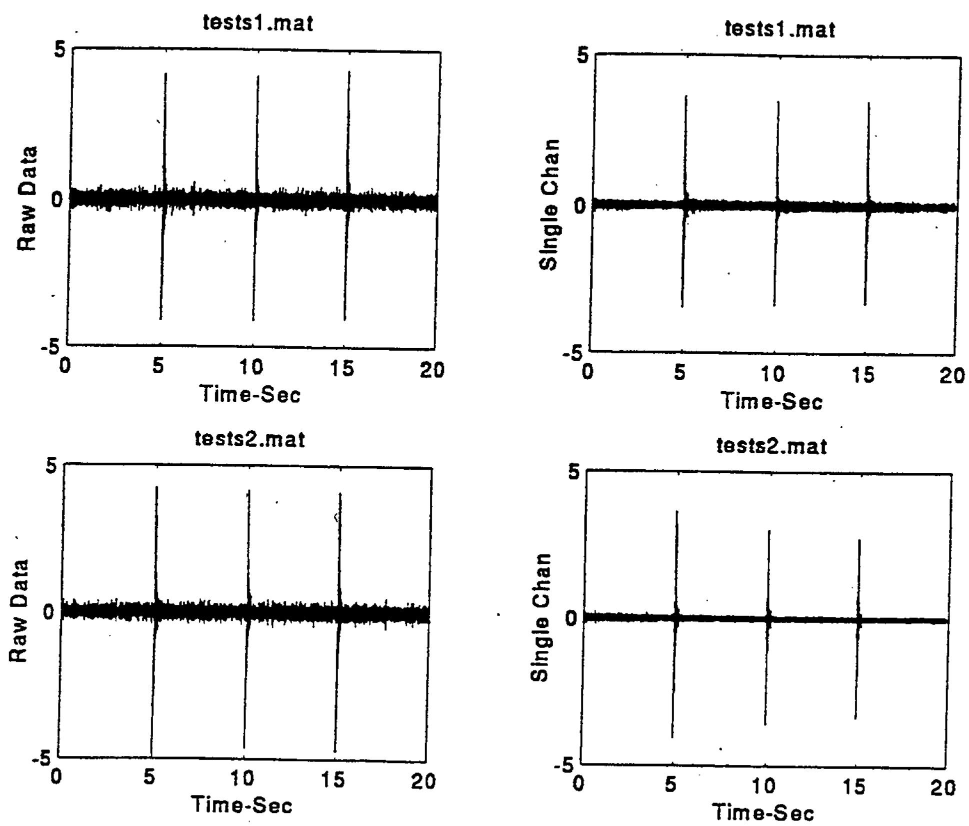

Figure 13. Single-channel Noise Cancellation Test S1 & S2

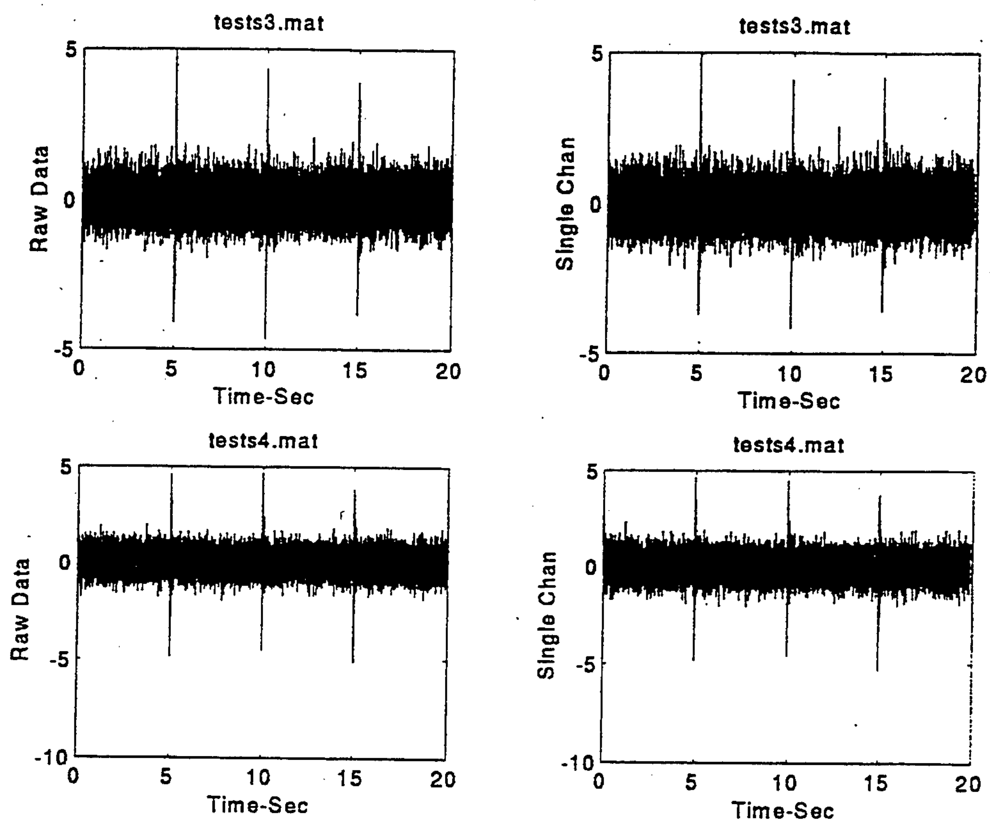

Figure 14. Single-channel Noise Cancellation Test S3 & S4

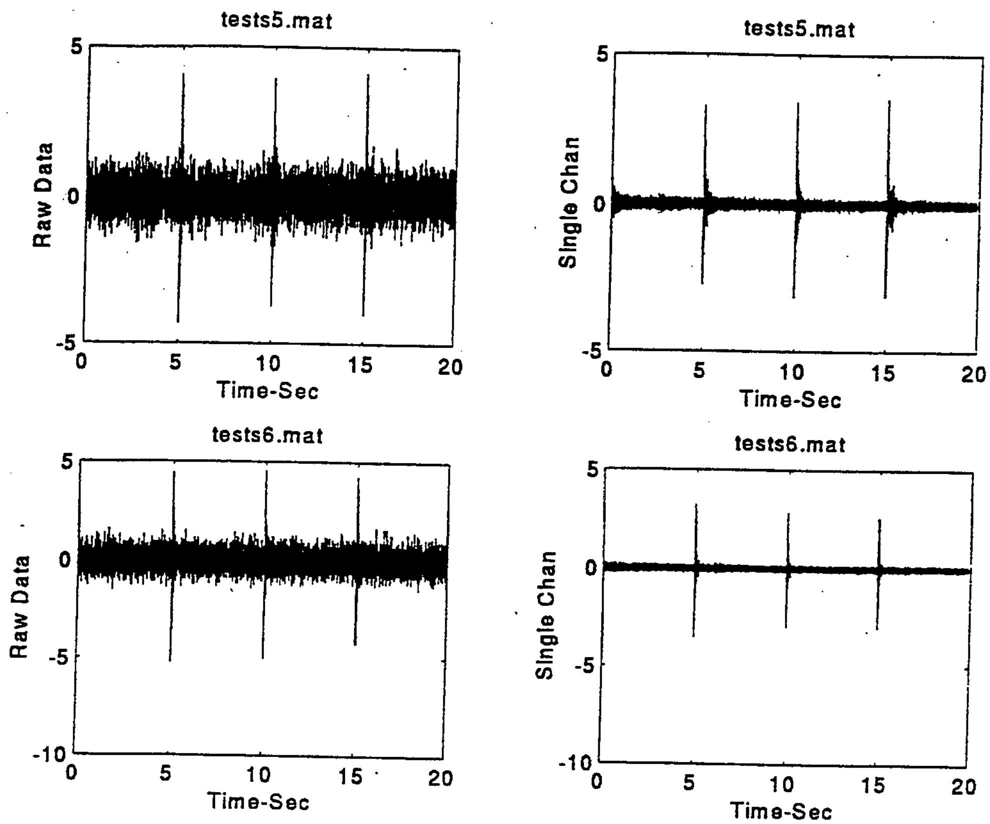

Figure 15. Single-channel Noise Cancellation Test S5 & S6

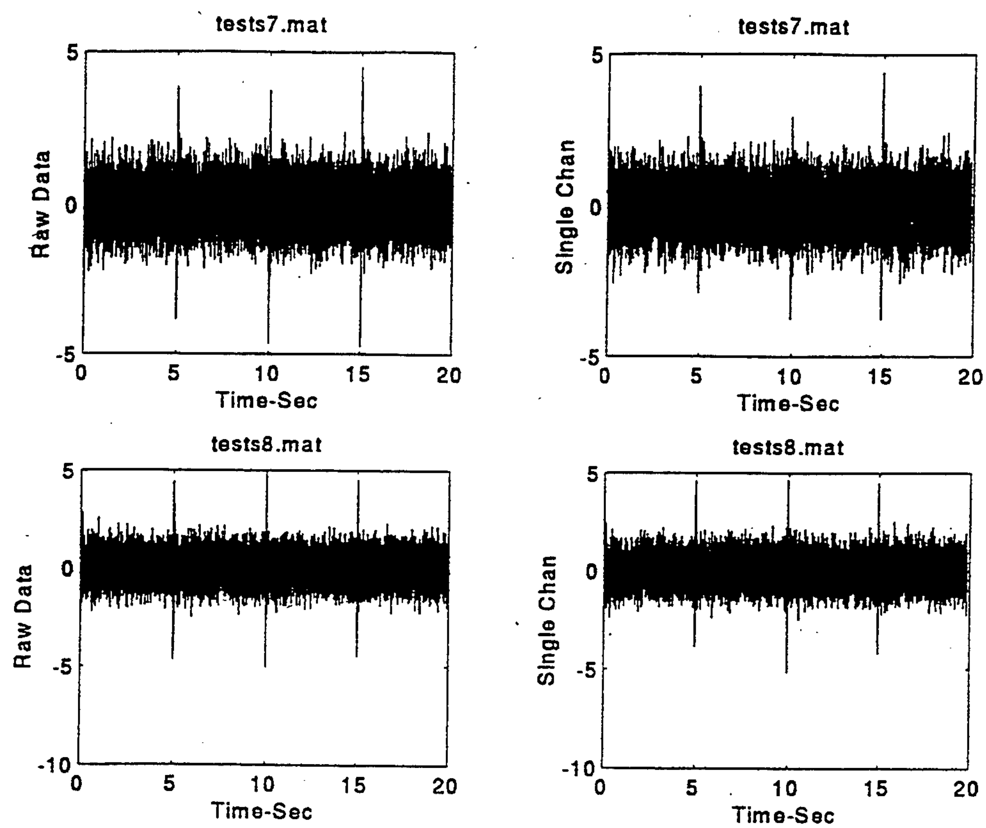

Figure 16. Single-channel Noise Cancellation Test S7 & S8

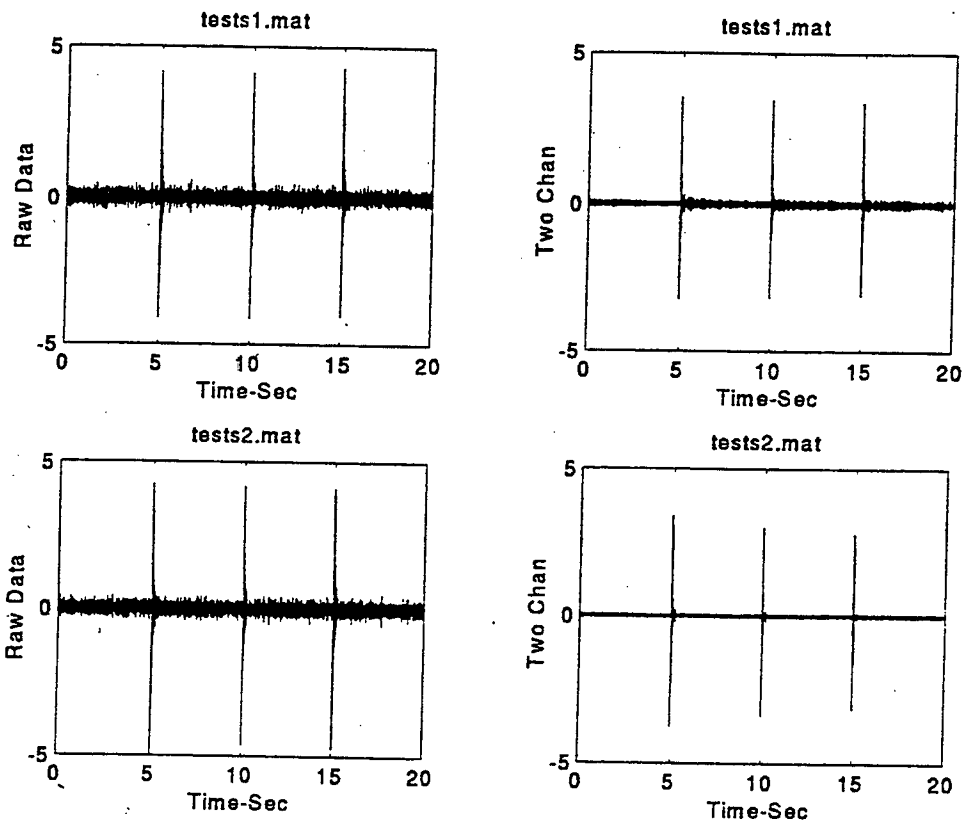

Figure 17. Two-channel Noise Cancellation Test S1 & S2

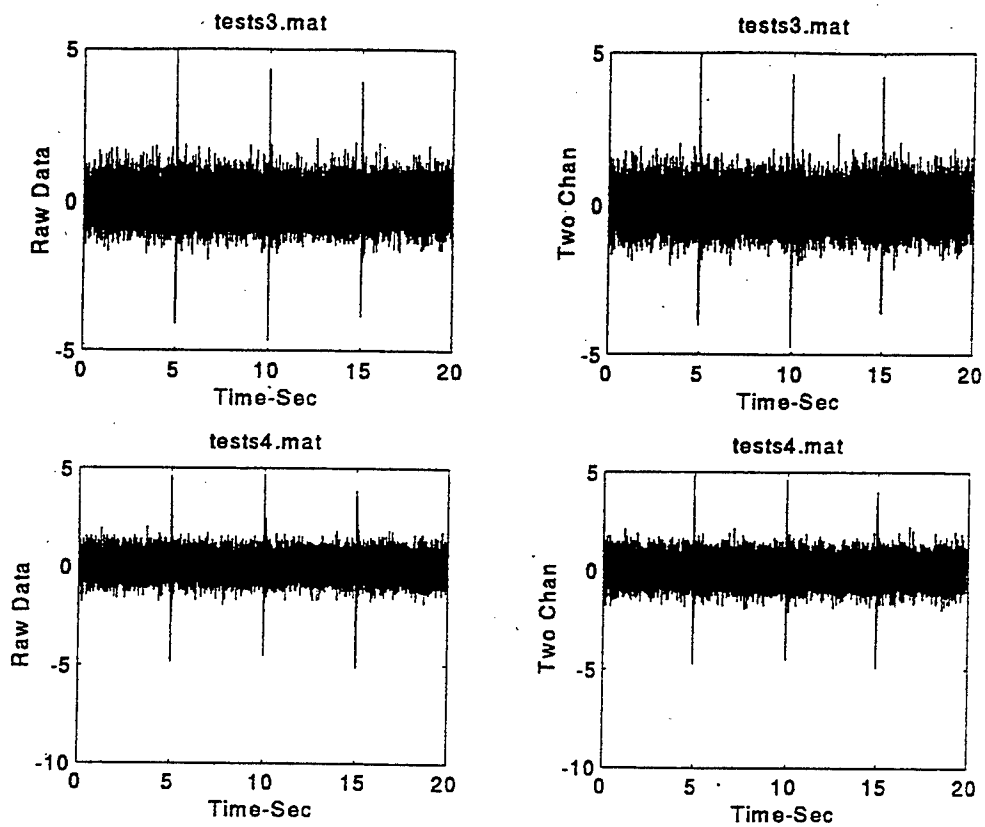

Figure 18. Two-channel Noise Cancellation Test S3 & S4

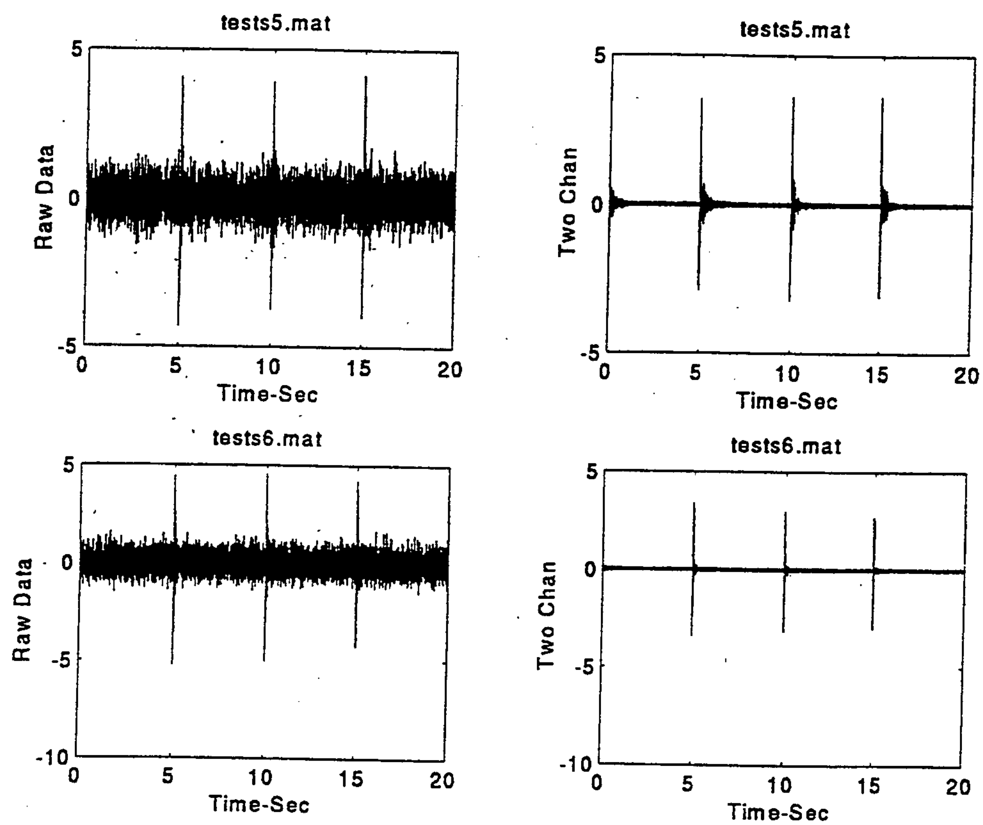

Figure 19. Two-channel Noise Cancellation Test S5 & S6

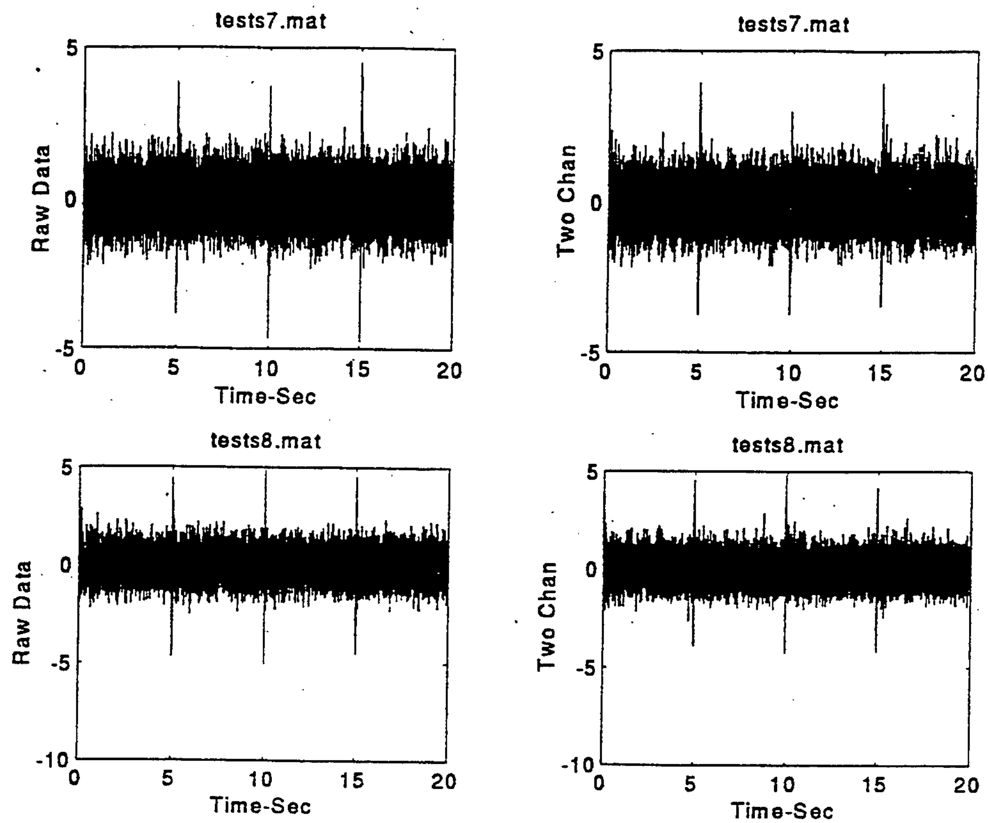

Figure 20. Two-channel Noise Cancellation Test S3 & S4

4 Field Test

Data from a field test was used to evaluate the effectiveness of these noise cancellation algorithms on actual data. Field test data were selected and digitized at 1000 samples/sec. These data were processed similarly to the previously discussed simulated tests.

Description of Field Test

Field tests were conducted at the Mississippi University for Women in Columbus, Mississippi. A device to simulate a thumping distress call was constructed and placed in a multistory office building which was then collapsed. The building was then demolished in a manner to simulate an earthquake induced failure. Signals from the "thumpers" were then recorded on digital tape using geophones at various selected locations near and on top of the building debris. The tests using the most responsive geophones were digitized and processed. These tests used two transducers placed .5, 1., 2., 4., and 6. feet apart. The placement of sensors is a very critical consideration in noise cancellation problems. If the two sensors are too close, both signal and noise will be correlated. If they are too far apart the transfer function, H(z), between the two systems may become too complex and nonlinear and therefore too difficult to learn.

In order to study the character of these tests the sections of the time history containing the signal (thumping) were identified and separated from the raw data. The noise and signal could then be studied separately or together. Plots of the Power Spectrum of each channel and the Coherence between each channel were calculated for the signal, the noise, and the raw data of signal plus noise. These plots are shown in Appendix B.

Field tests were made with three main configurations of sensors. The three diagrams are shown in Figure 21.

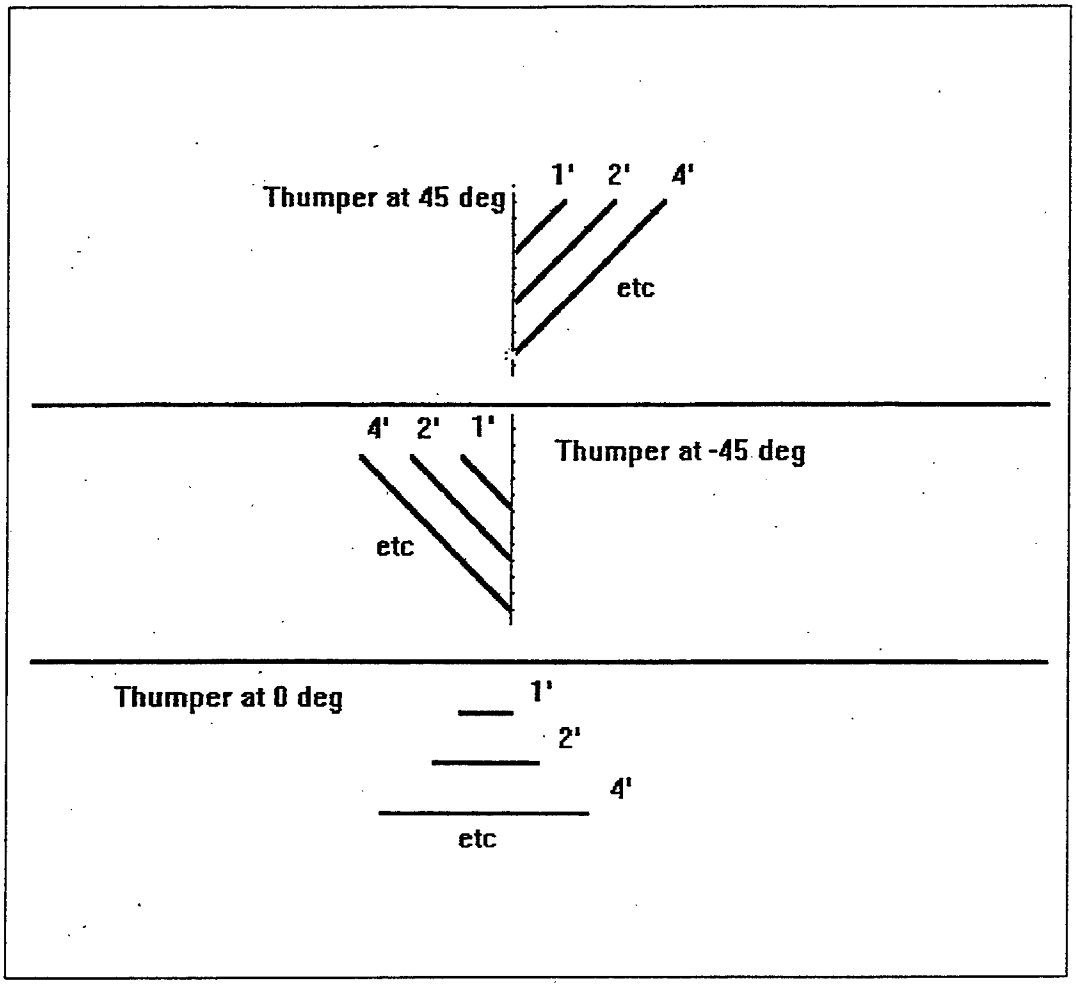

Figure 21. Site diagram

Digital Filtering Results

The data from the field test was also used with the band-pass filtering scheme discussed in Chapter 3. These results also show very good noise reduction. Examples are shown in Figure 22.

It can be seen from the spectral plots in Appendix B that the noise is separated from the signal in frequency. The signal is approximately 90 Hz and the noise is a narrow band around 150 Hz. This would indicate that the selective digital filtering method should provide excellent results.

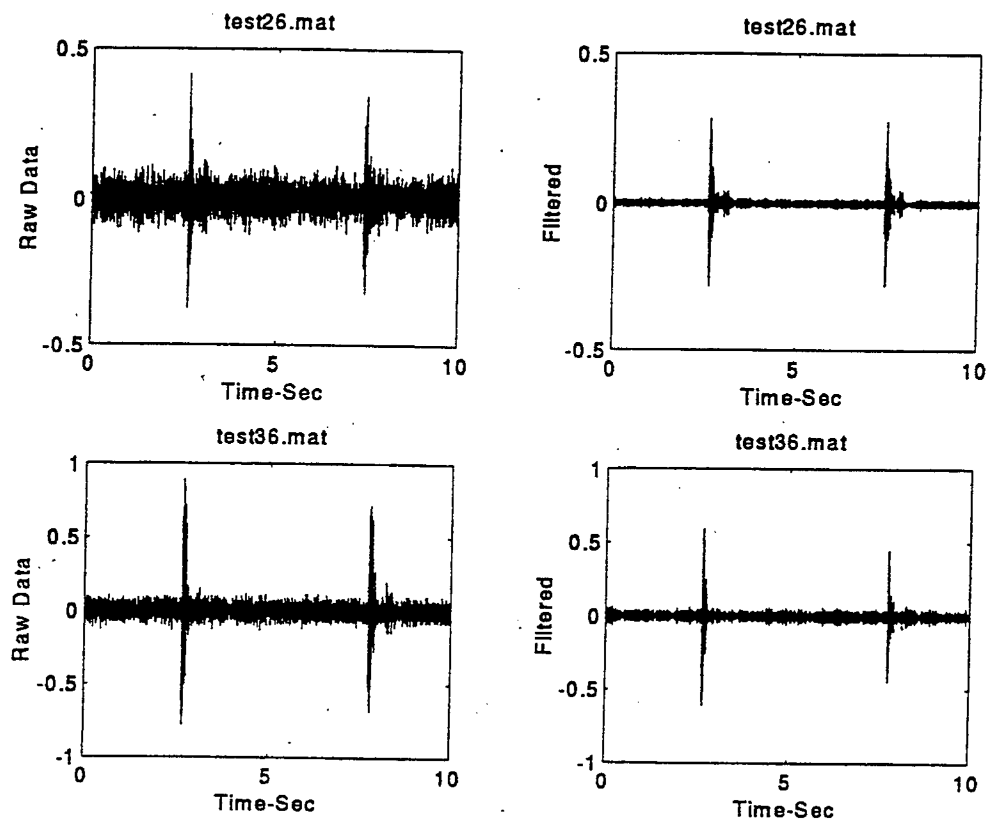

Figure 22. Filtered Output Test 26 & 36

Noise Cancellation Results

Both the single-channel predictor and the two-channel noise cancellation algorithms were used to test their results on the field data. To get good results from the two-channel noise cancellation algorithm, it is required that the noise signals be correlated and that the signal not be strongly present on both channels. This was not the case with this field data. The signal was highly correlated on each of the channels. It can be seen from the spectral plots in Appendix B that both signal and noise are strongly correlated. To get good results from the single-channel noise cancellation algorithms, it is required that the noise signal be highly periodic. Again this was not the case with this field data. As can be seen in Figure 23 and 24, the noise cancellation techniques produce poor results with this data.

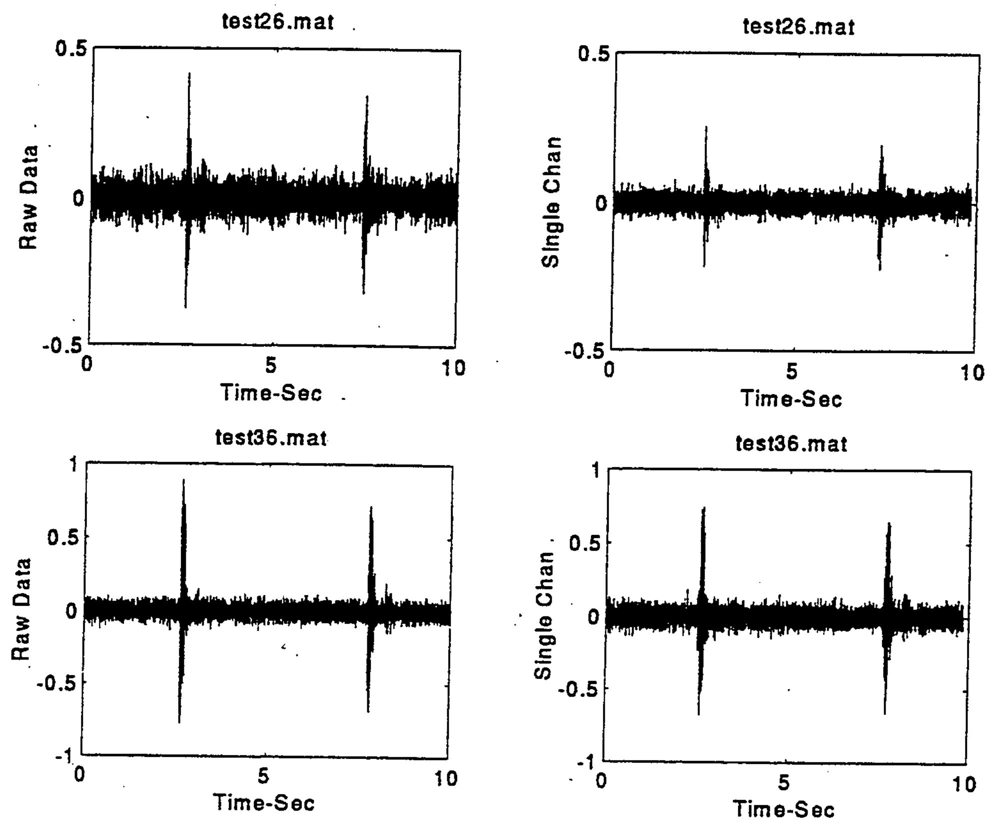

Figure 23. Single-channel Noise Cancellation Test 26 & 36

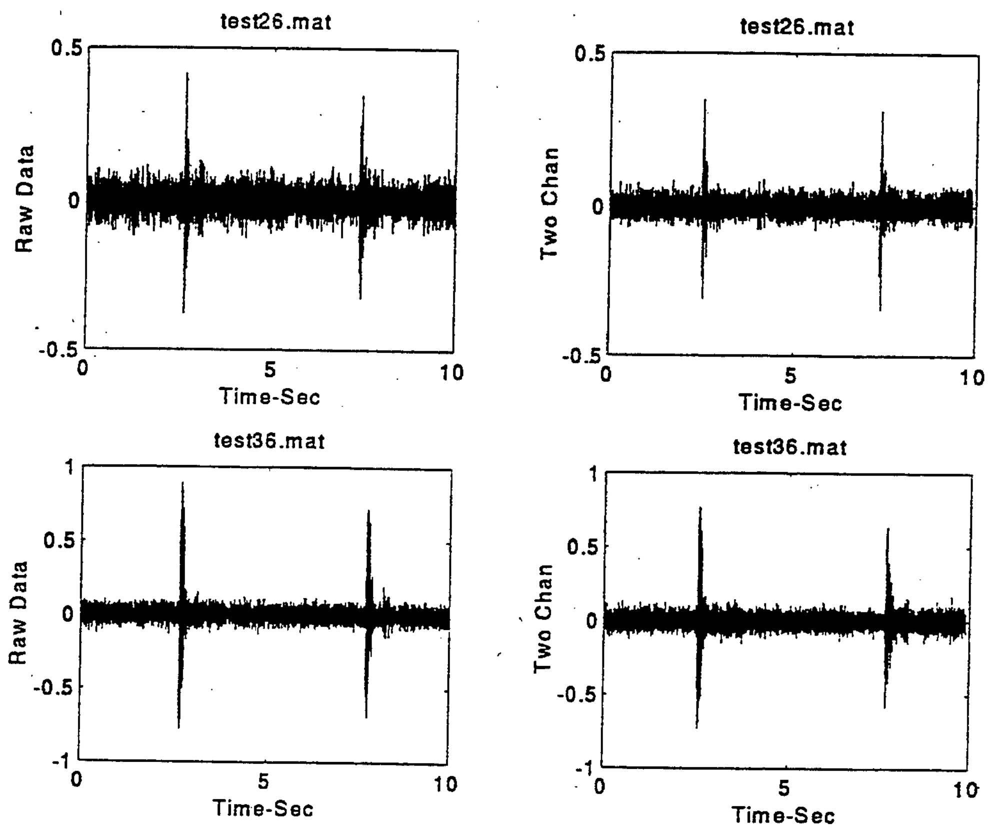

Figure 24. Two-channel Noise Cancellation Test 26 & 36

5 Comparison of Results

When the signal is correlated on both channels the output of the filter can be saved instead of the error. This provides a time history of the correlated signals. If the noise is uncorrelated, the results will give improved SNR. However, if both signal and noise are correlated or if neither is correlated no SNR improvement can be realized.

The three methods of processing the data were compared to try and determine which method produces the best results. In order to make a comparison in a form that is representative of the volume levels a person might hear, the data was converted to a decibel scale. This was done by processing the data through an RMS filter and converting this amplitude into decibels. The following formulas were used.

$$OUTPUT_{RMS} = \sqrt{\frac{1}{N}\sum_{i=1}^{n} X_i^2} \tag{8}$$

and

$$Decibels = 20\ LOG10(OUTPUT_{RMS}) \tag{9}$$

This measure of decibels should not be confused with the exact audio decibels that might be calculated for a microphone. But the change in decibels should be comparable to that of a microphone. Decibel plots for all three methods were made and compared with that of the raw data. These plots for all simulated tests and field tests are shown in Appendix C.

Data was extracted from these plots and tabulated in Tables 2 and 3. These tables show a comparison of the three methods used to process the data. They indicate that the Selective Filtering method works well with all of the data. The single-channel and two-channel noise cancellation routines work well when the levels of uncorrelated noise are low. In field tests where the signal and noise were both well correlated the noise cancellation routines gave no improvement in performance.

These results indicate that the "best-method" of filtering differs from site to site. One algorithm works best when the signal and noise are separated in frequency, while another works well when the signal and noise are not both correlated. How well the signal and noise are separated in frequency and how well they are correlated are functions of the site, the transducer placement, the structure, and the type of urban noise. The responders have little control over these conditions. The "best-method is therefore one which can select between multiple different methods of noise reduction.

Table 2 SNR Levels Before and After Processing				
Raw Data Ch 1 db	One Chan Noise Can db	Two Chan Noise Can db	Filter db	Test
13.2894	13.6256	15.6528	28.4677	26
20.5432	17.4282	20.3388	31.4466	28
16.7020	16.0571	20.2530	28.6329	29
18.8568	14.8417	16.4318	29.4628	30
16.5906	15.1096	16.0942	27.6939	31
18.0985	16.3148	17.0262	27.6135	32
18.9397	18.1548	19.9984	29.4657	33
17.7485	18.7080	20.3775	30.6381	34
27.9245	24.8290	26.5698	28.0173	35
20.5917	19.0165	19.9874	24.4057	36
20.9519	15.4994	20.9057	31.1359	37
20.8629	17.6962	20.6946	30.4611	38
17.8353	15.0043	17.4284	21.0796	39
19.6213	18.0286	22.1001	26.5954	40
18.8190	17.4334	12.6307	6.5276	41
23.5342	27.8079	28.7579	33.7639	S1
23.0693	30.9092	34.6914	29.3173	S2
11.7452	10.8272	11.3803	19.9742	S3
11.8002	10.7441	10.4280	19.8004	S4
13.2299	25.8739	34.8918	32.9168	S5
13.5019	25.7274	31.3524	47.6927	S6
9.3654	7.4533	6.6000	14.5801	S7
9.7874	8.9595	9.1001	19.9054	S8

Table 3
Signal to Noise Improvement

One Channel Noise Can. db	Two Channel Noise Can. db	Filter Noise Can. db	Test No
0.3362	2.3633	15.1782	27
-3.1150	-0.2044	10.9034	28
-0.6448	3.5510	11.9309	29
-4.0151	-2.4250	10.6060	30
-1.4808	-0.4962	11.1033	31
-1.7849	-1.0723	9.5149	32
-0.7849	1.0586	10.5277	33
0.9595	2.6290	12.8895	34
-3.0954	-1.3547	0.0927	35
-1.5752	-0.6043	3.8140	36
-5.4525	-0.0461	10.1840	37
-3.1667	-0.1683	9.5981	38
-2.8310	-0.4069	3.2443	39
-1.5927	2.4788	6.9740	40
-1.3855	-6.1882	-12.2913	41
4.2737	5.2237	10.2297	S1
7.8399	11.6221	6.2480	S2
-0.9180	-0.3648	8.2289	S3
-1.0560	-1.3721	8.0002	S4
12.6440	21.6618	19.6868	S5
12.2254	17.8504	34.1907	S6
-1.9120	-2.7653	5.2146	S7
-0.8279	-0.6861	10.1179	S8

6 Additional Signal Conditioning

In addition to filtering the data other methods of improving the quality of the data should be employed. It is recommended that Automatic Gain Control (AGC) be employed in the data conditioning process if the multi-bandpass filtering scheme is employed. The AGC should be used to make sure that the signal is always at a proper level for listening. Additionally, a method of replaying data should be employed so if a signal is heard the responder has the option of replaying a section to hear what is thought to be distress signals.

Automatic Gain Control

AGC is accomplished by periodically measuring the average level of a signal and continually adjusting the gain so it is always at approximately the same sound level. Considerable discretion must be used in determining exactly which AGC algorithm to use. If all signals, no matter how small, were amplified to the same level, very distant signals might be received and confused with signals from the location being investigated.

Delayed Playback Schemes

Delayed playback schemes can be easily employed with today's portable computers. Inexpensive Random Access Memory (RAM) and hard disk storage can be used to store substantial amounts of data for later listening.

Visual Aids

In addition to hearing the signal it is always helpful to have visual displays. Active matrix color displays offer the user the ability to view plots of time histories of data which often reveal qualities of the signal that are difficult to hear.

Beamforming

The signal-to-noise ratio of a data acquisition system can be improved by employing Beamforming techniques. Beamforming is a method of using combinations of sensors to provide improved results. A detailed discussion of Beamforming is found in Haykin [7]. When the direction of an approaching wave and its speed of propagation are known, delays between samples of channels can be inserted in order to align each channel as if it were parallel to the approaching wave. When this is accomplished all signals reach the array of sensors at the same time. An average of the sensors is then taken. Since the steered wave is in phase and other signals are out of phase, the desired wave will be amplified while other noise is attenuated.

Although Beamforming seems theoretically applicable, it has practical problems that prohibit its use. Exact placement of transducers must be made for beamforming to work effectively. In the awkward setting of the rubble pile of a collapsed structure, fast deployment of a beamformer array is not feasible. However, some advantage could be realized by deploying additional geophones, accelerometers, or microphones so extra channels could be processed and tested for distress taps.

7 Potential Hardware Implementations

Several methods of implementing a system to acquire seismic data, filter it and reproduce the filtered data are presented. The first method is to design a system that is unique for this application. Developmental cost for this method would be high. However, if many units were to be produced the total implementation cost would be low. Another method would be to equip a portable programmable computer (PC) with commercially available expansion printed circuit cards to acquire, process, and output the analog signal. Such a system would be much less expensive to develop.

Integrated Circuits

Reports of many systems have been published in which special digital signal processing (DSP) integrated circuits have been used to implement digital FIR, IIR, or adaptive filters. Examples of these integrated circuits include the Texas Instruments TMS32010, TMS3020, TMS320C25, the Motorola DSP56000, the Analog Devices ADSP-2100, and the NEC's uPD77230 [5]. These chips can be used with analog to digital conversion (ADC) integrated circuits, and digital to analog conversion (DAC) integrated circuits to acquire, process, and reproduce a filtered signal in real-time. These DSP integrated circuits can be preprogrammed to perform many different algorithms.

Printed Circuit Cards

An easy system to implement would be to equip a portable IBM compatable PC with a printed circuit card that performs the analog to digital and digital to analog conversions. These cards are available from many different vendors. If high speed 80386 or 80486 microprocessors were used in the PC the computer itself might be able to process all the required filtering. However, if the speed of the computer is not sufficient to perform all the necessary processing, DSP cards are currently available that utilize DSP integrated circuits like the ones discussed previously.

Example of Proposed System

It is recommended that the noise reduction system outlined in the previous section be implemented. This system is being proposed as a prototype and developmental system to be used in both simulated and actual rescue attempts. Hardware, that utilizes PC's to control the filtering, offers the advantage of being easily programmable. It also has the advantage of providing a large number of filtering options to the user. It provides storage for data samples that may be acquired and archived for future study. In addition to the real-time operation of the system, it would be beneficial to save an occasional sample of data so other filtering routines could be evaluated on real data after the emergency response has ended.

Once the routines that provide the best results are determined, the practicality of designing an even less expensive system utilizing DSP integrated circuits should be considered. A typical system could be configured as follows:

a. Portable IBM computer with 2 full length expansion slots; 16 Mbytes of RAM; 200 Mbytes of disk storage; Floppy Disk storage; 486 Processor running at 66MHz; Active Matrix Color VGA display;

b. National instruments AT-DSP2200 Dynamic Signal Acquisition and DSP Board for PC AT. [6]

c. Current Corps of Engineers STOLS unit. [10]

The DSP2200 can provide both ADC, DAC, and DSP capabilities in one card. All this equipment can be commercially purchased with maintenance available from the vendor. No hardware development time would be required, and the software development time could be performed in a high level language like FORTRAN or PASCAL. As the system is used in real or simulated rescue attempts, the effectiveness of the different filters could be evaluated and modified as necessary.

The DSP2200 operates at 25 million floating point operations per second (MFLOPS). A 486DX running at 33MHz can perform at approximately 2.4 MFLOPS. One FIR filter using 128 points would require 128 multiplications and 128 additions or 256 floating point operations for each sample of data. If data was sampled at 1000 samples per second per channel, it would require 256,000 floating point operations for one channel or 512,000 floating point operations for both channels. This indicates that at least 4 different filters could be implemented simultaneously on the 486 computer, requiring 4*(.512 MFLOPS) or 2.048 MFLOPS. The sampling rate of 1000 samples per second and the filter size of 128 points is the maximum that would be required. If the number of points was dropped from 128 to 32 the total number of FIR bandpass filters could be quadrupled. The use of the DSP220 would then increase the complexity or number of filters by a factor of 10.

This proposed system will provide an important tool for evaluating many different noise reduction algorithms in both simulated and actual rescue attempts. It should provide considerable improvements over existing commercially available noise reduction systems. It also provides considerable flexibility and can be modified as continued research reveals more information concerning noise cancellation, the nature of the distress calls, and the nature of the site environments.

8 Conclusions and Recommendations

Three ways of adaptively filtering data have been presented and tested. The first, single channel adaptive filtering, assumes that the noise is periodic and the desired signal is not. When using this method geophones should be placed as close to the signal source as possible. The second method, two channel adaptive filtering, assumes that the signal received on one channel is not correlated with the signal received on the other channel, and that the noise received on one channel is correlated with the noise received on the other. When using this method one geophone should be placed near the signal source and the other in a location that is distant from the signal source but in the same general location as the noise. The third method assumes that the frequency of the signal is different for that of the noise, and that the character of the signal is known to be an impulse. The data can then be filtered into several frequency bands and the ratio of peak signal to standard deviation will provide an estimate of how well the noise has been removed. The frequency band in which this ratio is highest should be monitored. Any number of channels can be used with this method.

Results show that the third method provided the best results in most simulated and field tests. In the field tests both data and noise were correlated so well that the two channel adaptive filtering provided no improvement and usually worsened the quality of the data.

It is recommended that further study focus and include the implementation of a system like that described in Section 7.3. This method demonstrates the best noise reduction data processing method as applied to the detection of trapped victims. Preliminary software should be developed to provide a wide variety of filtering options as addressed in this study and be adaptable to include any new data processing methodologies. Data should be obtained from additional field tests and processed in real-time to demonstrate the robustness of this signal conditioning. The results of each of the filter types or level of complexity should be evaluated for its effectiveness. Such a system would operate substantially better than the current commercially available noise reduction systems. It should be deployed to field crews and serious consideration should be given to manufacturing a specialized system like that described in Chapter 7.

9 References

1. Bernard Widrow, S.D. Stearns, Adaptive Signal Processing, Englewood Cliffs, NJ; Prentice Hall, 1985.

2. A.V. Oppenheim, R.W. Schafer, Digital Signal Processing, Englewood Cliffs, NJ; Prentice Hall, 1975.

3. M.J. Al-Kindi, J. Dunlop, "A Low Distortion Adaptive Noise CancellationStructure For Real Time Applicatons" IEEE Transactions 1987, 49.15.1-49.15.4

4. P.M. Grant, C.F.N. Cowan, "Design and Application of Adaptive Filters" Electronics & Power, Feb. 1985, p157-163.

5. Hen-Geul Yen, "Adaptive Noise Cancellation For Speech With a TMS32020" IEEE Transactions 1987, 27.14.1-27.14.3

6. National Instruments 1994 Data book.

7. S. Haykin, Array Signal Processing, Englewood Cliffs, NJ; Prentice Hall, 1985.

8. Daniel Graupe, Adam J. Efron, "An Output-Whitening Approach to Adaptive Active Noisse Cancellation" IEEE Transactions 1919, 1306-1313.9.

9. G.A. Powell, P. Darlington, P.D. Wheller, "Practical Adaptive Noise Reduction in the Aircraft Cockpit Environment" IEEE Transactions, 1987, 6.2.1-6.2.4.

10. R. Lewis, L. Smithhart, "Technical Urban Search and Rescue System to Locate Survivors (STOLS) Units. Operators Manual", USAE-WES-GL-93-**.

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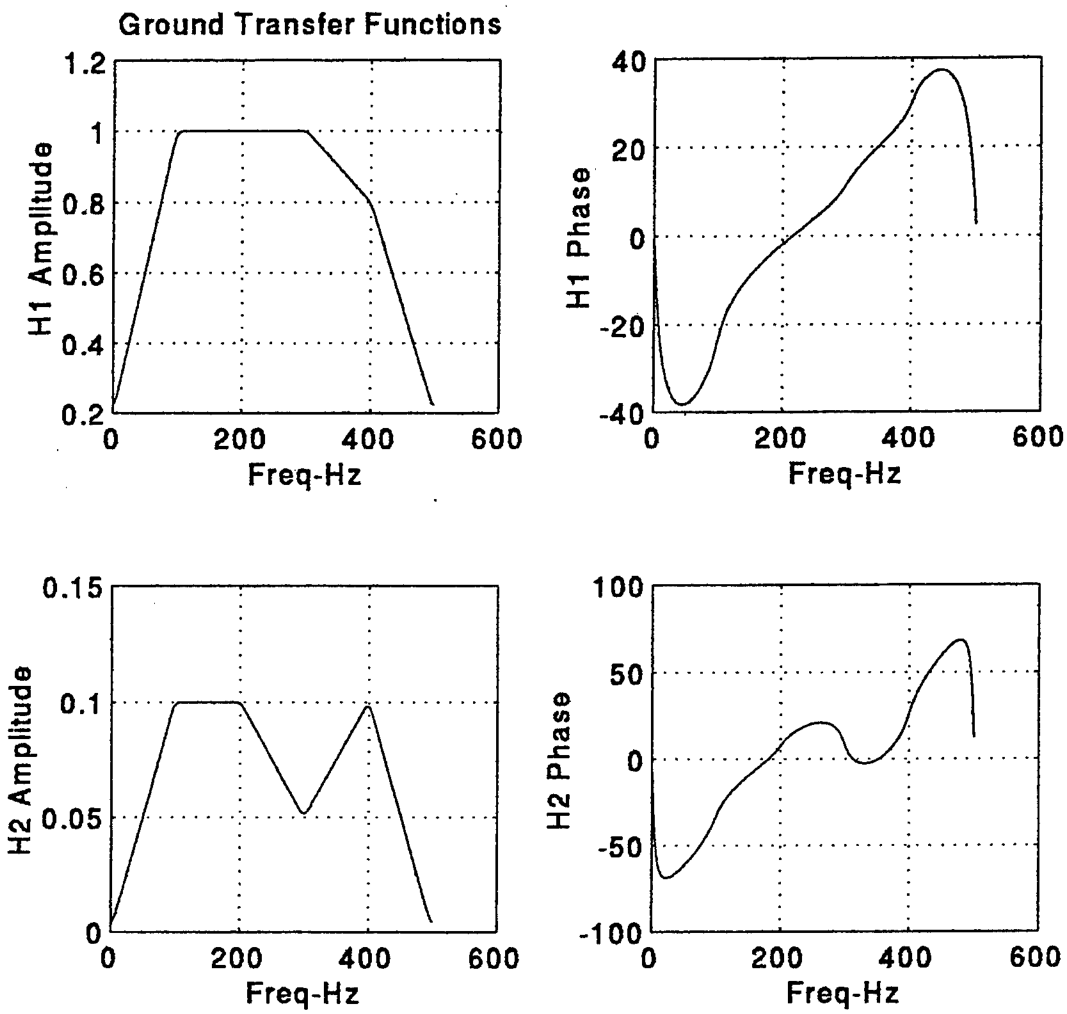
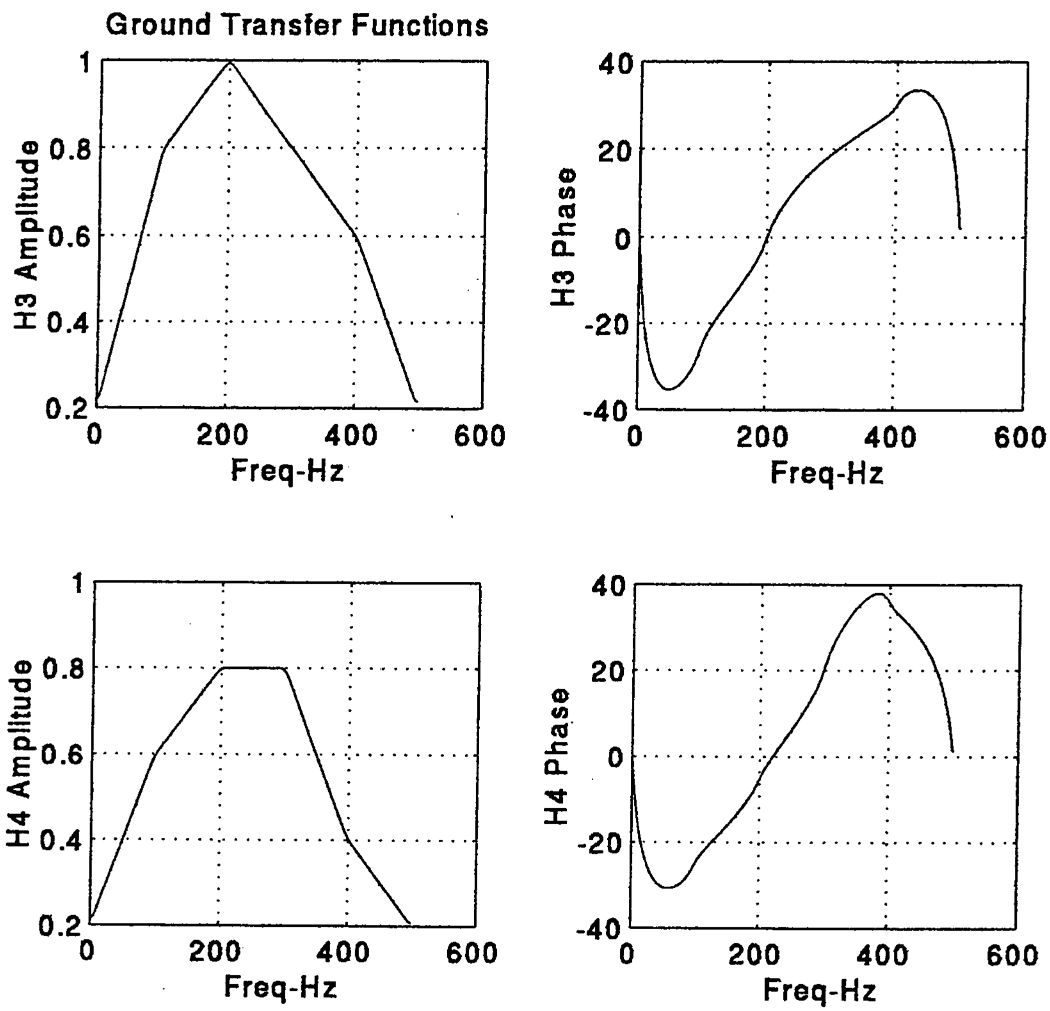
Appendix A
Analysis of Simulated Data

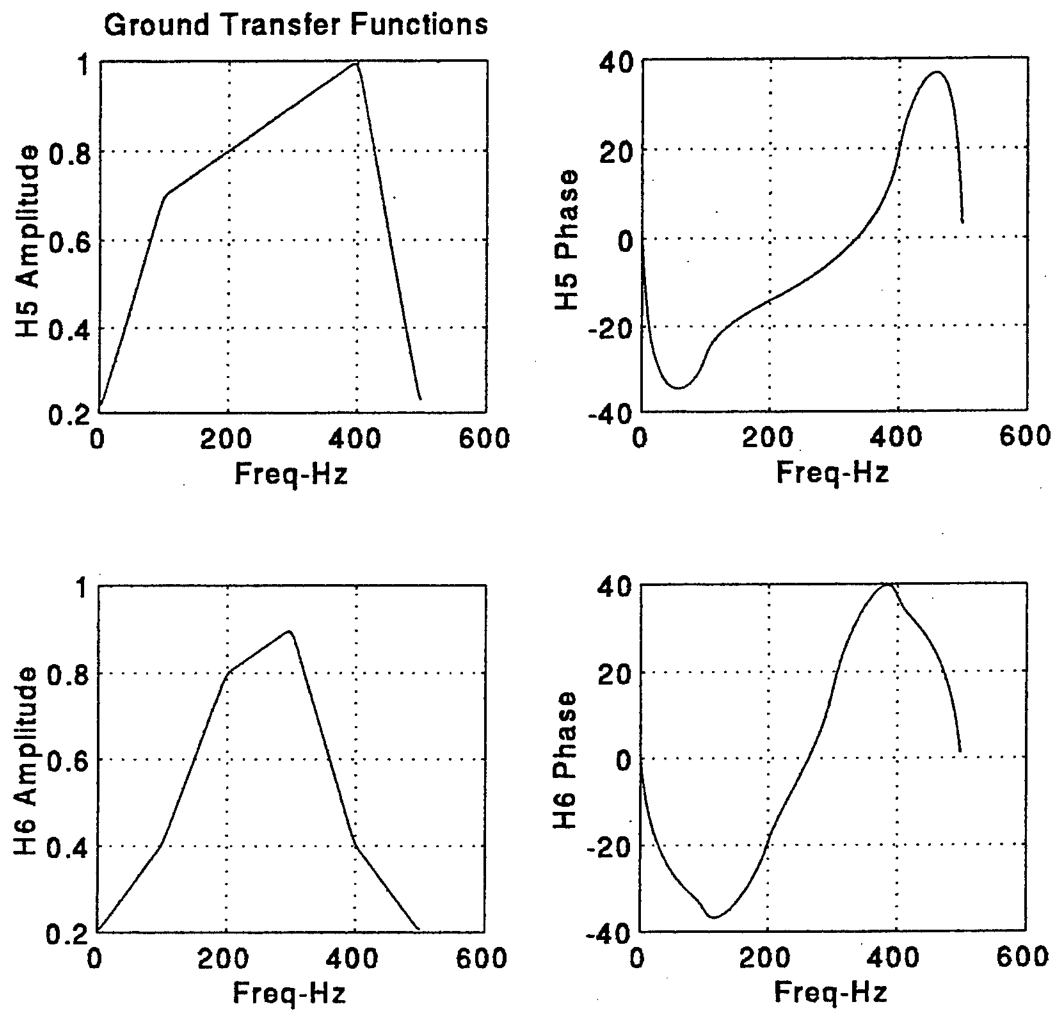
Appendix A contains plots of the simulated ground transfer functions used to generate two channels of simulated data. (Referenced on page 10).
This appendix also contains the simulated data's time history, PSD, coherence, and transfer function. (Referenced on page 12).

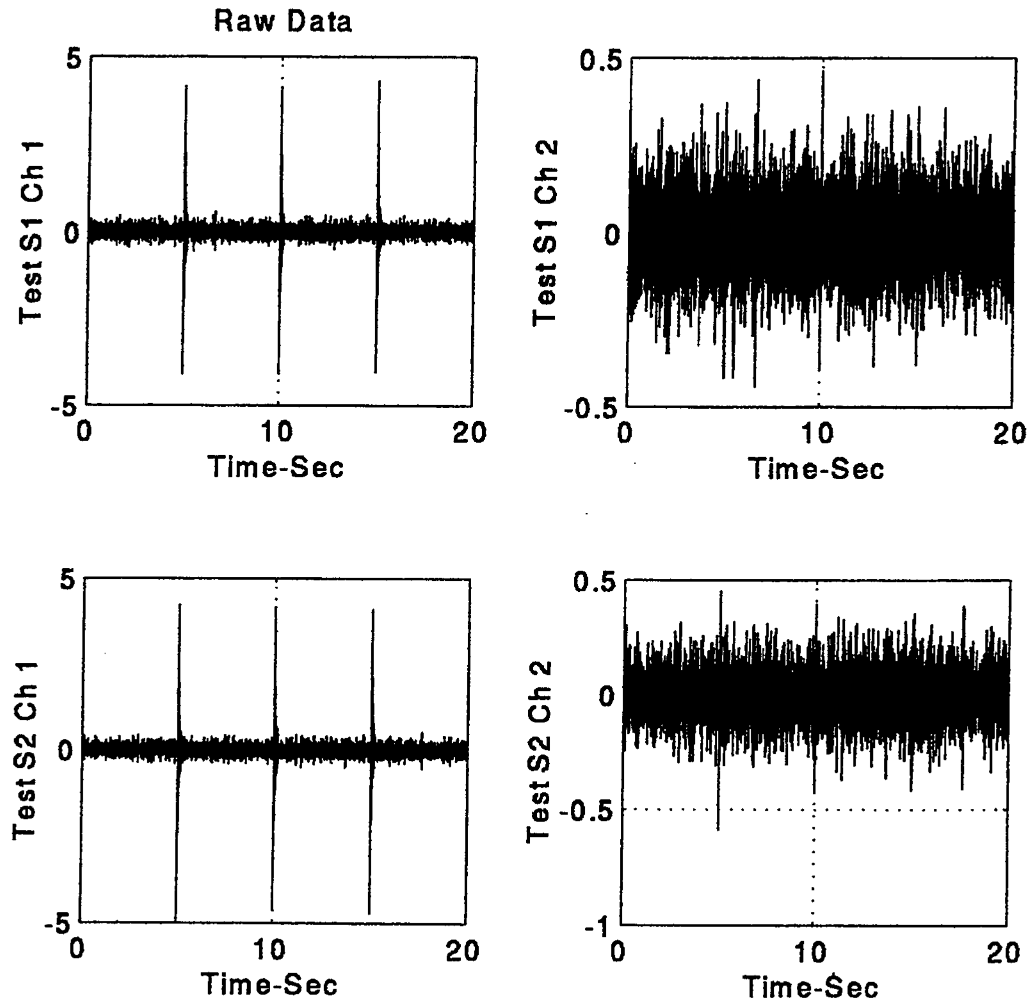


Ground Transfer Functions
1.2
1
0.8
0.6
0.4
0.2
H1 Amplitude
0
200
400
600
Freq-Hz
40
20
0
-20
-40
H1 Phase
0
200
400
600
Freq-Hz
0.15
0.1
0.05
0
H2 Amplitude
0
200
400
600
Freq-Hz
100
50
0
-50
-100
H2 Phase
0
200
400
600
Freq-Hz

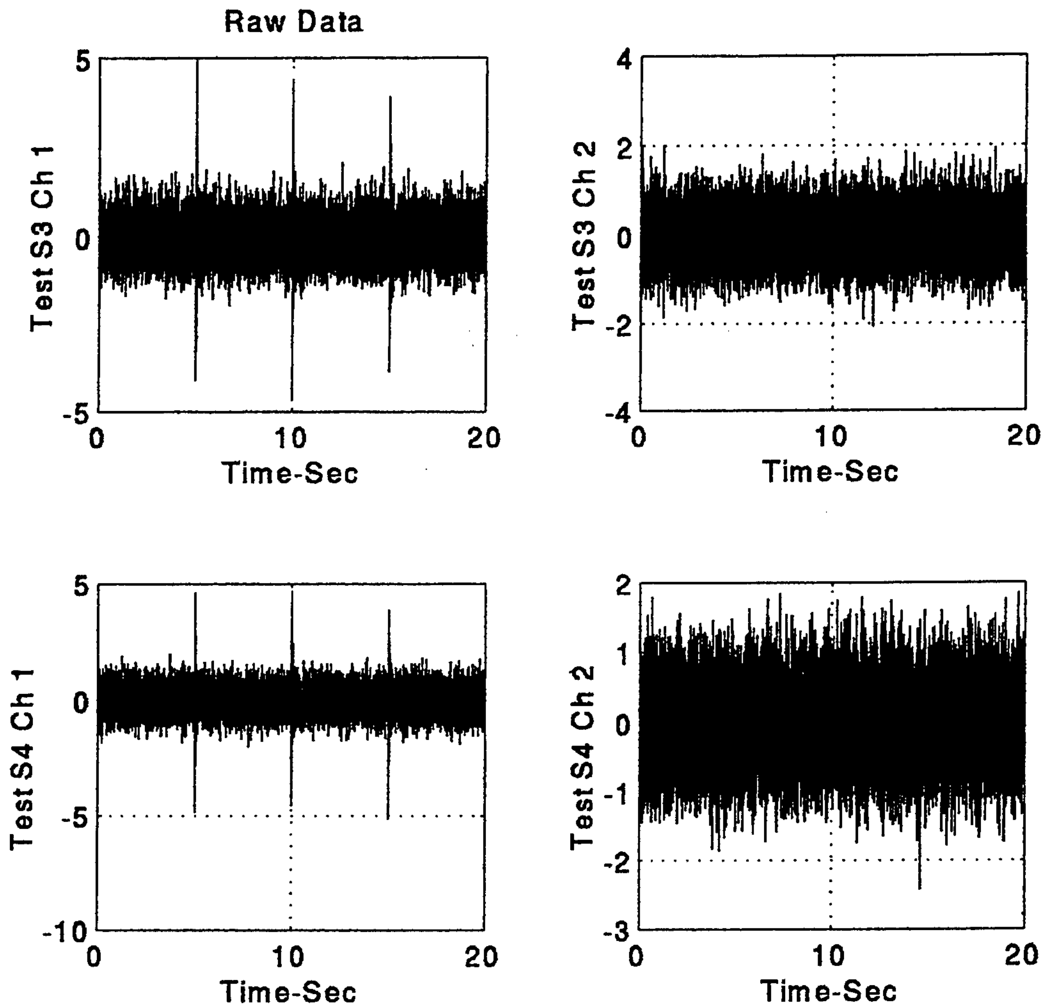


Ground Transfer Functions
H3 Amplitude
H3 Phase
H4 Amplitude
H4 Phase
Freq-Hz
0
200
400
600
1
0.8
0.6
0.4
0.2
40
20
0
-20
-40


Ground Transfer Functions
H5 Amplitude
H5 Phase
H6 Amplitude
H6 Phase
Freq-Hz
Freq-Hz
Freq-Hz
Freq-Hz
1
0.8
0.6
0.4
0.2
40
20
0
-20
-40
0
200
400
600

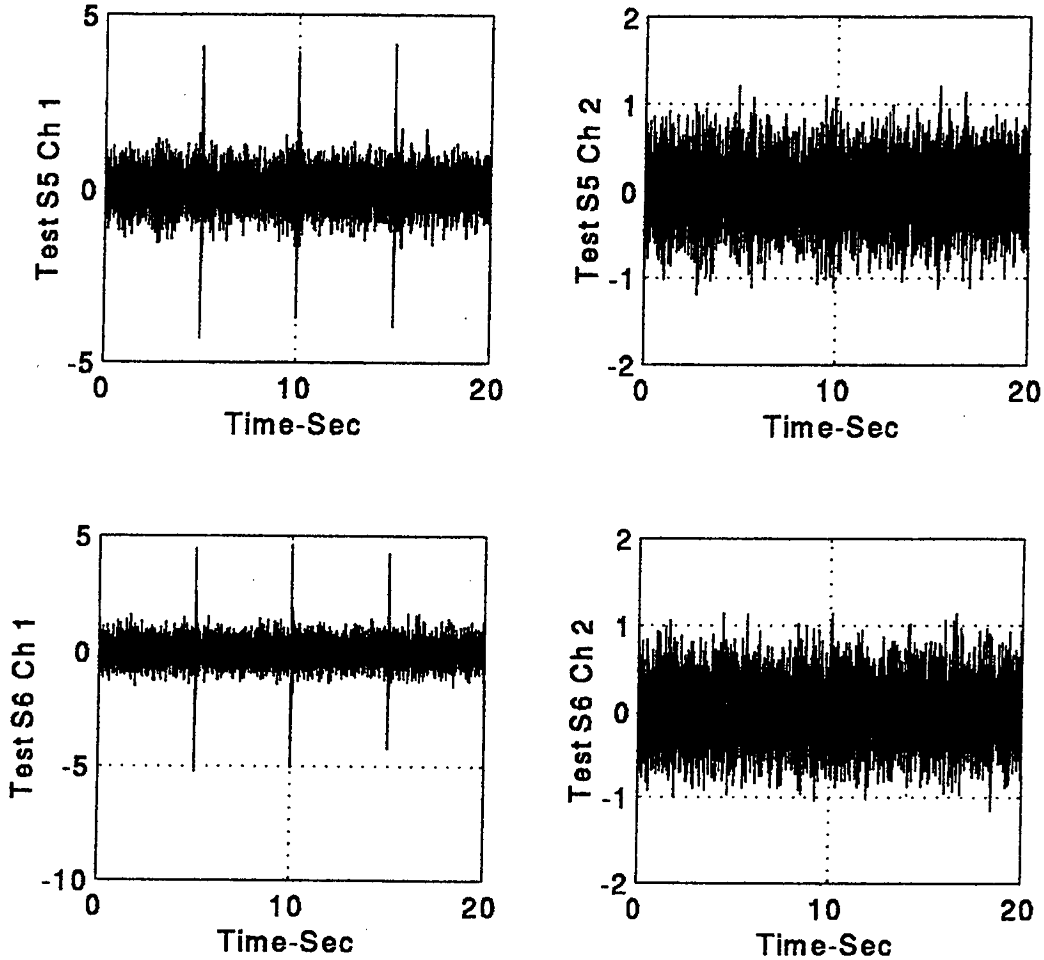


Raw Data
Test S1 Ch 1
Test S1 Ch 2
Test S2 Ch 1
Test S2 Ch 2
Time-Sec
5
0
-5
0.5
-0.5
-1
10
20

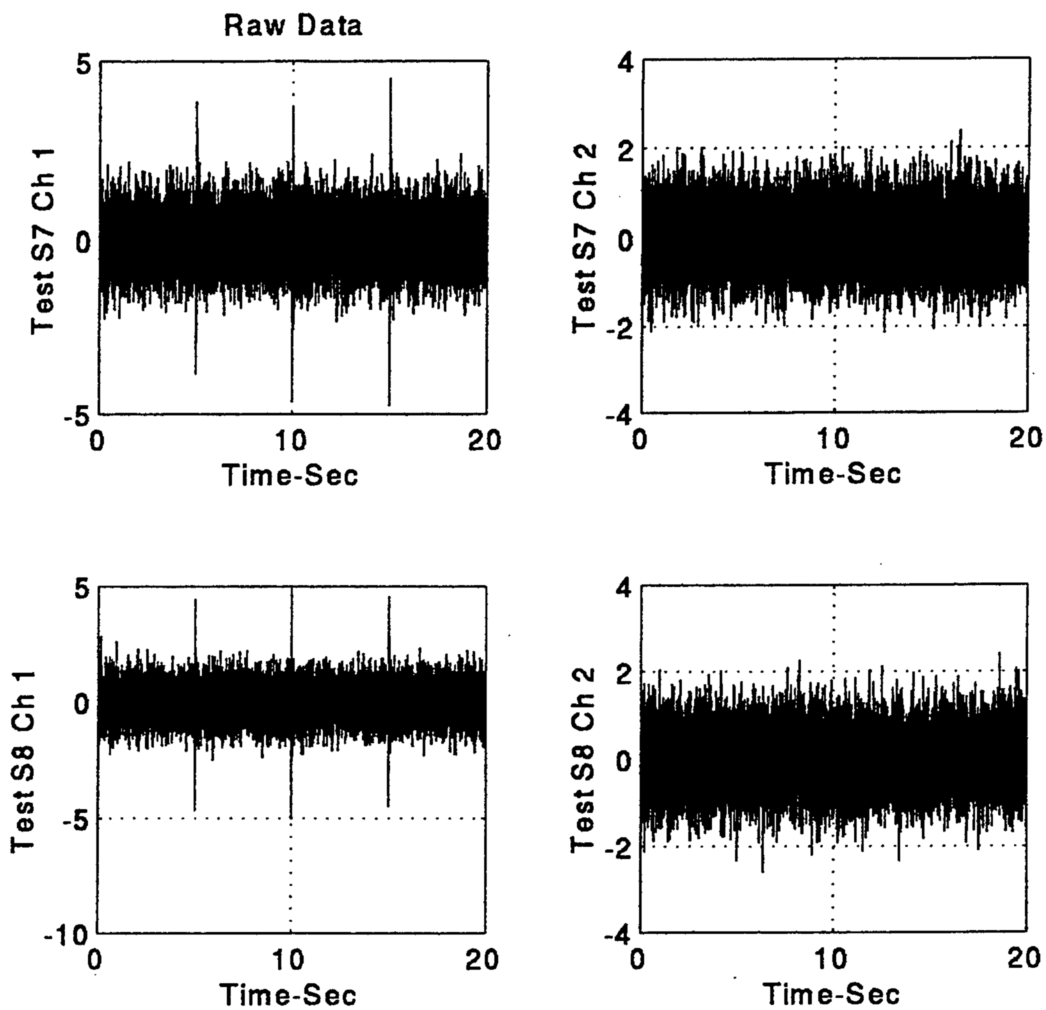


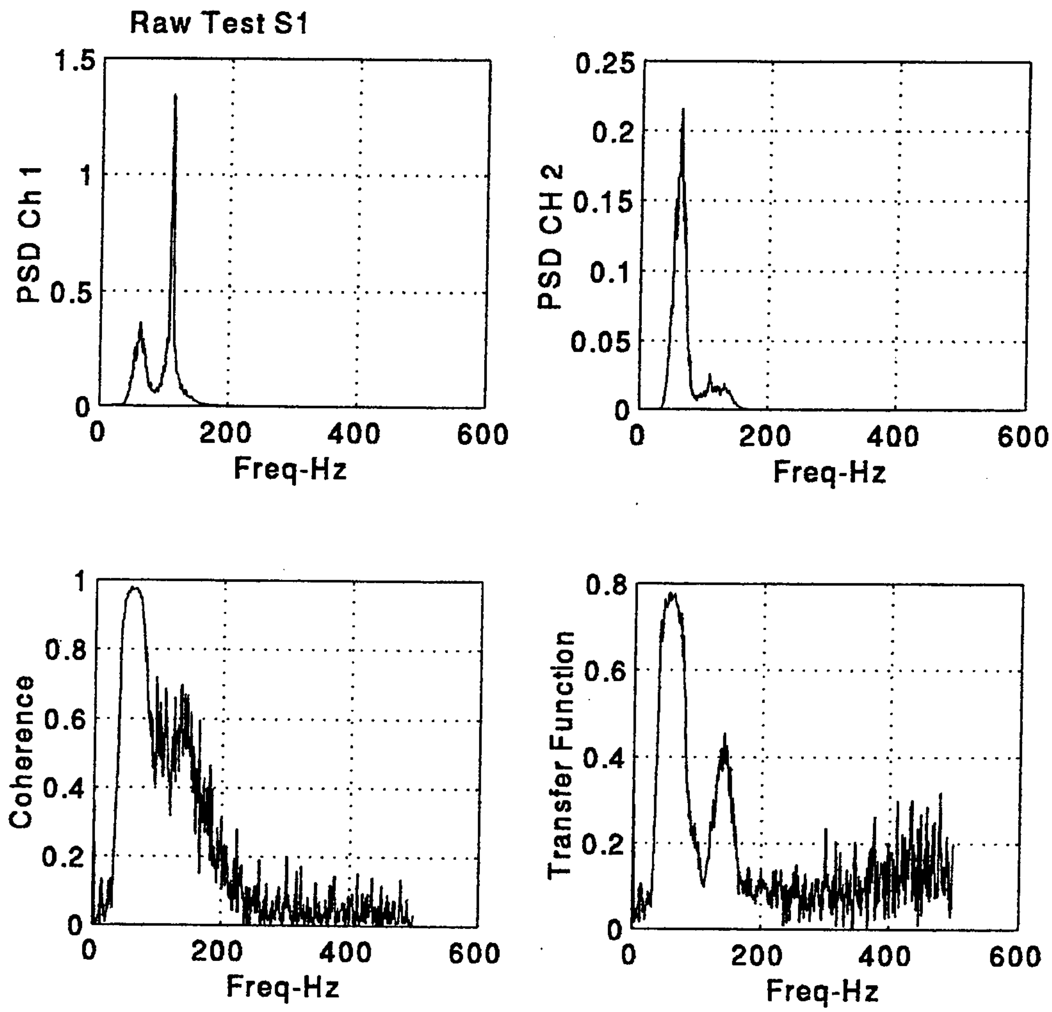
Raw Data
Test S3 Ch 1
Time-Sec
Test S3 Ch 2
Time-Sec
Test S4 Ch 1
Time-Sec
Test S4 Ch 2
Time-Sec

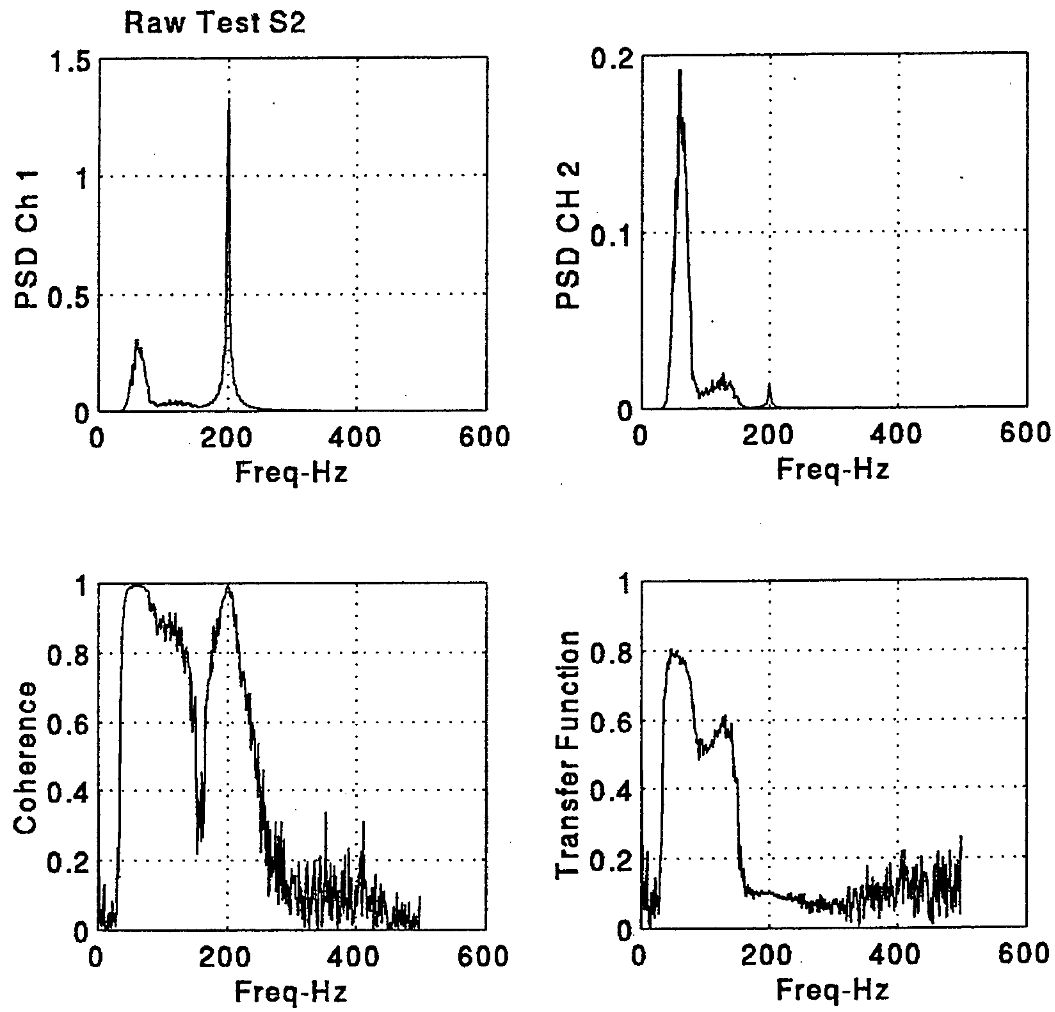
Raw Data

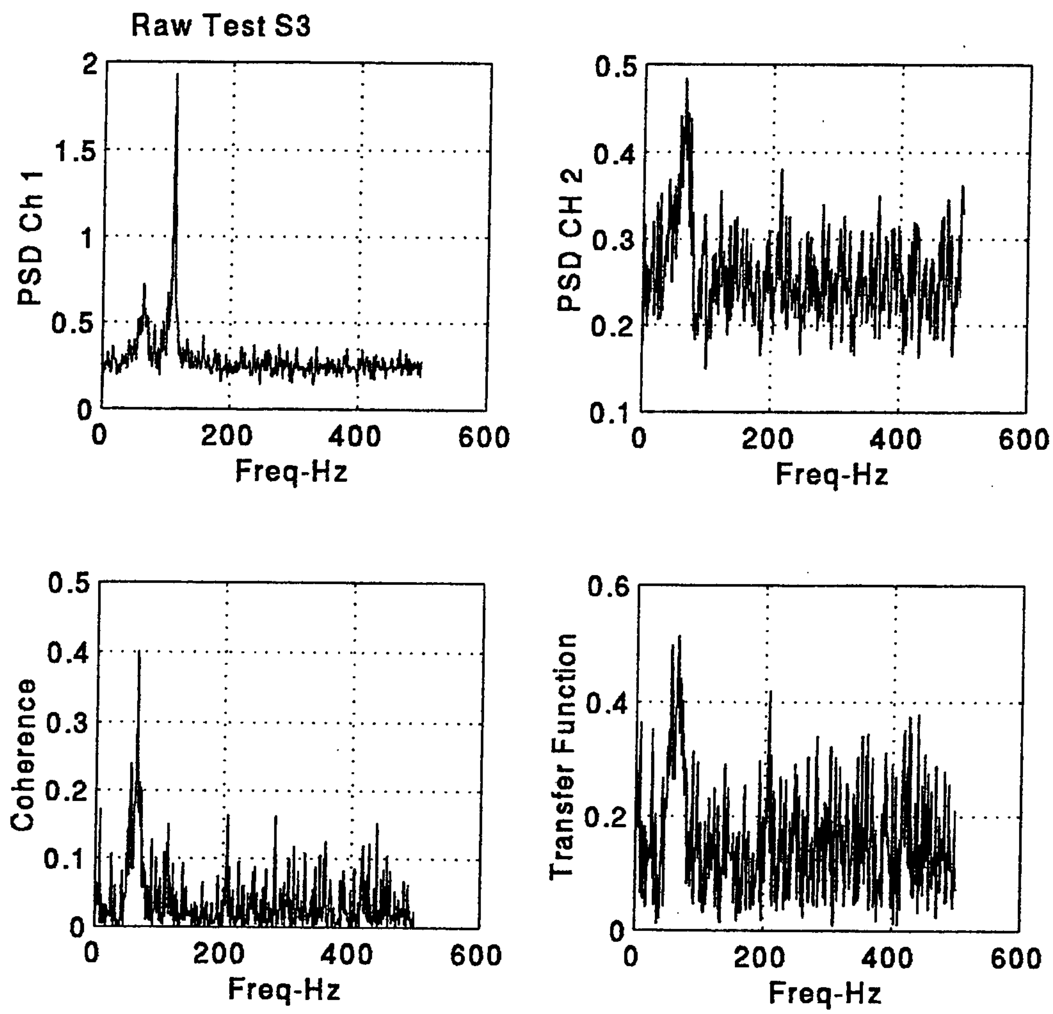


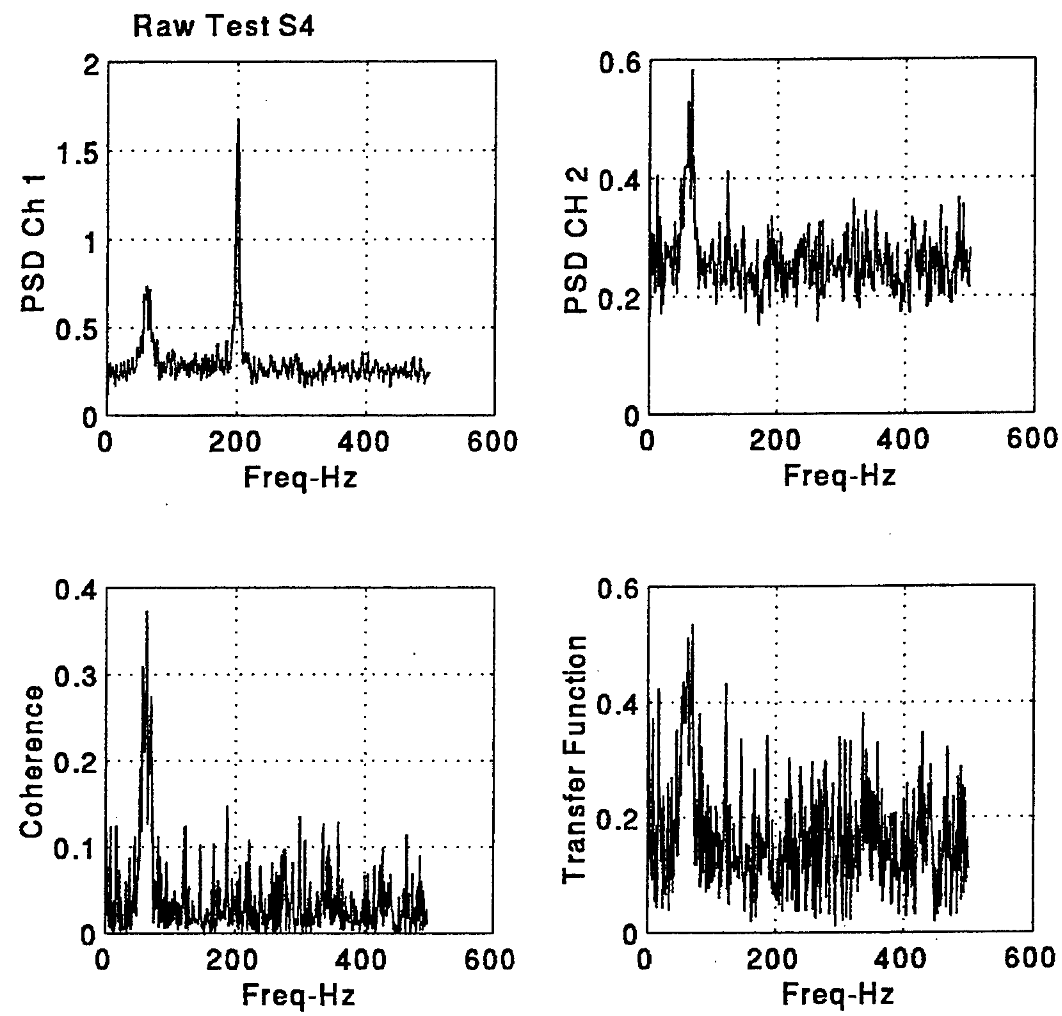
5
0
-5
0
10
20
Time-Sec
Test S5 Ch 1
2
1
0
-1
-2
0
10
20
Time-Sec
Test S5 Ch 2
5
0
-5
-10
0
10
20
Time-Sec
Test S6 Ch 1
2
1
0
-1
-2
0
10
20
Time-Sec
Test S6 Ch 2

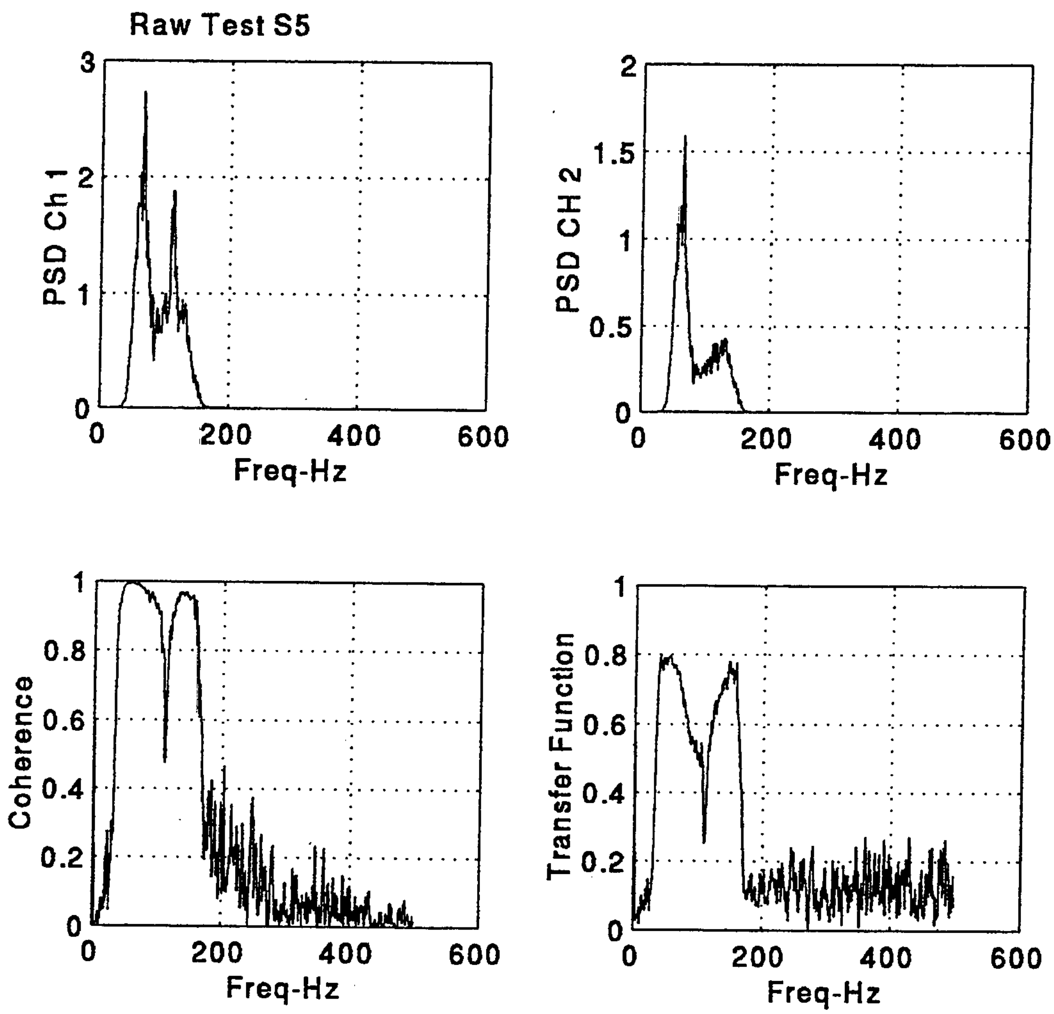

Raw Data
Test S7 Ch 1
5
0
-5
0
10
20
Time-Sec
Test S7 Ch 2
4
2
0
-2
-4
0
10
20
Time-Sec
Test S8 Ch 1
5
0
-5
-10
0
10
20
Time-Sec
Test S8 Ch 2
4
2
0
-2
-4
0
10
20
Time-Sec

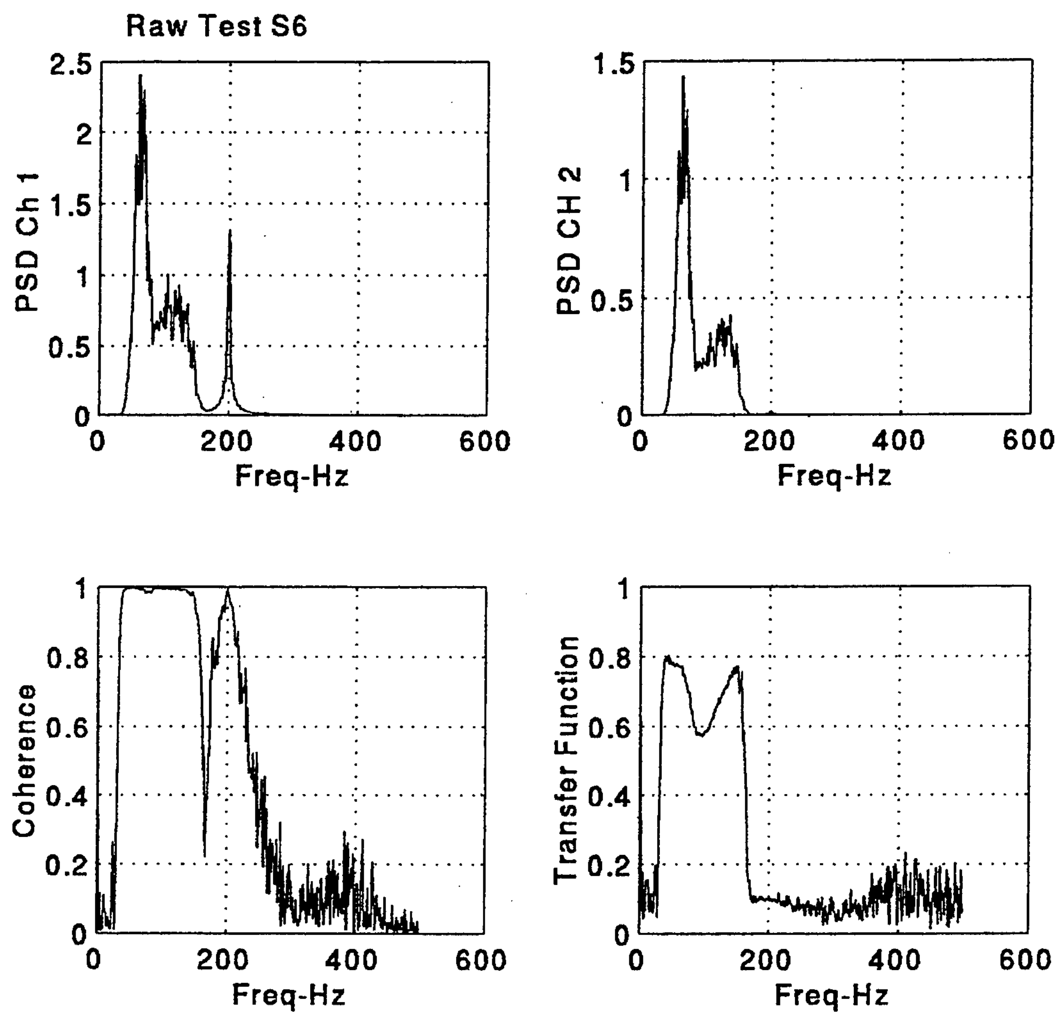

Raw Test S1
PSD Ch 1
PSD CH 2
Coherence
Transfer Function
Freq-Hz
Freq-Hz
Freq-Hz
Freq-Hz


Raw Test S2
PSD Ch 1
PSD CH 2
Coherence
Transfer Function
Freq-Hz
Freq-Hz
Freq-Hz
Freq-Hz

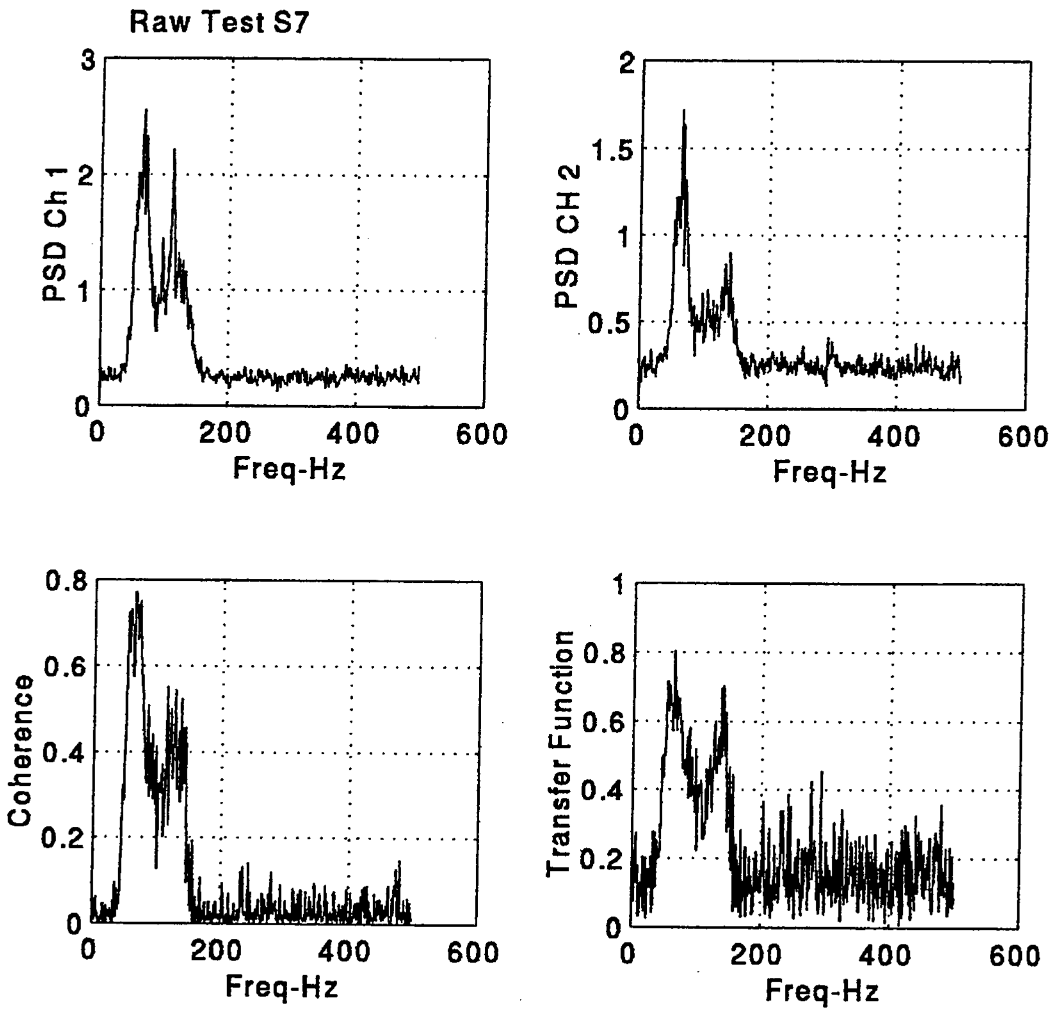

Raw Test S3
PSD Ch 1
Freq-Hz
PSD CH 2
Freq-Hz
Coherence
Freq-Hz
Transfer Function
Freq-Hz

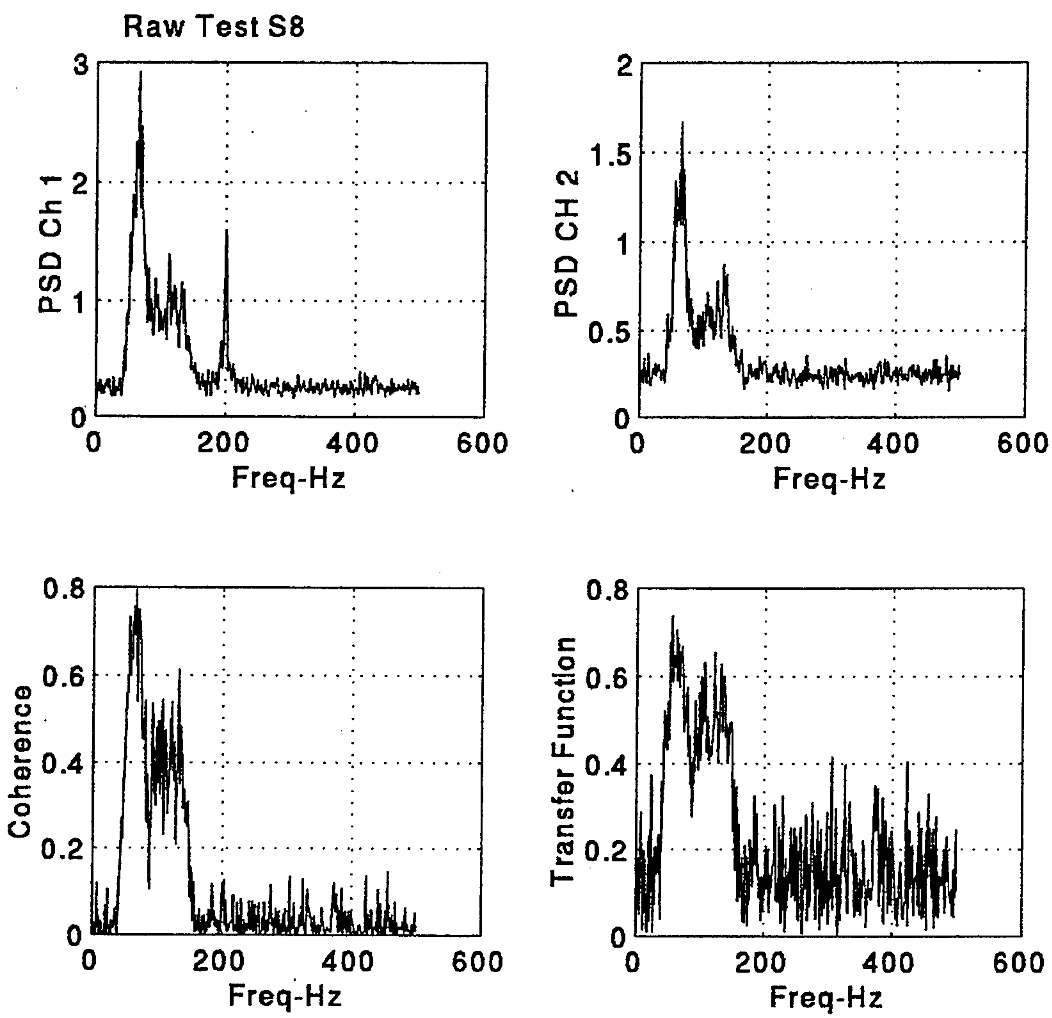

Raw Test S4
PSD Ch 1
PSD CH 2
Coherence
Transfer Function
Freq-Hz
Freq-Hz
Freq-Hz
Freq-Hz


Raw Test S5
PSD Ch 1
PSD CH 2
Coherence
Transfer Function
Freq-Hz
Freq-Hz
Freq-Hz
Freq-Hz
0
200
400
600



Raw Test S6
PSD Ch 1
PSD CH 2
Coherence
Transfer Function
Freq-Hz
Freq-Hz
Freq-Hz
Freq-Hz

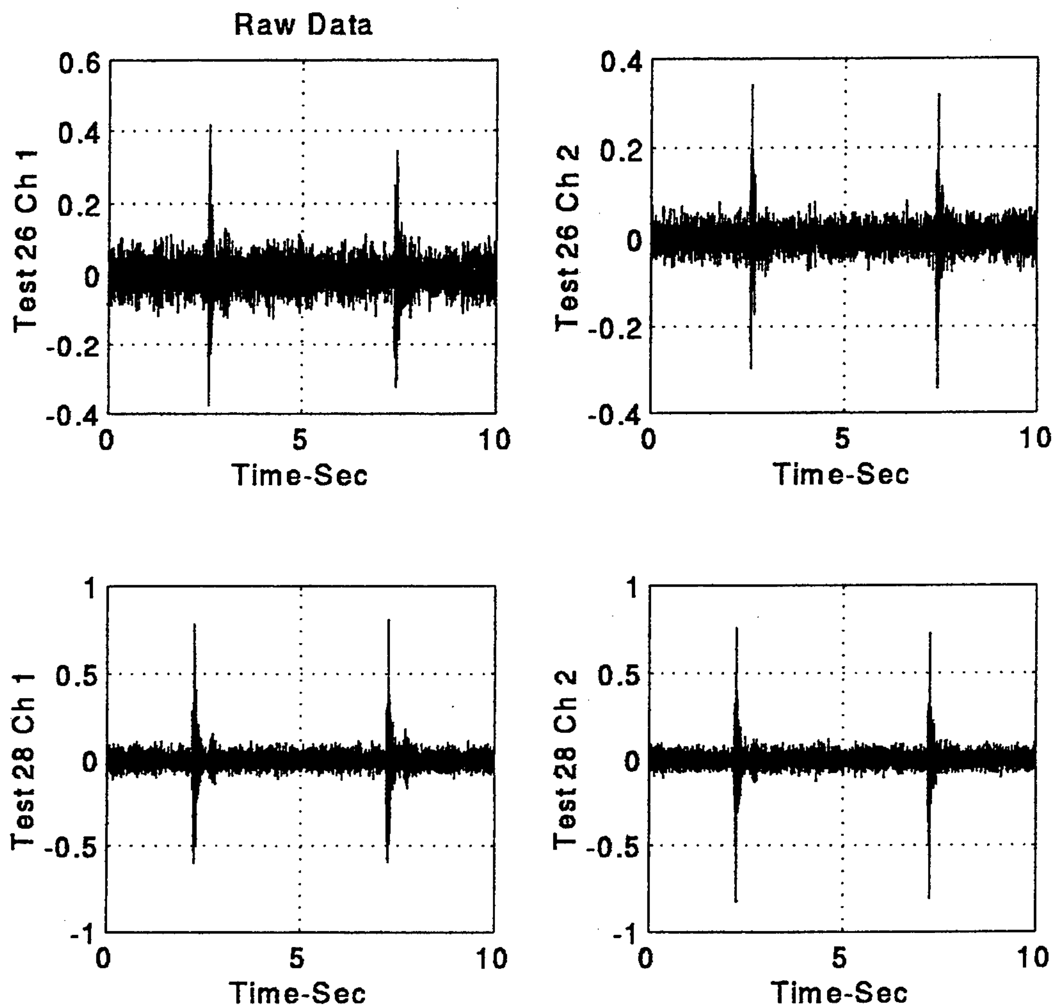

Raw Test S7
PSD Ch 1
PSD CH 2
Coherence
Transfer Function
Freq-Hz
0
200
400
600

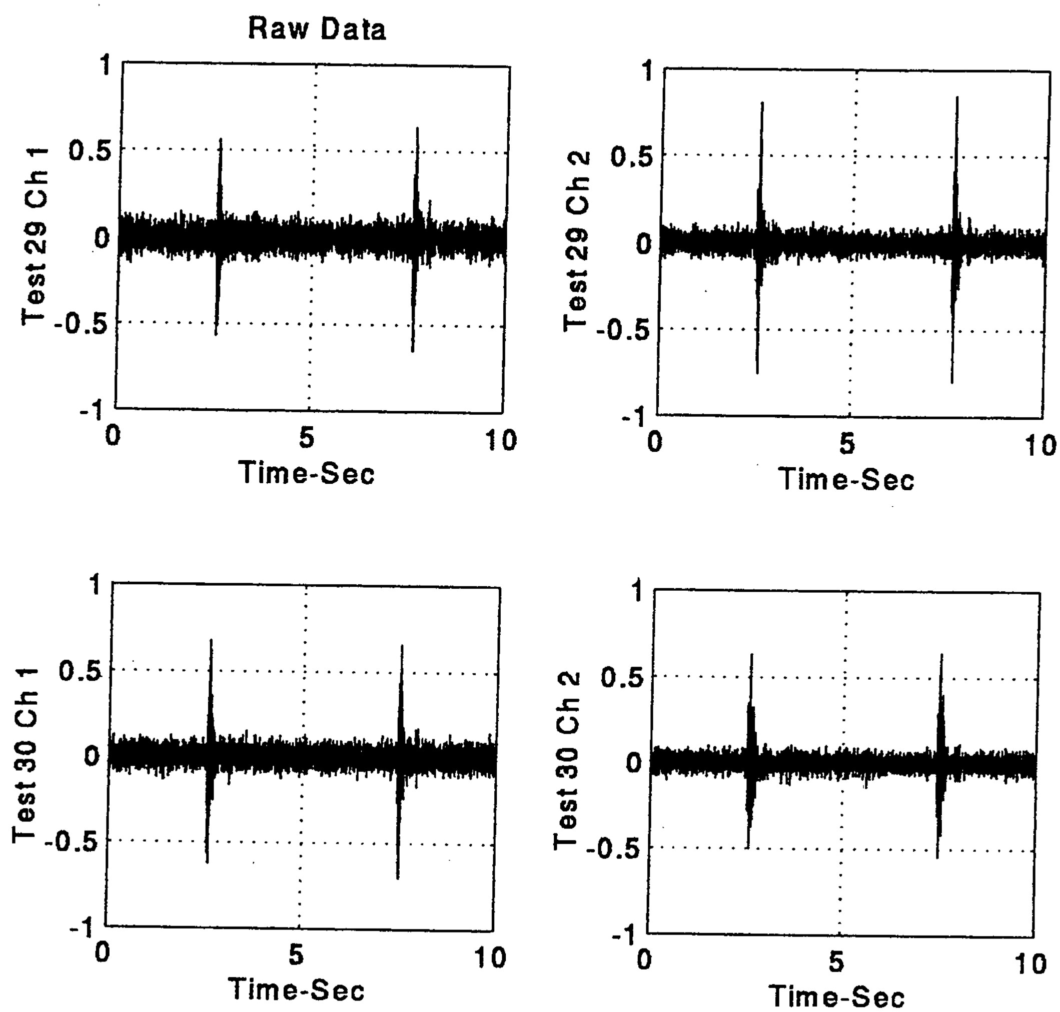


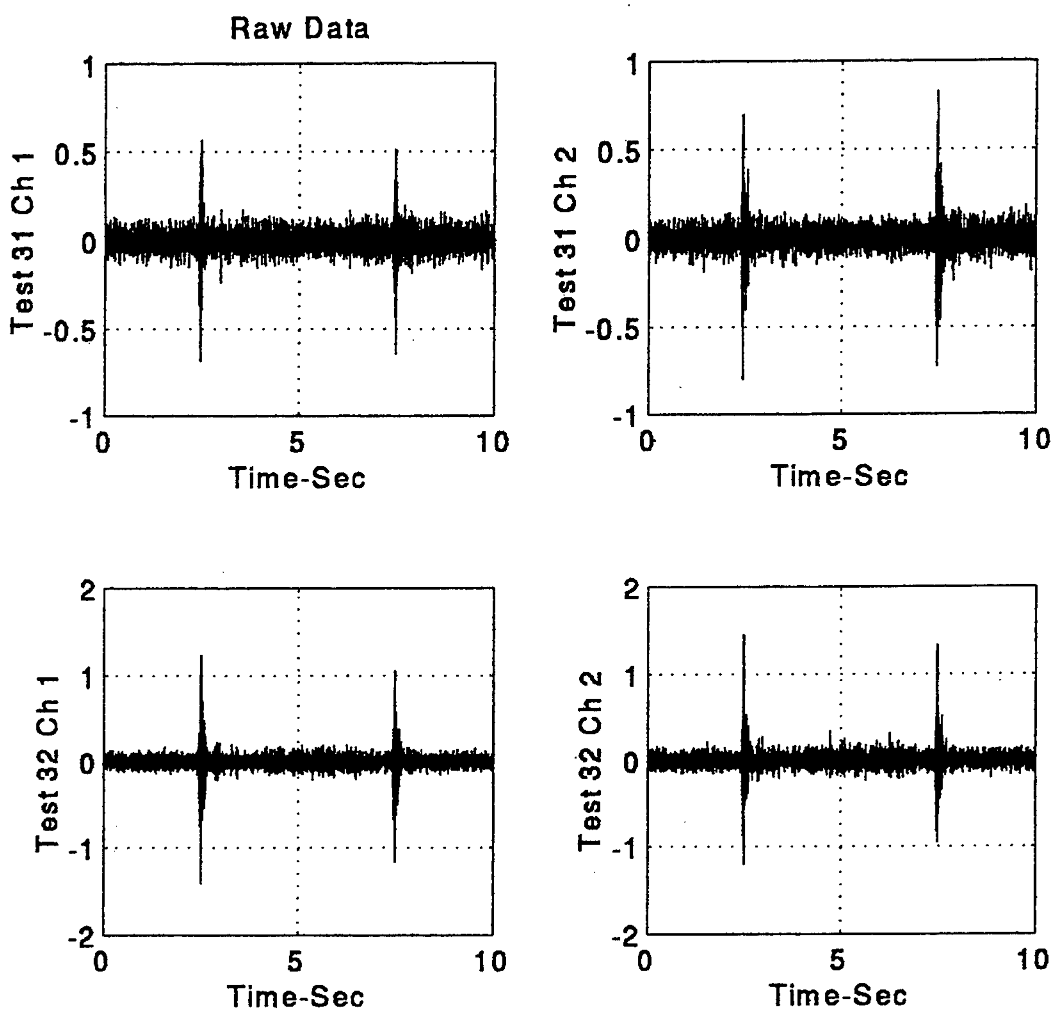
Raw Test S8
PSD Ch 1
Freq-Hz
PSD CH 2
Freq-Hz
Coherence
Freq-Hz
Transfer Function
Freq-Hz

Appendix B
Analysis of Field Data

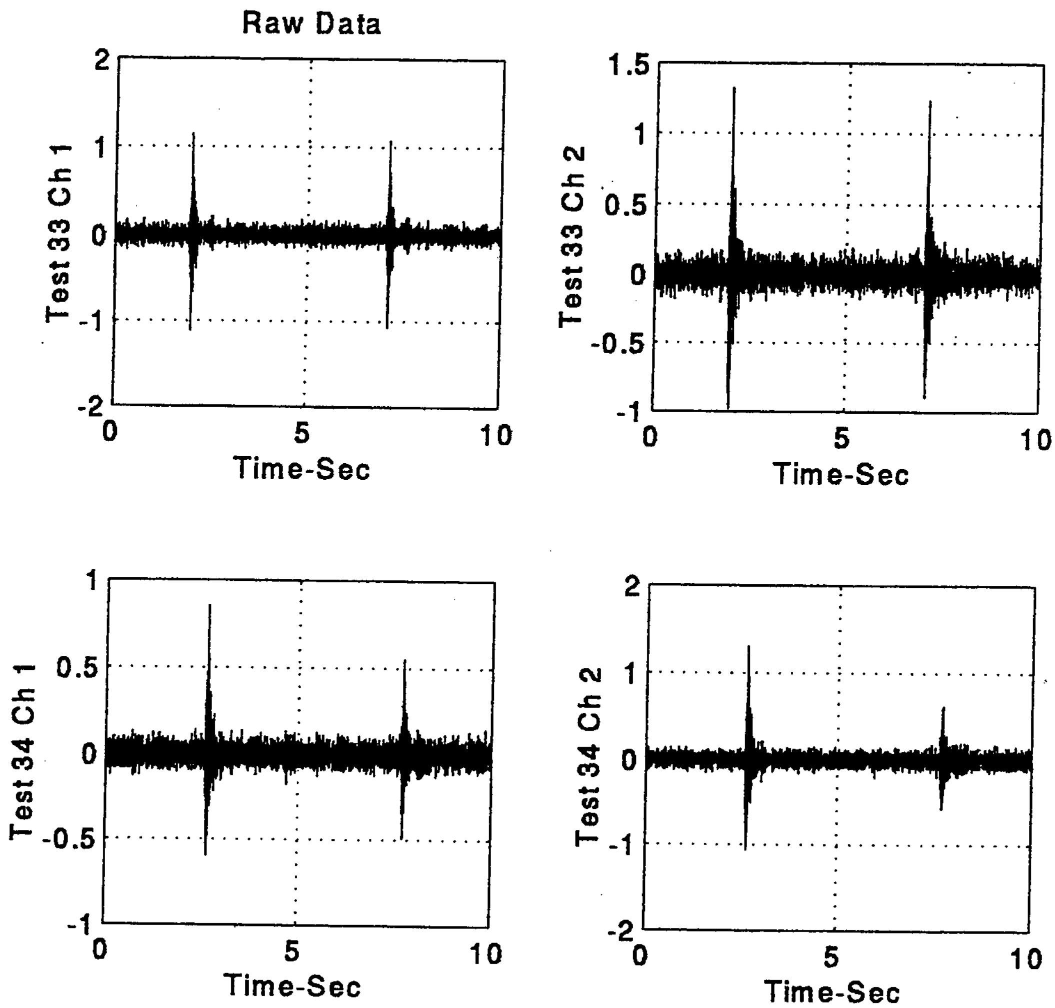
Appendix B contains plots of the raw time history for two channels of the field data. It also contains the raw data's PSD, coherence, and transfer functions. (Referenced on page 32).
This appendix also contains these same plots for its signal component and its noise component. (Referenced on pages 33 and 35).

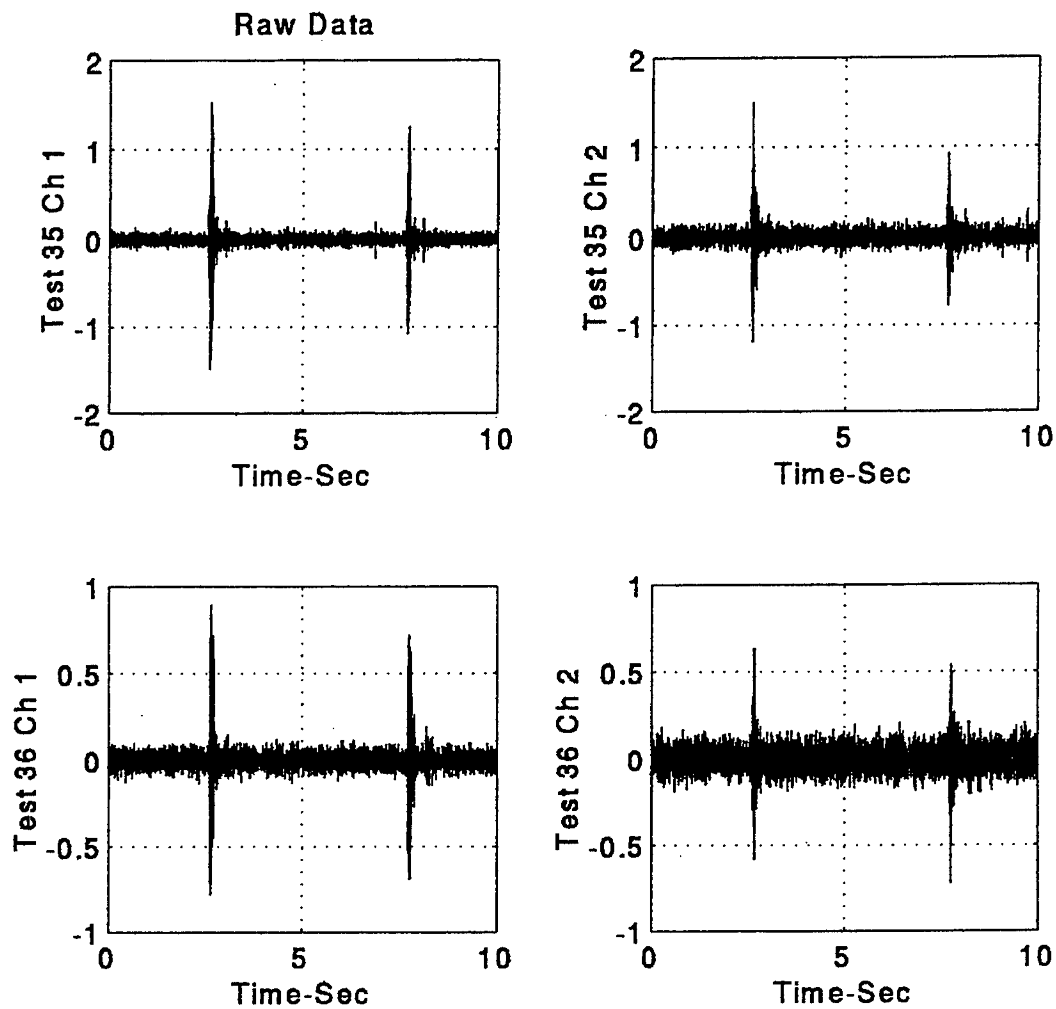

Raw Data
Test 26 Ch 1
Test 26 Ch 2
Test 28 Ch 1
Test 28 Ch 2
Time-Sec
Time-Sec
Time-Sec
Time-Sec

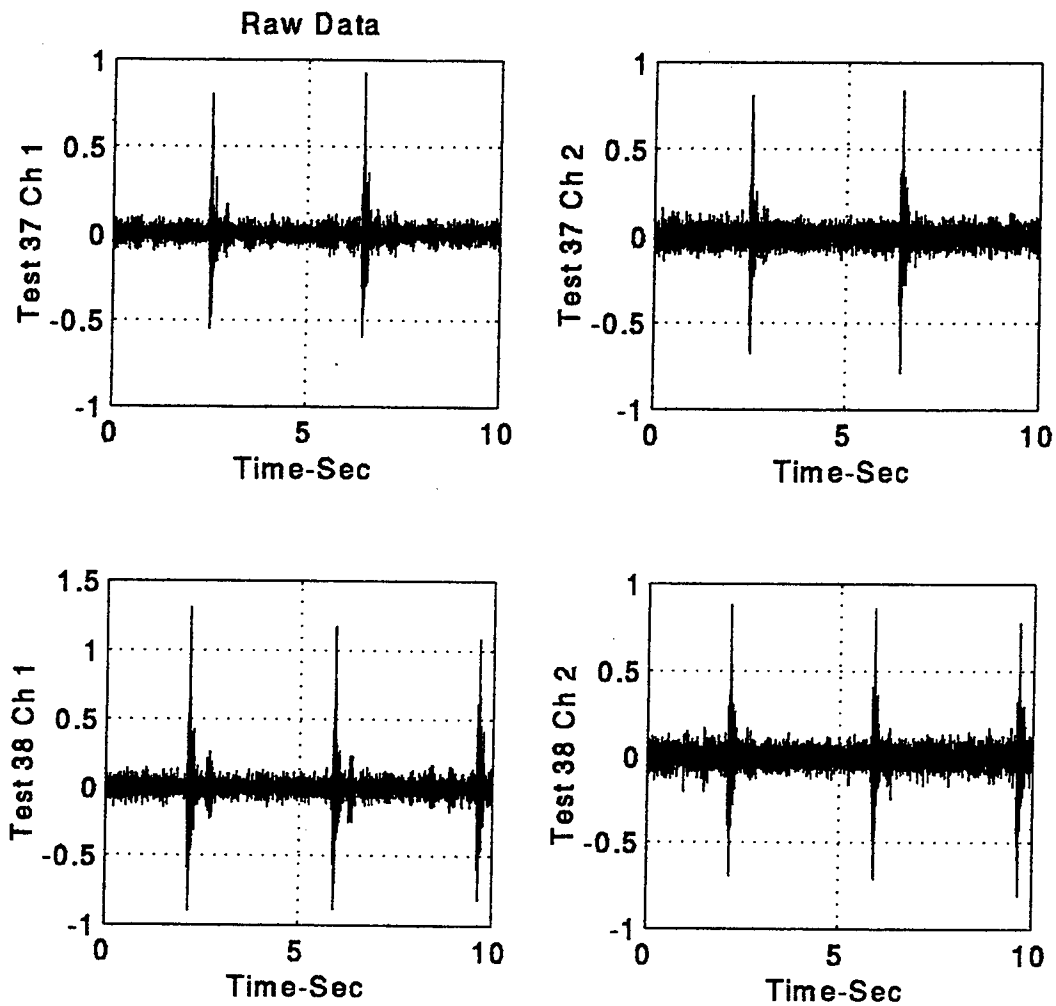

Raw Data
Test 29 Ch 1
Test 29 Ch 2
Test 30 Ch 1
Test 30 Ch 2
Time-Sec
1
0.5
0
-0.5
-1
0
5
10

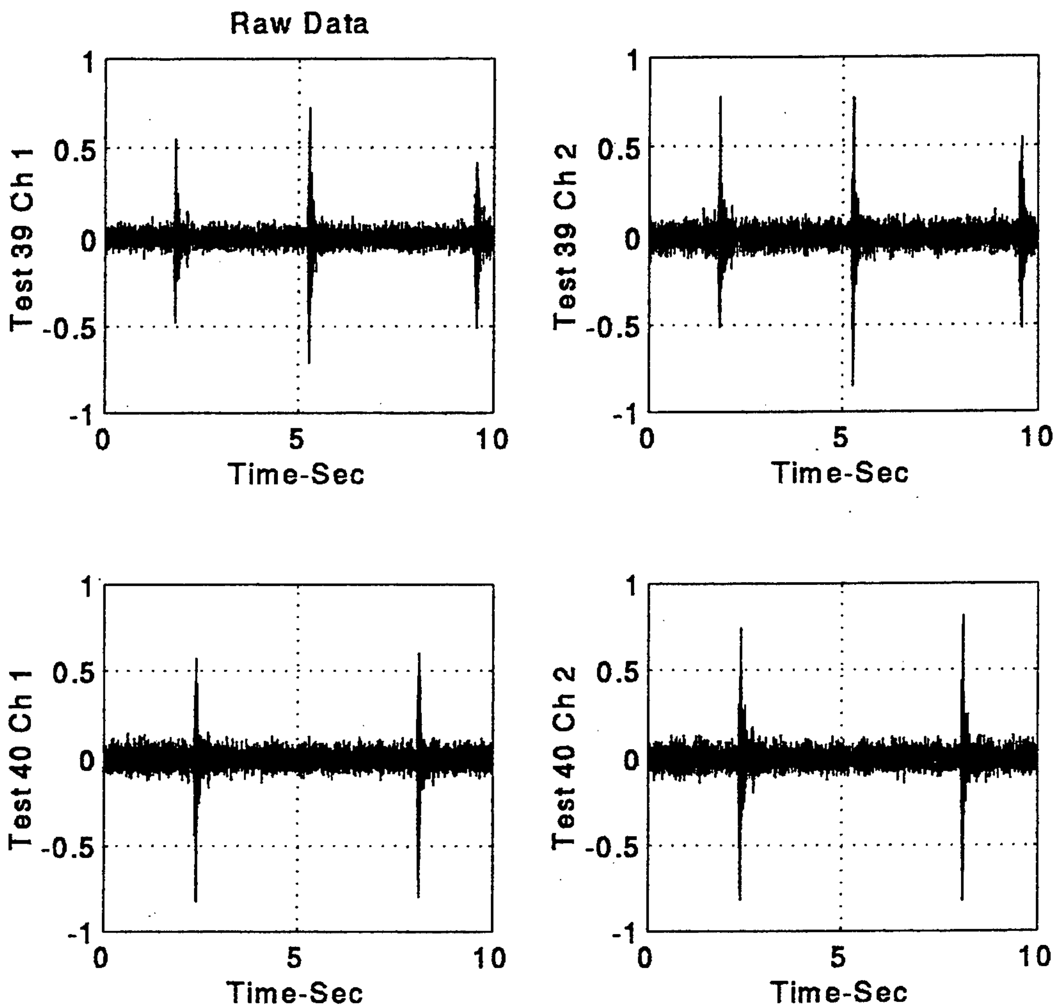


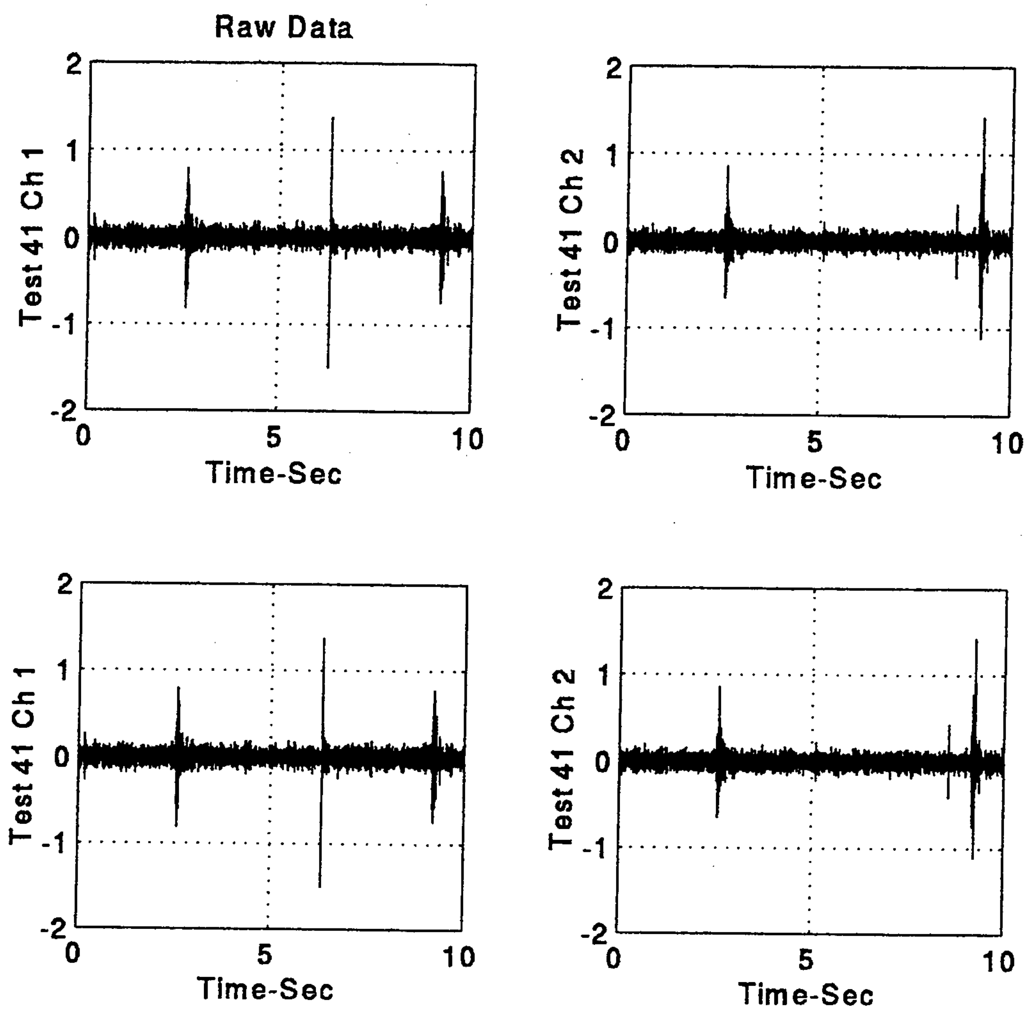
Raw Data
Test 31 Ch 1
Test 31 Ch 2
Test 32 Ch 1
Test 32 Ch 2
Time-Sec
Time-Sec
Time-Sec
Time-Sec


Raw Data
Test 33 Ch 1
Time-Sec
Test 33 Ch 2
Time-Sec
Test 34 Ch 1
Time-Sec
Test 34 Ch 2
Time-Sec

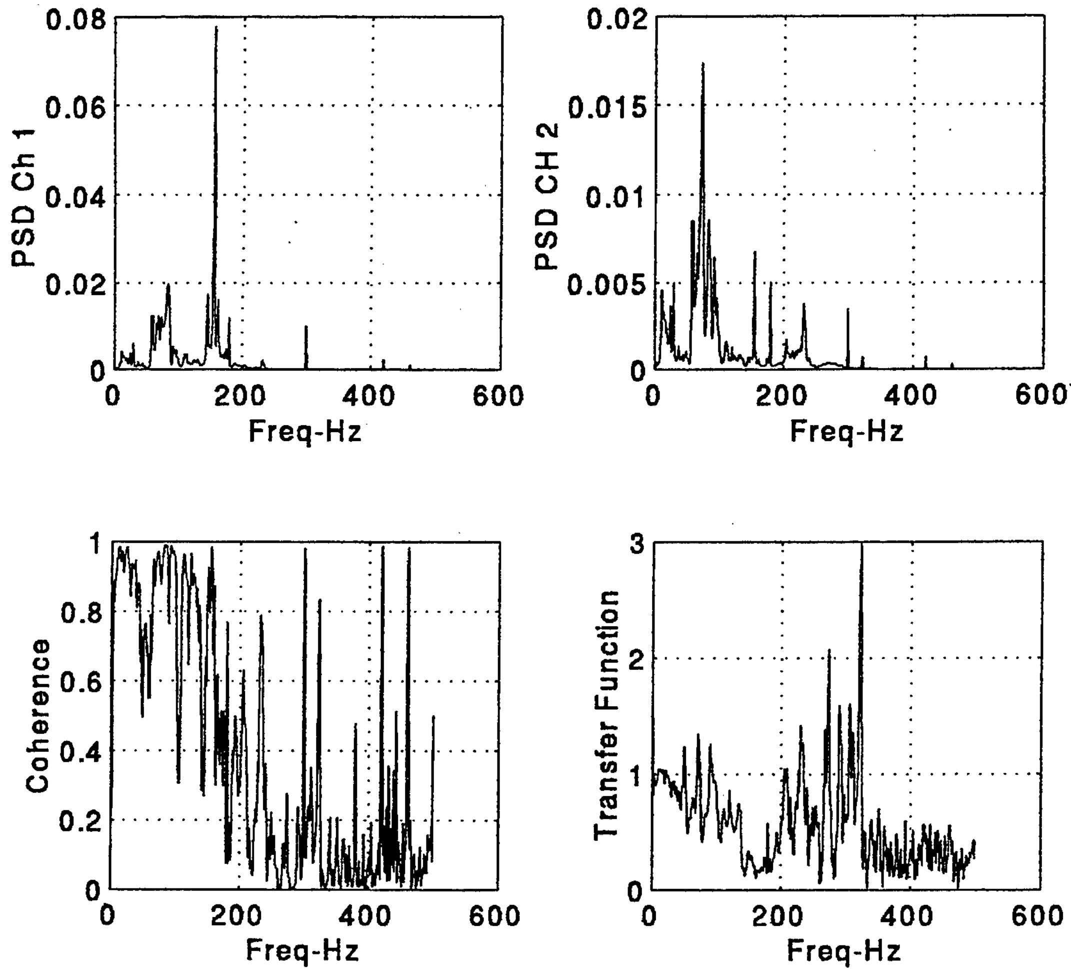

Raw Data
Test 35 Ch 1
Test 35 Ch 2
Test 36 Ch 1
Test 36 Ch 2
Time-Sec
2
1
0
-1
-2
0.5
-0.5
5
10

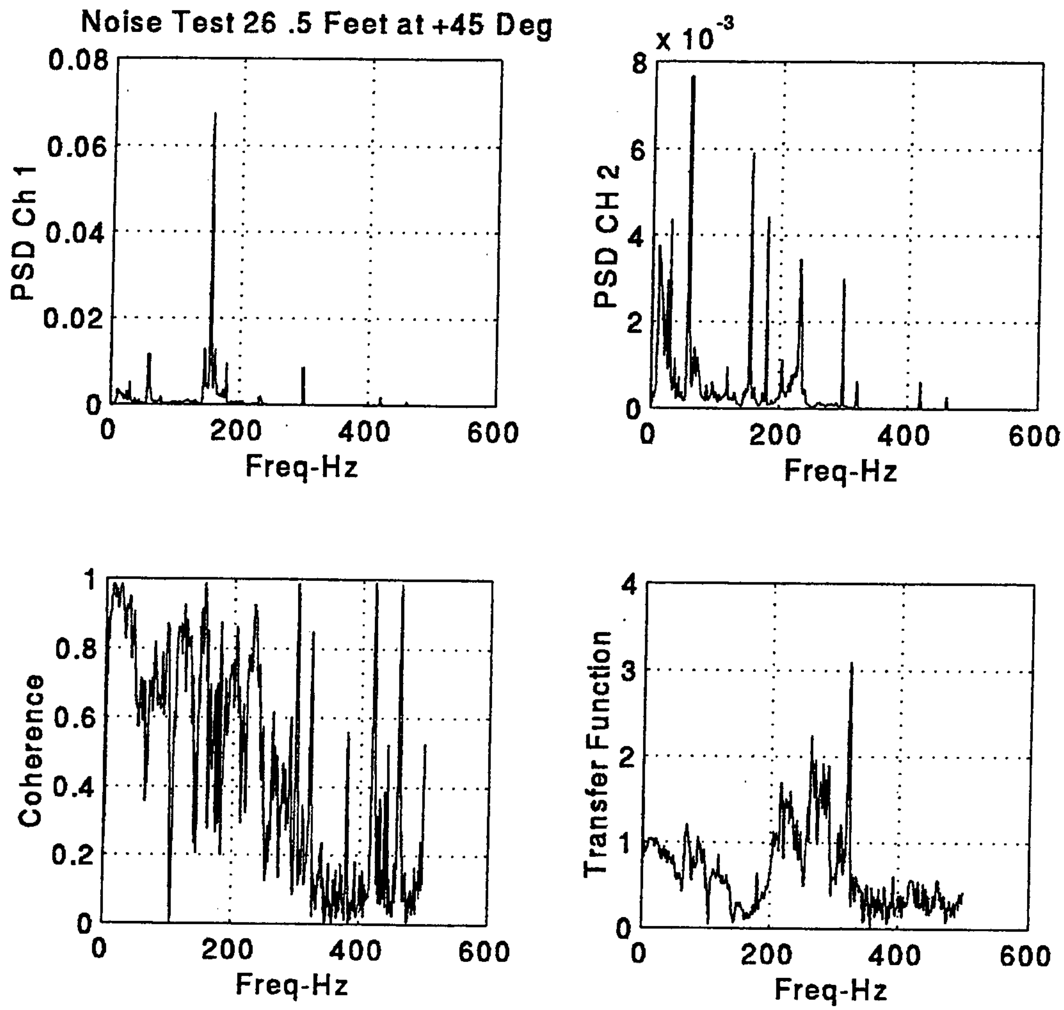

Raw Data
Test 37 Ch 1
Test 37 Ch 2
Test 38 Ch 1
Test 38 Ch 2
Time-Sec
Time-Sec
Time-Sec
Time-Sec
1
0.5
0
-0.5
-1
1.5
0
5
10

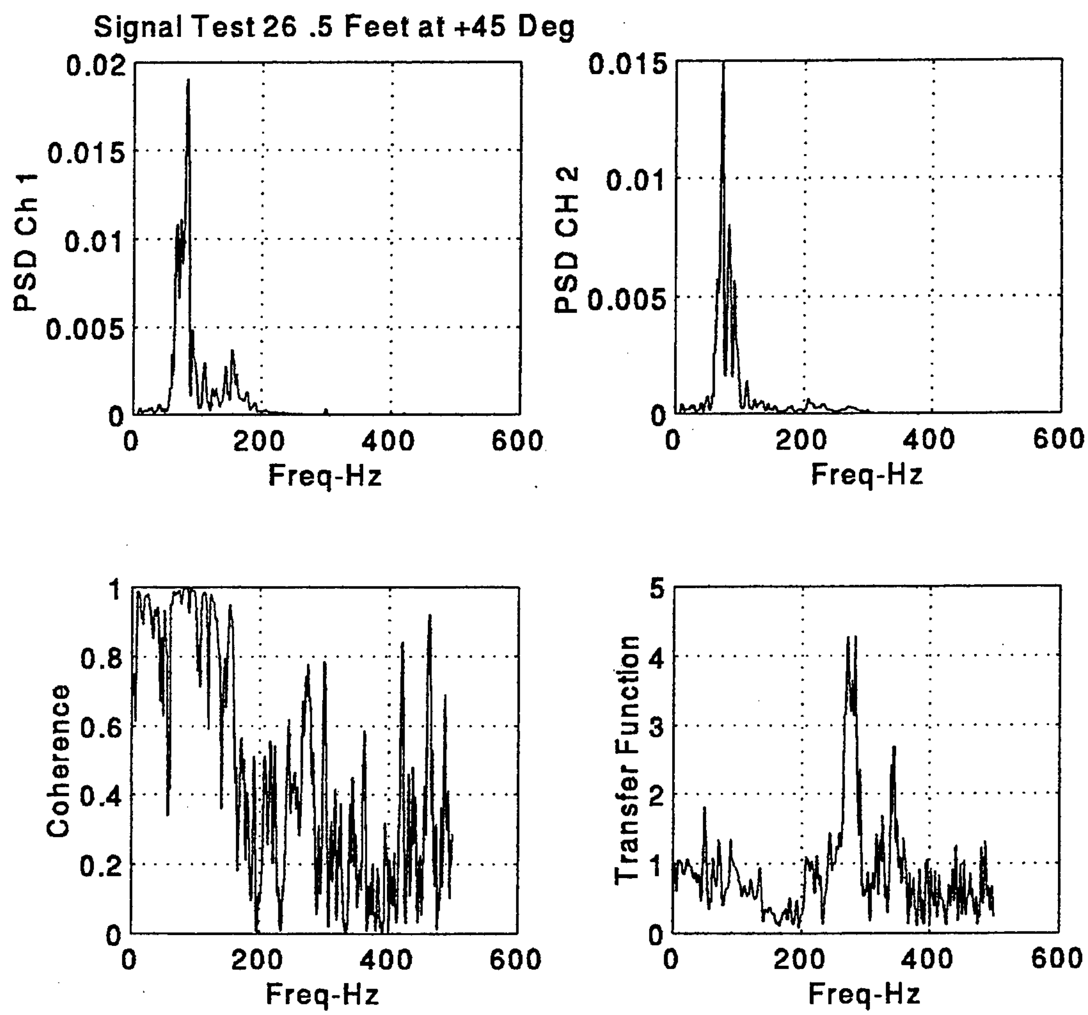

Raw Data
Test 39 Ch 1
Test 39 Ch 2
Test 40 Ch 1
Test 40 Ch 2
Time-Sec
Time-Sec
Time-Sec
Time-Sec


Raw Data
Test 41 Ch 1
Test 41 Ch 2
Test 41 Ch 1
Test 41 Ch 2
Time-Sec
Time-Sec
Time-Sec
Time-Sec

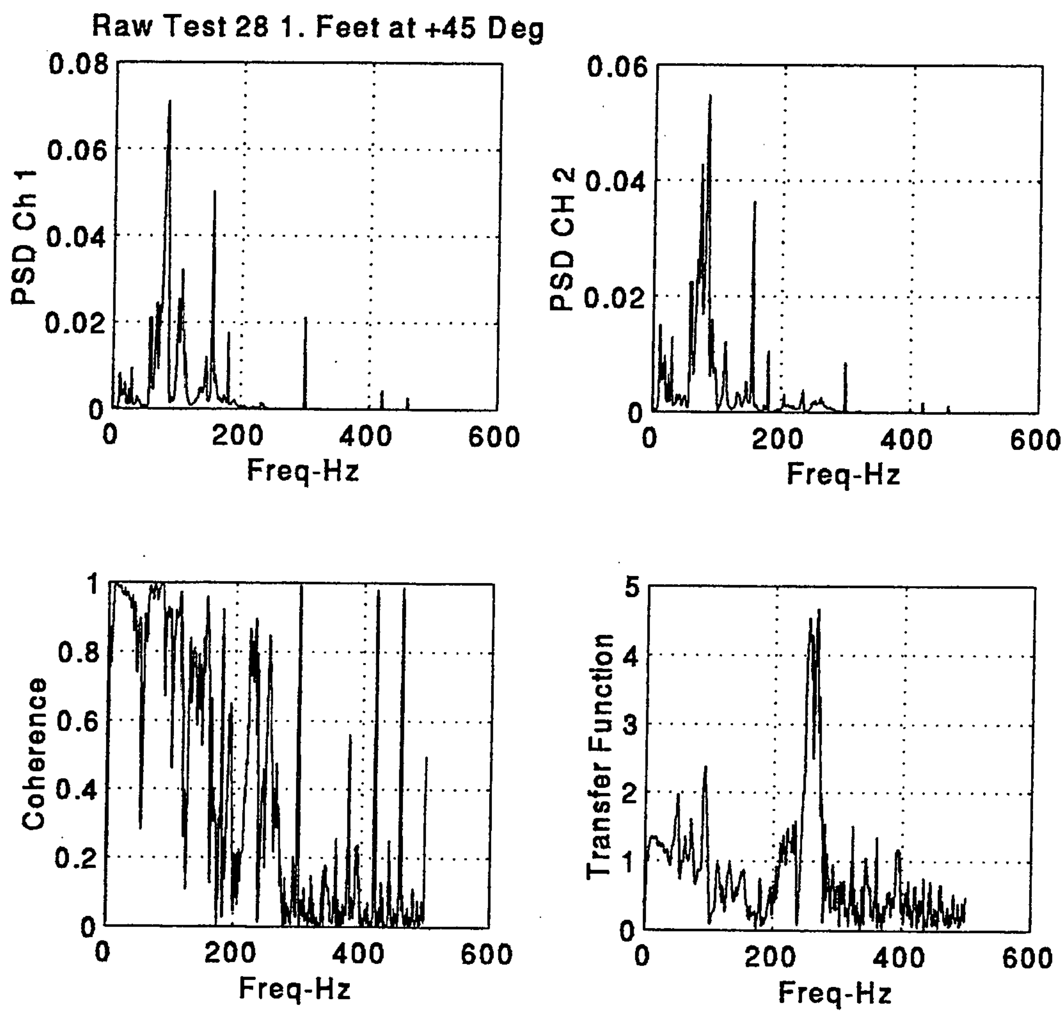
Raw Test 26 .5 Feet at +45 Deg

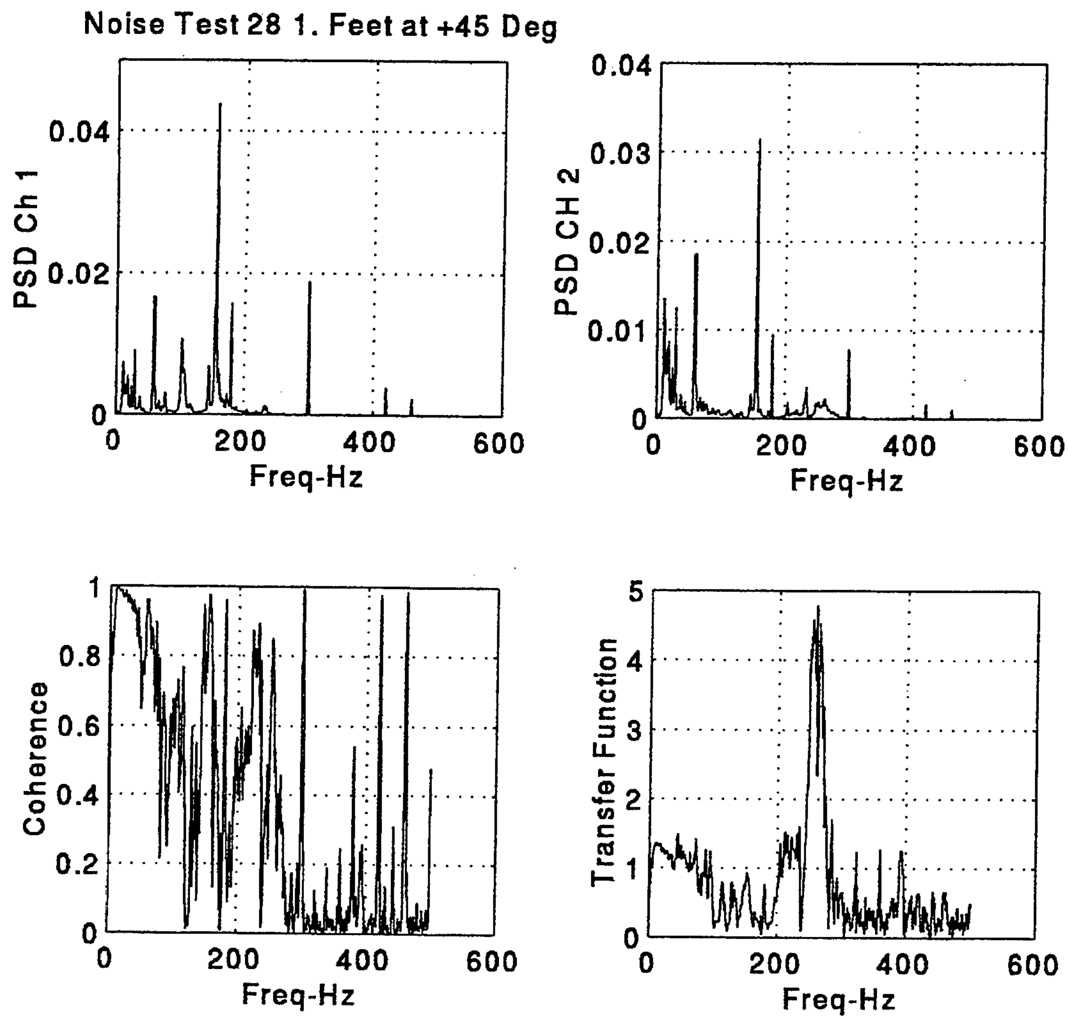


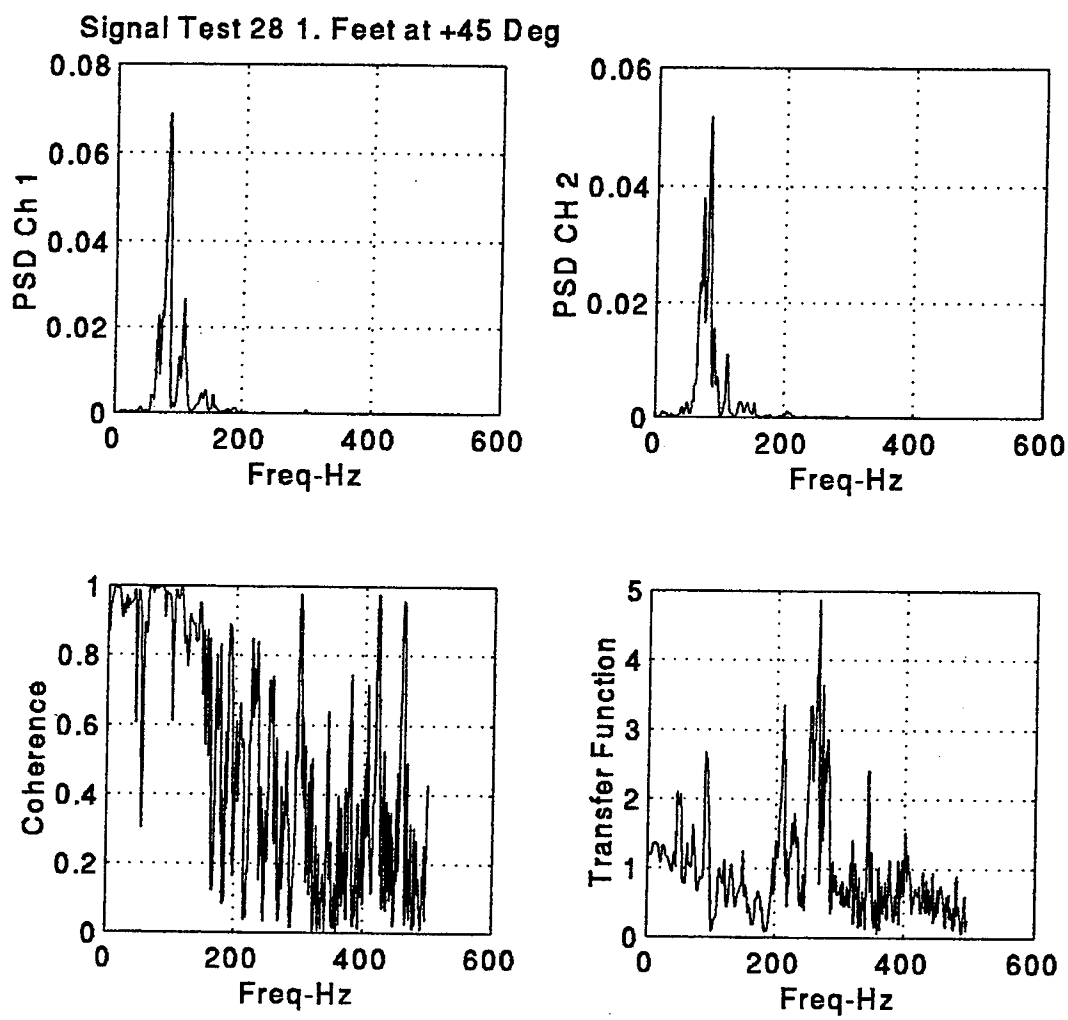

Noise Test 26 .5 Feet at +45 Deg
0.08
0.06
0.04
0.02
0
PSD Ch 1
0
200
400
600
Freq-Hz
x 10^-3
8
6
4
2
0
PSD CH 2
0
200
400
600
Freq-Hz
1
0.8
0.6
0.4
0.2
0
Coherence
0
200
400
600
Freq-Hz
4
3
2
1
0
Transfer Function
0
200
400
600
Freq-Hz

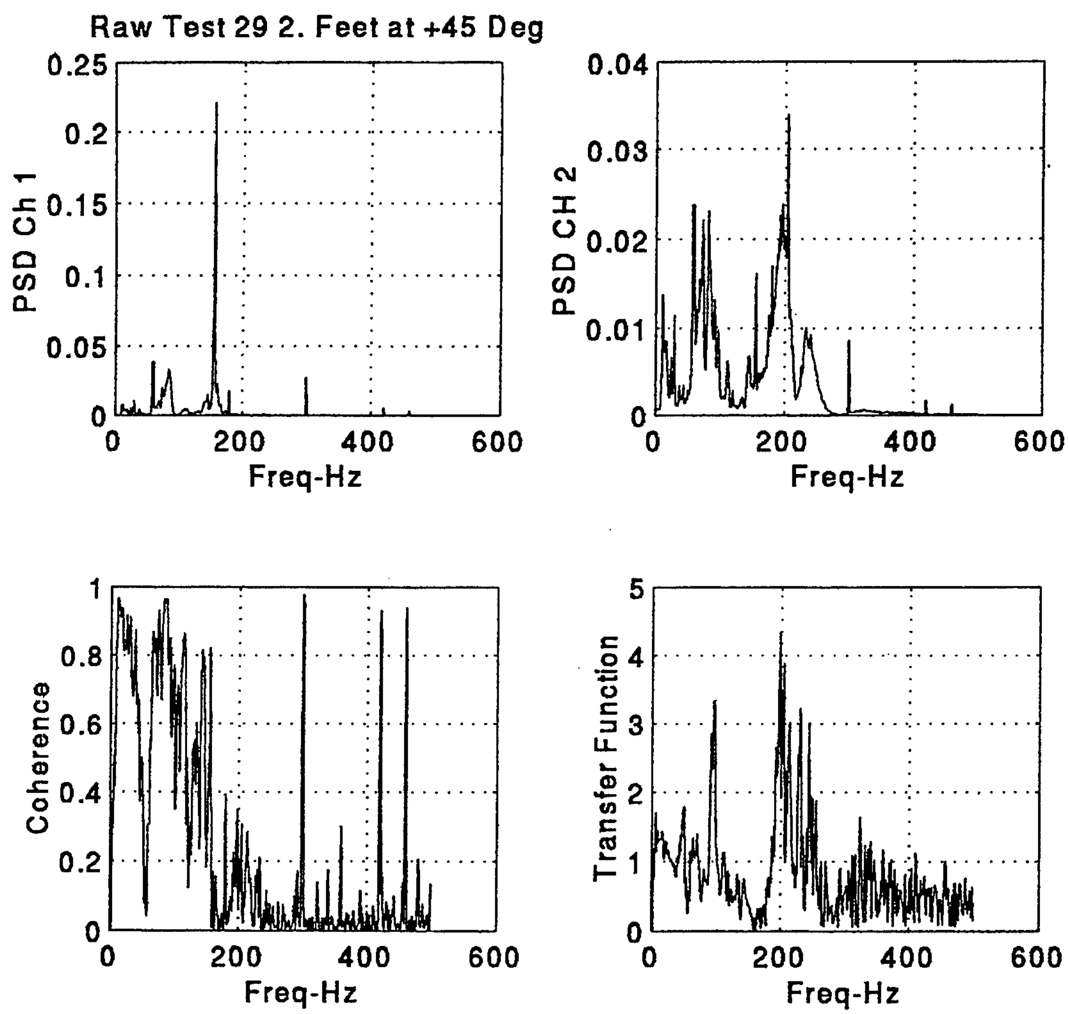


Signal Test 26 .5 Feet at +45 Deg
PSD Ch 1
0.02
0.015
0.01
0.005
0
200
400
600
Freq-Hz
PSD CH 2
0.015
0.01
0.005
Coherence
1
0.8
0.6
0.4
0.2
Transfer Function
5
4
3
2
1

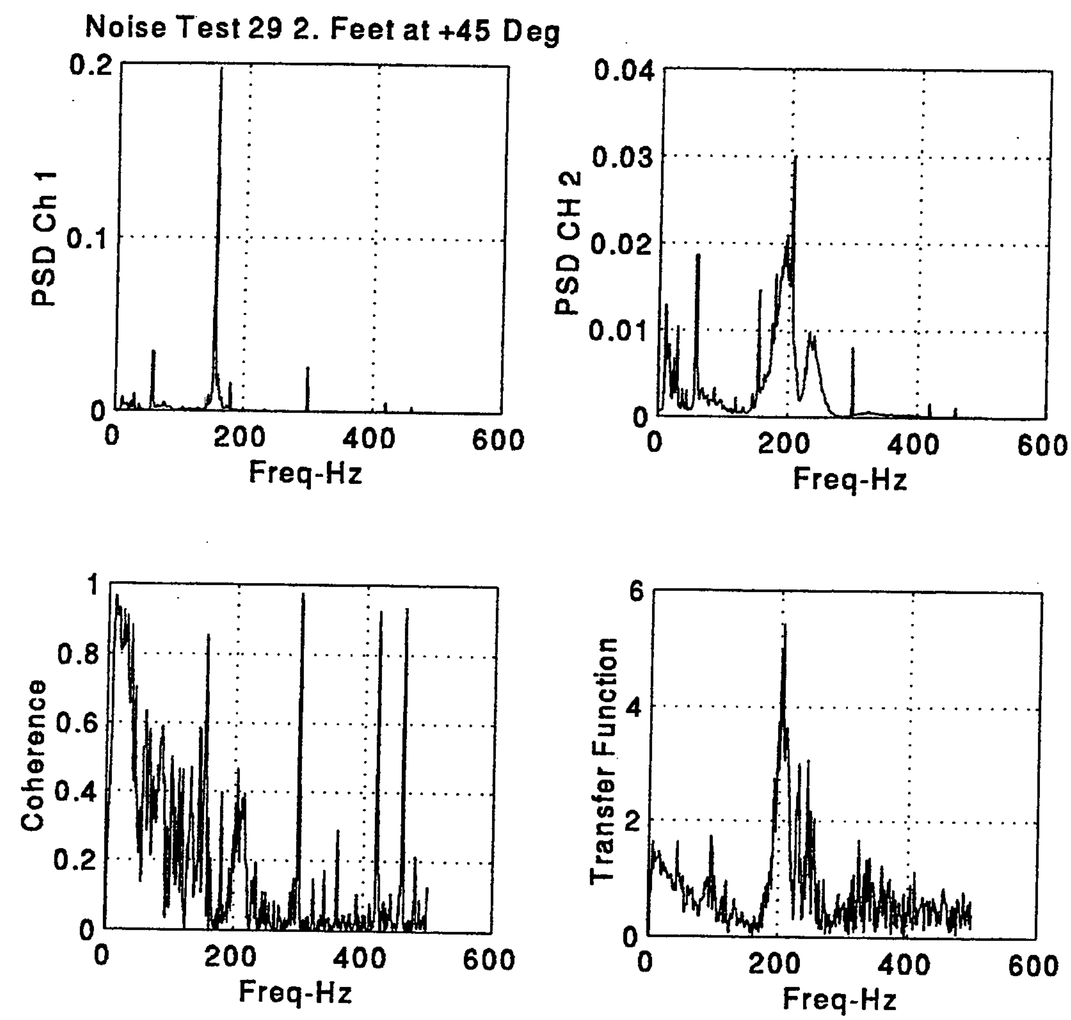

Raw Test 28 1. Feet at +45 Deg
PSD Ch 1
PSD CH 2
Coherence
Transfer Function
Freq-Hz
0
200
400
600
0.08
0.06
0.04
0.02
1
0.8
0.6
0.4
0.2
5
4
3
2

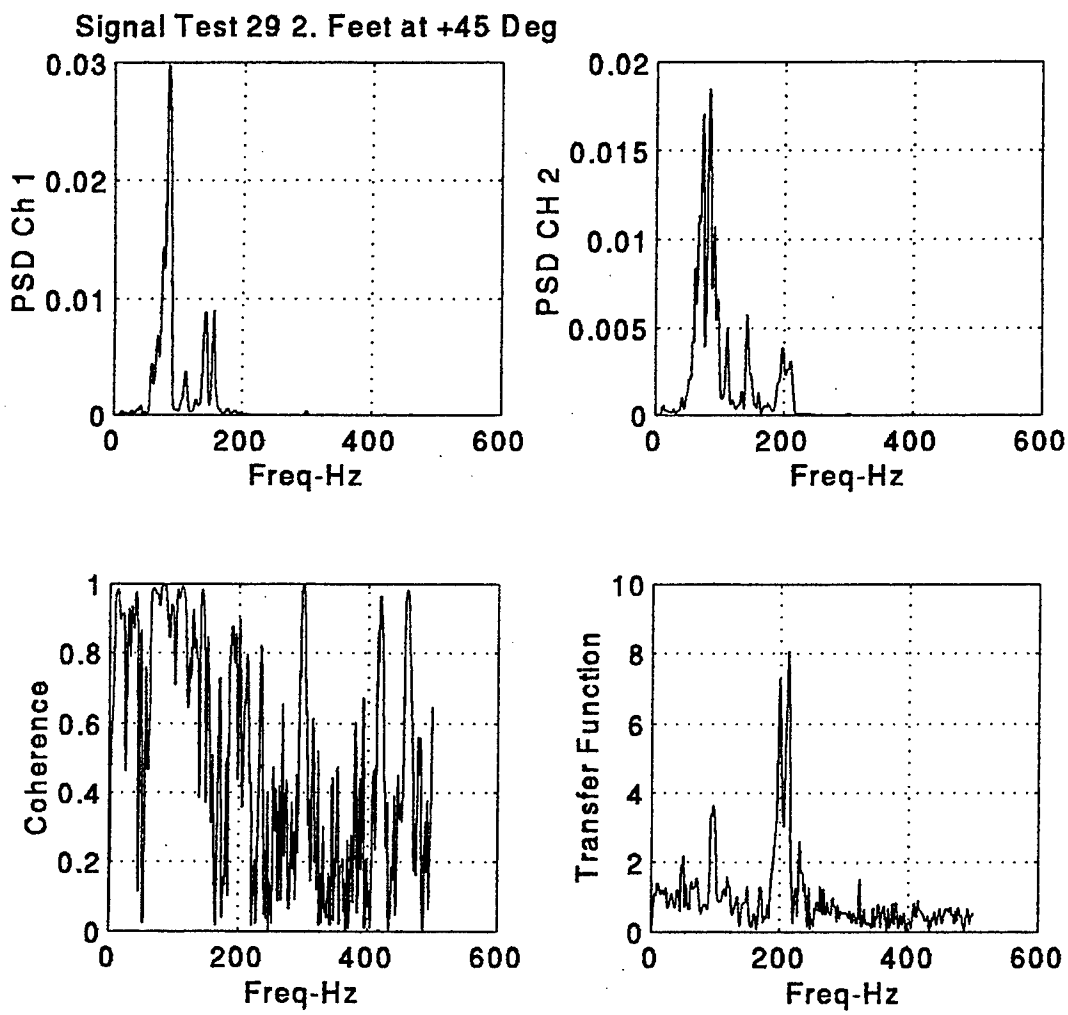

Noise Test 28 1. Feet at +45 Deg
PSD Ch 1
0.04
0.02
0
0
200
400
600
Freq-Hz
PSD CH 2
0.04
0.03
0.02
0.01
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
5
4
3
2
1
0
0
200
400
600
Freq-Hz


Signal Test 28 1. Feet at +45 Deg
PSD Ch 1
0.08
0.06
0.04
0.02
0
PSD CH 2
0.06
0.04
0.02
0
Coherence
1
0.8
0.6
0.4
0.2
0
Transfer Function
5
4
3
2
1
0
0
200
400
600
Freq-Hz



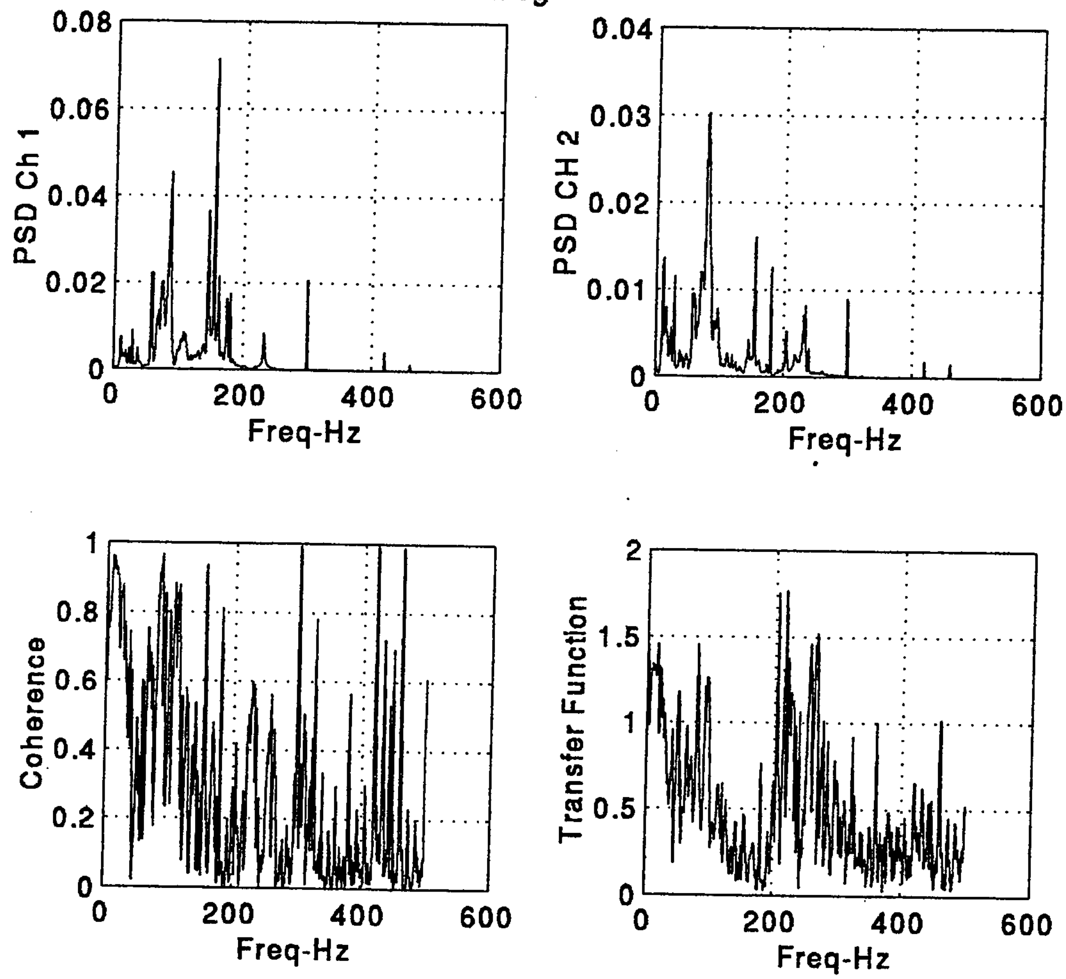
Raw Test 29 2. Feet at +45 Deg
PSD Ch 1
0.25
0.2
0.15
0.1
0.05
0
0
200
400
600
Freq-Hz
PSD CH 2
0.04
0.03
0.02
0.01
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
5
4
3
2
1
0
0
200
400
600
Freq-Hz

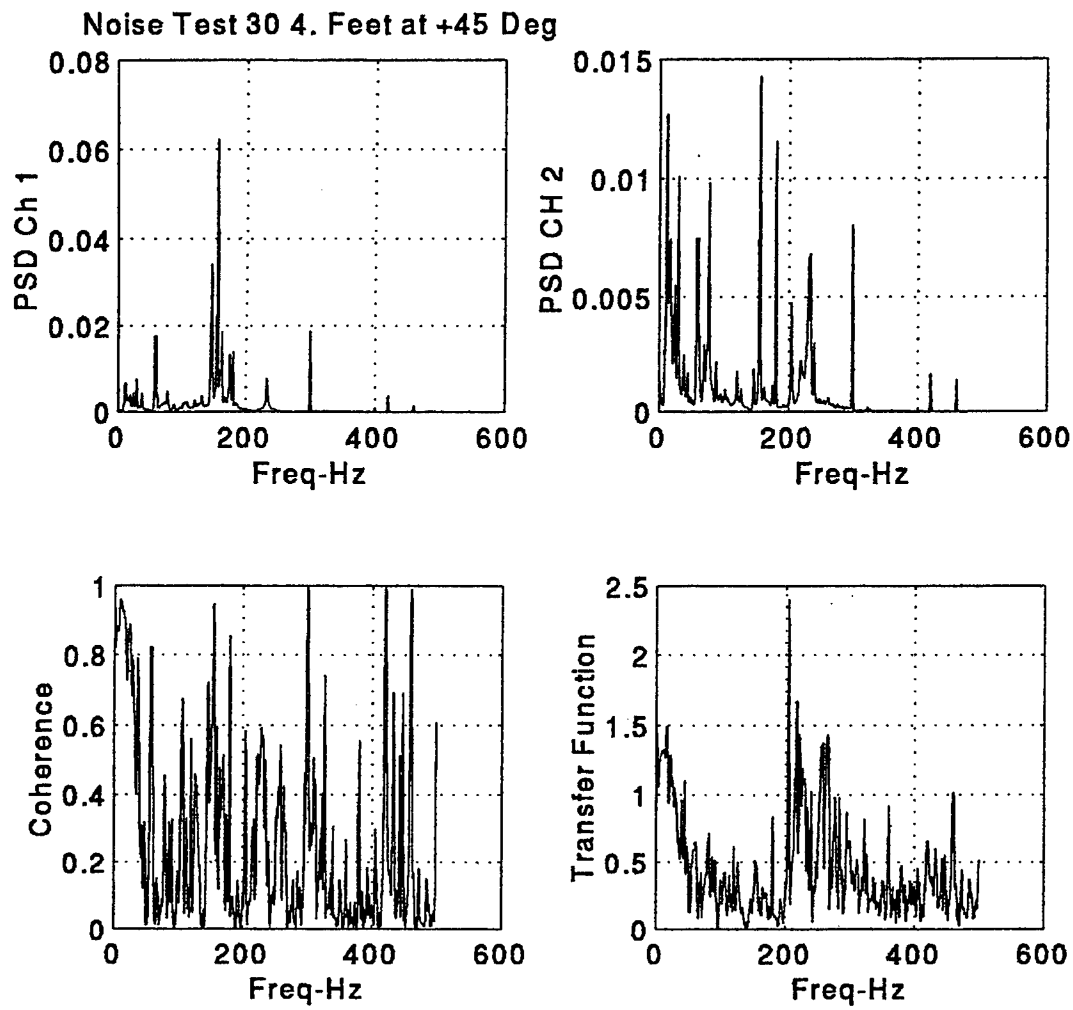

Noise Test 29 2. Feet at +45 Deg
PSD Ch 1
0.2
0.1
0
0
200
400
600
Freq-Hz
PSD CH 2
0.04
0.03
0.02
0.01
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
6
4
2
0
0
200
400
600
Freq-Hz



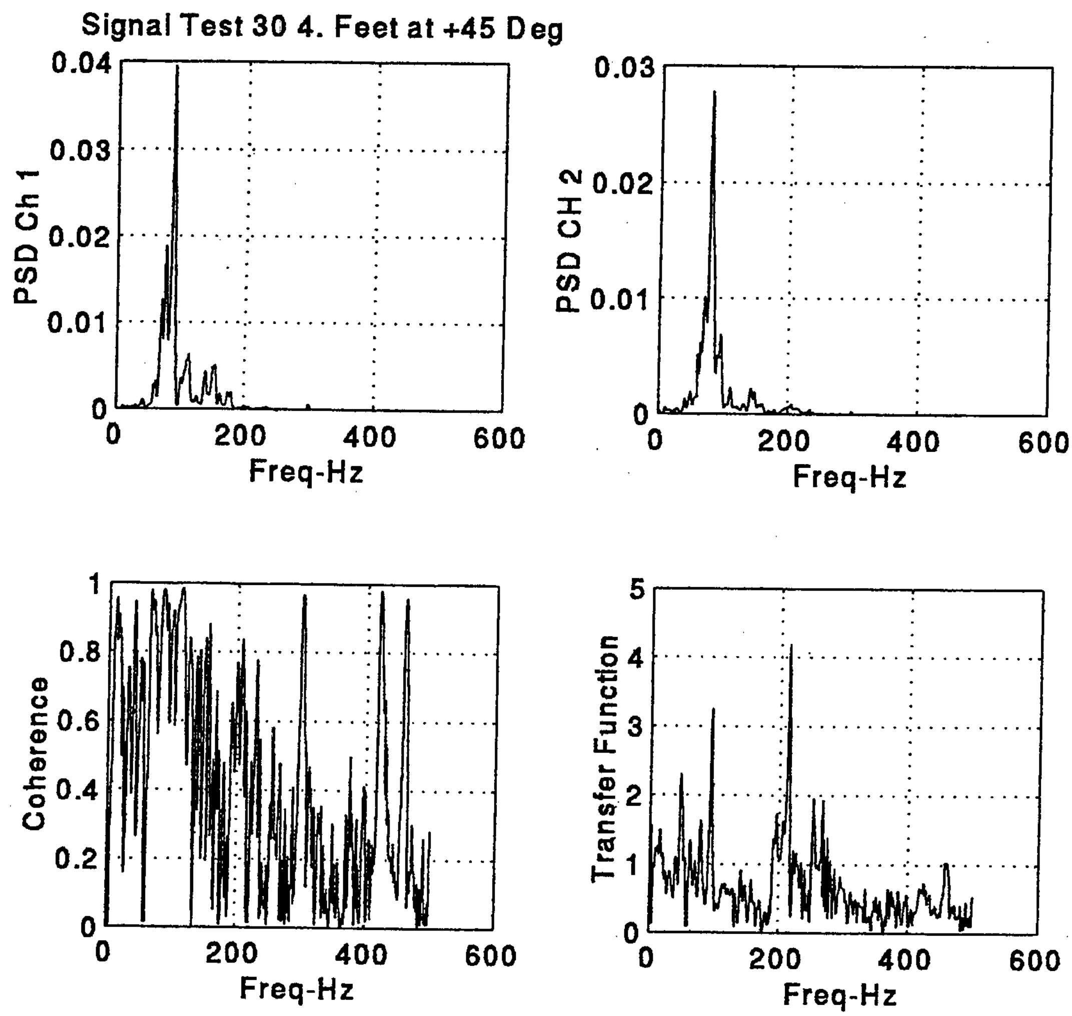
Signal Test 29 2. Feet at +45 Deg
PSD Ch 1
PSD CH 2
Coherence
Transfer Function
Freq-Hz

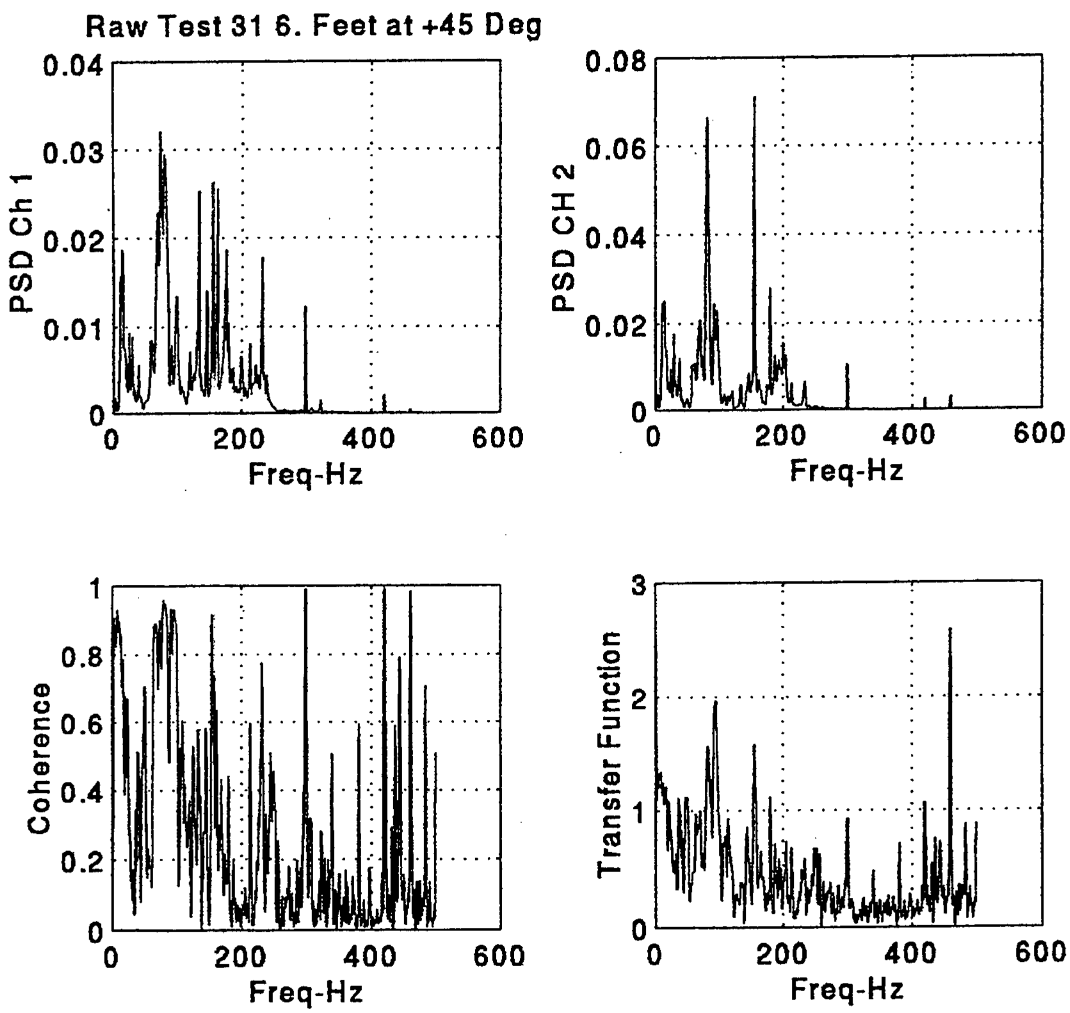
Raw Test 30 4. Feet at +45 Deg





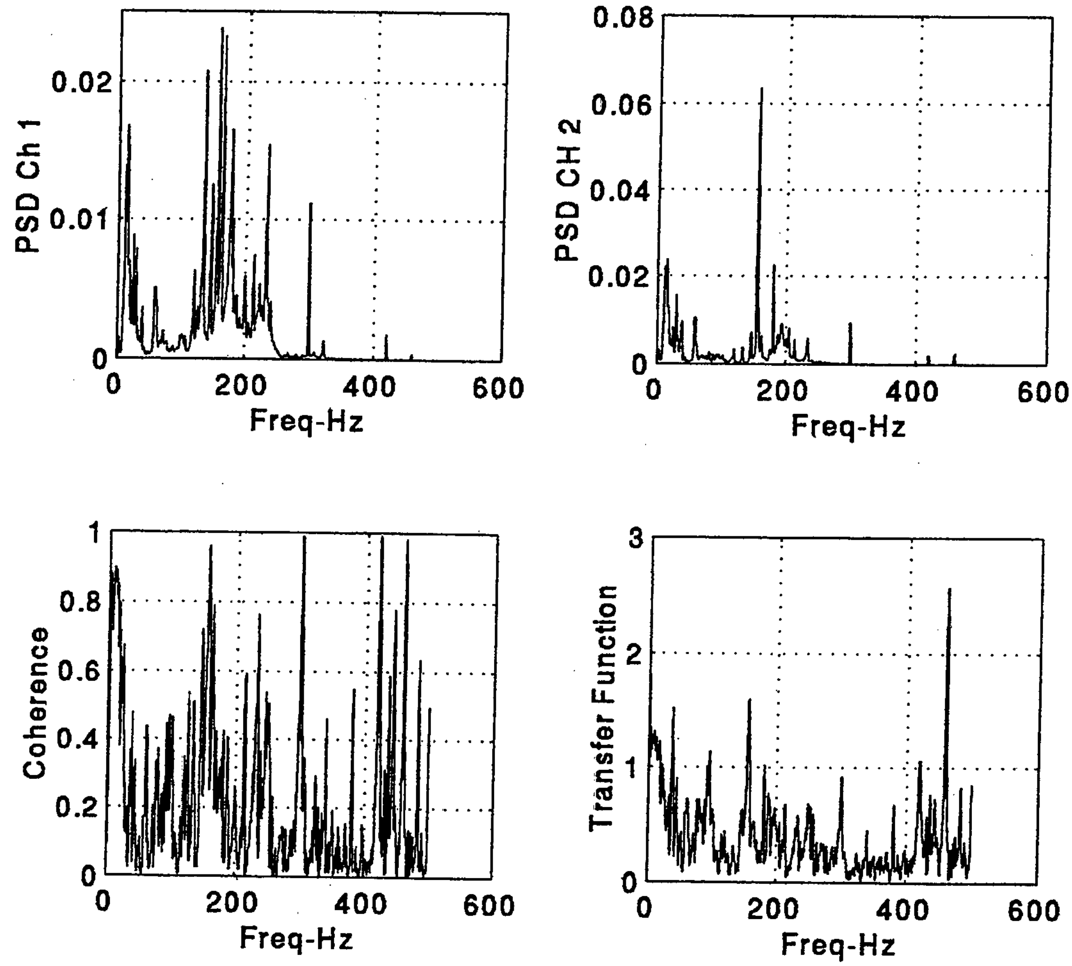
Noise Test 30 4. Feet at +45 Deg
PSD Ch 1
0.08
0.06
0.04
0.02
0
0
200
400
600
Freq-Hz
PSD CH 2
0.015
0.01
0.005
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
2.5
2
1.5
1
0.5
0
0
200
400
600
Freq-Hz

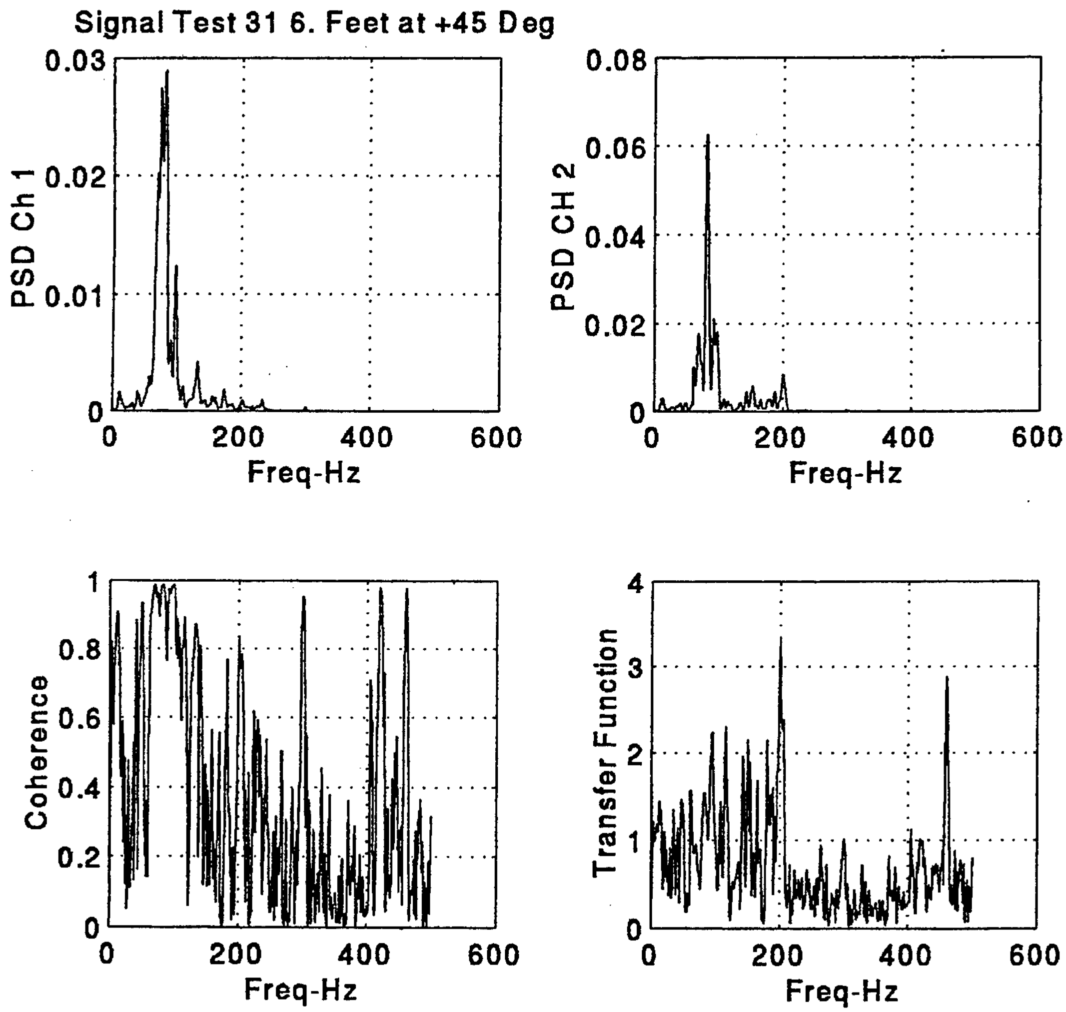

Signal Test 30 4. Feet at +45 Deg
PSD Ch 1
0.04
0.03
0.02
0.01
0
0
200
400
600
Freq-Hz
PSD CH 2
0.03
0.02
0.01
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
5
4
3
2
1
0
0
200
400
600
Freq-Hz

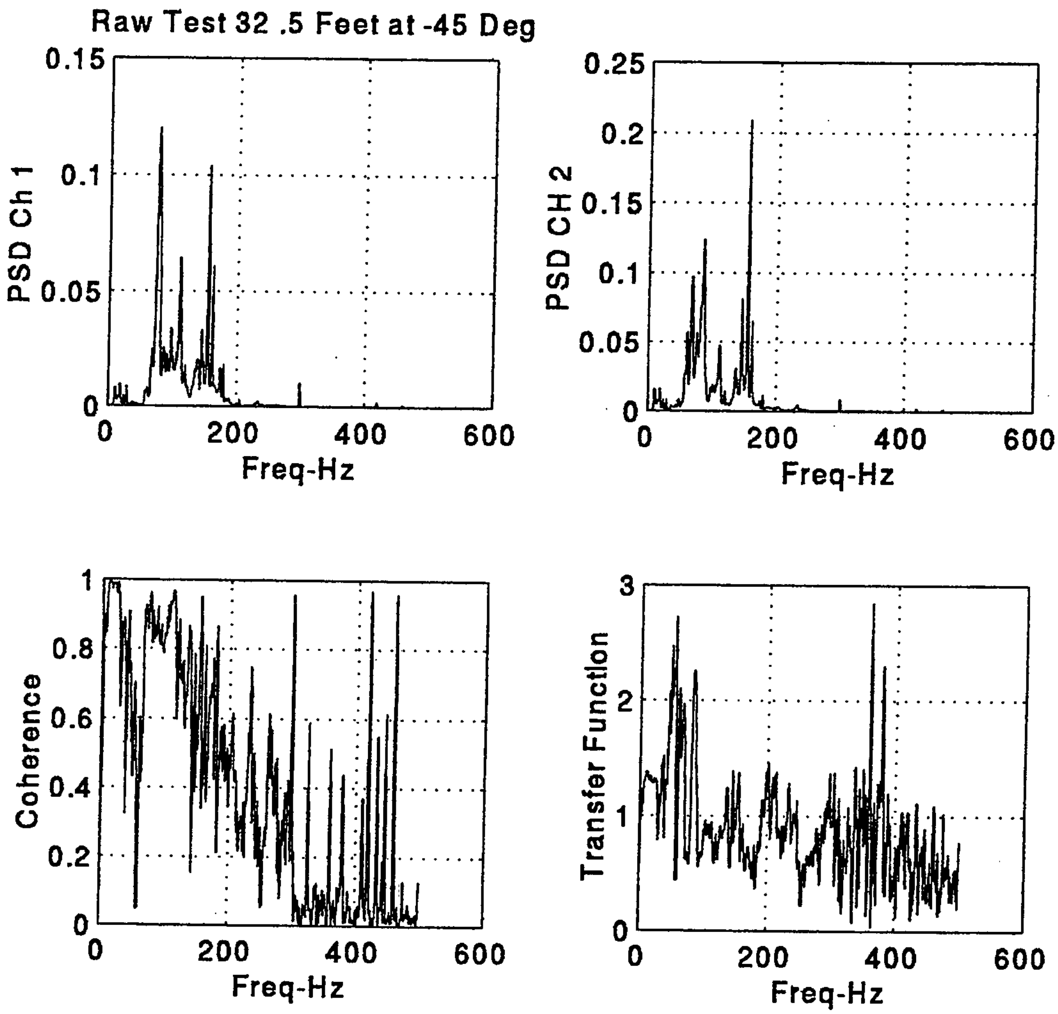


Raw Test 31 6. Feet at +45 Deg
PSD Ch 1
0.04
0.03
0.02
0.01
0
0
200
400
600
Freq-Hz
PSD CH 2
0.08
0.06
0.04
0.02
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
3
2
1
0
0
200
400
600
Freq-Hz

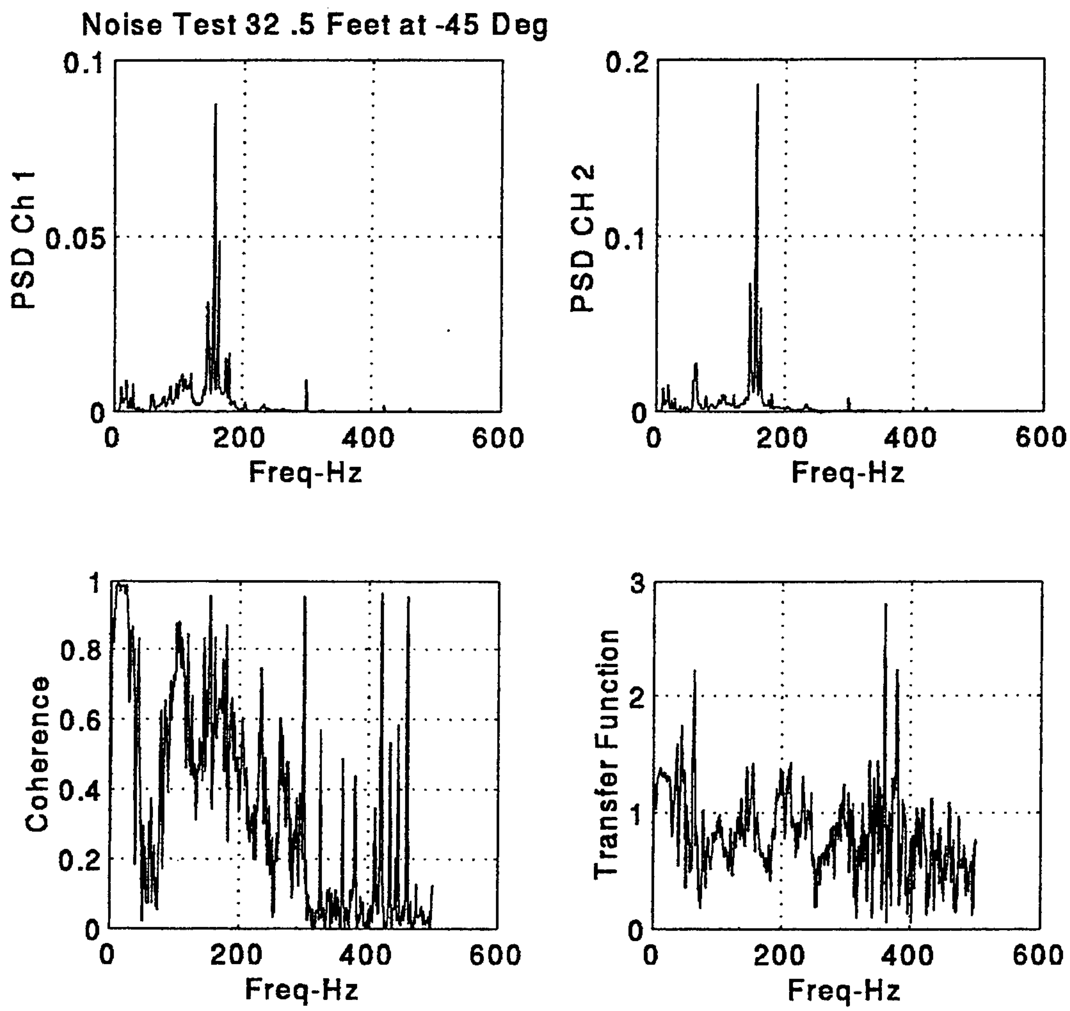
Noise Test 31 6. Feet at +45 Deg



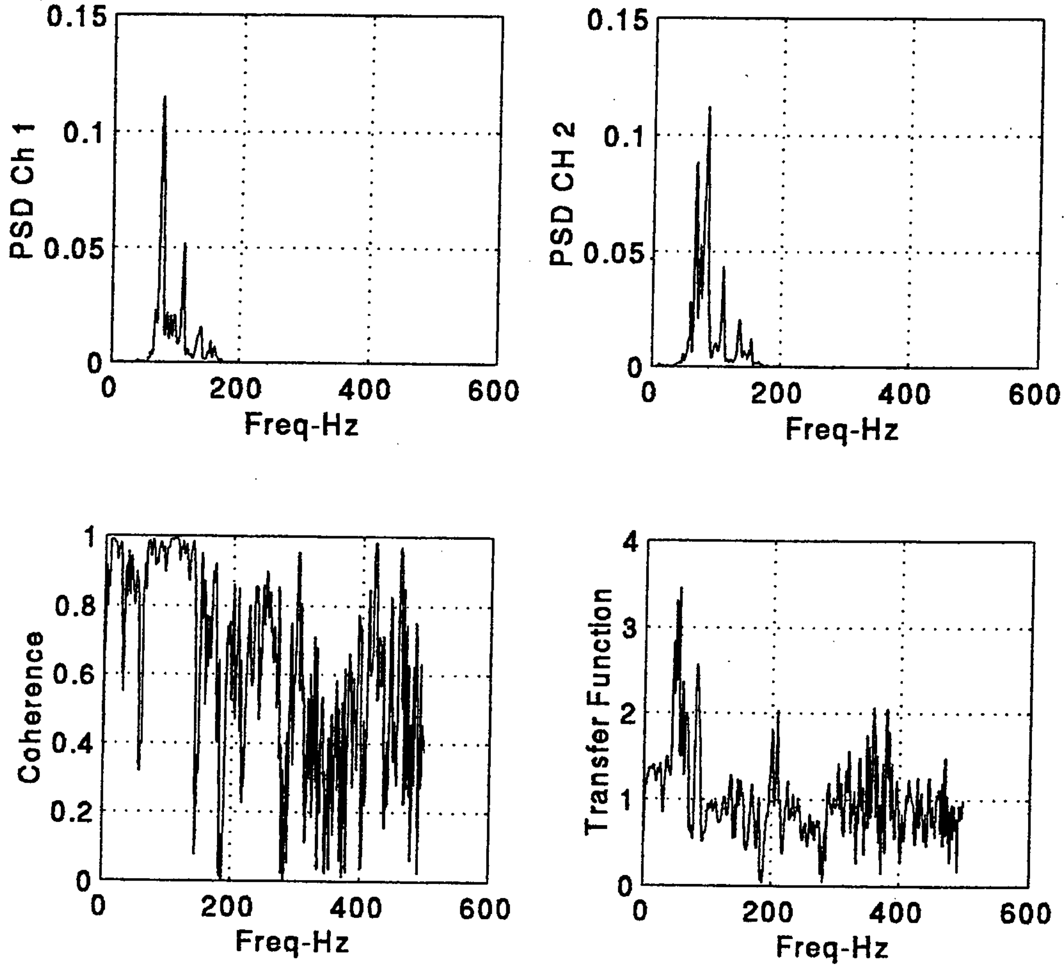

Signal Test 31 6. Feet at +45 Deg
PSD Ch 1
0.03
0.02
0.01
0
0
200
400
600
Freq-Hz
PSD CH 2
0.08
0.06
0.04
0.02
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
4
3
2
1
0
0
200
400
600
Freq-Hz

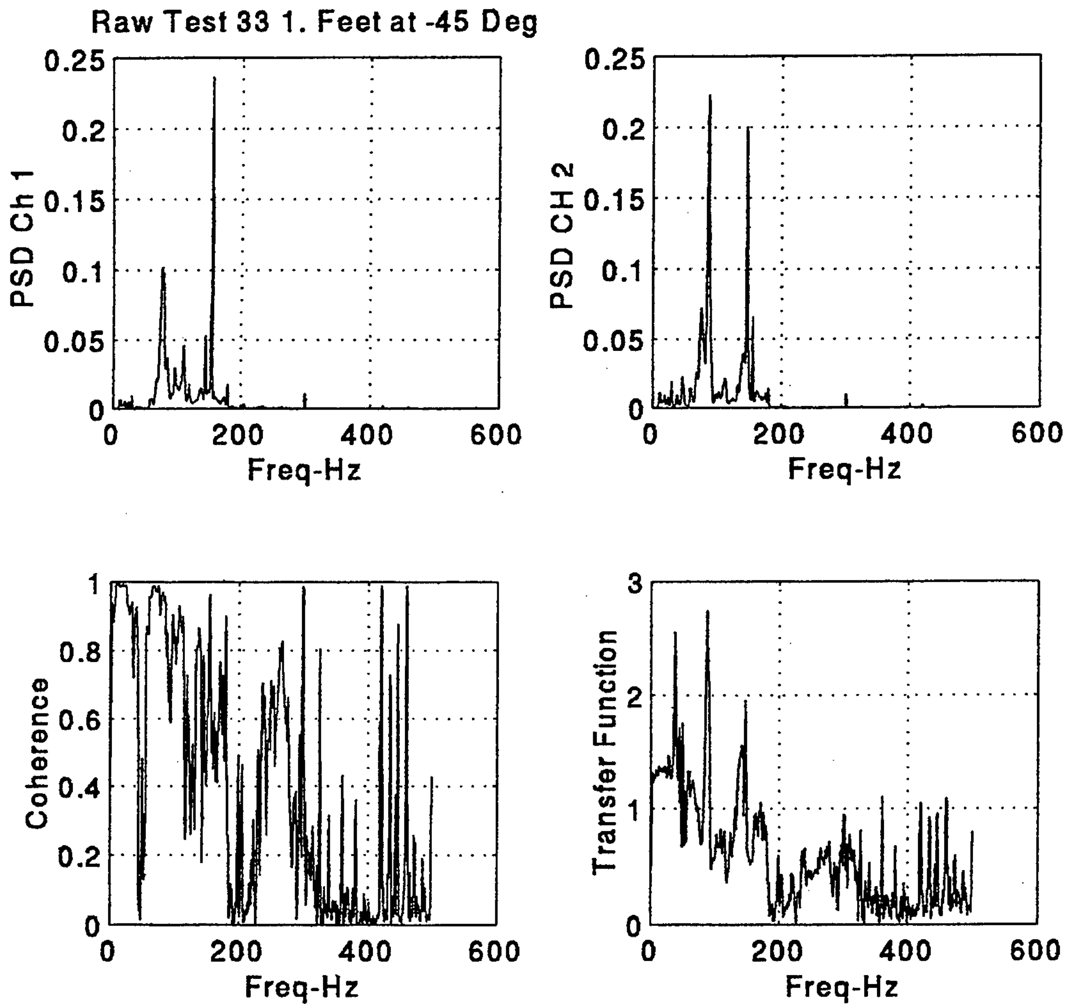


Raw Test 32 .5 Feet at -45 Deg
PSD Ch 1
0.15
0.1
0.05
0
0
200
400
600
Freq-Hz
PSD CH 2
0.25
0.2
0.15
0.1
0.05
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
3
2
1
0
0
200
400
600
Freq-Hz

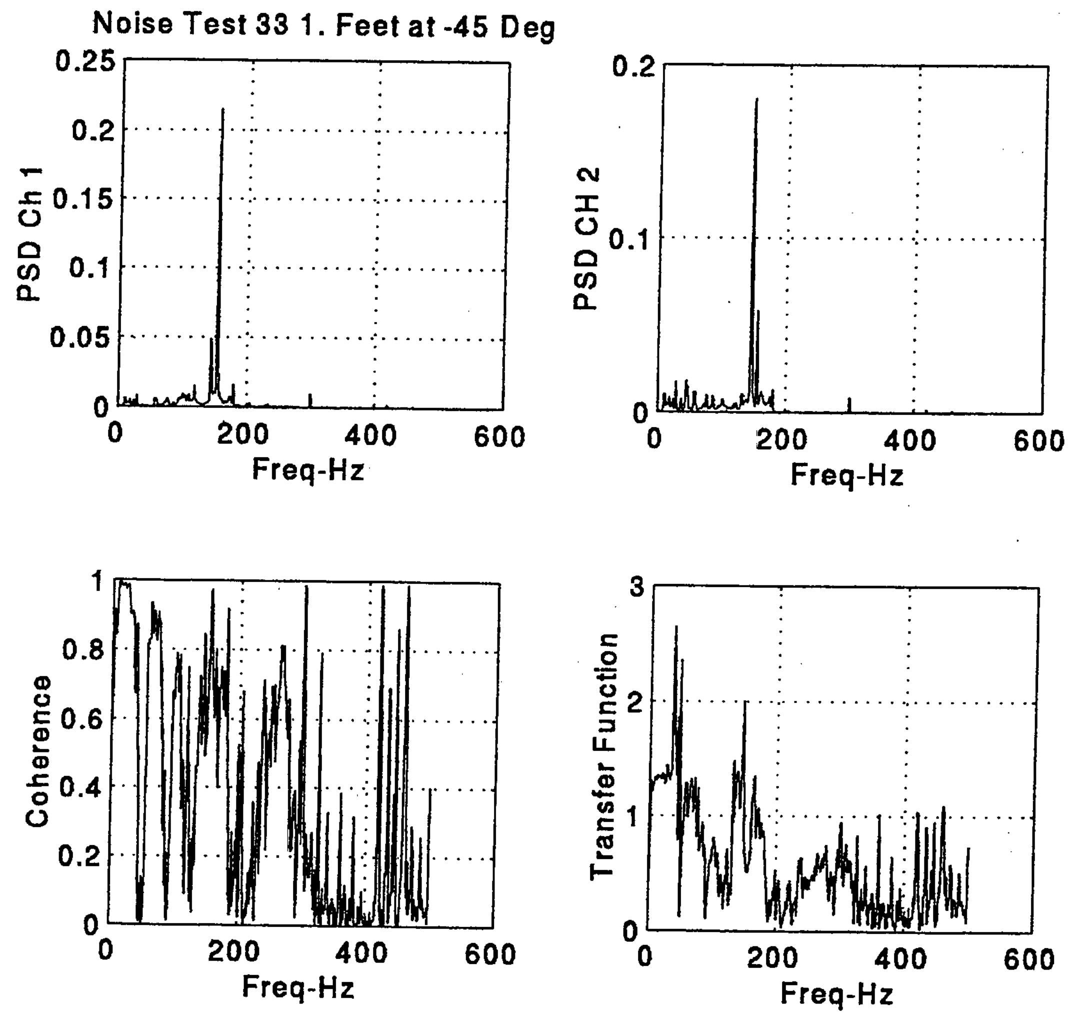

Noise Test 32 .5 Feet at -45 Deg
PSD Ch 1
0.1
0.05
0
0
200
400
600
Freq-Hz
PSD CH 2
0.2
0.1
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
3
2
1
0
0
200
400
600
Freq-Hz

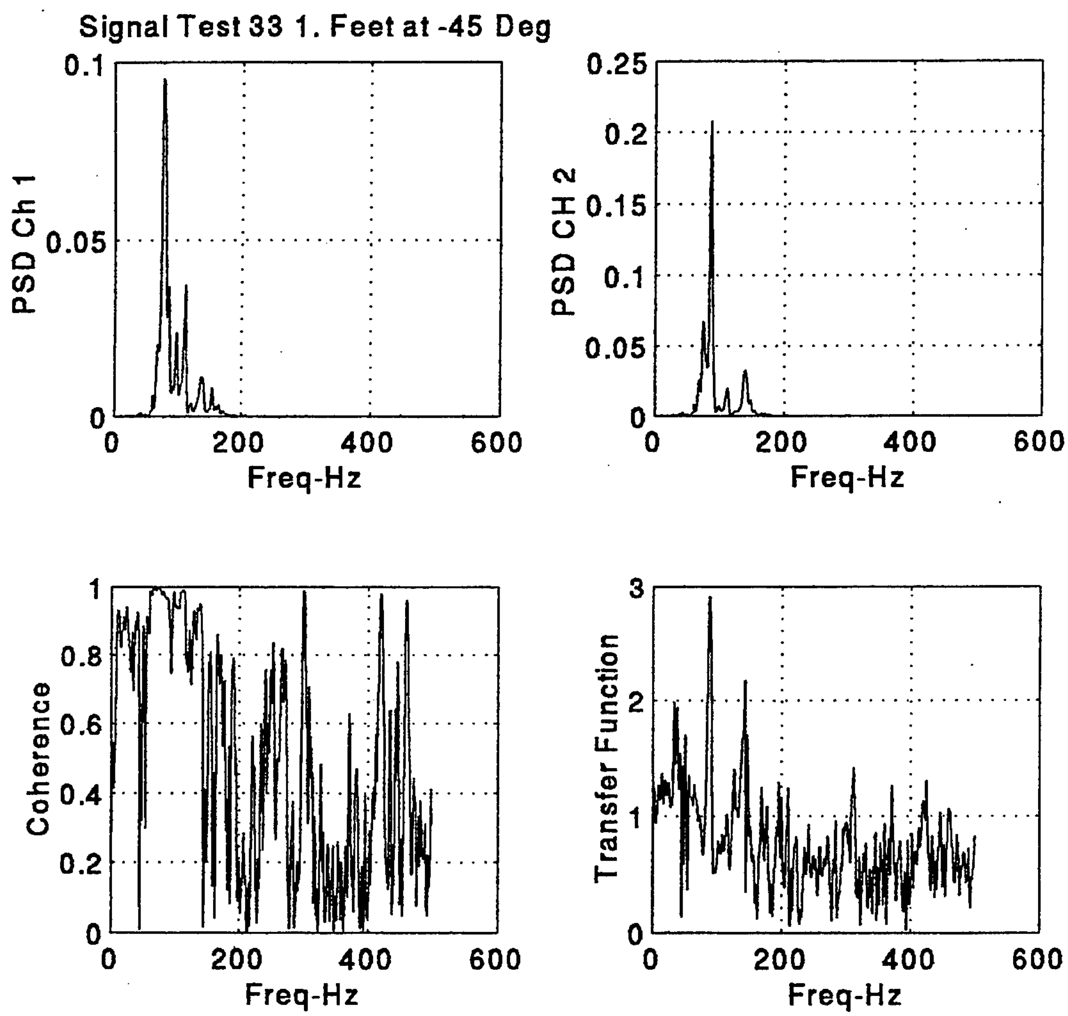
Signal Test 32 .5 Feet at -45 Deg

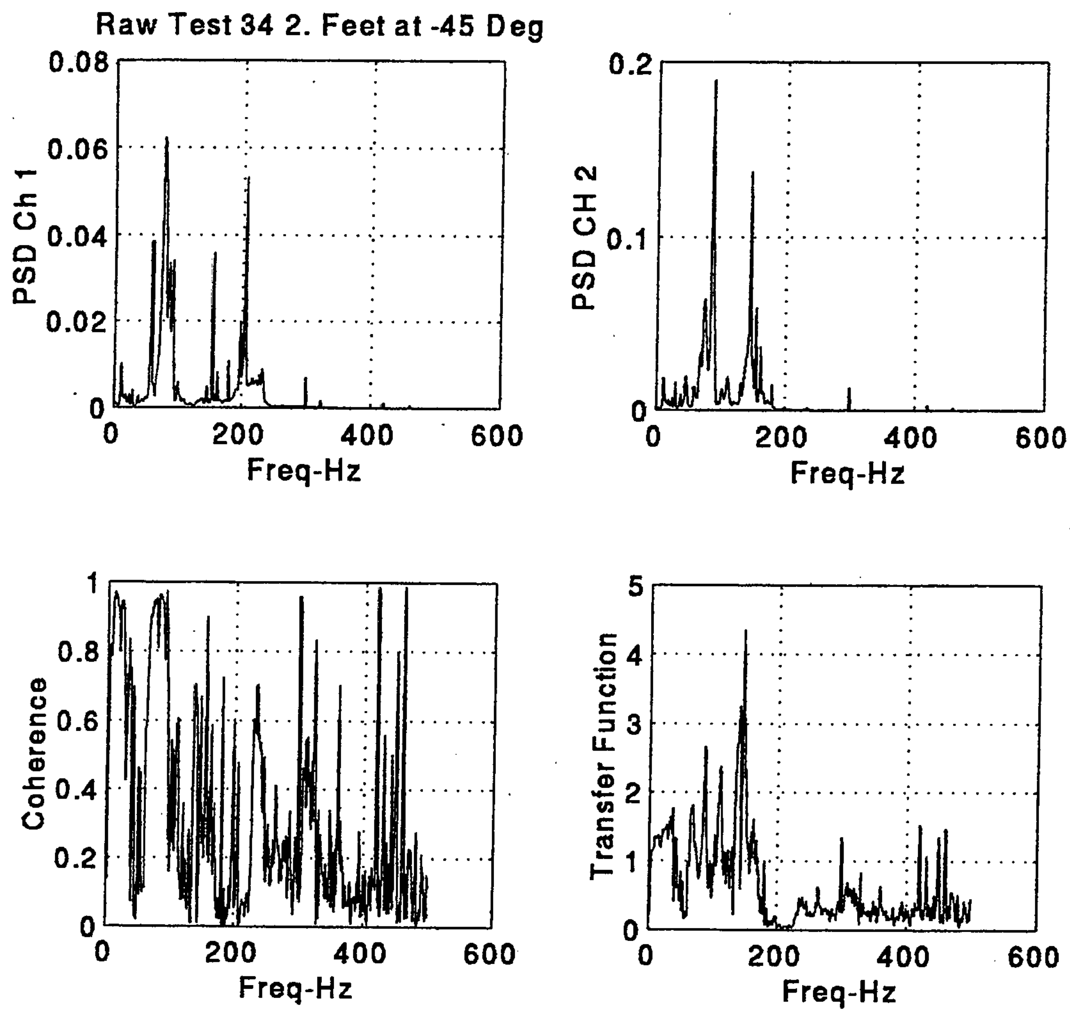




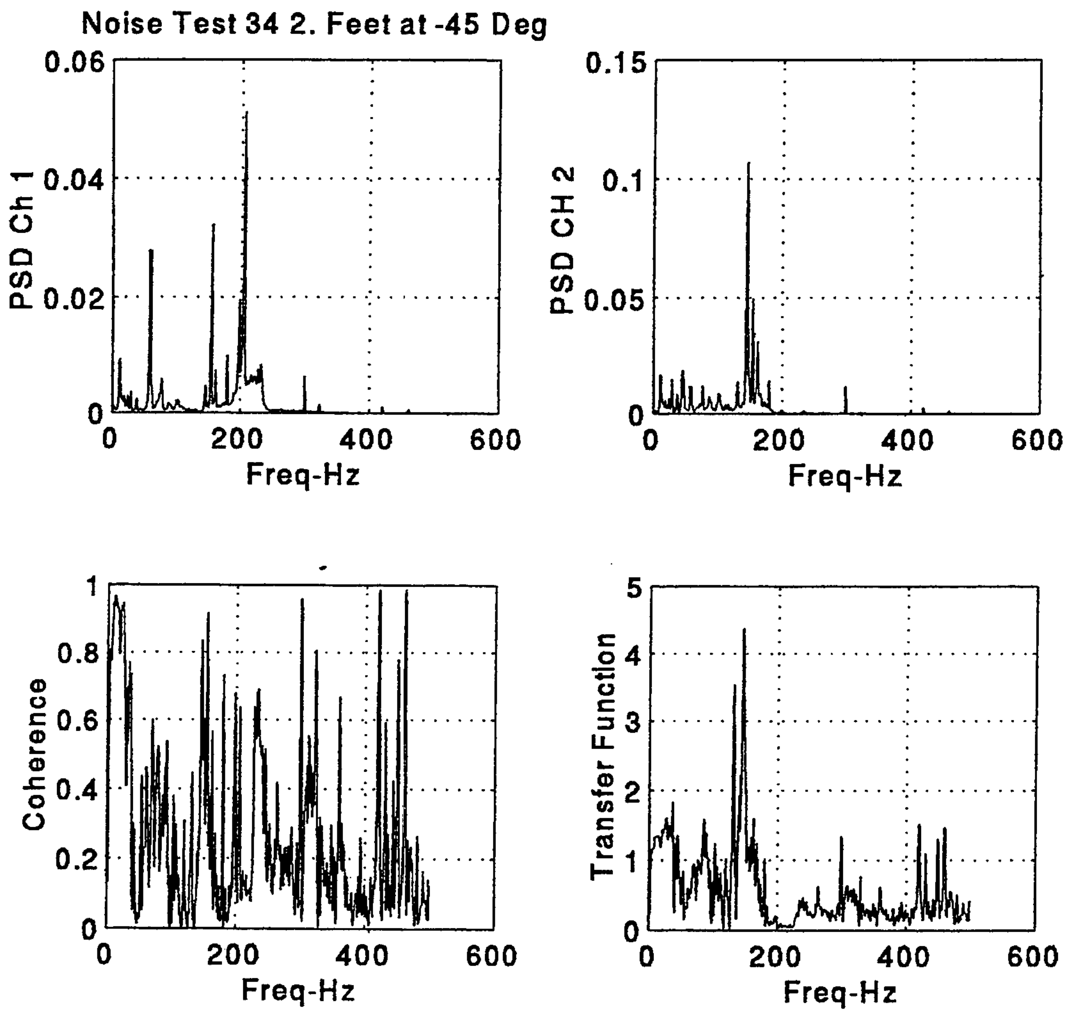
Raw Test 33 1. Feet at -45 Deg
PSD Ch 1
PSD CH 2
Coherence
Transfer Function
Freq-Hz

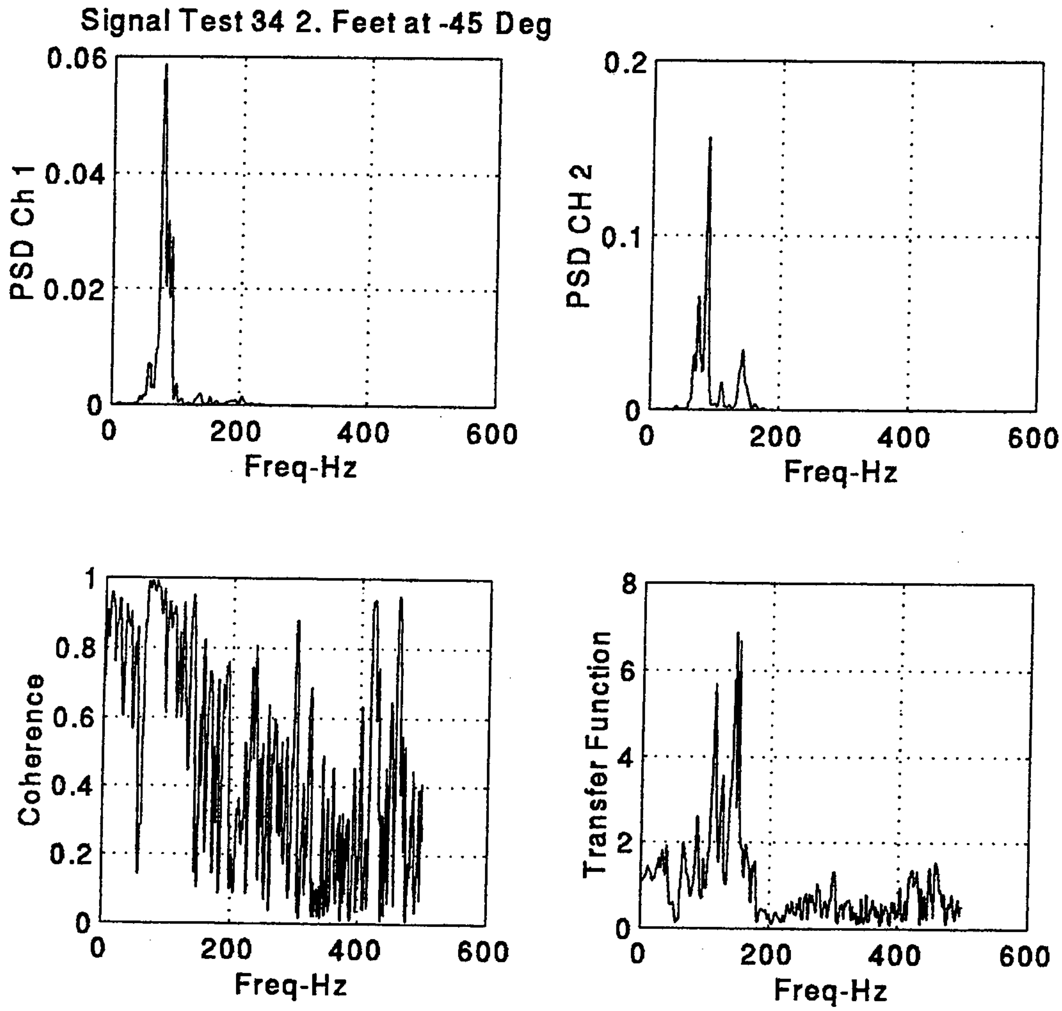

Noise Test 33 1. Feet at -45 Deg
PSD Ch 1
0.25
0.2
0.15
0.1
0.05
0
0
200
400
600
Freq-Hz
PSD CH 2
0.2
0.1
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
3
2
1
0
0
200
400
600
Freq-Hz


Signal Test 33 1. Feet at -45 Deg
PSD Ch 1
0.1
0.05
0
0
200
400
600
Freq-Hz
PSD CH 2
0.25
0.2
0.15
0.1
0.05
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
3
2
1
0
0
200
400
600
Freq-Hz

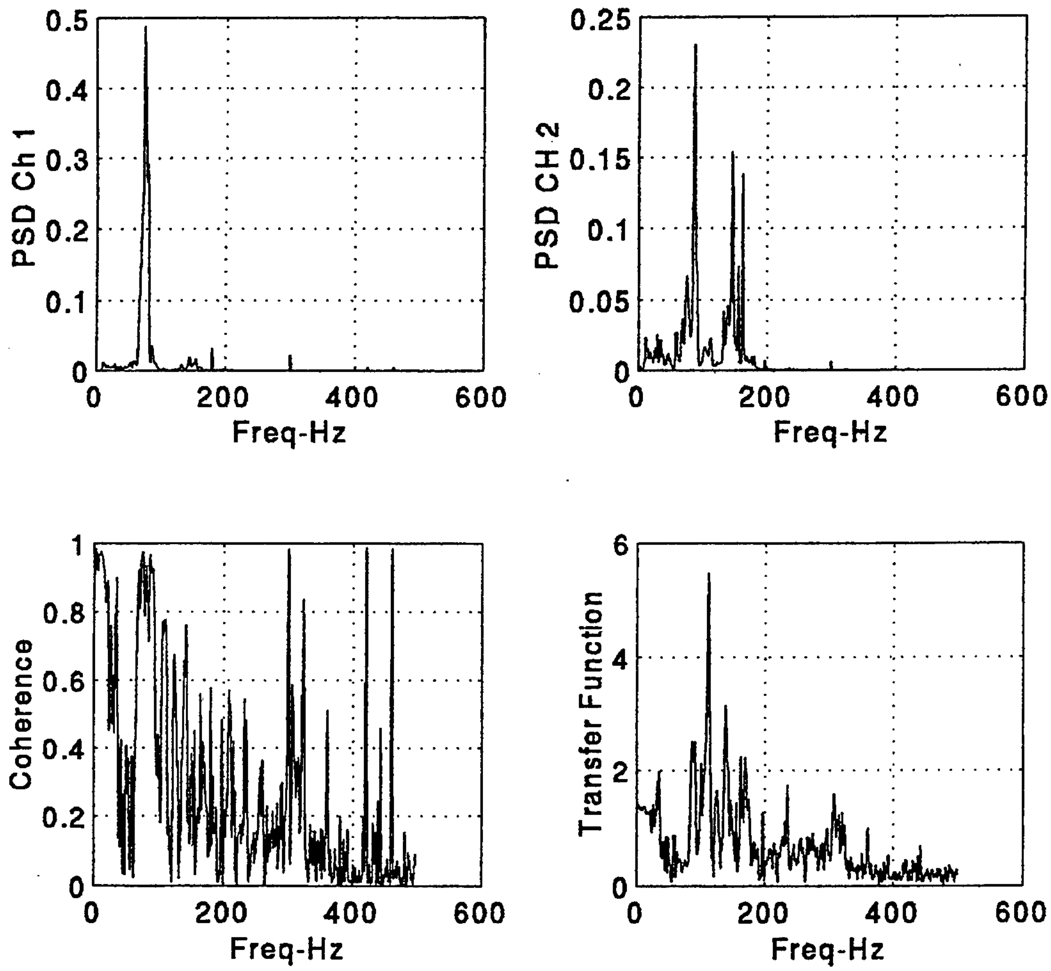


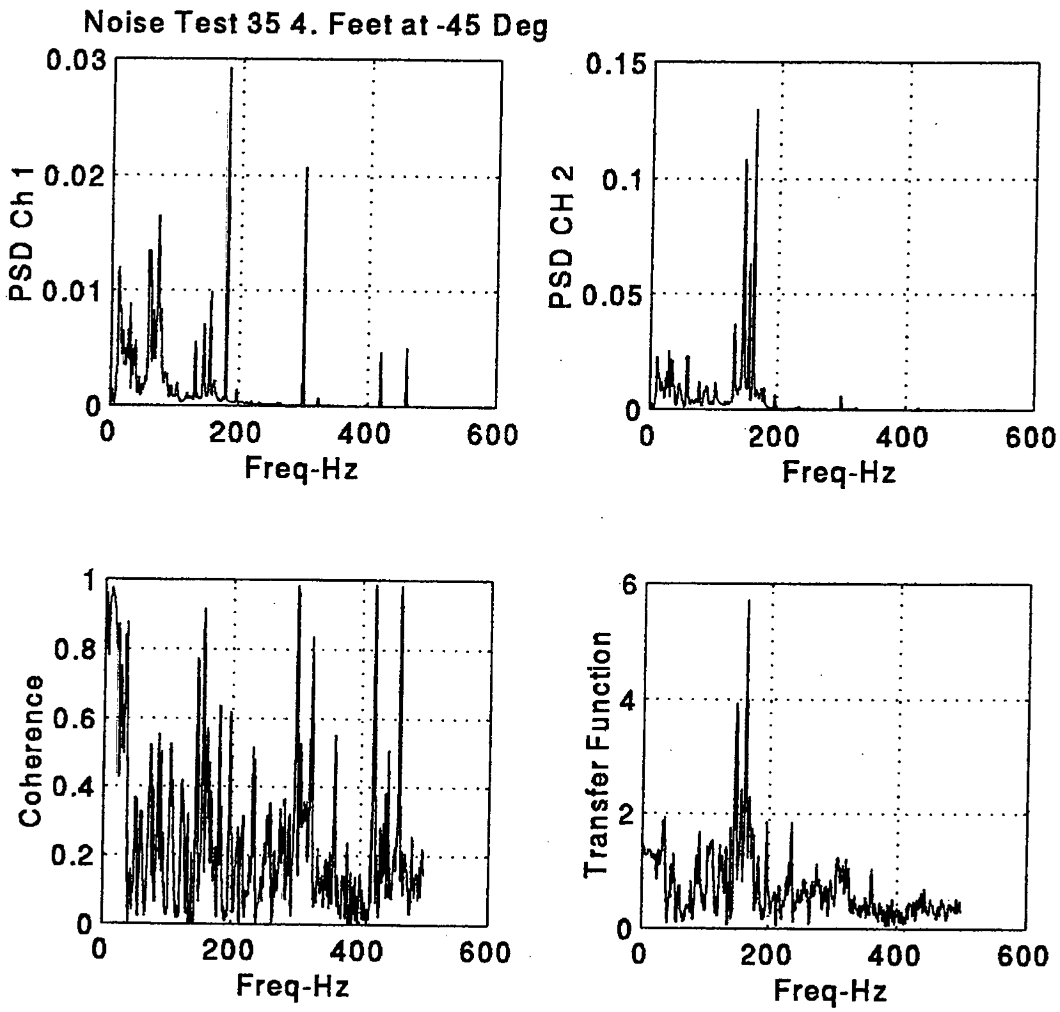
Raw Test 34 2. Feet at -45 Deg
PSD Ch 1
0.08
0.06
0.04
0.02
0
PSD CH 2
0.2
0.1
Coherence
1
0.8
0.6
0.4
0.2
Transfer Function
5
4
3
2
1
0
200
400
600
Freq-Hz

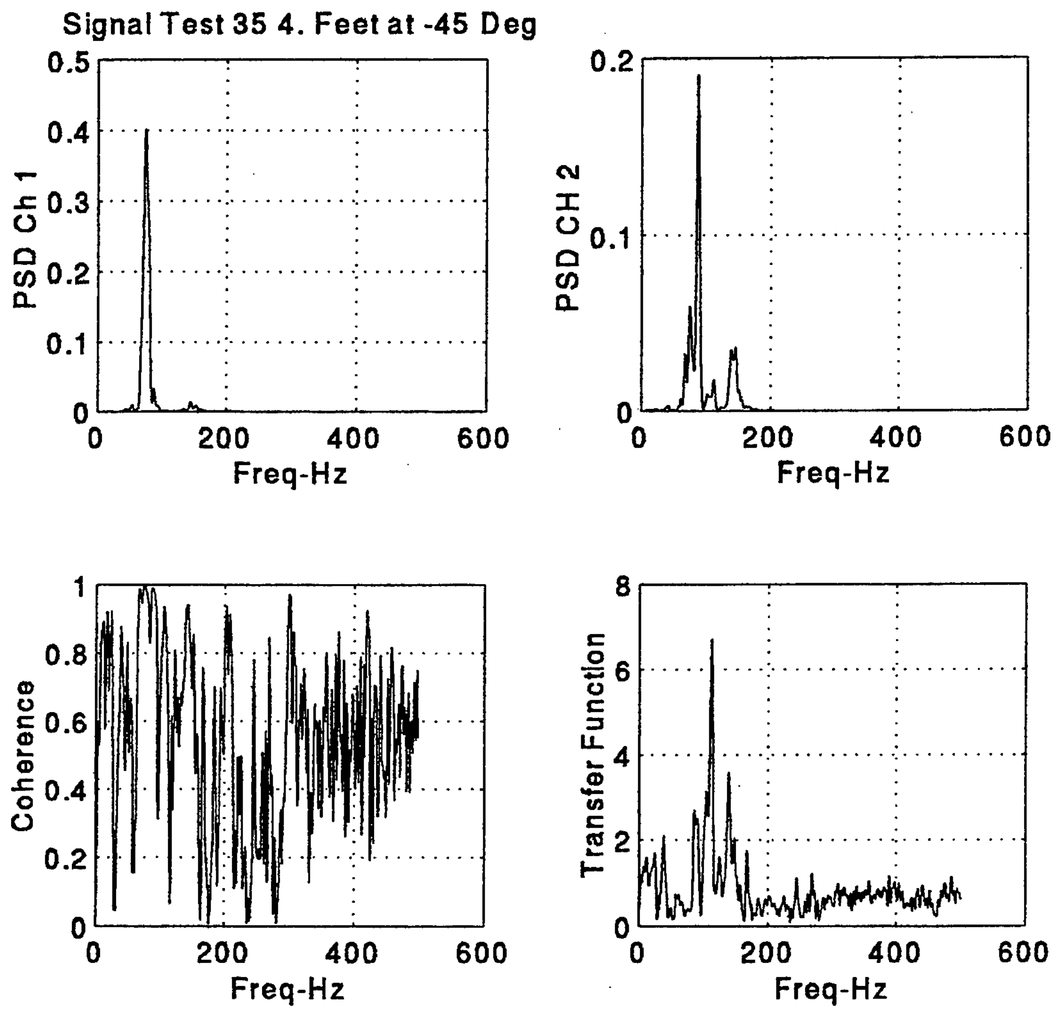

Noise Test 34 2. Feet at -45 Deg
PSD Ch 1
0.06
0.04
0.02
0
0
200
400
600
Freq-Hz
PSD CH 2
0.15
0.1
0.05
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
5
4
3
2
1
0
0
200
400
600
Freq-Hz

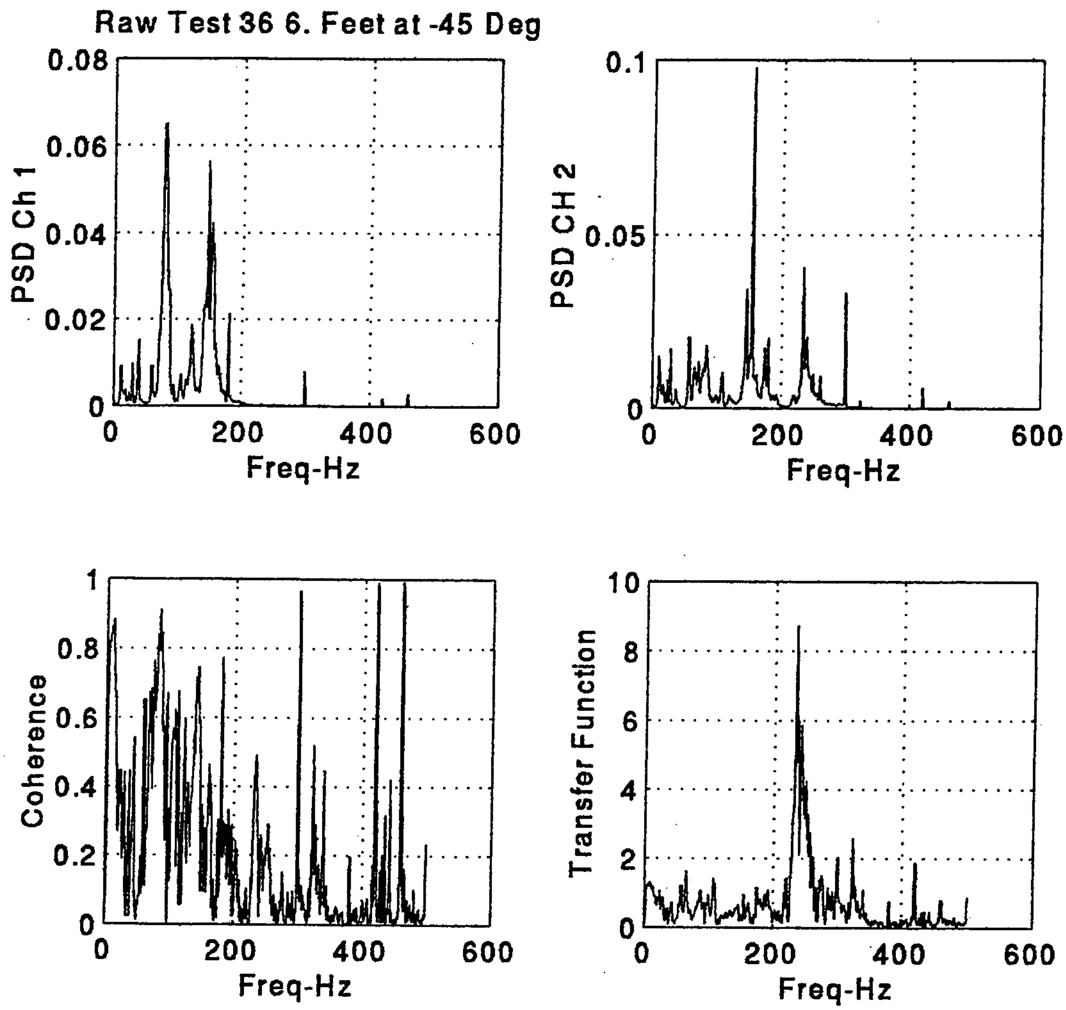

Signal Test 34 2. Feet at -45 Deg
PSD Ch 1
0.06
0.04
0.02
0
0
200
400
600
Freq-Hz
PSD CH 2
0.2
0.1
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
8
6
4
2
0
0
200
400
600
Freq-Hz

Raw Test 35 4. Feet at -45 Deg

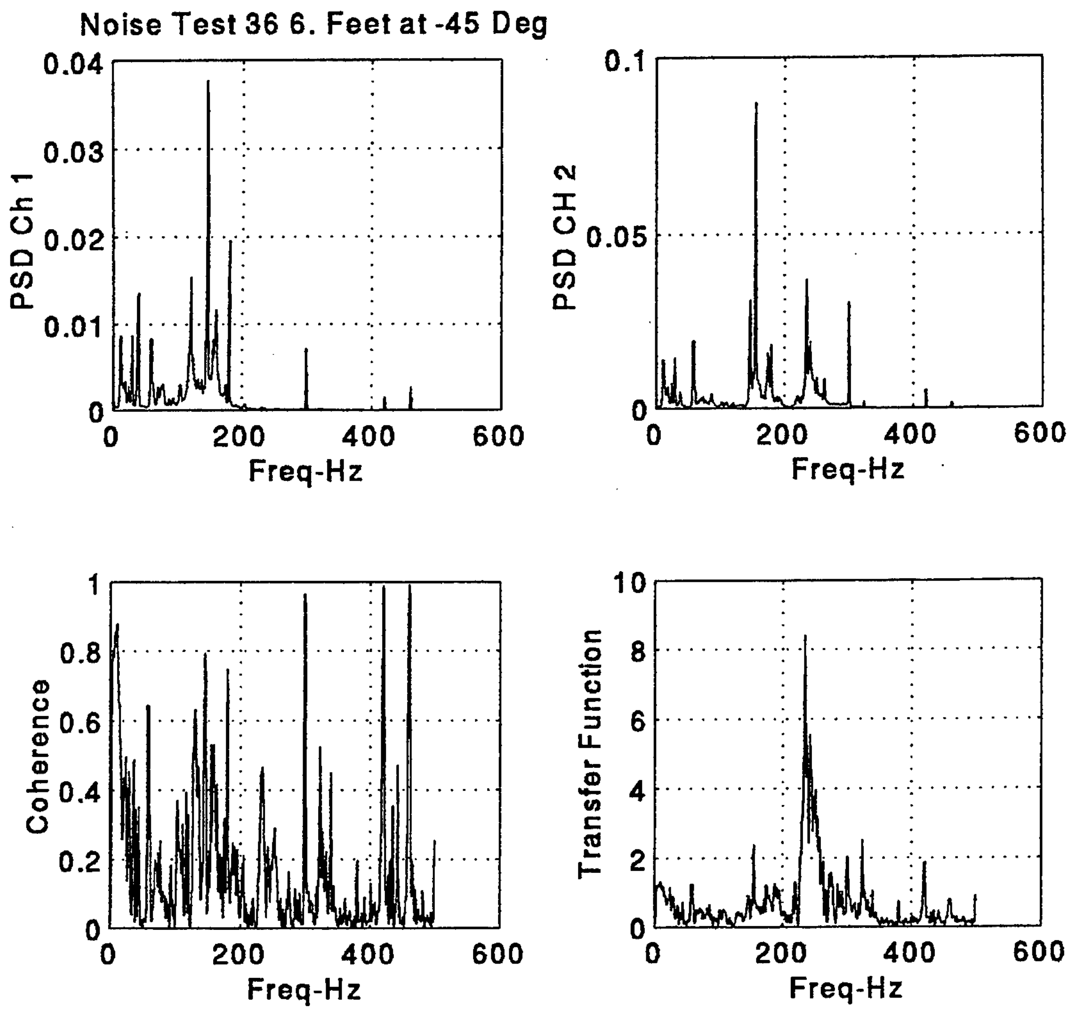


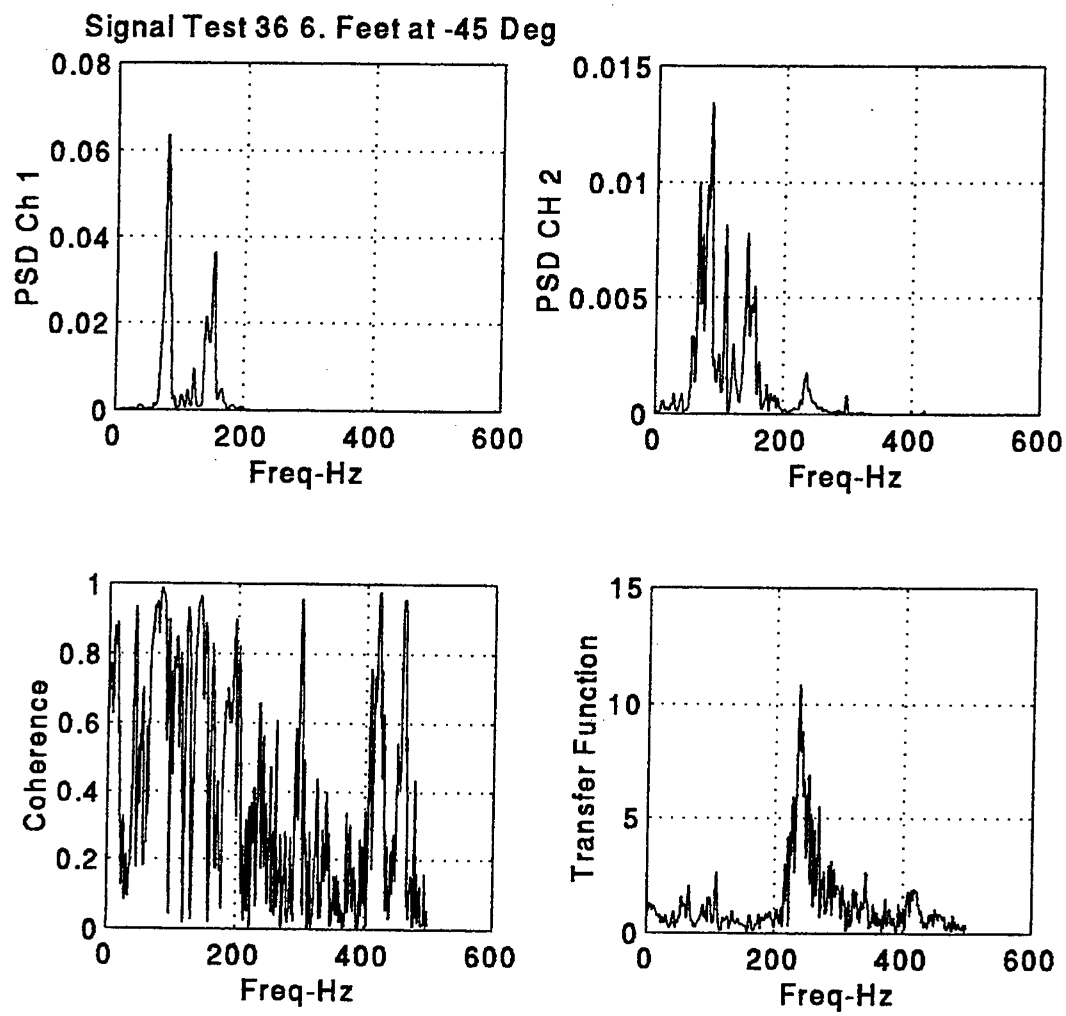

Noise Test 35 4. Feet at -45 Deg
PSD Ch 1
0.03
0.02
0.01
0
0
200
400
600
Freq-Hz
PSD CH 2
0.15
0.1
0.05
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
6
4
2
0
0
200
400
600
Freq-Hz

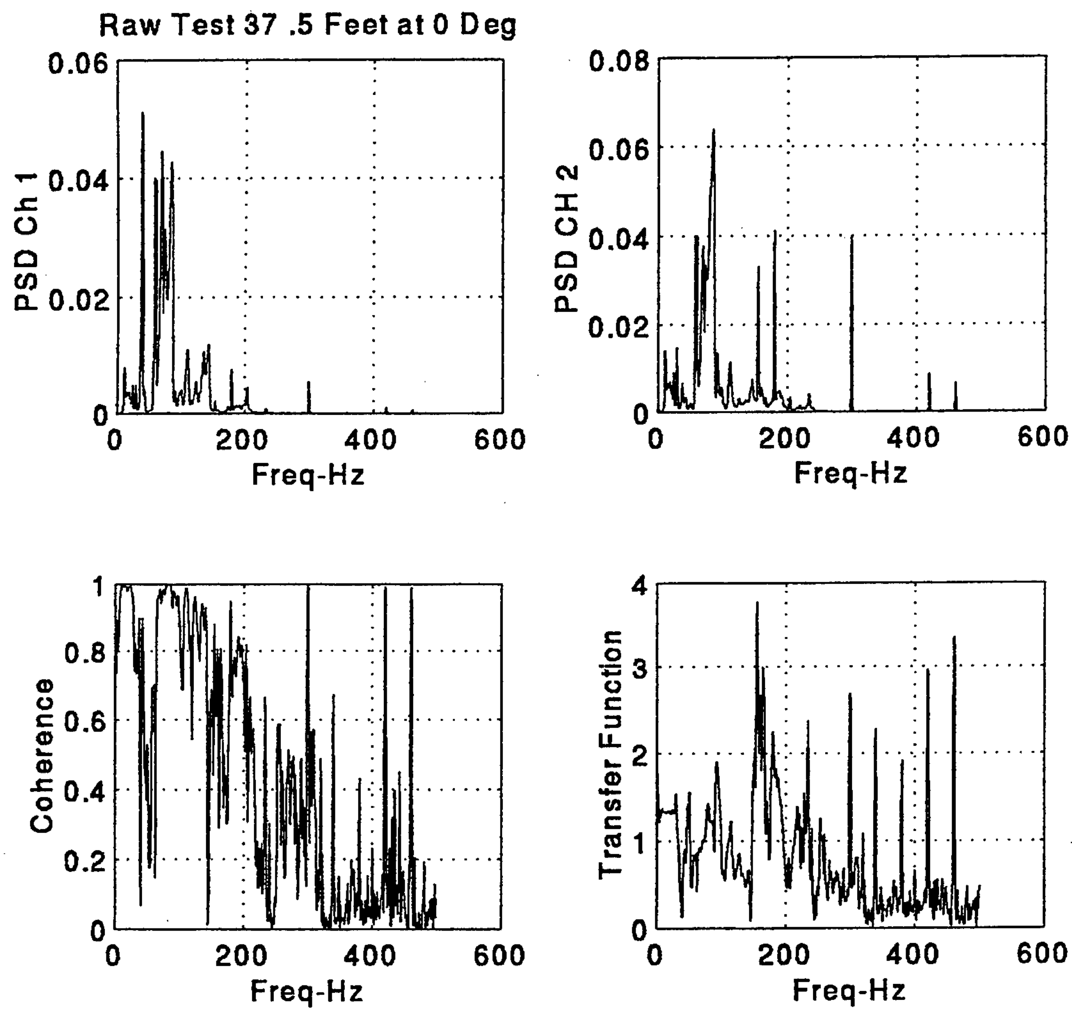

Signal Test 35 4. Feet at -45 Deg
PSD Ch 1
PSD CH 2
Coherence
Transfer Function
Freq-Hz
Freq-Hz
Freq-Hz
Freq-Hz

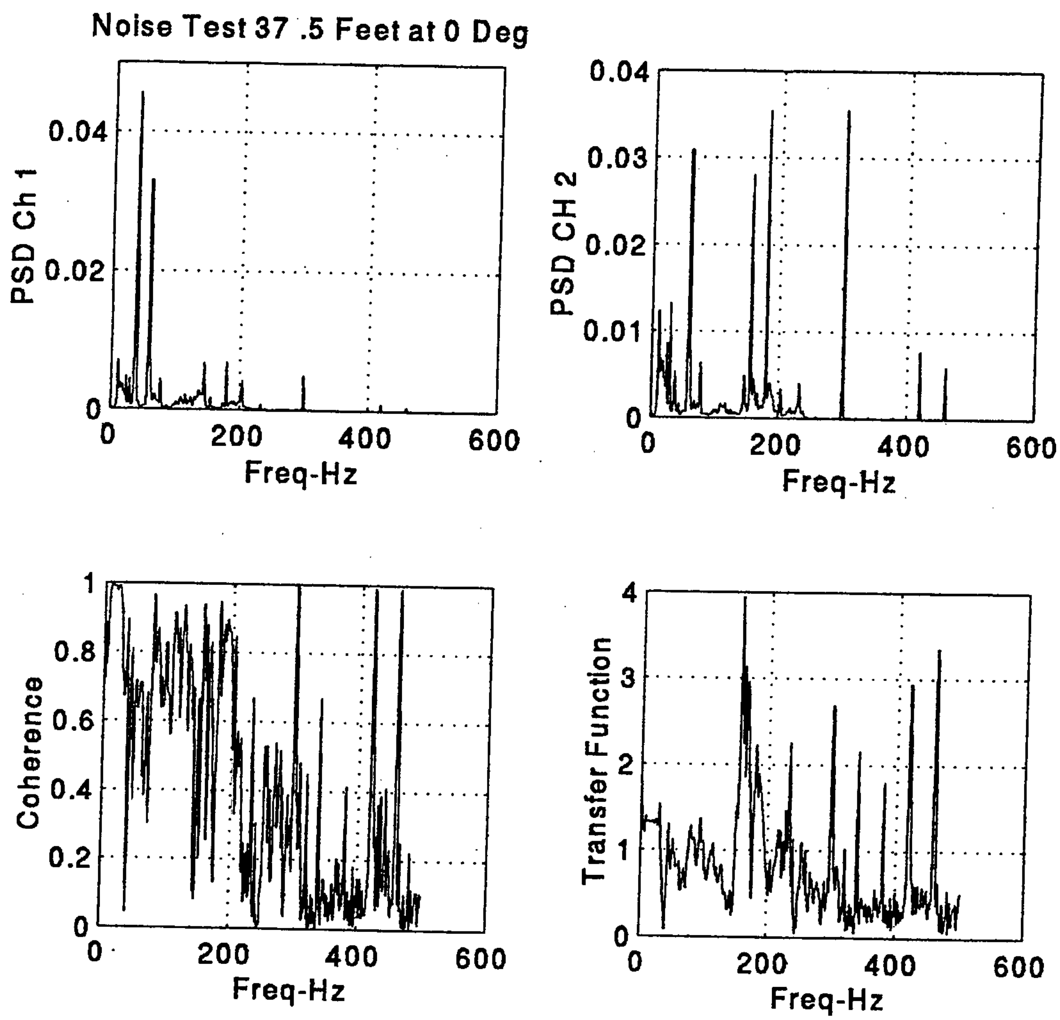


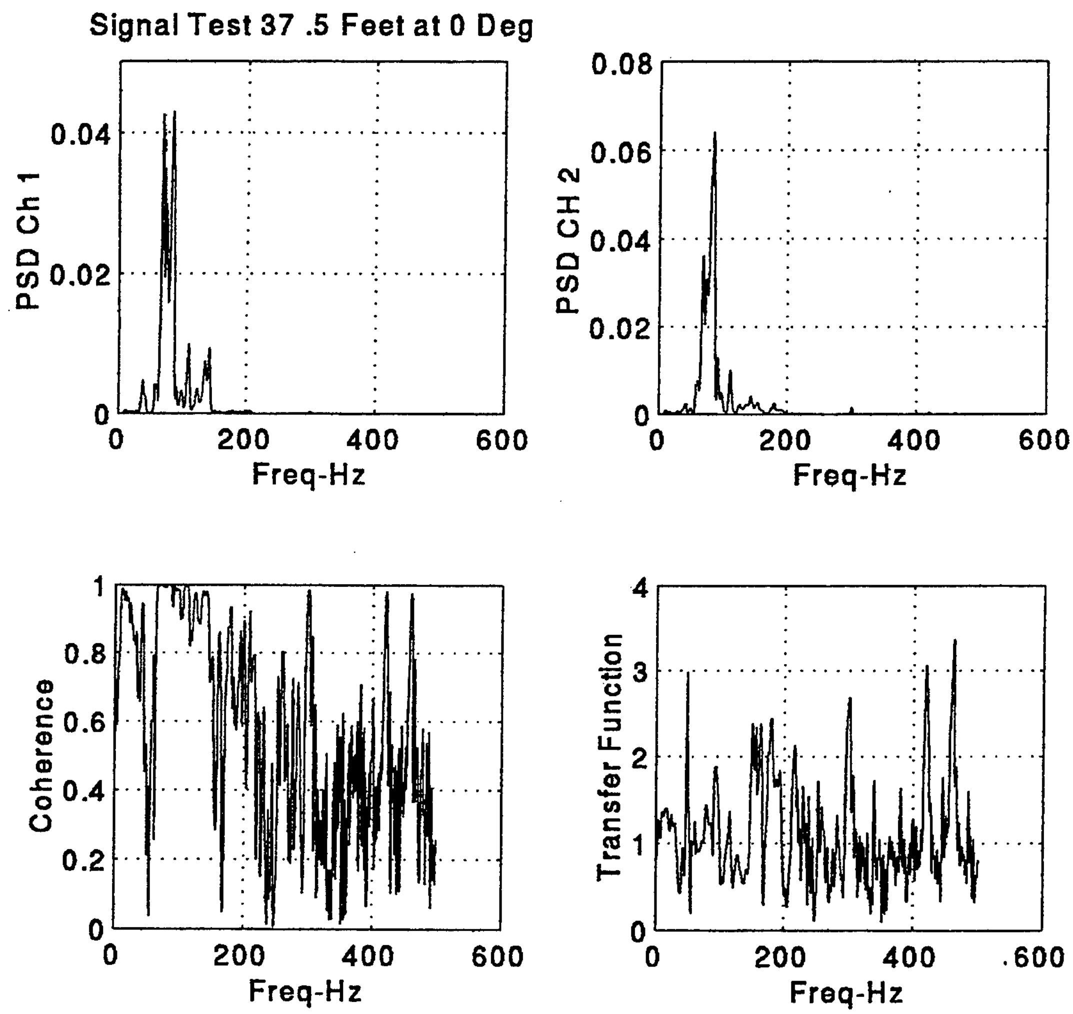
Raw Test 36 6. Feet at -45 Deg
PSD Ch 1
PSD CH 2
Coherence
Transfer Function
Freq-Hz


Noise Test 36 6. Feet at -45 Deg
PSD Ch 1
0.04
0.03
0.02
0.01
0
0
200
400
600
Freq-Hz
PSD CH 2
0.1
0.05
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
10
8
6
4
2
0
0
200
400
600
Freq-Hz

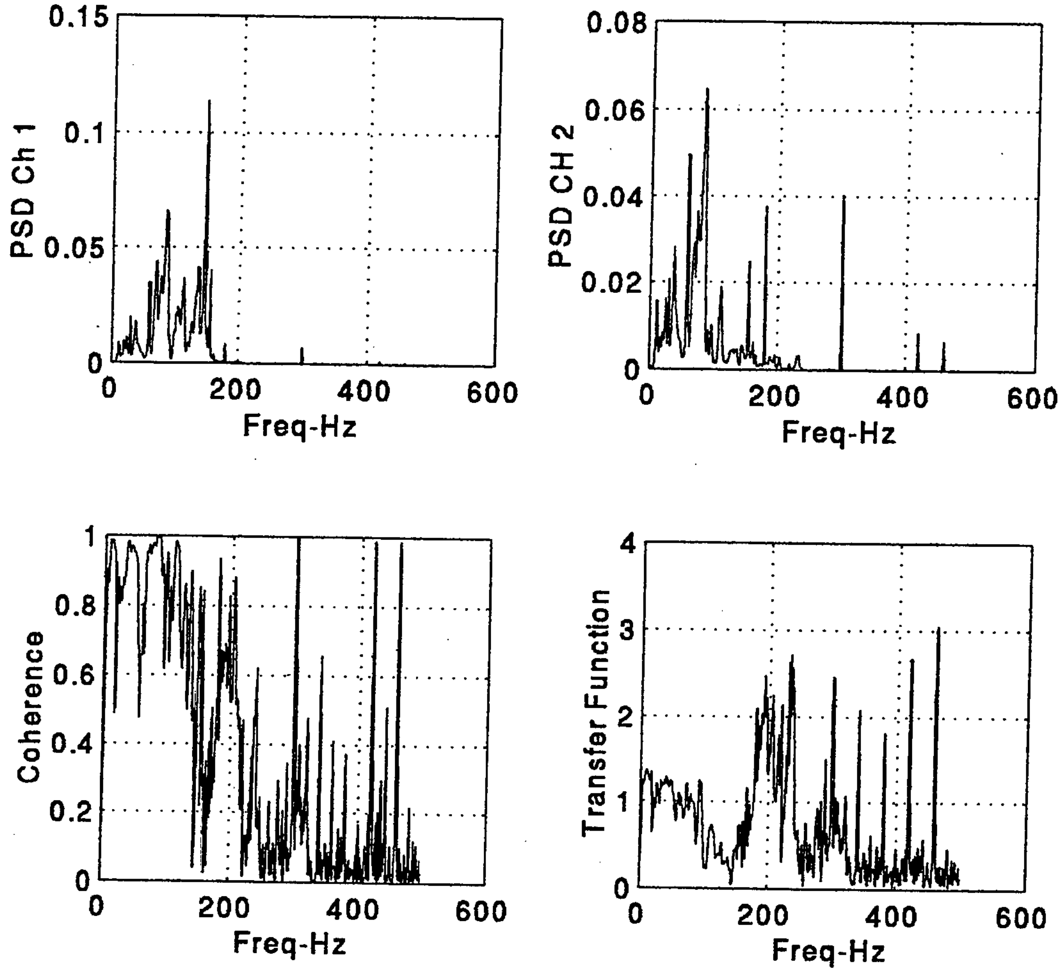

Signal Test 36 6. Feet at -45 Deg
PSD Ch 1
0.08
0.06
0.04
0.02
0
0
200
400
600
Freq-Hz
PSD CH 2
0.015
0.01
0.005
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
15
10
5
0
0
200
400
600
Freq-Hz

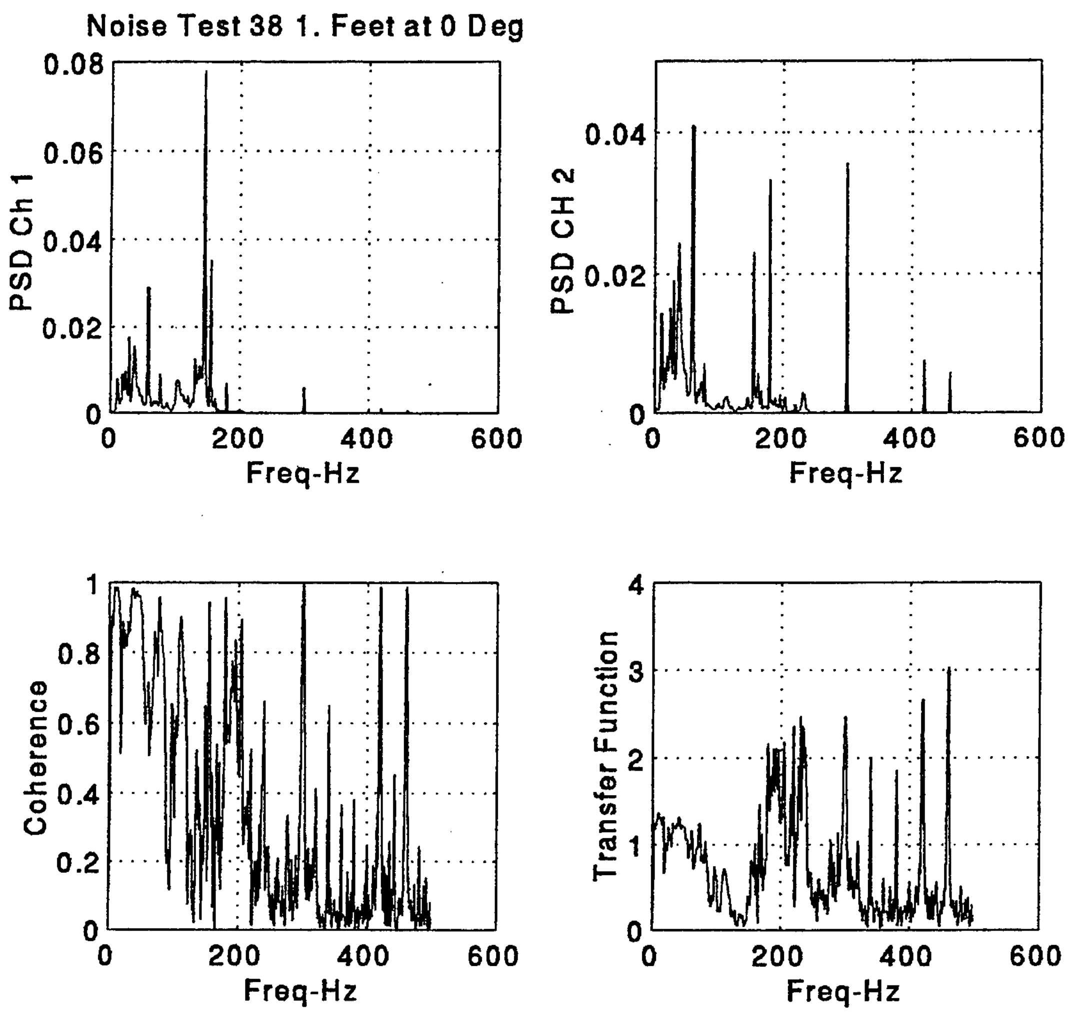

Raw Test 37 .5 Feet at 0 Deg
PSD Ch 1
PSD CH 2
Coherence
Transfer Function
Freq-Hz
0
200
400
600
0.06
0.04
0.02
0.08
1
0.8
0.6
0.4
0.2
4
3
2

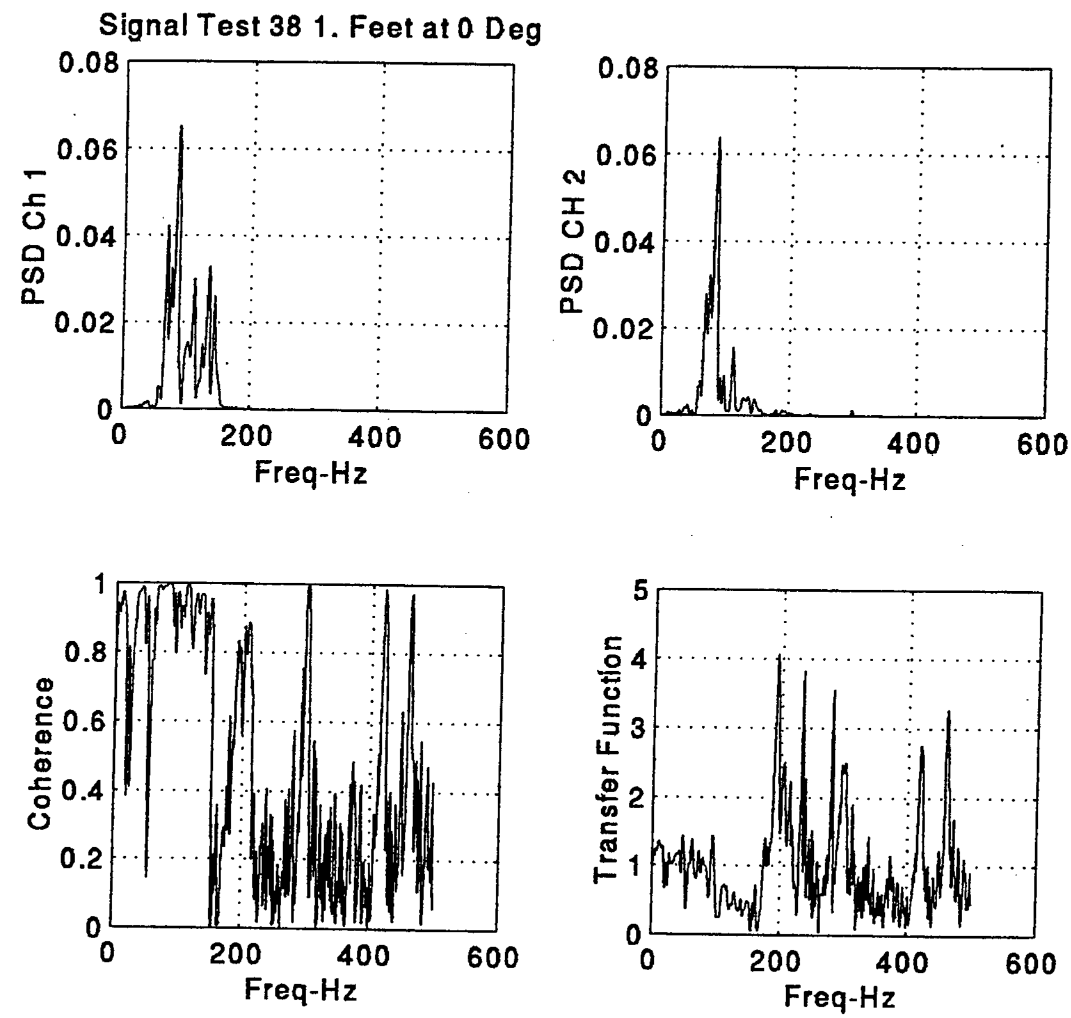

Noise Test 37 .5 Feet at 0 Deg
PSD Ch 1
0.04
0.02
0
0
200
400
600
Freq-Hz
PSD CH 2
0.04
0.03
0.02
0.01
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
4
3
2
1
0
0
200
400
600
Freq-Hz


Signal Test 37 .5 Feet at 0 Deg
PSD Ch 1
0.04
0.02
0
0
200
400
600
Freq-Hz
PSD CH 2
0.08
0.06
0.04
0.02
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
4
3
2
1
0
0
200
400
600
Freq-Hz

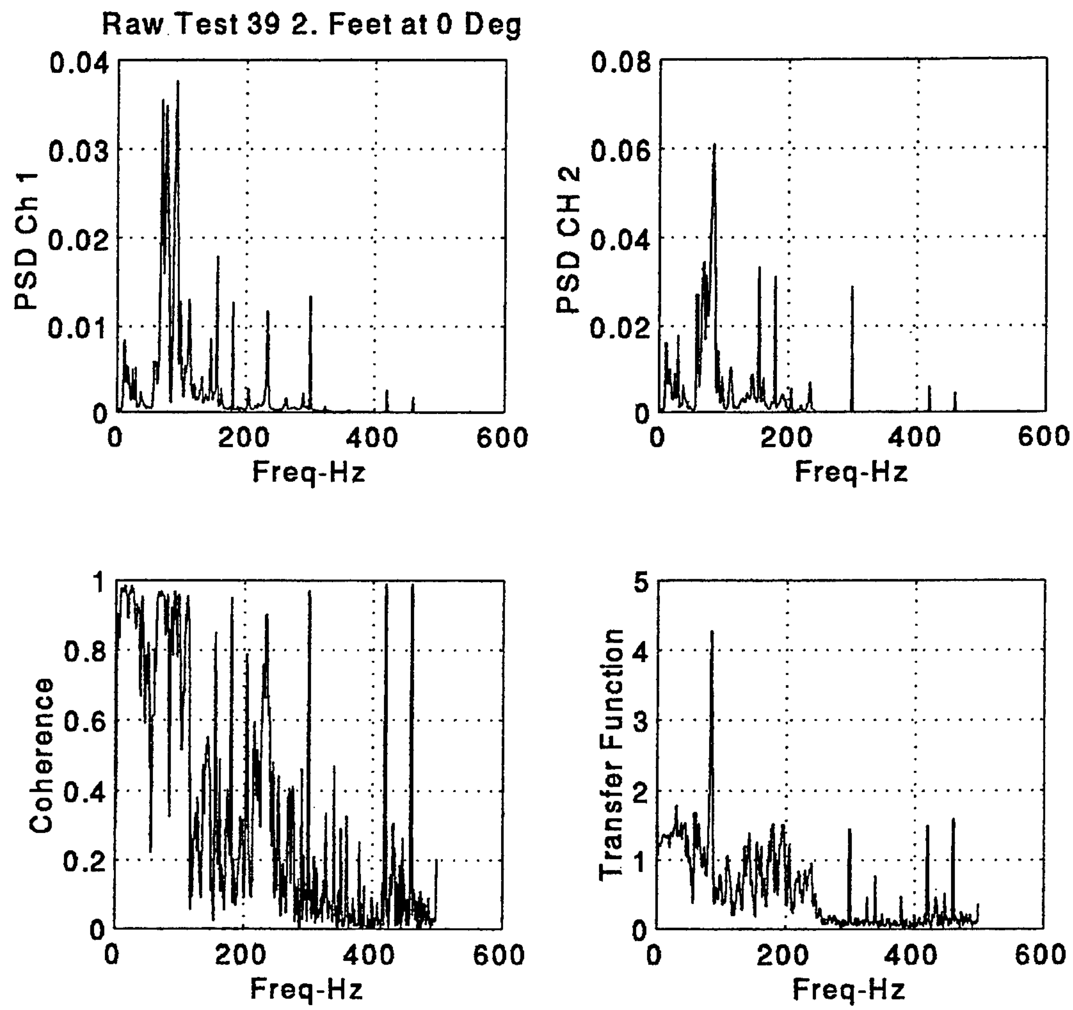
Raw Test 38 1. Feet at 0 Deg

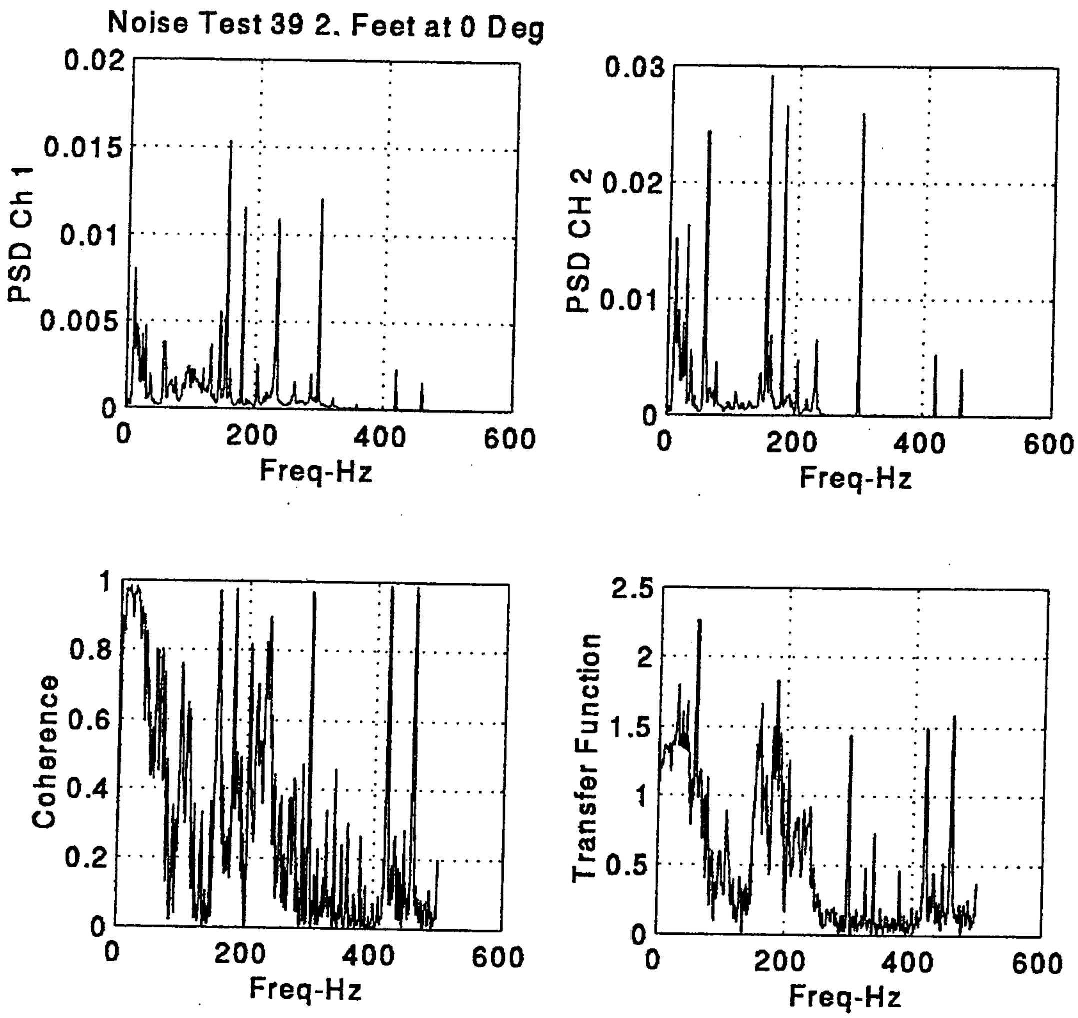



Noise Test 38 1. Feet at 0 Deg
PSD Ch 1
0.08
0.06
0.04
0.02
0
PSD CH 2
0.04
0.02
0
Coherence
1
0.8
0.6
0.4
0.2
0
Transfer Function
4
3
2
1
0
0
200
400
600
Freq-Hz

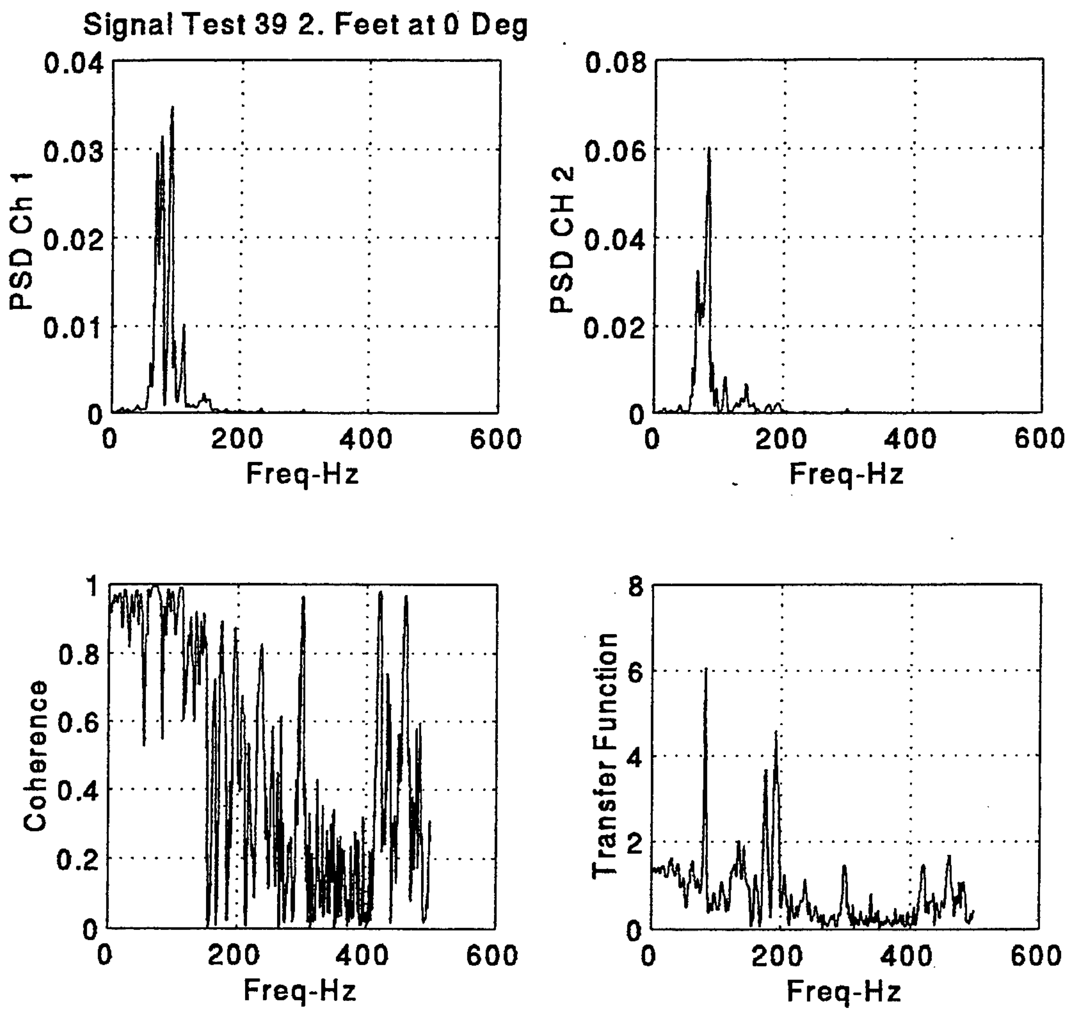


Signal Test 38 1. Feet at 0 Deg
PSD Ch 1
PSD CH 2
Coherence
Transfer Function
Freq-Hz
0
200
400
600
0.08
0.06
0.04
0.02
1
0.8
0.6
0.4
0.2
5
4
3
2

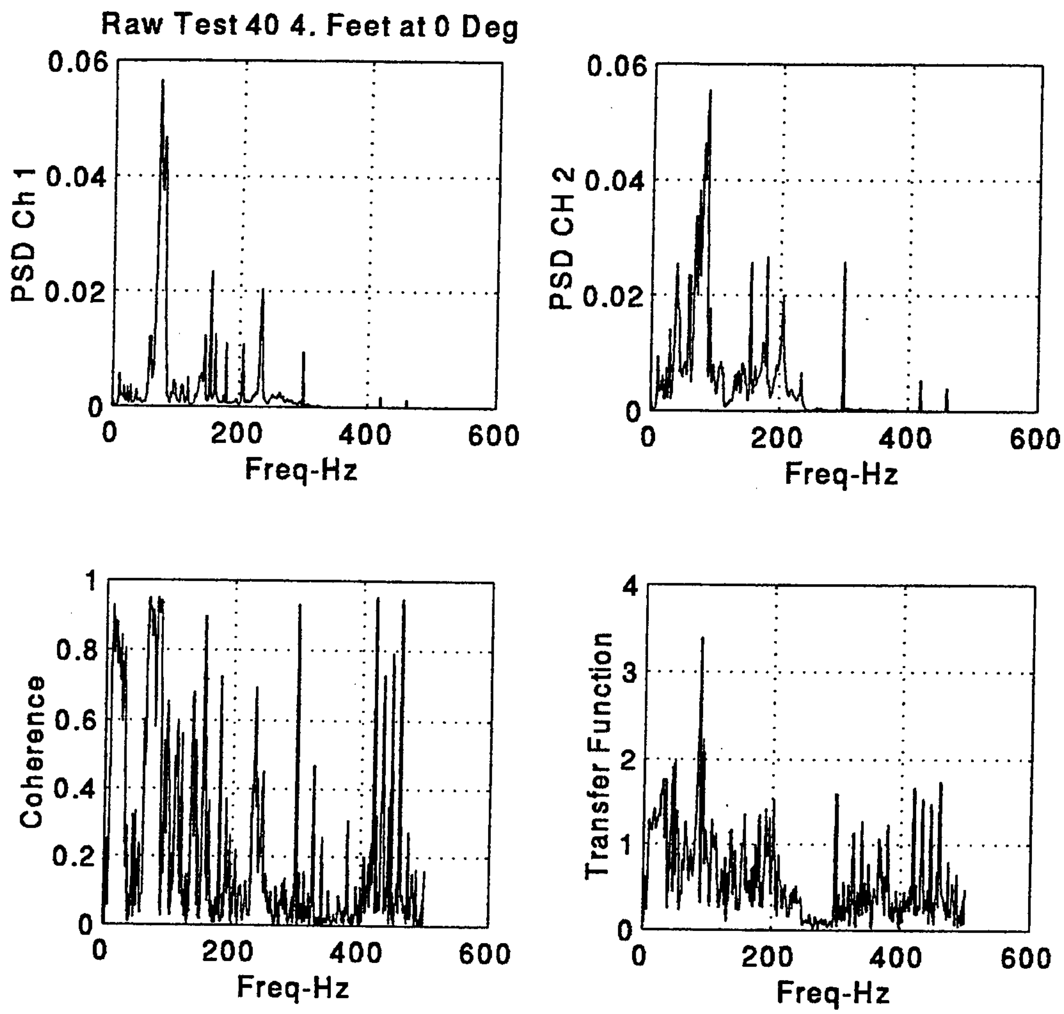

Raw Test 39 2. Feet at 0 Deg
PSD Ch 1
0.04
0.03
0.02
0.01
0
0
200
400
600
Freq-Hz
PSD CH 2
0.08
0.06
0.04
0.02
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
5
4
3
2
1
0
0
200
400
600
Freq-Hz

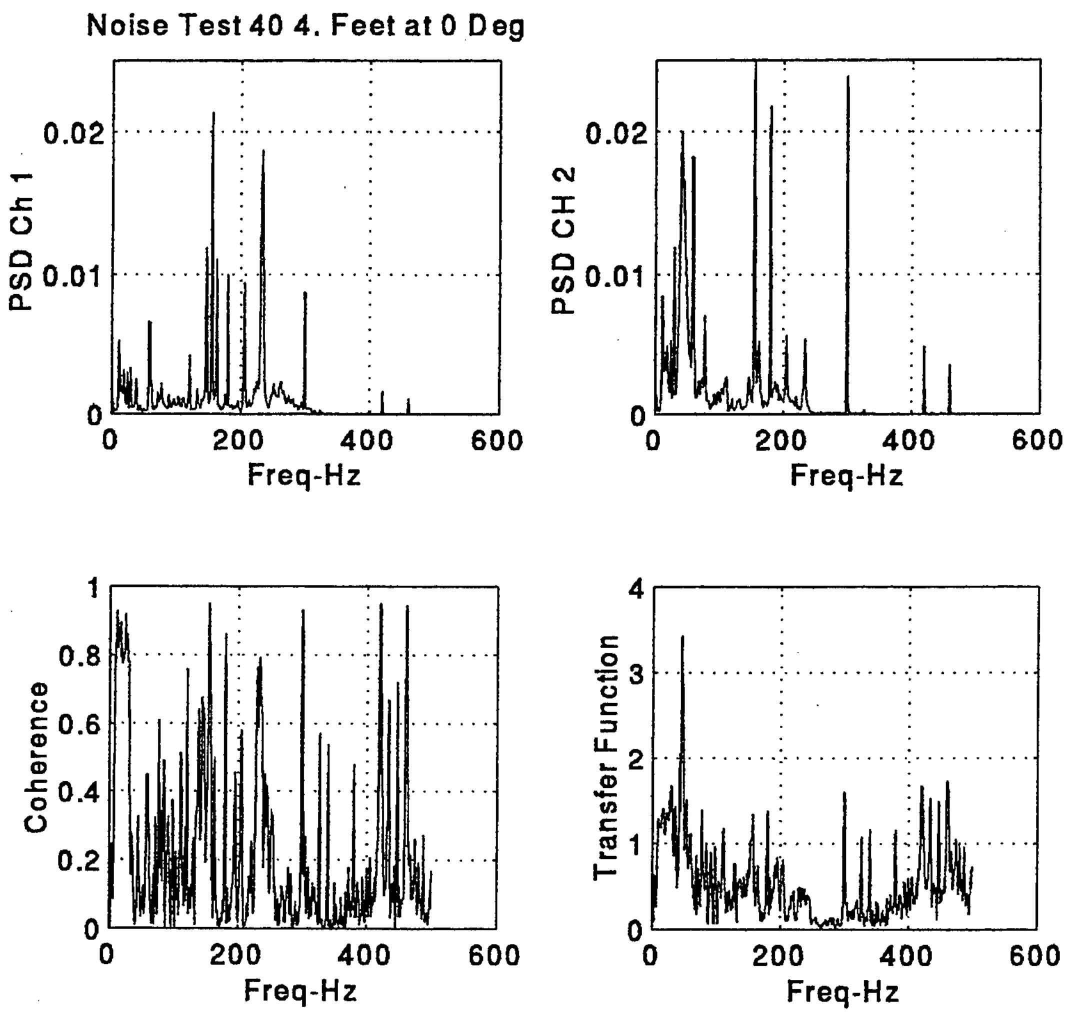


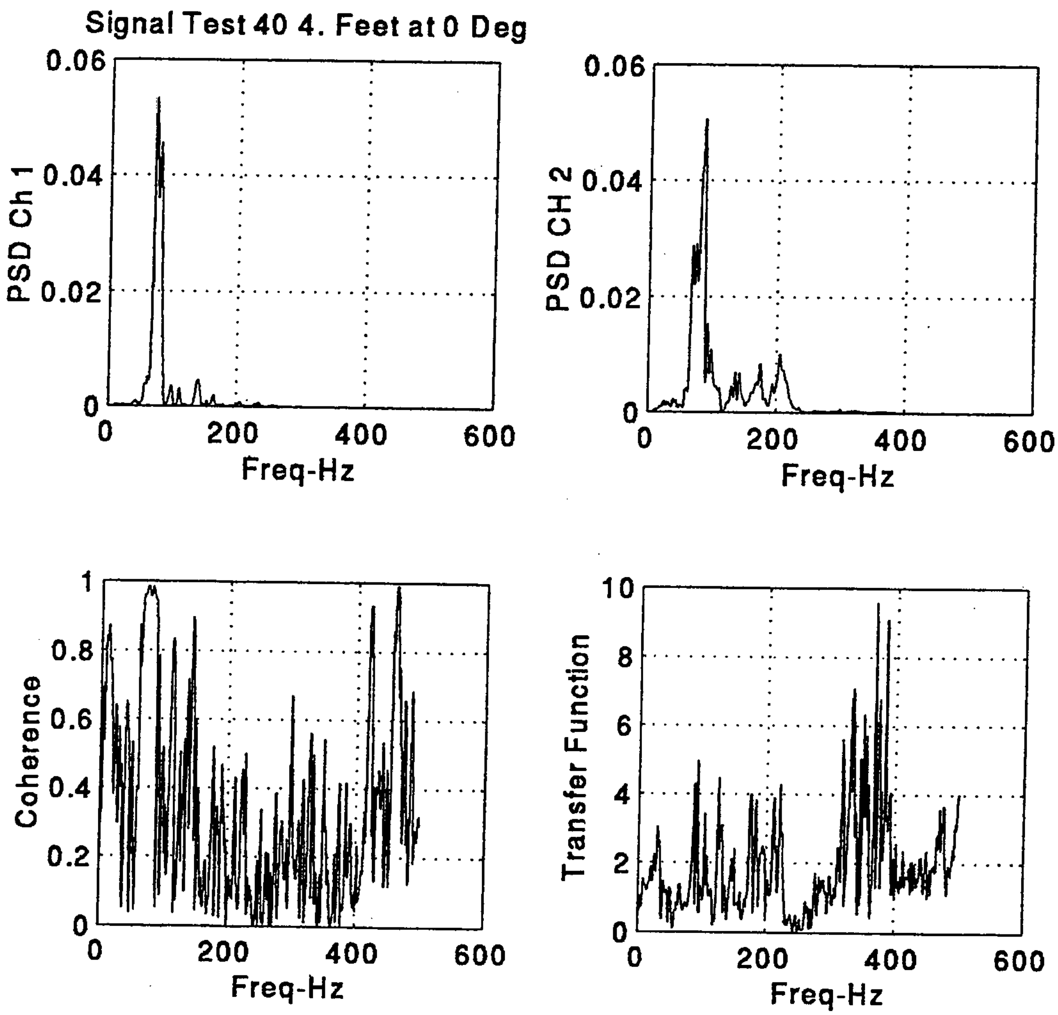
Noise Test 39 2. Feet at 0 Deg
PSD Ch 1
0.02
0.015
0.01
0.005
0
0
200
400
600
Freq-Hz
PSD CH 2
0.03
0.02
0.01
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
2.5
2
1.5
1
0.5
0
0
200
400
600
Freq-Hz

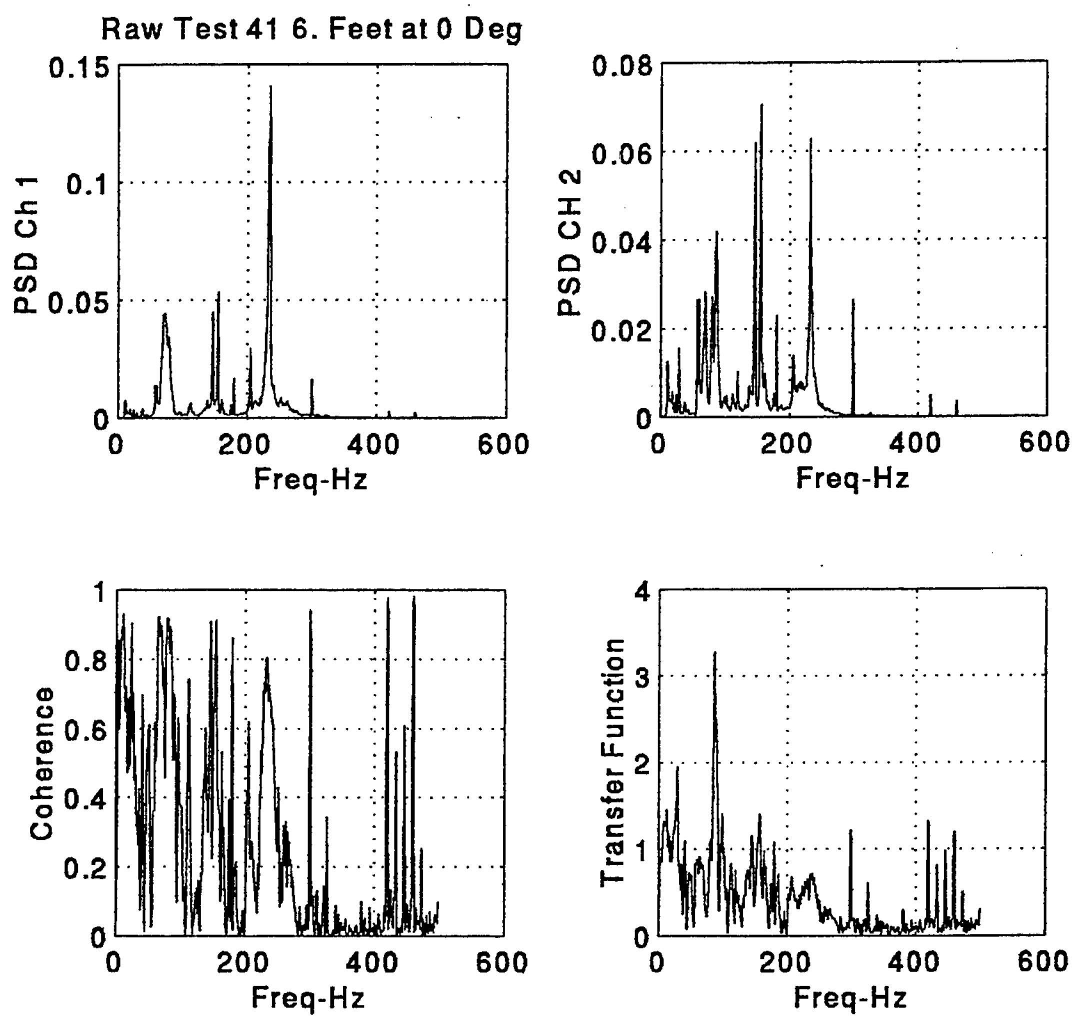


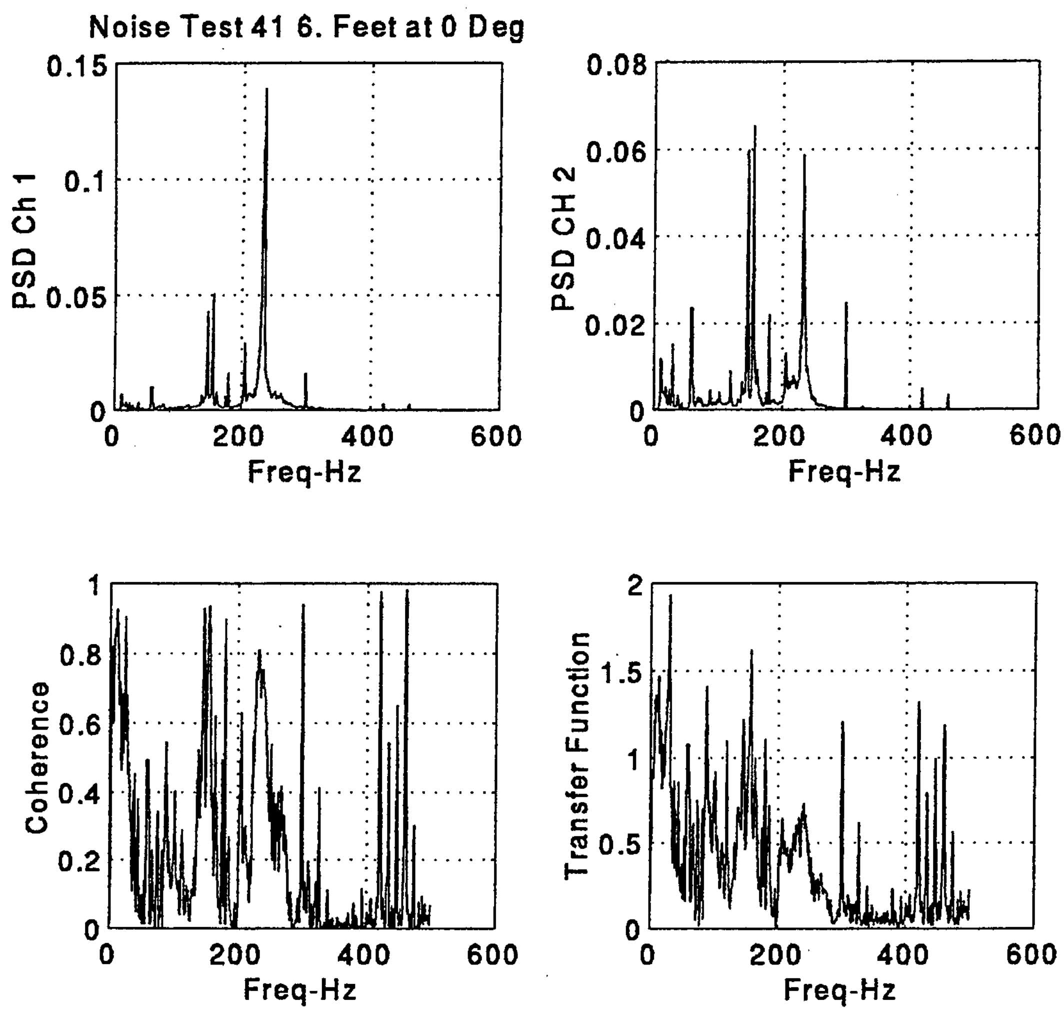
Signal Test 39 2. Feet at 0 Deg
0.04
0.03
0.02
0.01
0
PSD Ch 1
0
200
400
600
Freq-Hz
0.08
0.06
0.04
0.02
0
PSD CH 2
0
200
400
600
Freq-Hz
1
0.8
0.6
0.4
0.2
0
Coherence
0
200
400
600
Freq-Hz
8
6
4
2
0
Transfer Function
0
200
400
600
Freq-Hz

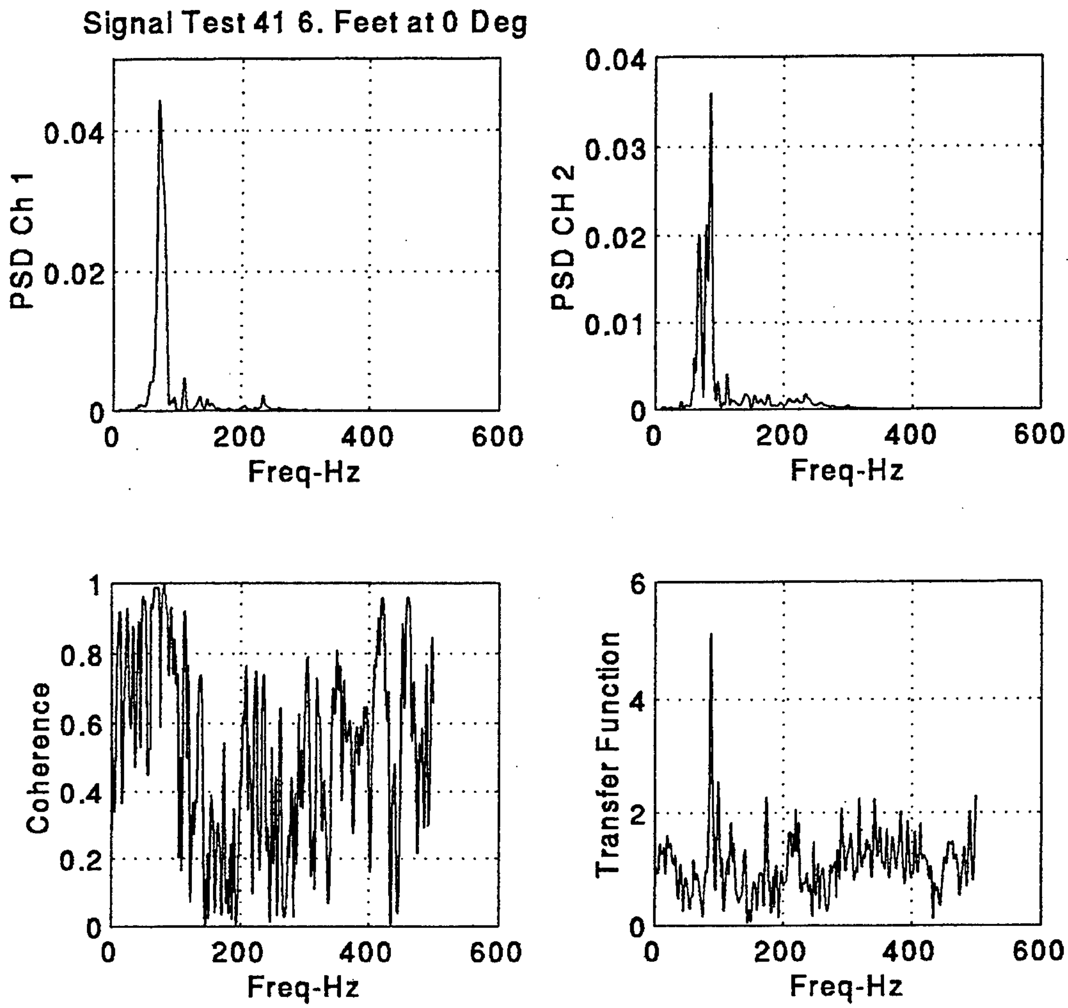

Raw Test 40 4. Feet at 0 Deg
PSD Ch 1
PSD CH 2
Coherence
Transfer Function
Freq-Hz
0.06
0.04
0.02
0
200
400
600
1
0.8
0.6
0.4
0.2
4
3
2
1


Noise Test 40 4. Feet at 0 Deg
PSD Ch 1
0.02
0.01
0
0
200
400
600
Freq-Hz
PSD CH 2
0.02
0.01
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
4
3
2
1
0
0
200
400
600
Freq-Hz


Signal Test 40 4. Feet at 0 Deg
PSD Ch 1
0.06
0.04
0.02
0
0
200
400
600
Freq-Hz
PSD CH 2
0.06
0.04
0.02
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
10
8
6
4
2
0
0
200
400
600
Freq-Hz

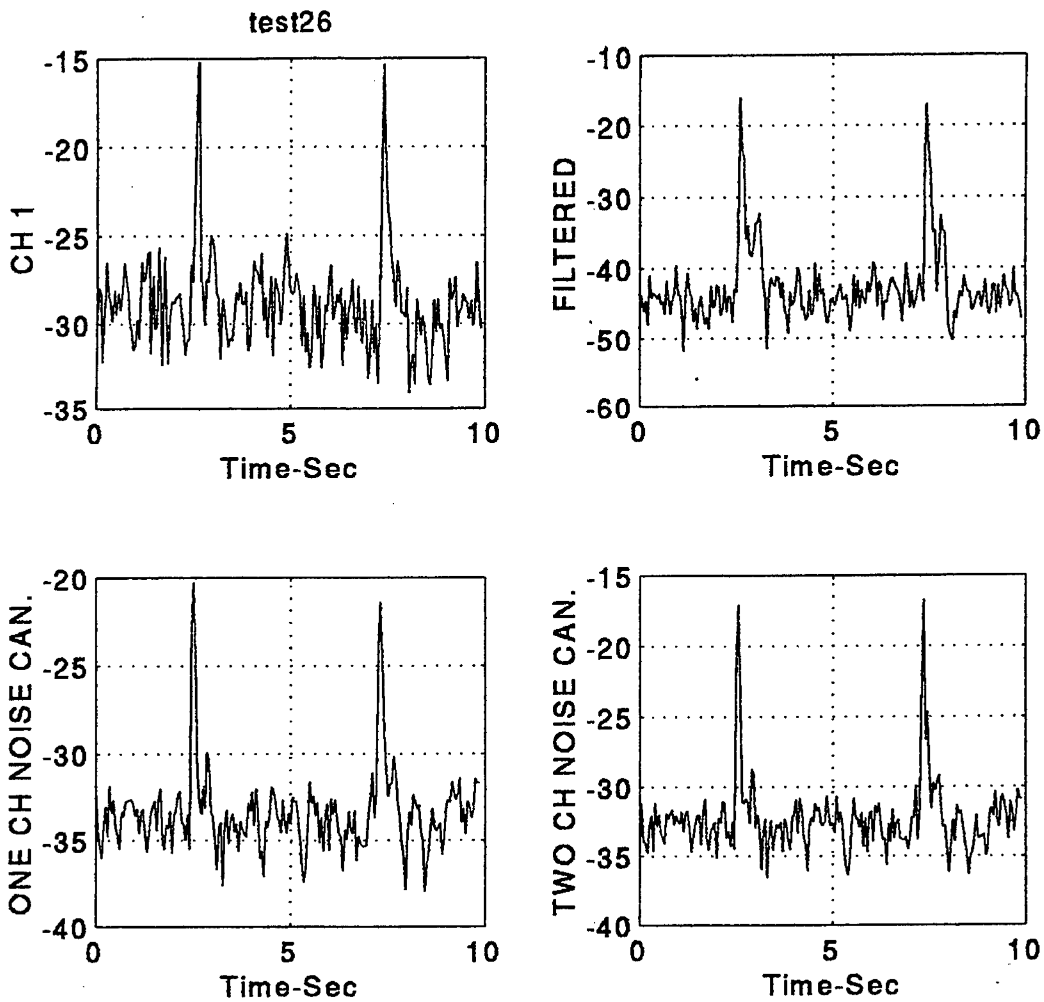

Raw Test 41 6. Feet at 0 Deg
PSD Ch 1
PSD CH 2
Coherence
Transfer Function
Freq-Hz
0.15
0.1
0.05
0
0.08
0.06
0.04
0.02
1
0.8
0.6
0.4
0.2
4
3
2
200
400
600

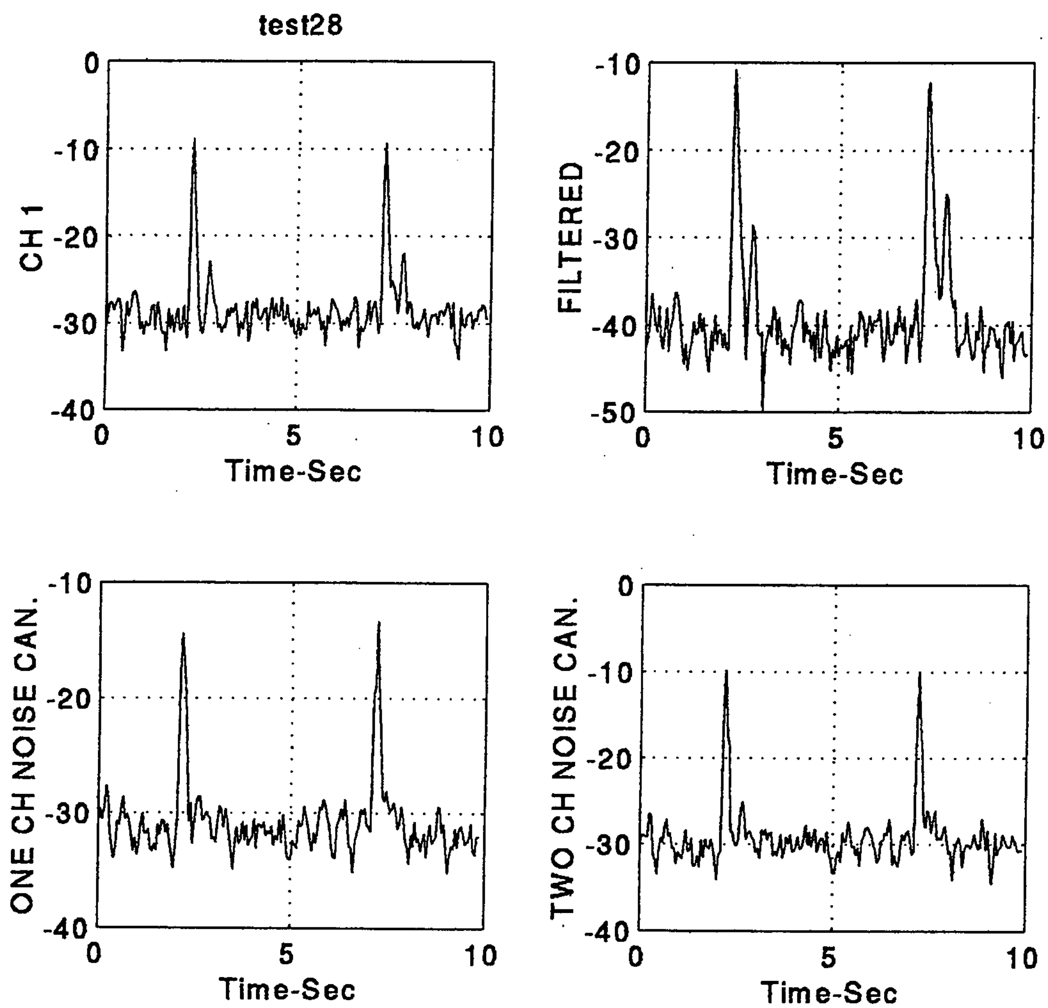

Noise Test 41 6. Feet at 0 Deg
PSD Ch 1
0.15
0.1
0.05
0
0
200
400
600
Freq-Hz
PSD CH 2
0.08
0.06
0.04
0.02
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
2
1.5
1
0.5
0
0
200
400
600
Freq-Hz

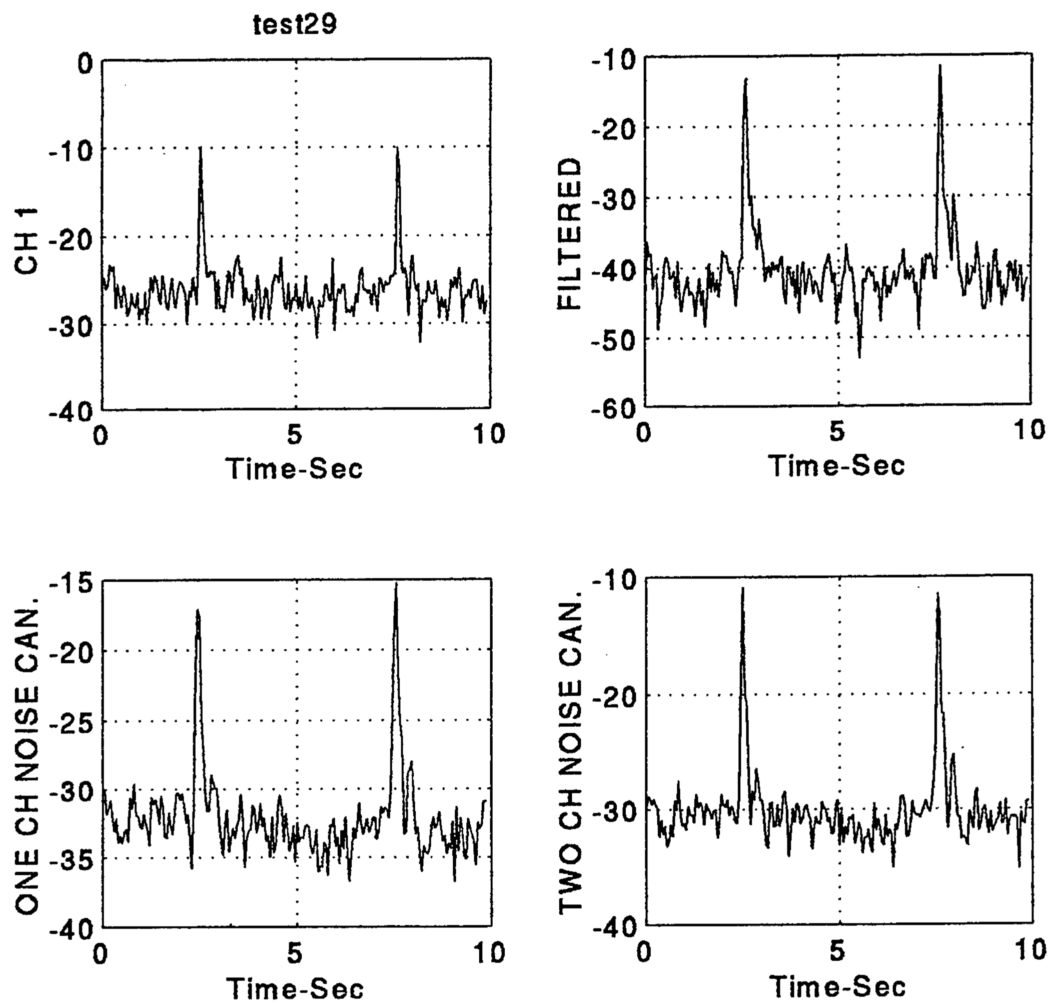

Signal Test 41 6. Feet at 0 Deg
PSD Ch 1
0.04
0.02
0
0
200
400
600
Freq-Hz
PSD CH 2
0.04
0.03
0.02
0.01
0
0
200
400
600
Freq-Hz
Coherence
1
0.8
0.6
0.4
0.2
0
0
200
400
600
Freq-Hz
Transfer Function
6
4
2
0
0
200
400
600
Freq-Hz

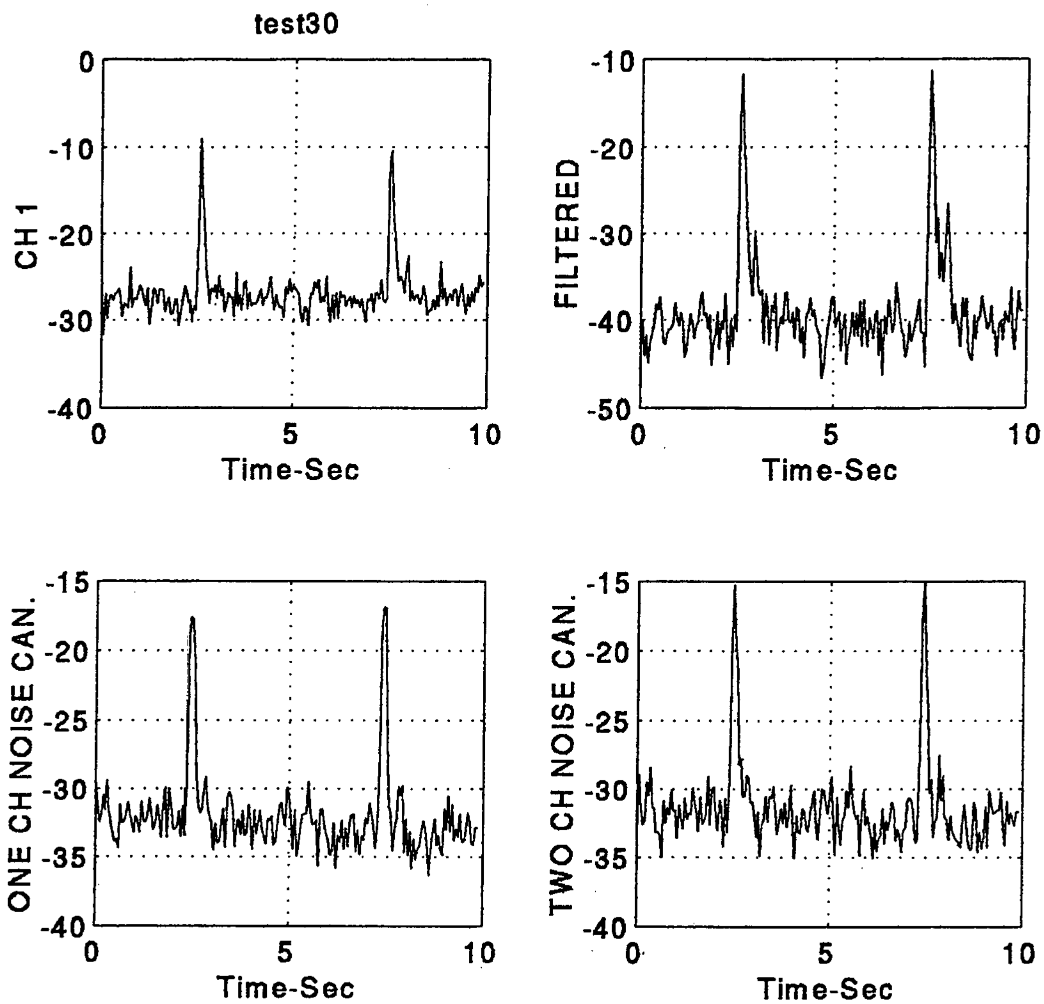
Appendix C
Decibel Plots of Simulated and Field Data

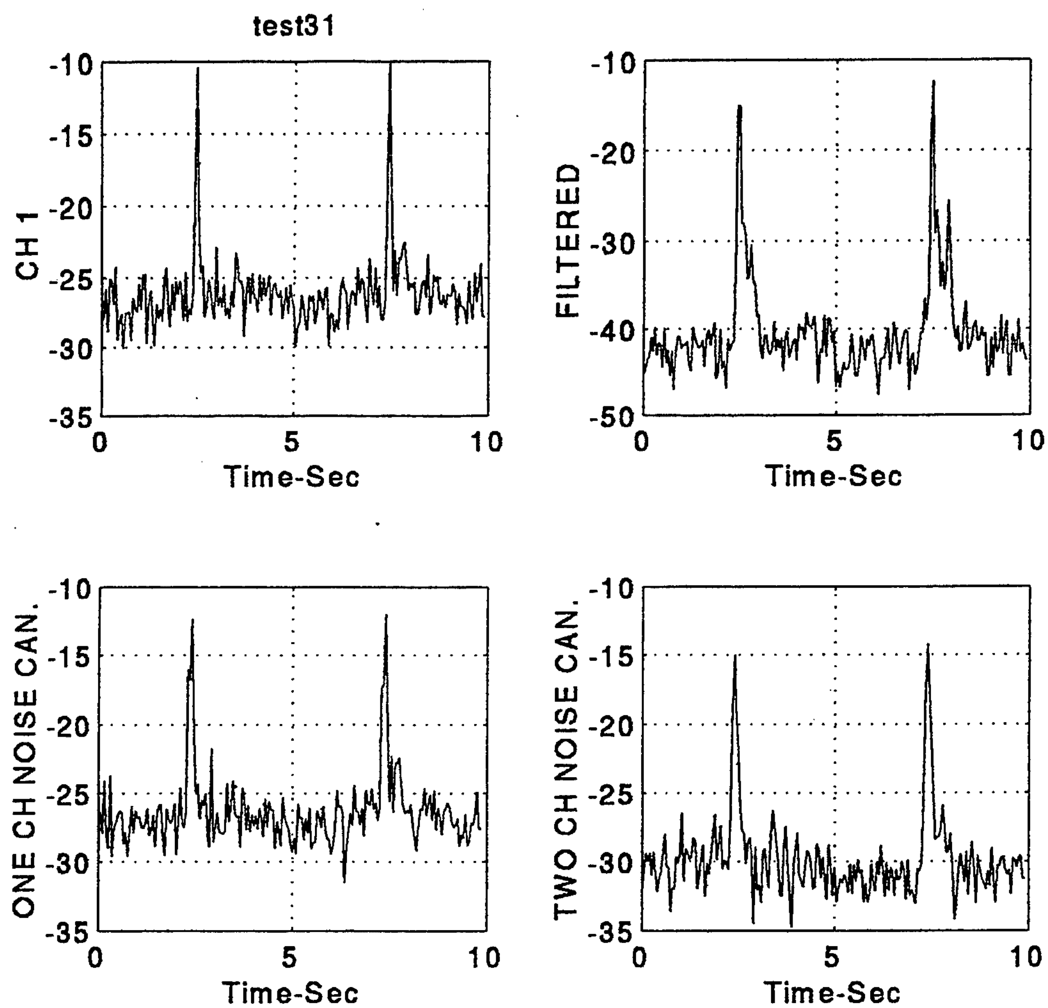
Appendix C shows plots of the processed results for both the field test and the simulated test. The plots show the results for band-pass filtering, one-channel noise cancellation filtering, two-channel noise cancellation filtering, and no filtering. (Referenced on page 38).

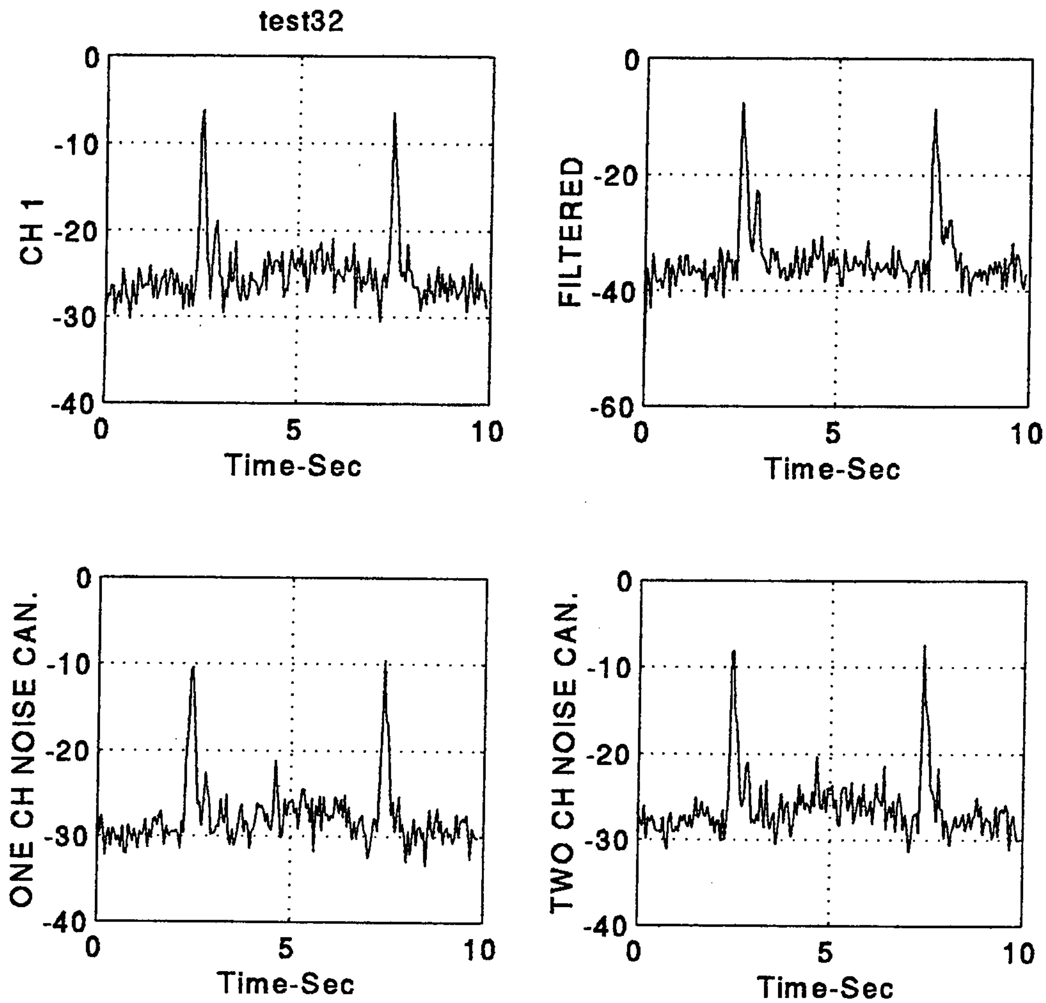

test26
CH 1
-15
-20
-25
-30
-35
0
5
10
Time-Sec
FILTERED
-10
-20
-30
-40
-50
-60
0
5
10
Time-Sec
ONE CH NOISE CAN.
-20
-25
-30
-35
-40
0
5
10
Time-Sec
TWO CH NOISE CAN.
-15
-20
-25
-30
-35
-40
0
5
10
Time-Sec

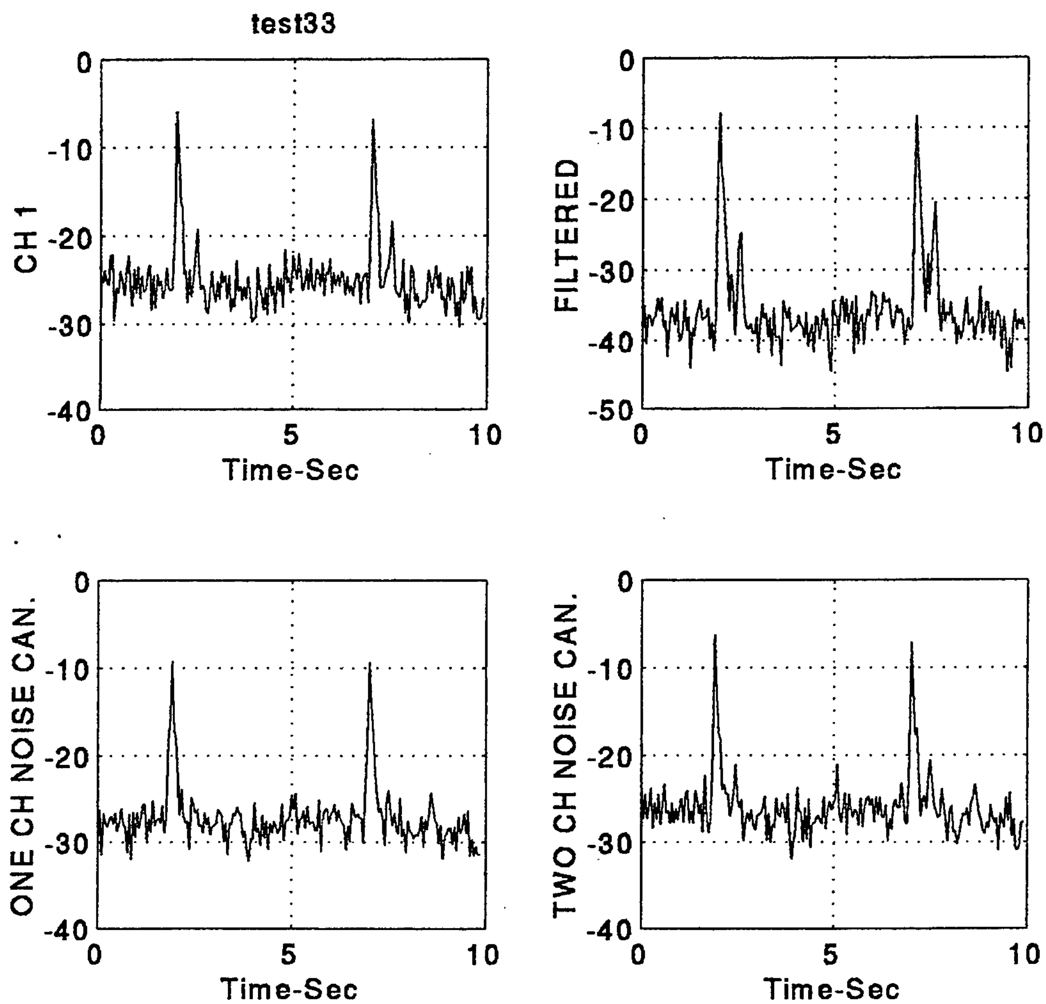

test28
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
Time-Sec
Time-Sec
Time-Sec

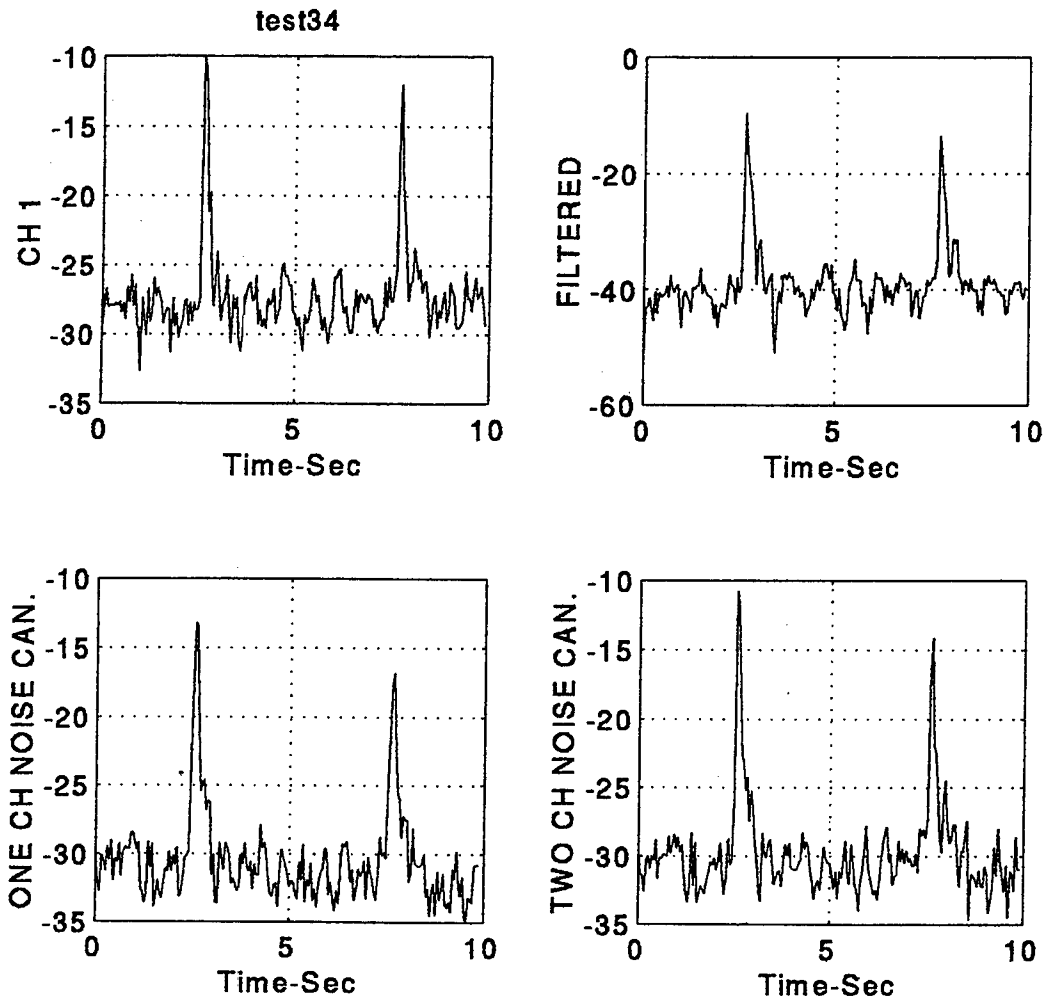

test29
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
Time-Sec
Time-Sec
Time-Sec

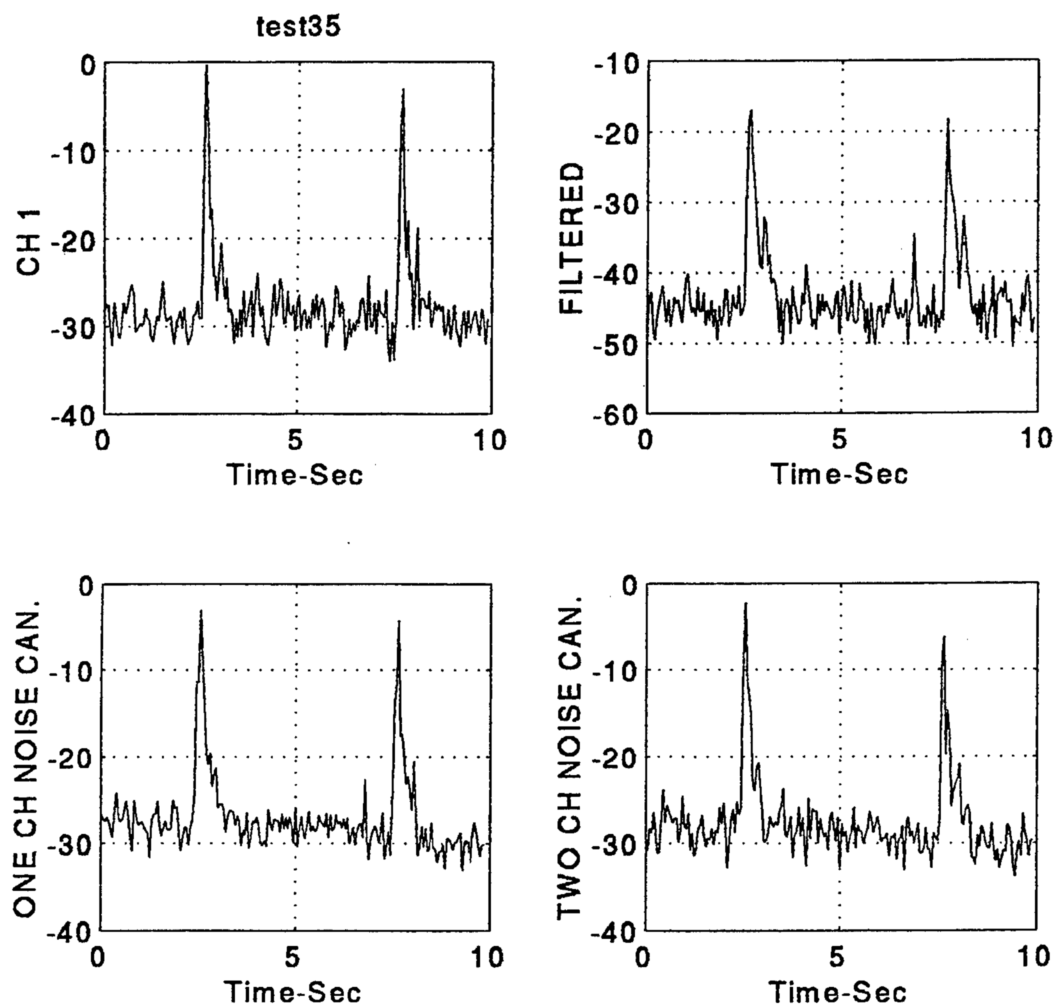

test30
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
Time-Sec
Time-Sec
Time-Sec

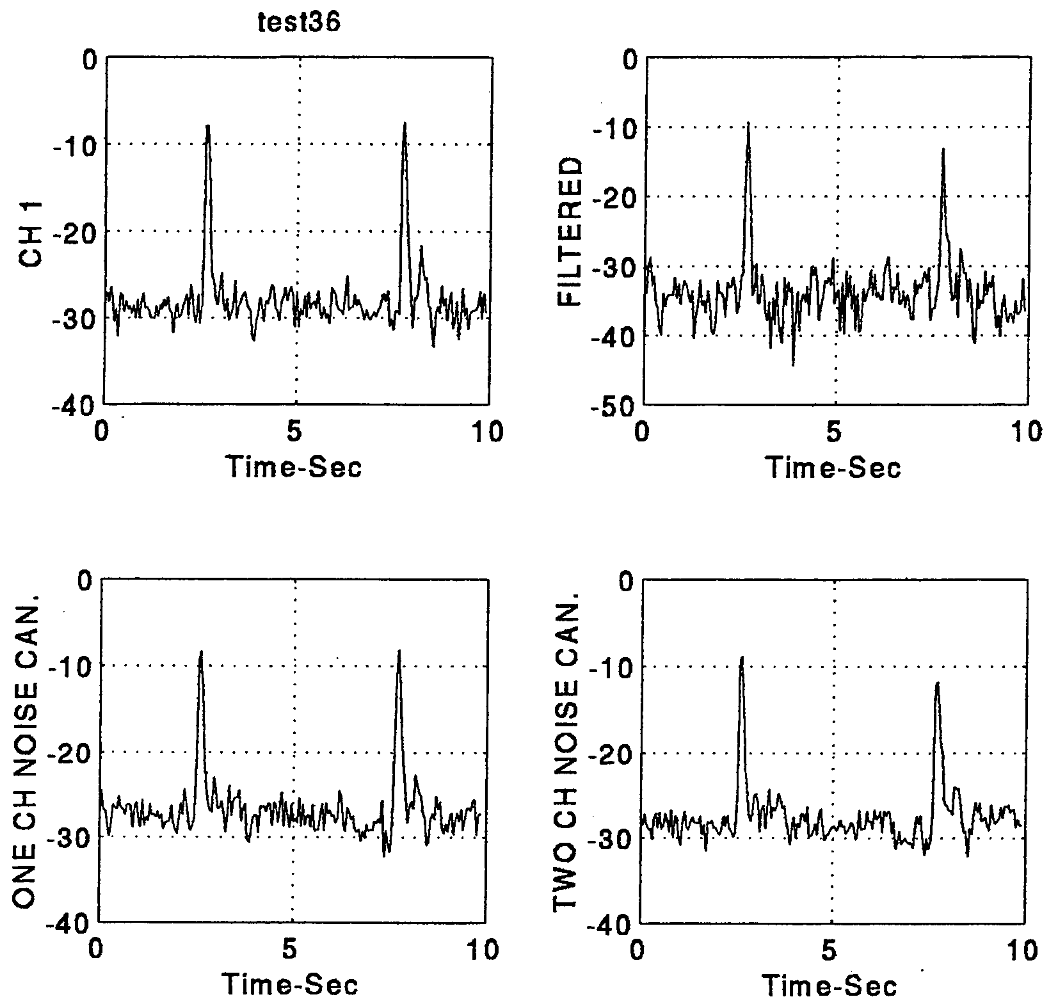

test31
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
Time-Sec
Time-Sec
Time-Sec

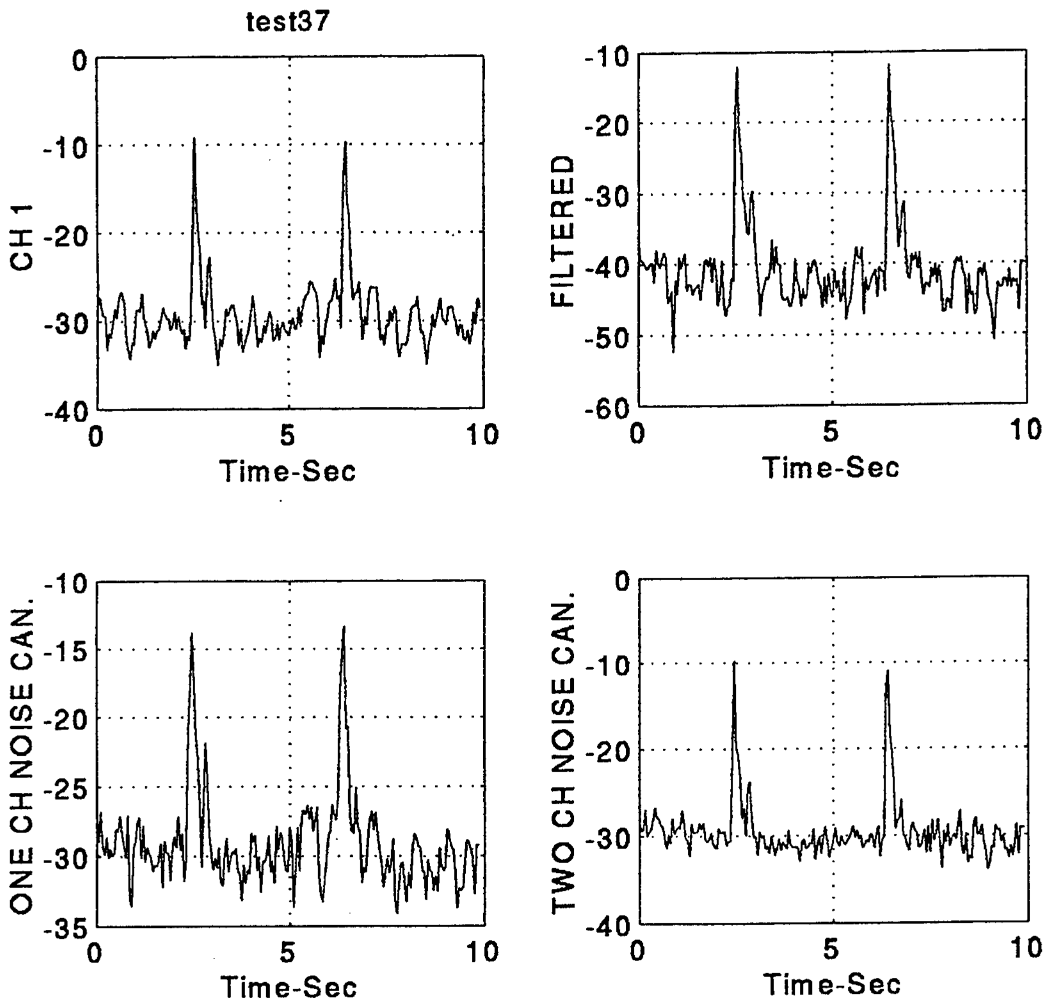

test32
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
0
-10
-20
-30
-40
-60
5
10

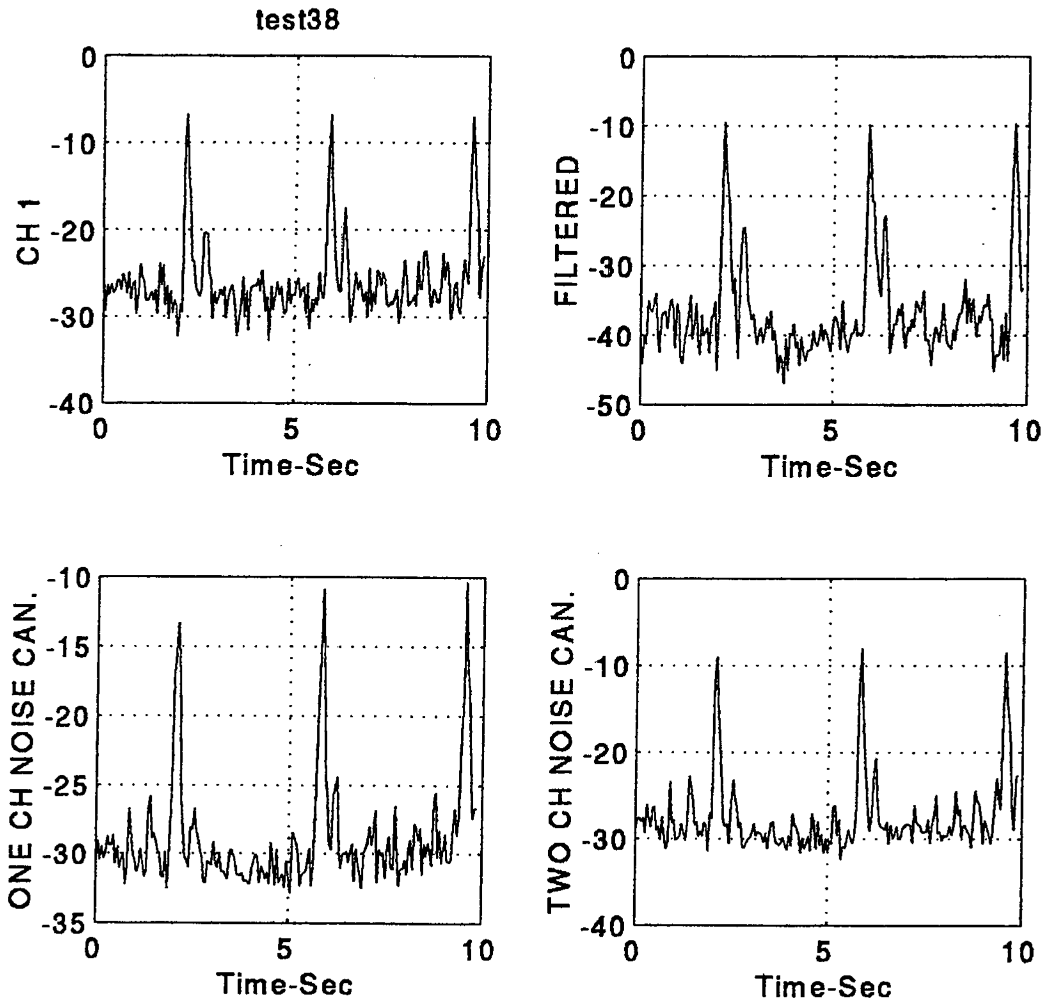

test33
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
Time-Sec
Time-Sec
Time-Sec
0
-10
-20
-30
-40
-50
0
5
10


test34
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
Time-Sec
Time-Sec
Time-Sec
-10
-15
-20
-25
-30
-35
0
-20
-40
-60
0
5
10

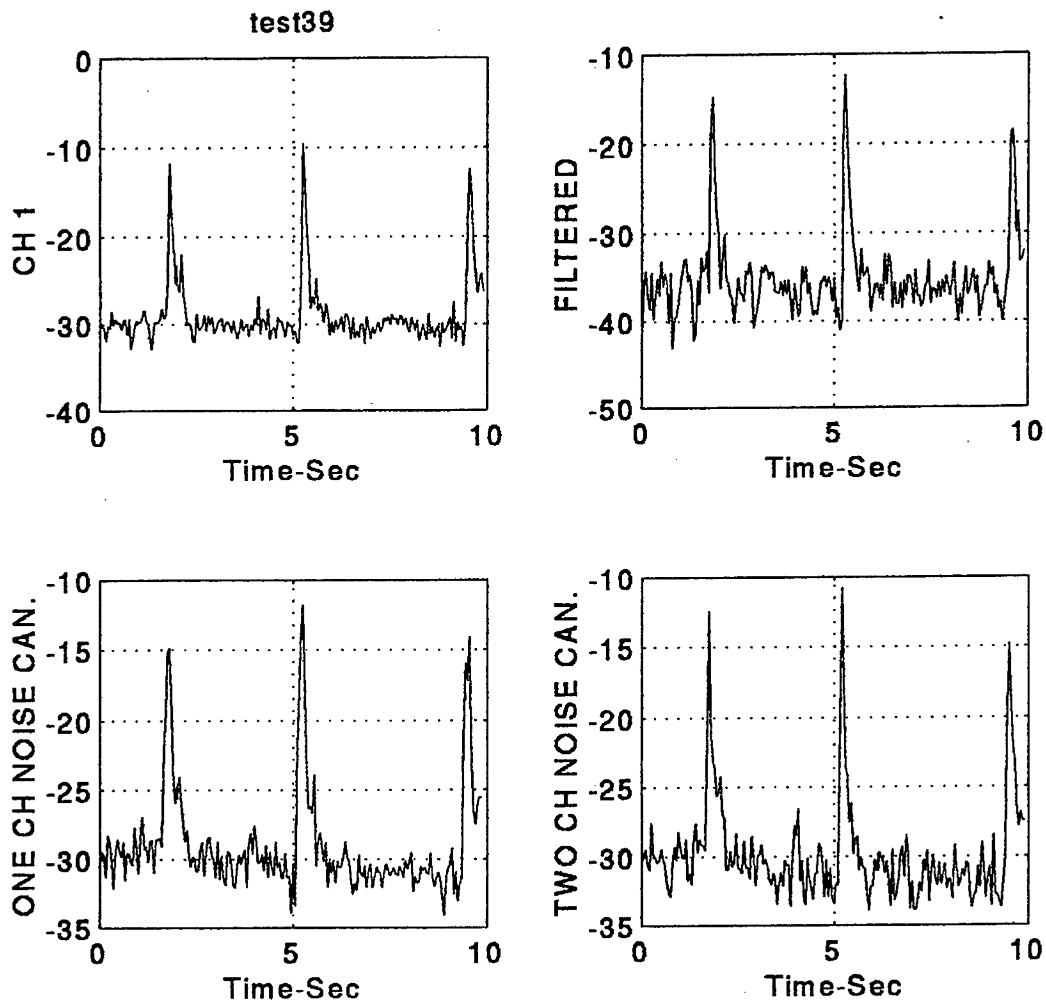

test35
CH 1
0
-10
-20
-30
-40
0
5
10
Time-Sec
FILTERED
-10
-20
-30
-40
-50
-60
0
5
10
Time-Sec
ONE CH NOISE CAN.
0
-10
-20
-30
-40
0
5
10
Time-Sec
TWO CH NOISE CAN.
0
-10
-20
-30
-40
0
5
10
Time-Sec

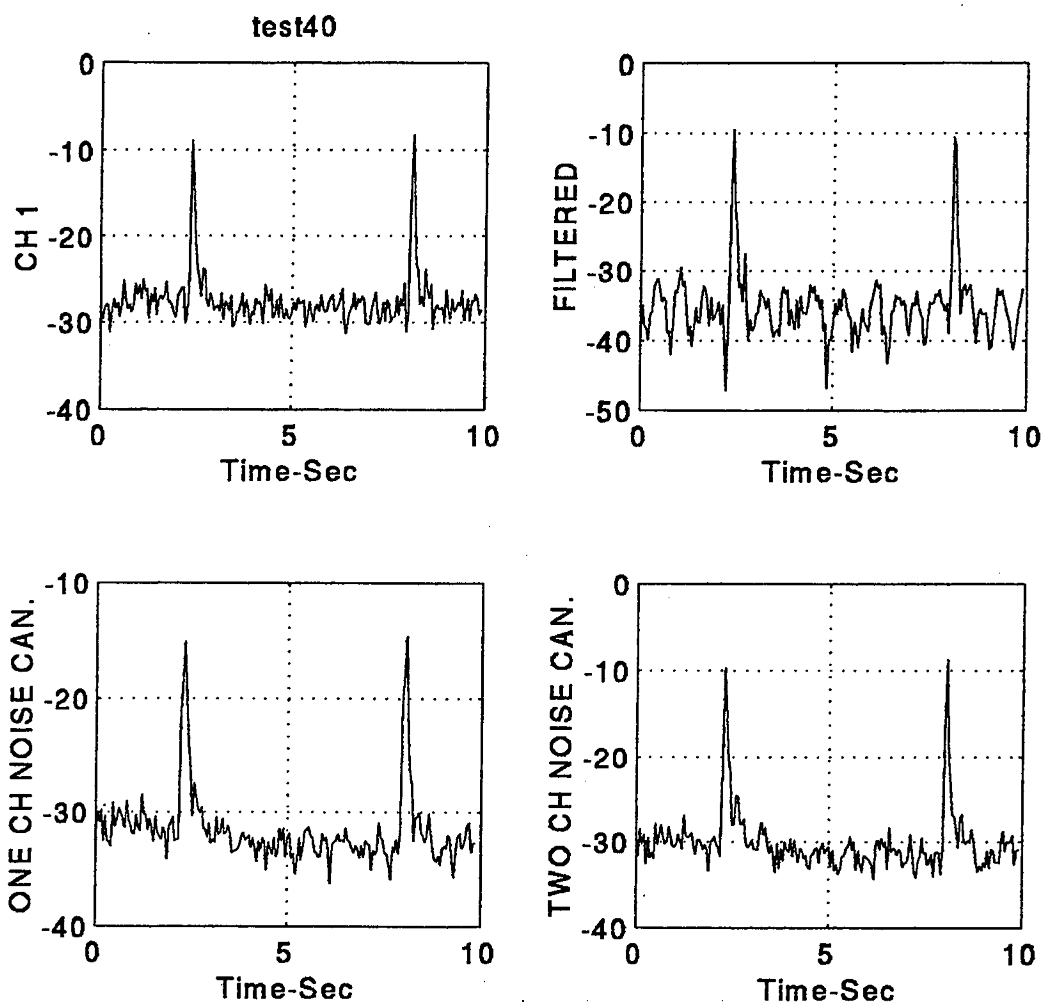

test36
0
-10
-20
-30
-40
CH 1
0
5
10
Time-Sec
0
-10
-20
-30
-40
-50
FILTERED
0
5
10
Time-Sec
0
-10
-20
-30
-40
ONE CH NOISE CAN.
0
5
10
Time-Sec
0
-10
-20
-30
-40
TWO CH NOISE CAN.
0
5
10
Time-Sec

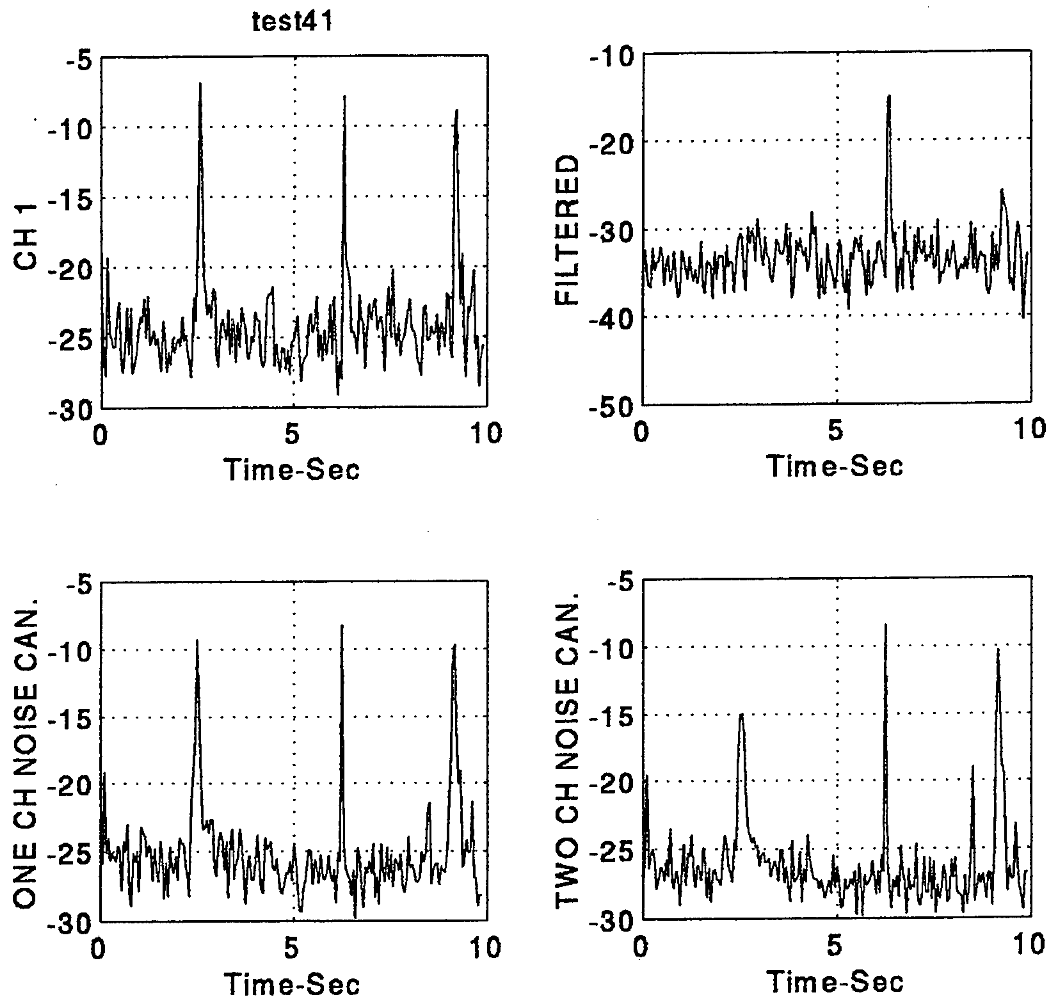

test37
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
Time-Sec
Time-Sec
Time-Sec

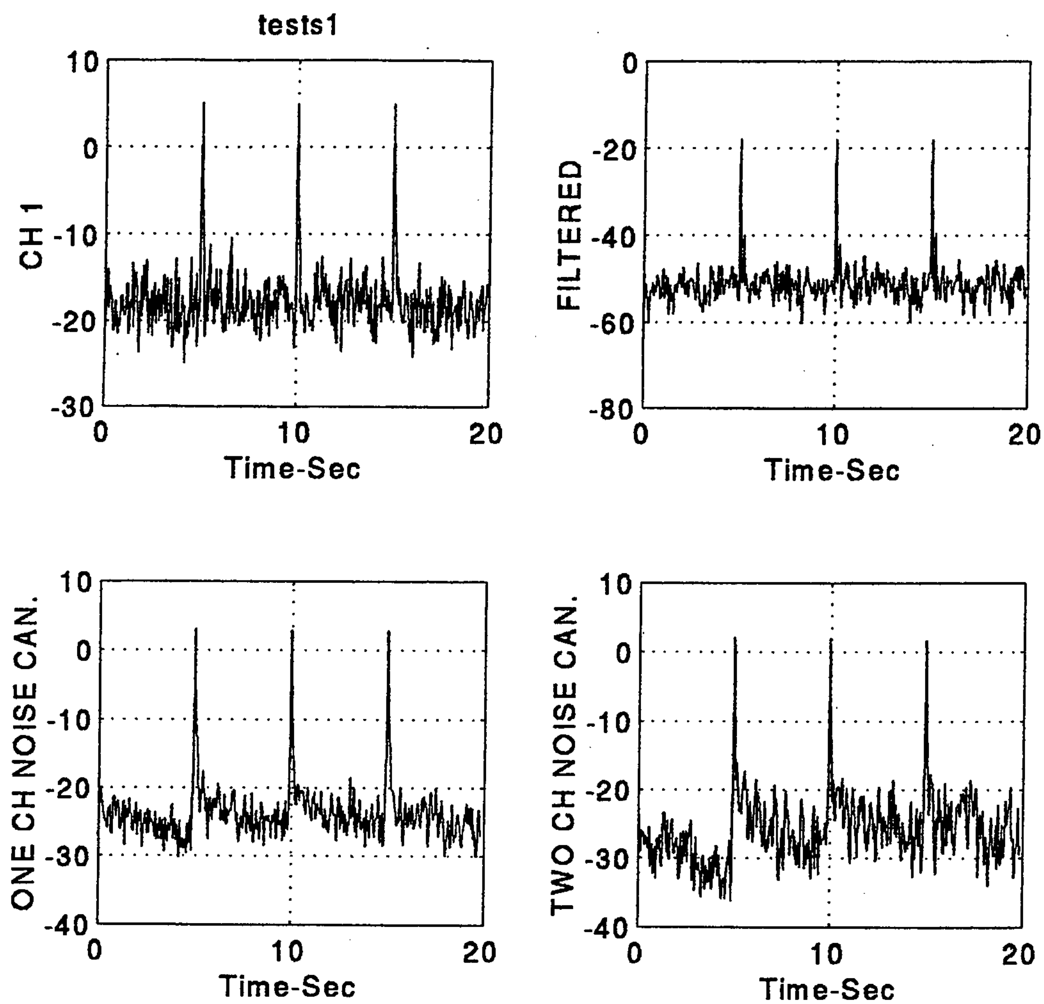


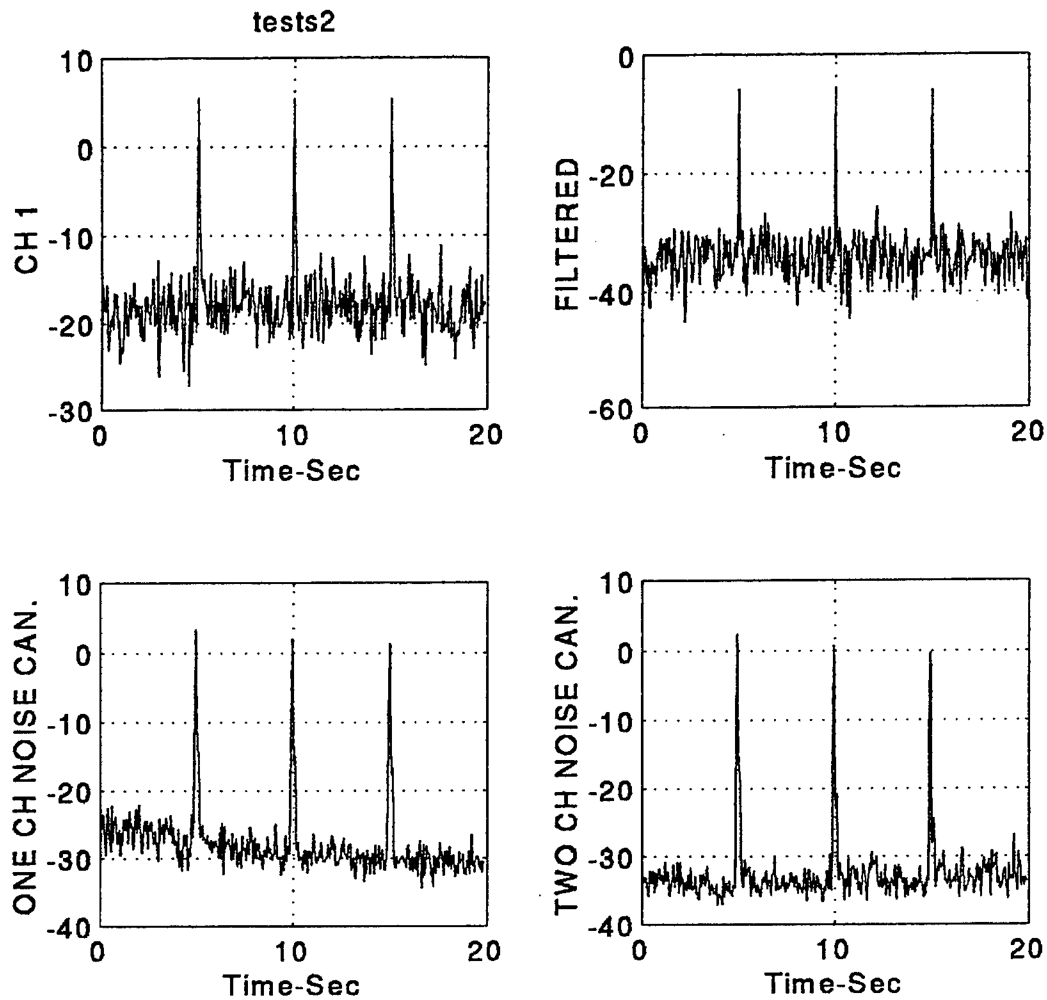
test38
CH 1
0
-10
-20
-30
-40
0
5
10
Time-Sec
FILTERED
0
-10
-20
-30
-40
-50
0
5
10
Time-Sec
ONE CH NOISE CAN.
-10
-15
-20
-25
-30
-35
0
5
10
Time-Sec
TWO CH NOISE CAN.
0
-10
-20
-30
-40
0
5
10
Time-Sec

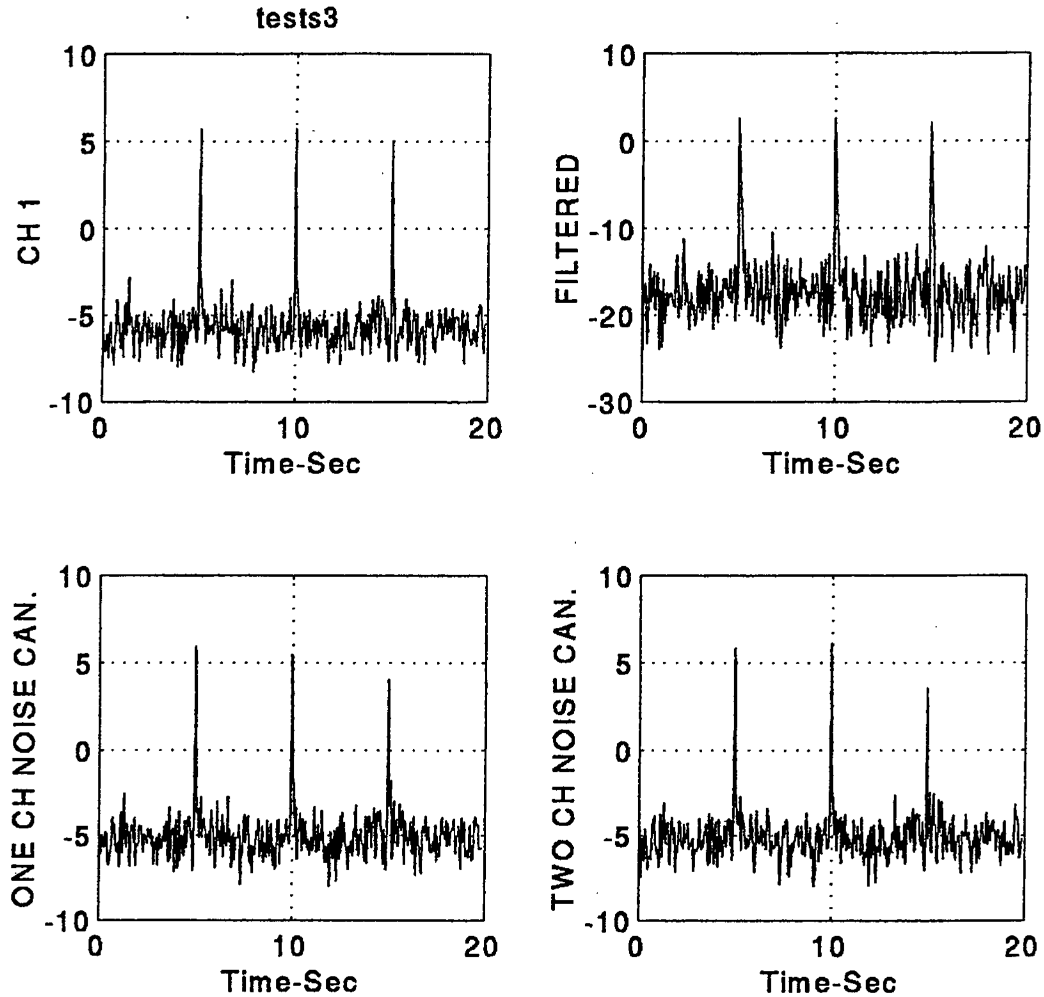

test39
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
Time-Sec
Time-Sec
Time-Sec
0
-10
-20
-30
-40
-50
-15
-25
-35
5
10

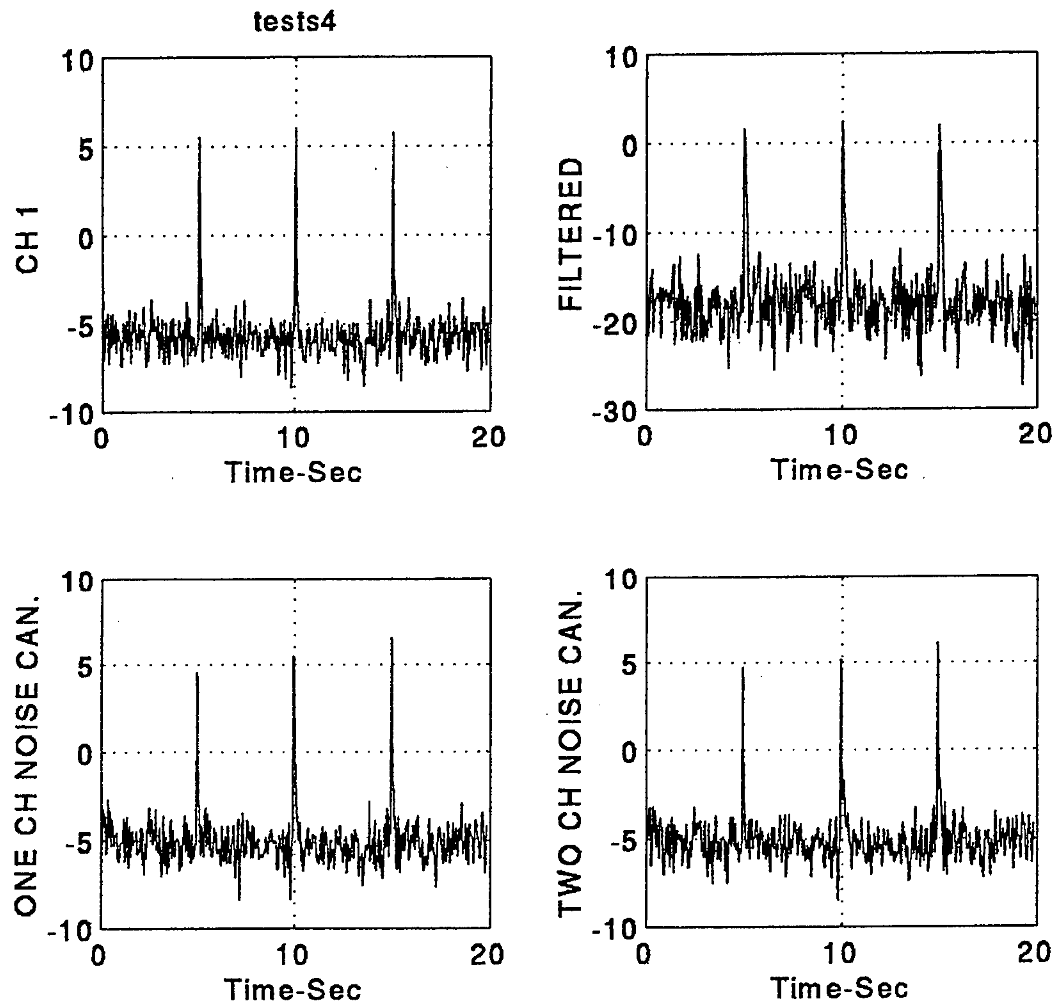

test40
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
Time-Sec
Time-Sec
Time-Sec
0
-10
-20
-30
-40
-50
5
10

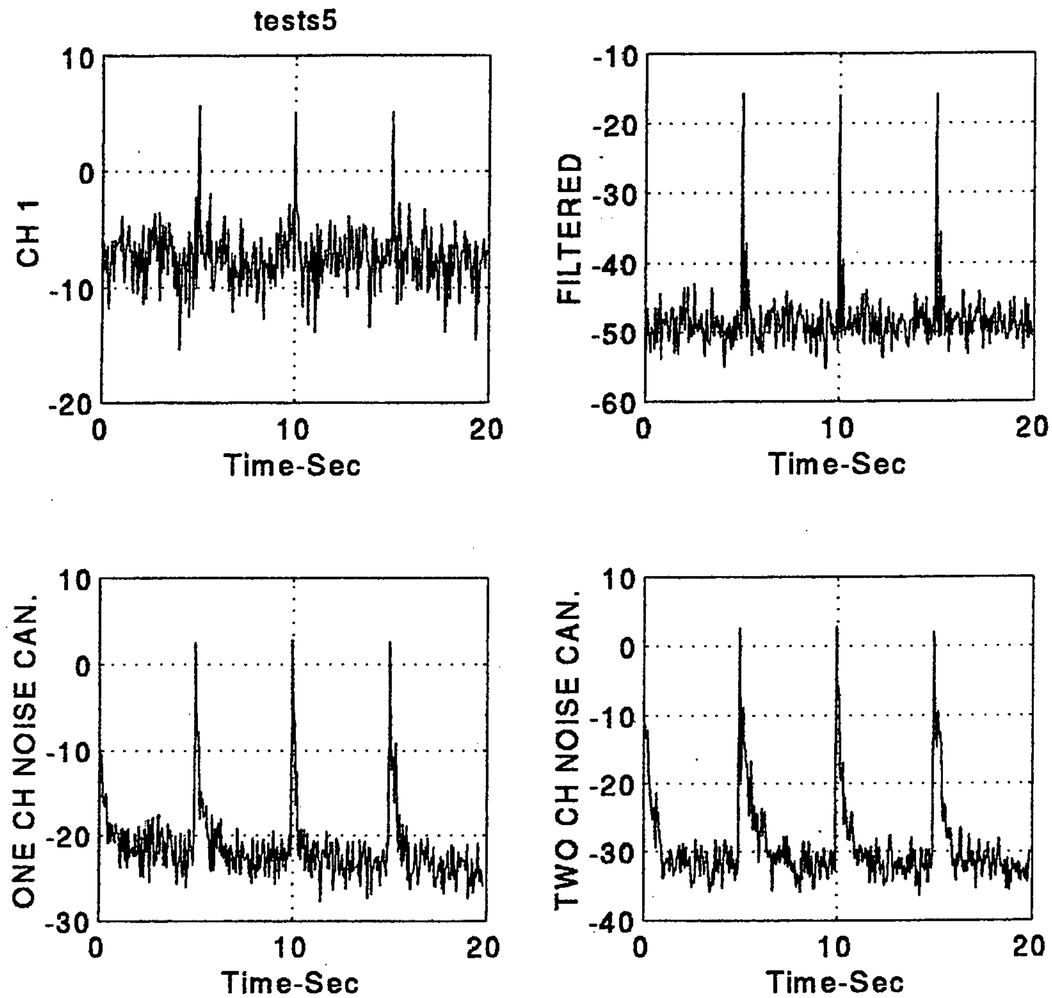

test41
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
-5
-10
-15
-20
-25
-30
-40
-50
0
5
10

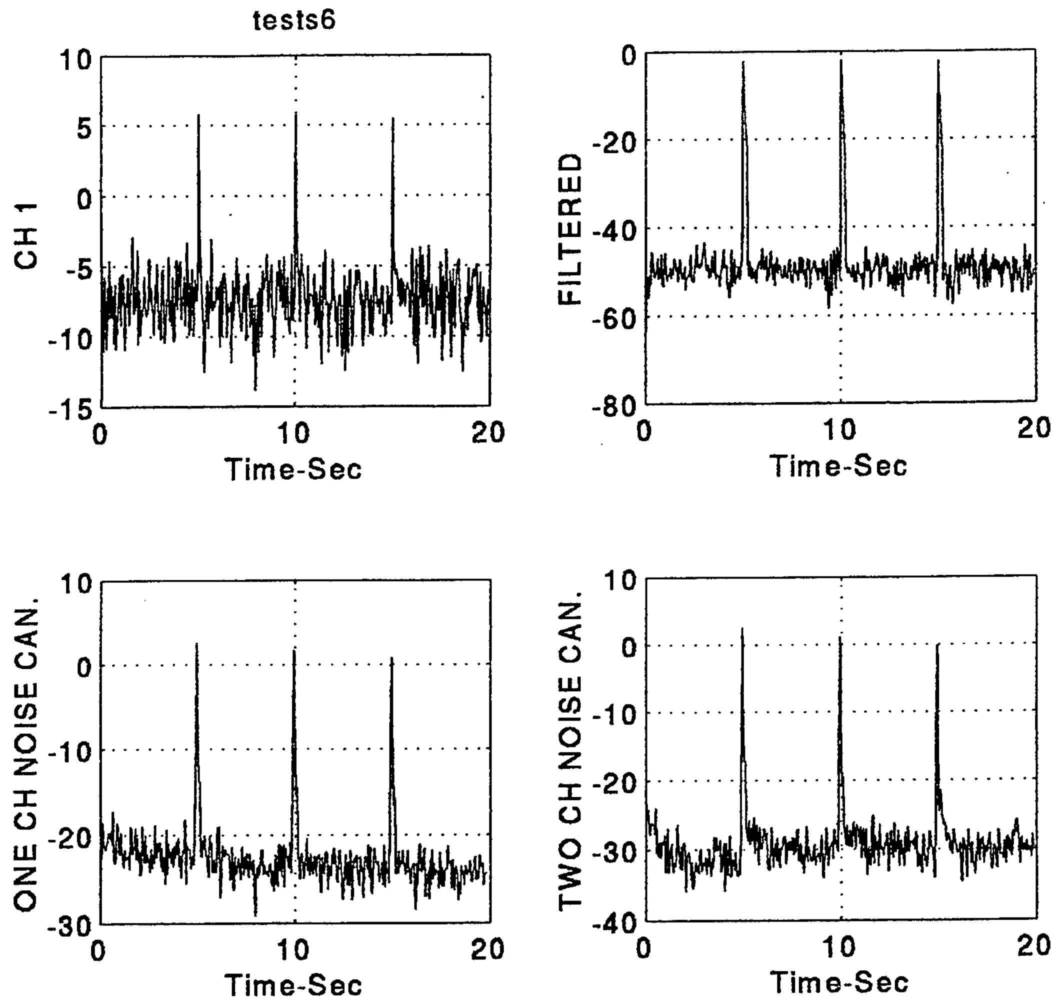

tests1
10
0
-10
-20
-30
CH 1
0
10
20
Time-Sec
0
-20
-40
-60
-80
FILTERED
0
10
20
Time-Sec
10
0
-10
-20
-30
-40
ONE CH NOISE CAN.
0
10
20
Time-Sec
10
0
-10
-20
-30
-40
TWO CH NOISE CAN.
0
10
20
Time-Sec

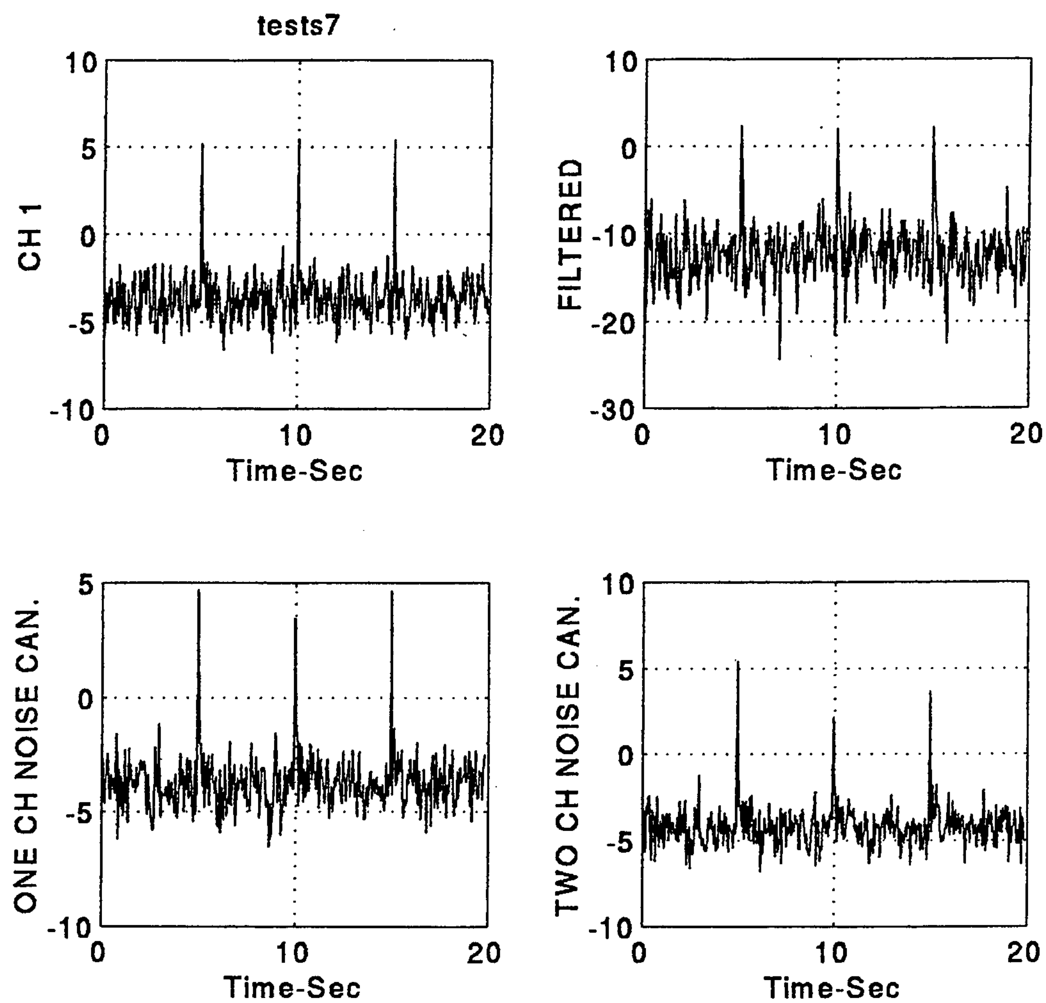

tests2
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
Time-Sec
Time-Sec
Time-Sec

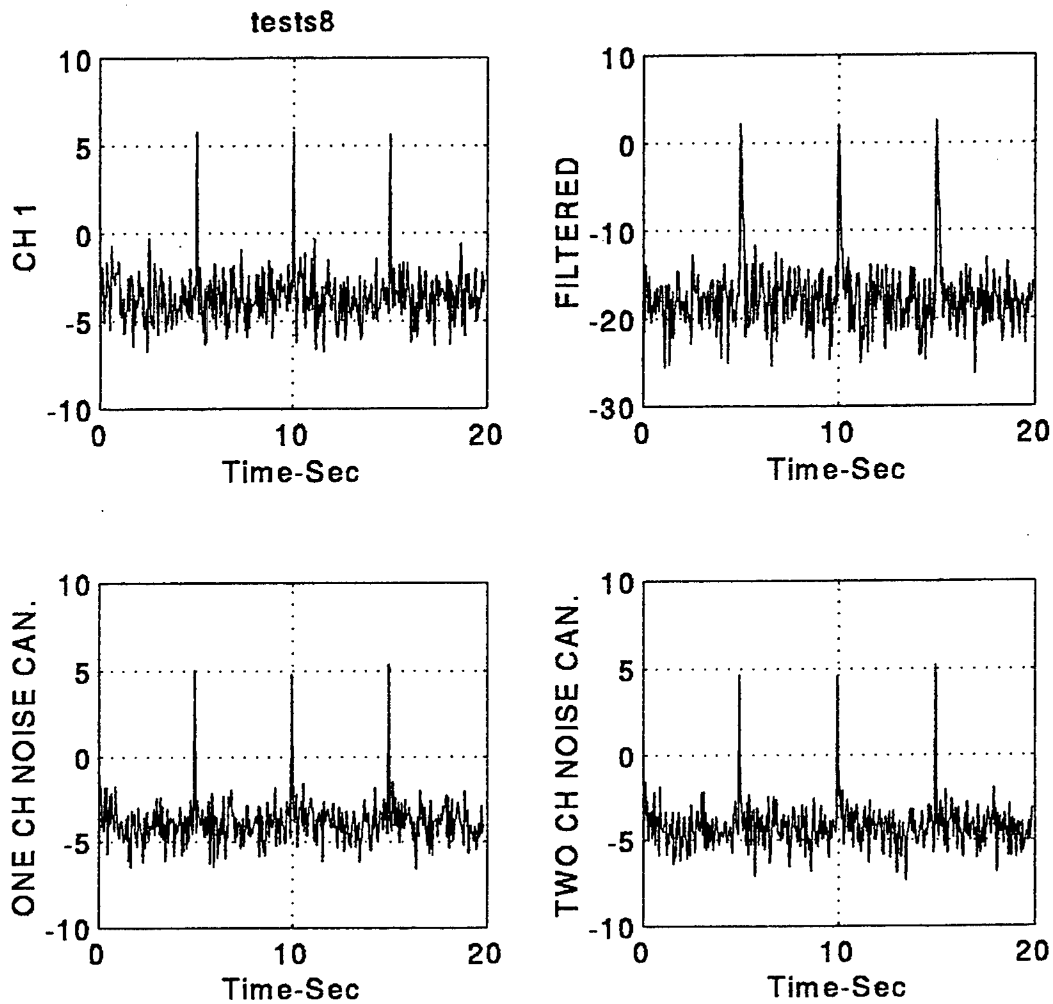

tests3
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
Time-Sec
Time-Sec
Time-Sec
10
5
0
-5
-10
0
-10
-20
-30
0
10
20


tests4
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
Time-Sec
Time-Sec
Time-Sec
10
5
0
-5
-10
-20
-30
20


tests5
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
10
0
-10
-20
-30
-40
-50
-60
20


tests6
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
Time-Sec
Time-Sec
Time-Sec


tests7
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
10
5
0
-5
-10
-20
-30
20


tests8
CH 1
FILTERED
ONE CH NOISE CAN.
TWO CH NOISE CAN.
Time-Sec
Time-Sec
Time-Sec
Time-Sec

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The problem of reducing background noise in seismic listening devices is of great importance to urban search and rescue teams using the Corps of Engineers STOLS system (System to Locate Survivors). Both single-channel and two-channel noise cancellation methods are compared with bandpass filtering methods. The results from both simulated data and field data are used to indicate the strengths and weaknesses of each method. Potential hardware implementations are presented.

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