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THESIS

DATA ACQUISITION, REDUCTION AND ANALYSIS
OF
SHOCK AND VIBRATION TESTING
USING A SUPER MICROCOMPUTER

by

Robert A. Shafer

June 1987

Thesis Advisor:

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prior to analysis. Thus the data can be analyzed within minutes of the completion of the experiment, giving immediate results at the testing site instead of an analysis facility. Another application of the data acquisition equipped microcomputer is in the analysis of the frequency response characteristics of structures. Programming the microcomputer to acquire the data from a vibrating structure in order to determine the dynamic signature is easily done. The Hilbert transform, identified previously as a means of detecting non-linearities in the frequency response, can also be included in the data analysis portion of the same program. Thus a complete package of signal analysis routines, including frequency response, power spectra, and system linearity, can be programmed into the microcomputer. Discussed in this study is the method utilized to validate a data acquisition equipped microcomputer for such applications.

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Data Acquisition, Reduction and Analysis
of
Shock and Vibration Testing
Using a Super Microcomputer

by

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Lieutenant, United States Navy
B.S.M.E., United States Naval Academy, 1981

Submitted in partial fulfillment of the
requirements for the degree of

MASTER OF SCIENCE IN MECHANICAL ENGINEERING

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June 1987

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ABSTRACT

The advent of small, portable super microcomputers has enabled the rapid analysis of mechanical signals. In addition to the microcomputer's reduction in cost, their capabilities have expanded to allow real time analysis of the frequency characteristics of vibrating structures. Data acquisition modules allow the addition of powerful components to a microcomputer. Modules consisting of the necessary Analog to Digital converters, Digital to Analog converters, and the hardware necessary to support sixteen channels of input data can easily fit inside desk top microcomputers. Fast 32 bit processors, small high volume hard disk drives and large RAM capacities complete the package, enabling a microcomputer to perform complex dynamic signal analysis. One application of the super microcomputer is in the field of underwater explosion shock testing. The combination of the acquisition and analysis capabilities in the same device eliminates the requirement to record the data to an intermediate device prior to analysis. Thus the data can be analyzed within minutes of the completion of an experiment, giving immediate results at the testing site instead of an analysis facility. Another application of the data acquisition equipped microcomputer is in the analysis of the frequency response characteristics of structures. Programming the microcomputer to acquire the data from a vibrating structure in order to determine the dynamic signature is easily done. The Hilbert transform, identified previously as a means of detecting non-linearities in the frequency response, can also be included in the data analysis portion of the same program. Thus a complete package of signal analysis routines, including frequency response, power spectra, and system linearity, can be programmed into the microcomputer. Discussed in this study is the method utilized to validate a data acquisition equipped microcomputer for such applications.

THESIS DISCLAIMER

The reader is advised that the computer codes presented in this research have not been tested under all possible operating conditions or the various operating systems for the MASSCOMP 5400 system. Effort has been made to run the programs on the most current operating system available and to the best of the researchers ability in the time available, to ensure that the programs are free of computational and logical errors. Any use of these programs without user verification is at the risk of the user.



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I. INTRODUCTION

A. BACKGROUND

The super microcomputer is a tool that enables the engineer or scientist to rapidly digest large quantities of information. Recent advances in computer technology have brought about a startling increase in the computing power of small "desktop" systems. The speed of the processor has significantly increased while the cost have dropped dramatically. Coupled with the ability to store large quantities of data in internal memory, the speed of computers have taken quantum leaps forward. Five years ago random access memory banks of 600 kilobytes (600K) were common for a desktop scientific computing system. Today systems with 10 Megabytes are readily available. Processor word size has increased from 8 and 16 bit word length to 32 bit in the same period. Data storage and retrieval systems have also become smaller and less expensive. Thus the ability to process information rapidly and store it has made micro and mini computer systems a valuable aid to the engineer.

One of the uses of the super microcomputer is the area of data acquisition. Experiments can be rapidly set up, the data can be collected and analyzed in near real time, and the results can be displayed in fraction of the time than was previously required. Evidence of the increase in computing power is evident in the increased ability of Analog To Digital conversion. Analog signals, frequently voltages from a set of remote sensing devices, can be automatically read and recorded. In 1973 the sampling rate of 20,000 samples per second was considered good for a microcomputing system [Ref. 1]. In 1986 the super microcomputer is capable of sampling 1,000,000 samples per second [Ref. 2]. The only limitation is the throughput capability of the data bus. This is a 50 fold increase in sampling rate and coupled with simultaneous sampling cards and multichannel A/D converters, super microcomputers are now able to analyze signals in wide band vibration experiments and shock analysis applications.

In the field of underwater explosion testing the engineer has been forced to record shock data to an intermediate device prior to analysis. This device, usually a wide band tape recorder, is a costly addition to the test equipment. If there was a small, portable device which could record the data and perform the analysis at the test site the engineer would have immediate results available. Not only would this save time, but it would

eliminate the need for the intermediate recording device. The properly equipped microcomputer would allow the engineer to gather the information, display the results, and record them for further analysis. This process would take seconds, where as the present alternative would take much longer.

In the study of the mechanical signature of structures, there has been a recent emphasis toward use of the Hilbert transform in order to determine if the frequency response of a structure is behaving in a non-linear manner. There are no dynamic signal analyzers available that include the capability of performing the Hilbert transform of the frequency response. In order to perform this operation the data would have to be transferred to a computer for further analysis. This is a costly and time consuming process. If a microcomputer with a data acquisition package could perform the dynamic signal analysis of mechanical signatures from a structure, then it could also perform the Hilbert transform of the frequency response of the structure. Again, this would result in a significant reduction in the time required for data analysis and include processes that are not available on the present generation of dynamic signal analyzers.

B. PURPOSE

The purpose of this study is to:

1. Program the properly equipped microcomputer to perform data acquisition, reduction, and analysis of underwater explosion shock data.
2. Perform dynamic signal analysis on mechanical signatures from structures, using both impact and steady state excitations.
3. Include in the dynamic signal analysis portion of the data analysis routines the Hilbert transform. This would place a complete signal analysis package in one device.

II. THEORY

A. DIGITAL SIGNAL PROCESSING FUNDAMENTALS

The Fourier transform is a method of calculating the frequency spectrum of a time signal. Fourier stated that any periodic time signal of fundamental frequency $\frac{1}{T_0}$ could be transformed into the summation of a series of sinusoidal signals of varying amplitude and frequency. The Fourier series of a periodic time signal of period T_0 is then:

$$x(t) = \frac{a_0}{2} + \sum_{n=1}^{\infty} \left\{ a_n \cos\left(\frac{2\pi n t}{T_0}\right) + b_n \sin\left(\frac{2\pi n t}{T_0}\right) \right\}. \quad [\text{Eqn. 1}]$$

The Fourier transform of a signal is a method of computing the frequency spectrum of a non-periodic infinite time signal, provided it satisfies Dirichlet's Conditions. It is defined as:

$$X(f) = \mathcal{F}\{x(t)\} = \int_{-\infty}^{\infty} x(t) e^{-j2\pi f t} dt. \quad [\text{Eqn. 2}]$$

The finite time signal is infinite in the frequency domain, the infinite time signal has a finite bandwidth of frequency components. The fast Fourier transform (FFT) is a recent mathematical technique allowing the rapid computation of the frequency components of a time signal. Programed into a microcomputer with the ability to convert a time signal into a set of digital samples, the FFT is a powerful tool that allows fast computations of a systems transfer function.

The process of digitizing an analog signal is to multiply it by a pulse train at a set sampling frequency. Given an analog signal of bandwidth F_{\max} , the Nyquist sampling frequency states that the sampling rate of the analog signal must be greater than or equal to $2 F_{\max}$ in order to avoid aliasing errors, thus:

$$F_s \geq 2 F_{\max}. \quad [\text{Eqn. 3}]$$

Multiplication in the time domain is equivalent to convolution in the frequency domain. This simply takes the frequency components of the signal and repeats them in the frequency domain at intervals of the sampling frequency. If the signal's bandwidth exceeds the sampling frequency then the higher frequency components overlap and sum to introduce errors in the frequency spectrum of the time signal. This is known as aliasing. Figures 1 through 3 illustrate the concept of aliasing and the Nyquist sampling rate.

When dealing with dynamic real time signals, the bandwidth of the signal may be quite large. In order to effectively sample the signal, an arbitrary frequency limit must be set by the programmer. Frequency components above this limit are usually not required for systems analysis and may be discarded prior to sampling. One of the easiest methods to do this is by an analog low pass filter. Setting the low pass filter at the desired frequency limit allows the higher frequency components of the signal to be ignored. Because the cut off frequency of a filter is defined as its half power or -3 db point, the sampling rate is usually somewhat greater than twice the cutoff frequency of the filter. This allows a reduced chance of aliasing the signal and optimum resolution in the frequency analysis.

Another method of avoiding aliasing is to sample the signal at three to four times its half power frequency component. This ensures that any aliasing that takes place is in the region of low power signals and that it does not affect the important frequency components of the signal. The major drawback to this method is that by sampling at a higher rate the portion time signal rate is decreased. This influences resolution of the spectrum.

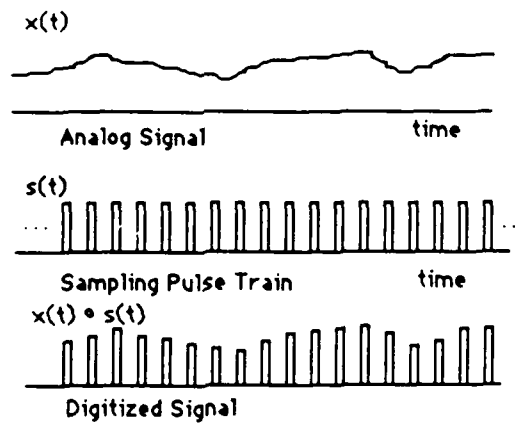


Figure 1: Sampled Analog Signal in the Time Domain

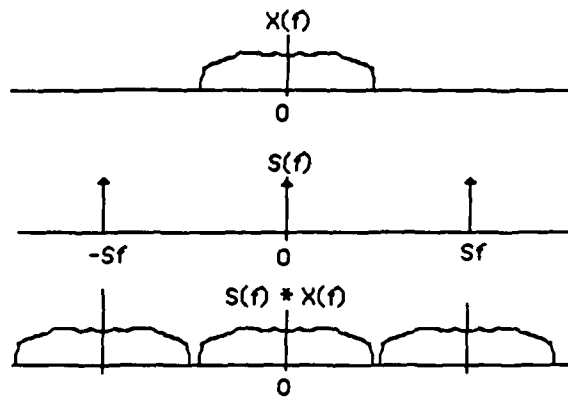


Figure 2: Correctly Sampled Signal in the Frequency Domain

Sampling Frequency less than Twice the Band Width

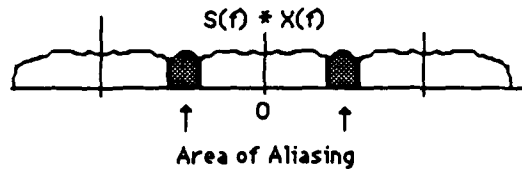


Figure 3: Signal Sampled at less than the Nyquist Sampling Frequency

When the signal is transformed into its frequency components via a discrete FFT, the minimum frequency resolution is inversely proportional to the time record length of the sampled signal. Thus if the sampling rate is F_s , and the time record consist of n samples, then the frequency resolution will be:

$$F_{\min} = \frac{1}{n \Delta t} = \frac{F_s}{n} \quad [\text{Eqn. 4}]$$

where n is the number of samples in the data set. The $n \Delta t$ term is the time record length of the time signal and is noted by T .

Thus, in order to achieve the best frequency resolution, the longest possible time record length is desired. To do this with a fixed number of samples, the slowest sampling frequency possible must be used. A filtered signal in conjunction with a slow sampling rate will yield non-aliased, high resolution frequency information.

Fast Fourier transform techniques assume that the time signal is periodic in nature. In order to meet this periodicity requirement, the time signal must be weighted, or windowed. This involves tapering the information by various methods without altering the frequency components of the signal. One of the primary windows used is the Hanning or cosine window. This multiplies the time signal by a sine wave of amplitude $\sqrt{2/3}$ and a period of $2T$. The time signal is then zero in the region of $t = 0$ and $t = T$. This introduces a frequency component at the frequency of half of the minimum frequency resolved by the Fourier transform method which will not affect the results obtained by the FFT. Thus the periodicity requirements are met and the frequency components of the time signal are not altered.

Important additional measurements performed on the mechanical signatures include the power spectra density function (or autospectra), the cross spectra density function, the system transfer function, and coherence. The input power spectra, $G_{xx}(\omega)$, is a measure of the energy content of the input signal at a frequency ω . Likewise, the output power spectra, $G_{yy}(\omega)$, is a measure of the energy content of the output signal. Both the input power spectra and output power spectra are real valued functions. The cross power spectra $G_{xy}(\omega)$, is a complex value,

$$G_{xy}(\omega) = G_{xyR}(\omega) + j G_{xyI}(\omega), \quad [\text{Eqn. 5}]$$

where $G_{xyR}(\omega)$ and $G_{xyI}(\omega)$ are the real and imaginary portions of the cross power spectra. The cross power spectra is a measure of the amount of energy transferred from the input point, x , to the output point, y , at a frequency ω .

The system transfer function, $G(\omega)$, is the ratio of the Fourier transform of the input and output time signals,

$$G(\omega) = \frac{\mathcal{F}\{x(t)\}}{\mathcal{F}\{y(t)\}} = \frac{X(\omega)}{Y(\omega)}. \quad [\text{Eqn. 6}]$$

It too is a complex number. It consist of

$$G(\omega) = G_R(\omega) + j G_I(\omega), \quad [\text{Eqn. 7}]$$

where $G_R(\omega)$ and $G_I(\omega)$ are the real and imaginary portions of the transfer function. One method of calculating the transfer function uses the auto spectra and cross spectra information.

$$|G(\omega)|^2 = \frac{G_{yy}(\omega)}{G_{xx}(\omega)},$$

and

$$\begin{aligned} G_R(\omega) &= \frac{G_{xyR}(\omega)}{G_{xx}(\omega)} \\ G_I(\omega) &= \frac{G_{xyI}(\omega)}{G_{xx}(\omega)}. \end{aligned} \quad [\text{Eqn. 8}]$$

The coherence, γ_{xy} , is a measure of the noise contamination of an output signal $y(t)$ from sources other than the input signal $x(t)$. The coherence function can be interpreted as the fractional portion of the output spectrum of $y(t)$ which is linearly due to $x(t)$ at frequency ω [Ref. 3]. A coherence of 1 indicates perfect transmission from the input point, x , to the output point, y . This indicates that there is no noise contamination. A coherence of 0 (zero) indicates severe contamination by noise at the output point. This indicates that none of the signal at y is due to excitation at point x . The coherence is calculated by

$$\gamma_{xy}^2(\omega) = \frac{|G_{xy}(\omega)|^2}{G_{xx}(\omega) G_{yy}(\omega)}. \quad [\text{Eqn. 9}]$$

B. THE HILBERT TRANSFORM AND THE FREQUENCY RESPONSE

The Hilbert transform has been identified as a means of determining the degree of linearity of a system's frequency response. It has been shown that accurate identification of a range of commonly occurring structural non-linearities such as friction, stiffness and power-law damping is possible in systems where the modes are well separated [Ref. 4]. By taking the Hilbert transform of a systems frequency response and comparing it to the original frequency response, the linearity of the system can be readily determined. For linear systems the Hilbert transform of the frequency response $H(\omega)$ will be identical to the frequency response of the system $G(\omega)$.

The Hilbert transform of a real-valued function $g(t)$ is defined by:

$$h(t) = \frac{1}{\pi} \int_{-\infty}^{\infty} \frac{g(\tau)}{t-\tau} d\tau . \quad [\text{Eqn. 10}]$$

In the frequency domain it is defined by:

$$H(\omega) = \text{PV} \int_{-\infty}^{\infty} \frac{G(\xi)}{\omega-\xi} d\xi, \quad [\text{Eqn. 11}]$$

where $H(\omega)$ is the Hilbert transform of the complex valued $G(\xi)$ and PV is defined as the Cauchy Principle Value for the integral.

$$G(\xi) = G_R(\xi) + j G_I(\xi) \quad [\text{Eqn. 12}]$$

where $G_R(\xi)$ and $G_I(\xi)$ are the real and imaginary parts of the $G(\xi)$. It is important to note that the Hilbert transform is in the same domain as the original function. Thus the Hilbert transform of a real valued time function is in the time domain, and the Hilbert transform of a function in the frequency domain is in the frequency domain. The Hilbert transform $H(\omega)$ is complex-valued in the frequency domain,

$$H(\omega) = H_R(\omega) + j H_I(\omega), \quad [\text{Eqn. 13}]$$

where $H_R(\omega)$ and $H_I(\omega)$ are the real and imaginary parts of $H(\omega)$.

In order to avoid the singularity point at $\xi = \omega$, the Hilbert transform the integration must be divided into two portions with limits as follows:

$$H(\omega) = \text{PV} \int_{-\infty}^{\infty} \frac{G(\xi)}{\omega-\xi} d\xi = \lim_{\epsilon \rightarrow 0} \int_{-\infty}^{\omega-\epsilon} \frac{G(\xi)}{\omega-\xi} d\xi + \int_{\omega+\epsilon}^{\infty} \frac{G(\xi)}{\omega-\xi} d\xi . \quad [\text{Eqn. 14}]$$

Because $G_R(\xi)$ is an even function and $G_I(\xi)$ an odd function,

$$G(-\xi) = G^*(\xi), \quad [\text{Eqn. 15}]$$

where * denotes the complex conjugate of the complex valued function $G(\xi)$, Equation 14 must be further subdivided into its real and imaginary parts:

$$H_R(\omega) = -\frac{2}{\pi} \text{PV} \int_0^{\infty} \frac{\xi G_I(\xi)}{\xi^2 - \omega^2} d\xi$$

$$H_I(\omega) = -\frac{2\omega}{\pi} \text{PV} \int_0^{\infty} \frac{G_R(\xi)}{\xi^2 - \omega^2} d\xi \quad [\text{Eqn. 16}]$$

One method utilized to calculate the Hilbert transform involves a numerical integration. Due to the finite frequency resolution of a discrete Fourier transform, a numerical integration of the frequency response will always yield inaccuracies at the singularity point $\omega = \xi$. The equations for evaluating the Hilbert transform numerically are given by [Ref. 5]:

$$H_R(\omega_j) = \frac{2}{\pi} \sum_{k=1}^n \frac{G_I(\omega_k) \omega_k \Delta \omega}{\omega_k^2 - \omega_j^2} + E_j^R$$

$$H_I(\omega_j) = \frac{2}{\pi} \omega_j \sum_{k=1}^n \frac{G_R(\omega_k) \Delta \omega}{\omega_k^2 - \omega_j^2} + E_j^I \quad [\text{Eqn. 17}]$$

where $H(\omega)$ denotes the Hilbert transform and $G(\omega)$ is the system's transfer function. The correction terms, E_j^R and E_j^I , are dependent upon the system's frequency response and the range of the frequency response under examination.

A correction term is needed to correct for resonant peaks outside the range under examination, and others to correct for the truncation of the frequency response information at the limits of the frequency band under examination. Corrections for multimode systems are usually limited to a shift of the imaginary portion of the Hilbert transform. The estimate of the shift necessary to correct this is given by [Ref. 6]:

$$C_{I_s} = \frac{j \omega X_s}{\omega_s^2 - \omega} \quad [\text{Eqn. 18}]$$

where C_{I_s} is the correction to the imaginary portion of the Hilbert transform due to the s^{th} resonant frequency, ω_s . X_s is the the imaginary portion of the modal constant for the s^{th} resonant frequency. An estimate of the value of X_s may be evaluated from the the fact that:

$$H_I(\omega_s) = \frac{X_s}{\delta_s \omega_s} \quad [\text{Eqn. 19}]$$

thus

$$X_s = H_I(\omega_s) \delta_s \omega_s. \quad [\text{Eqn. 20}]$$

δ_s , the structural damping ratio, may be estimated by using the half power bandwidth of the resonant frequencies outside the band under investigation. The value of $H_I(\omega_s)$ may be taken directly from the imaginary portion of the transfer function. Thus as the resonant frequencies outside the band under investigation, ω_s , get larger, their influence of it correction term C_{I_s} diminishes. For structures with many resonant modes, only those modes near the frequency band under investigation need be used as correction terms.

A second method of evaluating the Hilbert transform applied in this study is the use of the Fourier transform to perform the Hilbert transform. The integral

$$\frac{1}{\pi} \int_{-\infty}^{\infty} \frac{G(\xi)}{\omega - \xi} d\xi \quad [\text{Eqn. 21}]$$

is the convolution integral in the frequency domain. Since convolution in the frequency domain is equivalent to multiplication in the time domain, this is equivalent to taking the Fourier transform of $G(\omega)$ and multiplying it by the Fourier transform of $\frac{1}{\pi \omega}$.

The Fourier transform of $\frac{1}{\pi \omega}$ is:

$$\mathcal{F} \left\{ \frac{1}{\pi \omega} \right\} = -j \operatorname{sgn}(t) = \begin{cases} -j & t > 0 \\ 0 & t = 0 \\ j & t < 0 \end{cases}, \quad [\text{Eqn. 22}]$$

The Fourier transform of $G(\xi)$ is:

$$\mathcal{F}\{G(\xi)\} = g(t), \quad [\text{Eqn. 23}]$$

where both $G(\xi)$ and $g(t)$ are complex valued. Multiplying the Fourier transform of the transfer function, $G(\xi)$, by the Fourier transform of $\frac{1}{\pi \omega}$ yields:

$$\mathcal{F} \left\{ \frac{1}{\pi} \int_{-\infty}^{\infty} \frac{G(\xi)}{\omega - \xi} d\xi \right\} = \begin{cases} -j g(t) & t > 0 \\ 0 & t = 0 \\ j g(t) & t < 0 \end{cases} \quad [\text{Eqn. 24}]$$

Multiplying by j yields the j times the Fourier transform of the Hilbert transform:

$$\mathcal{F} \{ jH(\omega) \} = j \mathcal{F} \left\{ \frac{1}{\pi} \int_{-\infty}^{\infty} \frac{G(\xi)}{\omega - \xi} d\xi \right\} = \begin{cases} g(t) & t > 0 \\ 0 & t = 0 \\ -g(t) & t < 0 \end{cases} \quad [\text{Eqn. 25}]$$

Applying the property of linearity of the Fourier transform and adding the Fourier transform of the transfer function $G(\xi)$ to j times the Fourier transform of the Hilbert transform yields:

$$\mathcal{F} \{ G(\omega) + j H(\omega) \} = \begin{cases} 2g(t) & t > 0 \\ g(t) & t = 0 \\ 0 & t < 0 \end{cases} \quad [\text{Eqn. 26}]$$

and

$$G(\omega) + j H(\omega) = \mathcal{F}^{-1} \left\{ g(t) \circ \begin{pmatrix} 2 & t > 0 \\ 1 & t = 0 \\ 0 & t < 0 \end{pmatrix} \right\} \quad [\text{Eqn. 27}]$$

There exist many fast Fourier transform routines for complex vectors in computer libraries. Thus an easier method to calculate the Hilbert transform which avoids the numerical integration required of the Cauchy Method is to take the Fourier transform of a

real signal $G(\omega)$ and multiply it by $\begin{cases} 2 & t > 0 \\ 1 & t = 0 \\ 0 & t < 0 \end{cases}$. Then by taking the inverse Fourier transform of this modified step response, the transfer function $G(\omega)$ and its Hilbert transform $H(\omega)$ can be separated out [Ref 7]. Figure 4 summarizes the method used to calculate the Hilbert transform using the Fourier transform, Equations 21 through 27.

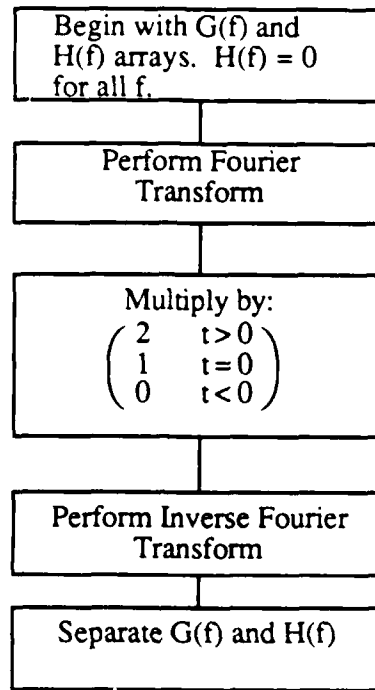


Figure 4: Fast Hilbert Transform Flow Diagram

When using the FFT to calculate the Hilbert transform a FFT routine capable of performing inverse Fourier transforms must be used. Because the FFT routine assumes convolution in the frequency domain has taken place, the Fourier transform is symmetric about the point $\frac{T}{2}$, where T denotes the time record length of the input signal. The inverse Fourier transform also assumes that convolution has taken place. An equivalent method of setting all negative frequency component to zero is by setting all components above $\frac{T}{2}$ equal to zero. All component from $f > 0$ to $\frac{T}{2}$ are multiplied by 2. The inverse Fourier transform

is then performed yielding the original function and its Hilbert transform. This method is faster and just as accurate as the numerical integration method for systems when applied properly. Correction terms are still required when using the FFT method of calculating the Hilbert transform.

III. EXPERIMENTAL PROCEDURE

A. EQUIPMENT UTILIZED

1. The MASSCOMP 5400

The MASSCOMP 5400 computer used to acquire the data and process it for the experimental work was configured as follows:

- 1 71 Mbyte hard disc drive
- 1 300 kbyte "floppy" disc drive
- 6 Mbyte Random Access Memory
- 1 16 channel, 1 MHz maximum aggregate sampling rate 12 bit resolution Analog to Digital Converter [AD12FA]
- 1 16 channel, 333 kHz maximum aggregate sampling rate 12 bit resolution Analog to Digital Converter [EF12M]
- 5 Timing Clocks, maximum frequency 6 MHz
- 1 (2) channel Digital to Analog Converter [EF12M]
- 1 16 channel sample and hold temporary storage card [SH16A]
- Various display and output devices

The capabilities of the various data acquisition devices installed in the computer are summarized in Appendix A.

The MASSCOMP 5400 is a Motorola 68020 based 32 bit word length system. It is run on the UNIX operating system. Language capabilities include "C" and FORTRAN. Software use to operate the system includes a graphics presentation package, Dynamic Signal Analysis Subroutines, Window Presentation Packages and the software necessary to operate the Data Acquisition devices.

The data analysis routines, except where noted, were copied from those provided by the MASSCOMP Dynamic Signal Analysis Package (SP-60). These were originally copied from the collection of Programs for Digital Signal Processing from the IEEE Acoustics, Speech and Signal Processing Society.

The programming necessary to collect and process the data was written in FORTRAN. All programs referred to in the remainder of this paper are documented in the appendices. It should be noted that the software packages for the data acquisition and presentation portions include subroutine calls that are unique to the MASSCOMP system and are written specifically to support MASSCOMP hardware. Further investigation into

the software documentation provided by MASSCOMP is warranted for a complete understanding of the data collection and presentation routines.

In addition to the MASSCOMP specific equipment, various recording devices, dynamic signal analyzers and signal generators were used to provide the necessary data and a means of verification of the accuracy and applicability of the MASSCOMP 5400 system.

2. Analog to Digital Converters

Analog to Digital converters work in several different manners. While the method used to digitize a signal is not important, the ultimate goal of a converter is to compare a sampled voltage and to digitize it to the nearest voltage value within its dynamic range. A twelve bit A/D converter is able to separate its dynamic range into 2^{12} individual levels. Thus a converter with a 20 volt dynamic range is able to obtain a resolution of $\frac{20}{4096}$ or 0.00488 volts. A 12 bit converter with only a 10 volt range will have twice this resolution. The speed of the conversion is dependent upon the method of digitization; the fastest devices having speed on the order of 20 nanoseconds and the slower ones of 1.2 microseconds [Ref. 8 p. 97].

The transfer rate of the data to a usable state is dependent upon the programming of the converter. There are three methods available to the programmer for data acquisition. The first is reading the information directly into an array for later use in the program. This is the easiest and usually the quickest. If the data is read into dynamic memory then the converter can be run at its maximum speed or at the data bus' speed whichever is less. The second method is a direct disc transfer. This is usually slower than the array transfer method. If the information is to be read into a hard disc for later retrieval and reduction then the programmer is usually limited to the transfer rate capability of the particular hard disc system available. One drawback of the direct to disc transfer is that the programmer has no access to the information until after the transfer is complete. The third and more complicated method is to read the data into memory buffers. The advantage of this method is that the data can be manipulated as the buffers are filled rather than having to wait until the final array or disc transfer is complete. Most micro computers allow a programmer to select any one of the three possibilities depending upon the application of the data.

Once the data was transferred to the hard disc in either the raw or reduced form it could be transferred to magnetic tape or floppy disc. This allows the volume of the information on the hard disc to be substantially reduced after the acquisition and analysis is

completed. If the data is required at a later date it is available for transfer from the permanent recordings (floppy disc or tape).

3. Digital to Analog Converters

Digital to Analog converters work in somewhat the same manner of the A/D converter, only their task is simpler. Because the goal is to send out an analog voltage, there is no need to compare the desired value to the output signal. Because the D/A converter is essentially a resistor capacitor circuit, the settling time requirement of the RC circuit, and thus the D/A converter determines its speed capability. Settling times are dependent upon the exact circuitry and the resolution of the converter. Eight bit converters may have settling times of 0.1 microseconds, while sixteen bit converters may have settling times of 75 microseconds [Ref. 8 p. 95]. The dynamic range of D/A converters are comparable to that of A/D converters. The programmer may also use the same three methods of transfer (array, disc and buffer) for digital to analog conversions.

Analog to Digital converters and Digital to Analog converters require timing clock signals in order to determine the frequency of sampling or output. Programmable clock chips are used to provide voltage pulse trains for the devices. Thus the programmer is able to set the sampling rate of an A/D device and the output speed of a D/A device. With clock chips capable of providing 6 MHz pulse trains, the programmer is limited only by the time characteristics of his conversion devices and the computers systems data bus capacity.

4. Verification Procedure

The process of determining whether the MASSCOMP 5400 system was suitable for the near real time processing of shock and vibration signals was completed in a series of steps. First, using analog data pre-recorded on magnetic tape from underwater explosions, the analog converters were tested for accuracy and speed. The MASSCOMP sampled, recorded, analyzed and displayed the information. Comparisons to the data analyzed by other means were used to verify the accuracy of the acquired data. Once validation of sampling speed and analysis of selected data sets was accomplished all data sets were transferred to the hard disc for archival purposes.

The second method of systems verification was completed by impact testing of an aluminum plate. Data analysis was expanded to include power spectral analysis and coherence measurements. Again the results were compared to those obtained by an separate HP 3562A Dynamic Signal Analyzer in order to verify the accuracy of the MASSCOMP system.

5. Application of the MASSCOMP 5400 System

Once the system was determined to operate satisfactorily, it was used on the vibrational analysis of a linear response system. In addition to the frequency response, power spectral density and coherence measurements, the Hilbert transform was used to examine the degree of linearity of the frequency response. This enabled a rapid identification of any nonlinearities in the system transfer function. The two methods of computing the Hilbert transform, numerical integration and use of the Fourier transform, were compared for accuracy. Once the Hilbert transform method was chosen, the frequency response characteristics of a non-linear system were evaluated to ensure the Hilbert transform identified the non-linearities.

B. SHOCK AND VIBRATION ANALYSIS

1. Analysis of Prerecorded Underwater Explosion Data

The analysis of prerecorded underwater explosion data involved the transfer of pressure and strain gage records from a Honeywell Model 101 sixteen channel FM tape recorder to the MASSCOMP system. Data was recorded at a tape speed of 120 inches per second. The AD12FA Analog To Digital converter card was used for the transfer of data. Figure 5 shows the schematic diagram of the connections used for the data transfer.

The bandwidth capability of the tape recorder was 250 kHz. The bandwidth of the pressure transducer data was 250 kHz, while the strain gage data was only 10 kHz. Location of data on the tape was noted by a trigger signal on one of the sixteen data channels. During the explosive testing procedure, an electrical wire with a DC voltage applied to it was wrapped around the explosive charge. As the charge was detonated, the cable was severed and the voltage sensed at the recorder dropped to zero. Monitoring the trigger channel for this drop in the voltage enabled the program to start the clock module. When the clock began sending pulses to the Analog To Digital converter, information was transferred off the tape into the data storage arrays in the computer program.

The Analog to Digital Converter section of the EF12M multifunction card is limited to a bandwidth of only 160 kHz. This required the use of the AD12FA Analog To Digital converter. This allowed sampling of the recorded data at a maximum frequency of 1.0 MHz, well beyond the bandwidth of the original signal. Figure 6 is the pressure transducer data set used as a reference to compare the data recorded by the A/D system. This was recorded by a HP5451C computer system at a sampling rate of 500 kHz. Data

was acquired on the MASSCOMP system at various sampling rates as shown in Table 1 and then compared to the reference record. The program utilized to acquire the data is shown in Appendix B.

TABLE 1:
SUMMARY OF SAMPLING RATES USED TO EXAMINE UNDERWATER
EXPLOSION DATA

Tape Speed (ips)	Sampling Rate
120	1 MHz
120	750 kHz
120	500 kHz
120	250 kHz

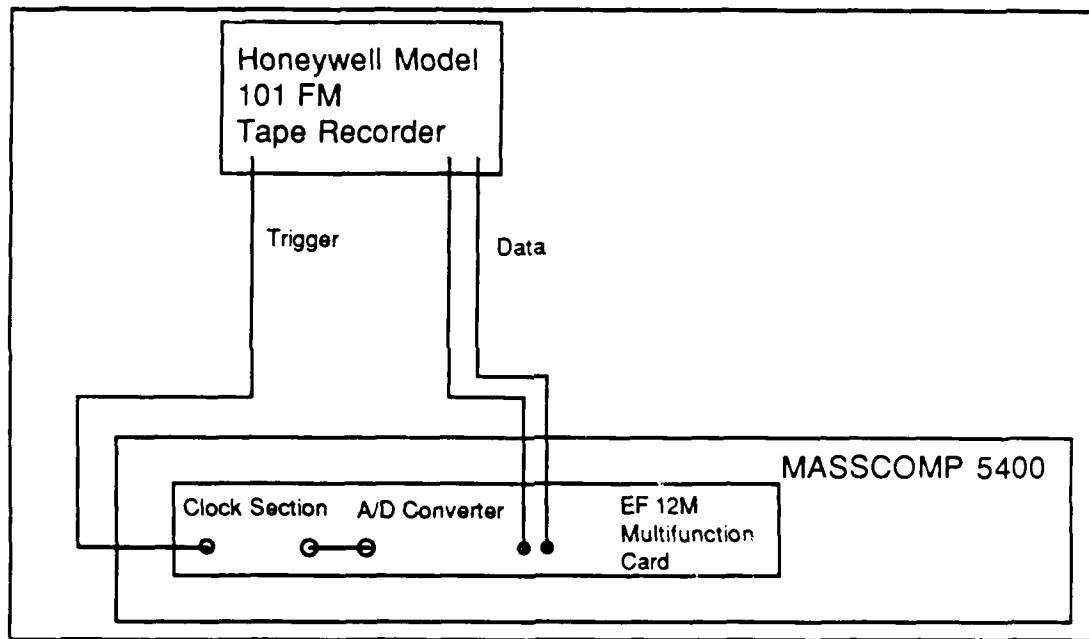


Figure 5: Schematic Diagram of Connections Used for Underwater Explosion Data Transfer

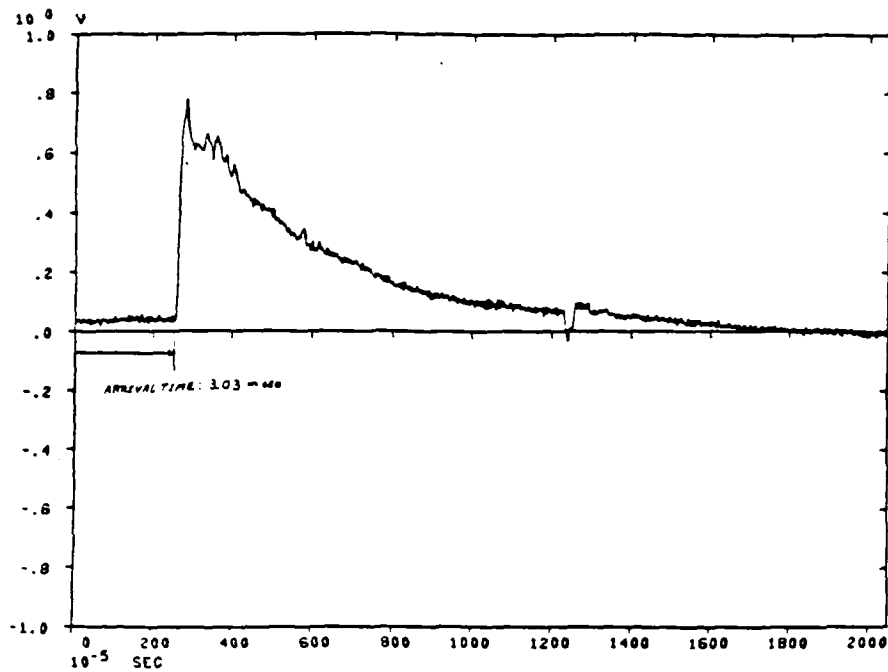


Figure 6: Reference Pressure Record Used as the Standard

Figures 7 through 10 are the data recordings from the same pressure pulse recorded at the sampling rates listed above. Comparisons of these Figures to the base line record revealed that the 500 kHz sampling rate matched the reference ideally. This was as predicted by the Nyquist sampling theorem. Recordings at rates less than the Nyquist frequency missed portions of the data that was available on the tape. Rates faster than the Nyquist frequency exceeded the bandwidth of the tape recorder and the reference record and subsequently appeared "noisy". Because there was no method to determine if this noise was actually recorded information or noise produced by the tape recording and play back process, the sampling rate for all further analysis of the prerecorded underwater explosion data was fixed at 500 kHz. The strain gage data was then acquired from the Honeywell tape recorder and transferred to the MASSCOMP hard disc.

The data was transferred to the the MASSCOMP system one track at a time. In order to utilize the system, as it is presently configured, for real time data acquisition in the field the sixteen channel sample and hold card (SH16F) and the single function Analog to Digital Converter (AD12FA) must be used. The only limitation is the data transfer bus, which is limited to 2 Megabytes per second. This allows one million sixteen bit words per

second to be gathered in real time. Table 2 summarizes the bandwidth limitations based upon the number of sampled channels.

TABLE 2: BANDWIDTH LIMITATIONS FOR MULTIPLE DATA CHANNELS

Number of Channels Sampled	Bandwidth Limit
1	500 kHz
2	250 kHz
4	125 kHz
8	62.5 kHz
16	31.5 kHz

Thus it would be impossible to record more than two pressure pulses in real time at a 500 kHz sampling rate. It would be possible to record up to sixteen strain gage traces at a 20 kHz sampling rate on the AD12FA A/D. The present system configuration limits the data acquisition to one type of data, either pressure or strain gage. The single sample and hold card can only be clocked at one frequency, limiting the choice to one type of data. A second sample and hold card, connected to the EF12M multifunction module, would allow the simultaneous sampling of both pressure and strain gage information.

In addition, to utilize the system for live data recording the program listed in Appendix B would have to be modified to include the use of the sample and hold card. This would necessitate a modification to the identification of the channels to be sampled and the clocking method used. Due to the increase in the volume of data, the storage arrays would have to be modified to allow up to sixteen channels of data. This would create the need for large arrays (16 [one for each channel] by 20,000 [one second of data at 20 kHz sampling rate]) for the raw data. Another method would be to read the information directly to the disc and then process it after the data is on the disc. This would still allow processing of the data in the field and allow analysis within minutes of the explosion.

Time History

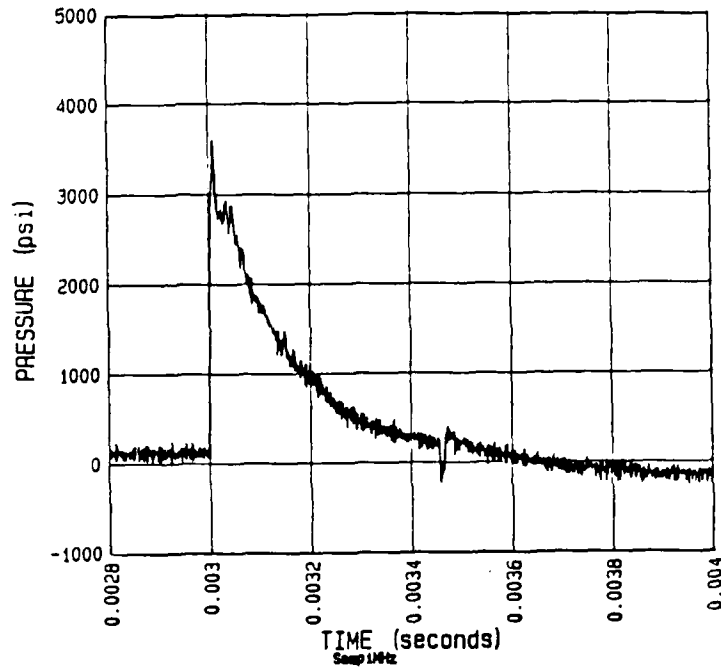


Figure 7: Pressure Recording at 1 MHz Sampling Rate

Time History

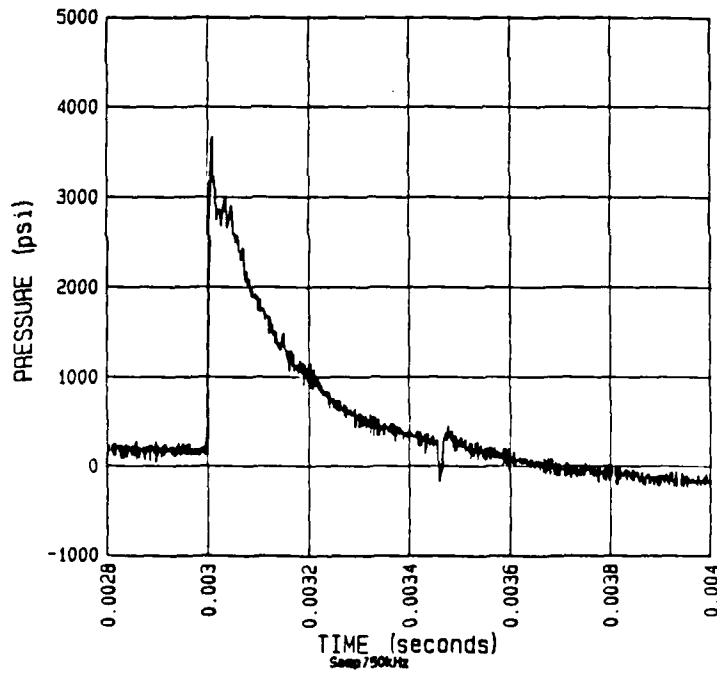


Figure 8: Pressure Recording at 750 kHz Sampling Rate

Time History

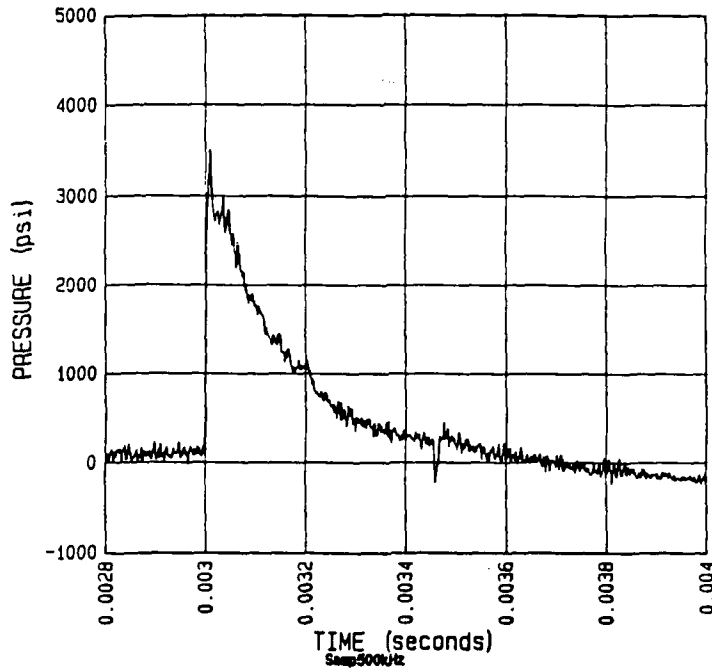


Figure 9: Pressure Recording at 500 kHz Sampling Rate

Time History

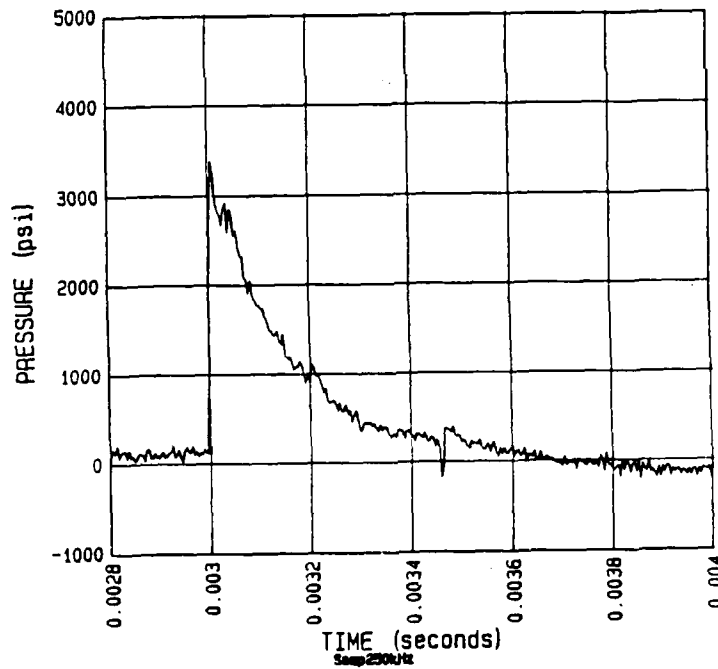


Figure 10: Pressure Recording at 250 kHz Sampling Rate

2. Impact Testing of an Aluminum Plate

Using the equipment schematic diagram shown in Figure 11, the digital signal analysis portion of the data acquisition system was tested. A modally tuned impact hammer with a bandwidth of 1,200 Hz was used to strike the aluminium plate. A force transducer located in the hammer provided a voltage signal to one channel of the A/D converter. An accelerometer located on the plate provided the plate acceleration data to a second channel of the A/D converter.

The data was sampled at a rate of 5,000 Hz per channel providing a 2,500 Hz bandwidth for the sampled signal. Because the energy level of the input signal was negligible beyond 2,500 Hz, a low pass filter was not required. Data blocks of 1024 samples were used yielding a frequency resolution of 4.88 Hz. This ensured that an adequate number of data points were in the frequency band width of the hammer. The program utilized to acquire the data is shown in Appendix C.

The time signals for the hammer and plate were recorded and averaged in the frequency domain. The frequency information was derived utilizing the FFT routine FFT842 [Ref. 9], while the power spectral information and coherence measurements were derived by utilizing the CCSE routine [Ref. 10]. Both of these routines were modified to suit the particular characteristics of the impact testing procedure and the microcomputing system.

The dynamic signal analysis was first performed on a single data record. The coherence, transfer function and frequency components of the signals were computed. As expected the coherence for a single impact was 1 throughout the frequency band under examination. The identical test was recorded using a HP3562A Dynamic Signal Analyzer for comparison. The analysis performed utilizing the super microcomputer yielded comparable results. Figures 12 through 14 display the recorded data for this single impact test from both sources.

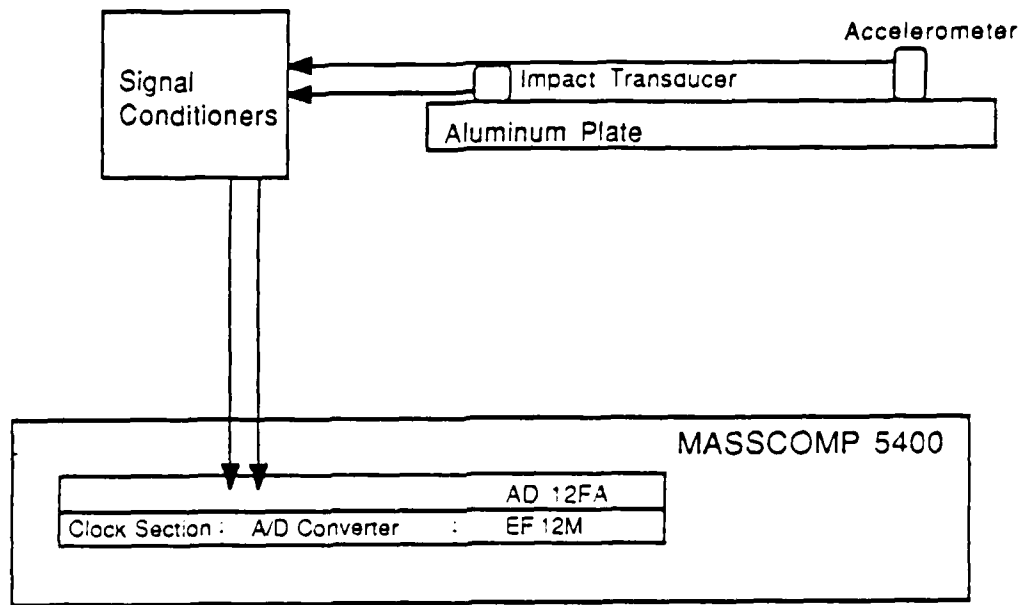


Figure 11: Experimental Set Up for Impact Testing of Aluminum Plate

The impact testing was then averaged to determine the influence of averaging on the transfer function. Figures 15 through 18 are the data recordings for sixteen impacts. As expected the frequency components of both the hammer and accelerometer data smoothed considerably when averaged. The coherence decreased in areas of noise contamination and non-linearities.

3. Steady State Vibration Response Measurement

Steady state vibrations were measured using the apparatus shown in the schematic diagram in Figure 19. An aluminum beam was mounted so that it could be excited by a shaker. The input acceleration was measured at the mounting jaws and the output acceleration was measured at the tip of the beam. The beam was excited using a random noise excitation of 1 kHz and 500 Hz bandwidths. The system transfer function was calculated, and the Hilbert transform of the real and imaginary portions of the frequency response was performed to determine if the system behaved in a linear manner.

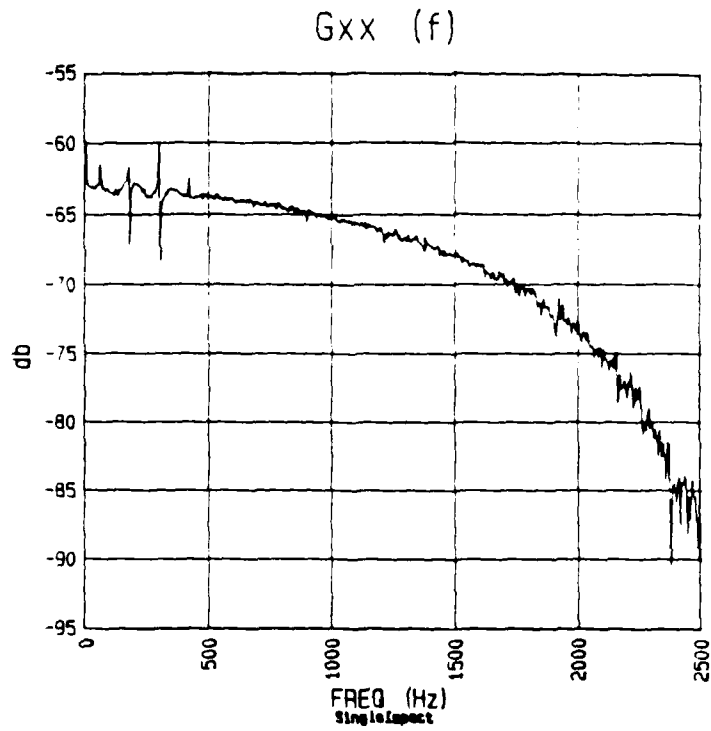


Figure 12: Input Power Spectra for Single Impact

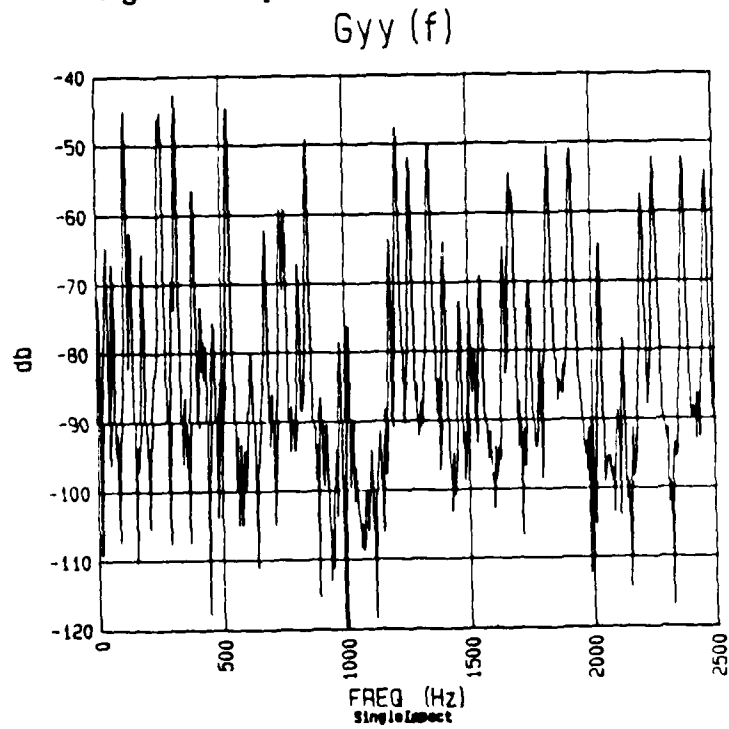


Figure 13: Output Power Spectra for Single Impact

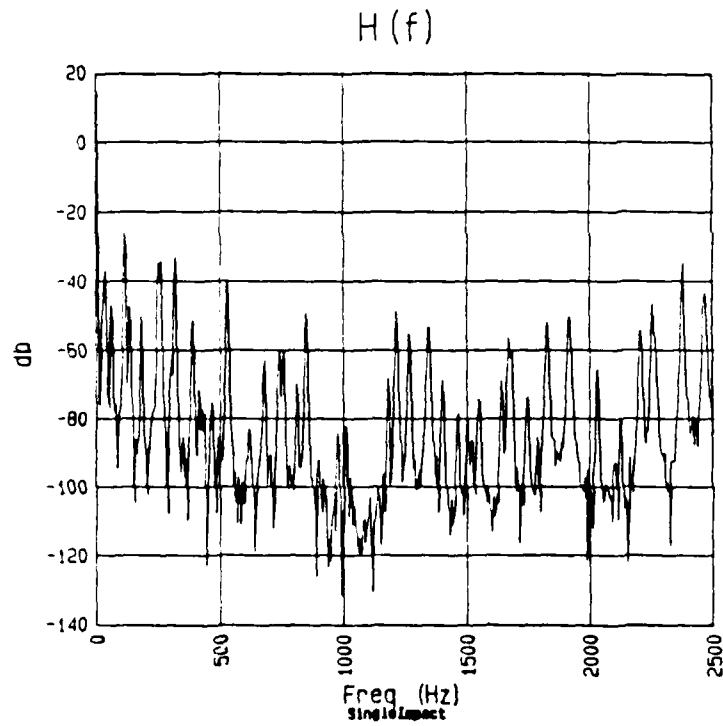


Figure 14: Transfer Function for Single Impact

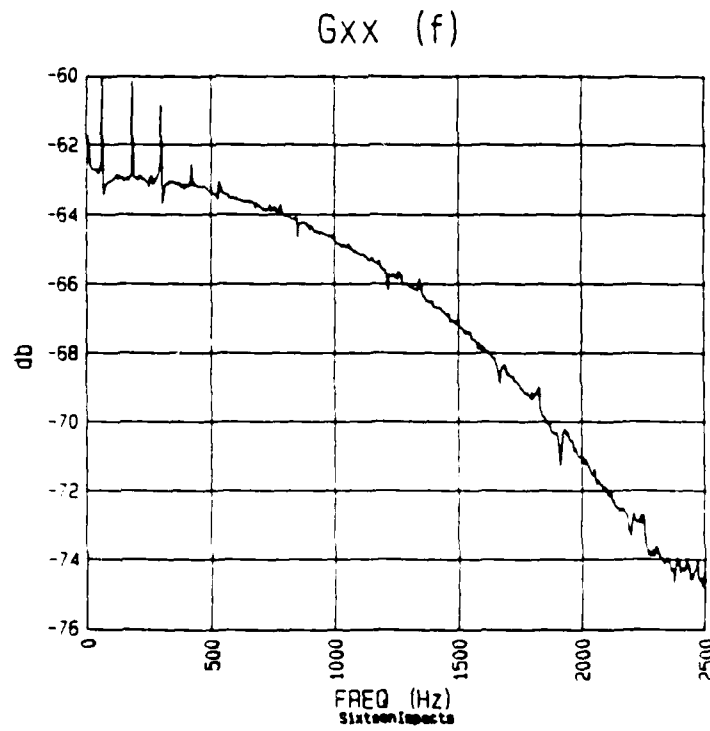


Figure 15: Input Power Spectra for Multiple Impact

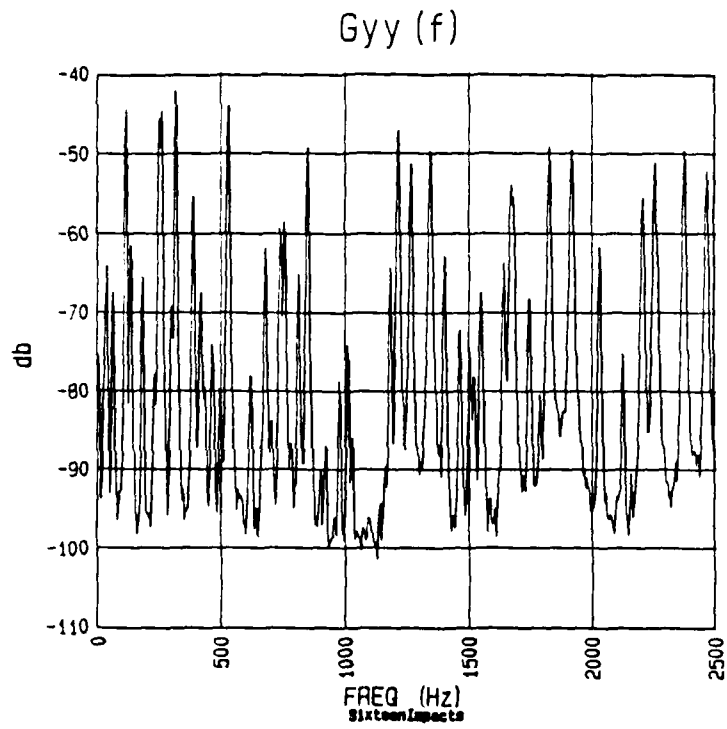


Figure 16: Output Power Spectra for Multiple Impact

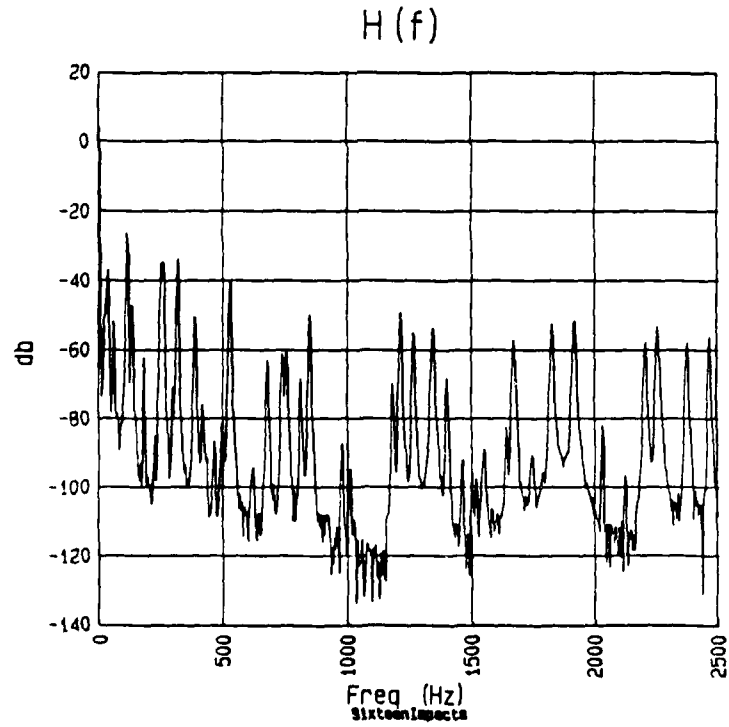


Figure 17: Transfer Function for Multiple Impact

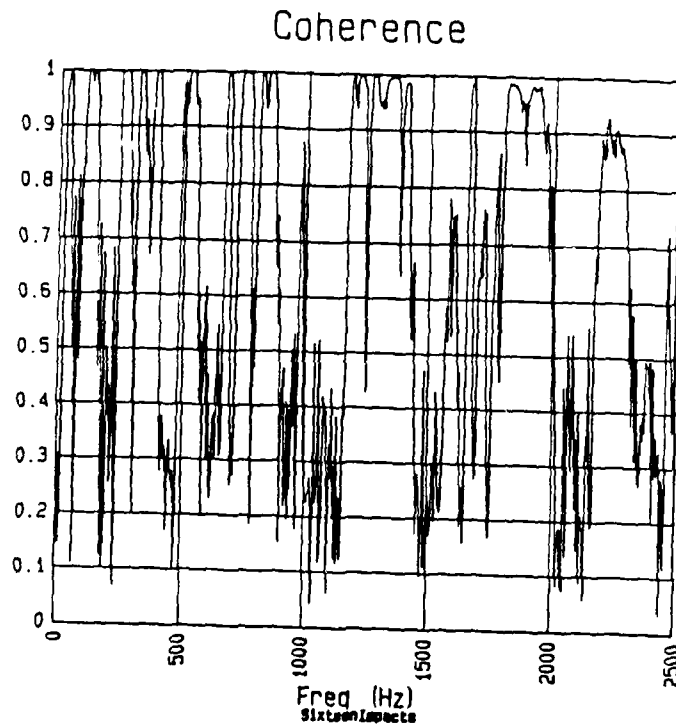


Figure 18: Coherence Measurement for Multiple Impact

A low pass filter ensured that no frequency components above the range of investigation contaminated the transfer function. The sampling rate was such that a frequency resolution of 1.3 Hz was obtained. For steady state vibration response measurements, a modified version of the program used for impact testing was used. The program is listed in Appendix D. Because the same data reduction and analysis was performed on the steady state vibrations testing as was done on the impact testing information the same reduction routines were used.

Figure 20 is the input power spectra for a random noise excitation at a 1 kHz bandwidth centered at 500 Hz. Figures 21 and 22 are the output power spectra and coherence measurements for the same excitation. The transfer function for this excitation is shown in Figure 23. The use of random noise produced a low coherence measurement in the region above 200 Hz. In order to improve the coherence measurements and smooth the transfer function in the low frequency area, a random noise excitation of only 500 Hz bandwidth starting at zero was used. The results of using this excitation are shown in Figures 24 through 29.

An attempt to utilize a swept sine wave as the excitation source was made. Two methods, one using a HP 3562A Dynamic Signal Analyzer and a second using a voltage

controlled oscillator driven by the digital to analog converter of the EF12M multifunction module, were used to sweep through a frequency band. Using the Hewlett Packard dynamic signal analyzer was not effective. As the dynamic signal analyzer swept through the frequency bandwidth, the MASSCOMP would sample the responses. However, due to the length of time required by the MASSCOMP to process the data, the MASSCOMP would miss some frequencies as the HP Dynamic Signal Analyzer swept through them. The method of using a voltage controlled oscillator was more effective; however, the processing time was such that only a 200 Hz bandwidth could be swept without skipping some frequencies. This required approximately twenty-five minutes to sweep this bandwidth, while with a Dynamic Signal Analyzer the entire bandwidth from 0 to 1 kHz could be sampled in the same amount of time. This necessitated the use of random noise as the only excitation source of the shaker.

The results of the steady state analysis were compared to the HP 3562A Dynamic signal Analyzer for accuracy. Figures 30 through 34 are the results of those test. When compared to the results gained via the microcomputer, the differences are quite noticeable. Of particular importance is the difference in magnitude in the input power spectra, the output power spectra and the cross spectra. While the shape of the measurement results are nearly identical, the magnitudes are different. This leads to an incorrect approximation of the real and imaginary portions of the transfer functions. This is evident in Figures 35 through 38. Figures 35 and 36 are the real and imaginary portions of the transfer function measured with the microcomputing system. The identical measurements using the dynamic signal analyzer are shown in Figures 37 and 38. The difference in magnitude is immediately apparent.

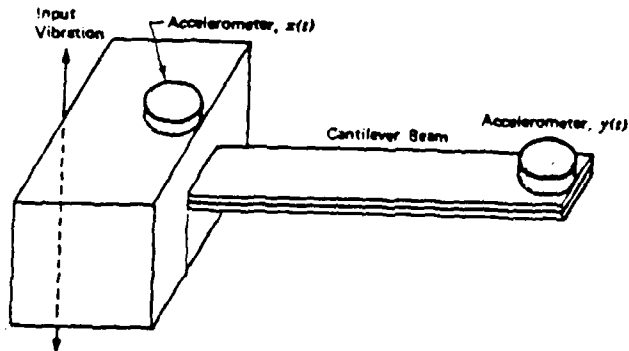
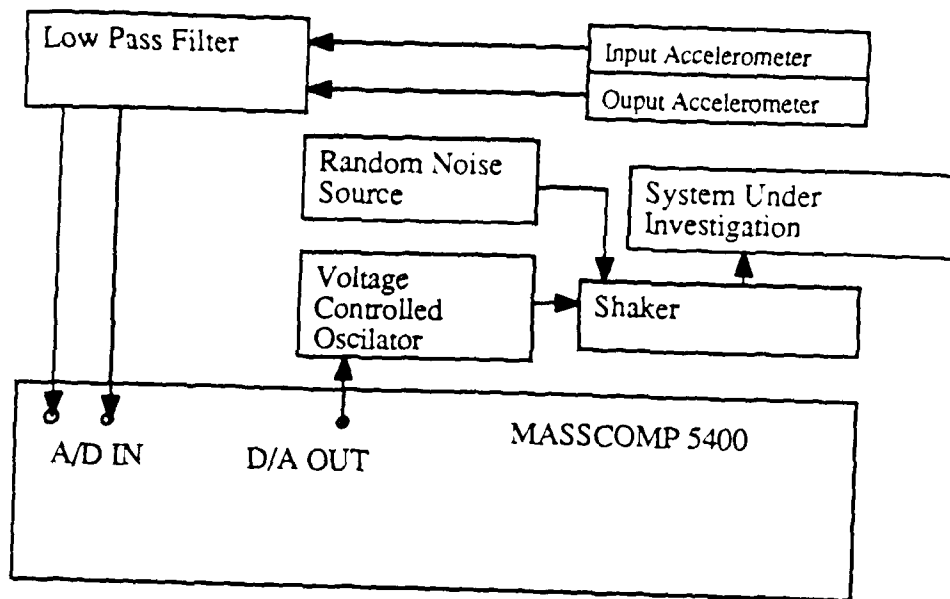


Figure 19: Steady State Vibration Measurement Schematic Diagram (upper) and Cantilevered Beam used in Experimental Procedure (lower).

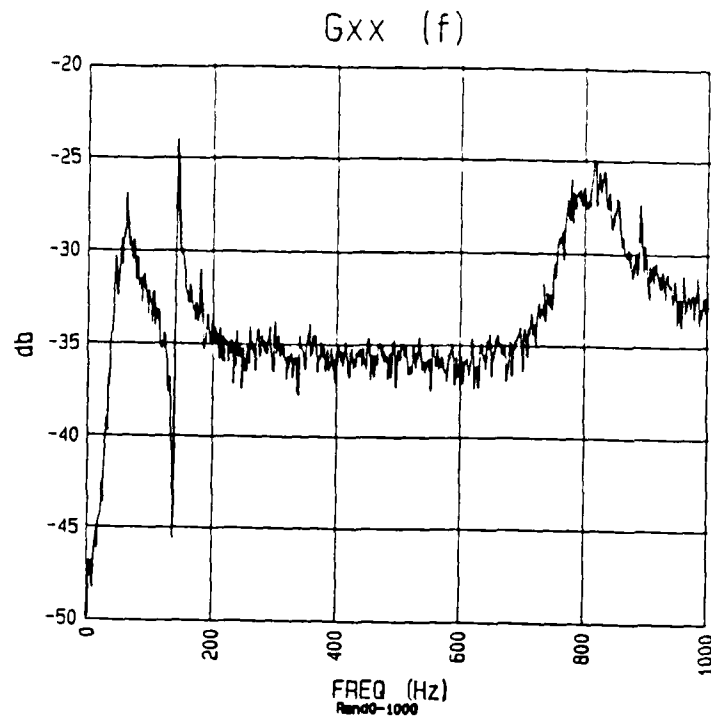


Figure 20: Input Power Spectra for 1 kHz Random Noise Excitation

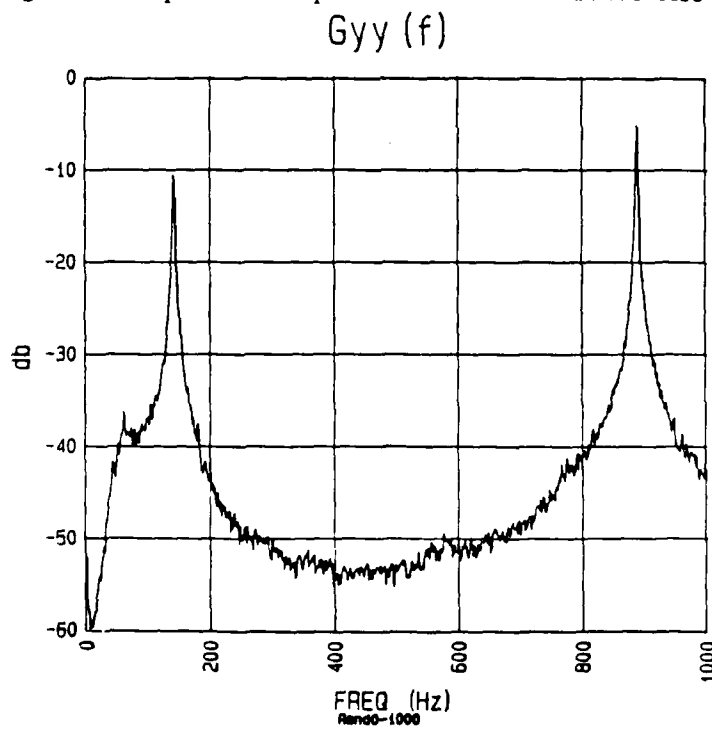


Figure 21: Output Power Spectra for 1 kHz Random Noise Excitation

Coherence

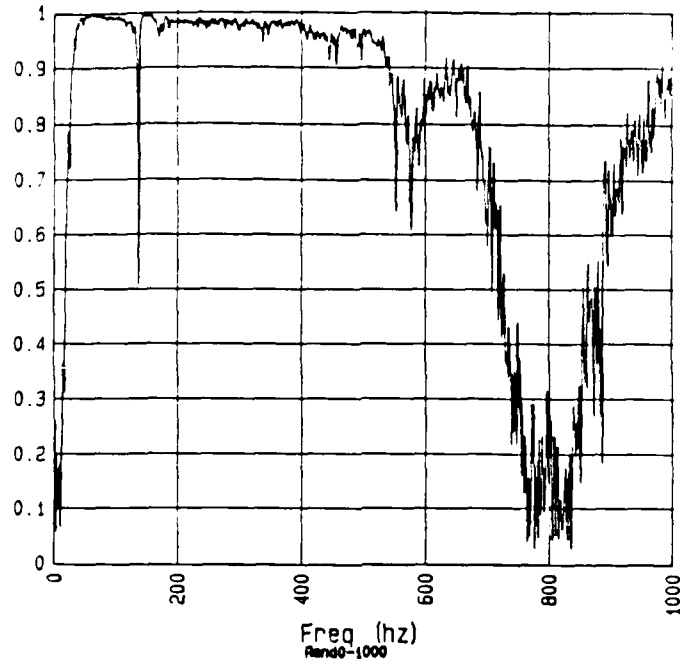


Figure 22: Coherence Measurement for 1 kHz Random Noise Excitation

$H(f)$

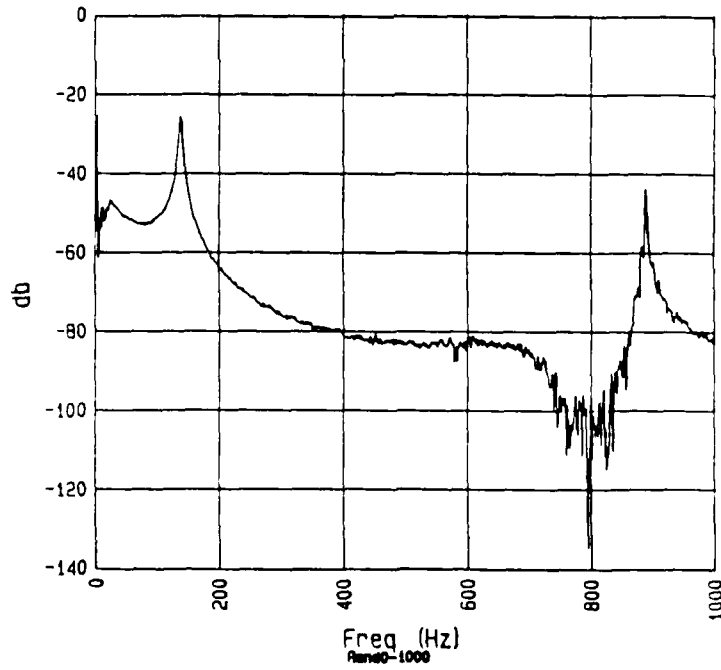


Figure 23: Transfer Function for 1 kHz Random Noise Excitation

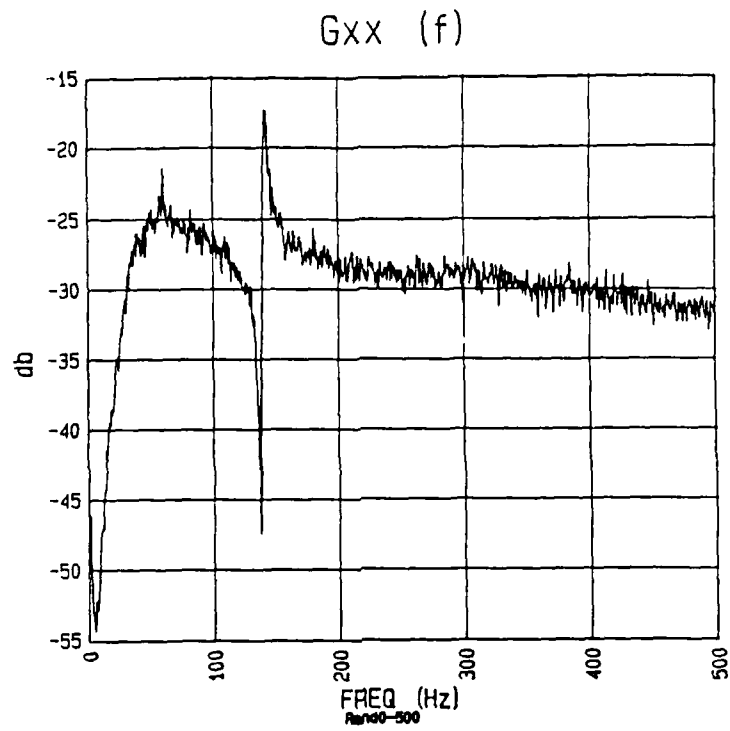


Figure 24: Input Power Spectra for 500 Hz Random Noise Excitation

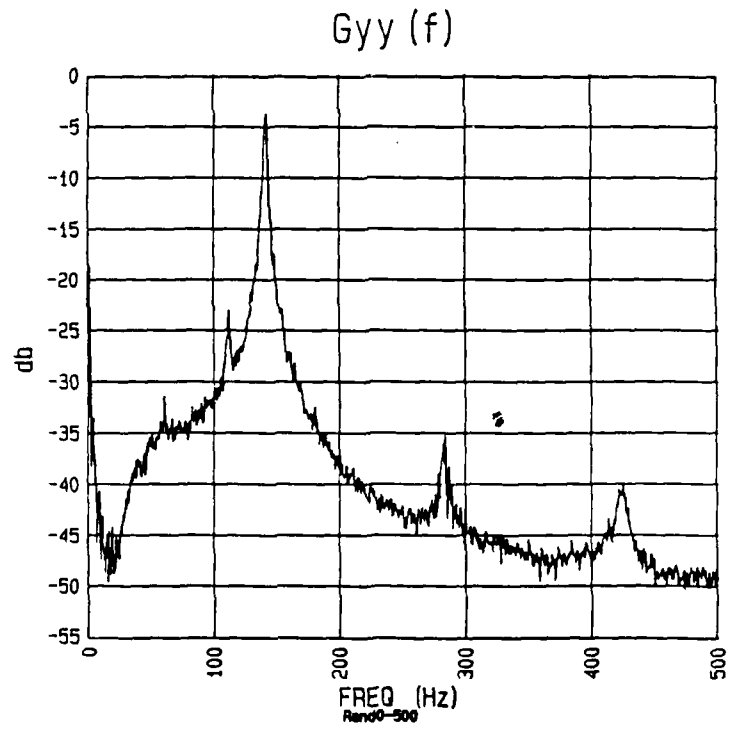


Figure 25: Output Power Spectra for 500 Hz Random Noise Excitation

Coherence

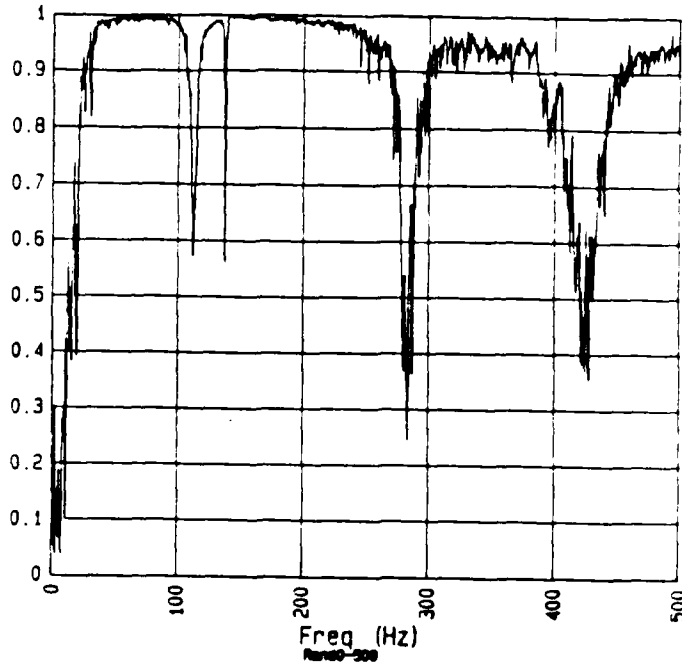


Figure 26: Coherence Measurement for 500 Hz Random Noise Excitation
 $H(f)$

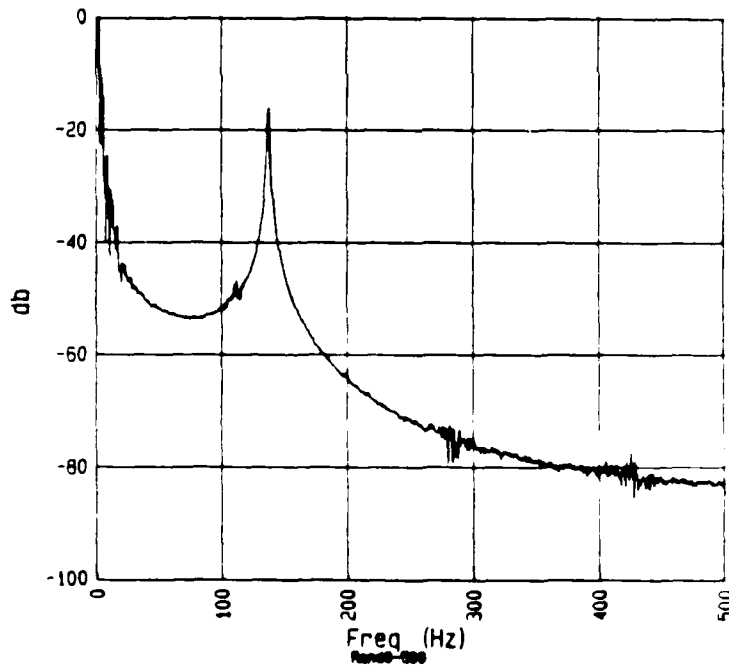


Figure 27: Transfer Function for 500 Hz Random Noise Excitation.

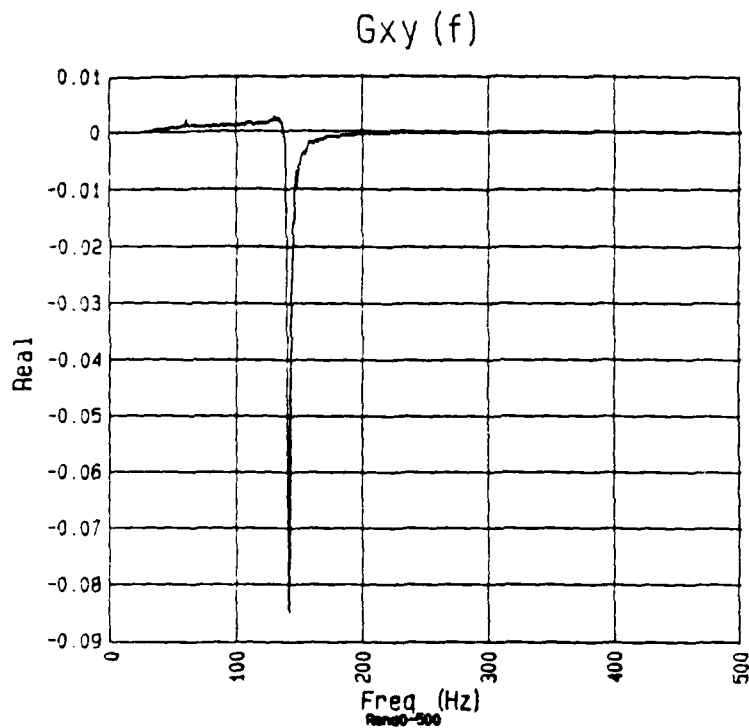


Figure 28: Real Portion of Cross Power Spectra for 500 Hz Random Noise Excitation

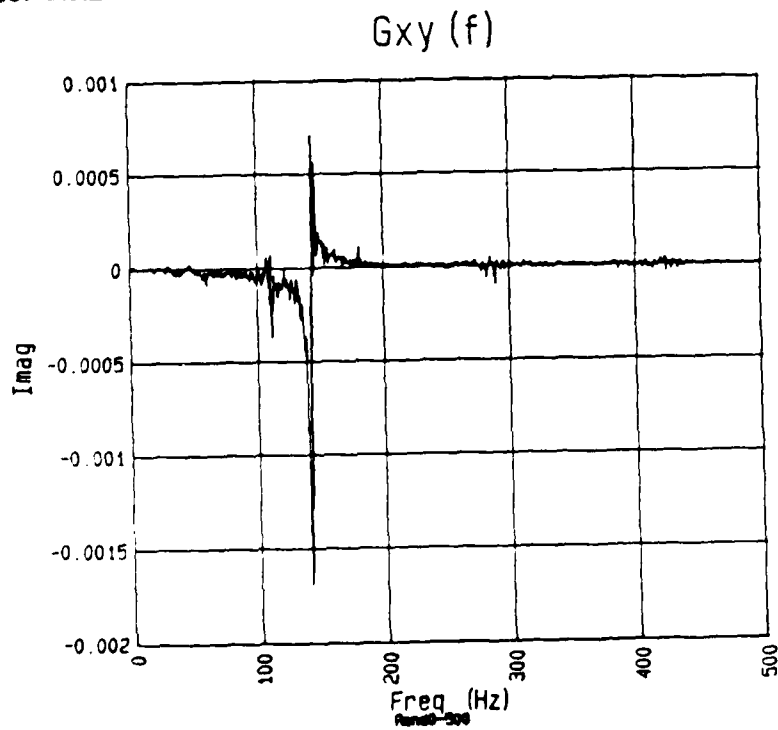


Figure 29: Imaginary Portion of Cross Power Spectra for 500 Hz Random Noise Excitation.

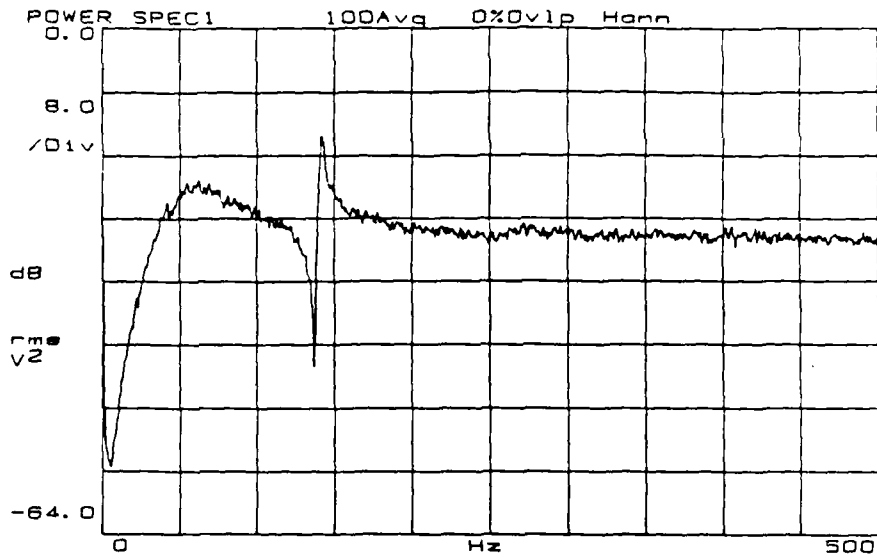


Figure 30: Input Power Spectra for 500 Hz Excitation Measured with HP 3562A

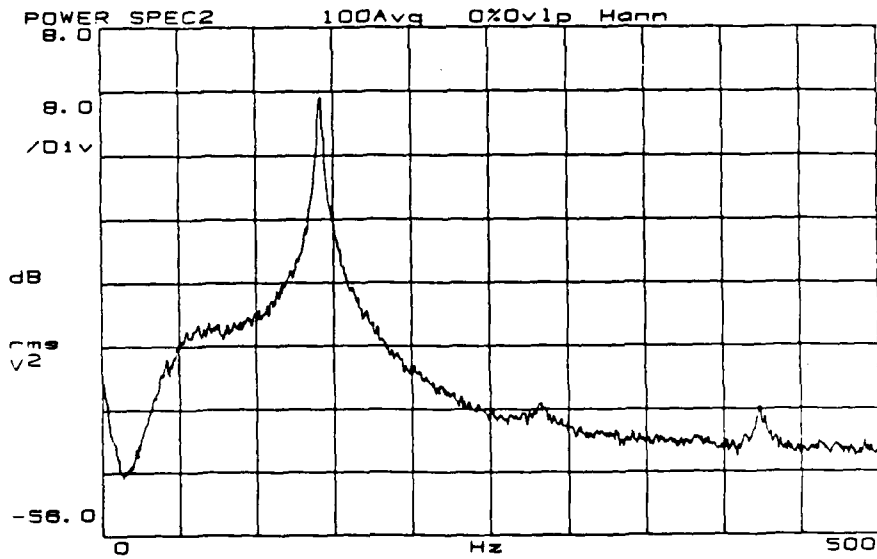


Figure 31: Output Power Spectra for 500 Hz Random Excitation Measured with HP 3562A

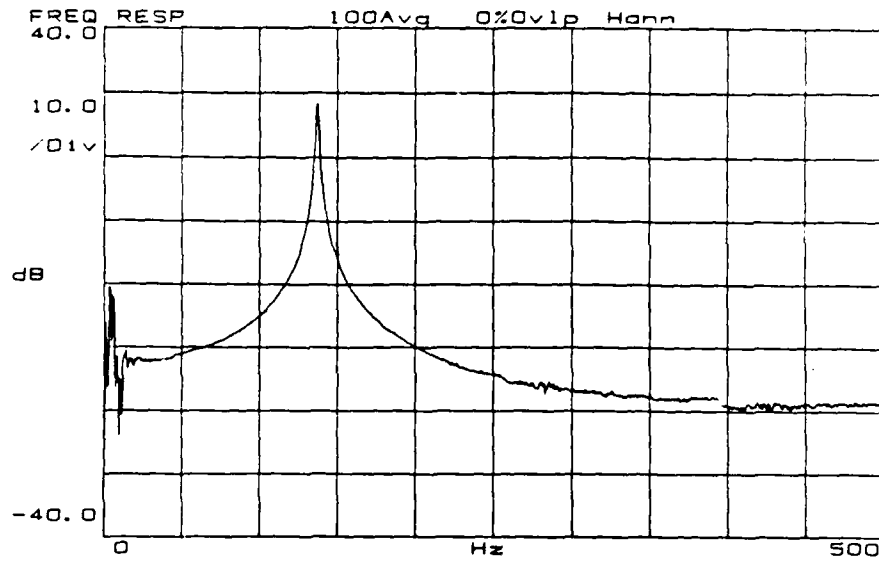


Figure 32: Transfer Function for 500 Hz Random Noise Excitation Measured with HP 3562A

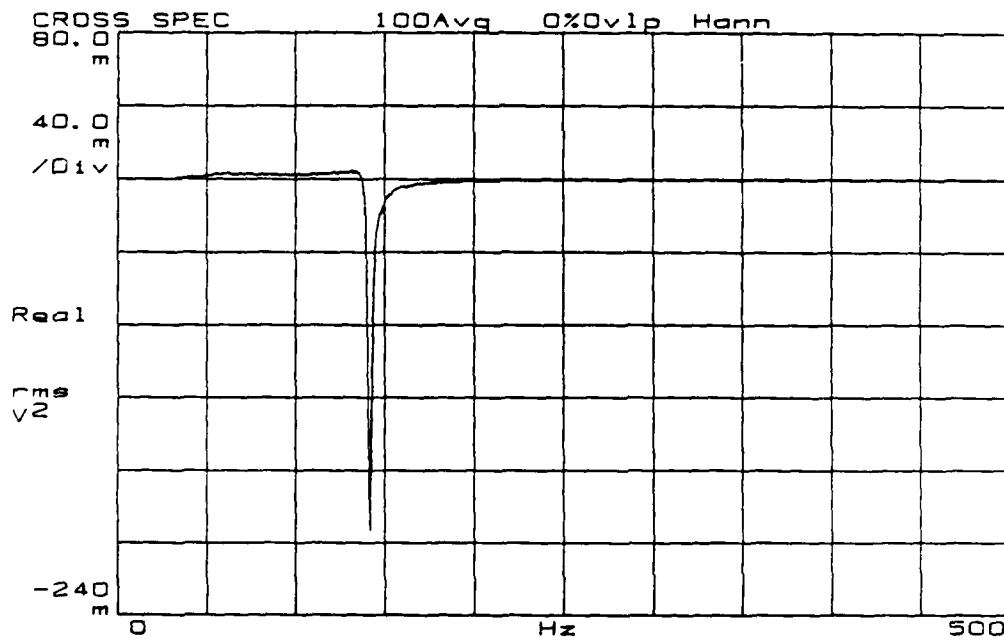


Figure 33: Real Portion of Cross Power Spectra for 500 Hz Random Excitation Measured with HP 3562A

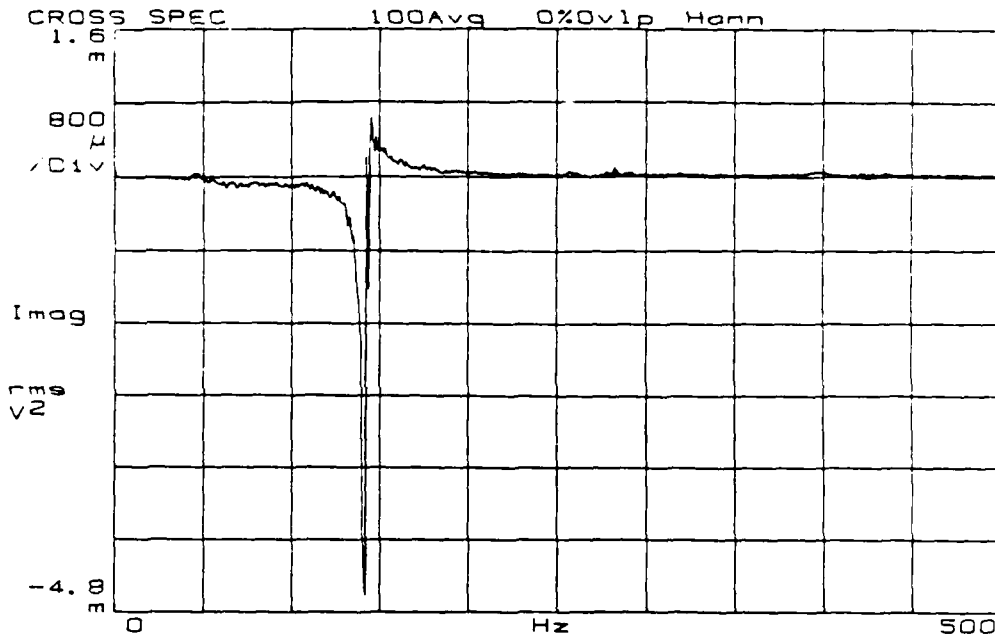


Figure 34: Imaginary Portion of Cross Power Spectra for 500 Hz Random Excitation Measured with HP 3562A

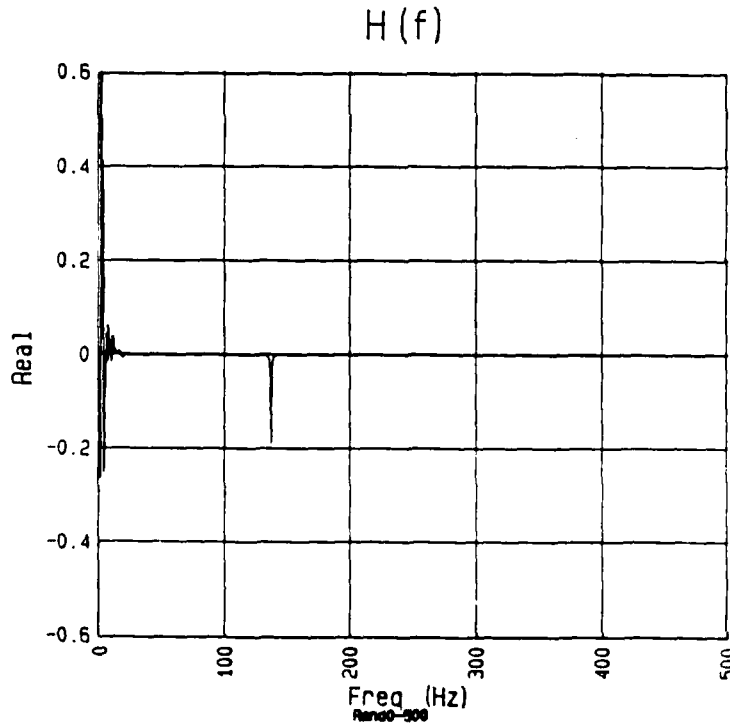


Figure 35: Real Portion of Transfer Function for 500 Hz Excitation

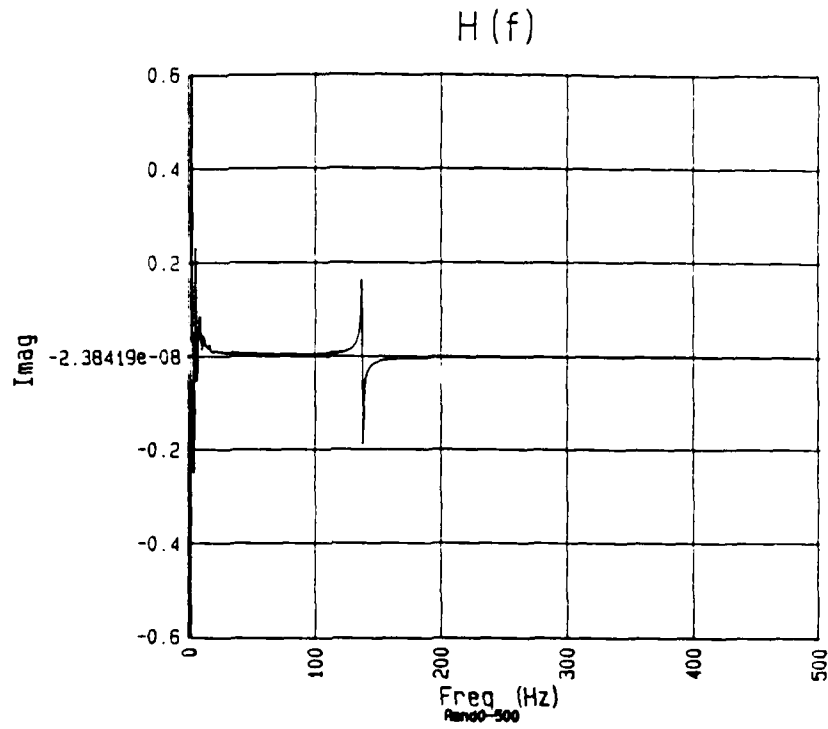


Figure 36: Imaginary Portion of Transfer Function for 500 Hz Excitation

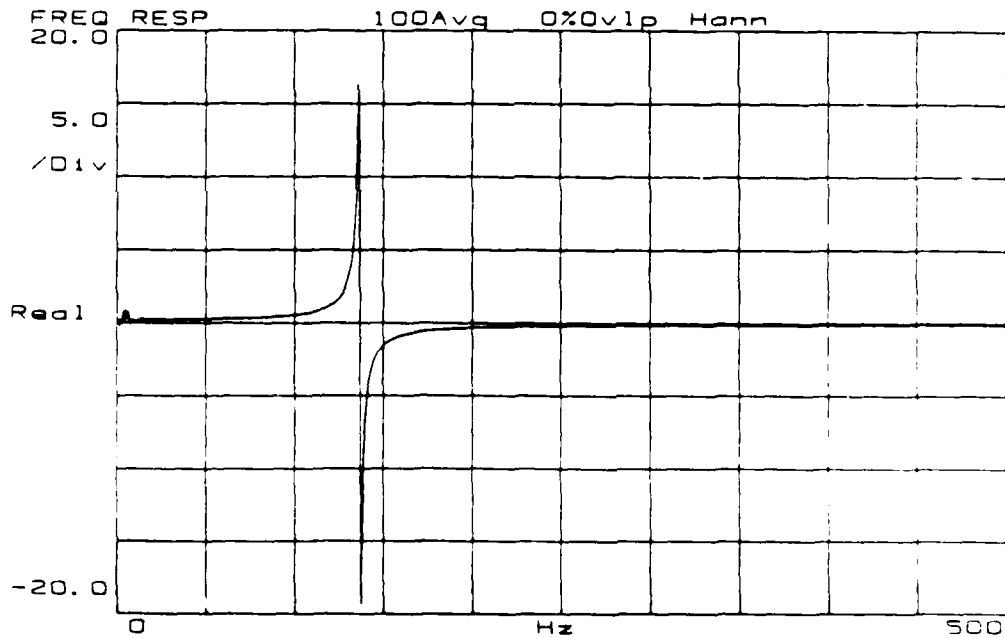


Figure 37: Real Portion of Transfer Function for 500 Hz Excitation Measured with HP 3562A

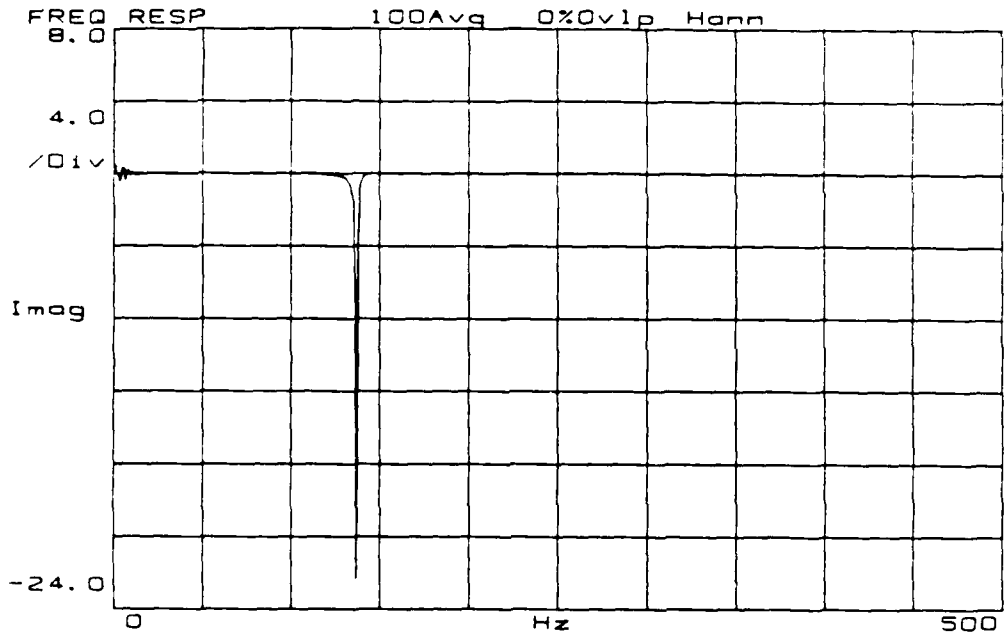


Figure 38: Imaginary Portion of Transfer Function for 500 Hz Excitation Measured with HP 3562A

IV. APPLICATION OF THE HILBERT TRANSFORM

The aluminum beam used for steady state vibrations analysis was also used to evaluate the feasibility of including the Hilbert transform as a method of identifying system irregularities. There are two resonant modes, one at 137 Hz and the second at 891 Hz, shown previously in Figure 23. Using the half power bandwidth method to estimate the viscous damping ratio at the second resonant frequency:

$$\zeta_s = \frac{1}{2} \left(\frac{\omega_2 - \omega_1}{\omega_s} \right) \quad [\text{Eqn. 28}]$$

where ω_2 and ω_1 are the half power frequencies for that resonant point and ω_s is the resonant frequency. Using the information for the second resonant frequency from Figure 39, the viscous damping ratio ζ_s was determined to be 0.00180. This damping ratio was then used to estimate the correction terms necessary for the Hilbert transform. Recalling that the structural damping ratio is proportional to the viscous damping ratio,

$$\delta_s = \frac{\zeta_s}{2} , \quad [\text{Eqn. 29}]$$

and substituting into Equation 20,

$$X_s = H_I(\omega_s) \delta_s \omega_s = H_I(\omega_s) \frac{\zeta_s}{2} \omega_s , \quad [\text{Eqn. 30}]$$

Substituting Equation 30 into Equation 18, the correction term for the imaginary portion of the Hilbert transform is then:

$$C_{1s}(\omega) = j \frac{\omega X_s}{\omega_s^2 - \omega^2} = j \frac{\omega H_I(\omega_s) \zeta_s \omega_s}{2 (\omega_s^2 - \omega^2)} = j \frac{\omega H_I(\omega_s) (\omega_2 - \omega_1)}{4 (\omega_s^2 - \omega^2)} . \quad [\text{Eqn. 31}]$$

Using the damping information from the second resonant frequency, the correction terms were less than 10^{-5} and were considered negligible and were not included.

The real and imaginary portions of the original transfer function and the Hilbert transform of the transfer function are shown in Figures 40 through 44. The numerical integration method of performing the Hilbert transform was not accurate in the area of the first resonant frequency. Figures 45 and 46 show that the Hilbert transform of the transfer function was not significantly different than the original transfer function. The numerical integration method will always yield incorrect answers for systems of low damping at resonance.

The FFT method of performing the Hilbert transform was also inaccurate. Due to the transform into the time domain and subsequent truncation, there appeared to be high frequency components introduced to the transfer function. This is evident in Figures 47 and 48. However, the shape of both the real and imaginary portions of the Hilbert transform are identical to the original transfer function.

When using the frequency response of a model of a linear system the Hilbert transform was able to indicate that the system behaved in a linear fashion. However, due to the inaccuracy in measuring the cross power spectra the transfer functions used in performing the Hilbert transform are inaccurate. Thus in order to ensure that the Hilbert transform in yielding correct information, the original transfer function must be correct; otherwise, the Hilbert transform will indicate a system that is in fact a linear system as a non-linear system.

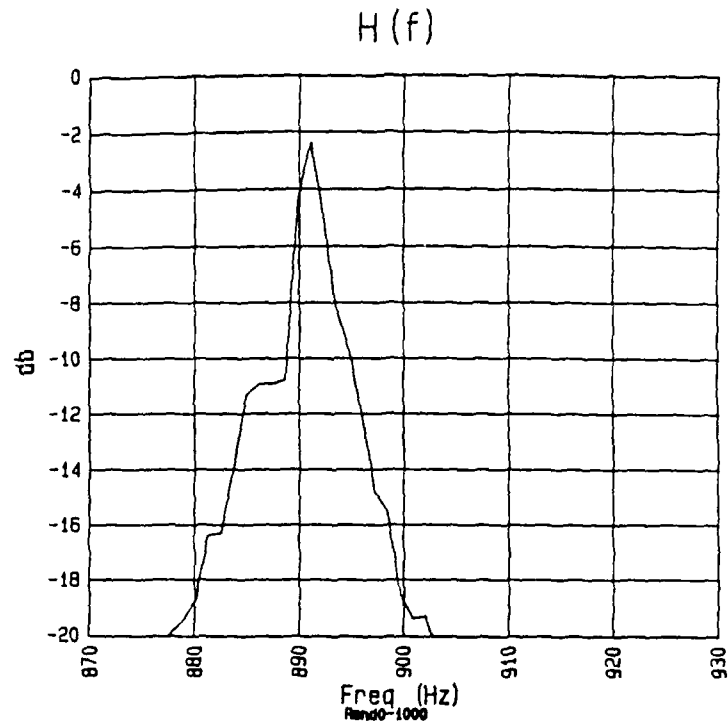


Figure 39: Second Resonant Frequency Identified using 1 kHz Random Noise Excitation

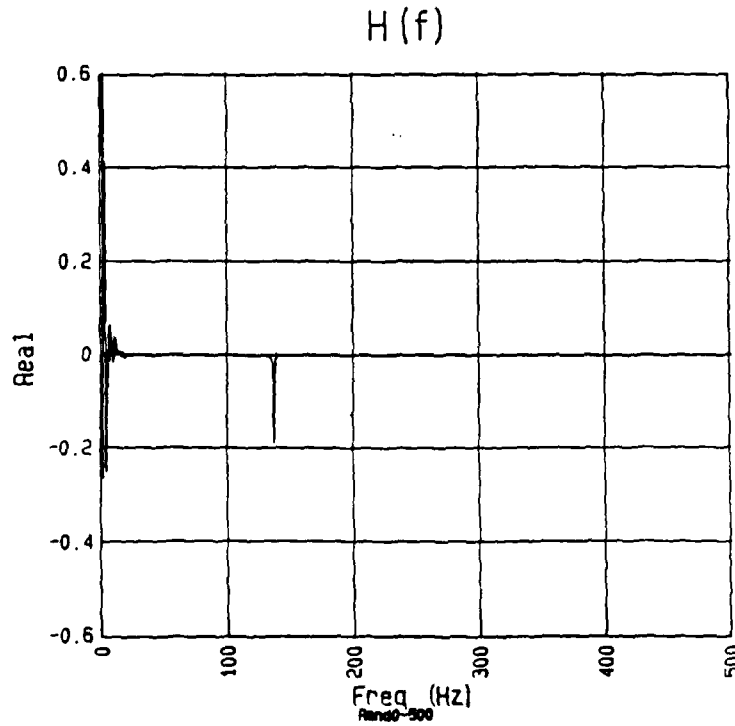


Figure 40: Real Portion of the Transfer Function of the First Resonant Frequency

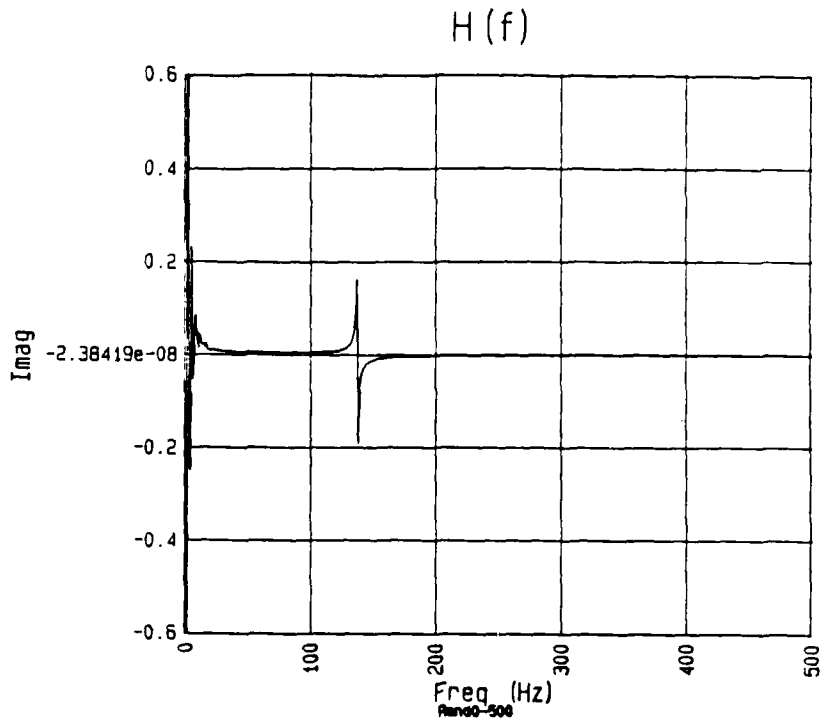


Figure 41: Imaginary Portion of the Transfer Function of the First Resonant Frequency Hilbert

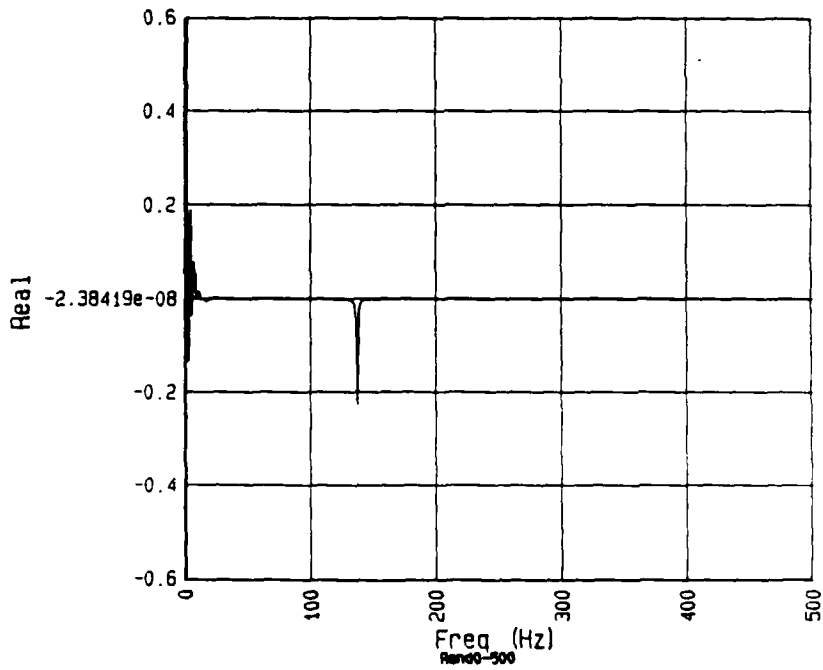


Figure 42: Hilbert Transform of the Real Portion of the Transfer Function using Numerical Integration

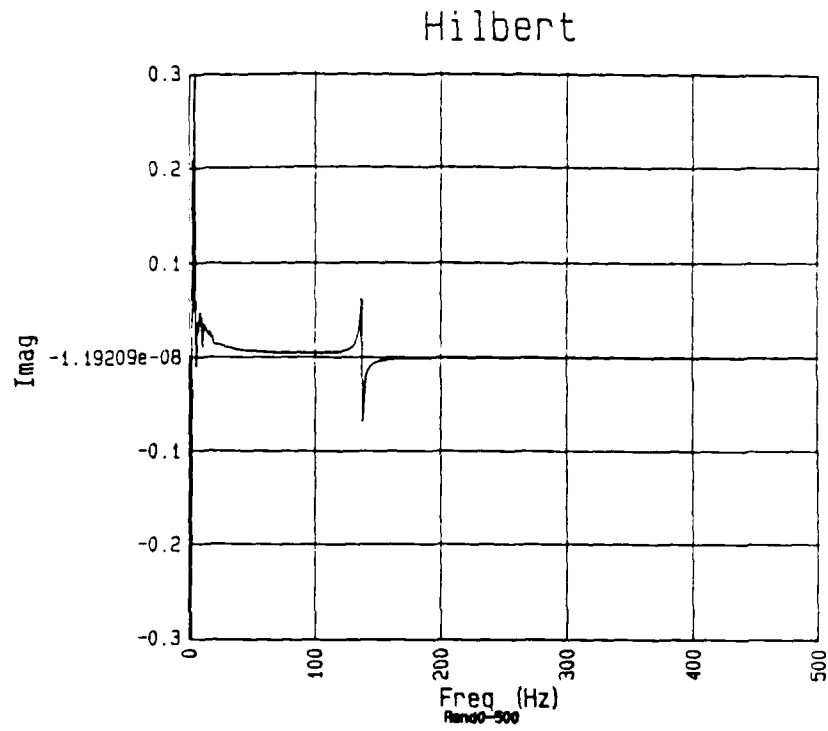


Figure 43: Hilbert Transform of the Imaginary Portion of Transfer Function using Numerical Integration

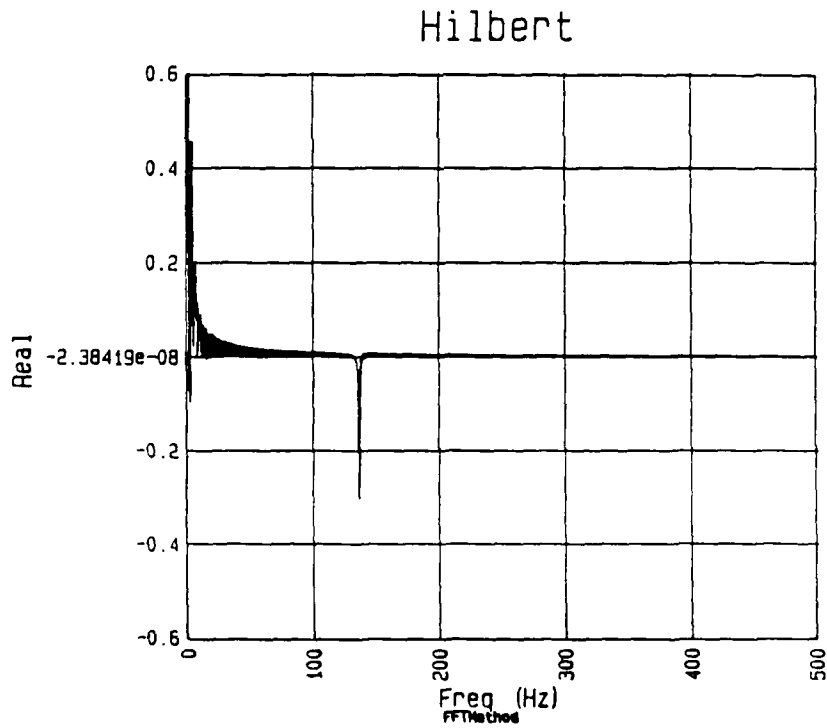


Figure 44: Hilbert Transform of the Real Portion of the Transfer Function using FFT Method

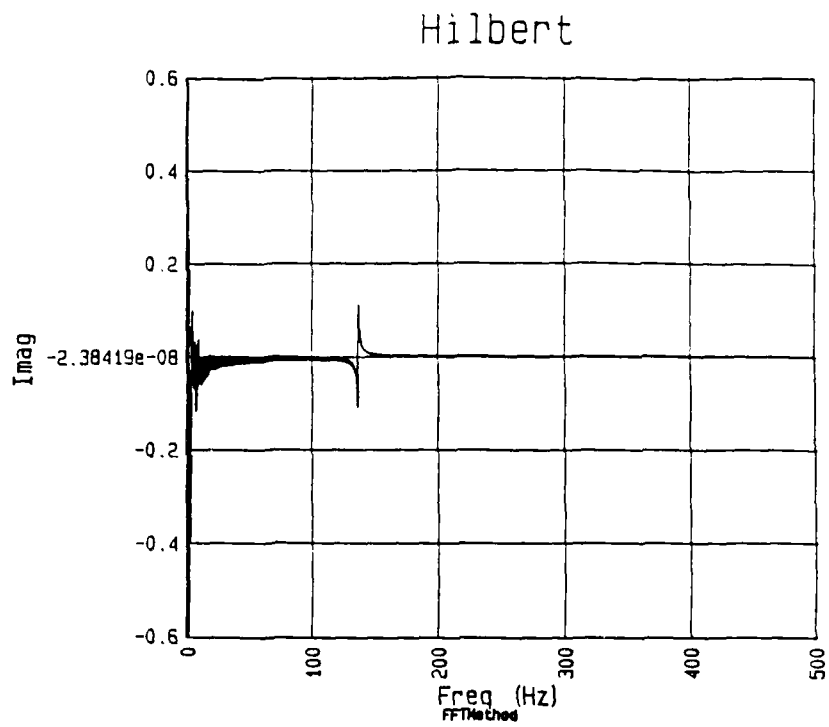


Figure 45: Hilbert Transform of the Imaginary Portion of the Transfer Function using FFT Method

V. CONCLUSIONS

A. UNDERWATER EXPLOSION TESTING

1. The super microcomputer is able to record up to two channels of shock pressure with 250 kHz bandwidth or sixteen channels of strain information with 10 kHz bandwidth directly to data arrays. It is possible to reduce and display the data prior to storage to a recording media.
2. In the present configuration of only one sample and hold device for the data acquisition hardware only one type of data, shock pressure or strain, can be recorded. The addition of a second sample and hold card would allow two different data types to be recorded and analyzed.

B. IMPACT TESTING OF AN ALUMINUM PLATE

1. The data acquisition capabilities, tied to the Digital Signal Processing software, allow for a complete dynamic signal analysis of a structure using a microcomputer.
2. Sampling rates of 32 kHz are sufficient for multi-channel low frequency vibrations analysis. However, the requirement to use an analog filter prior to sampling requires the use of a filter for each data channel monitored. This would require a rack of filters and reduce the ease of movement of the system. Setting up of a vibrations analysis station on a permanent basis would be required.

C. STEADY STATE VIBRATION ANALYSIS

1. The data acquisition techniques are sound; however, the microcomputer data acquisition equipment yields information that is significantly different in magnitude when compared to a proven measurement device. This is due to the 12 bit resolution over the 10 volt dynamic range of the analog to digital converter.
2. The coherence measurements using an external random noise source indicate that this is not the optimum method of performing response measurements. Attempts to improve the coherence measurements by using a voltage controlled oscillator controlled by the computer as a sinusoidal excitation source were unsuccessful.
3. The identification of non-linear response using the Hilbert transform is limited to systems with a few widely spaced resonant modes. It requires data reduction to be

performed by the user prior to attempting the identification of non-linearities. Because of the poor voltage resolution of the analog to digital converters, the system transfer information used as an input for the Hilbert transform is in error. Thus the reliability of the Hilbert transform for use determination of the linearity of the frequency response is poor.

4. The present data presentation in graphical format capability is rudimentary and requires improvement. The capability to include cursor input into the program for axes scaling and data zooming and recording would greatly improve the analysis capabilities available to the user. The capabilities of using a "mouse" or a digital graphics pad as an control device to delineate points of interest for further reduction would greatly enhance the capability to process information.

VI. RECOMMENDATIONS

A. UNDERWATER EXPLOSION TESTING

1. Include the use of a second sample and hold card. This would allow simultaneous recording of a single high frequency shock pressure signal and sixteen low frequency strain gage signals.

B. IMPACT TESTING

1. Improve the presentation software so that cursor input via a "mouse" can be used to select areas of interest for further analysis.

C. STEADY STATE VIBRATION ANALYSIS

1. Explore the possibility of using a proved data acquisition device, the HP 3562A, in conjunction with the programming capability of the MASSCOMP system. The EF12M Multifunction Module has an IEEE 488 interface capability. The HP 3562A may be able to acquire the data and transfer it to the MASSCOMP for exploration of the degree of linearity of the transfer function by use of the Hilbert transform.
2. Reduce the processing time requirements for large data arrays. FORTRAN programming is not the best method to process information in real time applications. Transferring the code to assembly language or machine code would greatly reduce the time requirements of data analysis and allow faster structure analysis.
3. Improve the presentation software as noted above for impact testing.
4. Develop a method of swept sine wave excitation of the structure using the microcomputer as the controller. The digital to analog converter allows for two channels of control information to be sent from the computer, yet it must operate independent of the analog To digital converter.
5. Add a signal conditioner to boost the level of the input signals to the analog to digital converter. This would result in more accurate calculation of the system transfer function and increase the reliability of the results of the Hilbert transform measurements.

APPENDIX A

MASSCOMP 5400 HARDWARE CAPABILITIES

The capabilities of the data acquisition devices installed on the MASSCOMP 5400 and the microcomputer itself are summarized below;

MASSCOMP 5400 Super Microcomputing system

- Motorola 68020 CPU module
- 6 Mbyte of system memory
- Data Acquisition Hardware
 - AD12FA Analog to Digital Converter
 - EF12M Multifunction Module
 - SH16F Sample and Hold Card
- Two Display Terminals
- HP plotter
- Dot matrix printer

AD12FA Analog to Digital Converter

- [Path identified as /dev/dacp0/adf1]
- 16 channel input single ended, 8 differential [channels 0 - 15]
- Programmable-gain amplifier (with gains selectable by software of 1,2,4, or 8)
- Sample and Hold circuit for connection to the SH-16F
- Maximum aggregate sampling rate: 1 Mhz
- Resolution: 12 bits (zero padding for total sample length of 16 bits)
- Input Voltages:
 - 0 to +10 V
 - 5 to + 5 V

EF12M Multifunction Module

- Analog to Digital Converter Section
 - [Path identified as /dev/dacp0/adf0]
 - 16 channel input single ended, 8 differential
 - Programmable-gain amplifier (with gains selectable by software of 1,2,4, or 8)
 - Maximum aggregate sampling rate: 333 kHz
 - Resolution: 12 bits (zero padding for total sample length of 16 bits)
 - Input Voltages:
 - 10 to +10V

Digital to Analog Converter Section

[Path identified as /dev/dacp0/daf0]

2 channels

Maximum clock rate per channel: 333 kHz

Resolution: 12 bits (zero padding for total sample length of 16 bits)

Output Voltages: -10 to +10 V

Parallel I/O Section

16 Channels Input and Output utilizing hardware handshake for transmission of data

Clock Section

[Path identified as /dev/dacp0/efclkn (n identifies clock 0-5)]

5 Counters with output, source input and gate input connections

Frequency capability ~0 Hz to 3 MHz

SH-16F Sample and Hold Module

[Path /dev/dacp0/adf1, channels 16 through 31 on the AD12FA card]

16 Single-ended Input Channels or,

8 Differential Channels

Maximum of 3 SH-16F modules may be connected to a AD12FA or EF12M to give the capability of sampling a maximum of 48 channels with a single A/D converter. The maximum aggregate sampling rate must never exceed the capability of the "host" A/D converter.

APPENDIX B

UNDERWATER EXPLOSION DATA TRANSFER PROGRAM

This appendix contains the program used to collect the data of the underwater explosion recordings from the FM tapes. All routines were written by the author. For further explanation of the data acquisition specific routines the reader is encouraged to refer to the MASSCOMP documentation "Data Acquisition User's Manual." Figure 46 is the program flow diagram. Sections of the program follow the general outline presented in the flow diagram. Specific subroutines for plotting the information are not shown in this appendix.

```
*
* THIS PROGRAM IS THE MAIN PROGRAM FOR DATA ACQUISITION
* OF DATA FROM STORED TAPES OF UNDERWATER EXPLOSIONS
* VERSION 1
* LT R. A. SHAFER, USN
* NAVAL POSTGRADUATE SCHOOL
* 28 APRIL 1987
*
*****
* SET UP VARIABLES
* MYBUFSZ IS THE BUFFER SIZE FOR TEMPORARY DATA STORAGE OF A/D DATA
* FACTOR IS A CONVERSION FACTOR FOR BUFFER MANAGEMENT
* N BUFFS      THE NUMBER OF BUFFERS FOR DATA COLLECTION
* NCLKS        THE NUMBER OF CLOCKS USED FOR A/D CONTROL
* NEARFREQ     PARAMETER FOR CLOCK CONTROL
* RONLY        SET A/D CONVERSION TO READ ONLY
* STARTLO      SET CLOCK WAVEFORM IN LOW (ZERO VOLTAGE) POSITION
* TIMEOUT      TIME LIMIT FOR DATA ACQUISITION
* FREQ         FREQUENCY OF SAMPLING CLOCK
* WIDTH        WIDTH OF SAMPLE PULSE (0.5 MICROSEC MIN)
*
* INTEGER      MYBUFSZ
* INTEGER      FACTOR
* INTEGER      NBUFS
* INTEGER      NCLKS, NEARFREQ, RONLY
* INTEGER      RDWRT, SLICE, SQUARE
* INTEGER      STARTLO, TIMEOUT
* REAL         FREQ, WIDTH
* PARAMETER    ( MYBUFSZ = 16384 )
* PARAMETER    ( FACTOR = 2 )
* PARAMETER    ( NBUFS = 3 )
* PARAMETER    ( NCLKS = 1 )
* PARAMETER    ( NEARFREQ = 0 )
* PARAMETER    ( RONLY = 1 )
* PARAMETER    ( RDWRT = 0 )
* PARAMETER    ( SLICE = 10 )
```

```

PARAMETER ( SQUARE = 4 )
PARAMETER ( STARTLO = 0 )
PARAMETER ( TIME0000 )
PARAMETER ( WIDTH = 0.0 )
*
* SET UP INPUT BUFFER LIMITS
* ENSURE BUFFER IN EVEN WORD BOUNDRY IN STACK
*
INTEGER*2 DATABLK(98304)
INTEGER DUMMY
EQUIVALENCE (DATABLK,DUMMY)
*
* DECLARE SERVICE ROUTINE AS EXTERNAL PROGRAM
*
EXTERNAL ADSERVICE
*
* SET UP VARIABLES FOR A/D CONTROL AND SETTINGS
*
* ADPATH MARKER FOR PATH TO A/D CONVERTER
* CLKPATH MARKER FOR PATH TO CONTROL CLOCK
* NCHAN NUMBER OF A CHANNELS TO BE SAMPLED
* FCHAN FIRST CHANNEL TO BE SAMPLED
* SCHAN SPACING OF CHANNELS IF MORE THAN ONE SAMPLED
* GAIN GAIN SETTING OF A/D CONVERTER
* INDEX ARRAY POINTER FOR BUFFER MANAGEMENT
* POWER NUMBER TO RAISE 2 TO THE POWER OF FOR SAMPLE BLOCK SIZE
* TYME ARRAY FOR TIME VALUES
* RECLN TIME RECORD LENGTH OF DATA SET
* RFREQ ACTUAL FREQUENCY CLOCK SET TO
* RWIDTH ACTUAL WIDTH OF SAMPLE PULSE
* SAMPLES ARRAY FOR TIME RECORD DATA
* LOW LOWER LIMIT FOR DATA STORAGE ON HARD DISK
* HIGH UPPER LIMIT FOR DATA STORAGE ON HARD DISK
INTEGER ADPATH
INTEGER CLKPATH
INTEGER STATWDS(2)
INTEGER I
INTEGER NCHAN
INTEGER FCHAN
INTEGER SCHAN
INTEGER GAIN
INTEGER INDEX
INTEGER POWER
REAL TYME(16384)
REAL RECLN
REAL RFREQ
REAL RWIDTH
REAL SAMPLES(16384)
REAL LOW
REAL HIGH
REAL TIMSCL
REAL SUM

CHARACTER ANSWER

```

COMMON SAMPLES, ADPATH, CLKPATH, INDEX, DATBLK, NITEMS

```
*
* WRITE HEADER ON SCREEN
*
      WRITE (6,9999)
9999  FORMAT (47H1UNDERWATER EXPLOSIONS DATA ACQUISITION PROGRAM)
*
* GET INPUT PARAMETERS FROM SCREEN
*
      PRINT *, "INPUT NUMBER OF CHANNELS TO BE SAMPLED"
      READ *, NCHAN
      PRINT *, "INPUT FIRST CHANNEL TO BE SAMPLED"
      READ *, FCHAN
      PRINT *, "INPUT SPACING OF CHANNELS ( 0 FOR NONE ) "
      READ *, SCHAN
      PRINT *, "INPUT GAIN FACTOR ( 0,1,2,3 = *1, *2, *4, *8 )"
      READ *, GAIN
*
* ENSURE ALL DACP DEVICES ARE CLOSED
*
      CALL MRCLOSALL
*
* OPEN A/D CONVERTER PATH
* SET FIRST CHANNEL, SPACING AND GAIN ON A/D CONVERTER
*
      ADPATH = -1
      CALL MROPEN(ADPATH, "/dev/dacp0/adf0", RONLY)
      CALL MRADMOD(ADPATH, 0, 0)
      CALL MRADINC(ADPATH, FCHAN, NCHAN, SCHAN, GAIN)
*
* IDENTIFY BUFFERS FOR DATA COLLECTION AND RELEASE THEM
* TO THE EMPTY BUFFER QUEUE OF A/D CONVERTER
*
      DO 200 I = 1, 81921, MYBUFSZ
      CALL MRBUFID (ADPATH, DATBLK(I), MYBUFSZ*2)
      CALL MRBUFREL (ADPATH, DATBLK(I))
200  CONTINUE
*
* OPEN CLOCK PATH FOR A/D CONVERTER CONTROL CLOCK
* AND INTERNALLY CONNECT TO THE A/D CONVERTER
*
      CLKPATH = -1
      CALL MROPEN (CLKPATH, "/dev/dacp0/efclk2", RDWRT)
      CALL MRWIRE (ADPATH, 0)
*
* GET CLOCK SETTINGS FROM SCREEN
*
      PRINT *, "INPUT DESIRED FREQUENCY IN HZ"
      READ *, FREQ
*
* DISARM CLOCK TO ENSURE IT IS NOT RUNNING
* AND SET UP CLOCK PARAMETERS
*
      CALL MRCLKDIS ( 1, CLKPATH )
```

```

        CALL MRCLKIGATED ( CLKPATH, NEARFRQ, FREQ, RFREQ, 1, 1.5, RWIDTH,
        STARTLO, 5)
*
* GET NUMBER OF DATA POINTS TO BE RECORDED
* AND TIME SCALING FACTOR
    PRINT *, "INPUT POWER OF TWO FOR NUMBER OF SAMPLES ( MAX = 14 )"
    READ *, POWER
    NITEMS = 2**POWER
    PRINT *, "NUMBER OF SAMPLES IS ", NITEMS
    RECLEN = NITEMS / RFREQ
    PRINT *, "ENTER TIME SCALING FACTOR"
    READ *, TIMSCL
    PRINT *, "SIMULATED SAMPLING FREQUENCY IS -> ", RFREQ * TIMSCL
    PRINT *, "TIME RECORD LENGTH IS ", RECLEN / TIMSCL
    PRINT *, "START TAPE NOW"
    PRINT *, "THEN ENTER 'y' AND HIT RETURN WHEN TAPE IS READY"
    READ *, ANSWER
*
* ARM CLOCK AND WAIT FOR TRIGGER TO DROP TO ZERO
*
    PRINT *, "ARMING CLOCK"
    CALL MRCLKARM ( 1, CLKPATH )
    PRINT *, "WAITING FOR TRIGGER PULSE"
*
* COLLECT DATA SET AND SEND BUFFER TO A/D SERVICE ROUTINE
*
    CALL MRXINQ ( ADPATH, NITEMS, NITEMS, ADSERVICE )
    CALL MREVWT ( ADPATH, STATWDS, 30000 )
*
* COMPUTE AVERAGE OF FIRST 1200 SAMPLES TO REMOVE ANY ZERO OFFSET
*
    PRINT *, "STOPPED READING"
    DO 250 I = 1, 1200
        SUM = SUM + SAMPLES(I)
250    CONTINUE
    SUM = SUM / 1200
*
* FILL TIME ARRAY AND REMOVE ZERO OFFSET FROM RECORD
*
    DO 300 I = 1, NITEMS
        TYME(I) = (I-1) / (RFREQ * TIMSCL)
        SAMPLES(I) = SAMPLES (I) - SUM
300    CONTINUE
9990    FORMAT (1X, G20.8, 10X , G20.8)
*
* CALL PLOTTING ROUTINE FOR DISPLAY OF DATA
*
    CALL PLOT1( NITEMS, SAMPLES, TYME, " Time History " , "Voltage" , "Seco
ds", "timhis.g" )
*
* WRITE DATA TO HARD DISK
*
    PRINT *, " INPUT LIMITS OF HARDDISK DATA RECORD "
    READ *, LOW, HIGH
    DO 350 I = 1, NITEMS

```

```

*   IF ( TYME(I) .GE. LOW .ME(I) .LE. HIGH) THEN
*       WRITE (7,9990) TYME(I), SAMPLES(I)
*   END IF
*350 CONTINUE

```

```

STOP
END

```

```

SUBROUTINE ADSERVICE

```

```

INTEGER    K
INTEGER    INDEX
INTEGER    POWER
INTEGER    ADPATH
INTEGER    NITEMS
INTEGER*2  DATABLK(98304)
INTEGER    GETINDEX
PARAMETER  (GETINDEX = 3)
REAL      DUMMY
REAL      SAMPLES(16384)

```

```

COMMON SAMPLES, ADPATH,CLKPATH, INDEX, DATABLK, NITEMS

```

```

IF (.EQ. 0) THEN
    NITEMS = 16384
END IF

```

```

*
* STOP CLOCK, GET BUFFER FROM DONE BUFFER QUEUE
* CORRECT DATA BY SCALING FACTOR (20/2^12)
* FILL SAMPLE ARRAY WITH TIME RECORD AND RELEASE BUFFER TO
* READY BUFFER QUEUE
*

```

```

    CALL MRCLKDIS( 1, CLKPATH )
    CALL MRBU GETINDEX, INDEX)
    TEMP = 1 / 204.8
    DO 300 K = 1, NITEMS
    DUMMY = FLOAT(DATABLK(K)) * TEMP
    SAMPLES(K) = DUMMY
300 CONTINUE
    CALL MRBUFREL (ADPATH, DATABLK(INDEX))
    RETURN
END

```

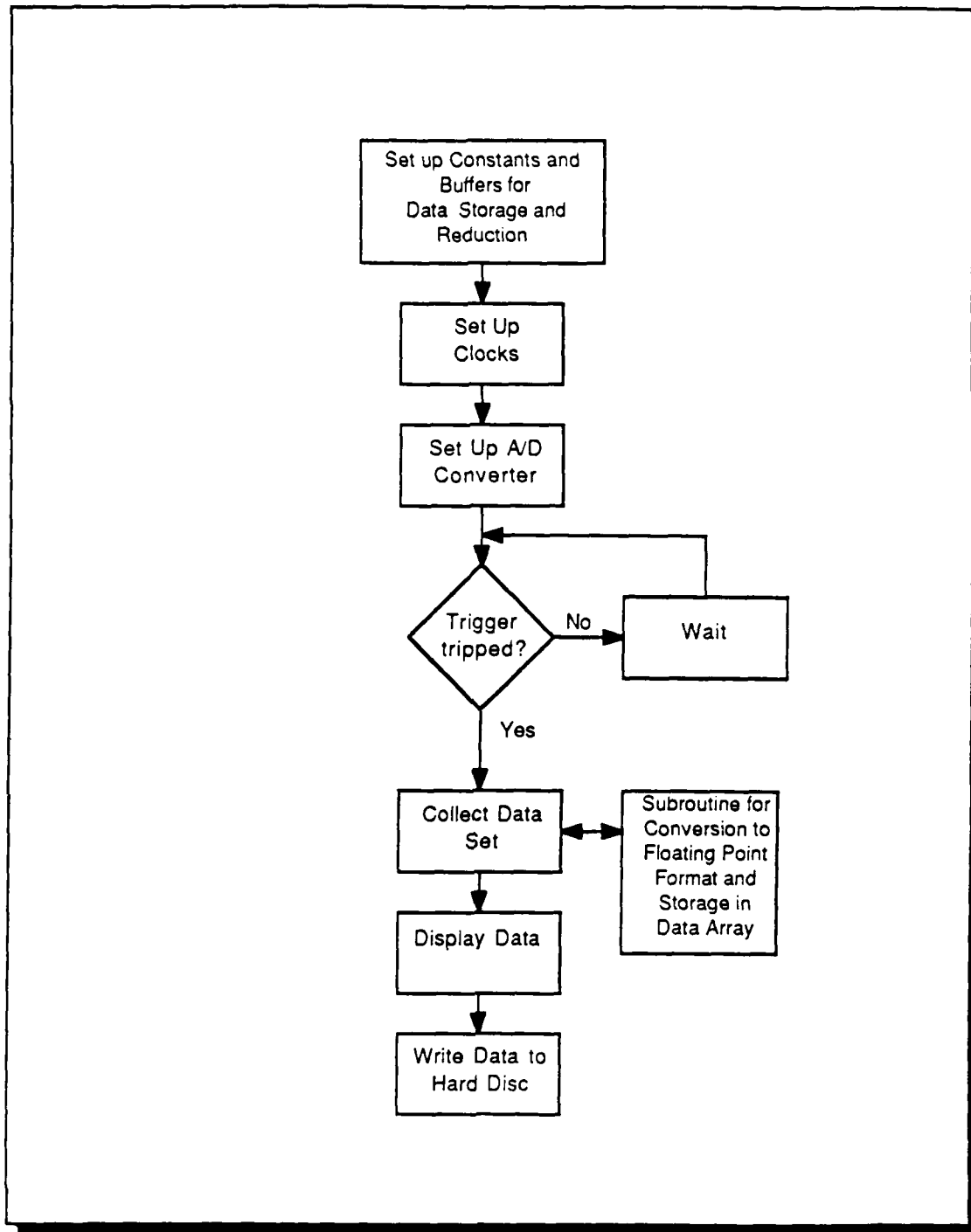


Figure 46: Underwater Explosion Data Transfer Program

APPENDIX C

IMPACT TESTING DATA ACQUISITION AND REDUCTION PROGRAM

This section contains the program listing used to acquire the data from the aluminum plate. All routines were written by the author except where noted. For further explanation of the data acquisition specific routines the reader is encouraged to refer to the MASSCOMP documentation "Data Acquisition User's Manual". Figure 47 is the program flow diagram. Sections of the program follow the general outline presented in the flow diagram. Specific subroutines for plotting the information are not shown in this appendix.

```
*
* THIS PROGRAM IS THE MAIN PROGRAM FOR DATA ACQUISITION
* AND REDUCTION OF IMPACT TESTING OF AN ALUMINUM PLATE
*
* R. A. SHAFER, LT, USN
* NAVAL POSTGRADUATE SCHOOL
* MONTEREY, CA 93940
* 27 MARCH 1987
*
*
* SET UP CONSTANTS AND VARIABLES
*
  INTEGER      MYBUFSZ, NBUFFS
  INTEGER      NEARFREQ, RDONLY
  INTEGER      RDWRT, SQUARE
  INTEGER      STARTLO, TIMEOUT
  REAL         FREQ, WIDTH
  REAL         PI
  INTEGER      OUTFILE
  PARAMETER    ( NBUFFS = 10 )
  PARAMETER    ( NEARFREQ = 0 )
  PARAMETER    ( RDONLY = 1 )
  PARAMETER    ( RDWRT = 0 )
  PARAMETER    ( SQUARE = 4 )
  PARAMETER    ( STARTLO = 0 )
  PARAMETER    ( TIMEOUT = 30000 )
  PARAMETER    ( WIDTH = 0.0 )
  INTEGER*2    DATABLK(100000)
  INTEGER      DUMMY
  EQUIVALENCE (DATABLK,DUMMY)
  EXTERNAL    ADSERVICE
  INTEGER      ADPATH
```

```

INTEGER      CLKPATH(2)
INTEGER      STATWDS(2)
INTEGER      I
INTEGER      NCHAN
INTEGER      FCHAN
INTEGER      SCHAN
INTEGER      GAIN
INTEGER      INDEX
INTEGER
REAL         TPI
REAL         WEGHT(513)
REAL         TYME(1024)
REAL         RECLEN
REAL         RFREQ
REAL         RWIDTH
REAL         XTR(1024), Y1TR(1024)
REAL         XTI(1024)
REAL         YITI(1024)
REAL         XX(1024)
REAL         YY1(1024)
REAL         XFR(1024),XFI(1024)
REAL         Y1FR(1024),Y1FI(1024)
REAL         GXX(513)
REAL         GYY(513)
REAL         GXYRE(513)
REAL         GXYIM(513)
INTEGER      NOSAMP
REAL         DELTAFRQ(1024)
REAL         DFREQ
REAL         HFR(513),HFI(513)
REAL         HF(513),HMR(513)
REAL         HMI(513),HM(513)
REAL         AVG(5)
CHARACTER    ANSWER
COMMON /BCR/ XTR, Y1TR, ADPATH, DATABLK, MYBUFSZ, NCHAN, AVG
COMMON /CCSE/ TYME, DELTAFRQ, OUTFILE
OUTFILE = 8
*
*   OPEN FILES FOR OUTPUT OF DATA
*
OPEN (7,FILE="/usr/shafer/plate/data",STATUS="NEW")
*
*   CALCULATE HANNING WINDOW WEIGHTING FACTOR
*
TPI = 8.0 * ATAN(1.0)
PI = 4.0 * ATAN(1.0)
TEMP = TPI / 1025.0
SCL = SQRT(2.0/3.0)
DO 40 I = 1,512
    WEGHT(I)= SCL * (1.0- COS(TEMP*FLOAT(I)))
40  CONTINUE
*
*   SET UP CONSTANTS FOR A/D CONVERTER SETUP AND BUFFERS
*
MYBUFSZ = 5000

```

```

NCHAN = 2
FCHAN = 16
SCHAN = 1
GAIN = 0
ADPATH = -1
*
*   OPEN A/D CONVERTER PATH AND SET CONVERTER MODE
*
CALL MROPEN(ADPATH, "/dev/dacp0/adf1", RONLY)
CALL MRADINC(ADPATH, FCHAN, NCHAN, SCHAN, GAIN)
CALL MRADMOD(ADPATH, 0, 0)
*
*   OPEN CLOCK PATHD PULSES AND SET CLOCK MODE
*
*   SET UP CLOCK FOR A/D CONVERTER
*
CLKPATH(1) = -1
CLKPATH(2) = -1
CALL MROPEN (CLKPATH(1),"/dev/dacp0/efclk0", RDWRT)
CALL MROPEN (CLKPATH(2),"/dev/dacp0/efclk4", RDWRT)
FREQ = 5000.
PRINT *, "SETTING UP CLOCKS"
CALL MRCLK2 (CLKPATH(1),CLKPATH(2),0,0,FREQ,RFREQ,0,200000.,RWIDTH,2,0)
CALL MRCLKTRIG (ADPATH, 2,CLKPATH)
print *, "Actual samp frequency per channel -- ",RFREQ
NITEMS = 60000
PRINT *, "NUMBER OF SAMPLES IS ",NITEMS
*
*   CALCULATE TIME RECORD LENGTH AND FREQUENCY RESOLUTION
*
RECLEN = 1024/RFREQ
PRINT *, "TIME RECORD LENGTH IS ",RECLEN
DFREQ= 1/RECLEN
DO 50 I = 1, 1024
TYME(I) = I / RFREQ
DELTAFRQ(I) = (I-1)*DFREQ
60  CONTINUE
*
*   GET NUMBER OF AVERAGES TO BE COMPUTED FROM USER
*
PRINT *, "INPUT NUMBER OF AVERAGES TO BE COMPUTED"
READ *,NOSAMP
*
*   SET UP BUFFERS AND RELEASE THEM TO A/D PATH
*
DO 200 I = 1, 45001, MYBUFSZ
CALL MRBUFID (ADPATH, DATABLK(I), MYBUFSZ*2)
CALL MRBUFREL (ADPATH, DATABLK(I))
200 CONTINUE
*
*   START DATA ACQUISITION
*
DO 250 L = 1, NOSAMP
65  PRINT *, L

```

PRINT *, "AWAITING TRIGGER ARM, TYPE ANY CHARACTER AND RETURN
WHEN READY"

READ *, ANSWER
AVG(1) = 0
AVG(2) = 0

*
* TAKE IN DATA RECORD
*

CALL MRXINQ (ADPATH, MYBUFSZ, NITEMS, ADSERVICE) MREATH,S, 30000)
PRSTOPPED READING"

*
* DISPLAY DATA RECORD FOR CORRECTNESS
* IF CORRECT CONTINUE, IF NOT DISCARD DATA SET AND TAKE ANOTHER
*

CALL PLOT1(1024, XTR, TYME, "DATA PREVIEW", "VOLTAGE", "TIME", "pre.g")
PRINT *, "Good data set?"
READ *, ANSWER
IF (ANSWER .EQ. "n") THEN
GOTO 65
END IF

*
* BEGIN DATA REDUCTION
*

DO 70 I = 1, 1024
XX(I) = XTR(I)
YY1(I) = Y1TR(I)
70 CONTINUE

*
* WEIGHT TIME RECORD WITH HANNING WINDOW TO ENSURE PERIODICITY
* REQUIREMENTS OF FFT ARE MET. XX, BEING AN IMPACT MEASUREMENT
* IS PERIODIC BY DEFINITION AND REQUIRES NO WEIGHTING
*

DO 80 I = 1, 512
ITMP = 1025 - I
YY1(I) = YY1(I) * WEGHT(I)
YY1(ITMP) = YY1(ITMP) * WEGHT(I)
80 CONTINUE

*
* CALCULATE POWER SPECTRA INFORMATION
* FOLLOWING PORTION TAKEN FROM CCSE PROGRAM
*

CALL FFT842(0,1024,XX,YY1,OUTFILE)

GXX(1) = GXX(1) + 4.0 * XX(1)**2
DO 90 K = 2, 513
J = 1026 - K
GXX(K) = GXX(K) + (XX(K)+XX(J))**2 + (YY1(K)-YY1(J))**2
GYY(K) = GYY(K) + (YY1(K)+YY1(J))**2 + (XX(J)-XX(K))**2
GXYRE(K) = GXYRE(K) + XX(K)*YY1(J) + XX(J)*YY1(K)
GXYIM(K) = GXYIM(K) + XX(J)**2 + YY1(J)**2 - XX(K)**2 - YY1(K)**2

90 CONTINUE
GYY(1) = GYY(1) + 4.0* YY1(1)**2
GXYRE(1) = GXYRE(1) + 2.0*(XX(1)*YY1(1))
GXYIM(1) = 0.0

```

CALL ZERO(XTI,1024)
CALL ZERO(YTI,1024)
*
* CALCULATE FREQUENCY RESPONSE OF INPUT AND OUTPUT
* SEPARATE OF THE POWER SPECTRA
*
DO 100 I = 1, 1024
XX(I) = XTR(I)
YY1(I) = Y1TR(I)
100 CONTINUE
*
* WEIGHT TIME RECORD WITH HANNING WINDOW TO ENSURE PERIODICITY
* REQUIREMENTS OF FFT ARE MET XX BEING AN IMPACT MEASUREMENT
* IS PERIODIC BY DEFINITION AND REQUIRES NO WEIGHTING
*
DO 110 I = 1, 512
ITMP = 1025 - I
YY1(I) = YY1(I) * WEGHT(I)
YY1(ITMP) = YY1(ITMP) * WEGHT(I)
110 CONTINUE
CALL FFT842(0,1024,XX,XTI,OUTFILE)
CALL FFT842(0,1024,YY1,Y1TI,OUTFILE)
TEMP = 1/FLOAT(NOSAMP)
DO 120 I = 1, 513
XFR(I) = XFR(I) + XX(I)*TEMP
XFI(I) = XFI(I) + XTI(I)*TEMP
Y1FR(I) = Y1FR(I) + YY1(I)*TEMP
Y1FI(I) = Y1FI(I) + Y1TI(I)*TEMP
120 CONTINUE
*
* TAKE NEXT DATA SET
*
250 CONTINUE
*
* WRITE DATA TO HARD DISK
*
DO 130 I = 1, 513
WRITE(7,9990) GXX(I),GYY(I),GXYRE(I),GXYIM(I)
130 CONTINUE
9990 FORMAT(1X,G16.10,1X,G16.10,1X,G16.10,1X,G16.10)
*
* WEIGHT POWER SPECTRA INFORMATION TO CORRECT FOR NUMBER OF DATA
SETS
*
CALL COHER(1024,RFREQ,NOSAMP,GXX,GYY,GXYRE,GXYIM,100)
*
* CALCULATE TRANSFER FUNCTION FROM POWER SPECTRA INFORMATION
*
DO 300 I = 1, 512
K = 1025 - I
HFR(I) = GXYRE(I)/GXX(I)
HFR(K) = HFR(I)
HFI(I) = GXYIM(I)/GXX(I)
HFI(K) = HFI(I)
HF(I) = 10* ALOG(SQRT(HFR(I)**2+HFI(I)**2))

```

```

      HF(K) = HF(I)
340  CONTINUE
      *
      *   CALCULATE THE MOBILITY TRANSFER FUNCTION
      *
      DO 350 I = 2, 1024
      HMR(I) = HFR(I)/DELTAFRQ(I)
      HMI(I) = HFI(I)/DELTAFRQ(I)
      HM(I) = 10 * ALOG(SQRT(HMR(I)**2+HMI(I)**2))
350  CONTINUE
      *
      *   CALCULATE THE FREQUENCY DOMAIN OF THE TIME SIGNALS
      *
      DO 400 I = 1, 513
      XFR(I) = SQRT(XFR(I)**2+XFI(I)**2)
      XFI(I) = 10* ALOG(XFR(I))
      Y1FR(I) = SQRT(Y1FR(I)**2+Y1FI(I)**2)
      Y1FI(I) = 10 * ALOG(Y1FR(I))
400  CONTINUE
      *
      *   DISPLAY INFORMATION ON SCREEN AND PLOT IF DESIRED
      *
      CALL PLOT1(513,HFR,DELTAFRQ,"H(f)","Real","Freq (Hz)","hfreq.g")
      CALL PLOT1(513,HFI,DELTAFRQ,"H(f)","Imag","Freq (Hz)","pfreq.g")
      CALL PLOT1(513,HF,DELTAFRQ,"H(f)","db","Freq (Hz)","Hf.g")
      CALL PLOT1(513,HMR,DELTAFRQ,"Mobility","Real","Freq (Hz)","Hft.g")
      CALL PLOT1(513,HMI,DELTAFRQ,"Mobility","Imag","Freq (Hz)","Hft.g")
      CALL PLOT1(513,XFR,DELTAFRQ,"X(f)","Mag","Freq (Hz)","xf.g")
      CALL PLOT1(513,XFI,DELTAFRQ,"X(f)","db","Freq (Hz)","logxf.g")
      CALL PLOT1(513,Y1FR,DELTAFRQ,"Y(f)","Mag","Freq (Hz)","yf.g")
      CALL PLOT1(513,Y1FI,DELTAFRQ,"Y(f)","db","Freq (Hz)","logyf.g")
      STOP
      END
      *
      *   SUBROUTINE TO TAKE FILLED BUFFERS FROM A/D CONVERTER AND
      *   CHECK TO SEE IF IMPACT HAS OCCURED. IF SO REMOVE DC COMPONENT
      *   AND TRANSFER TO ARRAY FOR FURTHER REDUCTION AND RETURN BUFFER
      *   TO READY BUFFER QUEUE AT A/D CONVERTER. IF NOT RETURN BUFFER
      *   TO READY BUFFER QUEUE AT A/D CONVERTER.
      *
      SUBROUTINE ADSERVICE
      INTEGER      K
      INTEGER      INDEX
      INTEGER      POWER
      INTEGER      ADPATH
      INTEGER      TEMP
      INTEGER      NCHAN
      INTEGER      MYBUFSZ
      INTEGER*2    DATABLK(100000)
      INTEGER      GETINDEX
      PARAMETER    (GETINDEX = 3)
      REAL         DUMMY
      REAL         AVG(5)
      REAL         DUMMY1
      REAL         THOLD

```

```

REAL      XTR(1024)
REAL      Y1TR(1024)
COMMON /BCR/ XTR, Y1TR, ADPATH, DATABLK, MYBUFSZ, NCHAN, AVG
TEMP = ADPATH
CALL MRBUFGET (1, GETINDEX, INDEX)
*
*   AVERAGE DATA SET TO DETERMINE DC OFFSET OF INFORMATION
*
IF (AVG(1) .EQ. 0.00) THEN
  DO 250 K = INDEX , INDEX + ( 99 * NCHAN), NCHAN
    AVG(1) = AVG(1) + FLOAT(DATABLK(K))/(2*20480)
    AVG(2) = AVG(2) + FLOAT(DATABLK(K+1))/(2*20480)
250  CONTINUE
  END IF
*
*   SET THRESHOLD TO 0.25V ABOVESET OF HAMMER
*
THOLD = AVG(1) + 0.25
DO 300 K = INDEX, INDEX + MYBUFSZ, NCHAN
  DUMMY = FLOAT(DATABLK(K))/(2*204.8)
*
*   IF THRESHOLD IS EXCEEDED GO BACK 200 INDEXES AND RECORD
*   INFORMATION TO ARRAYS FOR FURTHER REDUCTION
*
IF ( DUMMY .GE. THOLD) THEN
  DO 350 I = -200 , 0, NCHAN
    J = K + I
    L = (I/2) + 101
*
*   ENSURE BUFLAP IS CORRECT
*
    IF (J .LE. 0) THEN
      J = J + 100000
    END IF
    DUMMY = FLOAT(DATABLK(J))
    DUMMY1 = FLOAT(DATABLK(J+1))
    XTR(L) = (DUMMY/(2*204.8))-AVG(1)
    Y1TR(L) = (DUMMY1/(2*204.8))-AVG(2)
350  CONTINUE
    I = 50
*
*   READ IN REMAINING INFORMATION INTO DATA SETS FOR REDUCTION
*
  DO 400 J = K, K + 975*NCHAN, NCHAN
    I = I + 1
    DUMMY = FLOAT(DATABLK(J))
    DUMMY1 = FLOAT(DATABLK(J+1))
    XTR(I) = (DUMMY/(2*204.8))-AVG(1)
    Y1TR(I) = (DUMMY1/(2*204.8))-AVG(2)
400  CONTINUE
*
*   SET INDEX TO EXIT DATA TRANSFER LOOP
*
  K = INDEX + MYBUFSZ + 1
  END IF

```

```

300 CONTINUE
*
*   RELEASE BUFFER TO READY BUFFER QUEUE FOR A/D CONVERTER
*
  ADPATH = TEMP
  CALL MRBUFREL (TEMP, DATBLK(INDEX))
  RETURN
  END
  SUBROUTINE COHER(NNN,ISR,NDSJP,GXX,GYY,GXYRE,GXYIM,SFX,SFY)
C
C-----
C MAIN PROGRAM: A COHERENCE AND CROSS SPECTRAL ESTIMATION PROGRAM
C AUTHORS:      G. C. CARTER, J. F. FERRIE
C              NAVAL UNDERWATER SYSTEMS CENTER
C              NEW LONDON, CONNECTICUT 06320
C MODIFIED BY  R.A. SHAFER, LT USN
C              NAVAL POSTGRADUATE SCHOOL
C              MONTEREY CA
C              APRIL 1987
C              TO MEET THE REQUIREMENTS FOR ANALYSIS
C              OF REAL TIME DATA
C INPUT:       NNN IS THE NUMBER OF DATA POINTS PER SEGMENT
C              4 < NNN < 1025
C              ISR IS THE SAMPLING RATE
C              NDSJP IS THE NUMBER OF DISJOINT SEGMENTS
C              SFX IS THE SCALE FACTOR FOR THE INPUT DATA STORED IN
C              THE XX ARRAY
C              SFY IS THE SCALE FACTOR FOR THE INPUT DATA STORED IN
C              THE YY ARRAY
C-----
C
C SPECIFICATION AND TYPE STATEMENTS
C
  REAL ISR
  REAL XX(1024), YY(1024)
  REAL GXX(513), GYY(513), GXYRE(513), GXYIM(513)
  REAL WEGHT(513), PHI(513)
  REAL LINE(50)
  REAL DELTAFRQ(1024),TYME(1024)
  EQUIVALENCE (WEGI(1))

  COMMON /CCSE/ TYME, DELTAFRQ, OUTFILE
C
C NNN IS THE NUMBER OF DATA POINTS PER SEGMENT
C ISR IS THE SAMPLING RATE
C NDSJP IS THE NUMBER OF DISJOINT SEGMENTS
C SFX AND SFY ARE SCALE FACTORS FOR THE INPUT DATA

  NFFTS = NDSJP
  SMALL = .1E-20
C
C CALCULATE CONSTANTS
C
  TPI = 8.0*ATAN(1.0)
  DEG = 360.0/TPI

```

```

VARX = 0.0
VARY = 0.0
DT = 1.0/ FLOAT(ISR)
SF = SQRT(ABS(SFX*SFY))
NPFFT = NNN
NNNP1 = NNN + 1
NNND2 = NNN/2
NND21 = NNND2 + 1
NP2 = NPFFT + 2
ND2 = NPFFT/2
ND2P1 = ND2 + 1
DF = 1.0/(DT*FLOAT(NPFFT))
FNYQ = FLOAT(ISR)/2.0
CONST = 0.25*DT/FLOAT(NNN)
FLOW = 0.0
FHIGH = FNYQ
ISTRT = IFIX(FLOW/DF) + 1
ISTOP = IFIX(FHIGH/DF) + 1
C
C NORMALIZE ESTIMATES
C
FNSG = FLOAT(NFFTS)
OFNSG = 1.0/FNSG
TEMP1 = CONST*OFNSG*SFX
TEMP2 = CONST*OFNSG*SFY
TEMP4 = CONST*OFNSG*SF
TEMP3 = 2.0*TEMP4
DO 90 K=1,ND2P1
GXX(K) = GXX(K)*TEMP1
GYG(K) = GYG(K)*TEMP2
GXYRE(K) = GXYRE(K)*TEMP3
GXYIM(K) = GXYIM(K)*TEMP4
90 CONTINUE
VARX = 0.0
VARY = 0.0
DO 100 K=1,ND2P1
VARX = VARX + GXX(K)
VARY = VARY + GYG(K)
100 CONTINUE
VARX = VARX*DF*2.0/SFX
VARY = VARY*DF*2.0/SFY
C
C CONVERT GXX TO DB AND PLOT
C
DO 110 I=1,ND2P1
XX(I) = GXX(I)
PHI(I) = 10.0*ALOG10(AMAX1(GXX(I),SMALL))
110 CONTINUE
CALL PLOT1(ND2P1,PHI,DELTAFRQ,"Gxx (f)","db","FREQ (Hz)","loggxx.g")
C
C CONVERT GYG TO DB AND PLOT
C
DO 140 I=1,ND2P1
XX(I) = GYG(I)
PHI(I) = 10.0*ALOG10(AMAX1(GYG(I),SMALL))

```

```

140 CONTINUE
    CALL PLOT1(ND2P1, PHI, DELTAFRQ, "Gyy(f) [db], FREQ (Hz)", gyy, g)
C
C COMPUTE CROSS SPECTRUM AND MAGNITUDE SQUARED COHERENCE
C
    DO 230 K=1,ND2P1
        PHI(K) = GXYRE(K)**2 + GXYIM(K)**2
        XX(K) = SQRT(ABS(PHI(K)/(GXX(K)*GYY(K))))
230 CONTINUE
    CALL PLOT1(ND2P1, XX, DELTAFRQ, "Coherence", " ", "Freq (Hz)", "coher g")
C
C TERMINATE PROGRAM
C
    RETURN
    END
C-----
C SUBROUTINE: ZERO
C THIS SUBROUTINE STORES IN A FLOATING POINT ARRAY
C-----
C
    SUBROUTINE ZERO(ARRAY, NUMBR)
C
C INPUT: ARRAY = AN ARRAY OF FLOATING POINT VALUES TO BE
C         ZERO FILLED
C        NUMBR = NUMBER OF ARRAY VALUES
C
    DIMENSION ARRAY(1)
C
    DO 10 K=1,NUMBR
        ARRAY(K) = 0.0
10 CONTINUE
    RETURN
    END

```

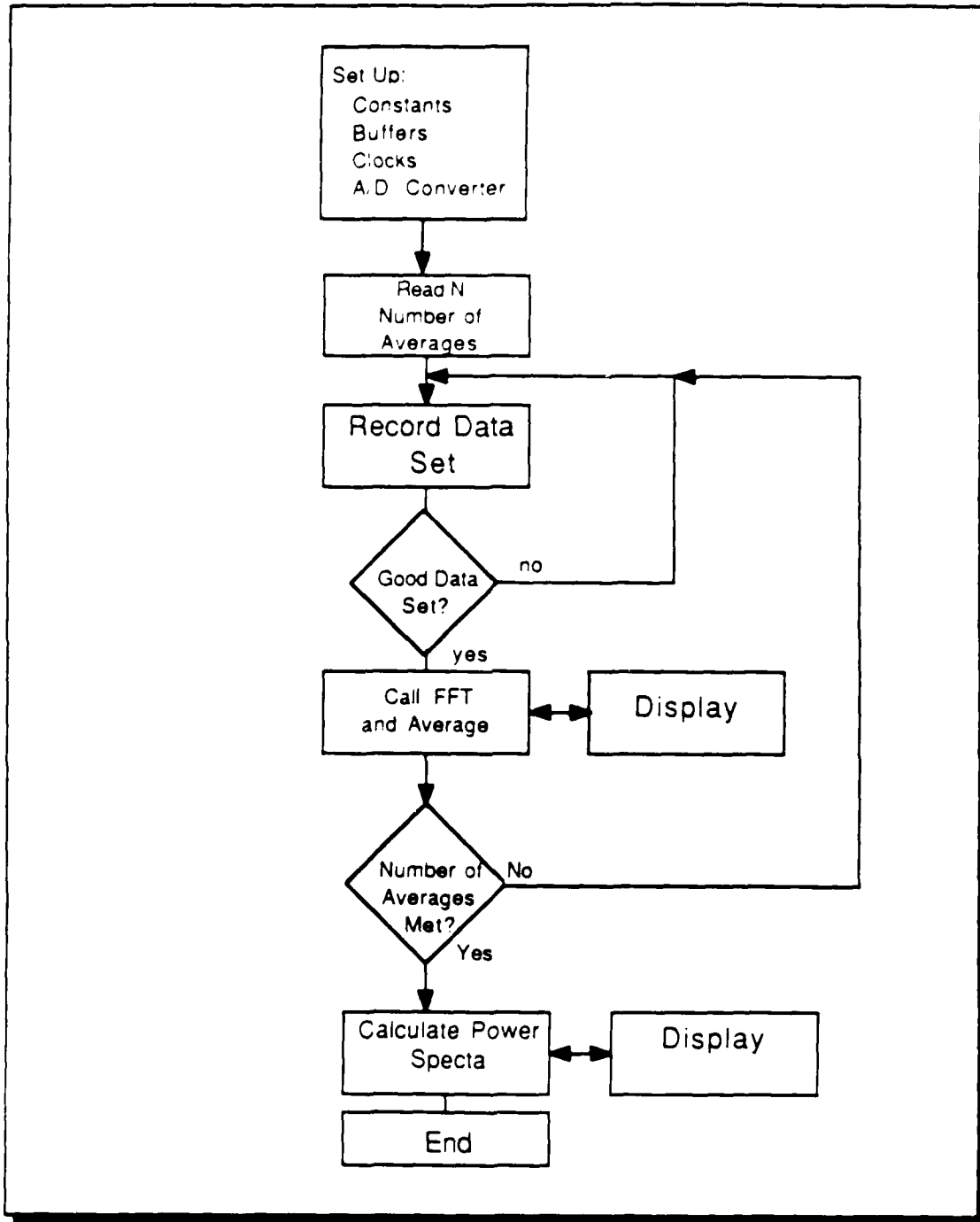


Figure 47: Impact Testing Data Acquisition and Reduction Flow Diagram

APPENDIX D

STEADY STATE VIBRATION DATA ACQUISITION AND REDUCTION PROGRAM

This Appendix list the computer program utilized to acquire the data for steady state vibrations analysis. All routines were written by the author except where noted. For further explanation of the data acquisition specific routines the reader is encouraged to refer to the MASSCOMP documentation "Data Acquisition User's Manual". Figure 48 is the program flow diagram. Sections of the program follow the general outline presented in the flow diagram. Specific subroutines for plotting the information are not shown in this Appendix.

*
* THIS PROGRAM IS THE MAIN PROGRAM FOR DATA ACQUISITION
* FOR STEADY STATE ANALYSIS OF A VIBRATING BEAM

*
*
* 07 MAY 1987
*
*

INTEGER MYBUFSZ, NBUFFS
INTEGER NEARFREQ, RONLY
INTEGER RDWRT, SQUARE
INTEGER STARTLO, TIMEOUT
REAL FREQ, WIDTH
REAL PI
INTEGER OUTFILE
PARAMETER (NBUFFS = 5)
PARAMETER (NEARFREQ = 0)
PARAMETER (RONLY = 1)
PARAMETER (RDWRT = 0)
PARAMETER (SQUARE = 4)
PARAMETER (STARTLO = 0)
PARAMETER (TIMEOUT = 30000)
PARAMETER (WIDTH = 0.0)
INTEGER*2 DATBLK(40960)
INTEGER DUMMY
EQUIVALENCE (DATBLK,DUMMY)
EXTERNAL ADSERVICE
INTEGER ADPATH
INTEGER CLKPATH(2)
INTEGER STATWDS(2)
INTEGER I

```

      INTEGER      NCHAN
      INTEGER      FCHAN
      INTEGER      SCHAN
      INTEGER      GAIN
      INTEGER      INDEX
      INTEGER      POWER
REAL      TPI
      REAL      WEGHT(1025)
      REAL      TYME(2048)
      REAL      RECLN
      REAL      RFREQ
      REAL      RWIDTH
      REAL      XTR(2048), Y1TR(2048)
      REAL      Y1TI(2048)
      REAL      XX(2048)
      REAL      YY1(2048)
      REAL      XFR(2048),XFI(2048)
      REAL      Y1FR(2048),Y1FI(2048)
      REAL      GXX(1025)
      REAL      GYY(1025)
      REAL      GXYRE(1025)
      REAL      GXYIM(1025)
      REAL      HWIN(2048)
      REAL      DELTAFRQ(2048)
      REAL      DFREQ
      REAL      MAG1(1025)
      REAL      MAG2(1025)
      REAL      MAG3(1025)
      REAL      HFR(2048),HFI(2048)
      REAL      HR(2048),HI(2048)
      REAL      AVG(5)
      REAL      PI2, TEMP
      INTEGER      AVGSAMP
      CHARACTER    ANSWER
      COMMON /BCR/ XTR, Y1TR, ADPATH, DATABLK, MYBUFSZ, NCHAN, AVG
      COMMON /CCSE/ TYME, DELTAFRQ, OUTFILE
*
*   OPEN DATA FILES FOR DATA STORAGE
*
      OUTFILE = 8
      OPEN (7,file="/usr/shafer/beam/simuldata")
*
*   CALCULATE WINDOWS AND CONSTANTS
*
      TPI = 8.0 * ATAN(1.0)
      PI = 4.0 * ATAN(1.0)
      TEMP = TPI / 2049.0
      SCL = SQRT(2.0/3.0)
      DO 40 I = 1,1024
          WEGHT(I) = SCL * (1.0 - COS(TEMP*FLOAT(I)))
40      CONTINUE
      PI2 = 2.0 * ATAN(1.0)
      TEMP = PI2 / 512.0
      DO 46 I = 2, 1024
          HWIN(I) = 2.0

```

```

46      CONTINUE
        DO 47 I = 1025,2048
          HWIN(I) = 0.0
47      CONTINUE
        HWIN(1) = 1.0
*
*      SET UP A/D CONVERTER AND BUFFERS FOR TEMPORARY STORAGE OF
DATA
*
        MYBUFSZ = 4096
        NCHAN = 2
        FCHAN = 16
        SCHAN = 1
        GAIN = 0
        ADPATH = -1
        CALL MROPEN(ADPATH, "/dev/dacp0/adf1", RONLY)
        CALL MRADINC(ADPATH, FCHAN, NCHAN, SCHAN, GAIN)
        CALL MRADMOD(ADPATH, 0, 0)
*
*      IDENTIFY BUFFERS AND RELEASE THEM TO READY BUFFER QUEUE
*
        DO 200 I = 1, 36865, MYBUFSZ
          CALL MRBUFID (ADPATH, DATBLK(I), MYBUFSZ*2)
          CALL MRBUFREL (ADPATH, DATBLK(I))
200     CONTINUE
*
*      SET UP CLOCK FOR A/D CONVERTER
*
        CLKPATH(1) = -1
        CLKPATH(2) = -1
        CALL MROPEN (CLKPATH(1), "/dev/dacp0/efclk0", RDWRT)
        CALL MROPEN (CLKPATH(2), "/dev/dacp0/efclk4", RDWRT)
        FREQ = 1000.
        PRINT *, "SETTING UP CLOCKS"
        CALL
MRCLK2(CLKPATH(1),CLKPATH(2),0,0,FREQ,RFREQ,0,200000.,RWIDTH,2,0)
        CALL MRCLKTRIG (ADPATH, 2,CLKPATH)
*
*      COMPUTE FREQUENCY RESOLUTION AND FILL FREQUENCY ARRAY
*      DISPLAY RESOLUTION ON SCREEN
*
        RFREQ = RFREQ
        PRINT *, "Actual sampling frequency per channel -- ",RFREQ
        PRINT *, "INPUT NUMBER OF AVERAGES TO BE COMPUTED"
        READ *,AVGSAMP
        NITEMS = 4096
        RECLN = 2048/RFREQ
        PRINT *, "TIME RECORD LENGTH IS PER CHANNEL IS ",RECLN
        DFREQ= 1/RECLN
        PRINT *, " FREQUENCY RESOLUTION IS ",DFREQ

        DO 60 I = 1, 2048
          TYME(I) = I / RFREQ
          DELTAFRQ(I) = (I-1)*DFREQ
60      CONTINUE

```

```

*
*   MAIN LOOP FOR DATA ACQUISITION AND PRELIMINARY REDUCTION
*
DO 250 L = 1 , AVGSAMP
CALL MRXINQ (ADPATH, MYBUFSZ, NITEMS, ADSERVICE)
CALL MREVWT (ADPATH, STATWDS, 0 )
PRINT *, AVG(3)

DO 70 I = 1, 2048
XX(I) = XTR(I)
YY1(I) = Y1TR(I)
70  CONTINUE
*
*   WEIGHT TIME RECORDS WITH HANNING WINDOW
*
DO 80 I = 1, 1024
ITMP = 2049 - I
XX(I) = XX(I) * WEGHT(I)
YY1(I) = YY1(I) * WEGHT(I)
XX(ITMP) = XX(ITMP) * WEGHT(I)
YY1(ITMP) = YY1(ITMP) * WEGHT(I)
80  CONTINUE
*
*   COMPUTE FORWARD FFT
*
CALL FFT842(0, 2048, XX, YY1,OUTFILE)
*
*   COMPUTE SPECTRA
*
GXX(1) = GXX(1) + 4.0*XX(1)**2
DO 90 K=2, 1025
  J = 2050 - K
  GXX(K) = GXX(K) + (XX(K)+XX(J))**2 + (YY1(K)-YY1(J))**2
  GYY(K) = GYY(K) + (YY1(K)+YY1(J))**2 + (XX(J)-XX(K))**2
  GXYRE(K) = GXYRE(K) + XX(K)*YY1(J) + XX(J)*YY1(K)
  GXYIM(K) = GXYIM(K) +XX(J)**2 +YY1(J)**2 -XX(K)**2 -YY1(K)**2
90  CONTINUE
GYY(1) = GYY(1) + 4.0*YY1(1)**2
GXYRE(1) = GXYRE(1) + 2.0*(XX(1)*YY1(1))
GXYIM(1) = 0.0

250  CONTINUE
*
*   DATA COLLECTION COMPLETE, COMPLETE REDUCTION AND DISPLAY
RESULTS
*
CALL COHER1(2048,RFREQ,AVGSAMP,GXX,GYY,GXYRE,GXYIM,1.0,1.0)
*
*   CALCULATE FREQUENCY RESPONSE (MOBILITY)
*
DO 600 I = 2, 1024
K = 2049 - I
HFR(I) = (GXYRE(I)/GXX(I))/DELTAFRQ(I)
HFR(K) = HFR(I)

```

```

      HFI(I) = (GXYIM(I)/GXX(I))/DELTAFRQ(I)
      HFI(K) = HFI(I)
600  CONTINUE
      HFR(1) = GXYRE(1)/GXX(1)
      HFR(2048) = HFR(1)
      HFI(1) = GXYIM(1)/GXX(1)
      HFI(2048) = HFI(1)
*
*   RECORD POWER SPECTRA INFORMATION TO DISK
*
      DO 700 I = 1, 1025
      WRITE (7,9990) GXX(I),GYY(I),GXYRE(I),GXYIM(I)
700  CONTINUE
9990  FORMAT (1X, G16.8,G16.8,G16.8,G16.8)
*
*   PLOT RESULTS
*
      CALL PLOT1(1024,HFR,DELTAFRQ,"H(f)","Real","Freq (Hz)","Hfr(g)")
      CALL PLOT1(1024,HFI,DELTAFRQ,"H(f)","Imag","Freq (Hz)","Hfi(g)")
      STOP
      END
      SUBROUTINE COHER1(NNN,ISR,NDSJP,GXX,GYY,GXYRE,GXYIM,SFX,SFY)
C
C-----
C MAIN PROGRAM: A COHERENCE AND CROSS SPECTRAL ESTIMATION PROGRAM
C AUTHORS:      G. C. CARTER, J. F. FERRIE
C              NAVAL UNDERWATER SYSTEMS CENTER
C              NEW LONDON, CONNECTICUT 06320
C MODIFIED BY  R. A. SHAFER, LT USN
C              NAVAL POSTGRADUATE SCHOOL
C              MONTEREY CA
C              APRIL 1987
C              TO MEET THE REQUIREMENTS FOR ANALYSIS
C              OF REAL TIME DATA
C INPUT:      NNN IS THE NUMBER OF DATA POINTS PER SEGMENT
C              4 < NNN < 1025
C              ISR IS THE SAMPLING RATE
C              NDSJP IS THE NUMBER OF DISJOINT SEGMENTS
C              SFX IS THE SCALE FACTOR FOR THE INPUT DATA STORED IN
C              THE XX ARRAY
C              SFY IS THE SCALE FACTOR FOR THE INPUT DATA STORED IN
C              THE YY ARRAY
C-----
C
C SPECIFICATION AND TYPE STATEMENTS
C
      REAL ISR
      REAL XX(2048), YY(2048)
      REAL GXX(1025), GYY(1025), GXYRE(1025), GXYIM(1025)
      REAL WEGHT(1025), PHI(1025)
      REAL LINE(50)
      REAL DELTAFRQ(2048), Tyme(2048)
      EQUIVALENCE (WEGHT(1),PHI(1))

      COMMON /CCSE/ Tyme, DELTAFRQ, OUTFILE

```

```

C
C NNN IS THE NUMBER OF DATA POINTS PER SEGMENT
C ISR IS THE SAMPLING RATE
C NDSJP IS THE NUMBER OF DISJOINT SEGMENTS
C SFX AND SFY ARE SCALE FACTORS FOR THE INPUT DATA

```

```

      NFFTS = NDSJP
      SMALL = .1E-20

```

```

C
C CALCULATE CONSTANTS

```

```

C
      TPI = 8.0*ATAN(1.0)
      DEG = 360.0/TPI
      VARX = 0.0
      VARY = 0.0
      DT = 1.0/FLOAT(ISR)
      SF = SQRT(ABS(SFX*SFY))
      NPFFT = NNN
      NNNP1 = NNN + 1
      NNND2 = NNN/2
      NND21 = NNND2 + 1
      NP2 = NPFFT + 2
      ND2 = NPFFT/2
      ND2P1 = ND2 + 1
      DF = 1.0/(DT*FLOAT(NPFFT))
      FNYQ = FLOAT(ISR)/2.0
      CONST = 0.25*DT/FLOAT(NNN)
      FLOW = 0.0
      FHIGH = FNYQ
      ISTRT = IFIX(FLOW/DF) + 1
      ISTOP = IFIX(FHIGH/DF) + 1

```

```

C
C NORMALIZE ESTIMATES

```

```

C
      FNNG = FLOAT(NFFTS)
      OFNNG = 1.0/FNNG
      TEMP1 = CONST*OFNNG*SFX
      TEMP2 = CONST*OFNNG*SFY
      TEMP4 = CONST*OFNNG*SF
      TEMP3 = 2.0*TEMP4
      DO 90 K=1,ND2P1
      GXX(K) = GXX(K)*TEMP1
      GYY(K) = GYY(K)*TEMP2
      GXYRE(K) = GXYRE(K)*TEMP3
      GXYIM(K) = GXYIM(K)*TEMP4
90 CONTINUE
      VARX = 0.0
      VARY = 0.0
      DO 100 K=1,ND2P1
      VARX = VARX + GXX(K)
      VARY = VARY + GYY(K)
100 CONTINUE
      VARX = VARX*DF*2.0/SFX
      VARY = VARY*DF*2.0/SFY

```

```

C

```

```

C CONVERT GXX TO DB AND PLOT
C
DO 110 I=1,ND2P1
  XX(I) = GXX(I)
  PHI(I) = 100*ALOG10(AMAX1(GXX(I),SMALL))
110 CONTINUE
CALL PLOT1(ND2P1,PHI,DELTA FRQ,"GXX (db)", " ", " ", " ")
C
C CONVERT GYY TO DB AND PLOT
C
DO 140 I=1,ND2P1
  YY(I) = GYY(I)
  PHI(I) = 100*ALOG10(AMAX1(GYY(I),SMALL))
140 CONTINUE
CALL PLOT1(ND2P1,PHI,DELTA FRQ,"GYY (db)", " ", " ", " ")
C
C COMPUTE AND DISPLAY PHASE FROM MAXIMUM TO MINIMUM
C
GXYIM(1) = 0.0
PHI(1) = 0.0
DO 200 K=2,ND2P1
  XX(K) = GXYRE(K)
  IF (XX(K) > 190, 170, 180)
170 IF (GXYIM(K) > 190, 180, 190)
180 XX(K) = 1.0
190 PHI(K) = DEG*ATAN2(GXYIM(K),XX(K))
200 CONTINUE
C
C PLOT PHASE FROM -PHLIM TO PHLIM
C
PHLIM = 180.0
DO 210 K=2,ND2P1
  X = PHI(K) - PHI(K-1)
  PHI(K) = PHI(K) - SIGN(360.,X)*AINT(0.5+ABS(X)/360.)
  IF (PHI(K).GT.PHLIM) PHI(K) = PHI(K) - PHLIM
  IF (PHI(K).LT.(-PHLIM)) PHI(K) = PHI(K) + PHLIM
210 CONTINUE
C
C COMPUTE CROSS SPECTRUM AND MAGNITUDE SQUARED COHERENCE
C
DO 230 K=1,ND2P1
  PHI(K) = GXYRE(K)**2 + GXYIM(K)**2
  XX(K) = SQRT(ABS(PHI(K)/(GXX(K)*GYY(K))))
230 CONTINUE
CALL PLOT1(ND2P1,XX,DELTA FRQ,"Coherence", " ", "Freq (hz)", "coher g")
C
C CONVERT GXY TO DB AND PLOT
C
DO 250 I=1,ND2P1
  PHI(I) = 5.0*ALOG10(AMAX1(PHI(I),SMALL))
250 CONTINUE
C
C TERMINATE PROGRAM
C
RETURN

```


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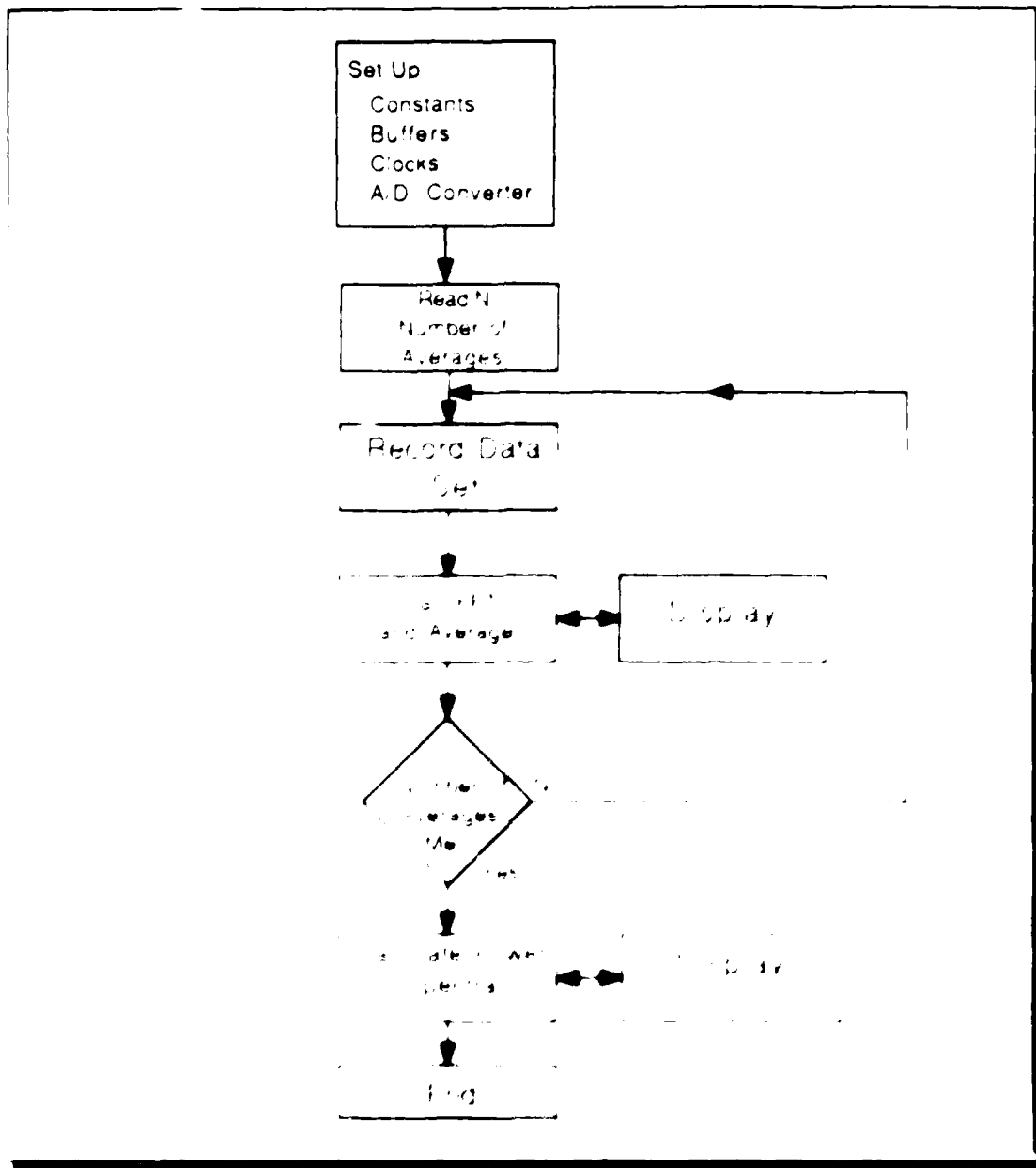


Figure 48. Steady State Vibrations Data Acquisition and Reduction Program Flow Diagram

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