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APPLICATIONS OF SIGNAL PROCESSING TECHNIQUES IN
MICROPROCESSOR BASED INSTRUMENTATION

by

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B.S., College of Engineering, Guindy, Madras, India, 1982

A MASTER'S THESIS

Submitted in partial fulfillment of the
requirements for the degree

MASTER OF SCIENCE

Department of Electrical Engineering

KANSAS STATE UNIVERSITY

Manhattan, Kansas

1984

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CHAPTER 1

INTRODUCTION

Signal processing techniques have definite applications in the area of microprocessor or computer-based instrumentation. A new approach has been evolved wherein a microprocessor-based system can accomplish data acquisition with subsequent processing by a computer utilising suitable processing algorithms. The algorithms used depend on what exactly needs to be measured or determined - a specific example considered here being the measurement of phase. Such a system possesses the inherent advantages of speed, automation, flexibility in type of algorithm used for the specific problem, and enormous computing and processing capabilities coupled with all the advantages which discrete time signal processing can offer.

A microprocessor based phasemeter has been developed which acquires data samples of the original signals and sends them to a computer for subsequent processing by suitable algorithms. These algorithms are based on Adaptive Filtering, FFT, Correlation and Numerical methods. The system was implemented on a North Star microcomputer in conjunction with an MC 6802 interface and HP Multiprogrammer; the whole forming an automated test system. The MC 6802 interface serves as a communication link between the North Star (NS) and HP Multiprogrammer (HPM). The sampling of the signals is initiated by a command from the NS software which invokes an assembly routine in the MC 6802 interface and uses the High Speed Analog to Digital converter (ADC) card on the HPM. These samples are then sent serially to the NS and subsequent

processing takes place by the algorithms stated above to determine the phase difference. The HP 9845B Desktop computer was used to study theoretical simulation results and for the development of algorithms.

Conventional phasemeters operate on the principle of measuring the time difference between the zero crossings of the two signals by "squaring up" the signals. Complex and sophisticated hardware consisting of amplifiers, precision reference generators, filters and networks are normally used. Also, manual controls and adjustments are required for making measurements. The automated microprocessor based system, however, employs more software and less sophisticated hardware. Though this system may not boast of the same accuracy obtainable with the dedicated and sophisticated phasemeter, it still has certain advantages and is capable of further development and improvement. The relative merits and demerits are discussed later.

Other applications can also be found for the system discussed above, examples being spectral estimation (FDS), frequency domain analyses, filtering, combating noise and determination of RMS value etc. Thus this project represents an initial step in a novel direction and the author is optimistic that further development and sophistication is possible in future.

CHAPTER 2

SOME PRELIMINARY CONSIDERATIONS

The specific application of signal processing techniques in phase measurement has been selected. There are some questions to be answered (before we proceed further) such as (a) Why measure phase? (b) What are the conventional methods of measurement? (c) How are the proposed methods different and what advantages do they have over the conventional methods.

The measurement of phase is accomplished by a phasemeter. Such measurements find applications in impedance meters, the measurement of amplifier phase shift, the transfer functions of networks, the measurement of group delay, the measurement of distance and directivity, and the testing and adjusting of filter networks.

The principle of phase measurement used in modern phase instrumentation is illustrated in fig. 2.1. through fig. 2.4. If "t" is the time interval between analogous zero crossings, and "T" is the wave period, then phase angle $P=360*(t/T)$ degrees. The signals are then squared up as shown in fig. 2.2. Most phasemeters use the squared off waves to generate the waveshape shown in fig. 2.3. The Dranetz 305 is one of a series of sophisticated phasemeters available. This phasemeter, after generating the waveform in fig. 2.3. standardizes the waveform amplitude and then filters the waveform to yield the DC value.

The filtering in the simplest case is accomplished by a low pass filter ie. an RC network. The time constant of this filter must be large enough to provide an accurate measurement.

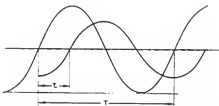


Fig.2.1. Signals with a certain phase difference.

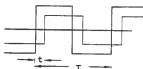


Fig.2.2. Clipping or squaring up the signals.



Fig.2.3. Triggering on and off at the zero crossings.



Fig.2.4. Standardizing the waveform amplitude.

Fig.2.1...2.4. Principle of phase measurement.

Vref, and all DC amplifiers following this circuit and other

circuitry must be stable and accurate with time and temperature. The speed of response is limited by the time constant and hence in practice it is made variable by automatic switching of filters with different time constants.

The presence of noise, interference, high harmonic content or various forms of signal modulation may impair accuracy. Also the input signals may vary appreciably in magnitude. This variation is counteracted by switching attenuators. Thus in the Dranetz system it is found that a great degree of complexity and sophistication in circuitry is used to combat all the above mentioned problems.

Having considered the conventional method of phase measurement attention is now focussed on the proposed techniques. Here the analysis of signals is carried out in the discrete time (DT) domain where we work with "samples" of the original signals as shown in fig. 2.5.

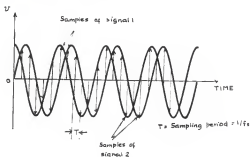


Fig.2.5. Samples of the signals in the DT domain.

It is necessary according to the Nyquist criterion that if

all information in the original signals has to be preserved then the sampling frequency " f_g " must be greater than the frequency of the signals " f_o " that is $f_g > 2 * f_o$. Once the signals are characterized by the "samples" then the next step is to develop some algorithm to process these samples to obtain the required characteristic (for example the phase) of the original signals. These samples may be obtained by using the Analog to Digital Converter (ADC).

The ADC produces a digital output proportional to the analog input. The parameters of importance here are the conversion time of the ADC (how fast it can sample) and its resolution (how well or accurately it represents the analog quantity in digital form). Some ADCs that are available have 8 or 12 bits resolution and are capable of moderate sampling rates. The "Flash" type of ADC is available with a high sampling rate (10 Mhz.) and 8 bits resolution. A more detailed discussion on ADCs is found later.

It is necessary at this stage to emphasize the fact that once the signals are "sampled" or "digitized" and the resulting samples are stored in memory then they are absolutely unaffected by any time or temperature changes. Hence the only caution that must be exercised is prior to and during digitizing when the effects of noise, temperature changes, offsets and all the problems associated with hardware exist (as in the conventional phase meter). Thus the single greatest advantage is due to the use of software algorithms which are unaffected by the problems that affect hardware. A computer can then obtain the samples

stored in memory and then process them with the pertinent algorithm and obtain the required result. All the advantages of software such as flexibility, use of statistical methods, computing capabilities become apparent.

The techniques discussed use Adaptive Filtering which uses digital filters (in contrast to the analog RC filters in the conventional phasemeter). Here the coefficients of the filters can be programmed and changes can be rapidly made without changing circuit components. Performance is not affected by wear and aging and is more reliable. The response characteristics can be easily changed by manipulating the coefficients which in turn leads to more flexibility. However, the finite resolution of the ADC, finite precision or word length of the computer, truncation and roundoff errors and overflow/underflow problems are associated with software processing. Nevertheless, by proper hardware/software tradeoffs, it is possible to obtain a suitable operating system.

Thus with these preliminary considerations we now proceed to describe the proposed algorithms.

CHAPTER 3

THE ADAPTIVE FILTERING ALGORITHM

3.1 INTRODUCTION

This method uses a signal processing technique familiarly known as the Widrow's LMS (least mean square) algorithm in the cancellation mode. The schematic is shown in fig 3.1. The two signals $y(k)$ and $x(k)$ form the primary and reference inputs to the adaptive filter configuration. Here $y(k)$ and $x(k)$ are derived by sampling the CT (continuous time) signals $x(t)$ and $y(t)$ and "k" is the discrete time index which represents the sample of the original signals at time $t=k$. The error $e(k)$ that is generated by comparing $g(k)$ and $y(k)$ is used to change the weights of the discrete time (DT) digital filter W such that the square of the mean error $e(k)$ is a minimum. Thus as the processing continues the error dynamically changes the weighting coefficients of the filter so that the error reduces in magnitude. After a sufficient number of processing steps the error tends to a small value, the weights of the filter are then used to calculate the phase difference between $y(k)$ and $x(k)$. Here $y(k)$ is assumed to lead $x(k)$.

3.2 THEORETICAL ASPECTS

Consider the adaptive scheme in fig. 3.1. It is seen that $y(k)$ and $x(k)$ form the primary and reference inputs and W_k is a $N+1$ weight filter. The error $e(k)$ obtained by comparing $y(k)$ with the filter output $g(k)$ is used to update the filter weights or coefficients. This process is carried on until the error reduces

to a small value.

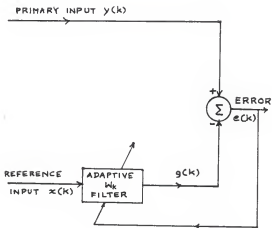


Fig.3.1. Adaptive filtering scheme in the cancellation mode.

Hence we have the following, in vector form:

$W' = (w_0 \ w_1 \ w_2 \ \dots \ w_M)$ considering $M+1$ weights.

$X_k' = (x(k) \ x(k-1) \ x(k-2) \ \dots \ x(k-M))$

where $x(k-1) \dots x(k-M)$ represent the M past values of $x(k)$.

$x(k) = \sin(w_0 kT)$ and $y(k) = \sin(w_0 kT + P)$

Now $g(k) = W'X_k = w_0 x(k) + w_1 x(k-1) + \dots + w_M x(k-M)$

$e(k) = y(k) - g(k)$

When the error $e(k)$ tends to a small value as a result of the adaptive process then $g(k)$ tends to cancel $y(k)$ at the summing device. Hence $g(k)$ "looks like" $y(k)$. At this stage taking the Z transform to obtain the filter transfer function we have the following:

$$G(z) = X(z) [w_0 + w_1 z^{-1} + \dots + w_M z^{-M}]$$

The transfer function $H(z)$ is then

$$H(z) = G(z)/X(z) \text{ and since } z = e^{jw_0^T} \text{ where } w_0 = 2\pi f_0$$

f_0 being the frequency of the signals and T being the sampling period.

Hence the value of transfer function at the frequency of the signals is given by

$$H(w_0) = G(w_0)/X(w_0) = w_0 + w_1 e^{-jw_0^T} + \dots + w_M e^{-jMw_0^T}$$

The phase transfer function $B(w_0) = \arctan[\text{Im}H(w_0)/\text{Re}H(w_0)]$

$$\text{Im}H(w_0) = [w_1 \sin w_0 T + w_2 \sin 2w_0 T + \dots + w_M \sin Mw_0 T]$$

$$\text{Re}H(w_0) = [w_0 + w_1 \cos w_0 T + w_2 \cos 2w_0 T + \dots + w_M \cos Mw_0 T]$$

The value of B at w_0 gives the required phase difference " P " between the two signals $x(k)$ and $y(k)$.

Thus from the above discussion it is apparent that after a

sufficient number of processing steps (or iterations) when the error decreases the phase difference "P" can be directly determined from the coefficients of the filter W. It must be noted that in the above expression for B it is required to know the values of f_0 and T for calculating the phase P.

3.3 SIMULATION AND VERIFICATION

As a first step, software development for the method was done on the North Star microcomputer system. The two signals $x(k)$ and $y(k)$ were generated as $x(k)=\sin(\omega_0 kT)$ and $y(k)=\sin(\omega_0 kT+P)$ where $\omega = 2\pi f_0$, f_0 being the frequency of the signals, T the sampling period, k the time index and P the phase difference in radians. The Widrow algorithm was implemented in software. The flow chart and BASIC program listing are found in Appendix 1a and 1b respectively.

The procedure followed is essentially iterative in nature and the weights are updated as follows for a M+1 weight filter.

$$w_m(k+1) = w_m(k) + v e(k) x(k-m); \quad 0 \leq m \leq M+1$$

where v is the convergence parameter and this parameter is dynamically updated as follows:

$$v = \alpha / \sigma_e^2(k); \quad \sigma_e^2(k) = \beta \sigma_e^2(k-1) + (1-\beta) e^2(k); \quad \sigma_e^2(0) = 0$$

The parameter α must be chosen with care for proper convergence and the phase is calculated as P given by the formula for B in section 3.2.

The most important parameters that are involved in this method are the following:

- a. the frequency of the signals (f_0)
- b. the sampling frequency ($f_s=1/T$)
- c. the convergence parameter (α)
- d. the phase difference involved (P)
- e. the number of iterations (K)
- f. the ratio (f_s/f_0)

Some of the important results obtained after considerable experimentation with the parameters stated above are as follows.

1. The theoretical value of sampling frequency $f_s > 2f_0$, but it was found that $f_s > 3f_0$ for proper working of the method.
2. For a 3 weight filter and with just 30 iterations convergence was established when the value of sampling frequency was equal to four times the signal frequency i.e. $f_s = 4f_0$ and the calculated phase agreed very closely with the actual phase (absolute error less than 0.2 degrees.)
3. However with a 16 weight filter convergence was easily obtained for a wide range of f_s/f_0 .
4. One of the theoretical aspects pertaining to convergence is that a greater number of weights leads to faster convergence. Also the convergence time is also dependent on the convergence parameter as $\tau = 1/\alpha = 1/(1-\beta)$.
5. Also the faster the convergence the greater is the residual error. Too fast a convergence can overdrive the filter W causing large residual errors. (i.e. magnitude of $e(k)$).
6. Hence the number of weights and the convergence parameter must be chosen carefully to obtain a small residual error and

hence a small error in the calculated phase.

7. The convergence also depends on the type of input.
8. The method works for any two zero mean signals.
9. Since the filter by itself cannot distinguish a phase shift of more than 90° , suitable modifications were made in the software which fixes the quadrant with respect to the sign of the numerator and denominator in the equation for phase as already shown. A phase of 360° could then be easily distinguished.
10. The above results were established assuming $y(k)$ leads $x(k)$.

Analysing the working of the method in the frequency domain, it can be said that the adaptive filter merely adjusts the amplitude and phase of all the components in its input signal $x(k)$ to cancel those in $y(k)$. This is a characteristic of the adaptive process.

3.4 DETERMINATION OF PHASE SIGN

In the above discussion it was assumed that $y(t)$ leads $x(t)$ by a phase difference P . When this is not known we resort to the following method to determine which is the leading or lagging waveform. If during the period $x(t)$ is positive there occurs a negative to positive transition of $y(t)$ then $x(t)$ leads $y(t)$ and if there occurs a positive to negative transition of $y(t)$ then $y(t)$ leads $x(t)$.

However the sign is automatically reflected in the present software. If $y(k)$ and $x(k)$ are interchanged (assuming $y(k)$ leads $x(k)$) then the calculated phase shows a value of $360^\circ - P$ where P is the actual phase difference. Numerical methods presented later

can be used to determine to a moderate degree of accuracy the zero crossing from which the sign could be determined.

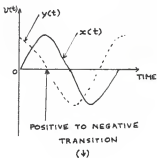
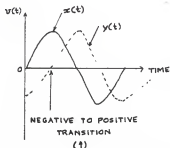


Fig.3.2. Determination of phase sign.

3.5 PRACTICAL ASPECTS AND SYSTEM IMPLEMENTATION

After having established the validity of the method in theory, it was extended to a practical situation. However, as is always the case, the transition from theory to practice is not smooth and many constraints had to be given due importance.

Before we discuss the practical system we must bear in mind the following aspects of the theoretical simulation carried out.

1. The samples generated are correct to 7 decimals.
2. f_0 and $T=1/f_g$ are quite accurate since they are just values given by us.
3. No noise is included in the signals.
4. Phase shift is constant with time.

However in a practical system, the situation is a bit different. The finite resolution of the ADC imposes a constraint on the accuracy with which a sample is represented. For example with a 12 bit ADC the accuracy is limited to around 3 decimal places. The frequency of the signals must be determined by some external means and the sampling frequency must be accurately known. The signals may be derived from a function generator which has some noise in its output and may also have some voltage offsets. The phase of the waveform generated may drift with time. Hence all these must be considered in a practical system.

The basic configuration required for system implementation of the method is shown in fig.3.3.

The ADCs must be controlled to sample both the signals $x(t)$ and $y(t)$ simultaneously. These samples may then be stored in memory. Once stored in memory the samples can be accessed by a

computer or microcomputer for processing the samples with the adaptive algorithm. The CRT terminal acts as an Input/Output device to provide interaction between the user and the system.

There are different ways of realizing the above system but the primary efforts were directed to utilise the facilities available. A practical system using the North Star microcomputer in conjunction with the MC 6802 interface and HP Multiprogrammer (HPM) being used as an automated test system. A brief discussion of the HPM and the interface follows.(fig.3.3.a.)

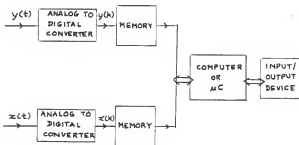


Fig.3.3. System implementation of the adaptive method.

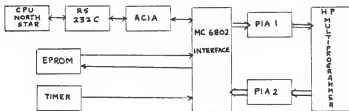


Fig.3.3.a. Schematic of the test system.

The HPM is a master control unit for bidirectional systems. It can be controlled using a computer. It contains many types of accessory cards which can be used depending on the requirement. For example the V/F card in the HPM can be programmed to make voltage or measurements on the device under test (DUT). The HPM also contains a High Speed Analog to Digital Converter card (ADC). Each card can be accessed by addressing the particular slot in which it resides. The various communication protocols are provided by the following four types of words - control word, output data word, input address word and return data word.

The MC 6802 interface acts as a communication link between the North Star (NS) and the HPM. There exists an assembly routine in the interface EPROM that can address the ADC in the HPM, obtain the samples and store them in the memory available in the interface RAM (1K). The procedure is to first set-up the HPM in the input mode. Then the ADC card is accessed by the proper slot address. The conversions are initiated by a gate signal and the end of conversion is indicated by a flag. The routine makes use of the above procedure to obtain the samples.

The ADC card is capable of 22K conversions per second. However due to the inherent limitations posed by the software assembly routine the present sampling frequency is 9130 samples per second. The number of samples that can be taken is fixed at 512. The ADC has three operating ranges ($\pm 10V$, $\pm 1V$, $\pm 100mV$). The interface accomplishes sampling of the signal and stores in memory. It then sends the data samples to the NS using the serial port. Thus the sampling is initiated by a command in the BASIC

program which invokes the assembly routine in the interface EPROM and the samples are then sent to the NS memory. The schematic of the interface and connections are shown in Appendix 2.

A look at fig. 3.3 shows that the two signals have to be sampled simultaneously requiring two ADCs. Since one ADC card is available on the HPM the following method was adopted to enable sampling of the two signals by one ADC and yet make it look like they were sampled simultaneously. The principle is illustrated in fig.3.4 and 3.5.

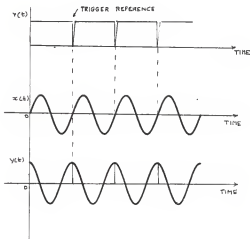


Fig.3.4. Generation of a trigger reference.

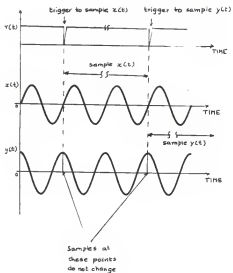


Fig.3.5. Use of trigger reference to sample $x(t)$ and $y(t)$.

The property of the two signals that is exploited in this method is that $x(t)$ and $y(t)$ continuous time signals and are periodic. Hence the signals by themselves do not change with time and with respect to themselves, a trigger reference may be used to initiate the sampling of the two signals.

In order to obtain the trigger reference a zero crossing detector was designed and built.(fig.3.6).

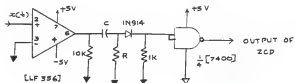


Fig.3.6. Circuit diagram of the zero crossing detector.

The output of the zero crossing detector (ZCD) was used to initiate the sampling of $x(t)$ and $y(t)$. In the assembly routine just before the sampling is actually initiated, a loop was interposed as shown below.

ADC assembly routine

.....

.....

BACK LDA CRB2

BPL BACK

LDA DDRB2

.....

.....

.....

Sampling initiated

We shall now discuss the operation of the above loop. A glance at the schematic of the interface (Appendix 2) shows that the CB2 terminal (control register B, [pin 18]) of PIA 2 in the interface can be used as an input terminal to which the negative pulses of the ZCD are given. The PIA 2 is initialized such that in the control register CRB2 bit 7 is set by negative transitions (↓) on the input terminal. Thus the output of the ZCD (the transitions) triggers the sampling as the execution of the loop is terminated when bit 7 is set by a negative transition. The MPU read of data register B clears bit 7 and the execution of the sampling process is done.

However, this method was later found to have certain

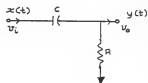
disadvantages since it introduced some errors. The MPU clock cycle time was found to be 1.1 us since it operates at 920.8 KHz. There is an ambiguity involved since the trigger pulse could arrive at any instant with respect to the two instructions forming the loop. An analysis will reveal that there could be a maximum error of 4 clock cycles (4.4 us) which at a signal frequency of around 1000 Hz. gives rise to a phase error of around 2.5 degrees.

Hence in order to get around this problem an interrupt service routine was used in the assembly program and the properties of the Wait For Interrupt (WAI) were utilised. When this instruction is executed then the uP waits indefinitely until an interrupt occurs and then proceeds to the interrupt service routine without any delay after which the sampling is initiated. Thus the source of ambiguity and error were removed totally. The assembly listing is given in Appendix 3.

The signals were derived using a function generator. In order to know the frequency of the signals a digital frequency counter was used. The ADC was switched to operate in the ± 10 V range keeping in view the effects of noise. It can be seen that if a sinewave has to be digitized properly then the amplitude of the signal must be adequate to cover the entire operating range of the ADC. This is discussed in a later chapter.

The signals $x(t)$ and $y(t)$ were derived using a simple RC circuit (fig.3.8). The phase shift of this circuit is given by $\arctan(1/2\pi f RC)$. This circuit also produces attenuation besides

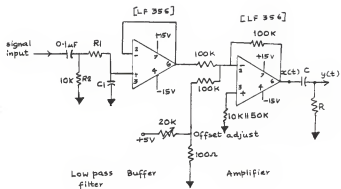
the phase shift. Hence when $y(t)$ is sampled the input is increased to a sufficient value so that the amplitude of $y(t)$ covers the operating range of the ADC.



$y(t)$ leads $x(t)$ by
 $\arctan [1/2\pi fRC]$

Fig.3.8. Use of RC circuit to generate $x(t)$ and $y(t)$,

Other considerations are that the HP3311 (function generator) produces a maximum sine output of ± 5 V and hence an amplifier using an OPAMP was used. Also since the output of the HP3311 was noisy a low pass filter (LPF) was used at the input stage. Offset adjustment was provided for the amplifier, since the method requires that $x(t)$ and $y(t)$ be zero mean signals. The circuit diagram is shown in fig.3.9.



R_1 and C_1 chosen for desired cutoff; $f_1 = 1/(2\pi R_1 C_1)$

Fig.3.9. Noise reduction and signal conditioning circuit.

Accomplishing all of the above enabled a working system to be developed which uses the adaptive algorithm. A block diagram of the system implementation is shown in fig. 3.10.

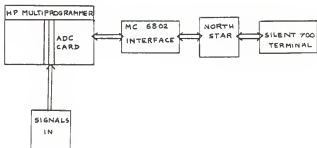


Fig.3.10. Block diagram of the test system.

3.6 WORKING OF THE METHOD AND THE TEST SYSTEM

A schematic of the test system is shown in fig. 3.11.

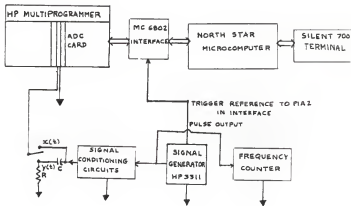


Fig.3.11. Configuration of the test system.

The Silent 700 terminal acts as the communicating device between the operator and the system. The entire operation is controlled by the software in BASIC. A command is sent via the serial port of the NS to the MC 6802 Interface . This consists of the character "A". The main assembly routine in the interface recognises this as a command to invoke the ADC routine and performs the sampling using this routine. The samples are then sent back via the serial port to the NS. The signals $y(t)$ and $x(t)$ are sampled successively. Before the processing starts the frequency of the signals indicated by the frequency is input to the NS by the use of the INPUT statement. The processing is then done and the results are displayed on the Silent terminal. The CRO was used to monitor the amplitudes of the signals to suit the range of the ADC.

A small modification that was carried out at a later stage was that although the ZCD was designed and used, the performance of the comparator was not up to the mark (probably due to noise) and hence a more accurate trigger reference was derived from the pulse output terminals of the HP3311. (this being generated internally in the HP3311).

After considerable experimentation, it was found that the ratio f_s/f_0 was very significant in determining convergence. The sampling frequency being fixed at 9130 Hz., the input signal frequency was varied to study the effects of the ratio over convergence and the calculated value of phase. This was done by comparing the phase obtained with the actual phase obtained by the formula $\arctan(1/2\pi f_0 RC)$ the values of R and C being

determined to some degree of accuracy by a digital resistance/capacitance meter. (Digibridge).

As discussed earlier in the section 3.2, the two parameters N (the number of weights), and the convergence parameter played a significant role on the convergence (K) and the calculated phase (P). The table below summarises the values of N, α , and K (the number of iterations needed for convergence) that enabled proper working of the method. These were incorporated in the software to cover the required frequency range.

TABLE SHOWING THE VALUES OF THE NUMBER OF FILTER WEIGHTS
CONVERGENCE PARAMETER AND ITERATIONS FOR THE COVERED
FREQUENCY RANGE

FREQUENCY	WEIGHTS	CONVERGENCE PARAMETER	ITERATIONS
F	N	V1	K
200	8	0.006	200
300	8	0.006	200
400	8	0.006	200
500	8	0.006	200
600	8	0.006	200
700	4	0.004	300
800	4	0.004	300
900	4	0.003	400
1000	4	0.003	400
2000	3	0.003	400

Thus the proper values of N, α , and K enable the filter to converge smoothly to the actual phase. The range covered in the table is from 2000 Hz. to 100 Hz. and the obtained readings are shown in the Appendix 4.

Frequencies less than 100 Hz. could not be considered since considerable attenuation was produced by the RC circuit and the output amplitude was too low to fit the range of ADC. This thus provides a frequency range f_B/f_0 from 9130/2000 to 9130/100. Hence if all the frequencies outside this range are to be covered then the sampling frequency must be made variable.

Some of the highlights of system performance are presented below:

1. The sampling frequency must be at least four times the frequency of the signals.
2. The convergence parameter and number of weights determine the number of iterations for convergence, as a function of the ratio f_B/f_0 .
3. The working of the method depends to a great extent on the performance of the ADC.
4. The amplitude of the input to the ADC must be properly scaled to suit the operating range of the ADC for proper digitizing.
5. The trigger reference must be as accurate as possible.
6. The data acquisition time for each signal is around 30 sec. and the processing time around 100 sec. Hence in order to decrease the time for one phase measurement the two times must be minimized.
7. The sampling frequency must be made variable to accommodate the

entire frequency range.

8. A 12 bit ADC has been used; a lower resolution would increase the number of iterations for convergence.

9. There is always some residual error due to the fact that the method does not converge to the Wiener solution.

The performance of the adaptive method was quite satisfactory. The main sources of error may be due to the characteristics of the input signal (noise, phase drift etc.) the requirement of an accurate trigger reference not being adequately met, shifts in frequency of function generator with changes in amplitude, accuracy of sampling frequency and finally the performance and repeatability of the ADC.

It may be concluded that this method is capable of being implemented and also capable of reasonable accuracy assuming all the sources of error are rooted out or minimized. A suitable system for implementation of this method is discussed in a later chapter.

CHAPTER 4

THE FFT AND CORRELATION ALGORITHMS

4.1 INTRODUCTION

The Fast Fourier Transform (FFT) and Correlation algorithms represent other suitable methods for the determination of phase. The FFT forms a very powerful tool in the frequency domain analysis of a sampled signal. It is possible to investigate certain characteristics of the original signal by studying the amplitude and phase spectra which the FFT makes possible. The Correlation technique is more or less used along with the FFT and is also discussed. It is shown that the FFT provides valuable information about the sampled signal which can be utilised to describe system performance and other characteristics of the signal.

4.2 THEORY OF THE FFT

Consider the Continuous time signal $x(t)$ sampled at the Nyquist rate to generate the sequence $x(m)$. The discrete Fourier transform (DFT) is a Fourier representation of the given sequence $x(m)$ $0 \leq m \leq N-1$ and is defined as $X(n) = \sum_{m=0}^{N-1} x(m)W^{nm}$, $0 \leq n \leq N-1$, where N is finite and $W = e^{-j2\pi/N}$ and $X(n)$ is the n th DFT coefficient. It is seen that the DFT coefficients are representative of the coefficients obtained by expanding the given signal in a Fourier series. This is shown in fig.4.1.

Thus the DFT coefficients $X(n)$ represent the sampled signal in the frequency domain as follows.

$$X(n) = \sum_{m=0}^{N-1} x(m)W^{nm} \quad 0 \leq n \leq N-1$$
$$W = e^{-j2\pi/N}$$

$$X(nw_n) = X(n)$$

which implies that the DFT coefficients yield the Fourier transform of a given sequence at a set of points that are $w_n = 2\pi/NT$ radians/sec apart, or $f_n = 1/NT$ Hz. apart.

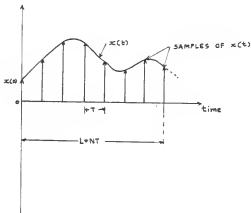


Fig.4.1. An N point DFT of a periodic DT signal.

Thus considering the DFT coefficients $X(n)$ we can see that "n" is the frequency index and that $X(n)$ is in general a complex quantity having a real and imaginary part. Thus we define the amplitude spectrum as $P_n = |X(n)|$

and the phase spectrum as $\phi_n = \arctan[\text{Im}(X(n))/\text{Re}(X(n))]$

Thus we see that in the frequency domain, $X(n)$ shows the frequency content of the original signal $x(t)$, except that this information is not available as a continuous function of frequency but only at discrete points given by $f_n = 1/NT$ Hz.

The FFT algorithm is merely an efficient and fast method developed to compute the DFT. Thus a N point FFT of a sequence $x(m)$ is obtained by running an FFT of N data points obtained by sampling $x(t)$. If the sampling frequency is made numerically equal to the value of N then we have $f_n = 1/NT = f_s/N$ since $T=1/f_s$ ie. $f_n = 1$ Hz.

Thus for $n=0$ we have $|X(0)|$ representing the DC component

for $n=1$ we have $|X(1)|$ representing the amplitude of the 1 Hz component and so on.

Similarly for the phase spectrum the phase of the different frequency components of the signal is obtained at $n=0,1,2,..$ Hz.

Now let us consider two signals $x(t)$ and $y(t)$ where $y(t)$ is the leading signal. If $x(t)$ and $y(t)$ are sine waves of some frequency f_0 then by running a FFT of the samples of $x(t)$ and $y(t)$ we have access to their phase spectra. We determine the phase of the frequency component f_0 for $x(t)$ and that for $y(t)$ and the difference yields the phase difference between $x(t)$ and

$y(t)$.

4.3 SIMULATION AND VERIFICATION

The simulation was carried out on the HP 9845B Desktop computer. Two sine waves of frequency equal to 24 Hz. were generated as follows and sampled at 128 Hz. ie. $f_s=128$ Hz. 128 samples ($N=128$) were taken for running the FFT.

$$x(k) = 0.5 \sin(\omega_0 kT)$$

$$y(k) = 2 \sin(\omega_0 kT + P)$$

$P=50$ degrees and its equivalent in radians in the expression for $y(k)$.

A 128 point FFT was run on the two sampled signals using the AC waveform analysis package available on the HP 9845B. The amplitude and phase spectra were obtained on the plots shown. Also a printout of the FFT coefficients was obtained. The plots and the printout are shown figs.4.2 through 4.6.

Let us consider the different spectra in turn. A look at the amplitude spectrum of $x(k)$ shows that there is a single frequency component at 24 Hz. with a magnitude of 0.5 which is justified since $x(t) = \sin(2\pi * 24 * t) * 0.5$

A look at the amplitude spectrum of $y(k)$ shows a peak at 24 Hz. again with an amplitude of 2 which is justified since $y(t) = 2\sin(2\pi * 24 * t + P)$.

By looking at the printout in fig.4.6 we see that the phase of the 24 Hz. component for $x(k)$ is -22.5 degrees and that for $y(k)$ is 27.5 degrees . Hence the phase difference is given by $27.5 - (-22.5) = 50$ degrees which is exactly the phase difference

between $x(t)$ and $y(t)$.

Thus the algorithm was verified by using the FFT package on the HP 9845B.

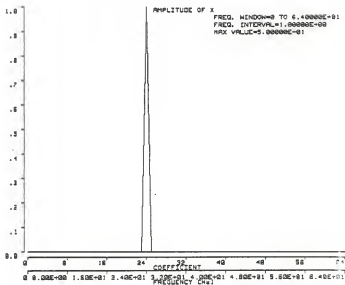


Fig.4.2. Amplitude spectrum of X.

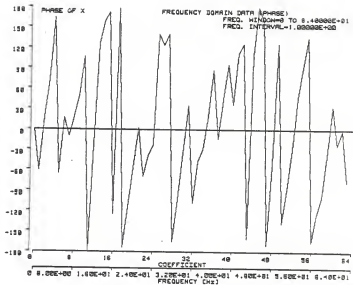


Fig.4.3. Phase spectrum of X.

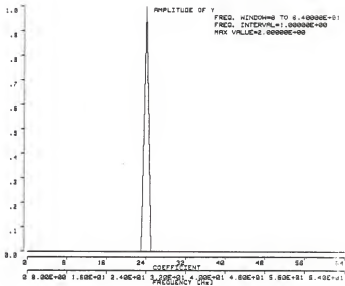


Fig.4.4. Amplitude spectrum of Y.

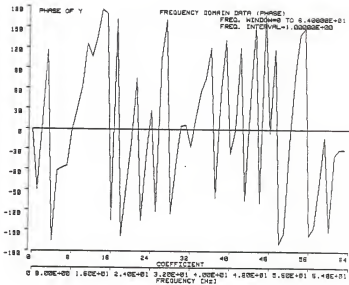


Fig.4.5. Phase spectrum of Y.

FREQUENCY DOMAIN DATA

FREQUENCY WINDOW=0 TO 6.48800E+01[Hz]
 FREQUENCY INTERVAL=1.00000E+00[Hz]

COEFF. DC TERM MAX FREQ.	FREQUENCY[Hz]	REAL	IMAG	MAGNITUDE	PHASE[DEG]
1	1.00000E+00	-1.04250E-11			
2	6.48800E+01	6.89375E-12			
3	1.00000E+00	1.03122E-11	-1.85450E-11	2.12193E-11	-60.92
4	2.00000E+00	1.41345E-11	4.47469E-12	1.48250E-11	17.27
5	3.00000E+00	3.49095E-12	1.89691E-11	1.15130E-11	72.31
6	4.00000E+00	-1.09111E-11	3.32410E-12	1.13774E-11	163.54
7	5.00000E+00	7.58508E-12	-1.62487E-11	1.79319E-11	-64.98
8	6.00000E+00	1.00810E-11	2.79179E-12	1.02154E-11	16.33
9	7.00000E+00	9.72328E-12	-1.81163E-12	9.89052E-12	-13.55
10	8.00000E+00	1.88418E-11	6.43141E-12	1.99805E-11	18.85
11	9.00000E+00	1.81196E-11	2.07836E-11	2.75731E-11	48.92
12	1.00000E+01	-1.21749E-11	4.06802E-11	4.23941E-11	106.69
13	1.10000E+01	-3.30581E-11	-1.96895E-13	3.30587E-11	-179.66
14	1.20000E+01	1.01196E-11	-4.26594E-12	1.09817E-11	-22.85
15	1.30000E+01	-9.59942E-12	1.28573E-11	1.66452E-11	126.75
16	1.40000E+01	-1.28977E-11	7.21818E-12	2.03037E-11	159.18
17	1.50000E+01	-2.13964E-11	3.01387E-12	2.16676E-11	171.98
18	1.60000E+01	-9.97835E-12	-1.43867E-11	1.75004E-11	-124.74
19	1.70000E+01	-3.95310E-12	2.12129E-13	3.95079E-12	176.93
20	1.80000E+01	-2.69206E-11	-2.62418E-12	2.70482E-11	-174.43
21	1.90000E+01	-1.66766E-11	-3.78805E-11	4.13944E-11	-113.76
22	2.00000E+01	2.25904E-11	-3.42549E-11	4.16376E-11	-56.59
23	2.10000E+01	2.36825E-11	1.81549E-12	3.97874E-11	2.81
24	2.20000E+01	3.46701E-12	-9.18176E-12	9.79344E-12	-69.64
25	2.30000E+01	2.96175E-11	-2.41783E-11	3.82333E-11	-39.23
26	2.40000E+01	4.61948E-01	-1.91342E-01	5.08908E-01	-22.56
27	2.50000E+01	-1.77575E-11	1.45137E-11	2.29323E-11	146.75
28	2.60000E+01	-6.15491E-12	9.72955E-12	1.08535E-11	124.55
29	2.70000E+01	-2.35421E-11	1.82666E-11	2.88599E-11	146.73
30	2.80000E+01	-2.96842E-11	-1.36412E-11	3.18841E-11	-184.89
31	2.90000E+01	-8.60485E-12	-3.87762E-11	3.97199E-11	-182.51
32	3.00000E+01	2.52641E-11	-1.73751E-11	3.26128E-11	-34.58
33	3.10000E+01	9.67410E-12	6.26559E-12	1.02271E-11	34.62
34	3.20000E+01	-4.21875E-12	-1.26563E-11	1.33489E-11	-108.43
35	3.30000E+01	1.07885E-11	-1.16101E-11	1.58434E-11	-47.12
36	3.40000E+01	1.96111E-11	-1.05227E-11	2.17290E-11	-28.96
37	3.50000E+01	2.63521E-11	9.26184E-12	2.79312E-11	19.37
38	3.60000E+01	9.84564E-13	1.73759E-11	1.74838E-11	86.74
39	3.70000E+01	8.84819E-12	-2.69727E-12	9.89335E-12	-13.33
40	3.80000E+01	1.61156E-11	1.66721E-11	2.27601E-11	44.92
41	3.90000E+01	-2.11871E-12	2.23652E-11	2.24654E-11	95.41
42	4.00000E+01	5.38888E-11	3.75088E-11	6.25088E-11	36.87
43	4.10000E+01	-1.72640E-12	4.13274E-12	4.97084E-12	112.67
44	4.20000E+01	-9.29969E-12	1.26325E-11	1.27825E-11	126.32
45	4.30000E+01	-1.46773E-11	-5.17347E-12	1.55624E-11	-160.58
46	4.40000E+01	-1.22805E-12	3.32848E-12	3.52719E-12	110.20
47	4.50000E+01	-7.09515E-12	1.22428E-13	7.06622E-12	179.66
48	4.60000E+01	-3.53890E-12	1.34160E-12	3.78467E-12	159.24
49	4.70000E+01	-1.36836E-11	-2.24751E-12	1.37874E-11	-170.62
50	4.80000E+01	7.47835E-12	-9.69916E-12	1.22474E-11	-52.37
51	4.90000E+01	-6.54820E-12	9.84351E-12	1.11653E-11	125.91
52	5.00000E+01	-9.99145E-12	-8.83381E-12	1.32539E-11	-138.26
53	5.10000E+01	1.78526E-12	-9.34736E-12	9.34892E-12	-88.91
54	5.20000E+01	1.07659E-11	-6.07366E-12	1.23618E-11	-29.43
55	5.30000E+01	1.11369E-11	1.16109E-11	1.60881E-11	46.20
56	5.40000E+01	-4.07135E-13	1.28334E-11	1.28398E-11	91.82
57	5.50000E+01	-1.81793E-11	1.80911E-11	2.56471E-11	135.14
58	5.60000E+01	-5.11848E-11	-1.29426E-11	5.27968E-11	-165.81
59	5.70000E+01	-1.45457E-11	-3.87893E-11	2.53732E-11	-125.88
60	5.80000E+01	-5.91462E-12	-3.33629E-11	3.38649E-11	-99.89
61	5.90000E+01	2.48785E-11	-1.98252E-11	3.18052E-11	-38.56
62	6.00000E+01	8.66474E-12	5.58911E-12	1.02606E-11	33.81
63	6.10000E+01	1.19597E-11	-5.27146E-12	1.35699E-11	-23.79
64	6.20000E+01	1.12365E-11	-3.16946E-13	1.13649E-11	-1.62
65	6.30000E+01	1.55675E-12	-7.43644E-12	7.26659E-12	-77.52

Fig.4.6. Printout of FFT data for X.

FREQUENCY DOMAIN DATA

FREQUENCY WINDOW=0 TO 6.0000E+01(Hz)
 FREQUENCY INTERVAL=1.0000E+00(Hz)

COEFF DC TERM MAX FREQ	FREQUENCY(Hz)	REAL	IMAG	MAGNITUDE	PHASE(DEG)
1	1.0000E+00	3.6804E-13	-2.47670E-11	2.47704E-11	-89.15
2	2.0000E+00	2.2486E-11	4.97397E-12	2.32276E-11	12.47
3	3.0000E+00	-1.58707E-11	3.67509E-11	3.42453E-11	116.11
4	4.0000E+00	-4.57583E-11	-1.18837E-11	4.72763E-11	-165.44
5	5.0000E+00	2.94318E-11	-5.35896E-11	6.11319E-11	-61.32
6	6.0000E+00	7.92928E-12	-1.21928E-11	1.44691E-11	-56.77
7	7.0000E+00	4.98659E-11	-5.63646E-11	6.91538E-11	-54.59
8	8.0000E+00	1.84142E-10	3.59254E-13	1.84142E-10	.20
9	9.0000E+00	7.49475E-11	4.47302E-11	8.72897E-11	38.03
10	1.0000E+01	6.25474E-11	1.20794E-10	1.35947E-10	62.61
11	1.1000E+01	-6.44133E-11	9.26721E-11	1.12525E-10	125.80
12	1.2000E+01	-6.87474E-12	3.27892E-11	2.38635E-11	164.79
13	1.3000E+01	-7.84951E-11	7.32328E-11	1.07532E-10	134.99
14	1.4000E+01	-8.49922E-11	3.97237E-12	8.52647E-11	175.93
15	1.5000E+01	-4.53158E-11	8.25448E-12	4.68686E-11	169.68
16	1.6000E+01	-3.44826E-11	-3.56039E-11	4.95450E-11	-134.68
17	1.7000E+01	-2.99088E-11	9.66780E-12	3.14112E-11	162.20
18	1.8000E+01	-1.24332E-10	-4.95360E-11	1.33837E-10	-158.20
19	1.9000E+01	8.22449E-12	-1.56238E-10	1.56454E-10	-86.99
20	2.0000E+01	1.18939E-10	-4.29784E-11	1.18973E-10	-21.18
21	2.1000E+01	1.49745E-11	5.74852E-11	5.96173E-11	75.38
22	2.2000E+01	-3.94323E-11	-4.87292E-11	5.66925E-11	-134.08
23	2.3000E+01	9.42521E-11	-8.66410E-11	1.28024E-10	-42.59
24	2.4000E+01	1.77492E+00	9.23497E-01	2.00016E+00	27.52
25	2.5000E+01	-2.22918E-11	-3.72859E-11	4.33729E-11	-120.93
26	2.6000E+01	-1.62648E-12	7.81844E-12	7.29443E-12	193.05
27	2.7000E+01	-8.92485E-11	2.97624E-11	9.48923E-11	161.56
28	2.8000E+01	-6.21247E-11	-1.24549E-10	1.50893E-10	-124.35
29	2.9000E+01	9.52150E-11	-1.38268E-10	1.61349E-10	-53.83
30	3.0000E+01	1.61482E-10	9.83276E-12	1.61957E-10	5.53
31	3.1000E+01	4.68154E-11	4.54929E-12	4.62397E-11	5.62
32	3.2000E+01	3.67188E-11	-1.84375E-11	4.18878E-11	-26.66
33	3.3000E+01	7.54866E-11	2.51156E-11	7.95492E-11	19.49
34	3.4000E+01	3.62475E-11	5.48154E-11	6.51623E-11	56.27
35	3.5000E+01	1.24373E-11	4.89946E-11	5.82522E-11	75.79
36	3.6000E+01	-3.16289E-11	5.19379E-11	6.85007E-11	126.86
37	3.7000E+01	-2.94997E-12	-1.68416E-11	1.10337E-11	-104.71
38	3.8000E+01	3.65730E-11	4.25438E-11	5.61831E-11	49.32
39	3.9000E+01	-4.68920E-11	5.23645E-11	7.02915E-11	131.84
40	4.0000E+01	2.18938E-10	-1.53122E-10	2.66657E-10	-35.98
41	4.1000E+01	3.52658E-11	-6.84525E-13	3.52718E-11	-.97
42	4.2000E+01	-2.76599E-11	4.72838E-11	5.47951E-11	128.32
43	4.3000E+01	-1.57932E-11	-5.87581E-11	6.48436E-11	-105.84
44	4.4000E+01	4.41818E-11	3.67949E-11	5.74969E-11	39.79
45	4.5000E+01	-4.82676E-11	2.17144E-11	5.22808E-11	125.91
46	4.6000E+01	-4.66132E-12	-1.42780E-11	1.50196E-11	-108.08
47	4.7000E+01	-1.65895E-11	3.18693E-12	1.68948E-11	169.12
48	4.8000E+01	3.46781E-11	-3.72891E-12	3.48692E-11	-5.81
49	4.9000E+01	-3.36364E-11	4.39259E-11	7.22617E-11	117.74
50	5.0000E+01	-4.44793E-11	-8.58738E-12	4.52256E-11	-169.17
51	5.1000E+01	-1.71285E-11	-8.58689E-12	1.91664E-11	-123.37
52	5.2000E+01	3.57403E-11	3.67949E-11	2.74211E-11	-29.16
53	5.3000E+01	8.41360E-12	4.17908E-11	4.36896E-11	78.61
54	5.4000E+01	-3.49198E-11	2.88754E-11	4.51214E-11	148.71
55	5.5000E+01	-5.68147E-11	3.41639E-11	6.63136E-11	148.99
56	5.6000E+01	-1.24454E-10	-5.27637E-11	1.35178E-10	-157.02
57	5.7000E+01	-1.30833E-10	-9.94259E-11	1.59661E-10	-145.35
58	5.8000E+01	1.84622E-11	-1.89539E-10	1.89432E-10	-84.43
59	5.9000E+01	1.12401E-10	-2.53973E-11	1.12524E-10	-12.73
60	6.0000E+01	-2.87452E-11	-1.24119E-11	2.92465E-11	-153.45
61	6.1000E+01	4.33981E-11	-3.48841E-11	5.86232E-11	-39.73
62	6.2000E+01	5.62394E-11	-3.24089E-11	6.49919E-11	-29.00
63	6.3000E+01	6.38797E-11	-3.74612E-11	7.48224E-11	-30.35

Fig.4.6.a. Printout of FFT data for Y.

4.4 PRACTICAL ASPECTS AND WORKING OF THE METHOD

In order to demonstrate the working of this method in a practical situation, use was made of the system consisting of the North Star microcomputer, the MC 6802 interface and the HP Multiprogrammer discussed in chapter 3. The same system was used to obtain samples of $x(t)$ and $y(t)$ generated using the function generator and the RC circuit. These samples were then keyed into the HP 9845B computer to form the data on which the FFT was to be run using the AC waveform analysis package.

The package is a standard one available on the HP 9845B. It has two modes of data input - manual and from disc storage. The simulation experiment was carried out by storing 128 samples of data of $x(t)$ and $y(t)$ on disc and then made use of by the FFT program in the package. However, when the actual data samples were taken using the NS system, the manual mode was used and data was input to the HP 9845B sample by sample in this fashion and finally stored on disc.

Certain aspects had to be given due importance and consideration. The sampling frequency is 9130 Hz. and just 128 samples were used which means that the frequency index "n" in the FFT components represents a frequency separation of $f_n = 1/(128 * [1/9130]) = 71.33$ Hz. Thus if the input signal frequency was to figure in the plots then some integral multiple of f_n had to be chosen as the value for the signal frequency. Hence utmost care was taken to accurately adjust the signal frequency to this

value before sampling was done.

The results are shown in Appendix 5. The phase difference by the FFT method was compared with the actual phase difference obtained by the formula for the phase shift of the RC circuit at that frequency.

These plots can be compared with those obtained by simulation. It can be seen in the amplitude plots for the simulation case there is a distinct peak at the signal frequency and the amplitude is zero at all points but this is not the case in the plots obtained by using actual data samples. These plots in fact show that the sampled signal contains a small amount of other frequencies as well. These effects are considered later in this chapter. It can also be seen that the error in the case of simulation was zero but in this case there is a finite phase error (obtained as the difference between the obtained phase and the actual phase of RC circuit). This may be attributed to the following factors.

1. Finite precision of ADC (12 bits)
2. Just 128 samples were used for the FFT. Since the sampling frequency is 9130 Hz, at least as many samples would have to be used for a very good frequency resolution which would minimise phase errors.
3. Presence of noise in the signals.
4. Inaccuracies in the trigger reference and the characteristics of the function generator.

However in spite of the above drawbacks the error was only

0.99 degrees as can be seen from the plots and improvements in the above stated factors would minimise this error further.

The FFT is a very powerful tool for analysis in the frequency domain. It is capable of a high resolution in frequency provided enough samples are used. As seen from the plots one can clearly see the noise that exists, the presence of harmonics (due to harmonic distortion of signal from generator etc.) and also any dc offsets that may be present. Thus apart from finding the phase the amplitude plots provide extra information as to the performance of the ADC, noise characteristics etc.

In the earlier discussion, it was mentioned that for a resolution of 1 Hz. the sampling frequency must be equal to the number of data points or samples. Thus in a practical system, the sampling frequency must be made variable to cover the entire frequency range. For example, the sampling frequency could be 256 Hz. to cover frequencies from 1 Hz. to 128 Hz. and a 128 point FFT is required. Similarly the value of sampling frequency could be stepped up to say 4096 Hz. and 4096 points are necessary. This means a memory of 4K would be required in hardware and the maximum signal frequency that can be analysed using the FFT would be 2048 Hz. If a memory of 16K could be afforded then f_s would be approximately 16 Khz. and the maximum signal frequency would then be 8 Khz. If such huge memory requirements cannot be met and assuming a 4K memory is available then the frequency range is limited to 2048 Hz.(assuming a resolution of 1 Hz. is necessary). In order to effectively increase the range the following approach is recommended.(fig.4.7).

In this schematic, the signals are heterodyned with a oscillator to provide an intermediate frequency of ≤ 2048 Hz. and

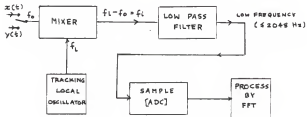


Fig.4.7. Scheme for increasing effective frequency range.

the resulting output is fed to a low pass filter. Once this frequency translation takes place the output of the LPF still contains all the phase information of either $x(t)$ or $y(t)$ and hence can be sampled and processed by the FFT algorithm. The mixer may be realised using an analog multiplier.

With the existing system the sampling frequency is fixed at 9130 Hz, and yet can be used to obtain the phase response of a RC filter from 0 Hz, to say $9130/2 = 4500$ Hz. The only restraint is that the signal frequencies must be raised in steps from 0 Hz, as integral multiples of $f_n = 1/NT$ Hz. Here $f_n = 71.33$ Hz, and the phase response of the filter can be determined.

It may be noted in passing that FFT chips are available to accomplish a very fast Fourier transform of the data samples. Also this may be achieved by an assembly routine. These may also be considered as possible methods in a practical system.

Thus the FFT method is another practical method that can be used in the determination of phase. As mentioned earlier a lot of extra information is also provided by this method and may be used to advantage.

4.5 THE CORRELATION TECHNIQUE

Efforts were also directed to obtain a suitable method for phase measurement using Correlation techniques. We first consider an attempt to use cross correlation in attaining our goal. By using the FFT along with this method a suitable method for phase

determination is described.

Theory of Correlation:

Consider two sequences $x(k)=V_x \sin(\omega kT)$ and $y(k)=V_y \sin(\omega kT+P)$.

The cross correlation sequence is given by the following:

$$C(mT) = [1/N] \sum_{k=0}^{N-1} x(k) * y(k+m) \quad 0 \leq m \leq N-1;$$

and with $m=0$ we have

$$\begin{aligned} C(0) &= (V_x V_y / N) \sum_{k=0}^{N-1} \sin(\omega_0 kT) * \sin(\omega_0 kT+P) \\ &= (V_x V_y / 2N) \sum_{k=0}^{N-1} [\cos P - \cos(2\omega_0 kT+P)] \end{aligned}$$

It can be shown that $\sum_{k=0}^{N-1} \cos(2\omega_0 kT+P)$ will go to zero if N is chosen to cover one full period or integral multiple of the period of $x(k)$ or $y(k)$.

$$\text{Hence } C(0) = (V_x V_y / 2) \cos P \quad \text{or} \quad \cos P = 2C(0) / V_x V_y$$

If V_x and V_y are known accurately along with $C(0)$ then the phase P can be determined using the above relation.

Attempts were made to utilise this relation and simulation was done on the HP 9845B. After the generation of $x(k)$ and $y(k)$ the two sequences were multiplied to obtain the cross correlation sequence C . This was then averaged over a period of $x(k)$ and $C(0)$ was computed. Some unreliable results were obtained (for eg. value of $\cos P > 1$) and the results were inconclusive and this approach was abandoned.

A slightly different approach produced encouraging results.

Consider the signals $x(t)=V_x \sin(\omega_0 t)$ and $y(t)=V_y \sin(\omega_0 t+P)$.

The cross correlation between $x(t)$ and $y(t)$ is $x(t)*y(t)$

$$\begin{aligned} C_{xy}(t) &= V_x V_y \sin(\omega_0 t) \sin(\omega_0 t+P) \\ &= [V_x V_y / 2] (\cos P - \cos(2\omega_0 t+P)) \end{aligned}$$

$$= [V_x V_y / 2] \cos P - [V_x V_y / 2] \cos(2\omega_0 t + P)$$

In the frequency domain the first term represents a dc term independent of frequency and the other term is a harmonic ($2\omega_0$) term. Now if an FFT is run on the cross correlation $C_{xy}(t)$ then the amplitude spectrum will show a dc term which is equal to that given by $[V_x V_y \cos P] / 2$. An FFT of $x(k)$ and $y(k)$ in turn would enable V_x , V_y , to be determined from the amplitude plots. Thus the dc term ie. $C(0)$, V_x , V_y , enable $\cos P$ and hence P to be determined.

4.6 SIMULATION AND VERIFICATION

Using the same example as was used for the verification of FFT by simulation the following were generated.

$$x(k) = 0.5 \sin(2\pi * 24 * k * T)$$

$$y(k) = 2 \sin(2\pi * 24 * k * T + P)$$

128 samples of $x(k)$ and $y(k)$ were used to generate the cross correlated sequence. An FFT of this sequence was run and the amplitude plot is shown in fig.4.8. From the printout (Appendix 6) the DC term is found (3.21394E-1) which gives $C(0)$. The values of V_x and V_y were obtained from the amplitude plots (0.5 and 2). Thus $\cos P = 2 * C(0) / [0.5 * 2]$ and from this P was calculated and found to be 50 degrees and this was exactly the phase difference between the two signals $x(t)$ and $y(t)$. Thus the algorithm was verified.

4.7 PRACTICAL ASPECTS

All the practical aspects discussed for the FFT method apply here. Other considerations are the presence of harmonics in

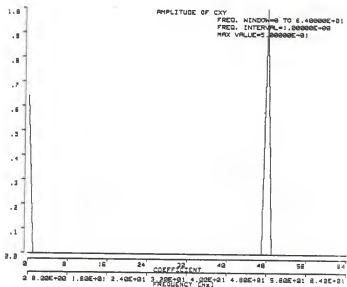


Fig.4.8. Amplitude spectrum of the correlated sequence.

the signals $x(t)$ and $y(t)$ due to the harmonic distortion present in the function generator output. The presence of harmonics will cause errors in the value of the DC term $C(0)$. Also additional frequencies will be introduced as a result of the mixing action produced by the cross correlation ($x(t)*y(t)$). In order to overcome this problem filtering has to be done. Use of analog filters is not practical and hence the use of digital filtering is recommended.

A simple low pass digital filter can be easily implemented. The digital filters have many advantages such as flexibility, reliability and ease with which response can be changed. Thus a LP digital filter can get rid of the harmonic content in $x(k)$ and $y(k)$ after which the correlation sequence may be formed and analysed using the FFT.

CHAPTER 5

NUMERICAL METHODS

5.1 INTRODUCTION

Numerical and statistical methods may also be used in the determination of phase. This method relies to a great extent on an accurate determination of zero crossings of the signals. This method outlines how the determination of zero crossings of the signals can be used to calculate the phase difference. Numerical methods like curve fitting and Newton's method for finding the root of a polynomial are described.

5.2 THEORY OF THE METHOD

The method uses the fact that the accurate determination of the zero crossings of the two signals enables the determination of phase difference. Consider the situation shown in fig. 5.1. Let us focus our attention to the zero crossing zone; we shall consider the positive to negative zero crossing for the sake of uniformity for both signals. Consider the samples being taken at times $t_1, t_2, t_i, t_{i+1}, \dots, t_n$ spaced by an amount T where T is the sampling period and $T=1/f_s$ where f_s is the sampling frequency.

Consider a straight line joining $v(t_i)$ and $v(t_{i+1})$ making intercepts x and y on the reference axis as shown. It is required to obtain the zero crossing time t_x . Here $x+y=T$, and using the property of similar triangles it can be shown that :

$$t_x = t_i + T[1/(1+y/x)] = t_i + T[1/(1+v(t_{i+1})/v(t_i))]$$

If $v(t_{i+1})$ and $v(t_i)$ are known then t_x can be evaluated. Similarly t_y the time of zero crossing for the second signal

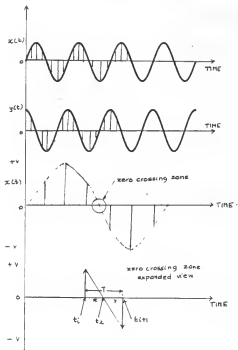


Fig.5.1. Use of zero crossings of signals to find phase.

$y(t)$ can be determined. Once t_x and t_y are known then the difference of time between zero crossings can be determined i.e. $\Delta t = t_x - t_y$. Once Δt is known the phase difference can be calculated as follows:

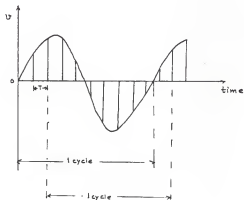


Fig.5.2. Phase determination for a periodic sine wave.

Since a sine wave is periodic the starting point is relatively unimportant as can be seen from fig.5.2. If the ratio of f_s/f_0 is say 6 then there would be 6 samples per cycle of the waveform. Hence $6T$ would represent 1 full cycle or a phase of 360° degrees. Hence by computing $t_x - t_y = \Delta t$, the phase would be given by $P = \Delta t (f_0/f_s) * 360$ in degrees. Thus the phase difference could be evaluated.

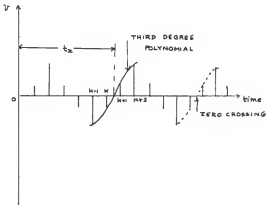
It can be seen from fig.5.1 that this method represents a straight line fit for the sine wave at the zero crossing. Thus this is an approximate fit for the sinewave at the zero crossing. Simulation was done to study the working of this method. It was found that the method produced large phase errors at some points possibly due to certain discontinuities. This may be due to the fact that the straight line fit is an approximate fit for the sine wave. Also further investigation of the expression for t_x reveals that if the ratio $v(t_{i+1})/v(t_i)$ tends to -1 then the denominator becomes zero and this causes a discontinuity. Hence this method was not considered further.

A slightly modified approach was then tried out. Since the straight line fit is approximate a higher degree polynomial is a more accurate representation of a sine wave. Hence a third degree polynomial was used to fit a curve to the sampled data at the zero crossing. The following procedure outlines the method.

1. Obtain samples of $x(k)$ and $y(k)$.
2. Find the values of k when there is a negative to positive transition (zero crossing) of $x(k)$ (this can be determined from a

sign change of the successive samples from -ve to +ve). Repeat this for $y(k)$.

3. Fit a third degree polynomial to the obtained samples at the zero crossings. This requires two data points before and two data points after the zero crossing point under consideration. Fig.5.3 illustrates this.



$p_3(x)=f(a_i); 0 \leq i \leq 3$, a_i are coefficients of the polynomial.

Fig.5.3. Fitting a polynomial at the zero crossings.

The third degree polynomial is fitted at each zero crossing using two data points to the left and to the right of the crossing. Repeat for $y(k)$.

The polynomial fit is done using the divided difference table as shown in fig.5.4.

THE DIVIDED DIFFERENCE TABLE

k	x(k)			
1	x(1)			
		$x'(1)=x(2)-x(1)$		
2	x(2)		$x''(1)=x'(2)-x'(1)/2$	
		$x'(2)=x(3)-x(2)$		$x''(1)=$
3	x(3)		$x''(2)=x'(3)-x'(2)/2$	$x''(2)-x''(1)$
		$x'(3)=x(4)-x(3)$		3
4	x(4)			

$$p_3(x)=x(1)+x'(1)(x-1)+x''(1)(x-1)(x-2)+x'''(1)(x-1)(x-2)(x-3)$$

which reduces to $a_0+a_1x+a_2x^2+a_3x^3$ where the coefficients are given by $a_i; 0 \leq i \leq 3$.

Fig.5.4. The divided difference table.

The coefficients of the polynomial are determined from the table and the polynomial is given by the expression shown. The same procedure is adopted for the signal $y(k)$ to obtain the polynomial at the zero crossings.

Once this is obtained then the Newton's method is used to obtain the root of this polynomial, in other words the zero crossing time. An iterative scheme is used to determine the root using the Newton's Method.

Thus the zero crossing of $x(k)$ and $y(k)$ are obtained and the value of Δt is used to calculate the phase.

5.3 SIMULATION

Simulation was done for this method and it was found to work reasonably well. The software used is shown in Appendix 7a and the results in Appendix 7b.

Certain very important results were deduced from this. The ratio f_g/f_o had to be fixed at 6 for the best results (least phase error). Any decrease or increase in this ratio increases the phase error. Also when random noise is injected into the samples the error increases as shown in the results. A phase of 360 degrees could be distinguished by suitable modifications in the software. This method is very sensitive to noise and the ratio of sampling frequency to the frequency of the signals is to be fixed at 6, since the polynomial represents the best fit when this condition is satisfied. However an averaging effect is produced by determining the phase at several crossings and then

obtaining the statistical average.

5.4 WORKING OF THE METHOD

When actual data samples were used (obtained using the system discussed in chapter 3) and processed using this method (the ratio f_s/f_0 was fixed at 6) the error obtained was around 4 degrees. However by running a predictor of the signals (discussed in chapter 6) which acts as a signal enhancer the error reduced to around 2 degrees.

As a conclusion this method has a constraint on the frequency ratio being fixed at 6 and also the method is sensitive to noise and needs a more accurate representation of the signals. Hence the method may not be very practical. It may be possible to use higher degree polynomials but this might require an increase in the frequency ratio (>6) thus requiring a higher sampling frequency and also limits the useful frequency range of operation of this method.

CHAPTER 6

USE OF PREDICTOR AND NOISE CONSIDERATIONS

6.1 INTRODUCTION

In this chapter the effects of noise on the signals that are used for sampling are studied. A suitable method for combating noise by the use of predictor is presented and also other effects like finite resolution of ADC, finite word length of computer, truncation etc. are also considered and studied.

6.2 USE OF A PREDICTOR

Sufficient care must be taken to ensure that there is no noise in the original signals. This can be done by the use of analog filters like low pass filters etc. In spite of this there still tends to be some noise which also gets "digitized" along with the signal. This noise is assumed have a zero mean and to be random. It is now shown that the adaptive predictor can be used as a signal enhancer and gets rid of the noise.

The schematic of the predictor is shown in fig.6.1. This scheme is similiar to the adaptive scheme in the cancellation mode (mentioned in chapter 3) except that the reference input to the filter W is a delayed version of the input $x(k)$. In other words $x(k-1)$ denotes that the input to the filter at time k is $x(k-1)$ ie. the past value. In vector form for a M weight filter ;

$$X_k^T = [x(k-1) \ x(k-2) \ \dots \ x(k-M)]$$

$$g(k) = W^T X_k$$

$$e(k) = x(k) + n(k) - g(k)$$

Minimizing the mean square error $e(k)$ causes $g(k)$ to look

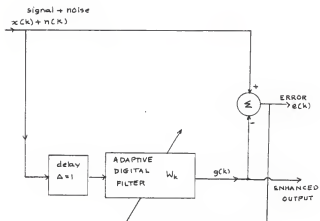


Fig.6.1. Schematic of the adaptive predictor.

like $x(k)$ or in other words the error $e(k)$ tends to look like the noise $n(k)$. Thus the filter W removes as much of the correlated energy from $x(k)$ and the enhanced output is available as $g(k)$. Since the filter takes a few iterations to adapt itself the output $g(k)$ must be taken after a few iterations for subsequent processing.

The predictor can also be implemented as an iterative scheme. Thus the predictor accomplishes enhancement of the signal corrupted by noise. An adaptive predictor with a 10 weight filter was used to clean up signals and the software is shown in Appendix 8. The predictor helped to enhance the signals used especially in the Numerical method and was instrumental in reducing the phase error.

6.3 OTHER CONSIDERATIONS

Certain considerations must be given due importance when dealing with implementation of DT (discrete time) digital systems. The following are the points to be noted.

1. Input quantization error is introduced due to a finite number of bits used to encode or digitize the signal $x(t)$ to form $x(k)$.
2. Coefficient inaccuracy error that results from using a finite number of bits to represent each coefficient w_k .
3. Round off errors that occur each time a product $w_k * x(k-1)$ is rounded to match a prescribed word length. (Truncation errors are worse than roundoff errors.)
4. Certain types of undesired oscillations that may occur due to limited accuracy and adder overflows.

These are the general considerations and we must take into account only the pertinent ones. Basically there are two types of arithmetic:- fixed point and floating point. It can be shown that if a satisfactory representation is required, the number of bits M required for an accuracy of D decimal places is given by $M > 3.3D$.

Input quantization error: This error occurs when an ADC with a finite resolution is used to sample the signal $x(t)$. If the encoded output is of B bits and the ADC has a step resolution of $q = V_{ref}/(2^B)$ then there is an ambiguity of $q/2$ referred to as the quantization error. Hence we represent the quantized value $x^Q(n) = x(n) + e(n)$ where $e(n)$ represents the error and is a random quantity which is uniformly distributed over $(-q/2, q/2)$. The overall effect of $e(n)$ is assessed in terms of its variance or power which is denoted by σ_e^2 . It can be shown that the value of this variance is equal to $q^2/12$ or $k \cdot 2^{-2B}/12$. Hence as B tends to ∞ the value of the variance or power of this noise tends to zero. When $x^Q(n)$ forms an input to a DT filter then the output noise is modified depending on the transfer function $H(z)$.

It can be shown that the effects of product roundoff errors, coefficient inaccuracies and adder overflows all contribute some amount of noise power to the overall signal content. These are given due consideration especially in special purpose hardware implementation of DT systems. However in a software implemented system only the pertinent errors need be considered.

CHAPTER 7

DESIGN CONSIDERATIONS

After having discussed the different algorithms it is essential to consider the design aspects of a suitable microprocessor based system which will accomplish data acquisition in the most efficient manner and also process the data with speed and accuracy. A discussion of the existing system is followed by the design considerations for a more efficient system.

In the existing system the configuration being indicated in fig.7.1 it can be seen that the microprocessor MC 6802 essentially acts as a communication link between the data acquisition device HP Multiprogrammer (HPM) and the data processing device North Star (NS). This important role of the microprocessor is elaborated later in this chapter. An estimation of the data acquisition time and the data processing time will enable us to make further improvements in the existing system. It can be seen that the data acquisition is initiated by a command from the NS via the serial port to the ACIA in the MC6802 interface. The main assembly routine recognises this command ("A") and branches to the ADC subroutine. The procedure is as follows. There are two PIAs (Peripheral Interface Adapter) in the interface each with two 8 bit ports available.(refer to the schematic in Appendix 9). The PIA1 is used to handle all the addressing to the HPM (16 bits) and the PIA2 is used to handle the data from the HPM (12 bits) by utilising the two ports of 8 bits each. When the data is from the HPM (12 bits) the remaining

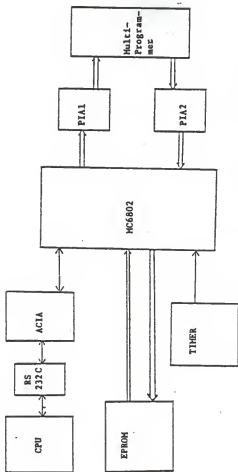


FIG 7.1

INTERFACE USING THE MC6802 MICROPROCESSOR

4 bits are masked off.

The ADC subroutine causes a control word and the slot address to be input to the HPM using PIA1 (address 2000) setting up the ADC card in the HPM for input. A gate signal is then given to the ADC to initiate conversion and at the end of the conversion signal indicated by the flag, the data in PIA2 data registers is stored in a convenient memory location. This process is continued until all the samples (512) are taken and stored in memory. These samples are then sent to the NS via the serial port using the ACIA in the interface. It was experimentally found that the time taken for the whole process is around 30 secs. Hence for sampling both signals the time taken is 60 secs. It has been found that the data processing time is around 100 secs. and hence the total time for a phase measurement is around 160 secs. A method to decrease the data acquisition time is now discussed.

It can be seen that the ADC samples at 9130 Hz. and hence the time for taking 512 samples is given by $512/9130 = 0.5$ sec. Hence for sampling the two signals it is around 1 sec. Thus it is apparent that the use of serial transmission is the main disadvantage since data is sent in series. Hence if the data acquisition time is to be minimised the parallel transfer of data is to be done. This may be accomplished by using the parallel port of the NS as shown in fig.7.2.

The data processing time however cannot be reduced unless a faster processing machine is used. Attention is now focussed on the data acquisition part and the errors generated and the improvements suggested for the same. It is shown that if an ADC



Fig.7.2. Use of the parallel port of the North Star.

has an operating range of $\pm 1V$ then the signal amplitude must also be of this range to enable proper digitizing of the signal. This is shown in fig.7.3.

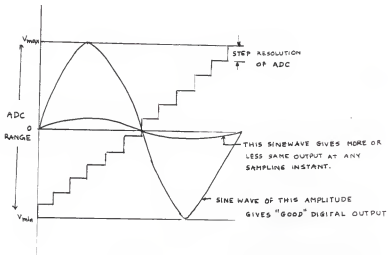


Fig.7.3. Effect of signal amplitude on "digitizing".

Hence in order to enable the "dynamic" adjusting of the signal amplitude for sampling by the ADC of a given operating range the following is suggested. (fig.7.4)

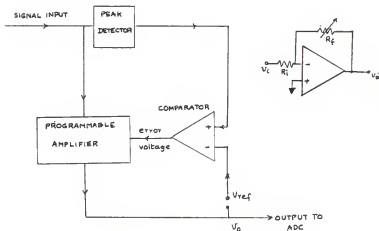


Fig.7.4. Dynamic adjusting of signal amplitude to suit ADC range.

The operation is as follows. The peak detector obtains a measure of the amplitude of the incoming signal. This is then compared with a standard reference V_{ref} and the resulting error voltage is used to control the gain of a programmable amplifier so that the output of the amplifier is now suitable for the operating range of the ADC. If instead of using a separate reference voltage the V_{ref} point is connected to V_o and derived from the output then we have a feedback control system. The programmable amplifier may be realized by using a digital attenuator or variable resistance as one of the resistances in a closed loop OPAMP circuit as shown. Thus irrespective of the signal amplitudes the output V_o is always scaled to suit the ADC range.

Errors due to delays in triggering are now considered. Since the output of the zero crossing detector or some suitable trigger reference is used it is of primary importance that these be very stable and accurate with respect to time. Fig. 7.5 illustrates the errors involved.

Let t_1 be the time after the zero crossing of the signal $x(k)$ when the trigger pulse initiates the sampling process for $x(k)$ and that for $y(k)$ be t_2 . If t_2 is different from t_1 then this leads to phase error given by $p=(t_2-t_1)*2\pi*f$ where f represents the frequency of the signals. For a difference of 4us between t_1 and t_2 and a signal frequency of around 1 KHz. the phase error can be as high as 2.5 degrees. This difference may arise due to the comparator characteristics in the ZCD or more possibly due to the noise at the zero crossings. Hence in order

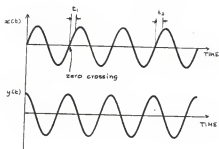


Fig.7.5. Phase errors due to inaccuracies in trigger reference.

to overcome this problem synchronous sampling is suggested using two ADCs unless of course a very stable and accurate trigger reference is provided.(fig. 7.6)

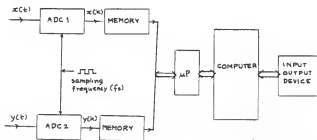


Fig.7.6. Synchronous sampling using two ADCs.

Also the use of an ADC with lower resolution (say 8 bits) would increase the convergence time and the phase error to a certain extent. In fig.7.6 the two signals $x(k)$ and $y(k)$ are simultaneously sampled and the samples are stored in memory 1 and 2 respectively which are then accessed by the microprocessor. The synchronous sampling would reduce the phase error to great extent.

Having considered all of the factors discussed a description of a suitable system for data acquisition and processing as follows. The schematic is shown in fig.7.7.

The two signals are initially conditioned. This includes reduction of noise and automatic magnitude scaling of signals to suit the ADC range. The control circuits and logic control the simultaneous sampling of the ADCs and storing of these samples in memory. The microprocessor then obtains these samples and sends them to the computer for processing. Interaction is provided by the Input/Output device. The most important part of this system is the portion dealing with the acquisition of samples and storing them in memory. This portion is now discussed.

There are basically two methods of controlling the data acquisition :

1. By using the microprocessor
2. Direct Memory Accessing (DMA)

The choice of the method depends on the sampling frequency required. At nominal values of sampling frequency the first method is possible but at higher sampling rates the second method

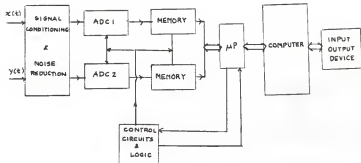


Fig.7.7. Suitable system for implementing the proposed methods.

is desirable. The first method has certain limitations which are now elaborated.

If it is assumed that the microprocessor operates with a clock of 1 MHz. (for the MC6802) then the MPU cycle time is 1 us. This is the time taken for one cycle and each assembly instruction takes a certain number of MPU cycles. The minimum software required to obtain the data samples is:

AGAIN	LDAA	SAMPLE	4 CYCLES
	STAA	SOMEWHERE	4 CYCLES
	INX		4 CYCLES
	CPX #	FIN ADDR.	3 CYCLES
	BNE	AGAIN	4 CYCLES
TOTAL	NUMBER OF CYCLES		19 CYCLES

We shall assume that the total number of cycles is approximately 20 which means that it takes a minimum time of 20us for executing these instructions. If the control signals (gate and flag) for initiating ADC conversion and checking end of conversion are also to be incorporated in software then a minimum of another 20 us may be necessary (20 cycles) for execution and hence the total execution time is 40 us. To this must be added the ADC conversion time which from the manual is given to be 50 us. Hence the overall time is 100 us. Thus the sampling frequency is the reciprocal of this time ie. $1/100\text{us} = 10 \text{ KHz}$. The value of sampling frequency experimentally determined is 9130 Hz. which is approximately the value obtained in theory. Thus higher rates of sampling are not possible with this method.

The DMA method which uses direct memory accessing is capable

of a much higher sampling frequency. Fig.7.8 shows the schematic.

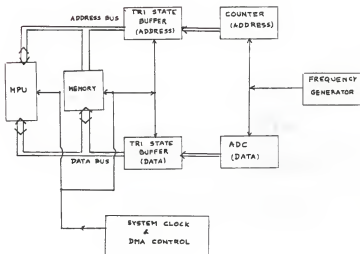


Fig.7.8. Direct Memory Accessing (DMA).

This method is especially suitable when using ADCs which can handle high sampling rates eg. the Flash ADC. Its sampling rate is around 10 MHz. However it is necessary at the same time to have a memory with a short enough access time so that it can take in the data samples given by the ADC at such a high rate.

In the schematic shown two 3-state buffers one for address and one for data are used to access the memory by the external devices. Here a high speed counter is used to provide the address to the memory. The counter is to be synchronized properly with the ADC conversion signals and the other buffer (DB) for data must also be simultaneously used to access the data bus. Thus the microprocessor must essentially disconnect itself from the address and data bus. This may be done by HALTING the uP or by other means depending on the uP used.(example cycle stealing, TSC control etc.) . The DMAC (direct memory access controller) could also be used for achieving DMA.

In order to make the sampling frequency variable a programmable frequency generator (VCO) may be used. A counter may be used to determine the signal frequency which in turn can be used to program the VCO to obtain the sampling frequency. It is essential for proper working of this system all the different parts must be properly synchronised and this requires suitable control logic.

However at high sampling rates the access time of the memory available is a very important aspect to be considered and poses the greatest problem in design and implementation. Also the rise and fall times, the propagation delays and the logic switching

delays must also be taken into account for the system to operate well. It is essential to study different types of memories available CMOS, Schottky TTL, ECL etc. and compare these with respect to speed, cost, amount of memory available, power dissipation etc.

In the system discussed the microprocessor acts as a control/communicating device. The uP is well suited for manipulation at the bit level and hence can do some scaling, bit manipulation, sign insertion etc. The data can then be transferred to a computer for processing and probably storage on disk for future processing.

As far as the processing portion of the system is considered there are again two methods 1. using assembly language 2. high level language. While the first method can handle only highly specific and dedicated algorithms the later has a lot of flexibility. The high level language algorithms in the Computer can be easily written and changed to suit the type of data involved and the processing algorithm itself may be changed for adifferent problem.

Thus by taking into account all these factors discussed the final design may be implemented.

CHAPTER 8

CONCLUSIONS

Before we attempt to conclude, we must make a comparative study of the four methods developed. The overall performance is not only dependent on the algorithm itself but also on the performance of the system used to implement it. However with the existing system the Adaptive method and the FFT method seem to perform well as compared to the other two methods. A comparison of all the methods with respect to a few features is now dealt with.

As far as simulation is concerned the Adaptive, FFT and Correlation methods performed very well. The numerical method had a larger phase error compared to the other three. When actual data samples were used the adaptive and FFT methods worked the best. The effects of noise induced large errors in the numerical method and had to be combated by the use of predictor. Harmonic distortion produced errors in the Correlation method but did not seem to affect the Adaptive and FFT methods. The FFT method has an advantage that it provides other valuable information regarding the signal. The adaptive method has an advantage that it automatically cancels the noise since it uses the cancellation mode. All the methods require that the sampling frequency be variable in order to cover the entire frequency range and be known accurately and also require that the frequency of the signals be known accurately. Having considered all of the above features it seems that the adaptive and FFT methods have the greatest advantages and accuracy. Hence the Adaptive and FFT

methods are recommended for an implementation for the measurement of phase.

CHAPTER 9

OTHER APPLICATIONS

There are several other applications of signal processing techniques in instrumentation. The use of a suitable system for data acquisition enables a variety of algorithms to be used for processing. The use of microprocessors in such applications is increasing and some new dedicated chips called "signal processor" chips are now making their appearance. A brief discussion of other possible applications follows.

Besides the phase and the FFT, the Power Density Spectrum (PDS) of the signal can also be obtained. Also digital filters can be used to shape the signal, alter its characteristics etc. The predictor can be used for signal enhancement and the combating of noise. The adaptive algorithm is a very powerful processing tool. A method for obtaining the amplitude of a sinusoidal signal is suggested as shown in fig.9.1.

It is required to find the amplitude of $x(k)$. Let $r(k)$ be the reference input generated within the computer at the same frequency as that of $x(k)$. The reference input $r(k)$ could also be derived by sampling a 1 volt reference signal of same frequency as that of $x(k)$.

As before when the error $e(k)$ decreases in magnitude to a sufficiently small value then $r(k)$ is transformed in amplitude and phase by W to cancel $x(k)$. We obtain the amplitude transfer function as:

$$H(w_0) = w_0 + w_1 e^{-jw_0 T} + \dots + w_M e^{-jw_0 M T}$$

The magnitude of the transfer function is equal to the square

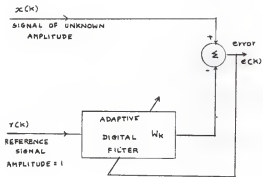


Fig.9.1.A method to find the RMS value of a sine wave.

root of $\text{Re}(H(\omega_0))^2 + \text{Im}(H(\omega_0))^2$.

Since $g(k)$ cancels $x(k)$ at the summing device the magnitude of $x(k)$ is given by the magnitude of the transfer function.

This actually gives the maximum value of the signal. So the maximum value divided by a factor of root 2 would yield the RMS value of $x(k)$. So this is one possible method to obtain the RMS value of a sinusoidal signal given its samples.

Other applications would be in other measurements and in transducer applications. The output of a transducer can be sampled and analysed in the frequency domain to obtain certain important characteristics like frequency content etc. Also after processing the sampled signal, the digital output can be reconverted to analog form by the use of DAC and used. Real time signal processing is now in the limelight.

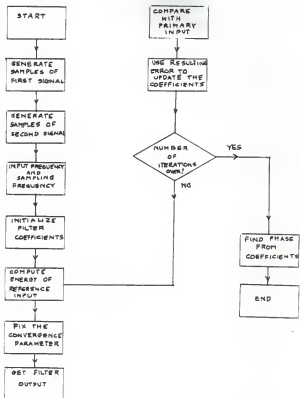
CHAPTER 10

PLOT PACKAGE

A plot package was developed for the HP 9845B which could be useful in plotting data. This data may be available on a floppy disk and the program can be modified to access this data from the disk for subsequent plotting of the data. The program may also be included in the main program as a subroutine and accessed whenever a plot is required. The program listing is given in Appendix 9. The program automatically finds the maximum and minimum values in the data block and plots the data and labels the axes etc. It can be used for plotting random data also. It was used extensively used to plot data obtained from simulation analysis of the different algorithms. It is capable of further modifications and improvements to achieve greater sophistication.

APPENDIX 1 a.

Flow chart for BASIC program. (Adaptive algorithm)



APPENDIX 1 b.

BASIC program listing. (Simulation)

```

10  REM *****
20  REM THIS IS A SIMULATION PROGRAM WHICH GENERATES TWO SINE
30  REM WAVES WITH A KNOWN PHASE DIFFERENCE. THE FREQUENCY OF
40  REM THE SIGNALS AND THE SAMPLING FREQUENCY CAN BE INPUT TO
50  REM THE PROGRAM AND IT USES THE ADAPTIVE FILTERING SCHEME
60  REM (WIDROW'S LMS ALGORITHM) TO COMPUTE THE PHASE DIFFERENCE.
70  REM THIS VALUE CAN BE COMPARED WITH THE ACTUAL PHASE INPUT TO
80  REM THE PROGRAM. THE AMPLITUDE OF THE SIGNALS CAN BE CHANGED IF
90  REM NECESSARY. ALSO THE AMPLITUDE TRANSFER FUNCTION AT THE
100 REM SIGNAL FREQUENCY IS ALSO DETERMINED.
110 REM *****
120 REM FORM THE ARRAYS FOR THE FILTER INPUT X(*) AND THE FILTER
130 REM WEIGHTS W(*)
140 REM *****
150 DIM X(16), W(16)
160 REM *****
170 REM INPUT THE VALUES OF THE SIGNAL FREQUENCY(F), THE SAMPLING
180 REM FREQUENCY(F1) AND THE REQUIRED PHASE DIFFERENCE(P) BETWEEN
190 REM THE GENERATED SIGNALS
200 REM *****
210 INPUT F, F1, P
220 F0=2*PI*F
230 T=1/F1
240 P0=P*PI/180
250 REM *****
260 REM INITIALISE COEFFS. OF THE FILTER
270 REM *****
280 REM *****
290 REM FIX THE NUMBER OF WEIGHTS(COEFFICIENTS OF THE FILTER) N1
300 REM *****
310 N1=16
320 FOR I=1 TO N1
330 W(I)=0
340 NEXT I
350 REM *****
360 REM FIX THE NUMBER OF ITERATIONS
370 REM *****
380 N1=50
390 REM *****
400 REM FIX THE VALUE OF CONVERGENCE PARAMETER (V1)
410 REM *****
420 V1=.002
430 REM *****
440 REM START PROCESSING
450 REM *****
460 FOR K=1 TO N1
470 REM *****
480 REM GEN Y THE PRIMARY INPUT TO THE FILTER
490 REM *****
500 Y=SIN(F0*K*T+P0)
510 REM *****
520 REM GEN X1 THE REFERENCE INPUT TO THE FILTER
530 REM *****
540 X1=SIN(F0*K*T)
550 REM *****
560 REM ESTIMATE THE INPUT SIGNAL ENERGY
570 REM TO UPDATE THE CONVERGENCE PARAMETER V
580 REM *****
590 V2=V2*(1-V1)+V1*X1*X1
600 IF V2<.001 THEN 430

```

```

510  U=U1/√2
520  GOTO 550
530  U=0
540  REM *****
550  REM FIND THE N PART VALUES OF X1 TO FORM X THE INPUT TO THE FILTER
560  REM TO FIND THE FILTER OUTPUT G
570  REM *****
580  FOR J=1 TO N
590  M=J-1
600  X(J)=SIN(F0*T+(K-M))
610  NEXT J
620  G=0
630  REM *****
640  REM FIND THE FILTER OUTPUT G
650  REM *****
660  FOR L=1 TO N
670  G1=W(L)*X(L)
680  G=G+G1
690  NEXT L
700  REM *****
710  REM COMP ERROR BETWEEN THE PRIMARY OUTPUT AND THE FILTER OUTPUT
720  REM *****
730  E=Y-G
740  PRINT E,
750  REM *****
760  REM UPDATE COEFFICIENTS OF THE FILTER USING THE ERROR E
770  REM *****
780  FOR I=1 TO M
790  W(I)=W(I)+U*E*X(I)
800  NEXT I
810  NEXT K
820  PRINT " "
830  PRINT "COEFFICIENTS"
840  FOR I=1 TO N
850  REM PRINT COEFFS.
860  PRINT W(I),
870  NEXT I
880  REM *****
890  REM FIND PHASE FROM COEFFS. OF THE FILTER. 0 IS THE REAL PART OF
900  REM THE TRANSFER FUNCTION AND Z IS THE IMAGINARY PART.
910  REM *****
920  S=0
930  FOR I=1 TO N
940  S=S+W(I)*COS(F0*T+(I-1))
950  NEXT I
960  Z=0
970  FOR J=1 TO N
980  Z=Z+W(J)*SIN(F0*T+(J-1))
990  NEXT J
1000 Z=Z/S
1010 PRINT "RESULT ",R
1020 C=ATN(R)
1030 C=C*180/PI
1040 REM *****
1050 REM ADJUST THE VALUE OF THE OBTAINED PHASE TO COVER THE PHASE RANGE
1060 REM FROM 0 THRU 360 USING THE INFORMATION OF THE SIGN OF S AND Z
1070 REM *****
1080 IF (S<0) AND (Z>0) THEN C1=180-C
1090 IF (S<0) AND (Z<0) THEN C1=180+C
1100 IF (S>0) AND (Z<0) THEN C1=360-C
1110 IF (S>0) AND (Z>0) THEN C1=C
1120 REM *****
1130 REM PRINT THE VALUE OF PHASE ON THE TERMINAL
1140 REM *****
1150 PRINT "PHASE=",C
1160 REM *****
1170 REM PRINT THE VALUE OF ACTUAL PHASE TO BE COMPARED WITH THE
1180 REM CALCULATED VALUE
1190 REM *****
1200 PRINT "THE ACTUAL PHASE DIFFERENCE IS",P
1210 REM *****
1220 REM FIND THE AMPLITUDE TRANSFER FUNCTION
1230 REM *****
1240 A=1/SQR(Z*Z+S*S)
1250 PRINT "AMPLITUDE TFN=",Amp1
1260 END

```

APPENDIX 1 c.

BASIC program listing (Actual data samples)

```

10 REM *****
20 REM THIS PROGRAM DETERMINES THE PHASE DIFFERENCE BETWEEN TWO
30 REM SIGNALS BY USING THE ADAPTIVE FILTER ALGORITHM.
40 REM ACTUAL SAMPLES OF THE TWO SIGNALS ARE TAKEN USING THE
50 REM ADC CARD IN THE HP MULTIPROGRAMMER AND SENT TO THE
60 REM NORTH STAR FOR PROCESSING BY THE ALGORITHM
70 REM THE SAMPLES OF THE FIRST SIGNAL ARE TAKEN INDICATED
80 REM BY THE FLICKERING OF LEDS ON THE HPM PANEL
90 REM THE SAMPLES ARE THEN SENT TO THE NS IN APPROX 30 SECS
100 REM AFTER WHICH THE SAMPLES OF THE SECOND SIGNAL ARE TAKEN
110 REM INDICATED BY THE LEDS ON THE HPM PANEL.
120 REM AFTER PROCESSING BY THE ALGORITHM THE RESULTS ARE DISPLAYED
130 REM ON THE SILENT TERMINAL.
140 REM IT IS ASSUMED THE FIRST SIGNAL LEADS THE SECOND SIGNAL
150 REM IN PHASE... THE RESULTS SHOW A VALUE OF 360-P WHERE
160 REM P IS THE PHASE DIFFERENCE IF THE SIGNALS ARE INTERCHANGED
170 REM *****
180 REM FORM THE ARRAYS FOR STORING THE SAMPLES OF THE TWO SIGNALS
190 REM*****
200 REM Y AND X1 ARE USED FOR THIS
210 REM X USED INITIALLY FOR STORING SAMPLES AND THEN THIS IS
220 REM STORED IN X1, X IS THEN USED FOR CURRENT PROCESSING
230 DIM X(512),Y(512),W(16),X1(512)
240 REM*****
250 REM CLEAR THE ACIA IN INTERFACE(MC 6802)
260 REM*****
270 FOR I=1 TO 10
280 INPUT#1,D$
290NEXT I
300 REM*****
310 REM SEND COMMAND TO INVOKE THE ADC SUBROUTINE IN INTERFACE
320 REM*****
330 PRINT#1,"A"
340 REM*****
350 REM GET THE 512 SAMPLES AND STORE IN Y(*)
360 REM*****
370 FOR I=1 TO 512
380 INPUT#1,D$
390Y(I)=VAL(D$)
400 NEXT I
410 REM*****
420 REM CLEAR THE ACIA AGAIN
430 REM*****
440 FOR I=1 TO 10
450 INPUT#1,D$

```

```

460 NEXT I
470 REM*****
480 REM ISSUE COMMAND TO SAMPLE THE SECOND SIGNAL
490 REM*****
500 PRINT#1,"A"
510 REM*****
520 REM GET 512 SAMPLES OF SECOND SIGNAL
530 REM*****
540 FOR I= 1 TO 512
550 INPUT#1,D#
560 X(I)=VAL(D#)
570 NEXT I
580 REM*****
590 REM SCALING THE SAMPLES TO OBTAIN THE ACTUAL DECIMAL VALUES
600 REM FOR THE FIRST SIGNAL
610 REM*****
620 FOR L=1 TO 512
630 IF Y(L) < 2048 THEN GOTO 650
640 Y(L)=Y(L)-4096
650 Y(L)=Y(L)*.005
660 NEXT L
670 REM*****
680 REM PRINT ABOUT 25 SAMPLES OF FIRST SIGNAL
690 REM*****
700 FOR L=1 TO 25
710 ! Y(L),
720 NEXT L
730 PRINT"/"
740 REM*****
750 REM SCALING FOR THE SECOND SIGNAL
760 REM*****
770 FOR I=1 TO 512
780 IF X(I) < 2048 THEN GOTO 800
790 X(I)=X(I)-4096
800 X(I)=X(I)*.005
810 NEXT I
820 REM*****
830 REM PRINT 25 SAMPLES OF THE SECOND SIGNAL
840 REM*****
850 FOR I=1 TO 25
860 ! X(I),
870 NEXT I
880 REM*****
890 REMSCALE THE AMPLITUDES OF THE SIGNALS TO 1V
900 REM*****
910 FOR I= 1 TO 512

```

```

920Y(I)=.1*Y(I)
930 X1(I)=X(I)*.1
940NEXT I
950 : " "
960 REM*****
970 REM GET THE FREQUENCY OF THE SIGNALS
980 REM*****
990 : "FREQ",
1000INPUT F
1010 REM*****
1020 REM START PROCESSING
1030 REM*****
1040 F0=2*(22/7)*F
1050 F1=9130
1060 T=1/9.13E3
1070 REM*****
1080 REM FIX THE VALUES FOR THE NUMBER OF WEIGHTS FOR THE
1090 REM DIGITAL FILTER
1100 REM FIND THE RATIO OF SAMPLING FREQUENCY TO THE SIGNAL FREQUENCY
1110 REM*****
1120 IF F1/F (<=15 THEN GOSUB 1160
1130 IF (F1/F (<=11 AND F1/F)15) THEN GOSUB 1200
1140 IF (F1/F(<=9 AND F1/F)11) THEN GOSUB 1240
1150 IF (F1/F(<=5 AND F1/F)9) THEN GOSUB 1280
1160 N=8
1170 V1=.006
1180 N1=200
1190 RETURN
1200 N=4
1210 V1=.004
1220 N1=300
1230 RETURN
1240 N=4
1250 V1=.003
1260 N1=400
1270 RETURN
1280 N=3
1290 V1=.003
1300 N1=400
1310 RETURN
1320REM*****
1330 REM START PROCESSING USING THE DETERMINED VALUES OF
1340 REM THE WEIGHTS(N), CONVERGENCE PARAMETER(V1),AND
1350 REM THE NUMBER OF ITERATIONS(N1)
1360REM*****
1370 INITIALIZE THE WEIGHTS(COEFFICIENTS) OF THE FILTER

```

```

1380REM*****
1390 FOR I= 1 TO 16
1400 W(I) = 0
1410 NEXT I
1420 FOR K=N TO N1
1430REM*****
1440 REM FIND INPUT SIGNAL ENERGY TO UPDATE THE
1450 REM CONVERGENCE PARAMETER
1460REM*****
1470 V2=V2*(1-V1)+V1*X1(K)*X1(K)
1480 IF V2<.001 THEN 1510
1490 V=V1/V2
1500 GOTO 1530
1510 V=0
1520REM*****
1530REM GEN. X
1540REM*****
1550 FOR J=1 TO N
1560 Y=J-1
1570 X(J)=X1(K-Y)
1580 NEXT J
1590 G=0
1600REM*****
1610 REM FIND THE OUTPUT OF THE FILTER G
1620REM*****
1630 FOR L= 1 TO N
1640 G1=W(L)*X(L)
1650 G=G+G1
1660 NEXT L
1670REM*****
1680 REM COMP. ERROR BETWEEN PRIMARY INPUT & FILTER OUTPUT
1690REM*****
1700 E=Y(N)-G
1710 FOR I=1 TO N
1720REM*****
1730 REM UPDATE COEFFS.
1740REM*****
1750 W(I)=W(I)+V*E*X(I)
1760 NEXT I
1770 NEXT K
1780 ! " / "
1790 ! "COEFFICIENTS"
1800 FOR I=1 TO N
1810 REM PRINT COEFFS.
1820 ! W(I).
1830 NEXT I

```

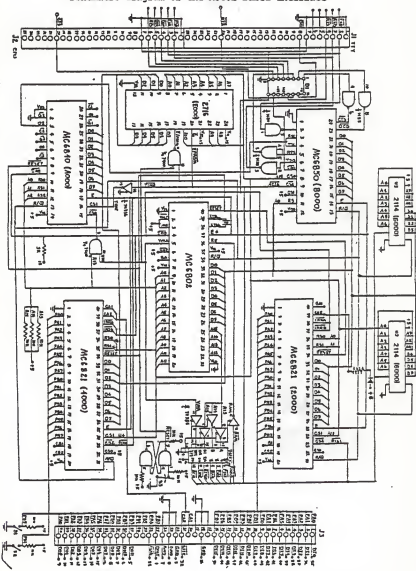
```

1840REM*****
1850 REM FIND PHASE FROM COEFFS.
1860REY*****
1870 REM S IS REAL PART OF THE TRANSFER FUNCTION
1880REM*****
1890 S=0
1900 FOR I=1 TO N
1910 S=S+W(I)*COS(F0*7*(I-1))
1920 NEXT I
1930REM*****
1940 REM Z IS THE IMAGINARY PART OF THE TRANSFER FUNCTION
1950REM*****
1960 Z=0
1970 FOR J=1 TO N
1980 Z=Z+W(J)*SIN(F0*7*(J-1))
1990 NEXT J
2000 Z=-Z
2010 R=Z/S
2020 : "RESULT", R
2030 C=ATN(R)
2040 C=C+180/(22/7)
2050REY*****
2060 REM ADJUST THE VALUE OF THE OBTAINED PHASE TO
2070 REM COVER THE ENTIRE RANGE (0 THRO 360) USING THE
2080 REM INFORMATION FROM S AND Z SINCE THE FILTER CAN
2090 REM CAN DISTINGUISH A RANGE (0 THRO 90 ONLY)
2100 REM PRINT THE OBTAINED PHASE AFTER ADJUSTING ON THE SILENT
2110 REM TERMINAL
2120REM*****
2130 IF S(0) AND Z(0) THEN ! "THE PHASE DIFFERENCE IS",180+C
2140 IF S(0) AND Z(0) THEN ! "THE PHASE DIFFERENCE IS",C+180
2150 IF S(0) AND Z(0) THEN ! "THE PHASE DIFFERENCE IS",360+C
2160 IF S(0) AND Z(0) THEN ! "THE PHASE DIFFERENCE IS",C
2170REY*****
2180 REM PRINT THE ACTUAL VALUE OF PHASE FROM RC CKT.
2190 REM HERE THE VALUES OF R AND C ARE FIXED
2200REY*****
2210 !ATN(1/(F0*101.15E3*4.394E-9))*180/(22/7)
2220REM*****
2230 REM THE ACTUAL PHASE CAN BE COMPARED WITH THE OBTAINED PHASE
2240REY*****
2250 END

```


APPENDIX 2

Schematic diagram of the M6802 Based Interface



APPENDIX 3

6802 Monitor Program Listing.

MOTOROLA M68SAM CROSS-ASSEMBLER

PAGE 1

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MOTOROLA M6800 CROSS ASSEMBLER, RELEASE 1.1

00001		NAM	MICRO1
00002	0050	STAKPT EQU	150
00003	A000	PTMC1 EQU	1A000
00004	A004	PTMB EQU	1A004
00005	A001	PTMC2 EQU	1A001
00006	0000	INCL EQU	100
00007	00E0	INCH EQU	1E0
00008	0011	MEM EQU	111
00009	0012	MEM1 EQU	112
00010	2001	CRA1 EQU	12001
00011	2003	CAB1 EQU	12003
00012	2000	DDR11 EQU	12000
00013	2002	DDR11 EQU	12002
00014	4001	CRA2 EQU	14001
00015	4003	CAB2 EQU	14003
00016	4000	DDR21 EQU	14000
00017	4002	DDR21 EQU	14002
00018	8000	ACIAS EQU	18000
00019	8000	ACIAC EQU	ACIAS
00020	8001	ACIARO EQU	18001
00021	8001	ACIAT EQU	ACIARO
00022	0010	CP EQU	110
00023	000E	VALL EQU	10E
00024	000F	VALH EQU	10F
00025	000C	THPL EQU	10C
00026	0000	THPH EQU	100
00027	0008	KNT EQU	108
00028	000A	MARK EQU	10A
00029	0006	LS03 EQU	106
00030	0007	LS02 EQU	107
00031	0008	LS01 EQU	108
00032	0009	LS0 EQU	109
00033	0001	SUBHI EQU	101
00034	0000	SUBLD EQU	100
00035	0003	HINH1 EQU	103
00036	0002	HINLO EQU	102
0003T	0004	NEG EQU	104
00038	0005	CARRY EQU	105
00039	0014	TEMP EQU	114
00040	0016	TYPE EQU	116
00041	0017	TVALL EQU	117
00042	0018	TVALL EQU	118
00043	0019	FFLAG EQU	119
00044	E000	ORG	1E000
00045	E000 00	FCB	100, 15F, 100, 117, 100, 105, 100, 102
	E001 5F		
	E002 00		
	E003 1T		
	E004 00		
	E005 05		

E006 00
E007 02

```

00046
00047
00048
00049
00050
00051 E008 8E 0050      LOS      #STAKPT
00052 E008 86 0C        LOA A   #S00
00053 E000 87 2001      STA A   CRA1
00054 E010 87 2003      STA A   CRB1
00055 E013 86 FF        LDA A   #SFF
00056 E015 87 2000      STA A   DDRAI
00057 E018 87 2002      STA A   OORB1
00058 E018 86 04        LOA A   #S04
00059 E010 87 2003      STA A   CRB1
00060 E020 86 3C        LOA A   #S3C
00061 E022 87 2001      STA A   CRA1
00062 E025 86 00        LOA A   #S00
00063 E027 87 4001      STA A   CRA2
00064 E02A 87 4003      STA A   CRB2
00065 E020 87 4000      STA A   OORA2
00066 E030 87 4002      STA A   OORB2
00067 E033 86 04        LOA A   #S04
00068 E035 87 4003      STA A   CRB2
00069 E038 86 3C        LOA A   #S3C
00070 E03A 87 4001      STA A   CRA2
00071 E030 86 03        LOA A   #S03
00072 E03F 87 8000      STA A   ACIAC
00073 E042 86 01        LOA A   #S01
00074 E044 87 8000      STA A   ACIA5
00075 E047 86 4002      LOA A   OORB2
00076 E04A 84 30        AND A   #S30
00077 E04C 44          LSR A
00078 E040 44          LSR A
00079 E04E 44          LSR A
00080 E04F 88 00        AOO A   #INOL
00081 E051 97 12        STA A   MEM1
00082 E053 86 E0        LOA A   #INOM
00083 E055 89 00        ADC A   #S00
00084 E057 97 11        STA A   MEM
00085 E059 0E 11        LOX   MEM
00086 E05B EE 00        LOX   0:X
00087 E050 86 93        LOA A   #S93
00088 E05F 87 A001      STA A   PTMC2
00089 E062 86 00        LOA A   #S00
00090 E064 87 A000      STA A   PTMC1
00091 E067 FF A004      STX   PTM8
00092 E06A 86 83        LDA A   #S83
00093 E06C 87 A001      STA A   PTMC2
00094
00095
00096
00097
00098

```

* PIA, ACIA, AND PTM INITIALIZATION *

POINT CRA1 AT OORA1

```

00099
00100
00101
00102
00103
-----
*
*
*           MAIN ROUTINE TO CHECK FOR VARIOUS COMMANDS C,?,@,%, 'A'
*
*
00104 E06F 80 E18F LOOKCO JSR   GETONE   GET THE NEXT CHAR FROM NS WHEN SENT
00105 E072 81 26      CMP A   #826
00106 E074 26 05      BNE   CKQM
00107 E076 80 E100    JSR   DOAT
00108 E079 20 F4      BRA   LOOKCO
00109 E078 81 3F      CKQM   CMP A   #83F
00110 E070 26 05      BNE   CKLB   IF NOT ? THEN CHECK IF IT IS #
00111 E07F 80 E200    JSR   RONSNO READ MP   AND SEND RESULT TO N.STAR
00112 E082 20 E8      BRA   LOOKCO
00113 E084 81 23      CKLB   CMP A   #823   CHECK IF CHAR EQUALS #
00114 E086 26 05      BNE   CKREL   IF NOT "8" THEN CHECK IF RELAY COM "8"
00115 E088 80 E400    JSR   TIMER   INITIALIZE V/F CARD & TIME I7
00116 E088 20 E2      BRA   LOOKCO
00117 E08D 81 25      CKREL   CMP A   #825   CHECK IF IT IS A RELAY COMMAND
00118 E08F 26 05      BNE   CKVCD   IF NOT RELAY THEN CHECK IF IT IS VCD COM
00119 E091 8D E500    JSR   RELAY
00120 E094 20 09      BRA   LOOKCO
00121 E096 81 21      CKVCD   CMP A   #821   CHECK IF VCD COMMAND IE " "
00122 E098 26 05      BNE   CADC    IF NOT VCD COMMAND THEN CHECK IF AOC COM
00123 E09A 80 E600    JSR   VCD
00124 E090 20 00      BRA   LOOKCO
00125 E09F 81 41      CAOC   CMP A   #841   CHECK IF A/O CONVERTER COMMAND. IE "A"
00126 E0A1 26 CC      BNE   LOOKCO
00127 E0A3 80 E198    JSR   ADC
00128 E0A6 20 CT      BRA   LOOKCO
-----
*
*
*           E N D   O F   M A I N   R O U T I N E
*
*
00130
00131
00132
00133
-----
*
*
*
*
00134
00135
00136
00137
00138
00139
00140
00141
00142
00143
-----
*
*
*           DOAT SUBROUTINE - TO ACCEPT CHARACTERS FROM N.STAR UNTIL $
*
*
*           CONVERT TO BINARY, AND SEND RESULT TO HP
*
*
00144
00145
00146
00147
-----
00148 E100          ORG   $E100
00149 E100 80 E108 DOAT   JSR   NSREAD  GET CHARS FROM N.STAR UNTIL "$"
00150 E103 96 10    LOA A  CP           IF NOTHING SENT THEN EXIT
00151 E105 27 03    BEQ   XDOAT
00152 E107 80 E168 JSR   STOMP
00153 E10A 39      XDOAT RTS          SEND VALUE IN VALL & VALH TO HP

```

```

00154
00155
00156
00157
00158
00159
00160
00161 E108 86 00
00162 E100 97 10
00163 E10F 97 0E
00164 E111 97 0F
00165 E113 97 0C
00166 E115 97 00
00167 E117 97 08
00168 E119 4C
00169 E11A 97 0A
00170 E11C 80 E18F
00171 E11F 81 24
00172 E121 27 08
00173 E123 80 30
00174 E125 3A
00175 E126 7C 0010
00176 E129 20 F1
00177 E12B 96 10
00178 E120 27 38
00179 E12F 32
00180 E130 06 08
00181 E132 26 00
00182 E134 97 0E
00183 E136 7C 0008
00184 E139 96 10
00185 E13B 91 08
00186 E130 27 28
00187 E13F 20 EE
00188 E141 5F
00189 E142 0C
00190 E143 48
00191 E144 59
00192 E145 97 0C
00193 E147 07 00
00194 E149 48
00195 E14A 59
00196 E148 48
00197 E14C 59
00198 E140 0C
00199 E14E 98 0C
00200 E150 09 00
00201 E152 7A 0008
00202 E155 26 EB
00203 E15T 98 0E
00204 E159 09 0F
00205 E158 97 0E
00206 E150 07 0F
00207 E15F 7C 000A
00208 E162 96 0A

```

```

*****
*
* NSREAD - SUBROUTINE TO READ N.STAR & CONVERT TO BINARY
* THIS ROUTINE WILL RECEIVE ASCII DIGITS FROM THE N.STAR UNTIL A "*"
* CONVERT TO 16 BIT BINARY & STORE RESULT IN VALL & VALM.
*
*****
NSREAD  LOA  A  #500
          STA  A  CP
          STA  A  VALL
          STA  A  VALM
          STA  A  TMPL
          STA  A  THPM
          STA  A  KNT
          INC  A
          STA  A  MARK
LOOKK2  JSR  GETONE  GET THE NEXT CHAR FROM NS WHEN SENT
          CMP  A  #824
          BEQ  CHECK  IF "*" THEN GET OUT
          SUB  A  #830
          FSN  A
          INC  CP
          BRA  LOOKK2
CHECK   LDA  A  CP
          BEQ  XNSRD
NEXT    PUL  A
          LOA  B  KNT
          BNE  KNTPOS
          STA  A  VALL
          INC  KNT
          LOA  A  CP
          CMP  A  KNT
          BEQ  XNSRD
          BRA  NEXT
KNTPOS  CLR  B
X10     CLC
          ASL  A
          ROL  B
          STA  A  TMPL
          STA  B  THPM
          ASL  A
          ROL  B
          ASL  A
          ROL  B
          CLC
          ADD  A  TMPL
          ADC  B  THPM
          OEC  KNT
          BNE  X10
          ADD  A  VALL
          ADC  B  VALM
          STA  A  VALL
          STA  B  VALM
          INC  MARK
          LDA  A  MARK

```

```

00209 E164 9T 08          STA A  KNT
00210 E166 91 10          CMP A  CP
00211 E168 26 C5          BNE  NEXT
00212 E16A 39             XNSRD RTS
00213
00214
00215                     *
00216                     * STOMP - SEND TO HP SUBROUTINE
00217                     * THIS SUBROUTINE WILL SEND 2 BYTES STORED IN VALL & VALM TO
00218                     * THE 16 BIT IN PORT OF HP - VALM TO HIGHER BYTE & VALL TO LOWER...
00219                     *
00218 E168 96 0E          STOMP LOA A  VALL
00219 E160 87 2000         STA A  OGRA1
00220 E170 96 0F          LOA A  VALM
00221 E172 8T 2002         STA A  OGRB1
00222 E175 02             NOP
00223 E176 02             NOP
00224 E177 86 4001         LOA A  CRA2
00225 E17A 84 F7          ANO A  #8F7
00226 E17C 8T 4001         STA A  CRA2
00227 E17F 86 4002         LOOKC3 LOA A  OGRB2
00228 E182 84 40          ANO A  #840
00229 E184 2T F9          BEQ  LOOKC3
00230 E186 86 4001         LOA A  CRA2
00231 E189 8A 08          ORA A  #108
00232 E18B 8T 4001         STA A  CRA2
00233 E18E 39             XSTOMP RTS
00234
00235
00236
00237
00238                     *
00239                     * GETONE - SUBROUTINE TO GET 1 CHAR (IN REG A) FROM NS, WHEN SENT
00240                     *
00239 E18F 86 8000         GETONE LOA A  AC1A5
00240 E192 4T              ASR A
00241 E193 24 FA          BCC  GETONE
00242 E195 86 8001         LOA A  AC1A0
00243 E198 39             XGETON RTS
00244
00245
00246
00247
00248
00249                     *
00250                     * AOC - SUBROUTINE TO COLLECT 512 (1/2 K) SAMPLES AND
00251                     * SEND THEM TO THE NORTH STAR. THE ADC CARD IS ASSUMED TO
00252                     * BE IN SLOT 405 OF THE MULTIPROGRAMMER
00253                     *
00251 E198                ORG  $E198
00252 E198 CE 80F0 AOC     LOX  #80F0   SEND CONTROL WORD "1111 0000 1011 0000"
00253 E19E 0F 0E          STX  VALL
00254 E1A0 80 E16B       JSR  STOMP
00255 E1A3 CE 0050       LOX  #8050   ADC CARD IS IN SLOT 405
00256 E1A6 0F 0E          STX  VALL
00257 E1A8 CE 6000       LDX  #6000   ← MODIFICATIONS START HERE.
00258 E1AB 80 E16B ANEXT JSR  STOMP   ADDRESS SLOT 5 WITH GATE SIGNAL
00259 E1AE 86 4000       LOA A  OGRA2  GET THE SAMPLE
00260 E1B1 A7 00        STA A  0,X    STORE IT AT LOCATIONS 6000 ONWARDS
00261 E1B3 08           INX
00262 E1B4 86 4002       LOA A  OGRB2  GET THE UPPER BYTE OF THE SAMPLE
00263 E1B7 A7 00        STA A  0,X

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00264 E189 08          INX
00265 E18A 8C 6400'   CPX   #86400   COLLECT 512 SAMPLES IN ALL.  IE 1K BYTES
00266 E180 26 EC      SNE   ANEXT
00267          *
00268          *
00269          *
00270 E18F CE 6000     LDX   #86000   GET READY TO SEND THE SAMPLES TO NORTH STAR
00271 E1C2 A6 00     AAGAIN LOA A 0,X   GET THE SAMPLE STORED IN MEMORY
00272 E1C4 43        COM A
00273 E1C5 97 00     STA A 00      TAKE 1'S COMPLEMENT OF THE LOWER BYTE
00274 E1C7 08        INX
00275 E1C8 A6 00     LOA A 0,X   GET THE HIGHER BYTE OF THE SAMPLE
00276 E1CA 43        COM A
00277 E1CB 84 0F     ANO A #80F   MASK OFF THE HIGHER 4 BITS
00278 E1C0 97 01     STA A 01
00279 E1CF 08        INX
00280 E100 0F 15     STX   $15    SAVE THE SAMPLE POINTER
00281 E102 CE 003F   LOX   #8003P
00282 E105 80 E4TE   JSR   TIMEOUT WAIT FOR ACIA TO GET READY
00283 E108 80 E224   JSR   WSEND   SEND THE SAMPLE
00284 E10B 80 E209   JSR   LOOKC1  SEND A CARRIAGE RETURN
00285 E10E 0E 15     LOX   $15
00286 E1E0 8C 6400   CPX   #86400
00287 E1E3 26 00     SNE   AAGAIN REPEAT UNTIL ALL 512 SAMPLES ARE SENT
00288 E1E5 39        XAOC   RTS
00289 E200           ORG   SE200
00290          *
00291          *
00292          *
00293          *   RONSND - READ AND SEND TO N.STAR SUBROUTINE
00294          *   THIS ROUTINE WILL READ NP ADDRESSED PORT AND SEND THE CORRESP
00295          *   DECIMAL VALUE -I IN ASCII)- TO THE NORTH STAR FOLLOWED BY "CR"
00296          *
00297          *
00298          *
00299          *
00299 E200 CE 0034   RONSND LOX   #80034   GET A DELAY COUNT
00298 E203 80 E4TE   JSR   TIMEOUT   GIVE N.STAR NANCICAP OF 1/10 SEC
00299 E206 80 E216   JSR   IOAT      READ NP AND SEND TO N.STAR
00300 E209 86 8000   LOOKC1 LOA A AC1AS
00301 E20C 46       ROR A
00302 E20D 46       ROR A
00303 E20E 24 F9    SCC   LOOKC1
00304 E210 86 00   LOA A #800
00305 E212 87 8C01 STA A AC1AT
00306 E215 39      XRONSN RTS
00307          *
00308          *
00309          *
00310          *   ROUTINE TO READ 16 BITS OF HP, CONVERT 12 BIT BINARY TO BCD TO
00311          *   ASCII, AND SEND IT SERIALLY TO NORTH STAR
00312          *
00313          *
00313 E216 86 4000   IOAT  LOA A ODRA2   READ LOWER BYTE OF MULTIPROGRAMMER
00314 E219 43       COM A
00315 E21A 97 00     STA A SUBLO    1'S COMPLEMENT OF LOW BYTE
00316 E21C 86 4002 LOA A ODRB2    LOW BYTE OF SUBTRANEND
00317 E21F 43       COM A
00318 E220 84 0F     ANO A #80F    READ UPPER BYTE OF MULTIPROGRAMMER
00318 E220 84 0F     ANO A #80F    1'S COMPLEMENT OF HIGH BYTE
00318 E220 84 0F     ANO A #80F    MASK OFF UPPER 4 BITS

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00319 E222 9T 01          STA A  SUBHI      HIGH BYTE OF SUBTRAHEND
00320 E224 CE 0006 MSEND  LOX   #LS03
00321 E227 86 03          LDA A  #803
00322 E229 C6 00          LDA B  #800
00323 E22B E7 00          INC   STA B  0xX      CLEAR CURRENT DIGIT
00324 E22D 08
00325 E22E 4A            DEC A
00326 E22F 2A FA          BPL   INC
00327 E231 86 E8          LDA A  #8E8
00328 E233 1T 02          STA A  HINLO
00329 E235 86 03          LDA A  #803
00330 E237 9T 03          STA A  HINH1      LOAD 1000 INTO HINHEND
00331 E239 8D E300 POS1   JSR   SUB16      BRANCH TO 16 BIT SUBTRACTION ROUTINE
00332 E23C 96 04          LDA A  NEG        CHECK TO SEE IF SUBTRACTION RESULT IS NEGATIVE
00333 E23E 81 0C          CMP A  #800
00334 E240 26 05          BNE  NEGR1
00335 E242 TC 0006       INC   LSD3
00336 E245 20 F2          BRA  POS1
00337 E247 96 00          NEGR1 LDA A  SUBLO     RESULT WAS NEGATIVE, ADD 1000 BACK
00338 E249 88 E8          ADD A  #8E8
00339 E24B 9T 00          STA A  SUBLO
00340 E24D 96 01          LDA A  SUBHI
00341 E24F 89 03          ADC A  #803
00342 E251 9T 01          STA A  SUBH1
00343 E253 86 64          LOA A  #864
00344 E255 9T 02          STA A  HINLO
00345 E257 86 DC          LOA A  #800
00346 E259 9T 03          STA A  HINH1      LOAD 100 INTO HINHEND
00347 E25B 8D E300 POS2   JSR   SUB16      BRANCH TO 16 BIT SUBTRACT ROUTINE
00348 E25E 96 04          LDA A  NEG
00349 E260 81 00          CMP A  #800
00350 E262 26 05          BNE  NEGR2
00351 E264 TC 000T       INC   LSD2
00352 E26T 20 F2          BRA  POS2
00353 E269 86 64          NEGR2 LDA A  #864     RESULT WAS POSITIVE, GO SUBTRACT ANOTHER 100
00354 E26B 98 00          ADD A  SUBLO     RESULT WAS NEGATIVE, ADD 100 BACK
00355 E26D 9T 00          STA A  SUBLO
00356 E26F 96 01          LDA A  SUBHI
00357 E271 89 00          ADC A  #800
00358 E273 9T 01          STA A  SUBH1
00359 E275 86 0A          LOA A  #80A
00360 E277 9T 02          STA A  HINLO     LOAD 10 INTO MINUEND
00361 E279 TF 0003       CLR   MINH1
00362 E27C 8D E300 POS3   JSR   SUB16      BRANCH TO 16 BIT SUBTRACT ROUTINE
00363 E27F 96 04          LDA A  NEG
00364 E281 81 00          CMP A  #800
00365 E283 26 05          BNE  NEGR3
00366 E285 7C 0008       INC   LSD1
00367 E28B 20 F2          BRA  POS3
00368 E28A 96 00          NEGR3 LDA A  SUBLO     RESULT WAS POSITIVE, SUBTRACT ANOTHER 10
00369 E28C 88 0A          ADD A  #80A
00370 E28E 9T 09          STA A  LSD
00371 E290 C6 04          LDA B  #804
00372 E292 09
00373 E293 86 30          ASC1I DEX          RESULT WAS NEGATIVE, ADD 10 BACK
                                LOA A  #830      RESULT IS LSD

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00374 E295 A8 00      ADD A 0,X      BCD TO ASCII
00375 E297 A7 00      STA A 0,X
00376 E299 5A        DEC B
00377 E29A 26 F6      BNE ASCII
00378 E29C CE 0006    LDX #LS03     GET READY TO SEND IT BACK
00379 E29F C6 04      LDA B #504
00380 E2A1 07 08      STA B KNT     INITIALIZE COUNTER KNT
00381 E2A3 A6 00      NEXT1        LOA A 0,X     COMPARE THE CHARACTER WITH "0"
00382 E2A5 81 30      CMP A #530
00383 E2A7 27 11      BEQ ZERO
00384 E2A9 F6 8000    LOOKC4       LOA B ACIAS   READ ACIA STATUS REG
00385 E2AC 56        RDR B
00386 E2AD 56        RDR B
00387 E2AE 24 F9      BCC LOOKC4
00388 E2B0 87 8001    STA A ACIAT   WRITE CHARACTER TO ACIA
00389 E2B3 08        INX
00390 E2B4 7A 0008    DEC KNT
00391 E2B7 26 11      BNE PATCH     NO MORE ZEROS CAN BE SUPPRESSED
00392 E2B9 39        XIGAT       RTS
00393 E2BA 06 08      ZER0        LOA B KNT
00394 E2BC C1 01      CMP B #501
00395 E2BE 26 04      BNE NOT
00396 E2C0 86 30      LOA A #530
00397 E2C2 20 E5      BRA LOOKC4
00398 E2C4 7A 0008    NOT        DEC KNT
00399 E2C7 08        INX
00400 E2C8 20 09      BRA NEXT1    ZEROS STILL BEING SUPPRESSED
00401 E2CA A6 00      PATCH     LOA A 0,X     NO FURTHER ZERO SUPPRESSION
00402 E2CC 20 08      BRA LOOKC4
00403 E300            ORG SE300
00404
-----
00405 *
00406 *      16 BIT SUBTRACTION SUBROUTINE
00407 *
-----
00408
00409 E300 96 03      SUB16       LOA A MINHI
00410 E302 36        PSN A
00411 E303 96 02      LOA A MINLO
00412 E305 36        PSN A
00413 E306 7F 0005    CLR CARRY   PUT HIGH AND LOW BYTE OF MINUEND ON THE STACK
00414 E309 7F 0004    CLR NEG
00415 E30C 43        CDM A
00416 E30D 0C        CLC
00417 E30E 88 01      ADD A #501   TWO'S COMPLIMENT OF MINLO
00418 E310 24 03      BCC LI
00419 E312 7C 0005    INC CARRY
00420 E315 06 03      LI         LOA B MINHI
00421 E317 53        CDM B
00422 E318 08 05      ADD B CARRY  TWO'S COMPLIMENT OF MINHI
00423 E31A 97 02      STA A MINLO
00424 E31C 07 03      STA B MINHI
00425 E31E 7F 0005    CLR CARRY
00426 E321 0C        CLC
00427 E322 96 00      LOA A SUBLO
00428 E324 98 02      ADD A MINLO  SUBTRACT THE LOW BYTES

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00429 E326 24 03      BCC    L2
00430 E328 TC 0005    INC    CARRY
00431 E328 06 01      LDA    B    SUBHI
00432 E320 08 03      AOD    B    MINHI      SUBTRACT THE HIGH BYTES
00433 E32F 08 05      AOD    B    CARRY
00434 E331 97 0C      STA    A    SUBLO
00435 E333 0T 01      STA    B    SUBHI      STORE THE RESULT
00436 E335 2A 03      BPL    L3
00437 E33T T3 0C04    CDH    NEG            SET NEG TO FF IF THE RESULT IS NEGATIVE
00438 E33A 32          PUL    A
00439 E338 9T 02      STA    A    MINLD
00440 E33D 32          PUL    A
00441 E33E 9T 03      STA    A    MINHI      RETRIEVE THE ORIGINAL MINUEND
00442 E340 39          XSUB16 RTS
00443 E400              ORG    $E400
00444
00445
00446
00447
00448
00449
*****
*
* ROUTINE TO MONITOR V/F CARD FOR SPECIFIED TIME AND READ IT
* BACK, AND SEND THE RESULT BACK TO THE NORTH STAR
*
*****
00450 E400 7F 0016    TIMER CLR    TYPE
00451 E403 80 E18F    JSR    GETDNE      GET THE NEXT CHAR FROM NS WHEN SENT
00452 E406 80 30      SUB    A    #330    GET THE HEX EQUIVALENT
00453 E408 48          ASL    A            AND MULTIPLY IT BY 2
00454 E409 CE E6EE    LDX    #CA00-2     GET INTO X ADDRESS TO 1ST COUNT MINUS 2
00455 E40C 0F 14      STX    TEMP
00456 E40E 9B 15      AOD    A    TEMP+1
00457 E410 9T 15      STA    A    TEMP+1
00458
*
00459 E412 80 E18F    JSR    GETDNE      GET THE NEXT CHAR FROM NS WHEN SENT
00460 E415 B1 30      CMP    A    #330    CHECK IF IT EQUALS ZERO
00461 E417 2T 04      BEQ    ARND
00462 E419 B6 20      LDA    A    #320
00463 E41B 9T 16      STA    A    TYPE
00464 E410 CE 60F0    ARND  LDX    #560F0  GET THE CONTRDL WRDR
00465 E420 0F 0E      STX    VALL
00466 E422 8D E168    JSR    STDHP       SEND VALUE IN VALL & VALH TO HP
00467
*
00468 E425 B6 00      LDA    A    #300    OD = 13 = SLDT ADDRESS FOR V/F CARD
00469 E42T 9T 0F      STA    A    VALH
00470 E429 B6 05      LDA    A    #305
00471 E42B 8D E4T6    JSR    SEND        SEND A 05 TO V/F CARD
00472
*
00473 E42E B6 01      LDA    A    #301    SEND A 01 TO V/F CARD
00474 E430 8D E4T6    JSR    SEND
00475
*
00476 E433 B6 09      LDA    A    #309    SEND A 09 TO V/F CARD
00477 E435 8D E4T6    JSR    SEND
00478
*
00479 E43B 0E 14      LOX    TEMP        GET THE POINTER TO THE COUNT
00480 E43A EE 00      LDX    0,X         GET THE CORRESPONDING COUNT IN X
00481 E43C 8D E4TE    JSR    TIMOUT
00482
*
00483 E43F C6 05      LDA    B    #305    7060 COUNTER TO BE READ 5 TIMES

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00484 E441 07 14          STA B TEMP
00485
00486 E443 86 00  AGA IN  LOA A #800  SEND A 13 TO V/F CARD
00487 E445 80 E4T6      JSR  SEND
00488
00489 E448 86 0F          LCA A #80F
00490 E44A 80 E4T6      JSR  SEND
00491
00492 E440 CE ACF0      LOX  #8A0F0  SEND ISL AND SYE FOR READING
00493 E450 0F 0E        STX  VALL
00494 E452 80 E16B      JSR  STOMP  SEND VALUE IN VALL & VALH TO HP
00495
00496 E455 CE 00D0      LOX  #80000  TC TRIGGER INPUT FROM V/F CARD
00497 E45B 0F 0E        STX  VALL
00498 E45A 80 E16B      JSR  STOMP  SEND A 9 TO THE V/F SLOT
00499
00500 E450 80 E466      JSR  READ
00501 E460 7A 00L4      OEC  TEMP
00502 E463 26 0E        BNE  AGAIN
00503 E465 39          XTIMER RTS
00504
00505
00506          *
00507          *
00508          *
00509          *
00510          *
00511          *
00512          *
00513          *
00514          *
00515          *
00516          *
00517          *
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00537          *
00538          *

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```

00509 E466 80 E2D0  READ  JSR  RONSNO  TIMEOUT, READ HP AND SEND TO NSTAR
00510 E469 CE 6CF0      LOX  #8A0F0
00511 E46C 0F 0E        STX  VALL
00512 E46E 80 E16B      JSR  STOMP  SEND VALUE IN VALL & VALH TO HP
00513 E471 86 0C        LOA A #800
00514 E473 9T 0F        STA A VALH
00515 E475 39          XREAD RTS

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00518          *
00519          *
00520          *
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00522          *
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00536          *
00537          *
00538          *

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```

00521 E4T6 9A 16      SEND  ORA A TYPE
00522 E4T8 97 0E      STA A VALL
00523 E4TA 8D E16B      JSR  STOMP  SEND A 5 TO THE V/F SLOT
00524 E4T0 39          XSEND RTS

```

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00526          *
00527          *
00528          *
00529          *
00530          *
00531          *
00532          *
00533          *
00534          *
00535          *
00536          *
00537          *
00538          *

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00530 E47E 86 5D      TIMEOUT LOA A #85D
00531 E480 4A          BACK  DEC A
00532 E481 26 F0      BNE  BACK
00533 E483 09          DEX
00534 E484 26 F8      BNE  TIMEOUT
00535 E486 39          XTINQU RTS

```

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00536
00537
00538

```

```

00539      *
00540      *-----*
00541      *
00542      * RELAY SUBROUTINE
00543      * THIS ROUTINE WILL ESTABLISH(WRITE) --OR-- READ BACK RELAY POSITIONS
00544      * TO WRITE :- IT ACCEPTS NEW POSITION FROM N.STAR ,CONVERTS TO
00545      * BINARY, AND SENDS IT OUT ON SLOT A (0001) OF HP INPUT PORT
00546      * TO READ:- FETCHES CONTENTS OF SLOT A OF HP AND SENDS IT TO NS
00547      *-----*
00548      *
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00594	E613	96	10		LOA	A	CP		
00595	E615	26	03		BNE		ARDUN	IF NOTHING SENT THEN ASSUME	
00596	E617	7C	0C19		INC		FFLAG	NO CHANGE IN FREQUENCY OF VCO	
0C597	E61A	96	0E	ARDUN	LOA	A	VALL		
00598	E61C	97	17		STA	A	TVALL	SAVE IT FOR NOW AND	
00599	E61E	80	E10B	GETAMP	JSR		NSREAD	GET THE AMPLITUDE FROM NORTH STAR	
00600	E621	96	10		LOA	A	CP	IF NOTHING SENT THEN	
00601	E623	27	1F		BEQ		NOCHNG	DO NOT CHANGE THE AMPLITUDE OF VCO OUTPUT	
00602	E625	06	0E		LOA	B	VALL	SAVE ONLY LAST 8 BITS OF AMPLITUDE	
00603	E627	37			PSH	B		ON STACK	
00604	E628	CE	60F0		LOX		#60F0	SEND SYE AND OTE TO HP	
0C605	E628	DF	0E		STX		VALL		
00606	E620	80	E168		JSR		STOHP	SEND VALUE IN VALL & VALH TO HP	
00607				*					
00608	E630	33			PUL	B		GET AMPLITUDE IN VALL	
00609	E631	02			NOP			MAY REQUIRE A COM B HERE	
0C610	E632	07	0E		STA	B	VALL		
00611	E634	96	16		LOA	A	TYPE	GET SLOT ADDRESS WITH RANGE	
0C612	E636	8A	01		DRA	A	#S01	MAKE WR(AMP) LOW	
00613	E638	57	0F		STA	A	VALH		
00614	E63A	80	E168		JSR		STOHP	SEND VALUE IN VALL & VALH TO HP	
0C615				*					
00616	E630	96	16		LOA	A	TYPE		
0C617	E63F	97	0F		STA	A	VALH	HOLD AMPLITUDE DATA	
00618	E641	80	E168		JSR		STOHP	SEND VALUE IN VALL & VALH TO HP	
0C619				*					
0C620	E644	96	19	NOCHNG	LOA	A	FFLAG		
00621	E646	26	10		BNE		XVCD	IF NO CHANGE IN FREQ THEN QUIT	
00622	E648	CE	60F0		LOX		#60F0		
00623	E648	0F	0E		STX		VALL		
00624	E640	80	E168		JSR		STOHP	SEND VALUE IN VALL & VALH TO HP	
0C625				*					
00626	E650	96	17		LOA	A	TVALL	GET THE FREQUENCY MAGNITUDE	
0C627	E652	43			COM	A			
00628	E653	97	CE		STA	A	VALL		
00629	E655	96	16		LOA	A	TYPE	GET SLOT ADDRESS AND RANGE	
00630	E657	8A	02		DRA	A	#S02	MAKE WR(FRD) LOW	
00631	E659	97	0F		STA	A	VALH		
00632	E658	80	E168		JSR		STOHP	SEND VALUE IN VALL & VALH TO HP	
0C633				*					
00634	E65E	96	16		LOA	A	TYPE		
00635	E660	97	0F		STA	A	VALH	HOLD FREQUENCY DATA	
00636	E662	80	E168		JSR		STOHP	SEND VALUE IN VALL & VALH TO HP	
00637	E665	39		XVCD	RTS				
00638	E6F0				ORG		#E6F0		
00639	E6F0	0288		CADD	FOB		\$0288		
0C640	E6F2	1958			FOB		\$1958		
00641	E6F4	F070			FCB		#F070		
00642	ETFE				ORG		SETFE		
0C643	ETFE	E0			FCB		SEQ,300		
	ETFF	0D							
0C644					END				

SYMBOL TABLE

STAKPT	0050	PTK1	A000	PTM8	A004	PTM2	A001	IN0L	0000	IN0H	00E0	MEH	0011	MEH1	C012	CRA1	2001
CRUI	2003	DURAI	2000	01M81	2002	CR42	4001	LR92	4003	00RA2	4000	00RB2	4002	ACIAS	8000	ACIAC	8000
ACTAKE	0001	ACTAT	0001	CP	0010	VALL	000E	VALM	000F	TMPL	000C	TMPH	0000	KNT	0000	MARK	000A
L303	0006	L302	0007	L301	0008	L50	0009	SUBH	0001	SUBL0	0000	M1NH1	0003	M1HLO	0002	NEG	000A
CARRY	0005	TEMP	0014	TYPE	0016	TVALL	0017	TVALH	0018	FFLAG	0019	LOOKCO	E06F	CRDP	E078	KALB	E084
CAREL	E080	CKVCO	E096	CAOC	E09F	00AT	E100	X00AT	E10A	MSREAD	E108	LOOKC2	E11C	CHECK	E128	NEXT	E12F
KNIPUS	E141	X10	E142	XMSR0	E16A	ST01P	E168	LOOKK3	E17F	XST0HP	E18E	GETONE	E18F	XGET0N	E198	A0C	E198
ANEXT	E1AB	AAGA IN	E1C2	XAOC	E1E5	R0NSH0	E200	LOOKK1	E209	XR0NSH	E215	IOAT	E216	MSEMD	E224	INC	E228
POST	E239	HEGR1	E247	P052	E258	NEUR2	E269	P053	E27C	MEGR3	E28A	ASC11	E292	NEXT1	E2A3	LOCKC4	E2A9
X10AT	E289	ZC00	E20A	NOT	E2C4	PATCH	E2CA	SUR16	E300	L1	E315	L2	E328	L3	E33A	X5UR16	E340
TIMER	E400	ARND	E410	AGAIN	E443	X11HER	E445	REAU	E466	VC0	E475	SENO	E476	XSENO	E47D	TIMOUT	E47E
BACK	E480	XTIMOU	E486	RELAY	E500	MOREL	E52T	XNEL	E53A	VC0	E600	GETFRQ	E610	AROUN	E61A	GETAMP	E61E
NCKING	E644	XVCO	E665	CA00	E6F0												

MODIFICATIONS MADE IN THE EXISTING EPROM IN MC 6802 INTERFACE

The following modifications were made in the existing assembly program in the MC 6802 interface EPROM. The modifications were made in the ADC assembly subroutine so that the ADC could be triggered by an external signal and sample be taken. The memory address at which the modifications were made are clearly indicated. The original listing of the assembly is also shown. The new EPROM contents are also shown.

MEMORY LOCATION E1 A8

JMP MODPRMST ! JUMP TO THE MODIFIED PROGRAM ADDRESS

MEMORY LOCATION E7 10

ORG MODPRMST ! START OF THE MODIFIED PROGRAM AT E7 10

LDX #*6000

SEI !SET INTERRUPT MASK

LDA A CRB2

ORA A #*01 ! SET BIT0 TO ENABLE INTERRUPT

STA A CRB2

LDA A DDRB2 ! CLEAR BIT7 OF CONTROL REG.

CLI ! CLEAR INTERRUPT MASK

WAI ! WAIT FOR INTERRUPT

SEI ! SET INTERRUPT MASK

JMP RTNADDR ! GO BACK TO THE ORIGINAL PRGM.

RTNADDR IS E1 A8 ! CONTINUE ORIGINAL PRGM. AT

! RETURN ADDRESS E1 A8

As soon as an interrupt occurs the microprocessor fetches the interrupt service vector and this is initialized at location E7 F8. The contents of this location are E5 50 and the interrupt

service routine starts at this location ie. E5 50.

INTERRUPT SERVICE ROUTINE STARTS AT E5 50

```
ORG   INROUTIN      : START OF INTERRUPT ROUTINE(E5 50)
      NOP            : NO OPERATION
      LDA A         DDRB2 : CLEAR BIT7 OF CONTROL REG.
      LDA A         CRB2  : LOAD ACC. WITH (CONTROL REG.B.)
      AND A         #0FE  : CLEAR BIT0 TO DISABLE INTERRUPTS
      STA A         CRB2  :
      RTI           : RETURN FROM THE INTERRUPT ROUTINE
```

Thus these modifications made in the existing ADC subroutine enabled the use of a trigger reference signal to initiate sampling. The modified EPROM contents are shown on the next page.

CONTENTS OF THE MODIFIED EDITION

000	00	5P	20	17	00	05	00	02	8E	00	20	86	00	87	20	01
010	87	20	03	86	FF	87	20	00	87	20	03	86	04	87	20	03
020	86	3C	87	20	01	86	00	87	40	01	87	40	03	87	40	00
030	87	40	02	86	04	87	40	03	86	3C	87	40	01	86	02	07
340	80	00	86	21	87	80	00	26	40	02	84	30	44	44	44	08
050	00	97	12	86	20	89	00	97	11	DE	11	EE	00	86	53	87
060	80	01	86	00	87	80	00	FF	80	04	86	83	87	80	01	80
070	E1	0F	81	26	26	05	80	E1	00	20	F4	81	3F	25	05	80
080	E2	00	20	E8	81	23	26	05	80	E4	00	20	E2	81	25	26
090	05	00	85	00	20	89	81	21	26	05	20	86	00	20	00	81
3A0	41	56	CC	8D	E1	89	20	C7	02	FF	FF	FF	FF	FF	FF	FF
0B0	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
2C0	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
3D0	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
0E0	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
0F0	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
100	80	E1	09	96	10	27	03	80	E1	60	39	86	00	97	10	97
110	0E	97	0F	97	0C	97	0D	97	05	4C	97	0A	8D	E1	8F	81
120	E4	27	04	80	30	36	7C	08	13	20	F1	96	10	27	33	32
130	D6	08	25	0D	97	0E	7C	00	08	96	10	91	09	27	28	20
140	EE	5F	0C	40	89	97	0C	07	8D	48	59	48	59	0C	98	0C
150	09	0D	7A	00	08	E5	E8	08	0E	D9	0F	97	35	D7	0F	7C
160	00	0A	96	0A	97	09	91	10	26	05	39	96	0E	87	20	00
170	96	0F	87	20	02	02	02	86	40	01	84	F7	87	40	01	86
180	40	02	84	40	27	F9	86	40	01	8A	08	87	40	01	39	86
190	80	00	47	24	FA	86	80	81	39	FF	FF	CE	80	F0	DF	0E
1A0	8D	E1	6B	CE	00	50	DF	0E	7E	E7	10	8D	E1	6B	86	40
1B0	00	07	00	08	86	40	0E	87	00	08	8C	64	00	26	8C	CE
1C0	62	00	A6	00	43	97	00	00	A6	00	43	84	0F	97	01	00
1D0	DF	15	CE	00	3F	8D	E4	7E	8D	E2	04	8D	E2	09	DE	15
1E0	40	02	7E	E1	80	FF	FF	FF	60	00	86	40	03	2A	FB	86
200	CE	00	3A	8D	E4	7E	80	E2	16	86	80	00	46	46	24	F9
210	8E	05	87	80	81	39	86	40	00	43	97	00	86	40	02	43
220	84	0F	97	01	CE	00	06	85	03	05	20	E7	00	00	4A	2A
230	FA	86	E3	97	32	86	03	97	03	8D	E3	20	96	04	81	00
240	26	05	7C	02	26	20	F2	26	00	88	E8	97	30	26	31	89
250	03	97	01	86	64	97	02	86	30	97	03	8D	73	00	86	34
260	81	00	26	05	7C	00	07	20	F2	36	64	20	00	97	00	86
270	01	89	00	97	01	86	0A	97	02	7F	02	02	30	E3	00	95
280	04	81	00	25	05	7C	00	00	20	F2	36	02	00	3A	97	09
290	C6	34	09	86	30	80	00	87	00	5A	36	76	CE	00	05	C5
300	84	07	06	06	00	81	30	37	11	76	02	02	56	56	24	F9
310	87	D0	01	20	7A	00	05	36	11	39	D6	08	C1	01	E6	04
320	85	20	20	E3	7A	00	00	08	20	D9	A6	00	20	D9	0F	0F
330	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
340	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
350	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
360	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
370	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
380	96	33	36	96	02	36	7F	00	05	7F	00	04	43	0C	33	01
310	24	03	7C	00	05	06	03	53	08	05	97	02	07	03	7F	00
320	05	0C	96	00	08	02	24	33	7C	00	05	D6	31	D8	03	D8
330	05	97	00	D7	81	2A	03	73	00	04	32	97	02	32	97	03
340	39	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
350	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
360	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
370	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
380	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF
390	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF	FF

3C3 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
1L3 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
1E3 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
3F3 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
430 7F 00 16 80 E1 87 83 30 48 CE 56 EE DF 14 9B 15
410 97 15 8D E1 87 81 38 27 94 35 28 97 16 CE 63 F3
429 DF 0E 8D E1 68 86 D8 97 0F 86 05 8D 24 76 86 31
438 8D E4 76 86 09 8D 24 76 DE 14 EE 08 8D 24 7E C6
443 05 D7 14 86 0D 8D 24 76 86 0F 8D E4 76 CE A0 F3
458 DF 0E 8D E1 68 CE 00 D8 DF 3E 8D E1 68 8D 24 66
460 7A 00 14 26 DE J9 8D E2 00 CE 60 F8 DF 0E 8D E1
478 68 86 D8 97 0F J9 9A 16 97 4E 8D E1 68 J9 86 5D
488 4A 26 FD 09 26 F8 J9 FF FF FF FF FF FF FF FF FF FF
498 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
4A8 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
4B8 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
4C2 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
4D8 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
4E3 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
4F3 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
508 8D E1 8F 81 57 26 28 8D E1 08 DE 0E DF 17 CE 78
518 F3 DF 0E 8D E1 68 DE 17 DF 0E 96 8F 84 0F 8A 10
528 97 0F 8D E1 68 28 13 CE 48 F8 DF 0E 8D E1 68 CE
538 08 10 DF 0E 8D E1 68 8D E3 30 J9 FF FF FF FF FF
548 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
553 81 86 48 82 86 48 83 84 FE 87 48 83 38 FF FF FF FF
568 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
578 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
588 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
598 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
5A8 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
5B8 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
5C3 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
5D8 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
5E3 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
5F3 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
603 8D E1 88 96 8E 84 83 48 48 8A 88 97 16 7F 88 19
618 8D E1 88 96 18 26 83 7C 88 19 96 8E 97 17 3D E1
628 88 96 18 27 1F D6 8E 37 CE 60 F8 DF 8E 8D E1 68
638 33 8E D7 8E 96 16 8A 81 97 0F 8D E1 68 96 16 97
643 8F 8D E1 68 96 19 26 1D CE 60 F8 DF 8E 8D E1 68
658 96 17 43 97 8E 96 16 8A 8E 97 0F 8D E1 68 96 16
663 97 0F 8D E1 68 39 FF FF FF FF FF FF FF FF FF FF
678 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
688 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
698 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
708 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
718 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
728 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
738 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
748 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
758 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
768 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
778 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
788 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
798 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
808 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
818 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
828 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
838 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
848 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
858 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
868 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
878 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
888 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
898 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
908 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
918 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
928 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
938 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
948 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
958 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
968 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
978 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
988 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
998 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF
008 FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF FF

APPENDIX 4

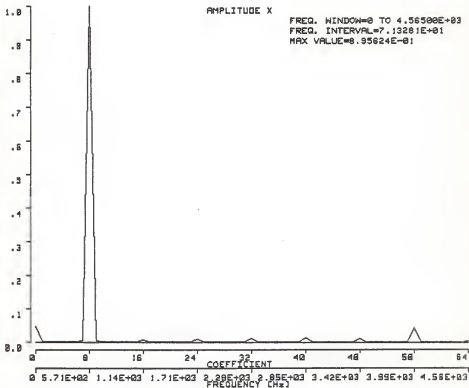
TABLE SHOWING THE READINGS OBTAINED FOR THE ADAPTIVE METHOD

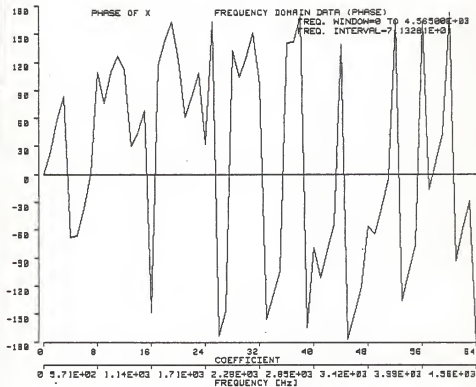
FREQUENCY	CALCULATED PHASE	ACTUAL PHASE	ERROR
F (Hz.)	P (Deg)	(Deg)	(Deg)
200	60.042437	60.78165	0.739213
300	50.935272	50.013108	0.922164
400	41.280897	41.807545	0.526648
500	34.611122	35.58437	0.973248
600	30.582739	30.807028	0.224289
700	26.862562	27.072203	0.209641
800	23.599043	24.095689	0.496646
900	21.049831	21.679956	0.630125
1000	19.271924	19.686781	0.414857
2000	10.005898	10.142929	0.137031

The actual phase was obtained from the RC circuit used. The values of R and C were found using a digital bridge. (Genrad RLC Digibridge). $R=101.15K$ and $C=4.394$ nF.

APPENDIX 5

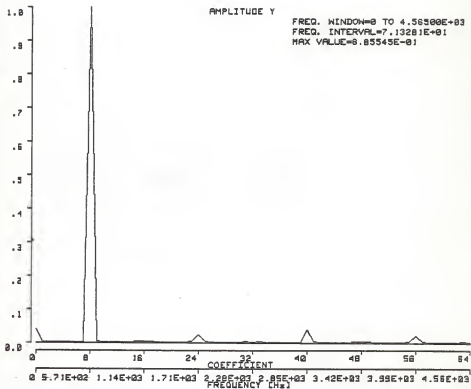
Plots obtained for the FFT method with actual data samples.

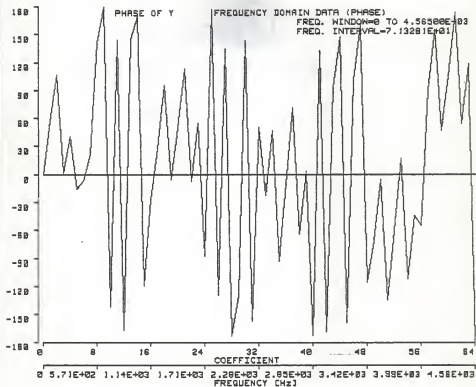




AMPLITUDE Y

FREQ. WINDOW=0 TO 4.56500E+03
FREQ. INTERVAL=7.13281E+01
MAX VALUE=8.85545E-01





Printout of the results obtained for X

FREQUENCY DOMAIN DATA

FREQUENCY WINDOW=0 TO 4.56500E+03[Hz]
 FREQUENCY INTERVAL= 7.13281E-01[Hz]

COEFF. DC TERM MAX FREQ.	FREQUENCY[Hz]	REAL	IMAG	MAGNITUDE	PHASE[DEG]
1	0.00000E+00	3.49154E-02			
1	4.56500E+03	-1.76563E-03			
1	7.13281E+01	1.32235E-03	1.87977E-03	2.24219E-03	56.76
2	1.42656E+02	-1.99268E-04	6.49395E-04	6.79280E-04	107.06
3	2.13984E+02	9.24246E-04	2.80697E-05	8.24724E-04	1.95
4	2.85320E+02	9.35504E-04	7.07742E-04	1.09499E-03	48.27
5	3.56641E+02	3.83316E-04	-1.68288E-04	3.95834E-04	-15.69
6	4.27969E+02	1.57744E-03	-1.79668E-04	1.58684E-03	-6.17
7	4.99297E+02	1.86564E-03	7.39181E-04	2.00618E-03	21.62
8	5.70625E+02	-6.86691E-01	5.59146E-01	9.85542E-01	140.05
9	6.41953E+02	-3.81833E-03	7.48294E-05	3.81966E-03	178.88
10	7.13281E+02	-4.75131E-04	-3.71329E-04	6.82969E-04	-142.80
11	7.84609E+02	-9.93946E-04	7.23897E-04	1.22962E-03	143.93
12	8.55947E+02	-1.04874E-03	-2.31644E-04	1.62720E-03	-168.97
13	9.27284E+02	-6.68485E-05	4.84291E-05	8.14089E-05	145.27
14	9.98622E+02	-1.28348E-03	2.54455E-04	1.31862E-03	168.79
15	1.06992E+03	-1.11979E-03	-1.99692E-03	2.20946E-03	-119.28
16	1.14125E+03	2.72667E-03	-1.35389E-03	3.04430E-03	-26.41
17	1.21258E+03	2.85029E-03	1.86402E-03	2.31106E-03	27.48
18	1.28391E+03	-1.17571E-04	1.20188E-03	1.28674E-03	95.59
19	1.35523E+03	1.38866E-04	-1.42189E-05	1.39591E-04	-5.84
20	1.42656E+03	6.49408E-04	7.67846E-04	1.09204E-03	49.75
21	1.49789E+03	-2.58167E-04	5.99686E-04	6.53863E-04	113.29
22	1.56922E+03	7.86887E-04	-1.93435E-04	7.93856E-04	-7.49
23	1.64055E+03	1.45348E-03	2.82629E-03	2.49267E-03	54.35
24	1.71187E+03	9.20344E-04	-2.20526E-02	3.26718E-02	-87.61
25	1.78320E+03	-2.54125E-03	1.66112E-04	2.54667E-03	174.26
26	1.85453E+03	-6.29228E-04	-9.85747E-04	1.08646E-03	-128.69
27	1.92586E+03	-3.97233E-04	4.46247E-04	5.68182E-04	134.36
28	1.99719E+03	-1.16891E-03	-1.37416E-04	1.11739E-03	-172.94
29	2.06852E+03	-2.53258E-02	-3.18299E-04	4.88538E-04	-129.32
30	2.13984E+03	-9.11379E-04	6.77736E-04	1.35533E-03	143.38
31	2.21117E+03	-2.33126E-03	-9.72983E-04	2.52616E-03	-157.32
32	2.28250E+03	4.21875E-04	5.08881E-04	6.54281E-04	49.84
33	2.35383E+03	2.09628E-03	-0.86866E-04	2.27609E-03	-22.93
34	2.42516E+03	6.83843E-04	7.26162E-04	9.96902E-04	46.75
35	2.49648E+03	-1.73746E-05	-3.32178E-04	3.32624E-04	-92.99
36	2.56781E+03	8.91854E-04	-1.53992E-04	9.85815E-04	-9.86
37	2.63914E+03	1.23964E-04	3.61867E-04	3.82281E-04	71.22
38	2.71047E+03	3.98217E-04	-8.35888E-04	9.25691E-04	-64.52
39	2.78180E+03	2.24207E-03	1.20244E-04	2.24529E-03	3.87
40	2.85312E+03	-3.34958E-02	-4.32263E-03	3.37736E-02	-172.65
41	2.92445E+03	-1.74962E-03	1.90004E-03	2.58289E-03	132.64
42	2.99578E+03	-1.64267E-03	-2.68757E-04	1.66337E-03	-168.68
43	3.06711E+03	-3.46944E-05	5.16844E-04	5.17239E-04	93.85
44	3.13844E+03	-9.46894E-04	6.26212E-04	1.35533E-03	146.32
45	3.20977E+03	-6.39817E-04	-1.79258E-04	4.71881E-04	-158.78
46	3.28109E+03	-2.86128E-04	1.67738E-03	1.69492E-03	100.83
47	3.35242E+03	-2.41989E-03	8.77726E-04	2.57412E-03	166.86
48	3.42375E+03	-1.64852E-03	-3.46327E-03	3.83562E-03	-115.45
49	3.49508E+03	7.50394E-04	-2.26688E-03	3.38778E-03	-71.68
50	3.56641E+03	9.92867E-04	-8.35841E-05	8.96771E-04	-5.35
51	3.63773E+03	-3.86887E-04	-3.18747E-04	4.41867E-04	-134.21
52	3.70906E+03	4.51612E-04	-8.13781E-04	9.30692E-04	-68.97
53	3.78039E+03	2.98333E-04	8.68841E-05	3.03837E-04	186.85
54	3.85172E+03	-3.79183E-04	-6.14464E-04	8.86486E-04	-112.04
55	3.92305E+03	1.64882E-03	-1.59644E-03	2.28873E-03	-44.62
56	3.99437E+03	1.08919E-02	-1.57174E-02	1.91282E-02	-85.28
57	4.06570E+03	2.81785E-05	2.39354E-03	2.39362E-03	89.52
58	4.13703E+03	-1.52158E-03	4.86188E-04	1.13129E-03	154.55
59	4.20836E+03	2.72917E-04	2.89785E-04	3.98862E-04	46.72
60	4.27969E+03	-2.71833E-04	9.94925E-04	1.03815E-03	165.81
61	4.35102E+03	-4.95022E-04	6.26499E-05	4.92977E-04	172.78
62	4.42234E+03	5.43882E-04	7.53391E-04	9.25681E-04	54.22
63	4.49367E+03	-1.13886E-03	2.14277E-03	2.42287E-03	117.82

Printout of the results obtained for Y

FREQUENCY DOMAIN DATA

FREQUENCY WINDOW=0 TO 4.56500E+03(Hz)
 FREQUENCY INTERVAL= 7.13281E-01(Hz)

COEFF. DC TERM MAX FREQ.	FREQUENCY(Hz)	REAL	IMAG	MAGNITUDE	PHASE(DEG)
1	0.59980E+03	4.15313E-02			
2	4.56500E+03	-3.23430E-03			
1	7.13281E+01	1.66399E-03	7.46356E-04	1.82126E-03	33.99
2	1.42556E+02	6.59789E-04	1.65818E-03	1.24643E-03	58.39
3	2.13784E+02	7.18324E-05	6.28126E-04	6.32208E-04	83.48
4	2.85312E+02	1.49137E-04	-3.64704E-04	3.74421E-04	-47.76
5	3.56641E+02	5.77635E-04	-1.37463E-03	1.39806E-03	-45.61
6	4.27969E+02	1.25536E-03	-9.23592E-04	1.57081E-03	-36.34
7	4.99297E+02	2.56984E-03	3.51885E-03	2.57008E-03	.78
8	5.70625E+02	-3.32829E-01	8.42874E-01	8.95624E-01	109.76
9	6.41953E+02	6.88751E-04	2.73284E-03	2.81635E-03	78.01
10	7.13281E+02	-7.42573E-04	2.53421E-03	2.14554E-03	111.95
11	7.84609E+02	-9.44134E-04	1.27948E-03	1.58288E-03	126.62
12	8.55937E+02	-1.88723E-04	4.74429E-04	5.10587E-04	111.69
13	9.27266E+02	5.96451E-04	3.41098E-04	6.81894E-04	30.81
14	9.98594E+02	9.96977E-04	9.87556E-04	1.40329E-03	44.73
15	1.06992E+03	6.92318E-04	1.77616E-03	1.98632E-03	68.78
16	1.14125E+03	-5.88628E-03	-3.12894E-03	5.89856E-03	-147.96
17	1.21258E+03	-9.44978E-04	1.86353E-03	2.09433E-03	118.94
18	1.28391E+03	-1.27492E-03	9.68528E-04	1.59625E-03	143.01
19	1.35523E+03	-8.76743E-04	2.72215E-04	9.18931E-04	162.75
20	1.42656E+03	-1.11337E-04	1.76192E-04	2.38423E-04	123.29
21	1.49789E+03	3.89231E-04	7.85359E-04	8.85617E-04	61.11
22	1.56922E+03	1.64240E-04	1.54459E-03	1.55350E-03	83.93
23	1.64055E+03	-6.68238E-04	1.93674E-03	2.04618E-03	166.82
24	1.71187E+03	5.96611E-03	3.72849E-03	7.83026E-03	32.03
25	1.78320E+03	-1.88104E-03	5.92221E-04	1.97208E-03	162.52
26	1.85453E+03	-1.47615E-03	-1.79863E-04	1.48697E-03	-173.08
27	1.92586E+03	-6.41666E-04	-4.35621E-04	7.75549E-04	-145.83
28	1.99719E+03	-8.13537E-05	9.68938E-05	1.21328E-04	132.11
29	2.06852E+03	-2.87432E-04	8.15659E-04	8.41622E-04	184.27
30	2.13984E+03	-8.77118E-04	1.23319E-03	1.51331E-03	125.42
31	2.21117E+03	-1.62918E-03	8.88379E-04	1.85658E-03	151.48
32	2.28250E+03	-1.17188E-03	1.16625E-02	1.11244E-02	96.85
33	2.35383E+03	-1.69537E-03	-7.92281E-04	1.87136E-03	-154.95
34	2.42516E+03	-8.8658E-04	-1.10888E-03	1.41335E-03	-128.84
35	2.49648E+03	-1.71617E-04	-6.82583E-04	7.03826E-04	-184.11
36	2.56781E+03	-5.72362E-05	4.88932E-05	7.48338E-05	148.01
37	2.63914E+03	-6.94632E-04	5.52688E-04	8.87681E-04	141.49
38	2.71047E+03	-1.49144E-03	3.19623E-04	1.52338E-03	167.99
39	2.78180E+03	-1.83952E-03	-5.16691E-04	1.91655E-03	-164.33
40	2.85312E+03	2.61638E-03	-1.28053E-02	1.38893E-02	-78.47
41	2.92445E+03	-6.33988E-04	-1.64877E-03	1.76643E-03	-111.03
42	2.99578E+03	2.19332E-03	1.32674E-03	1.24338E-03	-81.88
43	3.06711E+03	3.81745E-04	-5.38089E-04	7.94699E-05	-54.84
44	3.13844E+03	-5.97700E-05	5.22614E-05	6.59768E-04	-54.84
45	3.20977E+03	-8.56893E-04	4.13355E-05	8.57899E-04	-132.19
46	3.28110E+03	-1.24583E-03	-7.44279E-04	1.45122E-03	-149.15
47	3.35242E+03	-9.77023E-04	-1.59961E-03	1.87371E-03	-121.38
48	3.42375E+03	5.93778E-03	-8.84769E-03	1.06305E-02	-56.13
49	3.49508E+03	7.81589E-04	-1.62352E-03	1.81888E-03	-84.43
50	3.56641E+03	1.11629E-03	-7.82558E-04	1.36351E-03	-59.92
51	3.63773E+03	6.74851E-04	-5.20857E-05	8.75685E-04	-6.41
52	3.70906E+03	-6.69471E-05	1.97127E-05	6.97889E-05	163.59
53	3.78039E+03	-5.78859E-04	-5.75828E-04	8.15928E-04	-132.19
54	3.85172E+03	-3.69291E-04	-1.37333E-03	1.42214E-03	-185.05
55	3.92305E+03	4.36710E-04	-1.73545E-03	1.78956E-03	-75.88
56	3.99437E+03	-3.69348E-02	1.16931E-02	3.87414E-02	162.43
57	4.06570E+03	1.66736E-03	-5.81385E-04	1.74113E-03	-16.74
58	4.13703E+03	1.33533E-03	3.88478E-04	1.37658E-03	-43.18
59	4.20836E+03	5.25438E-04	4.93018E-04	7.20544E-04	43.78
60	4.27969E+03	-8.36732E-05	1.25637E-05	8.46112E-05	171.46
61	4.35102E+03	-3.47331E-05	-7.69768E-04	7.80491E-04	-92.58
62	4.42235E+03	7.51246E-04	-1.19657E-03	1.41286E-03	-57.88
63	4.49367E+03	1.56498E-03	-8.83121E-04	1.79696E-03	-29.44

APPENDIX 6

Correlation (CYX) results.

FREQUENCY WINDOW=0 TO 6.45000E+01(Hz)		FREQUENCY DOMAIN DATA			
FREQUENCY INTERVAL=1.00000E+00(Hz)		REAL	IMAG	MAGNITUDE	PHASE(DEG)
COEFF.	FREQUENCY(Hz)				
DC TERM	0.00000E+00	3.21394E-01			
MAX FREQ.	6.40000E+01	-2.10753E-11			
1	1.00000E+00	3.45436E-11	5.65294E-11	6.62403E-11	50.57
2	2.00000E+00	-9.86193E-12	3.50774E-11	3.61794E-11	104.10
3	3.00000E+00	-1.00304E-11	1.80753E-11	2.61033E-11	133.69
4	4.00000E+00	-1.25033E-11	1.33105E-11	1.62709E-11	137.63
5	5.00000E+00	-6.25535E-12	9.84500E-12	1.10112E-11	123.54
6	6.00000E+00	-1.22730E-11	-1.97721E-12	1.24313E-11	-170.05
7	7.00000E+00	9.09935E-12	5.40730E-12	1.13105E-11	29.00
8	8.00000E+00	-1.96406E-11	9.32094E-12	2.10267E-11	154.14
9	9.00000E+00	-9.64620E-12	-1.23714E-11	1.56630E-11	-127.03
10	1.00000E+01	-1.05432E-11	-1.05196E-11	2.13192E-11	-150.43
11	1.10000E+01	-1.99400E-12	-1.66749E-11	1.67033E-11	-75.32
12	1.20000E+01	1.26333E-13	2.00401E-11	2.00401E-11	89.74
13	1.30000E+01	-4.61009E-11	-3.06646E-11	6.01684E-11	-140.01
14	1.40000E+01	3.69434E-11	-5.35326E-11	6.66984E-11	-56.37
15	1.50000E+01	3.13103E-11	0.70624E-13	3.13227E-11	1.61
16	1.60000E+01	6.79600E-11	-1.07500E-11	7.05075E-11	-15.42
17	1.70000E+01	3.10930E-11	2.30250E-11	3.06906E-11	36.32
18	1.80000E+01	-3.43066E-13	2.00992E-11	2.00107E-11	70.62
19	1.90000E+01	-2.33639E-12	1.45091E-11	1.47720E-11	90.79
20	2.00000E+01	-1.63219E-11	1.36301E-11	1.80403E-11	123.93
21	2.10000E+01	-1.77134E-11	-1.00770E-11	2.03792E-11	-150.36
22	2.20000E+01	1.68419E-11	-1.73246E-12	1.69300E-11	-5.30
23	2.30000E+01	-1.42060E-11	3.02054E-11	3.34517E-11	115.13
24	2.40000E+01	-1.57220E-11	-1.50373E-11	2.23159E-11	-134.79
25	2.50000E+01	-6.24630E-14	5.21005E-12	5.21042E-12	90.69
26	2.60000E+01	-6.34926E-12	2.21069E-12	6.34930E-12	170.00
27	2.70000E+01	-9.70391E-12	7.94904E-12	1.25014E-11	141.05
28	2.80000E+01	-9.40631E-12	-1.00201E-11	1.43403E-11	-130.90
29	2.90000E+01	2.91247E-11	-1.04532E-11	3.09430E-11	-19.74
30	3.00000E+01	2.59014E-12	1.03971E-11	1.05797E-11	01.96
31	3.10000E+01	-1.00310E-11	1.29769E-11	1.69000E-11	120.05
32	3.20000E+01	-2.65625E-11	1.56250E-12	2.66004E-11	176.63
33	3.30000E+01	-1.59152E-11	-1.51612E-11	2.19000E-11	-136.39
34	3.40000E+01	-1.09900E-12	-1.62355E-11	1.63150E-11	-93.66
35	3.50000E+01	5.56400E-12	-1.23275E-11	1.30525E-11	-65.17
36	3.60000E+01	1.00151E-11	2.05356E-12	1.02399E-11	9.00
37	3.70000E+01	-4.63243E-12	1.73009E-11	1.77646E-11	103.12
38	3.80000E+01	-1.30042E-11	-5.62071E-13	1.32162E-11	-177.56
39	3.90000E+01	-1.35251E-11	-1.04930E-11	1.71194E-11	-142.20
40	4.00000E+01	1.02134E-11	-2.20374E-11	2.42092E-11	-65.13
41	4.10000E+01	6.21642E-12	2.37291E-11	2.45299E-11	70.32
42	4.20000E+01	-3.59422E-11	-6.11353E-12	3.64504E-11	-170.25
43	4.30000E+01	9.30641E-12	-6.60295E-11	6.74047E-11	-60.00
44	4.40000E+01	5.64577E-11	-6.00726E-11	7.46726E-11	-40.00
45	4.50000E+01	8.15542E-11	4.49011E-11	9.30970E-11	20.04
46	4.60000E+01	-9.35407E-12	6.54026E-11	6.60264E-11	97.00
47	4.70000E+01	0.73909E-12	-3.36321E-12	9.36466E-12	-21.05
48	4.80000E+01	4.90899E-01	4.35779E-02	5.00000E-01	5.00
49	4.90000E+01	-6.54945E-11	-2.27697E-11	6.93397E-11	-160.03
50	5.00000E+01	-2.12350E-11	-1.42064E-11	2.25413E-11	-146.21
51	5.10000E+01	-3.55531E-11	-2.90265E-12	3.56700E-11	-175.00
52	5.20000E+01	-4.43093E-11	-4.31932E-11	6.19350E-11	-135.70
53	5.30000E+01	1.94630E-11	-6.90197E-11	7.17113E-11	-74.25
54	5.40000E+01	3.00924E-11	-1.04234E-12	3.00963E-11	-1.54
55	5.50000E+01	4.55570E-12	-1.65003E-11	1.70205E-11	-74.64
56	5.60000E+01	2.02742E-11	-1.16793E-11	3.05914E-11	-32.44
57	5.70000E+01	2.02272E-11	-7.99003E-13	2.02430E-11	-2.26
58	5.80000E+01	2.00999E-11	1.10146E-11	3.11303E-11	32.30
59	5.90000E+01	1.12513E-11	2.39436E-11	2.60554E-11	64.03
60	6.00000E+01	1.97191E-12	2.42575E-11	2.43075E-11	05.35
61	6.10000E+01	-1.25795E-11	2.33479E-11	2.65194E-11	110.31
62	6.20000E+01	3.20107E-12	7.37544E-12	8.09060E-12	66.07
63	6.30000E+01	-1.33102E-11	3.60033E-11	3.84600E-11	110.25

APPENDIX 7

Program listing and results obtained for the Numerical Method.

```

10 | THIS PROGRAM DETERMINES THE PHASE BY SIMULATION OF TWO SIGNALS
30 | X(K) AND Y(K) BY FITTING A THIRD DEGREE POLYNOMIAL AT THE ZERO CROSSINGS
30 | OF THE SIGNALS AND USES THE NEWTON'S METHOD TO FIND THE ROOT AND THEN
40 | CALCULATES THE PHASE... SHOWS ALSO THE ERROR BETWEEN THE ACTUAL PHASE
50 | AND THE CALCULATED PHASE... SHOWS ALSO THE EFFECT OF VARYING THE RATIO
60 | OF SAMPLING FREQUENCY TO THE FREQUENCY OF THE SIGNALS... SHOWS THAT
70 | THE ERROR IS MINIMUM AT A RATIO OF 6.
80 | GENERATE Y1(K) AND Y2(K) THE TWO SIGNALS OF INTEREST
90 | Y2 LEADS Y1
100 DIM Y1(200),Y2(200),X(200),I(50),J(50)
110 | SAMPLING FREQ. IS F1 AND THE FREQUENCY OF THE SIGNALS IS F
120 | BY VARYING F1 WE CAN STUDY THE EFFECTS OF THE RATIO F1/F.
130 F=100
140 INPUT F1
150 PRINT "SAMP.FREQ.IS ",F1
160 PRINT "FREQ. OF SIGNAL=",F
170 FOR P=0 TO 90 STEP 10
180 | HERE P IS THE PHASE IN DEGREES
190 F0=2*(22/7)*F
200 T=1/F1
210 F0=F0*(22/7)/180
220 FOR K=1 TO 199
230 Y1(K)=SGN(F0*K*T)
240 Y2(K)=SGN(F0*K*T+P0)
250 NEXT K
260 L1=1
270 L2=1
280 P3=0
290 | DETERMINATION OF ZERO CROSSINGS
300 FOR K=1 TO 197
310 IF SGN(Y1(K))-SGN(Y1(K+1))=-2 THEN GOTO 330
320 GOTO 350
330 I(L1)=K
340 L1=L1+1
350 NEXT K
360 FOR K=1 TO 197
370 IF SGN(Y2(K))-SGN(Y2(K+1))=-2 THEN GOTO 390
380 GOTO 410
390 J(L2)=K
400 L2=L2+1
410 NEXT K
420 | FIND THE PHASE AT THE FIRST 10 ZERO CROSSINGS AND TAKE THE AVERAGE
430 | FIRST SET UP THE REQD. VALUES FOR THE DIFFERENCE TABLE
440 FOR N=1 TO 10
450 K=I(N)
460 X(K)=Y1(K)
470 X(K-1)=Y1(K-1)
480 X(K+1)=Y1(K+1)
490 X(K+2)=Y1(K+2)
500 | USE SUBROUTINE FOR FITTING THE POLYNOMIAL
510 GOSUB 650
520 | HERE T1 DENOTES THE ZERO CROSSING OF FIRST SIGNAL
530 T1=X0
540 | REPEAT FOR SECOND SIGNAL
550 K=J(N)
560 X(K)=Y2(K)
570 X(K-1)=Y2(K-1)
580 X(K+1)=Y2(K+1)
590 X(K+2)=Y2(K+2)
600 GOSUB 650

```

```

410   I T2 REPRESENTS ZERO CROSSING OF SECOND SIGNAL
420 T2=X0
430 GOTO 990
440   I SUBROUTINE FOR FITTING THE POLYNOMIAL USING THE DIFFERENCE TABLE
450 REM FORM DIFFERENCE TABLE
460 IF ABS(X(K))<.001 THEN X(K)=0
470 IF ABS(X(K+1))<.001 THEN X(K+1)=0
480 A0=X(K-1)
490 A1=X(K)
500 A2=X(K+1)
510 A3=X(K+2)
520 B0=A1-A0
530 B1=A2-A1
540 B2=A3-A2
550 C0=(B1-B0)/2
560 C1=(B2-B1)/2
570 D0=(C1-C0)/3
580 M1=F0+T
590 M2=M1*M1
600 M3=M2*M1
610 E0=A0-B0*(K-1)+M1+C0*(K-1)*M2-D0*(K-1)*(K+1)*M3
620 E1=B0-C0*(2*K-1)+M1+D0*(3*K*(K-1))*M2
630 E2=C0-D0*(3*K)*M1
640 E3=D0
650 REM THE ABOVE ARE THE COEFFS. OF POLYNOMIAL.
660 REM NEWTONS METHOD
670 DEF FNP(V)=E0+E1*V+E2*V*V+E3*V*V*V
680 DEF FNG(V)=E1+E2*V+E3*V*V
690 X0=M1*K+M1/2
700 X1=X0-FNP(X0)/FNG(X0)
710 IF ABS(FNP(X0))>1E-4(0 THEN 950
720 M1=M1+1
730 IF M1>10 THEN 770
740 GOTO 900
750 M1=0
760 RETURN
770   I OBTAIN THE DIFFERENCE BETWEEN THE ZERO CROSSINGS OF THE TWO SIGNALS
780   I AND CALCULATE THE PHASE
790 P1=T1-T2
1000 P2=P1/W1*360*F/F1
1010 P3=P3+P2
1020 NEXT N
1030 PRINT "AVERAGE OF THE PHASE VALUES IS",P3/(N-1)
1040 PRINT "ACTUAL PHASE IS",P,"ERROR IS ",P-P3/(N-1)
1050 PRINT " "
1060 P3=0
1070   I REPEAT FOR NEXT VALUE OF PHASE
1080 NEXT P
1090 END

```

SAMP. FREQ. IS	600		
FREQ. OF SIGNAL=	100		
AVERAGE OF THE PHASE VALUES IS	0	0	
ACTUAL PHASE IS	0	ERROR IS	0
AVERAGE OF THE PHASE VALUES IS	10	9.77897919153	
ACTUAL PHASE IS	10	ERROR IS	.2210208085
AVERAGE OF THE PHASE VALUES IS	20	19.3094805375	
ACTUAL PHASE IS	20	ERROR IS	.6905194625
AVERAGE OF THE PHASE VALUES IS	30	28.7222999216	
ACTUAL PHASE IS	30	ERROR IS	1.2777000784
AVERAGE OF THE PHASE VALUES IS	40	38.1637038994	
ACTUAL PHASE IS	40	ERROR IS	1.8362961006
AVERAGE OF THE PHASE VALUES IS	50	47.7834279363	
ACTUAL PHASE IS	50	ERROR IS	2.2165720637
AVERAGE OF THE PHASE VALUES IS	60	59.9999991558	
ACTUAL PHASE IS	60	ERROR IS	.0000008442
AVERAGE OF THE PHASE VALUES IS	70	69.7789819638	
ACTUAL PHASE IS	70	ERROR IS	.2210180362
AVERAGE OF THE PHASE VALUES IS	80	79.3094892334	
ACTUAL PHASE IS	80	ERROR IS	.6905107666
AVERAGE OF THE PHASE VALUES IS	90	88.7222917675	
ACTUAL PHASE IS	90	ERROR IS	1.2777082325

APPENDIX B

Listing of the program used for the Adaptive Predictor.

```

10  REM*****
20  REM THIS PROGRAM IS AN ADAPTIVE PREDICTOR WHICH ACTS AS A
30  REM SIGNAL ENHANCER.
40  REM THE INPUT SIGNAL IS STORED IN THE ARRAY X) WHICH
50  REM CONTAINS 500 SAMPLES OF THE INPUT SIGNAL X
60  REM THE ARRAY X1(*) IS USED TO GENERATE THE N PAST VALUES
70  REM OF THE INPUT AND THIS ACTS AS THE INPUT TO THE FILTER
80  REM THE ARRAY G(*) IS USED TO STORE THE OUTPUT OF THE
90  REM FILTER WHICH IS AN ENHANCED VERSION OF THE INPUT
100 REM SIGNAL - GETS RID OF THE NOISE IN INPUT SIGNAL
110 REM SINCE THE ADAPTIVE FILTER TAKES A FEW ITERATIONS TO
120 REM ADAPT ITSELF , THE OUTPUT G(*) MUST BE TAKEN AFTER A
130 REM FEW ITERATIONS FOR SUBSEQUENT PROCESSING.
140 REM*****
150 REM FORM THE DIFFERENT ARRAYS
160 REM*****
170 DIM X(500), X1(16), G(500), W(16)
180 REM*****
190 REM INITIALIZE FILTER COEFFICIENTS
200 REM*****
210 FOR I=1 TO 16
220   W(I)=0
230 NEXT I
240 REM*****
250 REM DEFINE THE NUMBER OF WEIGHTS OF THE FILTER TO BE
260 REM USED (N)
270 REM*****
280 N=10
290 REM*****
300 REM DEFINE THE CONVERGENCE PARAMETER V1
310 REM*****
320 V1=.002
330 REM*****
340 REM START PROCESSING
350 REM*****
360 FOR K=N TO 499
370 REM*****
380 REM COMPUTE INPUT SIGNAL ENERGY AND USE IT TO UPDATE
390 REM THE CONVERGENCE PARAMETER
400 REM*****
410 V2=V2*(1-V1)+V1*X (K)*X (K)
420 IF V2<.001 THEN 450
430 V=V1/V2
440 GOTO 470
450 V=0

```

```

460 REM*****
470 REM GENERATE X1 THE INPUT TO THE FILTER
480 REM*****
490 FOR J=1 TO N
500 M=J-1
510 X1(J)=X (K-M)
520 NEXT J
530 G0=0
540 REM*****
550 REM FIND THE OUTPUT OF THE FILTER G0
560 REM*****
570 FOR L= 1 TO N
580 G1=W(L)*X1(L)
590 G0=G0+G1
600 NEXT L
610 REM*****
620 REM COMP. ERROR
630 REM*****
640 E=X(K+1)-G0
650 REM*****
660 REM STORE THE OUTPUT OF THE FILTER IN G(*)
670 REM*****
680 G(K+1)=G0
690 FOR I=1 TO N
700 REM*****
710 REM UPDATE COEFFS. USING THE ERROR E
720 REM*****
730 W(I)=W(I)+V*E*X1(I)
740 NEXT I
750 NEXT K
760 REM*****
770 REM AFTER THE REQUIRED NUMBER OF ITERATIONS G(*)
780 REM CONTAINS THE ENHANCED OUTPUT... ALLOW ABOUT 100
790 REM ITERATIONS FOR THE ADAPTATION OF THE FILTER.
800 REM*****
810 RETURN

```

APPENDIX 9

Program listing for the PLOT package.

```

NAME PRD TYPE R10  REM THIS PROGRAM SHOWS HOW THE Plot SUBROUTINE CAN BE USED
20  REM TO PLOT THE REQUIRED DATA
30  REM HERE  RANDOM IS TO BE PLOTTED
40  OPTIMH BASE 1
50  DIM He(100),Ve(100)
60  RANDOMIZE
70  INPUT Hseinte
80  FOR I=1 TO 100
90  He(I)=I
100  Ve(I)=SIH(2*PI*10*I)
110  NEXT I
120  GOSUB Plot
130  END
140 Plot: REM  THIS IS A SUBROUTINE TO PLOT DATA IN THE ARRAYS
150  REM He AND Ve . THE PROCEDURE IS TO FILL THE ARRAYS AND THEM
160  REM CALL THE SUBROUTINE Plot
170  REM THE DATA MAY BE ON FLOPPY DISK OR OTHER STORAGE DEVICE
180  REM THIS DATA HAS TO BE ACCESSED AND USED TO FILL THE ABOVE
190  REM HENTIDHED ARRAYS
200  REM Find max and min values of the data in the arrays
210  Hemin=Vemin=-1E99
220  REM Hpoints IS THE NUMBER OF POINTS TO BE PLOTTED
230  FOR I=1 TO Hpoints
240  IF He(I)<Hemin THEN Hemin=He(I)
250  IF Ve(I)<Vemin THEN Vemin=Ve(I)
260  NEXT I
270  Hmax=Vmax=-1E99
280  FOR I=1 TO Hpoints
290  IF He(I)>Hmax THEN Hmax=He(I)
300  IF Ve(I)>Vmax THEN Vmax=Ve(I)
310  NEXT I
320  PRINT Hemin,Hmax,Vemin,Vmax
330  REM TO Find limits Hpr,Hol,Ver,Ved
340  GOTO 420
350  ! ACTUALLY A SUBROUTINE
360  REM THIS SUBROUTINE USES THE VALUE IN Value AND RETURNS A
370  REM NEW VALUE DEPENDIND WHICH OF THE LIMITS Hpr,Hol,Ver,Ved
380  REM IS TO BE DETERMINED
390  IF ABS(INT(Value))-ABS(Value)=0 THEN Return
400  IF ABS(Value)>1 THEN There
410  IF Value=1 THEN Return
420  GOTO 430
430  Decimal=1
440  Dplace=10Decimal
450  Dplace=1/Dplace
460  IF ABS(Value)>Dplace THEN Here
470  Decimal=Decimal*10
480  GOTO 440
490  There: Dplace=1
500  Here:  Mark=Dplace
510  !
520  IF Lin=1 THEN Getmax  *
530  IF Lin=0 THEN Getmin
540  Getmax: Maxad=Value DIV Dplace=1
550  IF Maxad=0 THEN Maxad=-2
560  Value=Maxad*Dplace
570  GOTO Return
580  Getmin: Minad=Value DIV Dplace=1
590  IF Minad=0 THEN Minad=-9
600  Value=Minad*Dplace

```



```

418 Return: RETURN
420 | Find the limits
430 | Her
440 Value=Hmax
450 Lim=1
460 GOSUB 350
470 Her=Value
480 | Hel
490 Value=Hmin
500 Lim=5
510 GOSUB 350
520 Hel=Value
530 | Vet
540 Value=Vmax
550 Lim=1
560 GOSUB 350
570 Vet=Value
580 | Veb
590 Value=Vmin
600 Lim=5
610 GOSUB 350
620 Veb=Value
630 REM THE TIC SPACINGS FOR DRAWING THE AXES ARE THEN FOUND
640 | FIND Xtic,Ytic ETC
650 Hetic=(Her-Hel)/20
660 Hnej=5
670 Vetic=(Vet-Veb)/20
680 Vnej=5
690 PRINT Hel,Her,Veb,Vet,Hetic,Vetic
700 REM THESE SUBROUTINES DETERMINE THE DIFFERENT PARAMETERS TO BE
710 REM USED IN FINDING INTERSECTIONS AND THE SPACINGS NEEDED FOR
720 REM LABELING THE AXES
730 IF (Hel(0) AND (Her(0) AND ((Veb(0) AND (Vet(0) THEN GOSUB One
740 IF (Hel(0) AND (Her(0) AND ((Veb(0) AND (Vet(0) THEN GOSUB Two
750 IF (Hel(0) AND (Her(0) AND ((Veb(0) AND (Vet(0) THEN GOSUB Three
760 IF (Hel(0) AND (Her(0) AND ((Veb(0) AND (Vet(0) THEN GOSUB Four
770 IF (Hel(0) AND (Her(0) AND ((Veb(0) AND (Vet(0) THEN GOSUB Five
780 IF (Hel(0) AND (Her(0) AND ((Veb(0) AND (Vet(0) THEN GOSUB Six
790 IF (Hel(0) AND (Her(0) AND ((Veb(0) AND (Vet(0) THEN GOSUB Seven
800 IF (Hel(0) AND (Her(0) AND ((Veb(0) AND (Vet(0) THEN GOSUB Eight
810 IF (Hel(0) AND (Her(0) AND ((Veb(0) AND (Vet(0) THEN GOSUB Nine
820 PRINT Xint,Yint
830 FOR I=5 TO 20 STEP 5
840 PRINT Hel+Hetic*I
850 PRINT Veb+Vetic*I
860 NEXT I
870 REM START PLOTTING OPERATIONS
880 PLOTTER IS 7,5,"9372A"
890 GRAPHICS
900 LIMIT 30,300,50,200
910 FRAME
920 LOCATE 25,145,25,90
930 FRAME
940 SCALE Hel,Her,Veb,Vet
950 AXES Hetic,Vetic,Xint,Yint,Hnej,Vnej,5
960 FRAME
970 REM ACTUAL PLOTTING OF DATA
980 FOR I=1 TO Hpoints
990 PLOT Hel(I),Ve(I)
1000 NEXT I

```

```

1210 GOTO 1566
1220 One: Xint=Nor
1230 Yint=Veb
1240 RETURN
1250 Two: Xint=0
1260 Yint=Veb
1270 RETURN
1280 Three: Xint=Hel
1290 Yint=Veb
1300 RETURN
1310 Four: Xint=Nor
1320 Yint=0
1330 RETURN
1340 Five: Xint=Yint=0
1350 RETURN
1360 Six: Xint=Hel
1370 Yint=0
1380 RETURN
1390 Seven: Xint=Nor
1400 Yint=Veb
1410 RETURN
1420 Eight: Xint=0
1430 Yint=Veb
1440 RETURN
1450 Nine: Xint=Hel
1460 Yint=Veb
1470 RETURN
1480 REM THE VALUES OF Sgnx AND Sgny DETERMINE THE RELATIVE MOVEMENTS
1490 REM OF THE PEN FOR LABELING THE AXES
1500 IF Hel=0 THEN Sgnx=-1
1510 IF Hel=0 THEN Sgny=1
1520 IF Veb=0 THEN Sgny=1
1530 IF Veb=0 THEN Sgny=-1
1540 IF (Xint=0) AND (Yint=0) THEN Sgnx=Sgny*-1
1550 REM LABEL THE X AXIS
1560 MOVE Hel,Veb
1570 FOR I=0 TO 20 STEP 5
1580 Xd=Hel-I*Snetic
1590 Yd=Yint+I*S*Vetic*Sgny
1600 MOVE Xd,Yd
1610 LABEL USING "HD.DBE";Xd
1620 NEXT I
1630 FOR I=0 TO 20 STEP 5
1640 Xd=Xint+I*S*Netic*Sgnx
1650 Yd=Veb+I*Vetic
1660 MOVE Xd,Yd
1670 LABEL USING "HD.DBE";Yd
1680 NEXT I
1690 MOVE Hel+S*Netic,Yint+4*Vetic*Sgny
1700 PRINT "X LABEL"
1710 INPUT $
1720 LABEL $
1730 REM LABEL THE Y AXIS; ROTATE MOTION OF PEN BY 90 DEGREES
1740 DEG
1750 LDIR 90
1760 MOVE Xint+3*S*Netic*Sgnx,Veb+S*Vetic
1770 PRINT "Y LABEL"
1780 INPUT $
1790 LABEL $
1800 LDIR 0

```



```

1810 RAD
1820 REM INITIALIZE PEN DIRECTION
1830 REM END OF PLOTTING OPERATION
1840 PEN 0
1850 RETURN

```

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ACKNOWLEDGEMENTS

I take this opportunity to express my sincere thanks to my major advisor Dr.M.S.P. Lucas. His great encouragement, views and stimulating discussions have contributed abundantly towards this entire project. His timely help and above all his remarkable sense of humor are deeply appreciated.

I also wish to express my sincere thanks to Dr.G.L. Johnson for his valuable support and suggestions and for extending his help in initiating me on the spinwriter which made the typing of this report a pleasure. Special thanks to Dr.P.M. Young for his support and suggestions.

I would like to express my gratitude to my parents who are a perpetual source of inspiration and encouragement.

APPLICATIONS OF SIGNAL PROCESSING TECHNIQUES IN
MICROPROCESSOR BASED INSTRUMENTATION

by

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AN ABSTRACT OF A MASTER'S THESIS

Submitted in partial fulfillment of the
requirements for the degree

MASTER OF SCIENCE

Department of Electrical Engineering

KANSAS STATE UNIVERSITY

Manhattan, Kansas

1984

ABSTRACT

The applications of signal processing techniques in microprocessor or computer based instrumentation were studied and a specific application in the measurement of phase was considered. The analyses were carried out in the discrete time (DT) domain using "samples" of the original signals. Algorithms based on Adaptive filtering, FFT, Correlation and Numerical methods have been developed. These methods were verified by using computer simulation and implemented on an automated test system.

The algorithm based on Adaptive filtering was developed using the Widrow's LMS (least mean square) method in the cancellation mode. Samples of the original signals form the primary and reference inputs to the adaptive filter and after a sufficient number of processing steps the phase difference is obtained from the coefficients of the filter. The FFT method uses the information from the phase spectra of the two signals in the frequency domain and determines the phase difference. The Correlation approach uses the amplitude spectra of each signal and also that of the cross correlated sequence obtained by multiplying the two signals in the DT domain and then calculates the phase difference. The Numerical method involves curve fitting and Newton's method to obtain an estimate of the zero crossings of the signals from which the phase difference is obtained.

The software development and system implementation was done on a North Star Horizon II, Z80 based microcomputer (NS) in conjunction with an MC6802 interface and a Hewlett-Packard Multiprogrammer (HPM); the whole forming an automated test

system. The MC 6802 interface serves as a communication link between the NS and the HPM. Data samples of the signals are obtained by using the High Speed Analog to Digital Converter card available on the HPM and these samples are then transmitted to the NS for subsequent processing by the algorithms discussed earlier. Computer simulation was done on a Hewlett-Packard 9845B Desktop minicomputer.

The design aspects of a suitable system for implementing these methods is also discussed. The role of the microprocessor as the control/communicating device is emphasized. Other applications in instrumentation are considered. A general purpose plot package was developed for the HP 9845B.