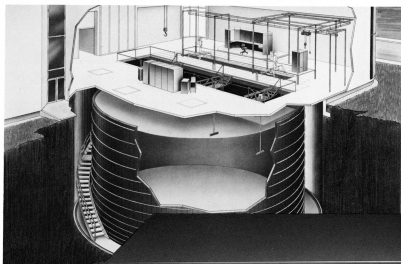


Using the HP 5183T to Characterize Sonar Transducers and Systems



*Tank illustration on cover
courtesy of Westinghouse*

TABLE OF CONTENTS

	Page
1.0 Introduction	2
2.0 HP 5183T Contributions	3
3.0 Sonar Transducer Characterization	5
3.1 Types of Transducers	5
3.2 Test Configurations	6
3.3 Tests Performed	8
3.3.1 Hydrophone Sensitivity	9
3.3.2 Projector Response	11
3.3.3 Complex Impedance	12
3.3.4 Directivity Pattern	17
3.4 Phase Measurement	20
3.5 Frequency Measurement	21
3.6 Pseudorandom Noise Sources	21
4.0 Sonar System Measurements	23
4.1 Projector Power Amplifiers	23
4.2 Sonobuoys	23
4.3 Array Gain	25
5.0 References	28
Appendices	29
A. Digitizing Analog Signals	29
B. Calibration Methods	29
B.1 Hydrophones	29
B.2 Projectors	31
C. Triggering	32

1.0 Introduction

The design and operation of effective sonar systems requires the use of calibrated transducers. Since many measurements have to be made as rapidly as possible when calibrating sonar transducers, an automatic, computer-controlled system is essential. The HP 5183T Digitizing Oscilloscope can be used as the principal component of an HP Series 200 computer-controlled measurement system that will completely characterize the behavior of sonar transducers. Features of the HP 5183T such as large memory and fast HP-IB interface enable the overall test system to achieve a high measurement throughput. For example, data for 720 different angles in a transducer directivity measurement can be captured and stored as fast as the transducer can be pulsed and rotated. The testing process is also speeded up by the ability of the HP 5183T to digitize and store only the specific signal parts of interest and to analyze this stored data.

Hewlett-Packard application note 205-2, "Sonar Transducer Calibration" [1], describes a sonar transducer test system that uses the HP 3042A Automatic Network Analyzer. The solution discussed in that application note is adequate for testing transducers in large ponds or pools; however, it is not as flexible and does not meet the needs of small tank testing as well as the solution using the HP 5183T.*

The main parts of a sonar system are the transducer array, the transmitter electronics, and the receiver electronics. Transmitter systems generate appropriate signals for the transducer array, and receiver systems process signals from the transducer array to extract information about underwater objects such as their range, Doppler shift, and bearing [2]. Although Chapter 4 describes examples of measurements relating to sonobuoys and transducer power amplifiers, the primary focus of this application note is on sonar transducer characterization.

For reference purposes, Table 1 presents the main features of the HP 5183T Digitizing Oscilloscope. The HP 5183T Literature Package contains more detailed information. Throughout this application note, we assume that the HP 5183T includes options 512 (512K samples of memory) and 301 (Adaptive Sample Rate (ASR)).

*For a variety of reasons, the HP 3577, which supercedes the obsoleted HP 3042A, cannot be used to test transducers in small tanks where acoustic reflections reach the test transducers more quickly than in larger tanks. The minimum pulse signal duration that can be measured with the HP 3577 is about 1.5 msec, but pulses in small tanks often have to be less than or equal to one msec.

- Two simultaneous input channels; HP 5183U allows four channels
- Differential or single-ended inputs
- Input amplifier bandwidth: nominal: 1 MHz (-1 dB), 3 MHz (-2 dB)
with anti-alias filter: 1 MHz (-4 dB), 3 MHz (-65 dB)
- Full-Scale Range (FSR): ± 100 mV to ± 50 V
- Resolution: 12 bits = 0.02% FSR
- Dynamic performance: At 1 MHz, 10 effective bits
- Memory: standard: 64K samples/channel
option 512: 256K samples/channel
- Digitizing rate: 4 MHz maximum
Can use an external timebase
- Enhanced triggering: Dropout, Delay, High frequency transient, External
- Analysis functions: FFT, RMS, Waveform math (+, -, *), Pulse Analysis
- HP-IB interface

TABLE 1. HP 5183T Digitizing Oscilloscope Features

2.0 HP 5183T Contributions

The capabilities of the HP 5183T to digitize, store, and process large amounts of test signal data make it an ideal instrument for the testing of sonar transducers and systems. In a typical sonar test system, data are collected and analyzed by a network analyzer, spectrum analyzer, voltmeter, or impedance analyzer depending on the particular measurement being performed. A versatile test system usually includes at least three of the above instruments and a controller. In contrast, the HP 5183T and an HP Series 200 computer perform most of the necessary measurements in such a test system. Thus, the HP 5183T replaces several instruments in this application.

The HP 5183T makes a major contribution to the testing of sonar transducers in small water-filled tanks. Since small tanks have more easily controlled conditions, are less expensive, and more convenient than large test pools or lakes, they are a more desirable transducer test environment. However, a major problem with small tanks is that acoustic reflections reach the test transducer in less time and with more power than they do in larger tanks. It is difficult to use continuous test signals because of interference between reflections and the test signals. If pulsed signals are used, the duration of the pulses must be long enough for a transducer to reach steady state, but short enough to keep the pulsed signals from overlapping with reflections (Figure 2.1) [3]. The HP 5183T easily implements pulsed-signal measurements. The HP 5183T digitizes signal pulses which are much shorter than 1 msec, provided that these signals contain frequency components no higher than 2 MHz (see Appendix A).

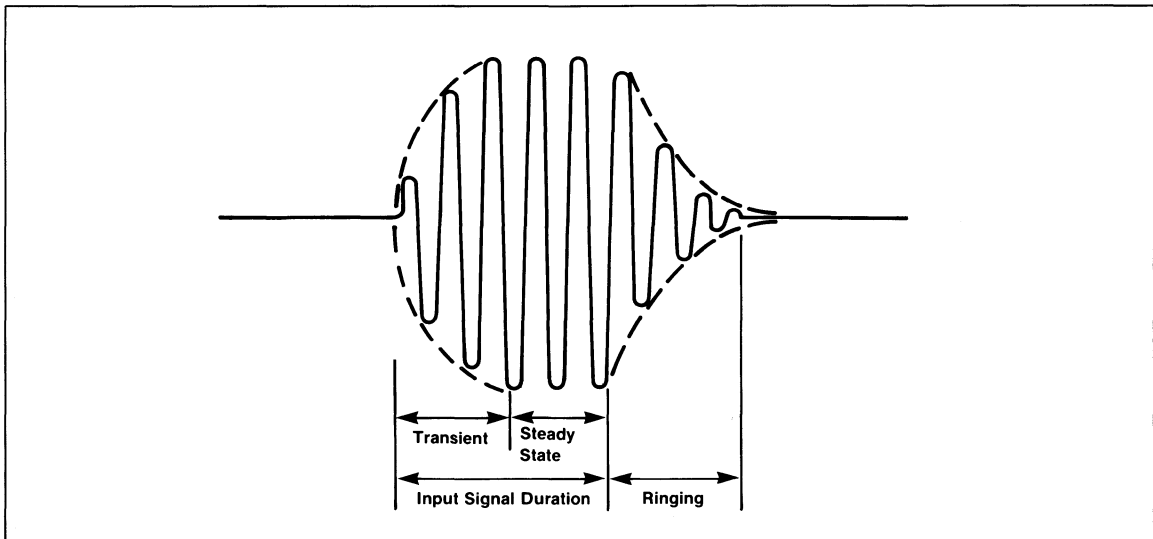


Figure 2.1. Sonar Transducer Test Signal Pulse

Besides allowing small tank testing, pulsed signals also allow high power level testing. In actual operation, some sonar projector transducers are driven with a pulse (tone-burst) of high power (e.g. 10 kW) to achieve increased range. The duty cycle of the driving signal must be set so that even though the power in the pulse exceeds the maximum transducer drive power, the average signal power does not exceed this maximum. Because it is desirable to test transducers at their operating power levels, the case described above requires low duty cycle pulsed signal testing. Since it is necessary to measure RMS voltage (e.g. projector response—section 3.3.2) and phase (e.g. complex impedance—section 3.3.3) of the signal bursts, the process of digitization, storage, and analysis with the HP 5183T provides the best solution.

The large memory—up to 512K samples—of the HP 5183T is useful in applications which require measurements to be made as parameters are varied. For example, the HP 5183T can capture data to calculate RMS voltage at 512 different frequencies to measure hydrophone sensitivity over some frequency range.

Built-in analysis functions relieve the system controller from some of the signal processing burden. These functions and other contributions of the HP 5183T to the testing of sonar transducers and systems are summarized below.

FEATURE

- (1) Fast Fourier Transform (FFT)
 - Power Spectrum (dBm)
 - Magnitude Spectrum (dBV)
 - Phase Spectrum
- (2) Root-mean-square (RMS)
- (3) Waveform mathematics
(multiplication, addition, subtraction)
- (4) Differential Inputs
- (5) Two simultaneous input channels;
Four channels with the HP 5183U.
- (6) Auto Advance;
Burst Mode Timebase
- (7) External Timebase
- (8) Adaptive Sample Rate (ASR)
- (9) Dropout Trigger
- (10) Delay trigger; External trigger;
Post-trigger delay
- (11) Resolution: 12 bits = 0.02% FSR
- (12) HP-IB interface

CONTRIBUTIONS

- Analyze complex signal bursts, such as chirp or FM signals. Perform precise phase calculations on short signal bursts. Calculate transducer magnitude response and impedance over a frequency range when a single burst of noise is used as the test signal.
- Used for transducer response, impedance, and directivity measurements.
- Phase difference between two signals. Power calculations. Array gain calculations.
- Reduce effects of noise on the signals. The long cables and low signal levels from hydrophones make noise more significant.
- Phase measurements. Power calculations. Impedance measurements. Array gain calculations. Increased throughput for response measurements.
- Capture successive signals as a parameter is varied. Use memory efficiently. Burst mode eliminates delays between records.
- Used to set frequency points for FFT calculations.
- Use memory efficiently when Burst mode cannot be used because of unknown signal durations.
- Capture power amplifier shutdown.
- Eliminates the need for a special receiver gating signal in pulsed-signal measurements.
- Up to 72 dB dynamic range for frequency domain analysis and directivity patterns.
- Allows automatic, computer-controlled operation.

3.0 Sonar Transducer Characterization

3.1 TYPES OF TRANSDUCERS

Sonar transducers perform conversions between sound energy and electric energy. A transducer that transforms acoustic signals into electric signals is called a receiver or hydrophone. The converse transformation uses a transducer called a projector. In some sonar systems, the same transducer acts as both the hydrophone and the projector.

Although research and calibration applications often use single transducer elements, most practical sonar systems require arrays of transducers. The advantages of using an array of transducers instead of a single transducer are increased sensitivity, enhanced directional capability, and improved signal-to-noise ratio. Given the same sound field, the sensitivity of an array is better than that of a single element since a number of elements will generate more voltage when connected in series or more current when connected in parallel. By comparing the phase of acoustic signals arriving at different elements in a hydrophone array, a sonar system can distinguish sounds coming from different directions. Similarly, a projector array can send a sound beam in a desired direction if time delays are added to the electrical signals sent to its transducer elements. The HP 5183T evaluates beam-forming delay networks by simultaneously measuring the response of individual array elements as described in sections 3.4 and 4.3. Signal-to-noise ratio is improved for arrays because their directional properties allow them to reject isotropic noise to a greater extent than the desired signal. A cylindrical array is shown in Figure 3.1.

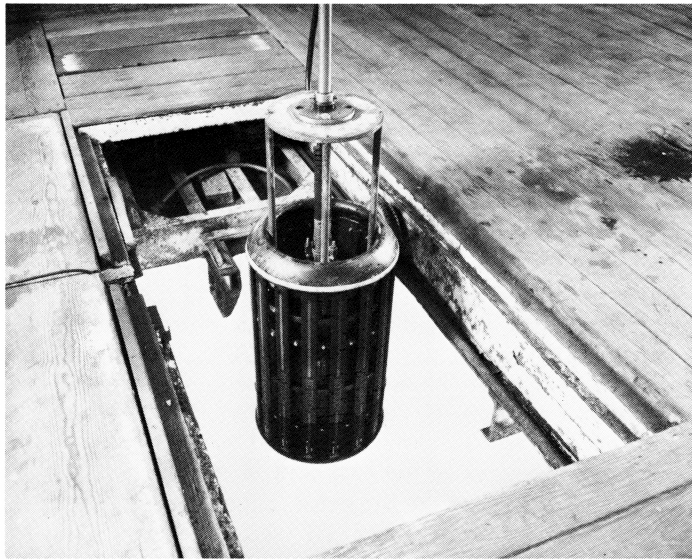


Figure 3.1. Cylindrical Transducer Array

Because transducer array elements are sometimes tested individually, the calibration process requires many measurements. Therefore, reduction in measurement time and complexity are major goals. Features of the HP 5183T which help achieve these goals are the 512K sample memory (option 512), built-in waveform processing, and the capability (HP 5183U) to digitize four channels simultaneously.

3.2 TEST CONFIGURATIONS

Evaluation of sonar transducers consists of testing acoustical and electrical characteristics while the transducers are immersed in water and operated at normal power levels. A variety of test configurations are used, including small pressurized tanks (approximately ten feet in diameter), large pools or ponds, and special anechoic pools or tanks.

Since test signals are corrupted by reflections from the surface, bottom, and sides of the testing medium (Figure 3.2) and by impurities within the water, test facilities strive to reduce the effects of these undesirable factors. One approach to reducing these effects is to use an anechoic test pool or tank to achieve adequate free-field* conditions by attenuating signal reflections [4]. A sound field in a tank approximates a free-field only when the length, width, and depth of the tank are large as compared with the wavelength of the sound in water. This condition sets a test signal low frequency limit that varies depending on tank size.

*A free-field is a homogeneous isotropic medium without boundaries.

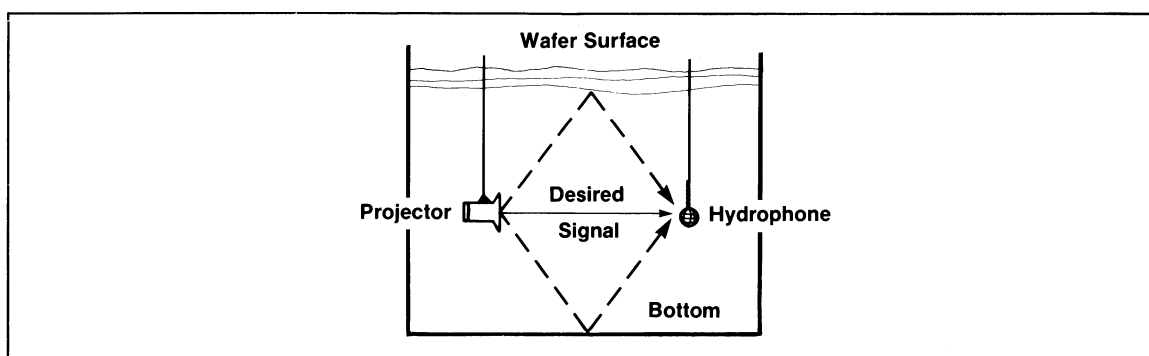


Figure 3.2 Interference of Reflections with Desired Signal

To eliminate the effects of acoustic reflections in the measured transducer data, the test system usually employs pulses of sound. The response is measured before pulse reflections get back to the receiving transducer. Therefore, it is advantageous to use large test pools so that reflections take a long time to get back to the center. However, as discussed in Chapter 2, large pools are expensive, so testing in smaller tanks is desirable.

Figure 3.3 shows the block diagram of a pulsed-signal transducer test system which is based on the HP 5183T, and has a configuration similar to the one at TRANSDEC (Transducer Evaluation Center at the Naval Ocean Systems Center (NOSC) in San Diego, CA [1, 4]). This system is appropriate for use with small tanks and has the capability to perform all of the measurements described in this application note. The test system is controlled by the HP Series 200 computer with an HP 98622A GPIO Interface card to produce the gain control signal. The HP 5182A Waveform Recorder/Generator produces test signals for the projector. Either the signal burst or source sync pulse may be used as an external trigger for the HP 5183T (see Appendix C).

The HP 5182A has a 10 MHz bandwidth, 16K sample memory, and ten-bit D/A converter and is used as a digital arbitrary waveform synthesizer. The HP 5182A offers great flexibility in the types of test signal waveforms that may be generated (e.g. tone-bursts and chirp pulses), without sacrificing frequency resolution and accuracy. Since a typical sonar transducer test facility requires a signal source with 1 Hz resolution and accuracy over a frequency range from DC to 1 MHz, conventional function generators are not acceptable. Some facilities use a frequency synthesizer and a custom circuit to gate the output to produce phase coherent tone-bursts. This approach is inconvenient and is still not as versatile as using an HP 5182A.

The measurements described in this application note assume the use of a frequency synthesizer or generator that can produce phase-coherent tone-bursts. Measurements over a frequency range are accomplished by varying the frequency of the synthesizer and recording data at a set of frequencies covering the range of interest. An alternative (which is discussed in section 3.6) to the swept frequency approach is to use pseudorandom noise (PRN) as the test signal (the HP 5182A can also produce PRN). However, frequency tones are assumed to be the test signals for measurements described in the remainder of this chapter because they can be used in all test situations, and they achieve better signal-to-noise ratio at each measurement point than PRN signals. Changes in measurement procedures that would be required when using a PRN test signal are discussed in section 3.6.

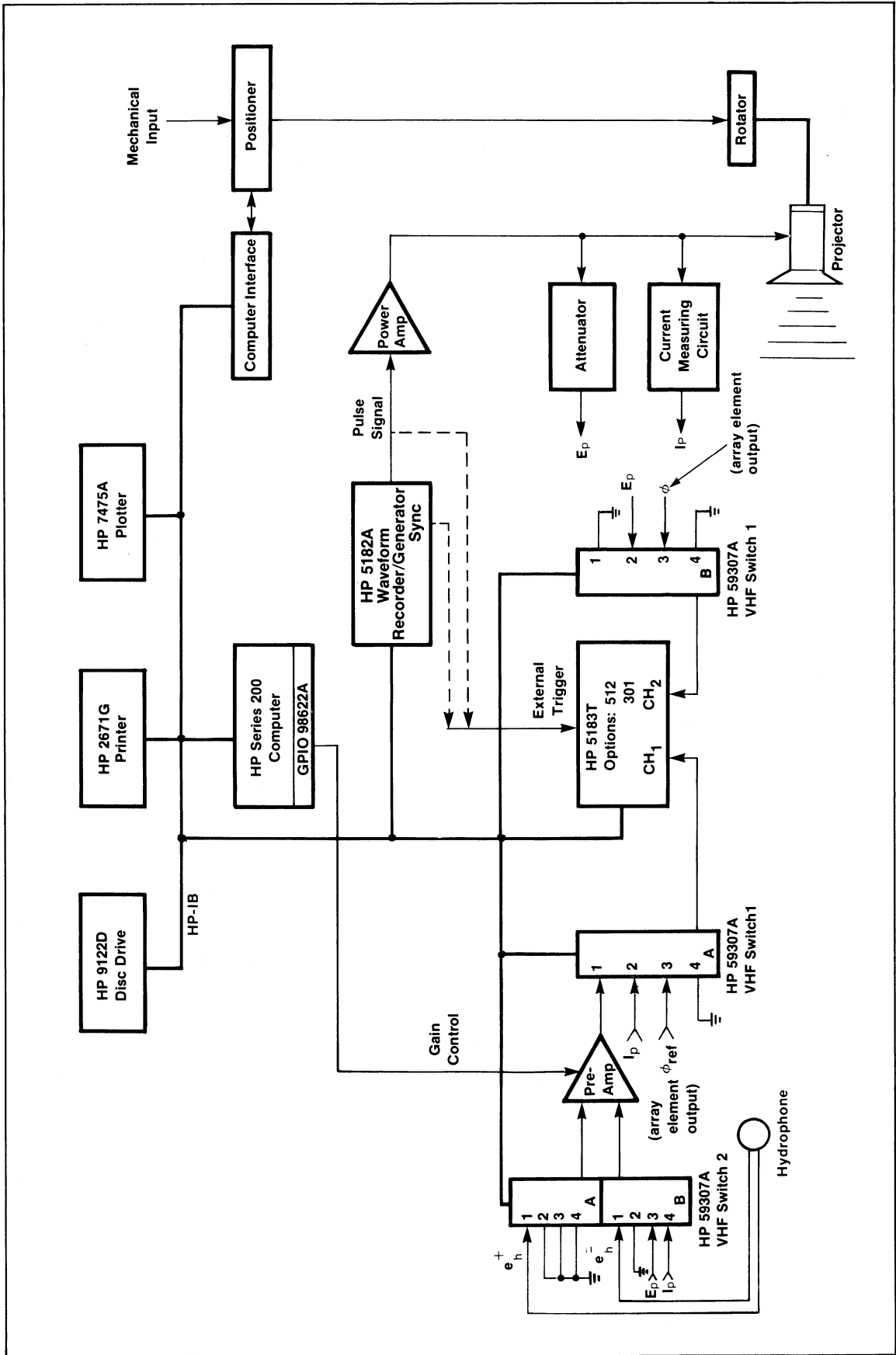


Figure 3.3 Sonar Transducer Test System

3.3 TESTS PERFORMED

Calibration of a sonar transducer consists of determining its response as a function of frequency and direction. The response of a transducer is defined as the proportionality factor with which a hydrophone converts sound into electricity or a projector converts electricity into sound. Figure 3.4 shows an example of a hydrophone response plot. Although there are many methods for determining transducer responses [5, 6], most of these methods produce measurement results in one of the following forms:

- 1) Hydrophone Sensitivity
- 2) Projector Response
- 3) Projector Complex Impedance
- 4) Directivity Pattern
- 5) Phase difference between two transducers

This application note focuses on the data collection and analysis aspects of transducer calibration, not on the variety of overall calibration methods.

To produce transducer evaluation results in one of the forms listed above, measurements of current, voltage, and phase are required. Hydrophone open-circuit voltage is measured from the output of an amplifier that has its high-impedance input connected to the hydrophone terminals. Figure 3.5 shows examples of circuits from which projector voltage and current may be measured. The HP 5183T measures phase from voltage or current waveforms captured on two simultaneous channels. Table 2 summarizes typical characteristics of transducer test signals.

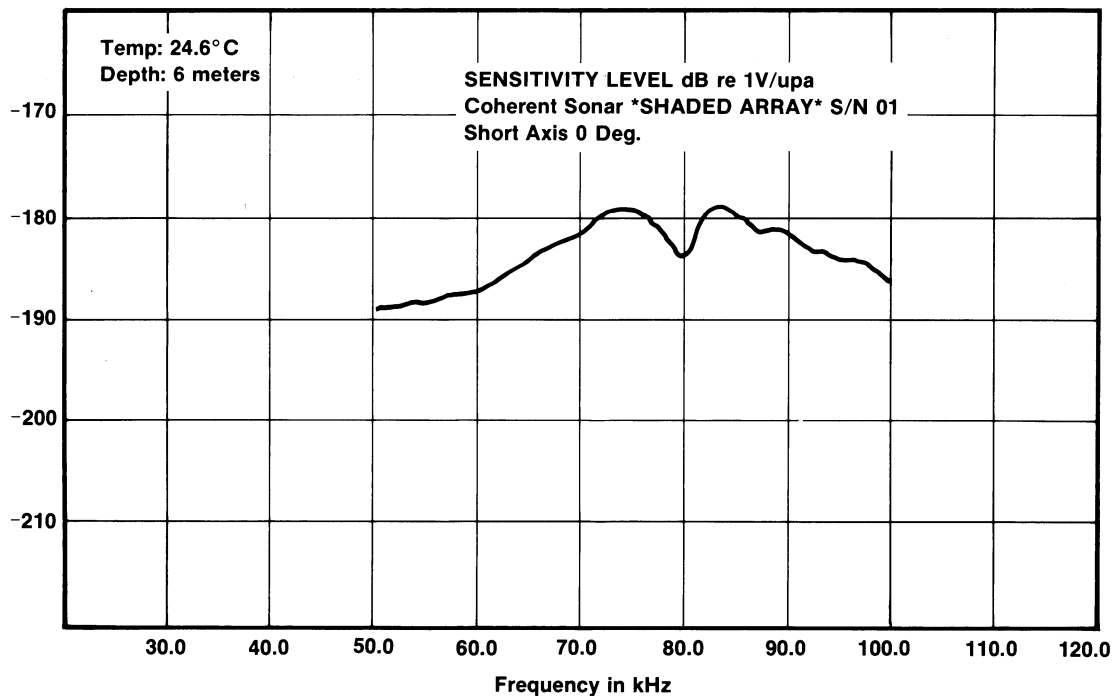


Figure 3.4 Hydrophone Sensitivity

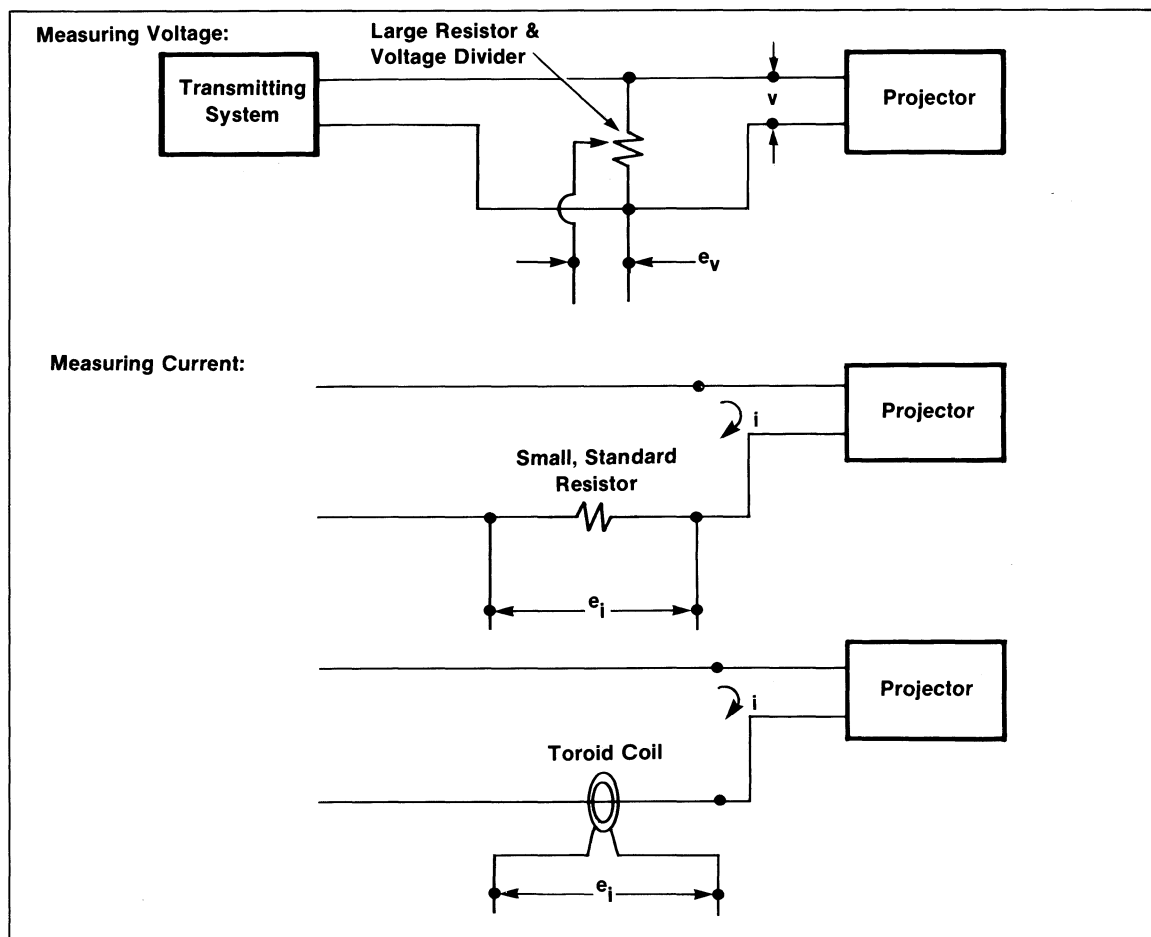


Figure 3.5 Circuits for Measuring Projector Voltage and Current (after Bobber, [5])

Hewlett-Packard makes available example programs that produce hydrophone and projector response, complex impedance, and directivity pattern measurements using the HP 5183T, an HP 9816 computer, and an HP 5182A Waveform Recorder/Generator [7].

- Pulsed or CW
- Pulse widths: 1 msec to 10 msec
- Pulse form: single-tone or multi-tone
- Frequency range of pulse tones: 50 Hz to 500 kHz (low power)
50 Hz to 50 kHz (high power)
- Signal dynamic range: 50 dB

Table 2. Typical Characteristics of Sonar Transducer Test Signals

3.3.1 Hydrophone Sensitivity

Hydrophone response or sensitivity is calculated from measurements taken across the hydrophone terminals of the open-circuit voltage caused by plane waves of known acoustic pressure. This response is usually expressed in terms of decibels relative to one volt per micropascal and is most often measured as a function of frequency. The response must be measured under free-field conditions, and the temperature and pressure must be specified since the response is a function of these parameters in addition to frequency. Since signal levels from a hydrophone can be very low (-50 dB to -60 dB re 1 volt), some kind of amplifier for the hydrophone voltage output is necessary. Also, because hydrophone signals are measured in noisy environments, they are usually measured in a differential configuration.

Several methods are used to measure a hydrophone response characteristic. Two of these methods, comparison and reciprocity, are discussed briefly in Appendix B for readers unfamiliar with this topic. Reference [6] gives a concise description of standard calibration procedures.

Once the method for obtaining the hydrophone response has been determined, hydrophone open-circuit RMS voltages and projector RMS driving currents are measured. Projector driving currents are actually calculated from measurements of voltages produced by circuits such as those shown in Figure 3.5. Hydrophone voltage or the voltage produced by a projector current measuring circuit is measured by setting up the HP 5183T as follows:

- (1) Input: Depending on the particular test configuration, set the CHANNEL 1 input to either single-ended or differential mode. Set the input amplifier RANGE and OFFSET according to the maximum level of the signal. Set AC COUPLING. Connect the voltage signal to the CHANNEL 1 input.
- (2) Memory: Set the RECORD LENGTH to 1024 samples. With the maximum memory of 512K samples, it is then possible to collect response data for 512 points. For example, a hydrophone response could be calculated over a 50 kHz range, with one data point every 100 Hz.
- (3) Recording Mode: AUTO ADVANCE
- (4) Timebase: MAIN RATE = 500 nsec (Thus, 1 MHz is the highest frequency signal that can be accurately reconstructed—See Appendix A).
With this sample interval and a 1024-point record, 512 μ sec worth of data will be captured.
Set to MAIN timebase
- (5) Trigger: The INTERNAL trigger may be used if a voltage from a projector current measuring circuit is being recorded. Set the trigger LEVEL so that the HP 5183T triggers at the beginning of a pulse (tone-burst). Set trigger POSITION to delay recording until after the pulse transient has stabilized (see Figure 2.1). The transient is typically about 600 μ sec long. A trigger position of +1200 samples will provide a 600 μ sec delay (1200 samples * 500 nsec/sample = 600 μ sec), so that the record will be filled with the steady state portion of the pulse. Note that trigger POSITION may be set in terms of number of samples, percent of record length, or time.
Triggering is more complicated when recording a hydrophone voltage (see Appendix C). The best method is to use an EXTERNAL trigger. Set trigger POSITION so that only the steady state portion of the pulse is recorded. The use of INTERNAL trigger is difficult, since reflections of the initial pulse are also received by the hydrophone and could be recorded into the next record. If the initial pulse is larger in amplitude than the reflections and if the trigger LEVEL is set high enough so that only the initial pulse causes a trigger, then INTERNAL trigger can be used.
- (6) Sweep: NORMAL mode

The projector in the test system is now pulsed 512 times, each time at a frequency that is incremented from the previous one, so that an entire frequency range is covered. The time between pulses is determined by the particular test pool or tank. The next signal pulse cannot occur until reflections from the previous pulse have dissipated. In the case of a large pool such as the one at TRANSDEC, pulses may be about 200 msec apart.

After the HP 5183T memory has been filled with test waveform data, RMS values for each record are calculated. PERIODIC RMS is used to calculate the RMS value of an integral number of periods of the steady state portion of the pulse (Figure 3.6). About 400 μ sec of data should be used for the periodic RMS calculation. The number of periods is equal to the closest integer to $f_0 * 400 \mu$ sec, where f_0 is the frequency of the pulse. After each RMS value is calculated, it is transferred to the controller. Equations from the comparison, reciprocity, or some other method are used to compute the hydrophone free-field voltage sensitivity in decibels relative to one volt per micropascal, and a plot versus frequency is generated. A typical plot was shown previously in Figure 3.4. The reliable operating range of the hydrophone is the relatively flat region below resonance.

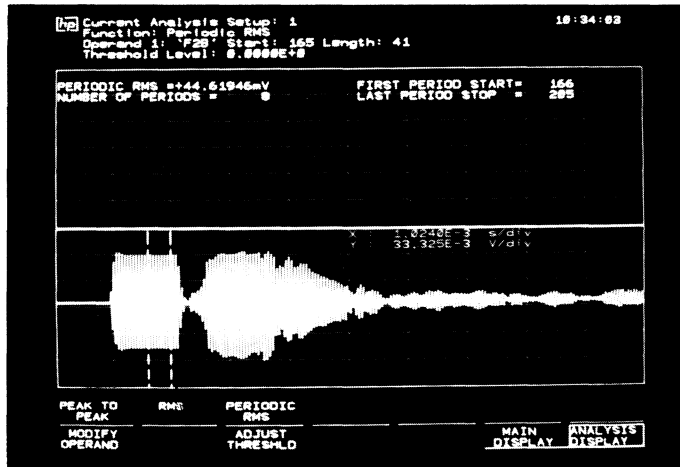


Figure 3.6 PERIODIC RMS of Received 20 kHz Signal Burst (Main Rate = 10 μ sec)

3.3.2 Projector Response

Projector responses are usually measured as transmitting current responses; however, transmitting voltage responses are also measured. The projector transmitting current response is defined as the acoustic pressure produced by the projector one meter from its center along its acoustic axis by one ampere of input current. This projector response is expressed in decibels relative to one micro-pascal per ampere and is most often measured as a function of frequency. As with hydrophone response calibration, there are many different methods for determining projector response characteristics. Appendix B gives a summary of the most common methods. All of the methods involve measurement of hydrophone open-circuit voltage output and projector input current or voltage. These measurements are made exactly as described in section 3.3.1 and are used to calculate projector response versus frequency. Figure 3.7 shows a typical plot.

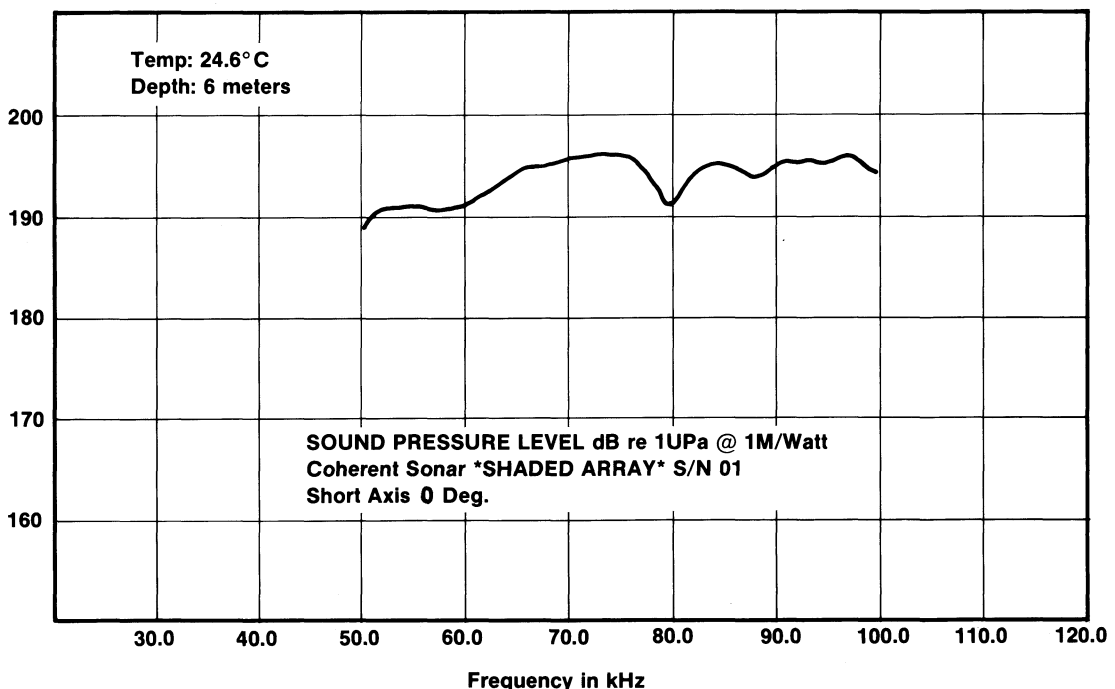


Figure 3.7. Projector Response

3.3.3 Complex Impedance

Complex impedance measurements are used for several purposes. Knowledge of a transducer's impedance is necessary for impedance matching between a transmitter or receiver and the transducer. Calculations of transducer efficiency and driving voltage from current responses (or *vice versa*) are made using impedance data. Using the self-reciprocity technique, projector or hydrophone response for a reciprocal transducer is calculated from the impedance measurement, the reciprocity parameter, and the distance between the transducer and the reflector. Finally, complex impedance measurements have been shown to be an effective tool for diagnosing transducer failures and impending failures.

Complex impedance at a transducer's electrical terminals is calculated from measurements of voltage, current, and phase. Magnitude of the impedance is calculated by dividing the RMS value of the projector driving voltage waveform by the RMS value of the projector driving current waveform. Phase of the impedance is the difference in phase between the projector voltage and current waveforms, using (arbitrarily) the current waveform as a reference. Although measured electrically, the transducer complex impedance is a function of the acoustical and mechanical characteristics of the transducer as well as the acoustical characteristics of the medium into which the transducer is radiating sound.

The transducer testing configuration can significantly influence the results of the impedance measurements. Waveform data are affected by the lengths of the cables used to connect test equipment with transducers as well as whether or not the cables are immersed in water. To acquire useful data, the testing configuration should be as similar as possible to the actual usage configuration (transducer immersed in water, minimum reflections from tank walls, etc.).

Using the HP 5183T, the impedance and admittance of a sonar transducer are measured as follows:

- (1) Input: Set the HP 5183T to record two channels simultaneously. Select either single-ended or differential mode depending on the particular test configuration. Set the RANGE and OFFSET of the input amplifiers according to the maximum level of the signals. Set AC COUPLING.
Connect the output of the projector driving voltage measurement circuit to CHANNEL 1, and the output of the projector driving current measurement circuit to CHANNEL 2.
- (2) Memory: Set the RECORD LENGTH to 1024 samples. With the maximum memory of 256K samples per channel, the instrument will be able to collect complex impedance data for up to 256 frequency points.
- (3) Recording Mode: AUTO ADVANCE
- (4) Timebase: Set to MAIN timebase.
Set MAIN RATE sample interval as long as possible—See Appendix A. For example, if 100 kHz is the highest frequency being tested, set the main rate to 4 μ sec.
- (5) Trigger: INTERNAL CHANNEL 1 (voltage waveform).
Set LEVEL and HYSTERESIS according to the signal.
POSITION: Set so that, using a 1K memory record and a sample interval of 4 μ sec, the steady state portion of the pulse will be recorded in the middle of the record. For a 1 msec pulse with 600 μ sec initial transient, the POSITION would be -312 samples. The 400 μ sec steady state portion of the pulse would be in samples 462 to 562 of the 1024 point record.
- (6) Sweep: NORMAL mode

The transducer is now given a pulsed input signal at up to 256 different frequencies covering the range of interest. To get the best accuracy in the phase measurements, select frequencies which are multiples of the frequency resolution of the FFT (see Section 3.4). For example, with a 250 kHz sample rate and 1024 point FFT, the frequency resolution is 244.14 Hz. Thus, test frequencies might be 50,049 Hz, 50,293 Hz, etc. After data have been captured, perform the following analysis:

- (1) FFT: Normally, each captured record contains data for a single frequency point (the pulse captured in that record was single-tone). The HP 5183T's built-in FFT function calculates the phase for the complex impedance as follows:
 - A) Configure the traces and operands so that the operands are CHANNEL 1, RECORD 1 and CHANNEL 2, RECORD 1. The analysis in steps B through E will be repeated for CHANNEL 1, RECORD 2 and CHANNEL 2, RECORD 2, etc., until all the records have been analyzed (Figure 3.8).

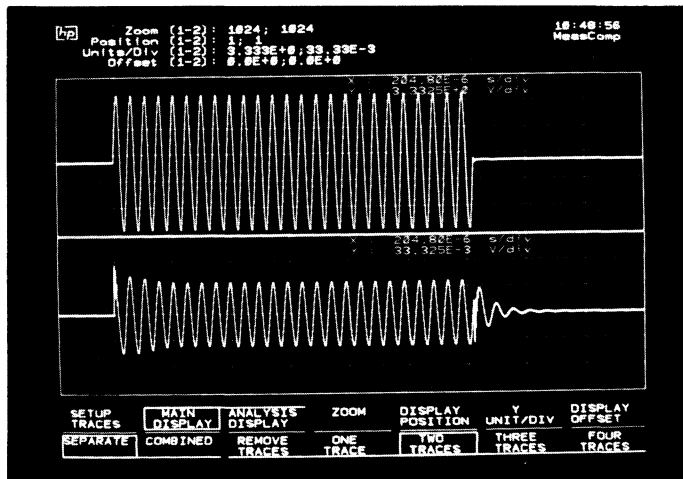


Figure 3.8 Projector Voltage (Top Trace) and Current Waveforms (20 kHz Bursts, Main Rate = 2 μ sec)

- B) In the FREQUENCY DOMAIN menu, use SELECT WINDOW to define a HANN window for each analysis setup. The Hann window provides better frequency resolution.
- C) Assign the PHASE SPECTRUM function to ANALYSIS SETUPS 1 and 2. There are four traces: one for each of the two channels, and one for the analysis result of each channel (Figure 3.9).

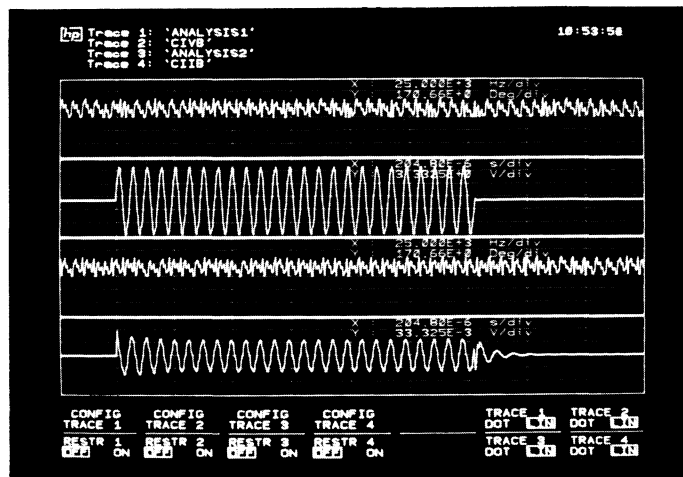


Figure 3.9 Phase of Projector Driving Voltage (Top Trace) and Current (Third Trace)

- D) The phase corresponding to the frequency of the captured records is now available at one of the points in each phase spectrum result. Use CURSORS to get the phase values for these points (Figure 3.10). See section 3.4 for a detailed discussion on phase measurement with the FFT function.

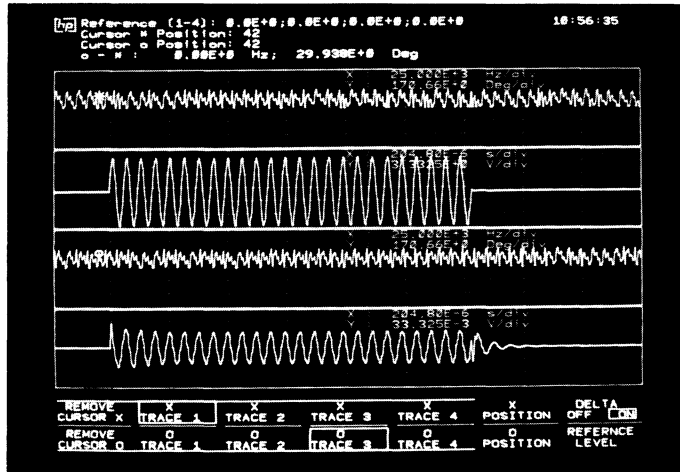


Figure 3.10 Using Cursors to get Phase Values at 20 kHz

- E) To calculate the phase for the complex impedance, use CURSOR DELTA to subtract the phase at the correct point in the phase spectrum (TRACE 3) of TRACE 4 (current) from the phase at the correct point in the phase spectrum (TRACE 1) of TRACE 2 (voltage) (Figure 3.10). Transfer this information to the controller. (The measurement process is shown in Figure 3.11).
- F) Repeat steps A to E for all corresponding pairs of records from CHANNELS 1 and 2 (record 1 from each channel, record 2 from each channel, ..., record 256 from each channel). The controller will now have the necessary phase data over the frequency range measured.

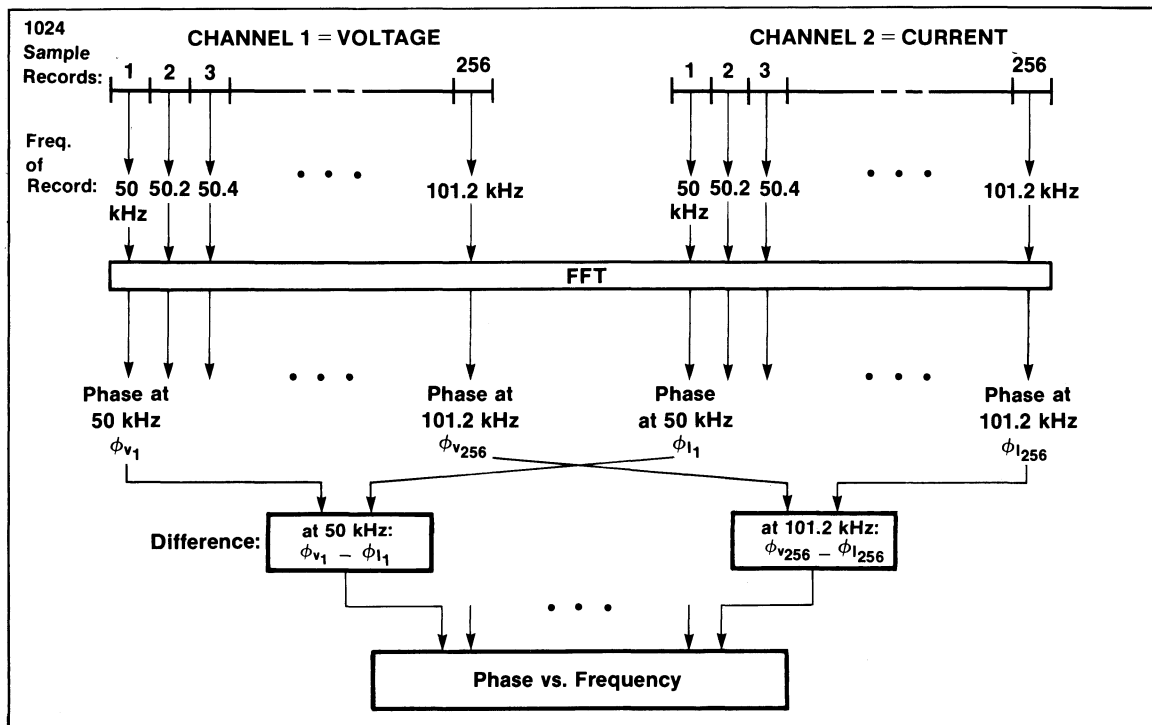


Figure 3.11 Phase Measurement Process for Complex Impedance Calculation

- (2) RMS: Follow the same procedure as explained for the FFT, except assign the RMS function in the VOLT METER menu to ANALYSIS SETUPS 1 and 2. Use PERIODIC RMS to get the RMS value of an integral number of periods in the steady state portion of the pulse. This corresponds to taking the RMS of x periods (where $x = \text{closest integer to } f_0 \cdot 400 \mu\text{sec}$ and f_0 is the pulse frequency) within sample numbers 462 to 562 (Figure 3.12). In step D above, transfer the RMS value for each of the two operand waveforms to the controller. In the controller, divide the RMS value of the CHANNEL 1 record (voltage) by the RMS value of the CHANNEL 2 record (current). The result is the magnitude $|Z|$ of the complex impedance at the frequency point corresponding to those records. Continue with step F as above. The controller will then have the necessary magnitude data over the frequency range measured.

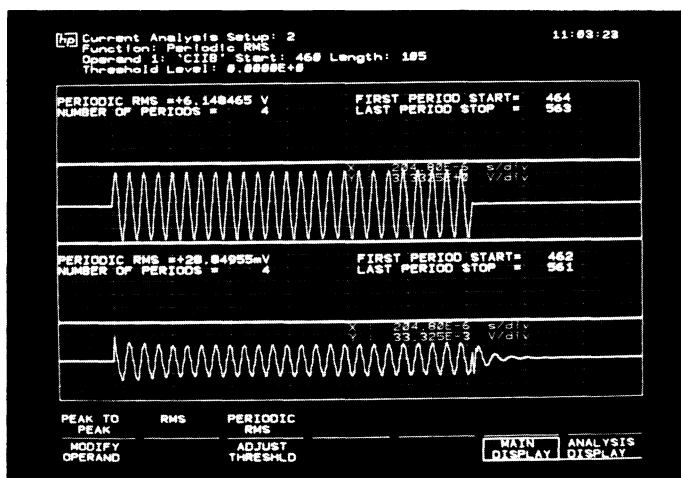


Figure 3.12 PERIODIC RMS of Captured Bursts (TRACE 2 = Voltage, TRACE 4 = Current)

The complex impedance data in the HP Series 200 controller may now be displayed in an appropriate format for interpretation. Complex impedance parameters are related as follows:

$$|Z| = V_{\text{rms}}/I_{\text{rms}} \quad \text{phase: } (\phi) = \phi_{\text{voltage}} - \phi_{\text{current}}$$

$$Z = R + jX \quad R = |Z| \cos\phi \quad X = |Z| \sin\phi$$

$$A = G + jB \quad A = 1/Z$$

where,

Z = complex impedance

A = complex admittance

R = resistance

X = reactance

G = conductance

B = susceptance

Plots of any of the real variables R, X, G, or B versus frequency are used in addition to conventional plots of magnitude and phase versus frequency. Examples of these types of plots are shown in Figures 3.13 and 3.14.

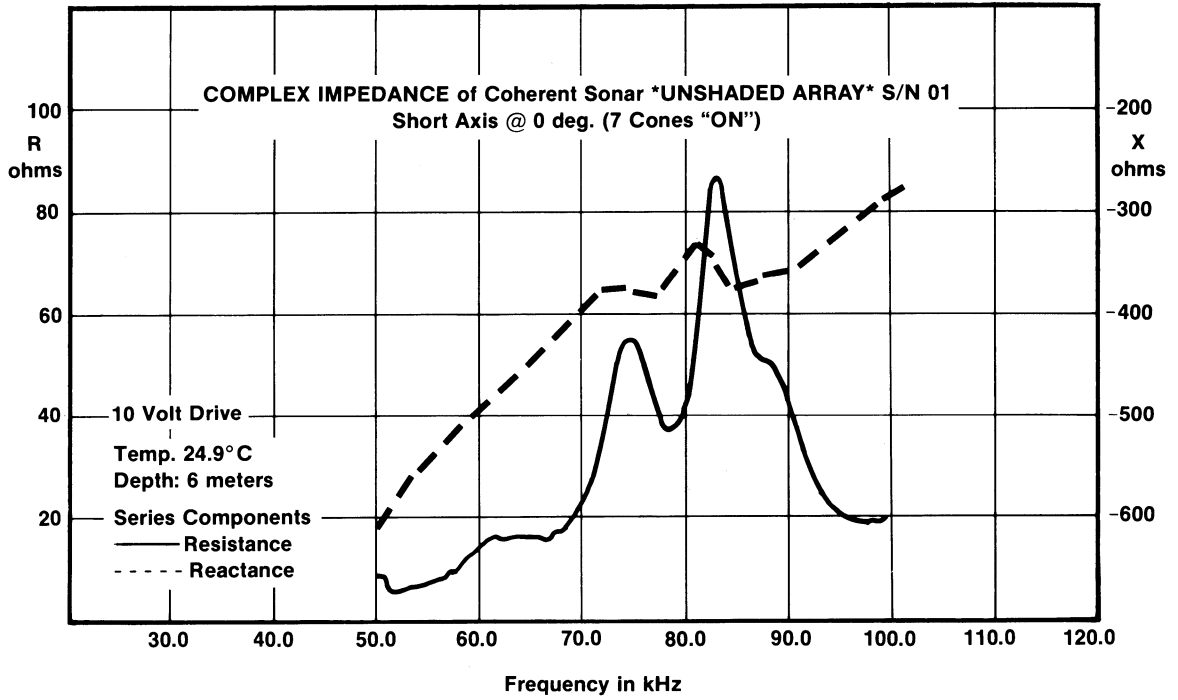


Figure 3.13 Complex Impedance

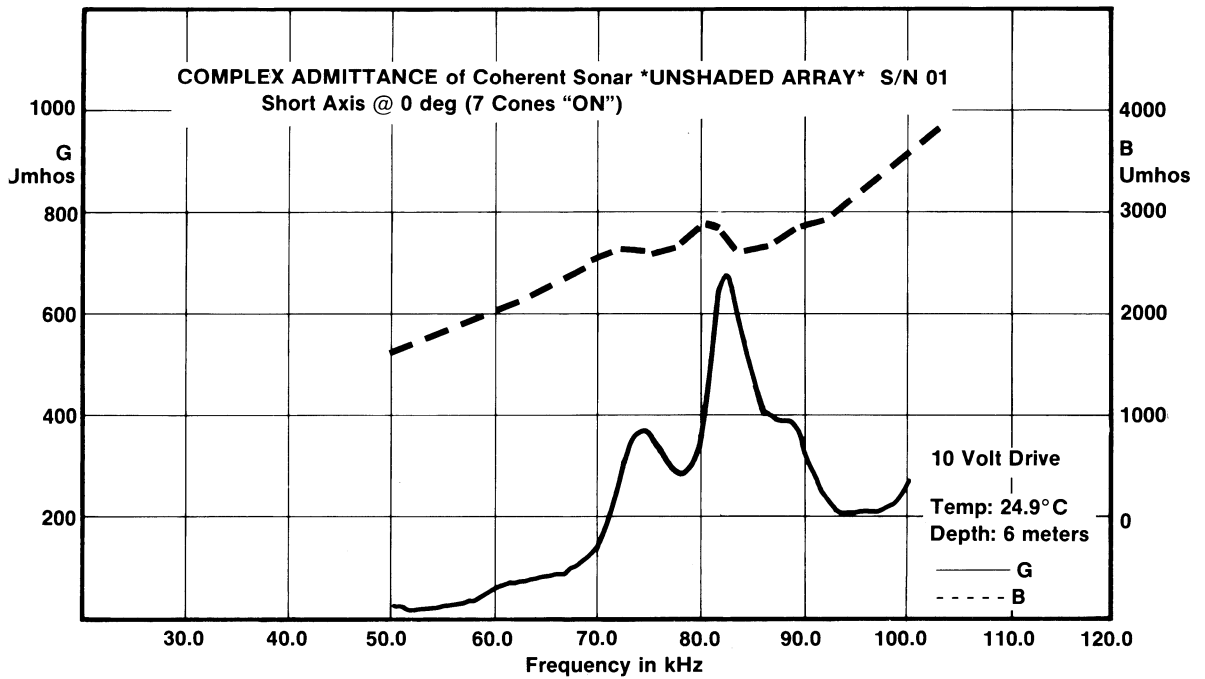


Figure 3.14 Complex Admittance

3.3.4 Directivity Pattern

A directivity pattern (or beam pattern) describes the directional characteristics of a transducer array. Specifically, the pattern is a plot of the transducer relative response as a function of the direction of a transmitted or received sound signal. For hydrophones, the response level is the open-circuit voltage at the hydrophone terminals; for projectors, the response level is the pressure produced at a specified distance from the projector when it is given a specified current. A reciprocal transducer has identical transmitting and receiving patterns. Although transducer directionality is three-dimensional, the directivity pattern is a two-dimensional (often polar) plot in a particular plane and at a single frequency. A left-handed polar coordinate system has been adopted as a standard for transducer calibration (Figure 3.15) [6]. The origin of the three axes corresponds to the acoustic center of the transducer, and the x-axis usually corresponds to the transducer acoustic axis. Thus, responses may be described as a function of two angles: $R(\theta, \phi)$. Typically, six to twelve beam patterns characterize one transducer.

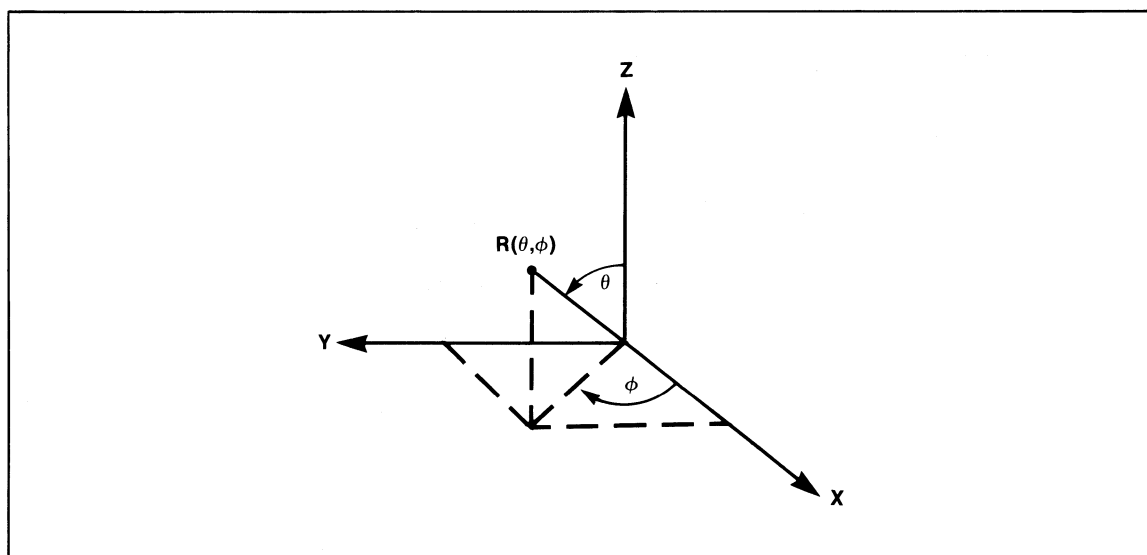


Figure 3.15 Standard Coordinate System

The directivity pattern is plotted in decibels and is normalized so that the direction of maximum sensitivity has a level of 0 dB. Thus, all other response levels at different angles are a certain number of decibels below 0 dB. For a regular array, the beam pattern is symmetrical, and the direction of maximum sensitivity (direction from which received sound has the greatest phase correlation) is the acoustic axis. The maximum response of an array can be steered by either rotating the array mechanically or by electrically changing the phase relationship among different elements.

Beam width and the level of the highest minor lobe are important characteristics of the directivity pattern. The beam width is defined as the total angle between two directions that are a certain number of dB (usually 3, 6, or 10) below the maximum. In addition, beam pattern data are used to calculate the directivity index and the directivity factor for a transducer array. These measures will be discussed further in chapter 4.

A technique for altering the directivity pattern of an array is known as shading. The amplitude response of particular elements in the array is changed in order to create the desired beam pattern. Usually, the level of minor side lobes is purposely reduced to increase the relative maximum sensitivity. Varying the spacing and phasing of array elements also changes the beam pattern. The disadvantage of shading is that it reduces the array gain when the operating environment involves the use of coherent signals in uniform incoherent noise. The HP 5183T analyzes array shading schemes using the procedures described in section 4.3.

To measure hydrophone directionality, the hydrophone is rotated and its response to sound waves from a fixed, calibrated projector is measured. Projector directionality is measured by rotating the projector and recording the response of a fixed, calibrated hydrophone to the projected sound waves. The angle of the test transducer must be known and controlled throughout the test, and the

distance between the test transducer and the fixed transducer should be large compared with the largest dimension of the test transducer. The long distance between transducers, coupled with the fact that differences in signal levels are being measured, makes it necessary to have better free-field test conditions than are required for response testing. These requirements make it difficult to do beam pattern measurements in small tanks. However, some near-field techniques have been developed to address this problem [5, 8, 9]. With one of these techniques, it is possible to use a test tank of diameter only 1.5 times that of the transducer array [10].

To measure transducer directivity with the HP 5183T, use the following procedure:

- (1) Input: The voltage produced at the output terminals of a hydrophone is the signal that must be measured whether recording a projector or a hydrophone beam pattern. Depending on the particular test configuration, set the CHANNEL 1 input to either single-ended or differential. Connect the hydrophone voltage output to CHANNEL 1 input. Set the input amplifier OFFSET and RANGE. Set AC COUPLING.
- (2) Memory: Configure the memory to one record of 512K samples.
- (3) Recording Mode: CAPTURE
- (4) Timebase: Set MAIN RATE to 2 μ sec.
Thus, 250 kHz is the highest frequency that may be accurately reconstructed— See Appendix A.
Select BURST mode, and set the number of samples per burst to 512. After each trigger, the HP 5183T will record 512 samples. With the maximum memory of 512K samples, directivity pattern data for up to 1024 angles can be recorded. Typically, data are taken every 0.5 degrees, so only 720 512-point bursts are required for a full beam pattern.
- (5) Trigger: Since a hydrophone signal is being recorded, reflections could cause false triggering. If the INTERNAL CHANNEL 1 trigger LEVEL and HYSTERESIS cannot be set to avoid triggering on reflections, an EXTERNAL trigger will have to be used (see Appendix C).
POSITION: When the HP 5183T timebase is in BURST mode, the trigger POSITION is automatically set so that the first sample in every burst is the trigger sample (POSITION = 0%).
- (6) Sweep: SINGLE mode.

As the test transducer is rotated, response data are recorded. Since there is only a two sample delay between burst measurements, the actual time necessary to record data for 720 different angles is $(512 + 2 \text{ samples/burst}) * (2 \mu\text{sec/sample}) * (720 \text{ bursts}) = 0.74 \text{ seconds}$. This time is insignificant compared with the minimum time between signal pulses imposed by power duty cycle limitations and reflection dissipation times. For example, a 5% duty cycle means that an overall period of 20 msec is required for a 1 msec pulse. Usually, the time required for reflections to dissipate is larger than the time delay imposed by the power duty cycle limitation. For example, a 200 msec delay between pulses (e.g. TRANSDEC) adds another $(720 * 200 \text{ msec}) = 144 \text{ seconds}$ to the overall data capture time.

The controller keeps track of the angular position of the rotating transducer. The beginning angle is assigned to the first data burst captured. To put these data into the proper format for a directivity pattern, analysis is performed as follows:

- RMS: To allow for transducer transient response settling, only about 300 μ sec of each burst are used for the RMS calculation. With the 2 μ sec sample interval, this 300 μ sec interval is between samples 300 and 450 in the 512-sample burst. PERIODIC RMS is used to calculate RMS data for the specified 300 μ sec interval (Figure 3.16).
Note that if the EXTERNAL trigger is used, it may be possible to set the trigger to occur when the signal pulse reaches steady state. In this case, cycles at the beginning of the burst could be used.

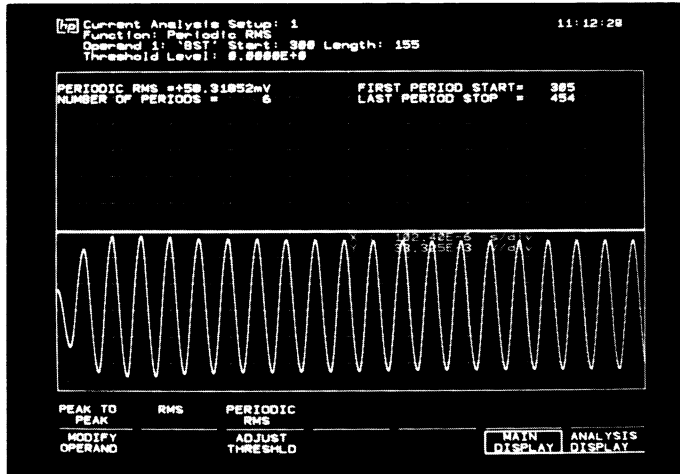


Figure 3.16 PERIODIC RMS on Six Periods of Received 20 kHz Burst

Once the RMS data for all 720 points have been transferred to the controller, these data points are normalized by dividing every point by the maximum RMS value for any point. The maximum RMS value will usually occur at the point corresponding to the transducer acoustic axis and be at zero degrees. The normalized values are converted to decibels by applying the formula: $20\log(\text{RMS value})$. The dB values are then plotted versus angle on either a polar or rectangular coordinate plot (Figure 3.17). The HP 5183T's 12 bit vertical resolution provides up to 72 dB dynamic range for the directivity plots.

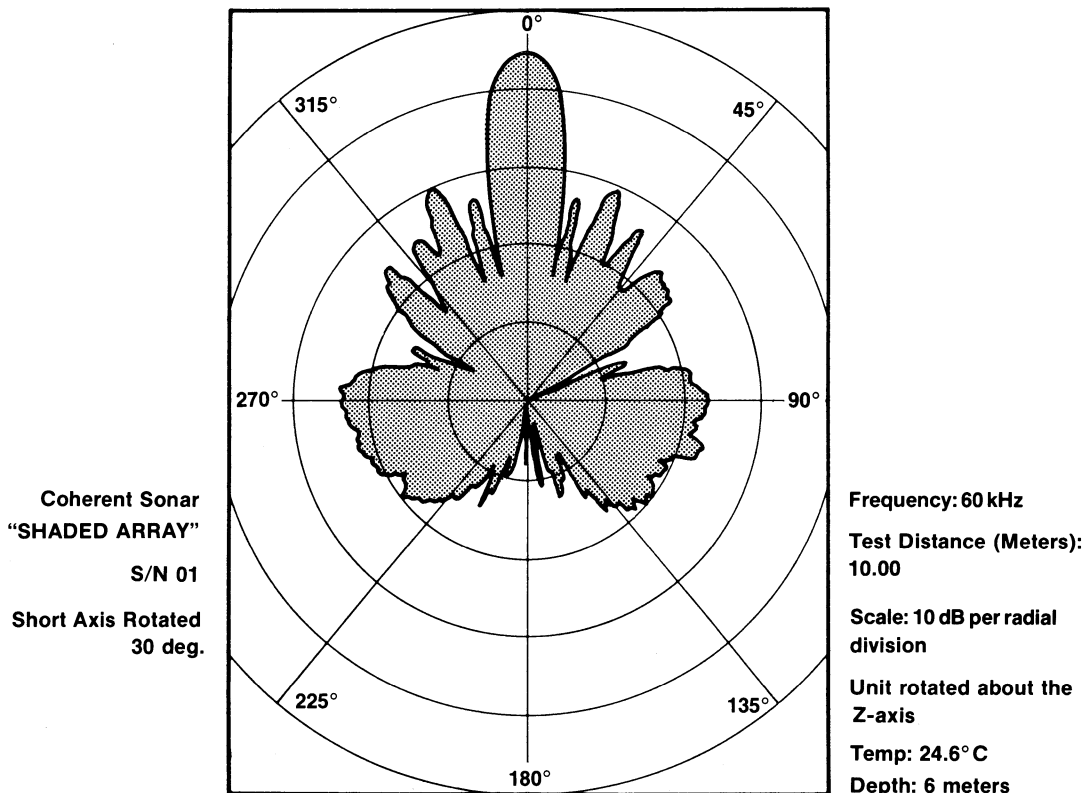


Figure 3.17 Directivity Pattern

3.4 PHASE MEASUREMENT

Phase measurements on transducer signals are necessary for calculating projector complex impedance and phase difference between signals received by different staves or elements in a hydrophone array. Knowledge of signal phase also is useful for analysis of sonar system power amplifiers.

To measure the phase difference between two signals, first capture the signals as described in section 3.3.3. The two signals will be projector driving current and voltage for complex impedance and will be hydrophone open-circuit output voltages for array characterization. Two alternatives for calculating the phase difference between corresponding records are described below. One method is to use the HP 5183T built-in FFT function; the other method is to transfer the raw data to the controller for processing. To get the most accurate measurements of phase, use frequencies which are multiples of the FFT frequency resolution.

The frequency resolution of an FFT depends on the sampling rate and the number of FFT points computed. For example, if the sampling rate is 100 kHz and the FFT length is 1024 (record length), then the frequency resolution is 100 kHz/1024 = 97.66 Hz. The values of phase and magnitude computed at each FFT point are analogous to those that would be obtained from a bandpass filter centered on the FFT frequency point. The bandwidth of the filter would be equal to the frequency resolution. The shape of the filter magnitude response can be changed by using various windows. Hann windows offer better frequency resolution, while uniform windows offer better amplitude accuracy. Computed phase values are most accurate when they correspond to a frequency at the center of one of these "filters". For the example above, these frequencies would be 97.66 Hz, 195.31 Hz, 292.97 Hz, etc. up to 50 kHz. The frequency resolution may be improved either by increasing the number of FFT points computed (i.e. increasing the record length) or by reducing the sampling rate. Neither one of these choices is very feasible, since record length is constrained by memory and sampling rate by signal frequency components (see Appendix A). Alternatively, the frequency spacing can be adjusted so that FFT points fall exactly on frequencies of interest. An external timebase is used to vary the sampling rate so that multiples of the FFT frequency resolution correspond to the desired frequency points. The easiest solution, however, is to restrict the frequencies at which phase is computed to those that are given by the spacing of the FFT points when the desired record length and sampling frequency are used.

Since phase measurements are required only at one frequency point of the phase spectrum (assuming signal pulses are at a single frequency) of each captured burst or record of data, an entire FFT calculation is not necessary. For cases in which a long FFT (e.g. 8K or 16K samples) is desired, it may be faster to compute a single frequency Discrete Fourier Transform (DFT) in the controller. Simply transfer a raw data record from the HP 5183T and apply the following formula:

$$X(k) = \sum_{n=0}^{N-1} x(n) * e^{-j2\pi kn/N} = \sum_{n=0}^{N-1} x(n) * [\cos(2\pi kn/N) - j\sin(2\pi kn/N)]$$

for desired k between 0 and N-1;

where N = record length; x(n) = data samples

For example: The phase of a signal at 25 kHz is desired. The signal is sampled at 100 kHz, so that there are 4 points per cycle. An 8192 point record is used (N = 8192).

$$k = \frac{25 \text{ kHz} * 8192}{100 \text{ kHz}} = 2048$$

$$X(2048) = \sum_{n=0}^{8191} x(n) * \exp \left\{ -j2\pi \frac{2048n}{8192} \right\}$$

The data samples, x(n), are put into the formula, and the result is a complex number, X(2048). Its phase is calculated from:

$$\phi = \tan^{-1} \left(\frac{\text{Im}[X(2048)]}{\text{Re}[X(2048)]} \right)$$

where Im[X(2048)] = Imaginary part of X(2048)

Re[X(2048)] = Real part of X(2048)

ϕ is the phase of the 25 kHz signal relative to the zero degree phase at DC.

Since the phase difference between two signals is desired, the above calculation is performed for the two data records of interest, and one phase is subtracted from the other.

3.5 FREQUENCY MEASUREMENTS

Sonar transducer signal analysis in the frequency domain is useful for characterizing transducer cavitation phenomena and multi-frequency (chirp) signals.

When projectors are driven in a pulsed mode at high power levels, an effect called cavitation—the formation of cavities in the testing medium—can occur. At the onset of cavitation, the projector output power versus input power characteristic becomes nonlinear, and there is a degradation in projector performance. These effects occur when the projector is driven at power levels beyond its cavitation threshold. Before cavitation occurs, projector output signal harmonics increase in level. Thus, Fourier Transform analysis may be used to monitor the harmonic content of projector current or voltage waveforms to measure the cavitation threshold. This threshold is raised by increasing the signal frequency, reducing the pulse width, or increasing the pressure on the transducer.

Sometimes sonar projectors are operated in their nonlinear range in order to achieve a desired power level. This high power operation requires the transmitted signal to be a short pulse in a low duty cycle. Since the projector is used in this nonlinear range, it must be calibrated in this range. Spectral analysis (via the FFT) of the projector driving current and voltage is then an important tool in the calibration process. The low duty cycle short pulsed signal makes it difficult to use conventional spectrum analyzers or dynamic signal analyzers. The HP 5183T Digitizing Oscilloscope easily handles such a measurement. Harmonic distortion in the digitized signal is measured using the HP 5183T built-in FFT function. The SPECTRUM dBm function in the FREQUENCY DOMAIN menu is selected to perform the FFT on a record of data which may range in length from 1024 to 16384 samples (Figure 3.18).

Spectral analysis of chirp signals also is useful. Chirp signals contain a range of frequencies, moving from lower to higher or *vice versa*, and are used so that the transmitted signal is less noticeable to other ships or submarines.

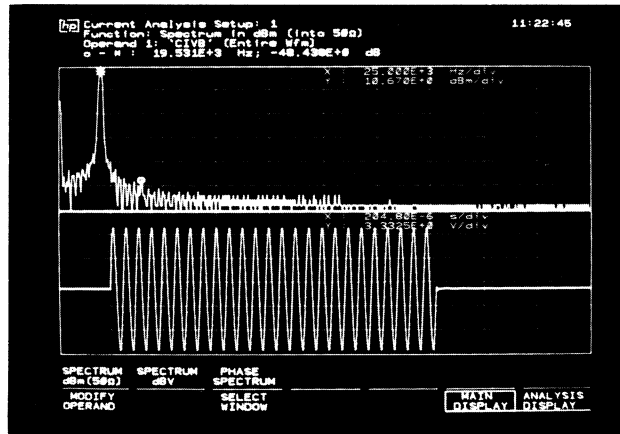


Figure 3.18 Harmonic Content of 20 kHz Projector Driving Voltage (Cursors Show Second Harmonic is 48 dB Down)

3.6 PSEUDORANDOM NOISE SOURCES

Since pseudorandom noise (PRN) contains all frequency components over a particular range, a single burst can be used to characterize completely a transducer response (Figure 3.19). The principal advantage of using a PRN source is that it reduces test time and complexity. For example, when measuring a transducer response over a 50 kHz range, data might be taken at 500 different frequencies. The HP 5183T has to store 500 records of data and process each record to produce the response. The signal source has to be programmed to step through the frequency range, producing a tone-burst at each frequency. In contrast, recording the transducer response to a single burst of PRN yields similar results with 500 times less processing and waveform storage. The HP 5183T FFT function obtains the magnitude response. The PRN should be band-limited to the frequency range of interest and should have a period greater than the duration of the pulse. The test time required to make directivity pattern measurements at multiple frequencies can be greatly reduced by using PRN bursts. By capturing the transducer response to a PRN burst at each angle, directivity plots at many different frequencies are generated simply by taking magnitude data at the selected frequency points on the FFT of the transducer response at each angle.

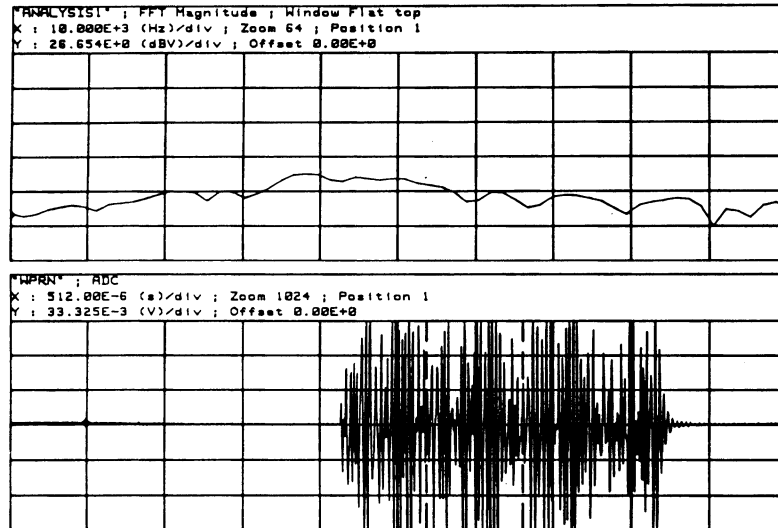


Figure 3.19 Received PRN Burst (0 to 100 kHz); 128 point FFT of samples 600 to 728 shown on top trace

Unfortunately, there are several disadvantages to using a PRN source. Even after bandlimiting the output of the PRN source to the frequency range of interest, signal power is still spread over this entire range. Consequently, signal-to-noise ratio (SNR) at any particular frequency is considerably less than a pure-tone synthesizer can achieve. Good SNR is important for such measurements as hydrophone sensitivity, since hydrophones are designed to detect signals only slightly above noise levels. Increasing the total power in the PRN signal driving the projector is not always feasible because of projector power limitations. It is beneficial to test projectors under conditions similar to actual use, and projectors are often used near full power at a single frequency. In this situation, PRN could not be used as a test signal because to get adequate signal power at one frequency, the noise power over the entire range would be at a level that would overload the projector. Another problem is that the resonance regions of transducer responses typically show large variations with change in frequency. Even a 16K-sample FFT of the PRN burst might not provide the frequency resolution required near resonance.

If the limitations involved with a PRN test source can be tolerated, it is possible to simplify the tests described in sections 3.3.1–3.3.3. Capturing only one record is sufficient. This record is the hydrophone open-circuit voltage or projector voltage or current caused by the PRN burst. The duration of the PRN burst depends on the tank size and is typically about 1 msec. If the sample rate is set to 2 MHz, then a 1024-point record captures 512 μ sec of data. As described in section 3.3.1, the trigger position is set so that steady state data are captured. The SPECTRUM dBV function gives the response voltage (equivalent to RMS) in decibels over the range DC to 1 MHz. If the PRN was bandlimited to the range of interest (e.g. 50 to 100 kHz), then the spectrum should show small magnitude outside this range. To get better frequency resolution in the range of interest, the sample rate should be reduced as described in sections 3.3.3. and 3.4. For complex impedance measurements (section 3.3.3), subtracting the SPECTRUM dBV of channel 2 (current) from that of channel 1 (voltage) yields impedance magnitude, and subtracting the PHASE SPECTRUM plots produces the impedance phase.

The test system shown previously in Figure 3.3 can generate either tone-burst or PRN signals. In addition, any other waveform of interest can be generated.

4.0 Sonar System Measurements

4.1 PROJECTOR POWER AMPLIFIERS

Sonar system projectors are often required to operate at high power levels in order to achieve sufficient transmission range and signal-to-noise ratio. A typical low power amplifier operates over a frequency range of 50 Hz to 300 kHz with less than 1% distortion, and a typical high power amplifier provides up to 10 kW in pulsed modes from 50 Hz to 30 kHz with less than 2% distortion.

Measurements of output signal level, input to output signal phase shift, and output signal distortion are useful for characterizing power amplifier performance. The output power level at desired frequencies is measured by recording attenuated versions of the amplifier voltage and current outputs, taking the RMS values of these two waveforms, and then multiplying them together. The procedure is the same as that for measuring complex impedance, except that the HP 5183T built-in multiplication function is used. Input to output signal phase shift is measured analogously to the phase measurement procedure described in section 3.4. The voltage waveform to the input of the power amplifier is recorded on channel 1, and the voltage waveform from the output of the power amplifier is recorded on channel 2. The FFT is used to compute phase difference as described in section 3.4. Finally, distortion of the power amplifier output signal is measured by performing an FFT on the recorded signal. The HP 5183T's SPECTRUM dBm function computes the power in harmonics up to 2 MHz when the digitizer is sampling at the maximum rate of 4 MHz.

The HP 5183T can conveniently characterize the power amplifier overload and shutdown process. This measurement involves ramping up the amplifier output level until overload circuitry automatically turns off the power. Because it is desirable to trigger on this turn off event, conventional triggering techniques will not work for this measurement. However, the DROPOUT trigger feature of the HP 5183T solves this problem. The trigger level is set slightly above the level that the signal will be at after the turn-off occurs. The dropout length (in seconds) is set so that it is longer than several cycles of the signal. Thus, while the output power ramps up, the sinusoidal signal is never below the trigger level long enough to cause a trigger. When the shutdown occurs, the signal drops below this trigger level for an extended duration, and the HP 5183T triggers. The amount of signal data captured before, during, and after the shutdown depends on the trigger position and record length (Figure 4.1).

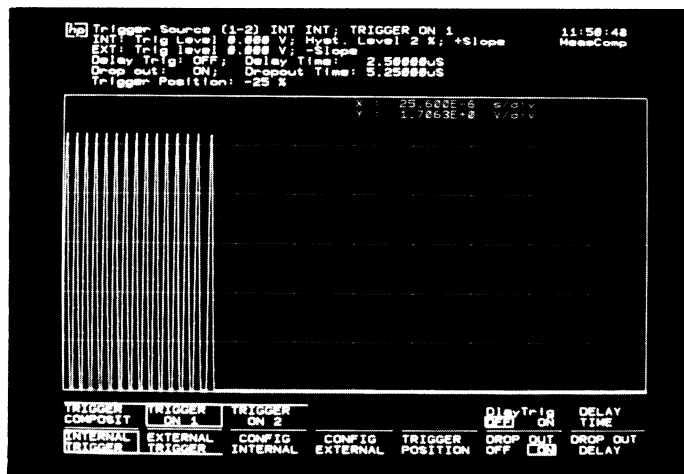


Figure 4.1 Dropout Trigger on Sinusoidal Waveform

4.2 SONOBUOYS

A sonobuoy is a portable, expendable sonar device that provides information about underwater objects such as submarines. Although there are both active and passive sonobuoys, passive sonobuoys are used most often since they do not transmit acoustic signals which would reveal their presence. Usually dropped from an aircraft, the sonobuoy receives sound signals via a submerged hydrophone attached to the buoy and then transmits the signals back to the aircraft. A VHF FM link might be used for sonobuoy transmissions to the aircraft. Sound signals to be received by the sonobuoy are sometimes generated by an explosive charge that is also dropped from the aircraft.

This procedure, called explosive echo ranging, is used with passive sonobuoys to obtain more detailed target direction and range information.

An alternative to using a passive sonobuoy and explosive echo ranging is to use an active sonobuoy which has a projector attached to it. Active sonobuoys are commanded via a UHF AM link from an aircraft (Figure 4.2). After these sonobuoys have been addressed, they are instructed to transmit specific acoustic signals.

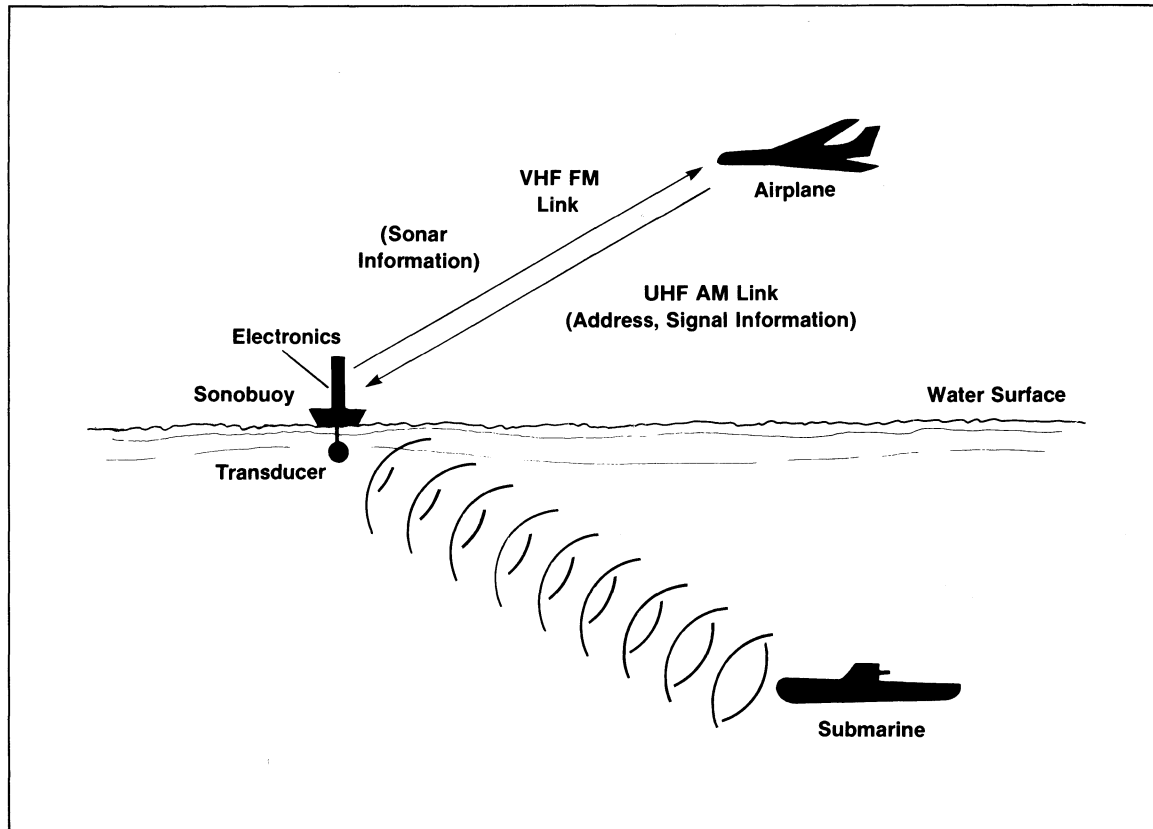


Figure 4.2 Active Sonobuoy

Testing of the signal which is used to address the sonobuoy is an ideal application for the HP 5183T. Each active sonobuoy is set to respond to particular tone-pairs (sums of two frequencies) which are transmitted over the UHF AM link from the aircraft. The address signal includes three tone-pairs and may be up to 1.5 seconds long. Amplitude, frequency, and duration of the tone-pairs are measured. The highest frequency component is about 50 kHz. This signal is completely characterized by digitizing it with the HP 5183T as follows:

- (1) Input: Set the input characteristics of HP 5183T CHANNEL 1 according to the signal level, offset, and desired input coupling.
- (2) Memory: Use only one record. With memory option 512, this will provide space for 524,288 samples.
- (3) Recording Mode: CAPTURE
- (4) Timebase: Set MAIN RATE to 4 μ sec (250 kHz sample rate). Set ASR (Adaptive Sample Rate)—ASR switches the sampling rate between the rate set and 64 times less than the rate set, depending on the frequency of the incoming signal. Complete time-interval accuracy is maintained. In addition, set the ASR NOISE REJECT to allow the ASR circuitry to reject signals below a specified energy level. Setting ASR will optimize the use of memory by switching the sample rate to 64 times less than 250 kHz when there is no signal present. Thus, the dead-time between tone-pairs will be recorded at a sample rate of only 3906.25 Hz. For

example, 0.75 seconds of dead-time in a 1.5 second address signal will occupy only about 3000 samples. The other 0.75 seconds of signal information (sampled at 250 kHz) will occupy 187,500 samples. Note that BURST mode would not conserve memory as well as ASR for this application since the duration of the tone-pairs is variable, and the number of samples per BURST would have to be set to capture the longest possible tone-pair.

- (5) Trigger: INTERNAL CHANNEL 1
Set the trigger LEVEL, HYSTERESIS, and POSITION according to the signal.
- (6) Sweep: SINGLE mode

After the signal has been captured, CURSORS X and O make some simple measurements. The duration of each tone-pair is measured by positioning the CURSORS at the beginning and end of the tone-pair and pressing the CURSOR DELTA softkey. Peak voltage values also are measured with the CURSORS, and RMS voltages are calculated as described previously in this application note. The FFT obtains frequency component amplitude information. The MODIFY OPERAND feature delimits portions (such as a 1024-point section of a captured tone-pair record) of the signal for the FFT. An easy way to increase the frequency resolution is to increase the number of FFT points computed (see section 3.4). For example, to get 15 Hz resolution with a 250 kHz sample rate, a 16,384-point FFT is computed.

4.3 ARRAY GAIN

One of the main advantages of using a transducer array instead of a single transducer is the improvement in signal-to-noise ratio that is obtained. The array gain is calculated to measure this improvement in signal-to-noise ratio and is defined (in decibels) as:

$$\text{Array Gain} = AG = 10 * \log \frac{(S/N)_{\text{array}}}{(S/N)_{\text{one element}}}$$

where S/N = signal-to-noise power ratio

A convenient formula for calculating the array gain can be used if the array output is derived from the series or parallel sum of the transducer element outputs (the following equations are from Urick [11]).

Equation 4.1:

$$AG = 10 * \log \frac{\overline{S_a^2} / \overline{N_a^2}}{s^2 / n^2} = 10 * \log \frac{\sum_i \sum_j (\rho_s)_{ij}}{\sum_i \sum_j (\rho_n)_{ij}}$$

To compute the array gain for an amplitude shaded array, the formula is modified slightly:

$$AG = 10 * \log \frac{\sum_i \sum_j a_i a_j (\rho_s)_{ij}}{\sum_i \sum_j a_i a_j (\rho_n)_{ij}}$$

Variable definitions are as follows:

$\overline{S_a^2}$ = average signal power at the array terminals

$\overline{N_a^2}$ = average noise power at the array terminals

s^2 = average signal power from one array element

n^2 = average noise power from one array element

a_i = rms voltage produced by the ith element due to the signal or noise

$(\rho_s)_{ij}$ = signal crosscorrelation coefficient between the ith and jth array elements

$(\rho_n)_{ij}$ = noise crosscorrelation coefficient between the ith and jth array elements

The crosscorrelation coefficient is calculated from:

$$\rho_{ij} = \frac{\overline{v_i(t) * v_j(t)}}{\{\overline{v_i^2(t)}\}^{1/2} * \{\overline{v_j^2(t)}\}^{1/2}}$$

where the bar indicates a time average, the denominator is a normalization factor, and v is a hydrophone output voltage produced by either a signal or noise.

When designing an array or experimenting with different delay networks among array elements (for beam-forming), it is useful to be able to calculate array gain values. These values are computed from equation 4.1 by calculating crosscorrelation coefficients for various pairs of array element hydrophones or separate hydrophones. A crosscorrelation coefficient varies between +1 and -1 and is a measure of the similarity between signals $v_i(t)$ and $v_j(t)$. Thus, for $\rho = +1$ the signals are the same, for $\rho = 0$ the signals are uncorrelated, and for $\rho = -1$ the signals have the same shape but are 180 degrees out of phase. The coefficients depend on the signal and noise fields which surround the array and on time delays inserted among the array elements to steer the array beam or response. Note that the array gain formula reduces to the directivity index formula when the signal is perfectly coherent (unidirectional plane wave) and the noise is isotropic.

$$\text{Directivity Index} = DI = 10 * \log \frac{4\pi}{2\pi \int_0^{2\pi} \int_{-\pi/2}^{\pi/2} b(\theta, \phi) * \cos \theta d\theta d\phi}$$

where $b(\theta, \phi)$ = directivity response

Using the HP 5183T, crosscorrelation coefficient values for array gain calculations are obtained as follows:

- | | |
|---------------------|--|
| (1) Input: | Set the input amplifier RANGE, OFFSET, and COUPLING according to the signals to be recorded. Set to simultaneously record the open-circuit voltages from two different hydrophones or array elements on CHANNEL 1 and CHANNEL 2. |
| (2) Memory: | RECORD LENGTH = 1024 samples |
| (3) Recording Mode: | CAPTURE (or AUTO ADVANCE for acquiring more than one record) |
| (4) Timebase: | Set as described in section 3.3.1. |
| (5) Trigger: | See section 3.3.1. |
| (6) Sweep: | SINGLE (or NORMAL if using AUTO ADVANCE) |

After capturing two records of data simultaneously, the crosscorrelation coefficient is computed.

Equation 4.2:

$$\rho_{xy} = \frac{\frac{1}{N} \sum_{n=0}^{N-1} x(n) * y(n)}{\left\{ \frac{1}{N} \sum_{n=0}^{N-1} x^2(n) \right\}^{1/2} * \left\{ \frac{1}{N} \sum_{n=0}^{N-1} y^2(n) \right\}^{1/2}}$$

n = sample number

N = record length = 1024

Let the waveform in CHANNEL 1 RECORD 1 be $x(n)$ and the waveform in CHANNEL 2 RECORD 1 be $y(n)$. The waveform MULTIPLY function is used to get $x(n)*y(n)$. This $x(n)*y(n)$ waveform is then transferred to the controller so that a time average over all N samples can be computed to give the numerator of equation 4.2. After the RMS function in the HP 5183T has been used to compute the RMS values of waveforms $x(n)$ and $y(n)$, these two values are multiplied together in the controller to produce the denominator of equation 4.2.

To compute actual array gain values, records are captured for all possible pairs of elements. The crosscorrelation coefficient values computed by the above procedure are then summed in the array gain formula (Equation 4.1). To see the effects of time shifts between certain pairs of elements (e.g. finding the time shift which gives the maximum ρ), the $y(n)$ sequence is shifted by m samples ($0 \leq m \leq N-1$) in the controller. Then,

$$\rho_{xy(m)} = \frac{\frac{1}{N} \sum_{n=0}^{N-1} x(n) * y(n+m)}{\left\{ \frac{1}{N} \sum_{n=0}^{N-1} x^2(n) \right\}^{1/2} * \left\{ \frac{1}{N} \sum_{n=0}^{N-1} y^2(n+m) \right\}^{1/2}} \quad 0 \leq m \leq N-1$$

The numerator is the average circular crosscorrelation between $x(n)$ and $y(n)$, and the denominator is a normalization factor. Array gain values corresponding to these new m -sample time shift crosscorrelation coefficients are now computed as before.

5.0 References

- [1] "Sonar Transducer Calibration", Application Note 205-2, Hewlett-Packard Co., USA, Sept. 1979.
- [2] Oppenheim, Alan V., editor, APPLICATIONS OF DIGITAL SIGNAL PROCESSING, Prentice-Hall, Inc., Englewood Cliffs, N.J., 1978.
- [3] Wallace, J.D., and E.W. McMorrow: Sonar Transducer Pulse Calibration System, Journal of the Acoustical Society of America (JASA), vol. 33, Jan. 1961, p. 75.
- [4] Green, C.E.: Anechoic Sonar Test Pool (Abstract), JASA, vol. 32, 1960, p. 1519.
- [5] Bobber, Robert J., UNDERWATER ELECTROACOUSTIC MEASUREMENTS, Naval Research Laboratory, Washington, D.C., July, 1970.
- [6] Procedures for Calibration of Underwater Electroacoustic Transducers, S1.20-1972, American National Standards Institute, New York, 1972.
- [7] Example Sonar Transducer Test Programs—available from Hewlett-Packard Santa Clara Division—contact application note author Bill Abbott—Telephone: (408) 246-4300.
- [8] Baker, D.D.: Determination of Far-Field Characteristics of Large Underwater Sound Transducers from Near-Field Measurements, JASA, vol. 34, 1962, p. 1737.
- [9] Horton, C.W., and G.S. Innis: The Computation of Far-Field Radiation Patterns from Measurements Made near the Source, JASA, vol. 33, 1961, p. 877.
- [10] Baker, D.D., and K. McCormack: Computation of Far-Field Characteristics of a Transducer from Near-Field Measurements Made in a Reflective Tank, JASA, vol. 35, 1963, p. 736.
- [11] Urick, Robert J., PRINCIPLES OF UNDERWATER SOUND, 3rd edition, McGraw-Hill Inc., USA, 1983.
- [12] Oppenheim, A.V. and R.W. Schaffer, DIGITAL SIGNAL PROCESSING, Prentice-Hall, Inc., Englewood Cliffs, New Jersey, 1975.

Appendices

A. DIGITIZING ANALOG SIGNALS

Most digital signal processing (DSP) systems may be represented by the diagram shown in Figure A.1.

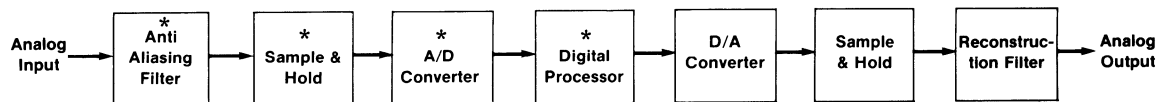


Figure A.1 Digital Signal Processing System

An asterisk in a block in the diagram indicates the functions that are performed by the HP 5183T. An important parameter that must be specified when using the HP 5183T is the sampling or digitizing rate. The period of this rate is set with the MAIN RATE softkey and takes values of $N \cdot 250$ nsec, where N is an integer in the range $1 \leq N \leq 4E+6$. Thus, the maximum sampling rate is 4 MHz.

In any DSP system, there is a minimum rate at which the analog signal must be sampled to avoid loss of information [12]. This minimum allowable sampling rate is called the Nyquist rate and is specified by the Sampling Theorem which may be summarized as follows:

When digitizing an analog signal, the sampling rate must be at least twice as great as the highest frequency component (f_0) in the spectrum of the signal that is being sampled.

Frequency components higher than f_0 will “alias” down into the frequency range below f_0 and will interfere with the original signal. For example, since a square wave can be represented as an infinite sum of sinusoids (Fourier Series) and thus contains very high frequency components, attempting to digitize this signal without an anti-alias filter on the input will result in severe aliasing in the captured signal. The anti-alias filter in the HP 5183T has a 1 MHz cutoff frequency (the filter response is 4dB down at 1 MHz) and may be either enabled or disabled. If the filter is disabled, the input amplifier bandwidth is down only 1 dB at 1 MHz and 2 dB at 3 MHz.

For most sonar applications, the 1 MHz bandwidth provided with the anti-alias filter is adequate. The sampling frequency usually should be set several times greater than the highest frequency component in the signal being measured. If the test signal is a pure sine wave, the sampling frequency may be set closer to the theoretical Nyquist rate. An easy way of checking the frequency content in the test signal is to sample at the maximum rate of 4 MHz and then use the HP 5183T’s SPECTRUM dBm function. This will show the power in test signal frequency components up to 2 MHz.

B. CALIBRATION METHODS

Examples of two basic methods for determining hydrophone and projector responses are described in this Appendix. As stated in section 3.3.1, complete descriptions of standard calibration procedures are given in reference [6].

B.1 Hydrophones

A comparison method (or substitution method) calibration is performed by recording the responses of a reference hydrophone and a test hydrophone when each is subjected to the same sound waves produced by a projector. The test hydrophone response is derived from a comparison with the reference response. The projector characteristics are not important as long as the right frequency is produced and the signal is strong enough. Usually, the two hydrophone responses are measured one at a time. The reference hydrophone is put into the test environment first and then is substituted with the test hydrophone. The measurement conditions should be the same for each hydrophone. In addition, the hydrophones should be as similar as possible to each other in size, shape, and design. It is important that the distance between the projector and hydrophone be great enough to cause the spherical sound waves transmitted from the projector to be received as plane waves by the hydrophone.

Once the open-circuit voltages of both the test and the reference hydrophones have been measured, the test hydrophone's voltage response is determined from the following equation:

$$M_x = (M_s * e_x) / e_s \quad \text{where} \quad \begin{array}{l} M_s = \text{voltage response of the reference (standard) hydrophone} \\ M_x = \text{voltage response of the test hydrophone} \\ e_x = \text{open-circuit output voltage of the test hydrophone} \\ e_s = \text{open-circuit output voltage of the reference hydrophone} \end{array}$$

The test hydrophone response also may be determined from known projector characteristics. After the test hydrophone's open-circuit voltage has been measured, its voltage response is calculated from the following equation:

$$M_x = (e_x * d) / (i_s * S_s) \quad \text{where} \quad \begin{array}{l} e_x = \text{open-circuit output voltage of the test hydrophone} \\ d = \text{distance in meters between the projector and the hydrophone} \\ i_s = \text{current driving the projector} \\ S_s = \text{transmitting current response} \end{array}$$

Although the comparison calibration method is straightforward and relatively easy to implement, it is prone to a variety of errors. Causes of these measurement inaccuracies include poor signal-to-noise ratio and free-field conditions, unstable reference hydrophones, and only approximate open-circuit conditions. It is sometimes difficult to achieve high enough signal-to-noise ratios to ensure reliable calibration data, since some hydrophones are designed to detect signals at the same level as noise. Free-field conditions are degraded by reflections from the bottom, surface, or sides of test tanks or lakes, and by impurities in the water. Another disadvantage of the comparison method is the requirement for a reference standard transducer. The reciprocity method does not have this disadvantage.

The reciprocity calibration method requires at least a reciprocal transducer in addition to the test transducer, unless the test transducer itself is reciprocal. If the test transducer is reciprocal, then it is possible to perform a self-reciprocity calibration, and no other transducers are required. For a transducer to be reciprocal, it must be linear, passive, and reversible. These requirements are necessary, but they are not sufficient since a transducer possessing them may not be reciprocal. There are a number of different techniques for performing a reciprocity calibration [5, 6], consequently only the basic principle is described below.

The behavior of a reciprocal transducer obeys the following equation:

$$M/S = J \quad \text{where} \quad \begin{array}{l} M = \text{voltage/pressure} = \text{transducer receiving response} \\ S = \text{pressure/current} = \text{transducer transmitting current response} \\ J = \text{reciprocity parameter} \end{array}$$

Depending on whether the test situation involves plane, spherical, or cylindrical waves, the reciprocity parameter J is calculated according to one of the three forms shown below:

$$\text{spherical: } J_s = 2d/\rho f$$

$$\text{cylindrical: } J_c = \{2L/\rho c\} * \sqrt{cd/f}$$

$$\text{plane: } J_p = 2A/\rho c$$

where (MKS units)

d = distance between two transducers

ρ = density of the water

f = frequency

L = length of cylindrical hydrophone

c = speed of sound in the water

A = surface area of transducer affected by plane wave

Thus, if a transducer is reciprocal and M or S, J, and the type of sound waves are known, then the other response that is not known can be calculated.

Using three transducers—a hydrophone, a projector, and a reciprocal transducer—the hydrophone or projector response is determined from a series of voltage measurements and the driving current, the appropriate reciprocity parameter, and the distance between the transducers. This three-transducer method was one of the first reciprocity methods to be used. Details on exactly how to use this calibration method are described in three of the references [5, 6, 11]. Reciprocity calibration of transducer arrays is more difficult and usually requires the use of near-field techniques [8, 9, 10].

B.2 Projectors

The comparison method for calibrating projectors is similar to the one for calibrating hydrophones. Either the transmitting current response or voltage response may be determined, but measurement of the transmitting current response is more common since it is reciprocally related to the receiving voltage sensitivity. The comparison method for projectors is not used as often as it is for hydrophones since it is more difficult to get a standard projector that is similar to a particular test projector. Because standard projectors are larger than hydrophones, they are more subject to diffraction effects and spurious resonances. In addition, projectors can be nonlinear and unstable at or near resonant frequencies. However, some projectors are very stable away from resonant frequencies and are used for comparison calibrations. The procedure is as follows:

- (1) Place the projector in a free-field.
- (2) Drive the projector with reference current i_x or reference voltage e_x .
- (3) Place a standard hydrophone (with known response M_s) at a distance d meters from the projector on the projector acoustic axis.
- (4) Measure the open-circuit voltage (e_s) at the hydrophone output terminals caused by the sound emitted from the projector.
- (5) Then, the transmitting current response of the projector is:

$$S_{i_x} = (e_s * d) / (M_s * i_x)$$

The transmitting voltage response is: $S_{v_x} = (e_s * d) / (M_s * e_x)$

Another way to do the comparison calibration is essentially the same as the hydrophone method. The procedure is as follows:

- (1) Record the response of a hydrophone (unknown M) to a standard projector transmission.
- (2) Replace the standard projector with the test projector (to be calibrated) and adjust the driving current or voltage until the hydrophone response is the same as before.
- (3) Calculate the test projector transmitting current or voltage response as follows:

$$S_{i_x} = (S_{i_s} * i_s) / i_x \quad \text{where} \quad \begin{array}{l} S_{i_s} = \text{standard projector transmitting current} \\ \text{response} \\ S_{v_s} = \text{standard projector transmitting voltage} \\ \text{response} \\ i_s = \text{standard projector driving current} \\ i_x = \text{test projector driving current} \\ e_s = \text{standard projector driving voltage} \\ e_x = \text{test projector driving voltage} \end{array}$$

To avoid the requirement for a standard transducer, different calibration methods such as the reciprocity method are used.

The reciprocity method discussed in section B.1 applies to projectors as well as to hydrophones. The same measurements are made, but different equations are used to derive the projector response. Since most projectors are reciprocal, their transmitting current response, S , can be obtained by calibrating their receiving response, M , and using the appropriate reciprocity parameter, J , in the equation: $S = M/J$.

C. TRIGGERING

While digitizing received signal bursts using the NORMAL sweep mode or BURST timebase, reflections from a previous signal burst could trigger the HP 5183T before the desired trigger point on the next signal burst. Listed below are several ways of solving this problem. This problem does not exist if reflections never reach the same amplitude as the initial received burst (i.e. the trigger level could be set high enough to prevent triggering on reflections), or if it is not necessary to capture successive memory records (e.g. when using noise as a test signal, only one burst is necessary for complete characterization).

- (1) Use SINGLE sweep mode with AUTO ADVANCE. The disadvantage of this approach is that the controller must send the START command before each record is captured.
- (2) If projector driving voltage or current is being recorded simultaneously on another channel, use this signal as the trigger. Both channels start recording as soon as the trigger is received. MIXED timebase or ASR timebase is used to prevent waste of memory during the delay between transmitted and received bursts (see Figure C.1).
- (3) If only the received hydrophone voltage is being recorded, either one of the following methods provides a convenient trigger for the HP 5183T.
 - A) Drive the EXTERNAL trigger input with the transmitted burst. This will work since transitions on the EXTERNAL trigger will always stop before capture of the record is complete. TRIGGER POSITION is set so that recording starts during the steady state portion of the received burst.
 - B) If the HP 5182A is the signal source, use its SYNC pulse to drive the HP 5183T EXTERNAL trigger input.
- (4) If BURST timebase is selected, use DELAY trigger. This special trigger mode overcomes the limitation of not being able to set a TRIGGER POSITION (In BURST timebase, trigger position is fixed at 0%). In DELAY trigger mode, the HP 5183T records data only after the following events occur. The transmit burst produces an EXTERNAL trigger. Then, after the delay time (which is set by the user) passes, the instrument waits for an INTERNAL trigger from the received signal to start recording the specified number of samples.

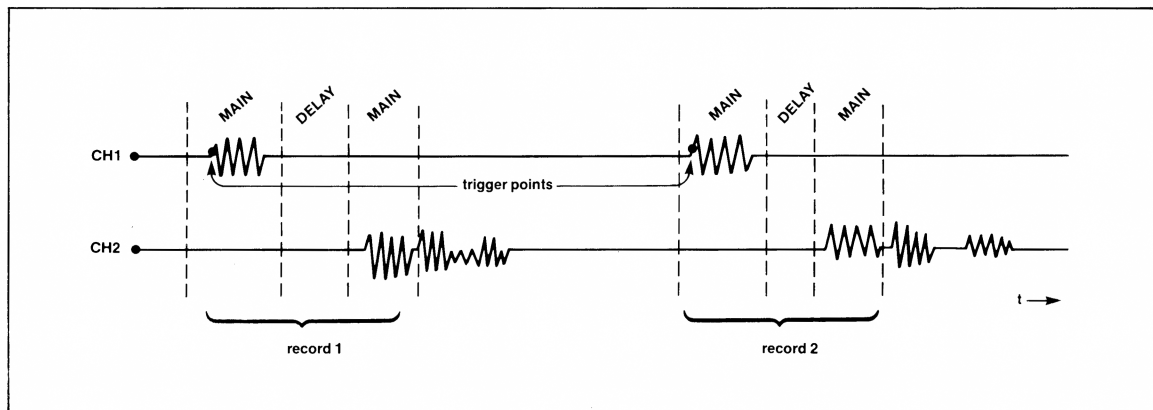


Figure C.1 CH 1 = transmit burst
 CH 2 = receive burst
 TRIGGER on INTERNAL CHANNEL 1
 MIXED timebase is shown with DELAY rate set so that only a few samples are taken between bursts.



For more information, call your local HP sales office listed in the telephone directory white pages. Ask for the Electronic Instrument Department, or write to Hewlett-Packard: **U.S.A.** - P.O. Box 10301, Palo Alto, CA 94303-0890. **Europe** - P.O. Box 999, 1180 AZ Amstelveen, The Netherlands. **Canada** - 6877 Goreway Drive, Mississauga, L4V 1M8, Ontario. **Japan** - Yokogawa-Hewlett-Packard Ltd., 3-29-21, Takaido-Higashi, Suginami-ku, Tokyo 168. **Far East** - Hewlett-Packard Asia Headquarters, 47/F China Resources Building, 26 Harbour Road, Wanchai Hong Kong. **Australasia** - Hewlett-Packard Australia Ltd., 31-41 Joseph Street, Blackburn, Victoria 3130 Australia. **Latin America** - Hewlett-Packard Latin America Headquarters, 3495 Deer Creek Rd., Palo Alto, CA 94304. For all other areas, please write to: Hewlett-Packard Intercontinental Headquarters, 3495 Deer Creek Rd., Palo Alto, CA 94304.

02-5952-7805

PRINTED IN U.S.A.