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A Digital Measuring System for Hydrophone Calibration

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Abstract

A computer-controlled measuring system that combines measurement and computation tasks to speed hydrophone calibration has been developed. The system incorporates one of the popular minicomputers now available. Specifications and developmental problems concerning the related hardware equipment are discussed. The small (4K) core memory size dictated specific approaches to programmed operation. Several modes of operation that meet the initial goals have been provided to make the system as flexible as possible. Programming requirements of these operational modes are reviewed.

Problem Status

This is an interim report on one phase of the problem; work is continuing on this and other phases.

Problem Authorization

NRL Problem S02-30

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A DIGITAL MEASURING SYSTEM FOR HYDROPHONE CALIBRATION

Introduction

Measurements in underwater acoustical experiments depend ultimately on the calibration of a standard reference hydrophone. Such a calibrated hydrophone is used to establish the absolute level of sound pressure (or some other characteristic) of the sound field. In turn, the standard reference hydrophone must be calibrated by one of the accepted primary or secondary procedures [1]. These calibration procedures are based on a series of electrical measurements made as a function of frequency through the range of interest. Traditionally, the measurements have been made at USRD by means of an analog system called an ATMS (Automatic Transmission Measuring Set). Signal levels processed by these sets have a wide dynamic range and are recorded on strip charts with a logarithmically scaled ordinate (decibels) and with frequency as the abscissa. The ordinate range of the chart is 50 dB with the absolute level adjustable in 10-dB steps by a separate gain control.

In the comparison calibration procedure, the sensitivity of the unknown hydrophone is determined at each discrete frequency of interest as the algebraic sum of values scaled from the recorded curves, usually two voltage curves. The voltage must be the equivalent open-circuit voltage as measured by a sufficiently high impedance.

In a reciprocity calibration, the basic measurements still are electrical ones (recorded voltage and current curves), but the computations require scaling a larger number of recorded curves--typically, ten or twelve.

In fulfillment of its mission, the USRD regularly calibrates hundreds of standard transducers that represent a variety of designs to meet different requirements. Many hundreds of manhours are dedicated to the tedious, repetitive, and time-consuming task of scaling the recorded charts to produce the needed calibration curves.

The appearance of minicomputers on the market stimulated thinking in terms of design objectives for a digital measuring system (DMS) that would by-pass or eliminate most of the time-consuming operations of the data-reduction processes. Early design objectives related to decreasing the time lag for data reduction, eliminating computational errors inherent in the manual data-reduction method, and more frequent calibration of standards, thereby providing statistical data for performance histories of individual standard hydrophones.

A set of initial performance specifications for the system evolved from the design objectives as follows:

- a. Operate in the frequency range 30 Hz to 150 kHz.
- b. Analog measurement accuracy ± 0.1 dB.
- c. Dynamic range of 50 dB for a single measurement.
- d. Produce a comparison or reciprocity calibration within 5-10 minutes after completing measurements.

System Hardware Description

Some sections of the digital measuring system are essentially identical to those of the early sweep-frequency analog systems. A simplified block diagram of the digital measuring system is shown in Fig. 1. The receive section, the tracking filter, and the transmit section are common to the two types of measuring systems. Of the remaining sections, the computer is used for central processing and control, and the input-output peripherals are used for storage and communication. Each section is described in detail in the following paragraphs.

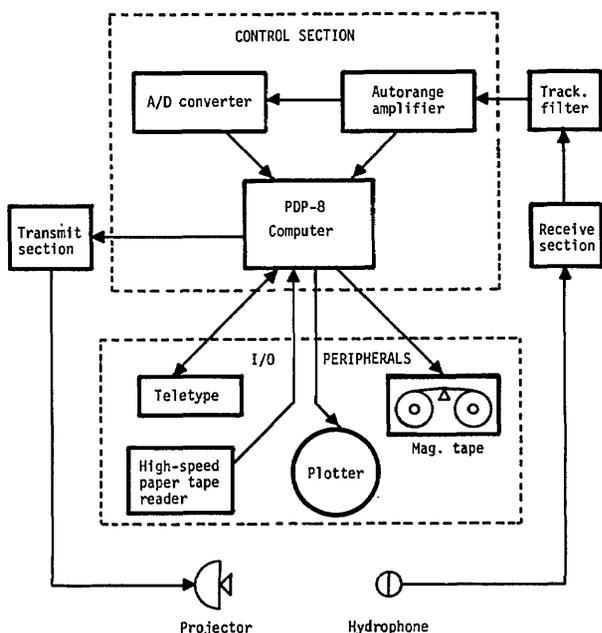


Fig. 1. Simplified block diagram of system.

Transmit Section

The nucleus of the transmit section is a frequency synthesizer, which is used because of its wide frequency range, its high stability, and its natural suitability for digital control. The output frequency of the synthesizer is controlled through an interface by signals from the computer. The desired frequency is derived first in the computer in floating-point form.¹ For output to the interface, the three-word floating-point number is converted to a binary-coded decimal (BCD) number having six decimal digits in two 12-bit words. When the new frequency is desired, the two words are transferred into two 12-bit registers in the interface.

When the command to change frequency is given, these registers are transferred to a 24-bit register where the BCD code is used to control a NIXIE-tube frequency display. The 24-bit BCD register output also is

¹Details of the various computer-oriented numbering systems are given in the section "Operation Under Programmed Control" and in Appendix A.

coded into six decimal outputs that are used to select the synthesizer frequency. The 24-bit buffer is used to hold the current frequency while a new frequency is being set up in the two 12-bit registers.

The remainder of the transmit system is similar to the older sweep-frequency analog systems, with manual attenuators, power amplifier, current- and voltage-monitoring networks, and dummy loads completing the list of equipment used. A block diagram of the transmit section is shown in Fig. 2.

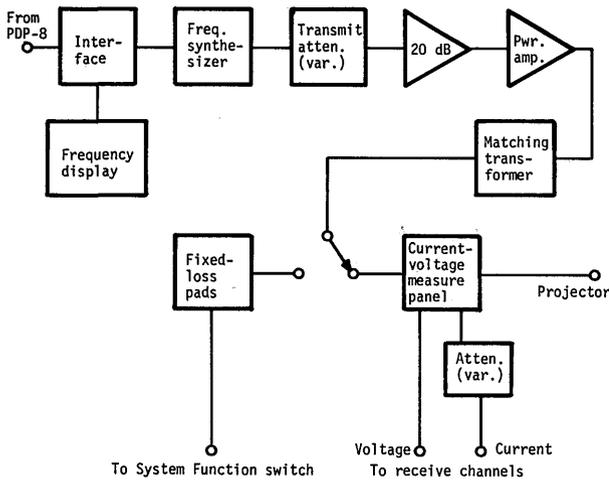


Fig. 2. Block diagram of transmit section.

for switching a filter into the channel, but only one filter is provided with the system. It is an off-the-shelf commercial tracking filter and usually is used in the response measurement channel. Bandwidths of 20 Hz, 100 Hz, and 200 Hz are selectable by a front-panel switch. The system's upper-frequency limit is reduced from 150 kHz to 50 kHz when the filter is used.

Control Section

The control section consists of four channels of autoranging amplifiers, an analog-to-digital converter, and the PDP-8 computer.

The analog-to-digital converter, a Digital Equipment Corp. model AF-01, produces a binary number that represents the magnitude of the d-c voltage at its input. The binary number is compatible with the number system employed by the computer. Word lengths of from 6 to 12 bits can be selected by a front-panel switch.

The accuracy and the time required for conversion depend upon the word length used. A table listing these characteristics can be found in reference [2]. The switching point error includes effects of nonlinearity and temperature. The over-all conversion error equals switching-point

Receive Section

The receive section is almost a direct supersedure of that in the analog systems, except that it contains four identical channels. These four channels allow the simultaneous measurement of four variables. The discrete channels are carried through to the input of the analog-to-digital converter, where a multiplexing switch allows them to be converted in sequential order.

Tracking Filter

Provision is made in each of the four receive channels

error plus a quantization error of plus or minus one half the digital value of the least significant bit (LSB). The digital value of the LSB for a 12-bit word is approximately 2.5 mV, which is found from the equation

$$DV_{\text{LSB}} = E/(2^n - 1) = (10 \text{ V})/(2^{12} - 1) = 2.5 \text{ mV},$$

where E is the full-scale input voltage and n is the word length. For a 12-bit word, then, $\frac{1}{2}\text{LSB} = 1.25 \text{ mV}$. This represents an error of 0.0125% to be added to the switching-point error. The over-all error could be 0.0375%, if the input voltage to the converter is the full-scale value 10 V. A signal level of 1 V to the input of the converter could produce a readout that would be in error by as much as 0.375%. Obviously, as the input level to the converter decreases, the accuracy of the result decreases. To obtain the greatest possible conversion accuracy, then, the input voltage to the converter must be kept in the range -1 to +10 V.

A block diagram of the autorange amplifier is shown in Fig. 3. The autorange amplifier in each channel automatically keeps the input voltage to the analog-to-digital converter within the -1 to +10 V range. The autorange amplifier contains an ac/dc converter that is a peak detector. This type of circuit has a much faster settling time than an averaging detector. The detector settling time is the limiting factor in the operating speed of the system (except when the tracking filter is used in its narrowest bandwidth position, in which case, it determines the rate of frequency switching that can be used).

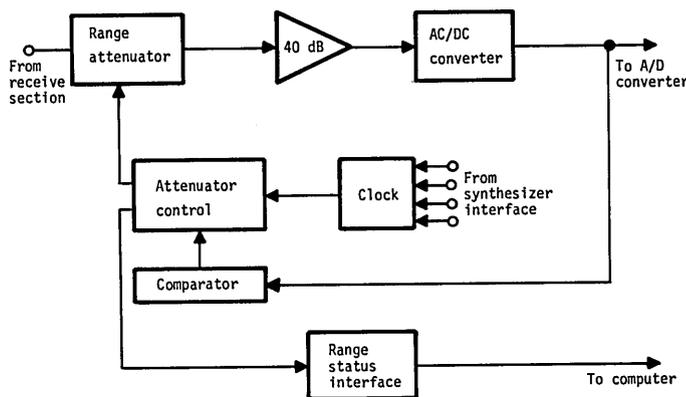


Fig. 3. Block diagram of autorange amplifier.

The input signal to the autorange amplifier is applied first to the range attenuator, which attenuates in four 10-dB steps from 0 to 40 dB. The signal is amplified and then converted to a d-c voltage. The converted output is applied to the analog-to-digital converter. This output is also monitored by a voltage comparator where it is compared with upper and lower threshold voltages. If the output voltage falls outside these limits, the comparator opens a gate in the attenuator control. This gate allows a clock pulse from a clock common to all four receive channels to switch the range attenuator to a new position and in the proper direction to bring the output within the threshold limits.

Sufficient time must be allowed for the peak detector to reach a new value following a range change. The fastest clock rate is 1000 pps,

which provides adequate time for the detector to settle whenever the signal frequency is 1 kHz or above. When the signal frequency is less than 100 Hz, the clock rate is switched to 10 pps. The rate 100 pps is used when the signal frequency is between 100 and 1000 Hz. It can be seen that the signal frequency determines to some extent how fast a range change can occur.

The over-all gain of each autorange amplifier channel is displayed on the autorange amplifier monitor panel (Fig. 4). This panel contains the following indicator and control functions:

- a. Range state display lights for each 10-dB step as well as "too high" and "too low" indicators for out-of-range conditions for each channel.
- b. On-off switch for each channel.
- c. Flag indicator light for each channel that indicates when the output is within the proper voltage range for the analog-to-digital converter.



Fig. 4. Receive section, input-output peripherals, and control section.

- d. A device flag light that indicates when all four channels have satisfactory outputs.
- e. Clock rate lights that indicate the slow, medium, or fast clock rate used for range changes.
- f. A manual range change pushbutton that can be used when setting up measurements to obtain a range change while the computer is halted.

The display monitor panel (Fig. 4) contains a rotary switch that is used to select any of the four input channels for display on the monitor oscilloscope. It also contains a lever switch for manually clearing the peak detectors of the autorange amplifiers during setting up operations. This switch is used in conjunction with the manual range-changing pushbutton on the autorange amplifier monitor panel. Another rotary switch is used to divide the basic clock rate supplied to the autorange amplifiers to accommodate the increased response time when narrow bandwidths are used in the tracking filter.

Input-Output Peripherals

The storage and communication devices used with the system are:

1. Teletype Corp. Model 33 ASR (Teletypewriter)
2. Cal Comp Model 565 Incremental Plotter
3. DECTape TU-55 Magnetic Tape Transports (2) and TC-01 Controller
4. Digital Equipment High-Speed Perforated Tape Reader, Model 750C

The Teletypewriter is used to enter necessary data for measurement runs and to call programs into memory from the magnetic tape storage. Results also can be typed out in tabular form on the Teletypewriter.

The incremental plotter can be used to generate a graphic plot of raw response data or data from any of the intermediate steps during processing that are stored in blocks on the magnetic tape system. All programs developed for the plotter are designed to utilize data that has been stored previously on magnetic tape. The plotter moves in 0.005-in steps and has a 10-in range of movement across the paper; it thus can employ the full integer range of the computer.

Each small (4-in diameter) reel of tape contains up to 1474 blocks of 128 words. Two transports are used and, generally, programs are written on one tape and data on the other. Each block of the tape is numbered and can be retrieved by any program that references the block number.

The high-speed paper-tape reader is a convenient device for entering programs or data into core from paper tape at high speed. Otherwise, paper tapes must be read in through the low-speed reader on the Teletypewriter, which operates at the speed one inch of tape per second.

Operation Under Programmed Control

In general terms, the computer-controlled calibration procedure was visualized to involve three modes of operation: data acquisition, data reduction, and data plotting or output. Each one of these operations has to be examined and thought out as a logical sequence of events that, when coupled with the interaction of the system operator, will ultimately produce the desired result--the hydrophone calibration. The sequence of events must be expanded into a definite flow chart that in turn can be translated or coded into a program for the computer.

A preliminary flow chart for the data-acquisition mode identifies the need for specific subroutines to service the components of the DMS as required and for mathematical routines that are needed in the process. In addition, the different data formats (discussed in Appendix A) need to be taken into account. New formats are created as required. Arithmetic operations are most convenient in a floating-point format that groups together three computer words (one word, 11-bit and sign for the exponent; two other words, 23-bit and sign for the mantissa). The readings obtained as output from the analog-to-digital converter are single-word 11-bit signed integers. The DECTape magnetic tape system uses 128-word blocks of 12-bit numbers. The operation of the frequency synthesizer is based on BCD numbers. An example of a new format is the 9-bit logarithm described below and in Appendix A.

A reasonable portion of the 4096-word memory must be allocated as a working buffer for the incoming data. When the buffer is identified as full, or at the end of a frequency sweep, the contents of the buffer are transferred onto the magnetic tape system for storage. Because of the large quantity of numbers to be handled in the calibration of wideband hydrophones, it is apparent that the measured data stored in core memory and subsequently on the magnetic tape should be in single-word (12-bit) format, if possible, for compactness and efficiency in the over-all operation.

The increment in frequency for the sweep can be taken as an additive constant to generate a linear frequency sweep or scale, or it can be taken as a multiplicative constant for a logarithmic scale. In a developing DMS, both types of frequency scales and their associated problems and limitations have to be investigated for possible use.

Within the data-reduction mode, there is hidden a problem that requires solution before the use of the DMS can achieve any degree of practicality. Reflections from the boundaries of the body of water (surface, bottom, etc.) create a multipath interference pattern on the recorded traces. Some practical method is needed to provide a useful estimate of the real signal in the presence of the multipath interference pattern.

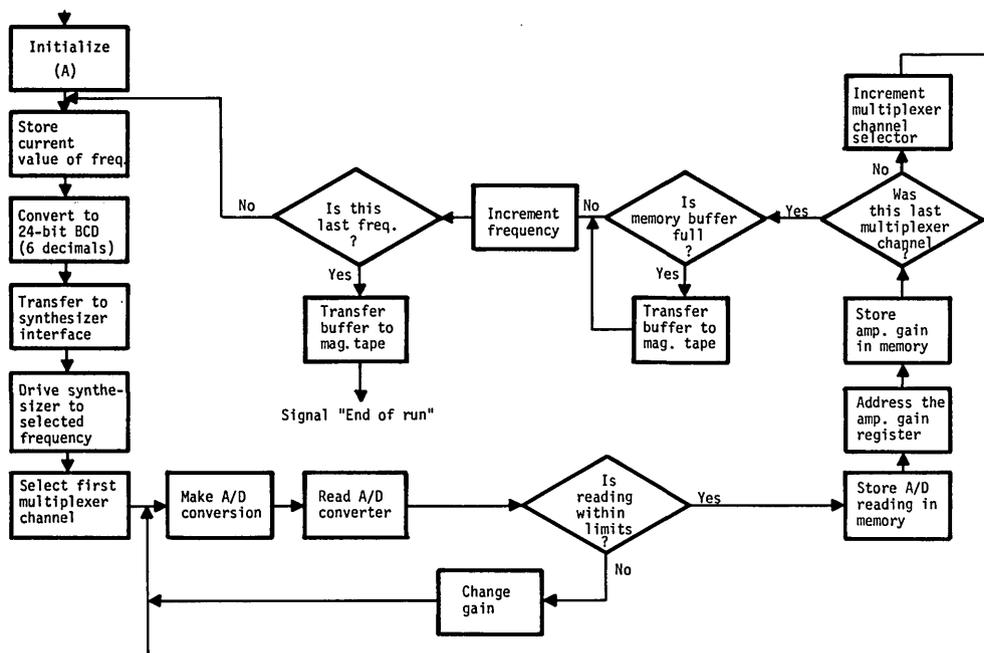
In the data-plotting mode, the programming problems center around identification of proper formats for storing the data and making them compatible with the characteristics of the basic software for the incremental plotter. Available programs have to be modified accordingly to reach compatibility.

Before any definite preparation of the series of calibration programs could be attempted, a great deal of programming work had to be done to satisfy the needs for specific subroutines and preliminary programs, and prior to this, familiarity with the computer and its programming techniques and languages had to be developed.

Concurrent with hardware development, special test programs were worked out as needed to exercise the system components while still in breadboard form--for example, the synthesizer interface, and the auto-range amplifiers.

Data-Acquisition Mode

Operation of the system in this mode can be illustrated in terms of Fig. 5. After the program has been called into core memory, the program is started and then it steps through the requested frequency stations making the measurements and recording the data at each station. The running time for completion of a run depends on the total number of frequency stations; typically, it is about two minutes. The data obtained within one such run are stored for subsequent processing with the other runs that are required to obtain the hydrophone calibration. The increment in frequency usually is taken as 10 Hz to obtain sufficient resolution in the low-frequency end, say at 100 Hz. For a run to 100 kHz, this results in nearly 10 000 frequency stations. With a logarithmic frequency scale produced by means of a multiplicative increment, the same frequency span (100 Hz to 100 kHz) can be covered adequately by about 120 stations, at the rate of 40 per decade.



(A) Dialog between operator and computer: set up frequency range and frequency increment.

Fig. 5. Flow chart, data-acquisition mode.

Output readings from the analog-to-digital converter are linear and each one is converted to logarithmic form by a subroutine that runs in about 1.8 msec to yield the common logarithm in 12-bit format. Truncation to 9 bits leaves 3 bits available to store the actual gain setting for the channel. Nine bits provides a resolution of 1 part in 512; thus, each 10-dB step of the autorange amplifier is resolved into 512 parts, which is adequate precision for the measurement. The result of this approach is two-fold: The data are recorded directly in logarithmic form, and they are stored in single-word format.

In the data-acquisition programs, the operator specifies the initial block number for the DECTape reel as part of the initial dialog at the Teletypewriter keyboard. This block number together with an identifying run number is recorded on the reel for logging purposes.

Data-Reduction Mode

The problem caused by surface and bottom reflections was investigated [3] simultaneously with the development of the DMS. Algorithms were developed to take into account the geometry (source-to-hydrophone distance and hydrophone depth) of the calibration experiment and to provide an estimate of the direct signal in the presence of the reflected signals.

Data reduction is inherently more difficult with the logarithmic frequency scale. The smoothing window has a constant frequency bandwidth, hence it includes a decreasing number of data points as the frequency is swept upwards. The data-smoothing programs provide for single-pass or double-pass smoothing. Double-pass smoothing is used for the higher values of interference-to-direct signal ratio, as occur when the separation distance is increased to 100 cm or more. Usually, measurements are made at 60 cm separation distance, and single-pass smoothing is sufficient.

A program for the comparison calibration procedure operates on a data run from the unknown hydrophone and a data run from a reference hydrophone to establish the calibration of the unknown hydrophone. A separate program measures and computes the hydrophone coupling data for those hydrophones that have a preamplifier. Results are stored on another section of the data DECTape for subsequent plotting.

Programs for reciprocity calibration are under development.

Data-Plotting Mode

For the data-plotting mode, the starting point is the sets of data stored on magnetic tape. A dialog with the computer at the Teletypewriter keyboard specifies the details of the plot such as the ranges for the ordinate and the abscissa. Eight versions of one basic plotting program are available to accommodate data format differences that depend upon the particular program used in the data-acquisition and the data-reduction modes.

A general-purpose labeling program is available to generate legends and titles on the plots from the incremental plotter. The usual plot of hydrophone sensitivity is made to the scale 4 dB per inch on the ordinate.

The entire calibration operation is shown in Fig. 6. This figure identifies some of the options available to the operator; for example, selection of logarithmic or linear frequency scaling, single-pass or double-pass smoothing. The separate programs are identified by the sequential number assigned to them for their development. Those programs that appear in Fig. 6 are the primary programs. The calibration procedure requires calling each program from the appropriate program DECTape, which is a time-consuming, repetitive operation by the operator. An approach was worked out² wherein the sequence of needed programs is specified at one time to an executive master program that calls for each program in the sequence.

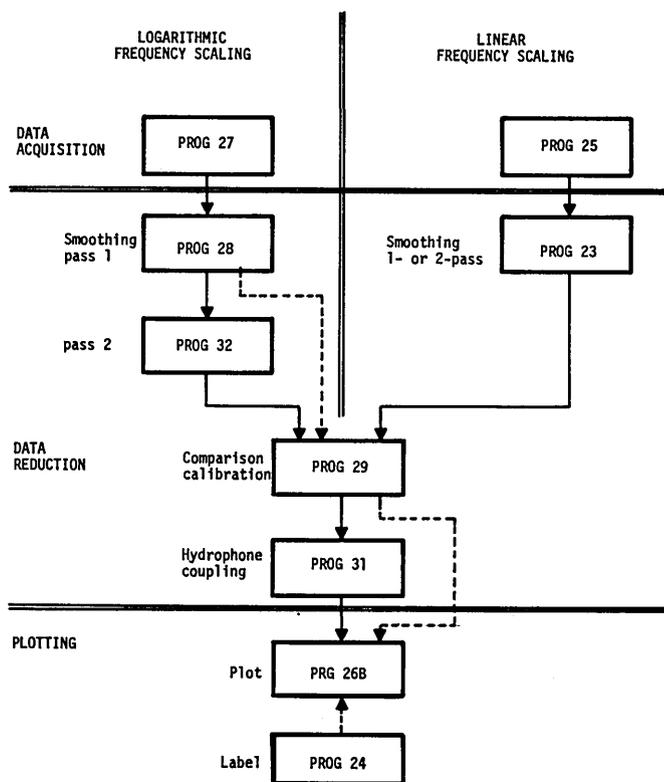


Fig. 6. Program application chart.

Memory-Oriented Mode

The description that has been given is that of a general approach to calibration work with the DMS that might be termed DECTape-oriented. Such a method of operation is quite general in that it is used for any reasonable combination of frequency range and increment. The experience gained with the DECTape-oriented system, particularly the success of the data-smoothing techniques, stimulated the development of a type of memory-oriented operation contained entirely within the 4K core memory of the minicomputer. The frequency range is covered by a discrete number of frequency points. For example, a list of 42 frequency values can cover a 3-decade span at the rate of 14 frequency values per decade. At each

²This valuable contribution was suggested and made operational by A. M. Young.

specific frequency value, the actual measurement is made at a band of frequencies that sequentially cover the smoothing window called for by the data-smoothing algorithm. Thus, the data-acquisition mode and the data-reduction mode are combined into a single operation. The data from one run are stored in core memory and the combination of the two proper runs yields the calibration of the hydrophone. It is more convenient in the memory-oriented system to print out in tabular form the calibration at each of the 42 discrete frequencies. For speed of operation, the operator obtains a stored list of the discrete frequencies from the DMS-program DECTape. Thus, the need for entering the list at the keyboard is eliminated.

The development of the memory-oriented system as a sequel to the DECTape-oriented system illustrates the flexibility available in the computer-controlled system.

Summary

The developmental system that has been described has been in regular use for about 12 months. More than 1200 hydrophones of various types have been calibrated with the system, as many as 60 in a single day. With large numbers of hydrophones of the same type, the point is reached where the calibration procedure is limited by the time required for rigging each hydrophone into position for measurement.

The development of this system was time-consuming because of the new technologies involved at the time that the system was started. This is particularly true of the problems associated with the generation of the software.

Acknowledgments

Several persons besides the authors were deeply involved in this project. Their contributions made completion of the project possible. Hardware was developed by R. J. Powelson. Software Packages were developed by R. W. Luckey. A. M. Young has used the system for a variety of measurement projects that established the usefulness of the system and made it possible to declare it operational. C. C. Sims had the initial responsibility for the project.

References

- [1] R. J. Bobber, *Underwater Electroacoustic Measurements* (Naval Research Laboratory, U. S. Government Printing Office, Washington, 1970), Chap. 2.
- [2] Digital Equipment Corporation, *Small Computer Handbook*, 1970 edition, p. 142.
- [3] L. B. Poché, Jr. and G. A. Sabin, "Multipath Interference Smoothing in Hydrophone Calibrations," *J. Acoust. Soc. Amer.*, in press (scheduled at this writing for September 1972 issue).

APPENDIX A

Computer Data Formats

Fundamentals

To become conversant with the operation of the PDP-8, one must be familiar with the several number systems employed. Numbers may be represented in decimal, binary, and octal number systems in the DMS; therefore, it frequently is necessary to convert from one number system to another. An excellent introduction to the necessary concepts is given in reference [A1], pages 1-5 to 1-14.

The PDP-8 uses 12-bit binary numbers, or words, to represent data. Also note that with 4096_{10} memory locations, each location can be specified by a 12-bit number, or address. This means that the contents of a location may represent either a data number or the address of another location. In addition, the *memory reference instructions* (reference [A1], Chapter 2) that control the arithmetic and logical functions of the computer are *coded* into 12-bit words, so that these instructions can be included among the possible contents of an address. In this manner, a program is constructed in which the memory locations are stepped through in sequence and the operations are performed upon the stored data. Both instructions and data are contained in the 4096_{10} memory locations.

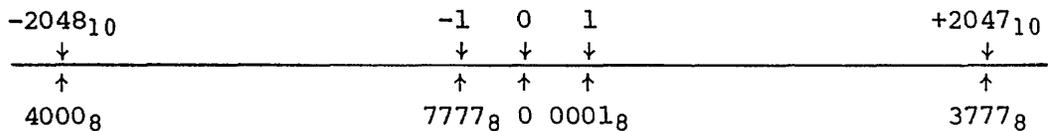
Several formats can be employed to represent numbers within the limitations imposed by 12-bit words. The following types of numbers will be encountered in the system:

- a. 11-bit signed integer
- b. 12-bit unsigned integer
- c. 3-word floating-point number
- d. A-D output number
- e. 9-bit logarithm

A brief explanation of each type is given below:

11-Bit Signed Integer

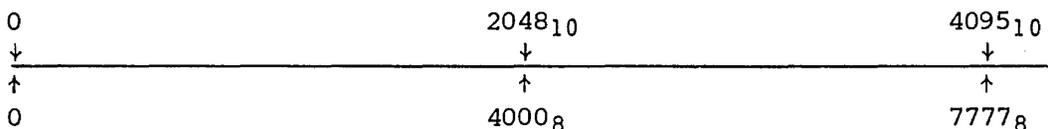
This is the form most commonly used for representing integers. It may be visualized by means of the diagram below:



If the first bit (Bit 0) of the word is used to denote the sense of the integer and the remaining 11 bits are used to represent the magnitude of the number, the scale of numbers will be split at the value 4000₈. If, in addition, complements are used to represent the negative numbers, a scale such as that shown will be obtained. A full explanation is given in reference [A1], page 1-34.

12-Bit Unsigned Integer

This form is encountered in counting operations (*e.g.*, block numbers on magnetic tape). Again, this form may be visualized by a diagram as below:



3-Word Floating-Point Number

To represent nonintegral, fractional, or integer numbers greater than 4096₁₀, more than one word must be joined and taken as a unit. Of the many possible combinations for achieving this end, the one chosen for use in the PDP-8 is as follows: A number to be converted is divided into two parts, a mantissa or fractional part and an exponent. As examples:

Number	=	Mantissa	Exponent
33.3	=	.333 × 10	2
.333	=	.333 × 10	0
.003	=	.300 × 10	-2

In the first case, the mantissa is .333 and the exponent is 2. Floating-point notation is similar except that the mantissa and exponent are binary numbers and the base of the exponent is 2 rather than 10, as in the example. The exponent and its sign are stored in one binary word, and the mantissa and its sign are stored in the remaining two words. Note that in each case, the mantissa is selected so that there are no leading zeroes, thus retaining the maximum number (23 bits) of significant digits. A full description of the floating-point system is given in reference [A2], Chapter 16.

A-D Output Number

The Model AF01 A-D Converter outputs a 12-bit word for each conversion, or reading. A d-c voltage between 0 and -10 volts at the input is represented by this word. The values are scaled according to the following diagram:

0	-4.998	-5	-5.002	-10 V
↓	↓	↓	↓	↓
↑	↑	↑	↑	↑
4000 ₈	7777 ₈	0	0001 ₈	3777 ₈

9-Bit Logarithm

Since the input to the A-D Converter is held within the range of -1 to -10 volts by the autorange amplifier, the a-d output will fall within a 20-dB range. The 12-bit a-d number is increased by 4000₈ to place it in the range 0000₈ - 7777₈. The resulting number is converted by a mathematical subroutine into a 12-bit logarithm, then truncated to 9 bits and rotated to the right, leaving the first 3 bits empty. Gain range information from the autorange amplifier is used to fill these bits. The 9-bit logarithm can resolve the 20-dB output range into 2⁹ or 512₁₀ steps. This provides a resolution of approximately 0.04 dB in the voltage data.

The "9-bit log" word is then arranged as follows:

Gain Range	9	-Bit Logarithm
0 1 2	3 4 5 6 7 8 9 10 11	

The gain range is coded as follows:

Binary	Range
000	TOO LOW
100	40 dB
010	30 dB
110	20 dB
001	10 dB
101	0 dB
011	TOO HIGH

The code 111 is used as a program function to indicate NO DATA exists in the channel. Note that if the 3 bits are read as a binary number from right to left (reversed) they are seen to increase from 0 to 7.

References

- [A1] Digital Equipment Corporation, *Introduction to Programming*, 1968.
- [A2] Digital Equipment Corporation, *Programming Languages*, 1970.

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13. ABSTRACT A computer-controlled measuring system that combines measurement and computation tasks to speed hydrophone calibration has been developed. The system incorporates one of the popular minicomputers now available. Specifications and developmental problems concerning the related hardware equipment are discussed. The small (4K) core memory size dictated specific approaches to programmed operation. Several modes of operation that meet the initial goals have been provided to make the system as flexible as possible. Programming requirements of these operational modes are reviewed.			

14. KEY WORDS	LINK A		LINK B		LINK C	
	ROLE	WT	ROLE	WT	ROLE	WT