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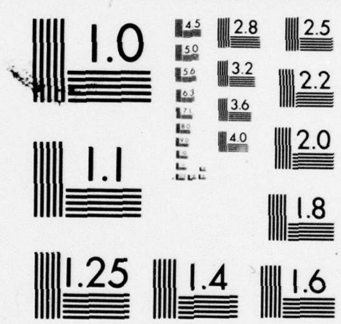


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The Application of Digital Techniques
to Sonar Systems

Prepared for

Sonar Branch
Bureau of Ships

under

Contract NObsr-85185
Problem 2

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by

TRACOR, Inc.
(Formerly Texas Research Associates Corp.)

March 29, 1962

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TRACOR, INC.

1701 Guadalupe St. Austin 1, Texas GR6-6601

**TEXAS RESEARCH ASSOCIATES** *1701 Guadalupe St. • Austin 1, Texas*

*A Division of* TRACOR, *Inc.*

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Prepared by

10 B. H./Deatherage,
E. A./Tucker,
A. F./Wittenborn
J. M./Young

12 27p.

11 29 Mar[redacted] 1962



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**TEXAS RESEARCH ASSOCIATES** *1701 Guadalupe St. • Austin 1, Texas*

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## Abstract

A discussion is given of several topics which are considered fundamental to a consideration of digital techniques. These include time sampling, amplitude quantitization, and selected comments on number systems and computers. Since the functional elements of sonar systems can be described in mathematical terms, it is inferred that these elements can, in principle, be implemented or simulated by either analog or digital hardware.

A discussion of specific applications is given to demonstrate that the choice of techniques depends on practical considerations. Two different uses of digital techniques are described, depending on whether or not the operation being performed must be carried out on a real time basis.

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*A Division of* TRACOR, *Inc.*

# THE APPLICATION OF DIGITAL TECHNIQUES TO SONAR SYSTEMS

## 1. INTRODUCTION

During the past decade very rapid advances have been made in digital computers and computing technology. These advances are evident in the development and manufacture of computing elements, in the assembly of these elements into large computers with fast operating times and in the development of programming techniques with which to apply the computers to the solution of problems. The large increase in computational capability, both per unit time and per unit price, produced by these technological advances has significantly influenced a number of areas of endeavor, e.g., scientific calculations, business record keeping, management decision processes, information storage and retrieval, and automated language translation. It is therefore of interest to consider to what extent these same technological advances can be advantageously applied to the field of sonar, in system design, in the implementation of existing system concepts, and in the implementation of system concepts heretofore not used or not considered to be practical.

The original purpose of this study was to conduct two parallel programs of work, one concerned with the application of digital techniques to sonar systems and the other concerned with the development of a method for comparative evaluation of sonar systems, specifically digital versus analog. Once an evaluation method has been worked out, possible system designs can be simulated and evaluated on a digital computer using recorded or computer generated input information. By carrying out a sequence of parameter studies, a detailed set of criteria which describe the relative effectiveness of digital and analog elements in a system can be established. However, due to fiscal considerations it was necessary to arbitrarily separate the program into two time phases. The present report is therefore an interim report. It contains only general conclusions which can be made without a complete evaluation technique and some of the preliminary detailed work which is necessary to develop the evaluation technique. A detailed discussion will be given when the evaluation technique has been established. The report has therefore been written to serve as an introduction to the use of digital techniques in sonar systems for readers who have not previously had the opportunity to consider the subject.

Section 2 contains a discussion of several topics which are considered fundamental and a necessary prelude to a discussion of digital techniques. Time sampling and amplitude quantitization are described and some selected comments concerning number systems are given. Then a discussion is given which concludes that, since the functional elements of sonar systems can be described in terms of mathematical equations, these elements, with the exception of the transducers, can, in principle, be simulated or implemented with either analog or digital techniques. It is inferred that a choice between these techniques in any specific instance usually depends on engineering and economic factors such as reliability, size, weight and cost. Two different uses of digital techniques are described, differing in whether or not the operation being performed must be carried out on a real time basis.

In Section 3 a more specific discussion of the application of digital techniques is given in order to illustrate the remarks of Section 2.

# 2. SELECTED FUNDAMENTALS

## 2.1 Time Sampling and Amplitude Quantitization

Almost universally, sensors of physical processes have analog outputs. For example, hydrophones have continuous electrical outputs which vary with time according to the variations of that portion of the sound pressure field in the water to which the hydrophone responds. It is intuitively obvious that in order to perform operations on the outputs of these sensors with digital equipment, the continuous time series must be replaced by an equivalent sequence of numbers.

The process of replacing a continuous series with an equivalent sequence of numbers is analogous to the process commonly used in making a table of values from a curve representing the continuous function. Values of the ordinate are read at time intervals (in general, intervals of the independent variable) sufficiently close together so that the continuous function can again be plotted from the table of values to within a prescribed degree of detail. The individual values of the ordinate (magnitude and sign) are called samples, and are usually taken at equally spaced intervals of time for convenience.

The accuracy to which the values of the samples can be determined depends on the scale to which the original curve is drawn. Although the physical process which produced the curve, say the sound pressure field in the water, had a precise and definite value, the limited resolution in the scale and recording equipment allows only a certain number of significant figures to be read. This results in a situation in which all values lying between two adjacent resolvable values within the minimum resolvable interval are rounded up or down in the usual manner. The smallest interval in magnitude which can be resolved is called the quantitization interval. Evidently, the coarser the scale resolution or the smaller the number of significant figures which can be obtained, the more a curve plotted from the table of numbers will differ from the original curve.

It is of interest to discuss how closely in time the samples must be spaced in order to insure that they represent the original time series to within a specified accuracy. (It should be noted that the sampling interval is short compared to the interval between samples, unlike the sampling process used in the conventional scanning sonar.) The required sampling rate can be conveniently defined if a high degree of precision is assumed in reading or measuring the instantaneous value of the sample. Then it is of interest to discuss the effect of quantizing the magnitude of the samples, i.e., assigning a numerical value to the sample to a prescribed number of significant figures, following the time sampling. These two processes, time sampling and amplitude quantitization will be recognized as the most fundamental factors affecting the application of digital techniques to sonar (or, for that matter, to any other signal or data processing operation where an analog signal is the basic input). From the point of view of equipment cost and complexity, it is desirable to minimize both the number of samples per unit time and the number of significant figures which must be used.

The lowest sampling rate or the maximum time interval between samples which can be used in representing a continuous time series by a sequence of samples can be determined from the well-known sampling theorem.[1] The usual and simplest way of expressing the sampling theorem applies to signals all of whose frequency components, f, are contained in a band of width B, where $0 \leq f < B$. For this case the theorem states that if a time series contains no frequencies higher than B cycles per second, it is completely determined by giving its value at a series of points 1/2B seconds apart, the series extending

[1] H. Nyquist, "Certain Topics in Telegraph Transmission Theory," Trans. AIEE 47, p 617 (1928).

throughout the time domain. Stated in another way, at least two samples per cycle of the highest frequency in the signal under consideration are required to provide an accurate representation of the signal. However, a precisely limited band of frequencies is not strictly attainable in practice. The effect of this on the actual choice of a suitable sampling frequency in a specific situation can be described in the following manner.

If a continuous function has a power spectrum $G(f)$, samples taken at intervals of $\Delta t$ seconds can be used to generate a new function, or time series whose power spectrum is $G_A(f)$. This new power spectrum is equal to zero for frequencies greater than B. The new power spectrum $G_A(f)$ corresponds to a new time series for which the equally spaced time samples give an exact representation, according to the sampling theorem as stated above. The two power spectra are related by the folding process indicated in Figure 1. The power originally contained in frequencies greater than B is folded back into the band $0 \leq f < B$. In any specific application, therefore, B must be chosen so that the power at frequencies greater than B is negligible compared to the power originally in the band $0 \leq f < B$ if distortion is to be avoided. The power folding process is commonly referred to as producing "aliasing" errors.

The distortion produced by amplitude quantitization or the finite resolution of the measuring device has been examined for a number of different situations, including band-limited random noise.[2] The distortion is not severe compared to aliasing, as can be understood qualitatively from the following considerations. Each sample value will usually be greater than or less than the correct value, depending on whether the prescribed number of significant figures were rounded up or down. If a continuous curve is drawn through a sequence

[2] W. R. Bennett, "Spectra of Quantized Signals," Bell System Technical Journal, 27, p 446 (1948).

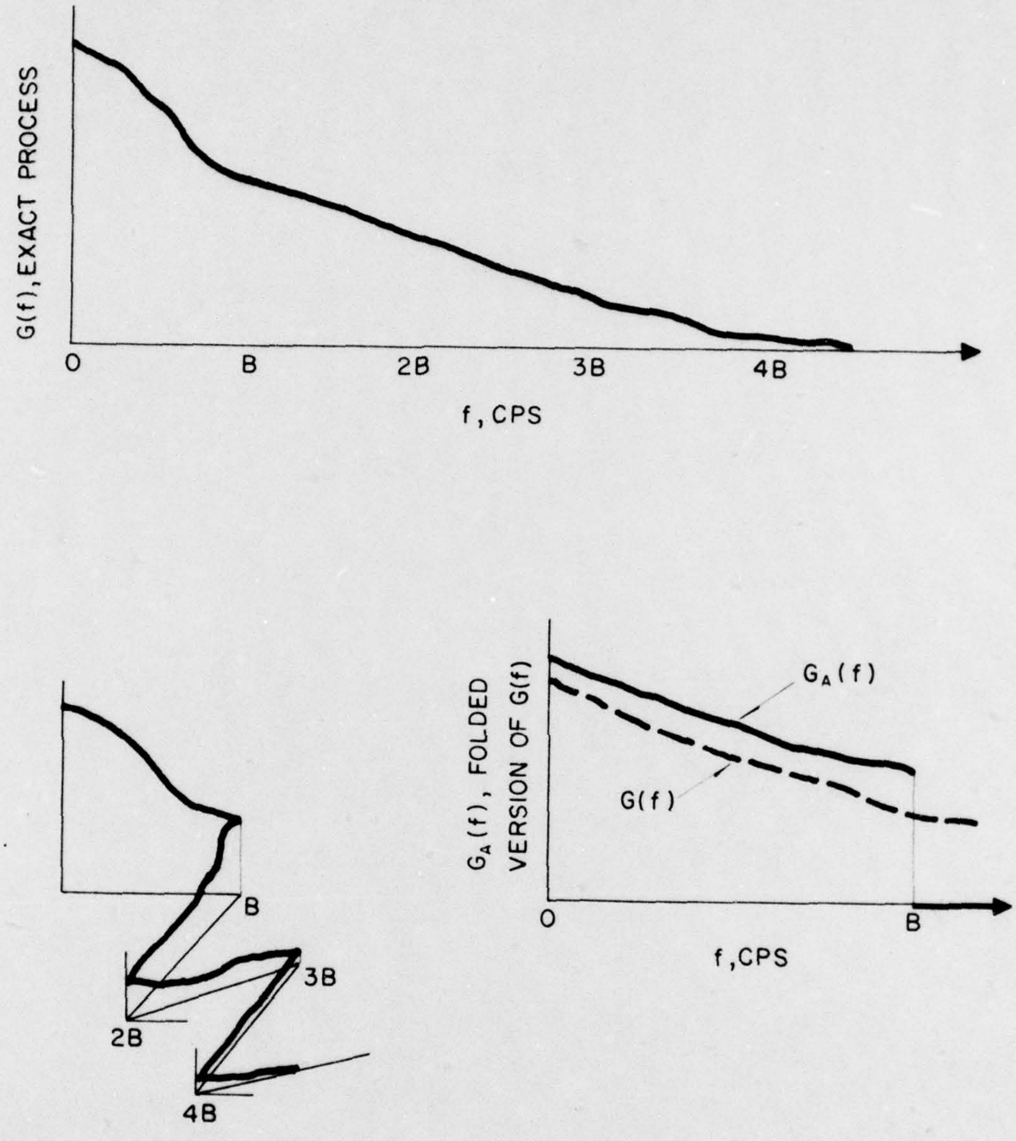


Fig. 1 - SCHEMATIC OF THE FOLDING OF A POWER SPECTRUM RESULTING FROM DISCRETE SAMPLING

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of such values, it will fluctuate above and below the correct value. Evidently the highest frequency which the fluctuations can have corresponds to the situation in which the fluctuation changes sign between adjacent samples, i.e., half the sampling frequency. All other fluctuations are at lower frequency. The entire fluctuation spectrum is thus in the frequency band $0 \leq f < B$, where the sampling frequency is 2B, and cannot be distinguished from the correct energy in the band after quantizing.

2.2 <u>On Number Systems</u>

It is of interest to digress briefly to consider some elementary concepts concerning numbers. In the decimal system commonly used for writing numbers, it is understood that the successive digits in a number are multiplied by powers of ten in descending order. Thus a decimal number such as 4289 is a shorthand method for writing

$$4289 = 4 \times 10^3 + 2 \times 10^2 + 8 \times 10^1 + 9 \times 10^0.$$

The number 4289 is expressed to the base 10. Any number can be written to the base ten if 10 symbols are available, specifically 0, 1, 2, 3, 4, 5, 6, 7, 8, 9. Now any other number besides 10 (with the exception of the integers 0 and 1) can be used as the base to form a new number system. If the base is the number b, then b symbols will be required. For example, if b = 8 (the octal system) the symbols 0, 1, 2, 3, 4, 5, 6, 7 are required and the decimal number 8 is written as 10 (usually $(10)_8$ to avoid ambiguity) since, analogous to the decimal system

$$(10)_8 = 1 \times 8^1 + 0 \times 8^0.$$

Essentially all electronic or high speed digital equipment or digital computing systems are constructed to use a number system to the base 2, called the binary system. For this system two symbols are required, namely 0 and 1. A number such as $(101101)_2$ is interpreted as

$$(101101)_2 = 1 \times 2^5 + 0 \times 2^4 + 1 \times 2^3 + 1 \times 2^2 + 0 \times 2^1 + 1 \times 2^0$$

$$= 32 + 8 + 4 + 1$$

$$= (45)_{10}.$$

A binary digit is commonly called a bit. The binary number system is of importance to digital equipment because it is necessary to resolve only one of two possible states. Thus a bit can be represented, for example, by a switch which is either open or closed, by a transistor which is either conducting current or quiescent, by a magnetic material which is saturated as either a north or a south pole. By paralleling a sufficient number of such two-state devices, any decimal number converted to the binary system can be represented with great accuracy. In addition, the number can be stored in this fashion for long periods of time without degrading the accuracy and can be readily retrieved for later use. One two-state device is always required for each bit.

The rules of arithmetic can be formulated for a number system to any base in a manner which is analogous to the formulation in the decimal system. These rules are especially simple in the binary system. Computing elements which add, subtract, multiply and divide can be constructed using devices which can recognize the presence or absence of the two states representing bits and respond with an output consisting of one of two possible states ("and" gates, "or" gates, etc).[3]

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[3]G. C. Gottlieb and J. N. P. Hume, High-Speed Data Processing, Chapter 3 (New York: McGraw-Hill Book Co., Inc., 1958).

When a collection of storage elements and arithmetic computing elements are organized and placed under the control of counters and gates, a digital computer is formed. In reality, the electronic digital computer is, therefore, a collection of analog devices which recognize, exist in, and respond to two possible states. Numbers can be received, stored and retrieved from storage accurately by representing them in the binary system or some system derived from it. Accurate arithmetic can be carried out by formulating the rules in terms of simple yes-no responses. Logical operations and number transfers can be carried out accurately by formulating these operations in terms of simple yes-no decisions.

### 2.3 On the Use of Digital Techniques

Sonar systems are built for the purpose of serving as a link to convey information contained in the sound pressure field in the water to a human operator. The information should be in the most suitable form for making operational decisions. To serve this purpose, sonar systems must perform several functions. They must first of all deliver information to determine whether or not a signal is present (detection). If a signal is present, they must deliver such information as may be required to determine whether or not the signal is a valid target (classification) and, if so, to supply such additional information as may be required to make a decision concerning a future course of action.

Under most operating conditions the signal from a potential target is masked by noise. In order to maintain a reasonable degree of functional efficiency, it is desirable to exploit the best known procedures for treating the signal and noise in the sonar system in order to aid the decision process. This can be achieved by designing equipment which instruments the best suited methods of signal processing, usually to maximize the output signal to noise ratio, and providing equipment to serve as the most suitable communications link between the output of the signal processing equipment and the human being. The latter equipment is called the display.

Most of the operations carried out in a sonar system can be described mathematically. Possible exceptions lie in the display or in the effect of the display on the operator. The mathematical descriptions of signal processing, on the other hand, have been developed to the extent that in some instances (e.g., linear filtering) optimum operations or procedures have been established.

It is logical to consider the mathematical description of the operations carried out in an analog sonar system as simulating the system. However, it is equally logical to consider the analog equipment as a simulator of the mathematical description of the operations. Then, in view of the fact that a continuous time series, such as the noise or signal and noise at some point in the system can be expressed in terms of an equivalent set of numbers or time samples, and in view of the known fact that electronic machines can be constructed to operate upon this sequence of numbers according to a program prescribed by a mathematical formulation, it is also logical to consider digital equipment as a simulator of the mathematical description of the operations. If, for the simulation of a specific operation the state of the art is such that it can be done by either type of equipment, the choice of equipment would be based on such factors as cost, reliability and possibly size and weight.

Digital equipments have a number of fundamental properties which, if they can be exploited, can lead to distinct advantages in certain operations. Some of these properties are the storage of data without degradation such that it can be readily retrieved, the possibility of time sharing of equipment between data channels because of the relatively short sampling time compared to the time between samples, and the potential stability in time which can be achieved primarily because of the storage.

Digital equipment can be applied in the field of sonar in two different ways. On the one hand, general purpose computers can be used for simulating signal processing and display in order to optimize design parameters. The mathematical modeling process as well as implementing the model in the form of a computer program can generally be carried out and subsequently modified more economically than the construction of an analog model or special digital model. Simulation can be carried out by using input data either generated by the computer or recorded at sea using a representative hydrophone array. An important feature of this type of simulation for design is that the computer need not perform the necessary calculations on a real time basis, i.e., the calculations can be made at a faster or slower rate than real time, depending on computer capability. The use of computers in design parameter study and evaluation is therefore straightforward and need not be considered further here.

On the other hand, the use of digital equipment in an operational system requires that all operations be carried out on a real time basis. In many instances this requirement may strain the state of the art in speed or capacity of digital equipment, and the performance of digital as compared to analog equipment will depend upon the cleverness and ability of the respective designers. Advances in the state of the art can change the relative standing at any time. A thorough discussion of the relative merits of the use of digital and analog equipment in synthesizing a sonar system cannot, therefore, be given without a comprehensive method for evaluating these systems. A study program for devising such a method of evaluation is in progress in connection with another problem assignment. A more complete discussion can be given when the evaluation study has been completed.

In order to illustrate the preceding general remarks, a more detailed discussion of the use of digital equipment in sonars is given in the next section in terms of conventional sonar system design procedures.

## 3. DISCUSSION

In this section the application of digital techniques to sonar systems will be considered more specifically by discussing briefly each of the elements in a system for which digital equipment can be made available. For receiving systems this will include those elements used in signal processing and display while for transmitting systems this will include array phasing and signal generation.

It will become evident from the discussion that, for the processes which are considered, either analog or digital techniques can, in principle, be used. Which equipment can, in practice, deliver the better performance and result in systems having better characteristics in terms of such factors as reliability, size, cost and weight is not so evident. General conclusions must await the development of a comprehensive relative evaluation technique. Even so, a comparison between specific competing designs at any given time, either between an analog and a digital element or between two digital elements, will not be independent of the state of the art nor will it necessarily be independent of the cleverness of application of state of the art devices by the design engineer. The following discussion will therefore be limited to considering how digital techniques can and, in some instances, have been used in sonar, and only preliminary remarks will be made concerning the practicality of this use. A more complete discussion will be given in connection with the discussion of an evaluation technique which is under development as a separate problem assignment.

### 3.1 Signal Processing

As is well known, signal processing gain in sonar systems for a given input signal-to-noise ratio, can be obtained by: (1) exploiting the difference in spatial coherence of signal and noise; (2) exploiting spectral differences in the signal and noise, which, through the Fourier transform, are equivalent to differences in the signal and noise as a function of time; and (3) providing means for increasing the observation time so that time averages of the noise or signal and noise can be used to smooth some of the naturally occurring fluctuations. The first of these is carried out by using a multiplicity of physically separated hydrophones and combining the separate outputs to form beams. The second of these is carried out by filtering, or the time equivalent process of correlation. The third of these is achieved by forming suitable time averages of either a set of correlation functions or of the rectified signal or signal and noise (detection and post-detection integration). These processes will now be considered in more detail insofar as digital techniques can be employed in their execution to maximize output signal-to-noise ratio.

Arrays and Beam Forming

The noise received by a hydrophone is generally the resultant of a large number of relatively small sources distributed in some fashion in the volume surrounding the hydrophone. The manner in which the sources are distributed has been investigated both theoretically and experimentally.[4] It has been observed to vary with environmental conditions such as sea state, depth of hydrophone, etc. The character of the resultant noise field is therefore also dependent on these factors. In general, however, because of attenuation effects the bulk of the resultant noise field at the hydrophone arises from sources relatively near it. If the noise field is examined at a number of different points in space, the noise at each of these points becomes statistically independent of the others if the separation of the points

[4] See, e.g., B.F. Gron and C.H. Sherman, "Noise Models for the Computation of Signal-to-Noise Gain of an Array of Elements," U.S. Navy Underwater Sound Laboratory USL Technical Memorandum No. 913-61-61, (26 May 1961) for some recent theoretical work.

is sufficiently great. It has been shown[5] that the required separation for band limited isotropic noise is approximately 1 to 1.5 wavelengths corresponding to the center frequency of the band.

The signal from a target normally arises from a number of relatively intense sources concentrated in a small solid angle as viewed from the receiving hydrophone. If the signal field is examined at a number of different points in space in the vicinity of any given point, the signals at pairs of these points will have a high correlation if one is time delayed with respect to the other to compensate for propagation time differences, provided the hydrophone separations are not so great that multipath effects and inhomogeneities of the medium can destroy this coherence.

The outputs of an array of hydrophones can be combined, usually by summing. When a number of statistically independent noise outputs are summed, the fluctuation, or standard deviation, $\sigma$, about the mean of the sum will increase, compared to the fluctuation of an individual output, $\sigma_k$ as the square root of the number, N, of outputs summed.[6] Thus

$$\sigma \approx \sqrt{N}\,\sigma_k \qquad (1)$$

[5] James J. Faran and Robert Hills, Jr., "The Application of Correlation Techniques to Acoustic Receiving Systems," Tech. Memo. No. 28, Acoustics Research Lab., Harvard University, pp 14-15 (Nov. 1, 1952).

[6] See, e.g., William Feller, An Introduction to Probability Theory and Its Applications, p 216 (New York: John Wiley and Sons, Inc., 1950). Equations (1) and (2) are often written in terms of an average, in which case Equation (1) becomes

$$\sigma \approx \frac{\sigma_k}{\sqrt{N}}$$

Equation (2), see below, becomes

$$S \approx \frac{NS_i}{N} = S_i$$

and Equation (3) remains unchanged.

On the other hand, when compensating delays are used to correspond to the direction from which a signal arrives, the signal components will add in direct proportion to the number of outputs summed, to give a resultant signal S

$$S \approx NS_i \tag{2}$$

The output signal-to-noise ratio, in terms of the input signal-to-noise ratio, as usually defined, is therefore

$$\frac{S}{\sigma} = \sqrt{N}\,\frac{S_k}{\sigma_k} \tag{3}$$

The process of beam forming by delaying and summing* is a convenient example for beginning a discussion of the application of digital techniques to sonar systems. It is necessary to store a sequence of numbers representing the output of each hydrophone for a period of time corresponding to the propagation time differences which must be compensated for. In forming a specific beam, a sample from each hydrophone to be used is selected such that the signal components are in phase, and these samples are summed to give the beam output. This process was originally investigated at the Marine Physical

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*Other methods of combining hydrophone outputs besides summing have been suggested, e.g., forming products,[7,8] but such methods are not in common sonar use.

[7]John B. Thomas and Thomas R. Williams, "On the Detection of Signals in Nonstationary Noise by Product Arrays," J. Acoust. Soc. Amer. 31, p 453 (1959).

[8]A. Berman and C. S. Clay, "Theory of Time-Averaged-Product Arrays," J Acoust. Soc. Amer. 29, p 805 (1957).

Laboratory and has been described by Anderson[9] and Rudnick.[10] It has been given the acronym DIMUS (digital multibeam steering). The stored data samples representing the hydrophone outputs consist of polarity samples. (Polarity samples will be considered further below.) This is significant from an engineering point of view since only one bistable device is required for storing each sample. Rudnick has shown that the processing gain of the array is not significantly degraded by the use of polarity samples in isotropic noise fields. However, since polarity sampling is equivalent to amplification and clipping, which is a nonlinear process, harmonics of the input noise and signal components are produced, thus increasing the effective bandwidth. This production of harmonics can be compensated for by increasing the sampling frequency by approximately a factor of three, corresponding to the third harmonic of the center frequency. This increase in the sampling frequency is also sufficient, as Anderson has indicated, for providing accurate phasing of the hydrophone outputs; since the time sampled data is obtained at discrete, uniformly spaced intervals, the precision with which a given delay can be obtained depends, on the average, on the size of the sampling interval.

In the DIMUS equipment, analog summing of the hydrophone outputs has been used up to the present time. Digital summing could, of course, also be used. A potential value of digital summing will be discussed further below under post-detection integration.

As is well known and will be pointed out below, the output (amplitude) signal-to-noise ratio is proportional to the square root of the time during which a signal is observed. The use of multiple

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[9] Victor C. Anderson, "Digital Array Phasing," J. Acoust. Soc. Amer. 32, p 867 (1960).

[10] Philip Rudnick, "Small Signal Detection in the DIMUS Array," J. Acoust. Soc. Amer. 32, p 871 (1960).

fixed beams, so arranged that continuous coverage is obtained over the solid angle of interest, is therefore a fundamental requirement in optimizing over-all processing gain. Since multiple fixed beams can, in principle, be formed by using either analog or digital techniques, the beam forming process will be an interesting area for further study and evaluation in order to determine which method or combination of the two methods can be made the more practical in any given case, including active and passive receiving systems as well as the phasing of directional transmitting arrays.

For example, the number of digital samples which must be handled in an active receiver for, say, a sinusoidal pulse can, in principle, be significantly reduced by heterodyning before sampling. The information contained in the pulse is in a bandwidth approximately equal to the reciprocal of its duration, and normally this bandwidth is small compared to the center frequency. No information is lost if the carrier frequency is heterodyned to as low a frequency as possible, consistent with good filtering techniques to avoid (analog) "aliasing."

Filtering and Correlation

The purpose of filtering is to admit into a receiver (or power amplifier of a transmitter) the minimum amount of noise while admitting a signal of a given bandwidth. The design of filters has been studied to the extent that for the conventional linear filter, optimized criteria (for maximizing output signal-to-noise ratio) have been developed.[11] This work has been extended to include time-variable,

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[11] Much of the work on filters stems from the work of Wiener; see, e.g., N. Wiener, Extrapolation, Interpolation, and Smoothing of Stationary Time Series (Cambridge, Mass. Technology Press, 1949).

Other descriptions of optimum filters and extensive lists of references may be found in:

[12] David Middleton, An Introduction to Statistical Communication Theory, Chap. 16 (New York: McGraw-Hill Book Co., Inc., 1960).

[13] J. H. Laning and R. H. Battin, Random Processes in Automatic Control, Chap. 7 (New York: McGraw-Hill Book Co., Inc., 1956).

[14] J. S. Bendat, Principles and Applications of Random Noise Theory, Chaps. 4, 9 (New York: John Wiley and Sons, Inc., 1958).

nonlinear and adaptive filters, although the theories and criteria have not been developed to the same degree as for linear filters. For the specific application to sonar systems, two types of optimum filters have been described, one for the active case and the other for the passive case. These will be considered below.

For the active case, where the nature of the signal is known (except for possible distortion caused by multipath effects, medium inhomogeneities and the reflection process, including doppler shifts from the target) the optimum filter for maximizing the signal-to-noise ratio is defined as a matched filter.[15,16] Its frequency response is the complex conjugate of the Fourier spectrum of the signal plus a time delay which is independent of frequency. The action of a matched filter can be understood if its time equivalent is considered in terms of correlation.[17] A length (in time) of received signal and noise or noise alone equal to the duration of the transmitted pulse is compared, using either the continuous signal or a sequence of discrete samples, to a replica of the transmitted pulse usually called a pilot signal. The integral or sum of the product of the two signals is determined. When only noise is present, the products will tend to be either positive or negative for equal amounts of time so that the integral will have a zero value on the average. When a signal is present, however, the products will tend to be more positive, on the average, than negative at those times when the signal is in phase with

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[15] D. O. North, "Analysis of Factors which Determine Signal-Noise Discrimination in Pulsed-Carrier Systems," RCA Technical Report PTR-6C (June 1943).

[16] W. T. Davenport, Jr., and W. L. Root, An Introduction to the Theory of Random Signals and Noise, pp 244-247 (New York: McGraw-Hill Book Co., Inc., 1958).

[17] The equivalence of these two processes has been demonstrated with a simple example in terms of likelihood ratio in Special Report No. 1, Problem 2, by C. W. Horton entitled "Some Notes on Communication Theory," NObsr-85185.

the replica, giving a positive integral. The integral will be negative, on the average, when the signal and replica are 180$^{o}$ out of phase. The integral will reach a maximum when the received signal is present during the entire integration interval.

The above matched filter-correlation process is well suited to digital techniques. It has been shown that the use of polarity samples does not result in significant degradation of the output.[18] Polarity coincidence correlators have existed for a number of years in the past and modern versions are in use at this time.[19] It can be noted that the beam forming process using polarity samples is analagous to the correlation corresponding to the matched filter except that the former process utilizes samples obtained from different points in space (and with a relative time delay to compensate for propagation time differences) while the latter process uses samples from a given beam obtained sequentially in time.

Doppler shifts in the echo pulse can be measured by performing a number of correlations with pilot pulses which have been shifted in frequency to correspond to the doppler shifts to be measured.[20]

The use of matched filter techniques can perhaps be significantly influenced by digital techniques. The general theory of the use of filters which are nonlinear, or time-variable, or both, has not been applied extensively to sonar systems. The general purpose computer can be used to investigate these problems. Furthermore, any resulting application to operational systems can probably be implemented more conveniently with digital equipment.

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[18] See footnote Reference 9; also James L. Lawson and George E. Uhlenbeck, Threshold Signals, p 57 (New York: McGraw-Hill Book Co., Inc., 1950).

[19] Victor C. Anderson, "The Deltic Correlator," Technical Memorandum No. 37, Acoustics Research Laboratory, Harvard University (January 5, 1956).

[20] R. D. Isaak, U.S. Navy Journal of Underwater Acoustics, 11, p 27 (Confidential).

For the correlation discussed above, the integration time was assumed to be equal to the signal duration. It should also be noted that the results of the integration have a value qualified by "on the average." Because of the presence of noise, each individual result is subject to deviations from the average result. This deviation or uncertainty is smaller for large integration times. The reason for this is analagous to the ones leading to Equations (1) and (2) above. Although the problem of compromise between peak transmitted power, total transmitted power and pulse duration will not be considered here, it is of interest to note that the storage capability of digital equipment can be utilized to obtain pulse to pulse integration in order to increase the observation time. In order to maintain coherence in the transmitted signal either the conventional analog oscillator or digitally generated pseudo-random noise can be used.[21]

For the passive case, the spectrum of the noise and the signal are essentially alike over the bandwidths usually of interest. For this case Eckart [22] has shown that the optimum filter is one which equalizes the usual decrease in spectrum level as a function of frequency, i.e., "whitens" the spectrum. Since the signal itself is noiselike, i.e., the nature of the signal is not known except for its spectrum, the techniques used for the active case cannot be used. Instead, one can observe the amount of power being received from each direction in space and attempt, by a succession of measurements of this power to distinguish between the two conditions: signal and noise or noise alone. The individual observations fluctuate about a mean value, so that an effort must be made to smooth these fluctuations in order to recognize small changes in the mean value of the noise power when a signal is also present.

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[21] A.W. Ellinthorpe, U.S. Navy Journal of Underwater Acoustics, 11, p 33 (Confidential).

[22] Carl Eckart, "Optimal Rectifier Systems for the Detection of Steady Signals," Scripps Institution of Oceanography, SIO Reference 52-11 (4 March 1952).

The power can be measured by rectifying the signal and noise as received by the hydrophone. The subsequent smoothing process is carried out by integrating or averaging the rectifier output. This is usually referred to as post-detection integration. The output signal (amplitude) to noise ratio is proportional to the square root of the observation time.[23]

It can be useful to use as long a post-detection integration time as possible, consistent with the duration of the signal. By exploiting the storage capability of digital equipment, the data samples from each increment of solid angle of interest, or for fixed beams from each data channel, can be integrated not only from a specific direction or channel but also over a number of different directions or channels in a manner corresponding to actual target paths in space. By sequentially comparing an average along a given path to a time variable threshold function, all integration times from a minimum value to a prescribed maximum can be computed and compared to the threshold.[24] The equipment stability required for carrying out such processes may be such that digital equipment will produce consistently better performance.[25]

Although it has been indicated above that much of the filtering could be carried out either with analog or digital equipment, it should be noted that good bandlimiting analog filtering will probably always be useful. It has not been either possible or convenient to construct hydrophones whose frequency response corresponds precisely to the frequency band to be monitored by the sonar. Without good bandwidth limiting, a higher sampling rate is required and more data samples must be processed by any digital equipment which might be used.

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[23] Middleton, op. cit., p 875.

[24] A. O. Christensen, A. F. Wittenborn and H. B. Patterson, Jr., Dresser Electronics, SIE Division, Contract NObsr-85189, May 5, 1961 (Confidential).

[25] A. F. Magaracci, USNUSL Technical Memorandum No. 921-048-61 (4 December 1961) (Confidential).

### 3.2 Display

A display, as defined in Section 2, is essentially the transducer at the transmitting end of a communications system for which a human being is the receiver. The purpose of the communications system is to couple, in an efficient manner, the electrical or numerical output of the signal processing equipment to the observer through his senses in order that he can make a decision.

Considerable analysis of decision processes and decision criteria has been carried out during recent years.[26,27,28] It has been possible to develop relatively complete mathematical formulations of some simple decision processes. Whenever one of these processes so formulated is applicable to a physical system, it can be implemented by hardware. Such a decision process would then properly fall under the category of signal processing. Evidently, each such decision process which is transferred from the human operator to signal processing equipment frees the operator to perform his remaining functions better or to assume new functions. The human nervous system is a very highly complex mechanism, so it is doubtful whether a complete transfer of responsibility can ever be made without degrading performance. Perhaps, because of human adaptability, as functions are transferred new skills will be developed.

For relatively complex decisions, good mathematical formulations are not available. Display is therefore a functional element which can be expected to continue to evolve with time. The role that either analog or digital techniques can play in display cannot now be defined because the basic psychophysical work required to do it is not available. The integration of equipment for display with human

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[26] Middleton, op. cit., Chap. 18.

[27] A. Wald, Statistical Decision Functions (New York: John Wiley and Sons, Inc., 1950).

[28] Carl W. Helstrom, Statistical Theory of Signal Detection (New York: Pergamon Press, Inc., 1960), especially Chap. XI.

capabilities has not yet reached a culminating state of development. For example, transmission of data to an operator in such a fashion that use is made of the fact that he has two eyes or two ears, with a fusion of information in the brain, has not been explored extensively. The exploitation of such human capabilities will no doubt require considerable data manipulation and storage, so that digital techniques can be expected to assume a major role in the process.