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A SONAR DESIGN TOOL: MULTI-CHANNEL MONTE CARLO SIMULATION.(U)

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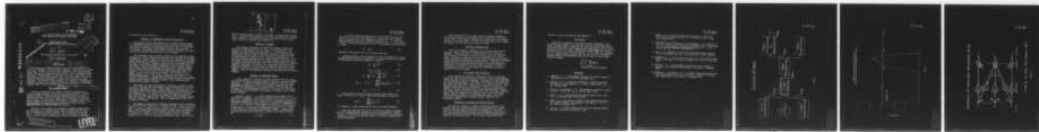
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A SONAR DESIGN TOOL:
MULTI-CHANNEL MONTE CARLO SIMULATION

by

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R. L. Gordon

USL Technical Memorandum No. 2242-380-68

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Technical memos
Introduction

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10 p.

The design of a combined passive and active sonar system in the form of a programmable special purpose computer is feasible with current technology. In fact, the flexibility offered by such a system brings to the foreground an important sonar problem: What algorithms are most effective in accomplishing a given task or mission? Since many of the algorithms are non-linear and may not be directly analyzed in terms of relevant performance parameters, a Monte Carlo simulation has been developed. The essence of the technique is to develop a representative stochastic sequence and carefully observe the algorithm (or algorithms) response under controlled conditions.

14

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The Software Concept

The aim in developing the computer simulation programs was to build in enough flexibility to provide for the simulation of several different types of signal processing hardware and to allow either synthesized signals and noise, (Monte Carlo Techniques) or taped sea data as input. By allowing for modification to key programs, direct system comparisons can be made with identical stochastic samples. The reproducibility and exact control of synthesized signal and noise characteristics are the most important features.

Detailed in this report are six specific computer programs which are the result of a multibeam sonar study in progress by Code 2242. The six programs are shown in Figure 1 and will be detailed in that order. The first three programs are used to synthesize a realistic acoustic field, and the last three are used to simulate the hardware

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and extract certain statistics.

Synthesis of Independent Filtered Random Noise

The purpose of the first program was to produce several channels of independent filtered random noise with a specified spectral shape. The noise is produced by initially generating a series of independent and normally distributed random numbers and then "providing time correlation or dependence" with digital filtering techniques to produce the desired spectral characteristics for each channel.

The techniques used to generate independent but repeatable random numbers by a digital computer are detailed in an excellent survey paper by Chambers¹. Besides describing the history and application of random number generation by computers, a survey was made of current algorithms and their shortcomings. Chambers' conclusion was that perhaps the best algorithm is the multiplicative congruential¹ (power residue) method. This method involves multiplying two thirty-five bit numbers together (one of them always being the same) and taking the least significant thirty-five bits of the result as the random number generated. This number is also used as the next multiplier. The success of this method depends on the choice of the constant and on the choice of a starting random number. The method is extremely fast (about 8 μ s as coded by the author on a UNIVAC 1108) and will with proper normalization present numbers evenly distributed between 0 and 1 with negligible dependence from number to number.

The numbers are then transformed to a Gaussian or normal distribution with a zero mean and unit variance using a polynomial fit to the cumulative distribution curve.² These numbers are now equivalent to the numbers that would be obtained after sampling a bandlimited low pass noise spectrum of 1 volt RMS at twice the cutoff (Nyquist) frequency. The remaining task is to produce correlation between the numbers so that they look like numbers which were obtained from sampling a bandpass process at several times Nyquist. This is accomplished by some form of digital filtering.

The theory and application of digital filtering has been documented many places³⁻⁵ and is founded on Z-transform theory⁶. The most efficient method in terms of machine cycles is usually a filter of the recursive type. Such an algorithm works with future, present and past samples to properly filter the data. It was desired for the initial simulation to synthesize a bandpass filter in the frequency domain with rather steep skirts so a 5th order Butterworth design was used. Such a filter

It turns out there are many \underline{A} matrices, (at least 2^N) which will satisfy (1) for a particular \underline{R} matrix. A particular \underline{A} matrix may be found by solving \underline{R} for its N eigenvalues, denoted as λ_i , forming a diagonal matrix \underline{D} whose main diagonal consists of the λ_i , and a matrix \underline{C} whose i th row are the elements of the i th eigenvector associated with \underline{R} . Then

$$\underline{A} = \underline{D}^{\frac{1}{2}} \underline{C} \quad (2)$$

where $\underline{D}^{\frac{1}{2}}$ is the square root of the diagonal matrix \underline{D} .

Another method for solving equation (1) due to Marsaglia¹⁰, provides an algorithm for finding the elements of a triangular matrix \underline{A} in terms of the original covariance elements as follows:

$$a_{11} = \sqrt{r_{11}} \quad (3)$$

$$a_{1j} = r_{1j}/a_{11} \quad (4)$$

$$a_{ii} = \sqrt{\left(r_{ii} - \sum_{m=1}^{i-1} a_{mi}^2 \right)} \quad i > 1 \quad (5)$$

$$a_{ij} = \left\{ r_{ij} - \sum_{m=1}^{i-1} a_{mi} a_{mj} \right\} / a_{ii} \quad j > i \quad (6)$$

$$a_{ij} = 0, \quad i > j \quad (7)$$

Using either of these solutions a network such as the one in Figure 3, may be used so that a particular hydrophone voltage $e_i(t)$

$$e_i(t) = \sum_{j=1}^N a_{ij} x_j(t) \quad (8)$$

is in general a linear combination of the N uncorrelated channels.

Since the correlated noise field is formed on a sample by sample basis the interhydrophone correlation may be programmed as a function of time. In addition this method allows the study of arrays operating in acoustic noise fields that have unequal correlation, between equidistant hydrophones.

The third program is also used to add the signals generated in the second program to the correlated noise at any desired S/N ratio at any phone. Multi-target situations are synthesized by the repeated addition of various signals, and time varying S/N ratios are produced by changing the S/N ratios as a function of time.

Forming of Multiple Beams

The fourth program is designed to form multiple beams from the noise and signals that have been synthesized in the first three programs. A pre-calculated time delay structure for the particular array geometry of interest is used to form any number of desired beams. In addition different non-linear operations may be performed on the data before beamforming so that different systems may be compared under identical noise and signal characteristics. The high order statistics at the beamformer output present considerable analytical difficulties for certain of the non-linear operations, but are of considerable interest in system design and performance, and may be estimated from this program.

Post Beamforming Processing

After beamforming a variety of algorithms exist for multibeam sonars that deal with normalization, detection, tracking and parameter estimation that are easily programmed. Simulations of these algorithms are not meaningful however if the high order statistics are unknown at the inputs, and what their relationship is to the input acoustic field. Therefore they must be an integral part of the total simulation. An example of the difficulties encountered is shown by considering the problem of trying to locate an echo on three adjacent beams. What is the probability of an echo being detected on two of the three adjacent beams? Do false alarms occur at the same time on all three beams? The problems are clearly of a multidimensional nature, where the efficiency of Monte Carlo techniques becomes competitive with theoretical methods.

Statistical Estimation and Utility Routines

The last main program is designed to provide estimates of the relevant statistics at various points in the algorithm or "system". Various plots and graphs are produced and summaries of each run. In addition, various routines are used for tape handling, since tapes are produced between each program so that different algorithms may be compared with the same inputs. Some of these routines have been

detailed in other USL Technical Memorandum^{11,12}

Conclusions

Monte Carlo simulations can provide detailed knowledge of how major and minor changes in an algorithm (and the corresponding special purpose hardware used to implement the algorithm) effect certain sonar performance parameters, under identical and controllable input conditions. Because of this it provides a unique sonar design capability that may be effectively used in trade-off studies. In most cases the simulations run on a general purpose computer will not run as fast as the special purpose hardware, and therefore should not be used in a major data reduction program.

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Existing Software

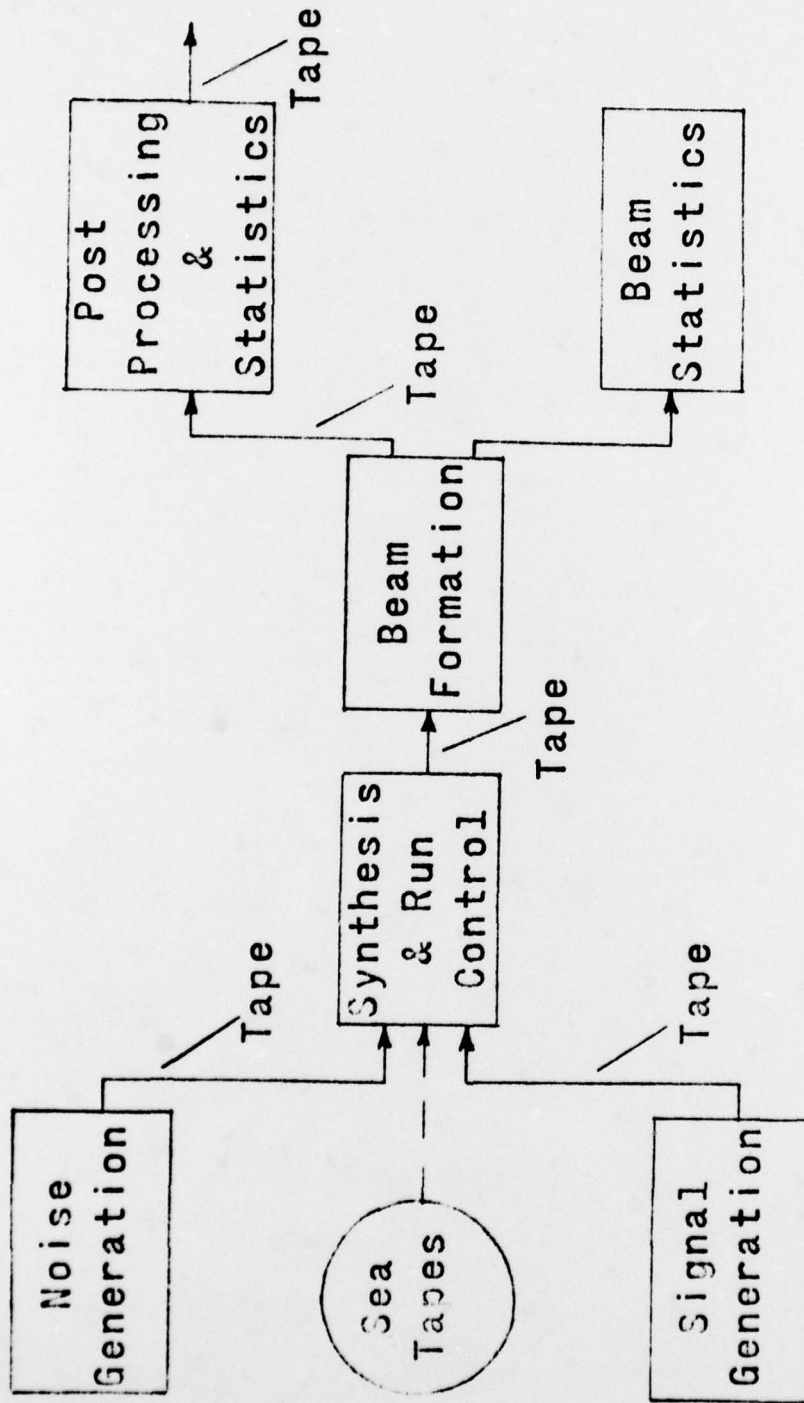
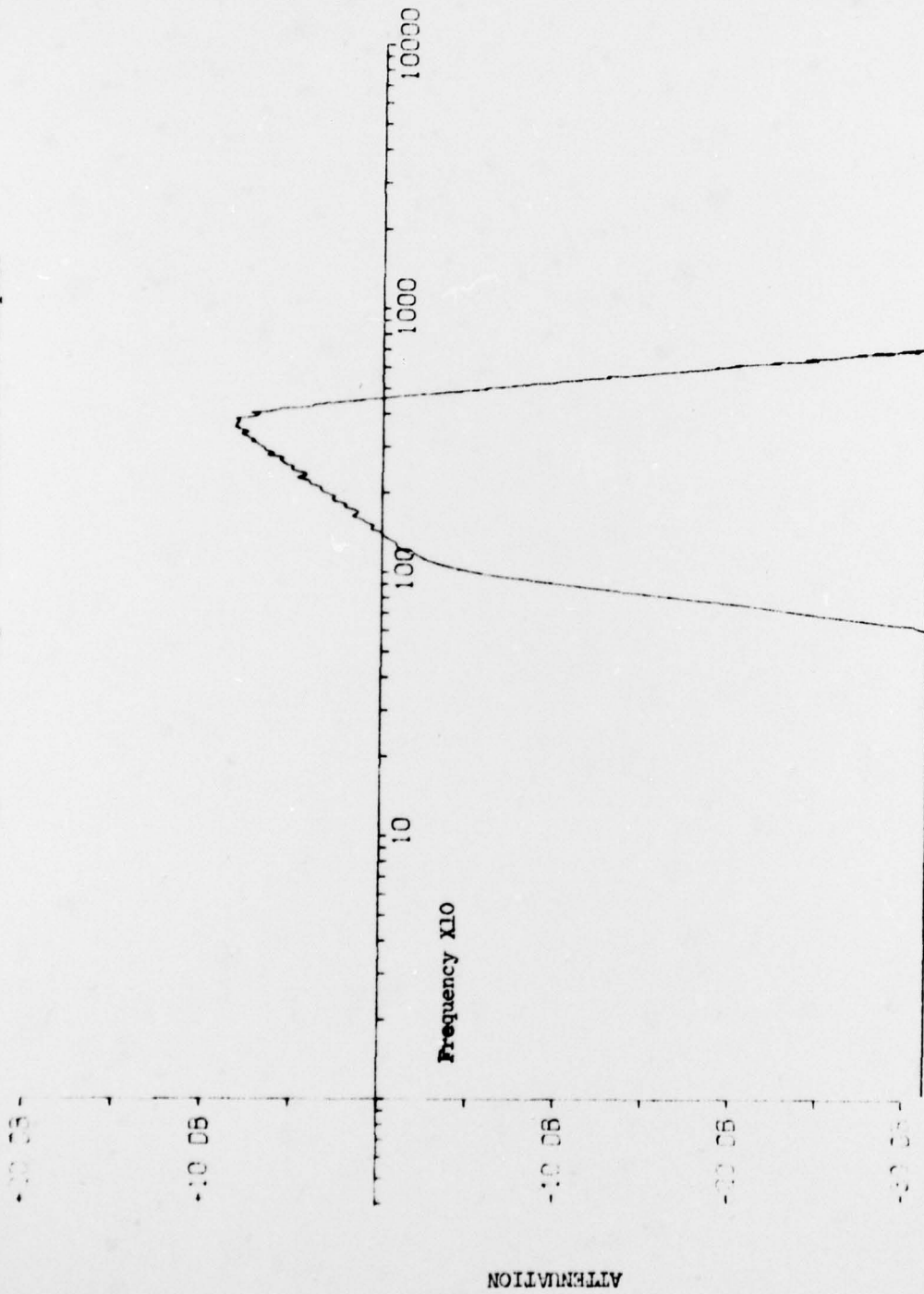


Figure 1.

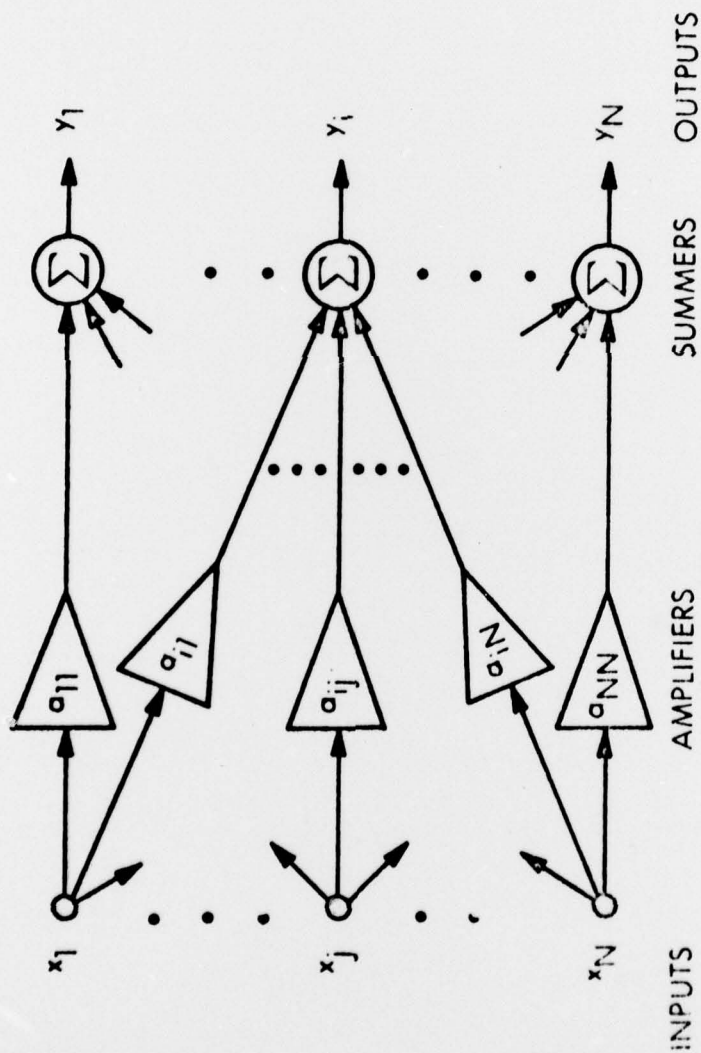
Input Noise Spectrum as Synthesized by
Digital Filtering of Uncorrelated Noise Samples



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No. 2242-380-68

Figure 2.

Synthesis of Spatially Correlated Noise



From E. S. Eby; USL Report 925

Figure 3.