AD-A060 234 NAVY UNDERWATER SOUND LAB NEW LONDON CONN F/G 17/1
A SONAR DESIGN TOOL: MULTI-CHANNEL MONTE CARLO SIMULATION.(U)
OCT 68 R L GORDON
UNCLASSIFIED USL-TM-2242-380-68 NL

| OF |
AD
A060234

END
DATE
FILMED
12-78
DDC

Code No. NW

Copy No. 43

U. S. NAVY UNDERWATER SOUND LABORATORY
FORT TRUMBULL, NEW LONDON, CONNECTICUT

A SONAR DESIGN TOOL: MULTI-CHANNEL MONTE CARLO SIMULATION

by

R. L. Gordon

USL Technical Memorandum No. 2242-380-68

14 October 1968

Technical memo 10p.

USL-TM-2242-380-68

Introduction

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.

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 synthsized 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 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 Chambers[1]. 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 congruental[1] (power residue) method. This method involves multiphying 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 μs 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 polynominal 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 places[3-5] and is founded on Z-transform theory[6]. 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 has the response shown in Figure 2 and needs 17 multiply adds per data point. Using the above techniques the author has been able to produce over 50 channels of independent filtered noise about 20,000 samples long in under two minutes on the UNIVAC 1108.

Synthesis of Signals

This program is designed to create multi-channel signals with any desired characteristics to be added to the synthesized noise. Normally in sonar applications the signal is distinguished by its spatial distribution being a plane wave, and by some spectral shape. To properly form the signal a time delay structure is specified and a single channel of filtered noise, sinusoidal components, etc. is split and properly delayed into the correct number of channels. Since the signal is virtually formed on a sample by sample basis the type of signals formed may vary with time. This allows simulation of fading channels, transients, and echoes. The time delay structure for the array under simulation and the type of signal are data inputs to this program. The signal to noise ratios are controlled at the time the signal is added to the spatially correlated noise produced in the third program.

Synthesis of Hydrophone Inputs

It was desired to structure the signal and noise for this simulation to have a realistic acoustic spatial structure as well as temporal structure. In order to accomplish this a method for producing interhydrophone correlation for the noise field was necessary since some arrays[7,8] are operating in such an environment. The technique for producing the interhydrophone correlation is detailed in a USL Report by Eby[9] and will be reviewed for the reader in a simple form, as used by this program.

Let $e_1(t), e_2(t), - - - , e_N(t)$ represent the voltage at the N hydrophones in question, having a specified covariance matrix $\underline{R} = (r_{ij})$ where $r_{ij} = \overline{e_i(t)e_j(t)}$. The results of the first program produced N independent channels of noise, which will be represented as $x_1(t), x_2(t)$. The noise (uncorrelated between channels) covariance is $\overline{x_i(t)x_j(t)} = 0$ for all $i \neq j$. In order to introduce correlation between the channels an NXN matrix of real frequency independent weights $A = (a_{ij})$, which will be called the connection matrix, must be found. The specified covariance matrix may be expressed in terms of the connection matrix as:

$$\underline{R} = \underline{A}\,\underline{A}^T \qquad (1)$$

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 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 C whose i^{th} row are the elements of the i^{th} eigenvector associated with $\underline{R}$. Then

$$\underline{A} = \underline{D}^{\frac{1}{2}}\underline{C} \tag{2}$$

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

Another method for solving equation (1) due to Marsaglia[10], provides an algorithm for finding the elements of a triangular matrix $\underline{A}$ in terms of the original covariance elements as follows:

$$a_{11} = \sqrt{r_{11}} \tag{3}$$

$$a_{1j} = r_{1j}/a_{11} \tag{4}$$

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

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

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

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

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

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.

R. L. Gordon

R. L. Gordon
Electronic Research Engineer

References

1. Chambers, R. P., "Randon-Number Generation on Digital Computers", IEEE Spectrum, Vol. 4, pp 48-56, February 1967.

2. Muller, M. D., "A Comparison of Methods for Generating Normal Deviates on Digital Computers", J. Assoc. Comput. Mach., Vol. 6, pp. 376-383, 1959.

3. Holtz, H., and Leondes, C. T., "The Synthesis of Recursive Digital Filters", J. ACM, Vol. 13, pp. 262-280, April 1966.

4. Kaiser, J. F., "Design Methods for Sampled Data Filters", 1963 Proc. 1st Allerton Conf., pp. 221-236.

5. Rader, C. M., and Gold, B., "Digital Filter Design Techniques in the Frequency Domain", Proceedings of the IEEE, Vol. 55, No. 2, pp. 149-171, February 1967.

6. Jury, E. I., "Theory and Application of the Z-Transform Method", New York, John Wiley & Sons, Inc., 1964.

7. Marsh, J. A., "Correlation Study on the AN/SQS-26 Sonar Array Aboard the USS McCLOY (DE-1038) (U), USL Report No. 814, 23 May 1967 (Confidential).

8. Schneider, W. A., Newkirk, W. H., and Hileman, J. A., "Analysis of Near-Field Submarine Radiated Noise Measurements from a Towed Array" (U), U. S. Navy Journal of Underwater Acoustics, Vol. 14, No. 4, October 1964, pp. 963-993 (Confidential).

9. Eby, E. S., "Synthesis of Multivariate Gaussian Random Processes with a Preassigned Covariance", USL Report 925, 17 June 1968 (U).

10. Marsaglia, G., "A Note on the Construction of a Multivariate Normal Sample", IRE Trans. on Information Theory, Vol. II-3, p. 149, June 1957.

11. Gordon, R. L., "A Random Number Generator for Use in Fortran V Programs", USL Technical Memorandum No. 2242-320-68, 6 September 1968.

12. Gordon, R. L. and Owsley, N. L., "A General Purpose Multidimensional Fast Fourier Transform for Use in Fortran V Programs*", USL Technical Memorandum No. 2242-319-68, 6 September 1968.

Existing Software

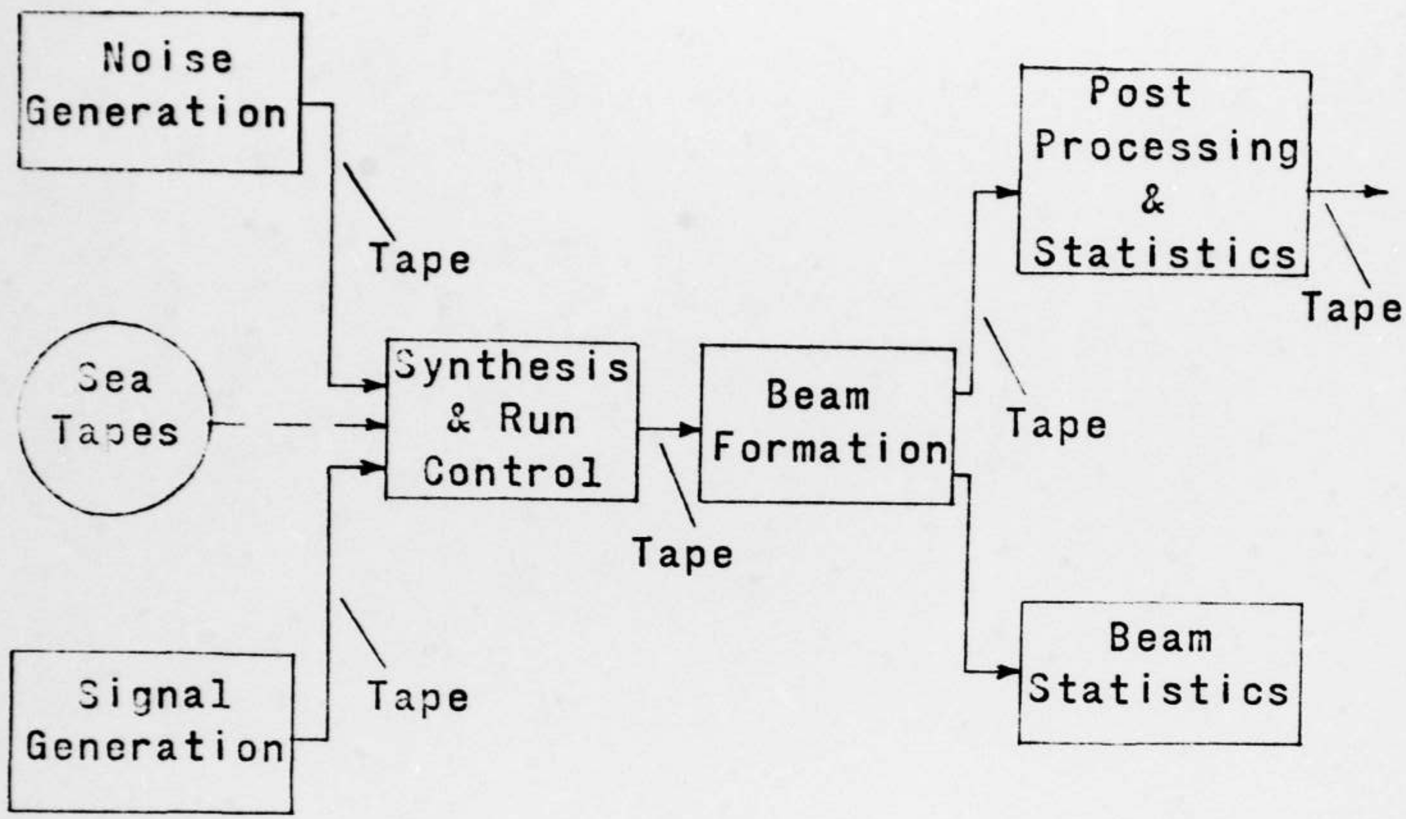


Figure 1.

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

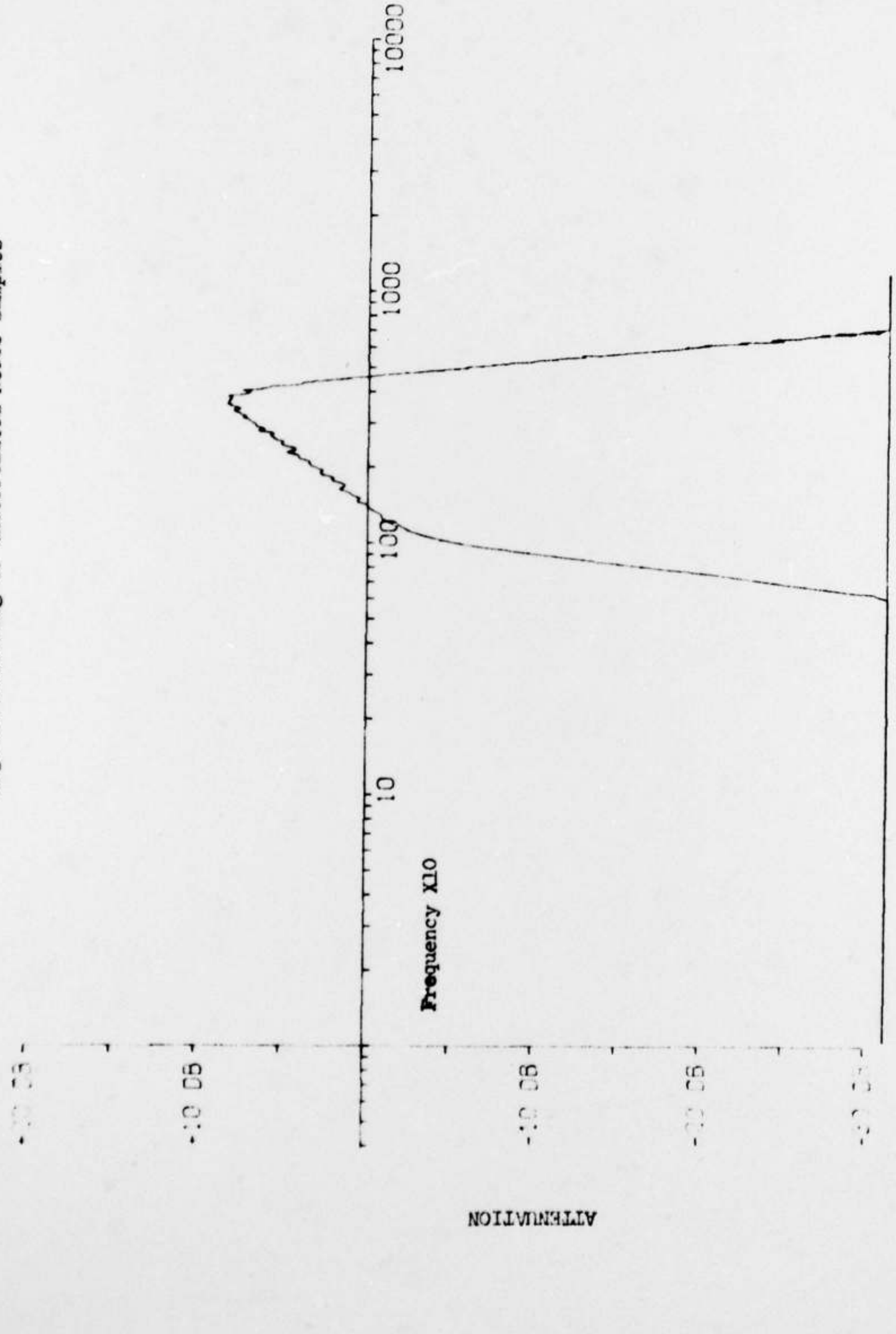
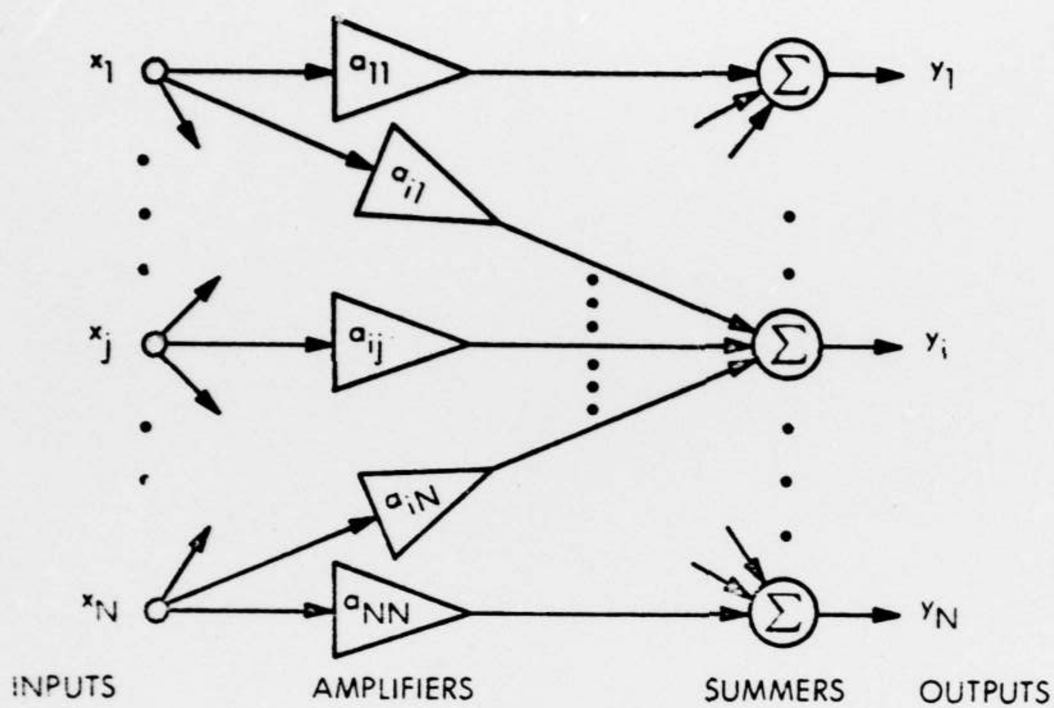


Figure 2.

Synthesis of Spatially Correlated Noise



From E. S. Eby; USL Report 925

Figure 3.