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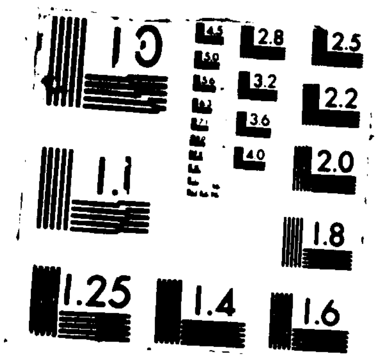
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19. ABSTRACT (Continue on reverse if necessary and identify by block number)

Research was conducted on mathematical problems related to signal processing. An explicit expression for the mean square error in the signal extrapolation problem has been obtained. An interactive algorithm of finding the minimal eigenvector of a symmetric matrix was developed. These and other results are dcoumented in the 15 journal publications, 17 conference papers, and 7 Ph.D. dissertations supported by this grant.

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FINAL SCIENTIFIC REPORT
Grant AF-AFOSR 81-0186
(1 August 1981 - 30 July 1986)

SIGNAL PROCESSING ALGORITHMS

AFOSR-TR- 87-0931

This report summarizes the research conducted under Grant AF-AFOSR 81-0186 during the period 1 August 1981 to 30 July 1986. The numbers in the bracket [ ] refer to publications listed in Appendix A that have been either published, or accepted for publication, or submitted for publication. The prefix j refers to a journal paper and the prefix c refers to a conference paper. Appendix B lists the Ph. D. dissertations supported by the Grant.

1. Signal Extrapolation

A problem often encountered in practice is that of extending a signal beyond the interval over which it is observed. Singular Value Decomposition (SVD) provides a means for determining the solution using the truncated Moore-Penrose inverse. However, this approach requires a large amount of computation and there is some uncertainty as to where the truncation should occur. We have derived an explicit expression for the mean square error, thus providing a guide for proper truncation. We also showed that decimation can be applied in the extrapolation problem to reduce the high computation load without degrading the extrapolation performance [j6,c9,c11].

The ill-posed nature of sub-Nyquist rate for signal reconstruction is studied [j15]. Conjugate gradient method for solving linear equations are attractive in such applications as signal extrapolation, because of its fast convergence relative to other descent methods. Another conjugate direction method, known as A-minimal iterations, is examined and shown to be more robust in the presence of noise [c15].

2. Solving Two Dimension Toeplitz Systems

Perhaps the most important statistical signal processing task is that of linear mean square filtering, which in the stationary case, leads to the familiar Yule-Walker equation of the form: $\mathbf{R}\,\mathbf{x} = \mathbf{y}$, where the correlation matrix $\mathbf{R}$ is Hermitian and Toeplitz. Classical methods requires $O(N^3)$, where $N \times N$ is the size of the matrix. Several $O(N^2)$

algorithms, due to Levinson and others, are known. In the two dimensional case **R** is a Toeplitz-block Toeplitz matrix, consisting of an $N \times N$ Toeplitz arrangement of blocks, each of which is an $M \times M$ Toeplitz matrix. The Levinson and Burg algorithms are not applicable to a Toeplitz-block Toeplitz matrix. A method proposed by us recently [j14] solves the Toeplitz system in two dimensions using the data *directly* without having to estimate the correlation coefficients first. The total number of computation count is $\frac{3}{4}N^2M^3$, a 25% reduction from the previous methods.

3. Computation of Minimum Eigenvalue and Harmonic Retrieval

An iterative algorithm of finding the minimal eigenvector of a symmetric matrix, which is applicable in adaptive methods, is developed. The required computation is $O(n\log n)$ multiplications per iteration using the FFT [c12]. One of the apparent drawbacks to eigenstructure-based (Pisarenko) spectrum estimation is the tendency of the resultant eigen-polynomial to have extraneous roots on the unit circle when the autocorrelation matrix is overdetermined. A complete eigenvector decomposition is computationally expensive. It is shown that good result can be obtained by simply subtracting a small perturbation from the (0,0) element of R prior to finding the minimum eigenvector [c16].

Recent works have described the application of gradient search techniques and iterative eigenvalue algorithms to the problem of implementing Pisarenko Harmonic Retrieval in an adaptive setting. New rotational search algorithms have been applied by us for the same problem and have resulted in roughly a 2/3 saving in computation over previous approaches [j8].

4. Adaptive Signal Processing

Adaptive signal processing has been applied to a wide range of problems, including the suppression of interference in radar processing, anti-jamming, and equalization of communication channels. The steady state output error of the most commonly used least mean square (LMS) adaptive algorithm due to the use of finite precision arithmetic is analyzed [j5,c6]. When a multiple of processors is used to implement the LMS adaptive algorithm in order to obtain high throughput rate, the response is often not immediate due to pipelining and other factors. A delayed gradient estimate LMS algorithm is investigated [j12].

The implementation of adaptive equalizers can be simplified significantly by employing a finite-bit power-of-two quantizer multiplier in which one multiplicant is converted to a 'word' with a single 'one', hence reducing a multiplication to a mere shift. It is shown that, in spite of its very simple implementation, the performance is comparable in many cases with the use of full multipliers [j9,c14].

The major drawback of the well known least mean square adaptive algorithm is its slow convergence when the input autocorrelation matrix has a large eigenvalue spread. A small step size would result in a small excess error, but the convergence is slower. We have studied one scheme with changing step size for each tap, analogous to the adaptive delta-modulation, and showed that the approach provides fast convergence initially and a small final error [c17].

5. Spectral Estimation

A great many problems of extraction information from sensor gathered data involves the central step of estimating the spectrum of some data sequences. It is shown that decimation can be applied to a number of spectral estimation methods to improve the resolution and/or to reduce computation. The reduction in computation is especially significant in dealing with multidimensional signals because of the burdensome computation [j3,c5,c13].

Although in principle, much of the known results in 1-D spectral estimation can be extended to 2-D, the actual carrying out of these extensions are usually not straightforward. Explicit expressions of spectral estimates using periodogram with the Bartlett window, the maximum likelihood method (MLM), and different linear prediction (LP) models are derived for input signal consisting of a finite number of sinusoids in white noise. These expressions are useful in the analytical investigation of the question of resolving closely located sinusoidal signals. Spurious and splitting peaks occur in all spectra using LP modeling, especially for high SNR and/or the model order. It is shown that this problem can be overcome by using a special combination of the LP spectra [j13].

6. Single Chip VLSI Implementation of Digital Filters

As integrated circuit technology enters the very large scale integration (VLSI) era, there is a considerable increase in the activity of implementing digital signal processors in a single VLSI chip. Several special purpose structures that implement bit serial recursive and nonrecursive digital filters have been developed. The input and output data rate of these structures is one bit per clock cycle. Our efforts have been directed toward simple and regular schemes that exhibit high word rate and are efficient in chip area utilization [j10,j11,c7].

Submitted by:

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January 1987

APPENDIX A
Publications Supported by Grant AF-AFOSR 81-0186

JOURNAL PUBLICATIONS

j1. "Transmultiplexer Design Using All-Pass Filters," R. Ansari and B. Liu, *IEEE Trans. Comm.*, Vol Com-30, No. 7, July 1982, pp. 1569-1574.

j2. "Generation of a Random Sequence Having a Jointly Specified Marginal Distribution and Autocovariance," B. Liu and D. C. Munson, Jr., *IEEE Trans. Acous. Sp. Sig. Proc.*, Vol ASSP-30, No. 6, Dec. 1982, pp.973-983.

j3. "Improving Resolution for Autoregressive Spectral Estimation," M. Quirk and B. Liu, *IEEE Trans. Acous. Sp. Sig. Proc.* Vol ASSP-31, June 1983, pp.630-637.

j4. "Efficient Sampling Rate Alteration Using Recursive (IIR) Digital Filters," R. Ansari and B. Liu, *IEEE Trans. Acous. Sp. Sig. Proc.* Vol. ASSP-31, Dec. 1983, pp.1366-1375.

j5. "A Roundoff Error Analysis of the LMS Adaptive Algorithm", C. Caraiscos and B. Liu, *IEEE Trans. Acous. Sp. Sig. Proc.* Vol. ASSP-32, Feb. 1984, pp.34-41.

j6. "On the use of singular value decomposition and decimation in discrete-time band-limited signal extrapolation," B.J. Sullivan and B. Liu, *IEEE Trans. Acous. Sp. Sig. Proc.* Vol. ASSP-32, December 1984, pp. 1202-1212.
This publication was selected for the 1986 IEEE-ASSP Best Paper Award.

j7. "A Class of Low-Noise Computationally Efficient Recursive Digital Filters with Applications to Sampling Rate Alterations", R. Ansari and B. Liu, *IEEE Trans. Acous. Sp. Sig. Proc.* Vol. ASSP-33, February 1985, pp. 90-97.

j8. "Rotational Search Methods for Adaptive Pisarenko Harmonic Retrieval", D.Fuhrmann and B. Liu, *IEEE Trans. Acous. Sp. Sig. Proc.* Vol. ASSP-34, December 1986, pp 1550-1565.

j9. "Adaptive Equalizer Using Finite-Bit Power-Of-Two Quantizer", P. Xue and B. Liu, *IEEE Trans. Acous. Sp. Sig. Proc.* Vol. ASSP-34, December 1986, pp. 1603-1611

j10. "From Digital Filter Flow-Graph to Systolic Arrays", C. Caraiscos and B. Liu, submitted to *IEEE Trans. Acous. Sp. Sig. Proc.*

j11. "Bit-serial VLSI Implementations of FIR and IIR Digital Filters", C. Caraiscos and B. Liu, submitted to *IEEE Trans. Cir. Sys.*

j12. "An LMS Adaptive Algorithm That Uses a Delayed Estimate of the Gradient of the Mean Square Error", C. Caraiscos and B. Liu, submitted to *IEEE Trans. Acous. Sp. Sig. Proc.*

j13. "On Resolving 2-D Sinusoids in White Noise Using Different Spectral Estimates", L. Zou and B. Liu, submitted to *IEEE Trans. Acous. Sp. Sig. Proc.*

j14. "An Efficient Algorithm for Two Dimensional Autoregressive Spectrum Estimation", B.F. McGuffin and B. Liu, submitted to *IEEE Trans. Acous. Sp. Sig. Proc.*

j15. "The Ill-posed Nature of a Method for Sub-Nyquist Rate Signal Reconstruction", B. Sullivan and B. Liu, submitted to *IEEE Trans. Cir. Sys.*

CONFERENCE PAPERS

c1. "A Class of Low Noise Computationally Efficient Recursive Digital Filters," R. Ansari and B. Liu, *1981 IEEE Int. Symp. on Circ. and Sys.*, Vol. 2 of 3, April 1981, pp. 550-553.

c2. "Interpolators and Decimators as Periodically Time-Varying Filters," R. Ansari and B. Liu, *IEEE Int. Symp. on Circ. and Sys.*, April 1981, pp. 447-450.

c3. "Two Dimensional DFT Using Mixed Time and Frequency Decimations," C. Caraiscos and B. Liu, *Proceedings IEEE Int. Conf. Acous. Sp. Sig. Proc.*, May 1982, pp. 24-27.

c4. "Quantization Effects in Computationally Efficient Realizations of Recursive Filters," B. Liu and R. Ansari, *Proceedings IEEE Int. Symp. Cir. & Sys.*, May 1982, pp.

c5. "On the Resolution of Autoregressive Spectral Estimation", M. Quirk and B. Liu, *IEEE Int. Conf. Acous. Sp. Sig. Proc.* April 1983, pp. 1095-1098.

c6. "A Round-off Error Analysis of the LMS algorithm", C. Caraiscos and B. Liu, *IEEE Int. Conf. Acous. Sp. Sig. Proc.* April, 1983, pp. 29-32.

c7. "Bit Serial VLSI Implementation of FIR and IIR Digital Filters," C. Caraiscos and B. Liu, *IEEE Int. Symp. Cir. Sys.* May 1983, pp. 717-721.

c8. "Approximating the Eigenvectors of a Symmetric Toeplitz Matrix", D.R. Fuhrmann and B. Liu, *Proc. 21st Ann. Allerton Conf. on Comm., Control, and Comp.*, Oct. 1983.

c9 "Solving ill-conditioned systems using singular value decomposition with application to signal extrapolation," B. J. Sullivan and B. Liu, *Proc. 21st Annual Allerton Conf. on Comm., Control, and Comp.*, Oct. 1983.

c10. "An Iterative Algorithm for Finding the Minimum Eigenvalue of a Class of Symmetric Matrices", D.R. Fuhrmann and B. Liu, *IEEE ICASSP 84*, March 1984.

c11. "Extrapolation of discrete-time band-limited signals using singular value decomposition with decimation," B. J. Sullivan and B. Liu, *Proc. IEEE Int'l. Conf. on Acoust., Speech, and Signal Proc.*, March, 1984;pp. 31.3.1-31.3.4

c12. "An Iterative Algorithm for Locating the Minimal Eigenvector of a Symmetric Matrix", D. Fuhrmann and B. Liu, *IEEE Int. Conf. Acous. Sp. Sig.Proc.*, March 1984; pp. 45.8.1 - 45.8.4

c13. "Improvement of Resolution and Reduction of Computation in 2D Spectral Estimation Using Decimation", L. Zou and B. Liu, IEEE Int. Conf. Acous. Sp. Sig. Proc., March 1984; pp. 4.7.1 - 4.7.4

c14. "Adaptive Equalizer Using Finite-Bit Power of Two Quantizers", P. Xue and B. Liu, IEEE Int. Conf. Acous. Sp. Sig. Proc., March 1984; pp. 46.9.1 - 46.9.4

c15. "A Robust Conjugate Gradient Method for Solving Linear Systems", B.J. Sullivan and B. Liu, Proc. 22nd Annual Allerton Conf. on Comm., Control, and Comp., Oct. 1984

c16. "A Perturbation Approach to Improving Pisarenko Harmonic Retrieval", D.R. Fuhrmann and B. Liu, 22bd Annual Allerton Conf. on Comm., Control, and Comp., Oct. 1984

c17. "On the Convergence of a Variable Step Size Adaptive Filter", P. Xue and B. Liu, IEEE Int. Symp. Cir. Sys., June 1985, pp. 1661-1662

APPENDIX B
Ph.D Dissertations Supported Fully or Partially by Grant AF-AFOSR 81-0186

1. "Low Noise Computationally Efficient Algorithms and Their Application"
   R. Ansari, 1981. (University of Pennsylvania, Dept. of Electrical Engineering)

2. "Efficient Computation of Narrowband Spectra"
   M. Quirk, 1982. (Institute for Defense Analysis, Princeton, NJ)

3. "Implementation Issues of Digital Signal Processing Algorithm"
   C. Caraiscos, 1983 (National Technical University, Athens, Greece)

4. "Fast Eigenvector Methods for Digital Signal Processing"
   D. Fuhrmann, 1984 (Washington University, Dept. of Electrical Engineering)

5. "Solving Ill-Conditioned Problems in Signal Processing"
   B. Sullivan, 1984 (Northwestern University, Dept. of Electrical Engineering)

6. "Adaptive Filtering - Implementation and Performance"
   P. Xue, 1985 (RCA Laboratories, Princeton, NJ)

7. "Aspects of Two Dimensional Spectrum Estimation"
   B.F. McGuffin, 1986 (Stanford Telecommunications, Washington, DC)

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