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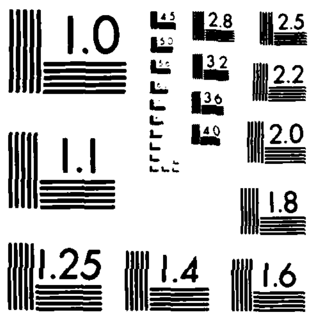

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Interim

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Grant AF-AFOSR 81-0186

SIGNAL PROCESSING ALGORITHMS

(1 August 1982 - 30 July 1983)

This interim scientific report summarizes the research conducted under Grant AF-AFOSR 81-0186 during the period 1 August 1982 to 30 July 1983. The number in the bracket [ ] refer to the items in the attached list of publications.

It is shown that using decimation by D with an autoregressive model of order M for spectrum estimation yields the same resolution as a model order MD used with the undecimated signal, and that decimation reduces the computation. An expression of the autoregressive spectrum for K complex sinusoids in additive white noise is derived. This expression is used to study the resolution question in terms of the model order and the required signal to noise ratio [1,2]. Extension to higher dimension is carried out [3], as well as extentions to the maximum likelihood method and the use of autoregressive moving average model [3,4].

The steady state output error of the least mean square (LMS) adaptive algorithm due to the use of finite precision arithmetic is analyzed [5]. The relationship between the quantization error and the error which occurs when adaptation may cease due to quantization is also investigated. 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 [6]. The implementation of adaptive equalizers can be simplified significantly by employing a finite-bit power-of-two quantizer. The performance of such an adaptive system is shown to be comparable with that using a full multiplier [7].

A common approach to extrapolate a bandlimited signal in discrete-time is to solve an under-determined system of linear equations. Singular value decomposition (SVD) provides a means for determining the solution using the Moore-Penrose inverse. An expression for the mean square error is derived [8,9]. The expression indicates that decimation can be applied in the extrapolation problem to reduce the high computation cost of SVD without degrading the extrapolation. Both one- and multi-dimension cases are investigated [9].

The minimum eigenvalue and its associated eigenvector of a symmetric Toeplitz matrix are of interest in a number of applications, including the sinusoidal in noise spectrum estimation problem. A fast iterative algorithm is developed for this purpose [10]. As in any iterative scheme, a good starting point would greatly enhance this method. A one-pass method for approximating eigenvectors associated with small eigenvalues is found [11], which requires $O(n^2)$ multiplications where n is the size of the Toeplitz matrix. As a by product, we get reasonable approximations to all the eigenvalues as well as upper and lower bounds on the smallest and largest eigenvalues respectively.

A preliminary investigation of implementation of digital filters directly on a single chip or a small number of chips connected in an array has begun [12]. The emphasis is on bit serial arithmetics and on the operation at the bit level.

Submitted by:

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Department of Electrical Engineering
and Computer Science
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October, 1983

PUBLICATIONS

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

2. "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.

3. "Improvement of Resolution and Reduction of Competition in 2D Spectral Estimation Using Decimation," L.H. Zou and B. Liu, submitted to the *1984 IEEE Int. Conf. Acous. Sp. Sig. Proc.*

4. *Efficient Computation of Narrow-Band Spectra*, M.Quirk, Dissertation, Princeton University, 1982.

5. "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, (also to appear in *IEEE Trans. Acous. Sp. Sig. Proc.*, Vol ASSP-31).

6. "A Delayed Gradient Estimate LMS Adaptive Algorithm," C. Caraiscos and B. Liu, to be submitted.

7. "Adaptive Equalizer Using Finite-Bit Power-of-Two Quantizers," P. Xue and B. Liu, submitted to the *1984 IEEE Int. Conf. Acous. Sp. Sig. Proc.*

8. "Solving Ill-Conditioned Systems Using Singular Value Decomposition with Application to Signal Extrapolation," B.J. Sullivan and B. Liu, *1983 Allerton Conf. on Comm. Contr. and Comp.*

9. "On the Use of Singular Value Decomposition and Decimation in Discrete-Time Band-Limited Signal Extrapolation," B.J. Sullivan and B. Liu, submitted to *IEEE Trans. on Acous. Sp. Sig. Proc.*

10. "An Iterative Algorithm for Finding the Minimum Eigenvalue of a Class of Symmetric Matrices," D. Fuhrmann and B. Liu, submitted to *1984 IEEE Int. Conf. Acous. Sp. Sig. Proc.*

11. "Approximating the Eigenvectors of a Symmetric Toeplitz Matrix," by D. Fuhrmann and B. Liu, *1983 Allerton Conf. on Comm. Contr. and Comp.*

12. "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.

13. "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.

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This report summarizes research under the grant during this period in the areas of autoregressive model for spectrum estimation, steady state output error of the least mean square, and extrapolation of bandlimited signal in discrete-time. Papers produced during this period are listed.

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