Digital Signal Processing Research Group

Academic and Research Staff
Professor Alan V. Oppenheim, Professor Arthur B. Baggeroer, Dr. Charles E. Rohrs.

Visiting Scientists and Research Affiliates
Dr. Dan E. Dudgeon[1], Dr. Yonina Eldar[2], Dr. Ehud Weinstein[3], Dr. Maya R. Said[4]

Graduate Students
Thomas Baran, Sourav Dey, Zahi Karam, Jon Paul Kitchens, Shay Maymon, Melanie Rudoy, Joseph Sikora III, Archana Venkataraman, Dennis Wei.

Technical and Support Staff
Eric Strattman

Introduction

The Digital Signal Processing Group develops signal processing algorithms that span a wide variety of application areas including speech and image processing, sensor networks, communications, radar and sonar. Our primary focus is on algorithm development in general, with the applications serving as motivating contexts. Our approach to new algorithms includes some unconventional directions, such as algorithms based on fractal signals, chaotic behavior in nonlinear dynamical systems, quantum mechanics and biology in addition to the more conventional areas of signal modeling, quantization, parameter estimation, sampling and signal representation. When developing new algorithms, we often look to nature for inspiration and as a metaphor for new signal processing directions.

[1] BAE Systems IEWS, Senior Principal Systems Engineer, Nashua, New Hampshire.
[2] Department of Electrical Engineering, Faculty of Engineering, Technion-Israel Institute of Technology, Israel.
[3] Department of Electrical Engineering, Systems Division, Faculty of Engineering, Tel-Aviv University, Israel; adjunct scientist, Department of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, Massachusetts.
[4] Visiting Scientist, Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, Massachusetts.

1. The Thermodynamics of Signal Flow Graphs

Sponsors
MIT Lincoln Laboratory Signal Processing Research P.O. No. 3118144
MIT Lincoln Laboratory Signal Processing Research P.O. No. 7000032361
BAE Systems, Inc. P.O. No. 112991
Texas Instruments, Inc. Leadership University Program
Bose Corporation

Project Staff
Thomas Baran, Professor Alan V. Oppenheim

Signal flow graphs are used widely to design, analyze and describe digital and analog systems. The representation is applicable to any system where a notion of causation is involved, so the concept is used in fields ranging from electrical engineering to cell biology. This project examines the relationship between thermodynamic systems and the signal flow graphs that represent them. In particular, we draw on a rich body of thermodynamic results to make useful statements about the behavior of systems described by signal flow graphs.

Recent focus has been directed toward the problem of system inversion. We show that systems represented by a particular class of signal flow graphs have inverse systems that are found by making simple graph manipulations. The result therefore implies a duality between two classes of flow graphs, a principle that may be applicable to the synthesis of linear and nonlinear systems.

2. Randomized Sampling and Filtering and Multiplier-Less Filtering

Sponsors
Texas Instruments, Inc. Leadership University Program
National Defense Science and Engineering Graduate Fellowship (NDSEG)
BAE Systems, Inc. P.O. No. 112991

Project Staff
Sourav Dey, Professor Alan V. Oppenheim

This project considers the benefits of randomization in two fundamental signal processing techniques: sampling and filtering. The first part develops randomized non-uniform sampling as a method to mitigate the effects of aliasing. Randomization of the sampling times is shown to convert aliasing error due to uniform under-sampling into uncorrelated shapeable noise. In certain applications, especially perceptual ones, this form of error may be preferable.

Two sampling structures are developed in this project. In the first, denoted simple randomized sampling, non-white sampling processes can be designed to frequency-shape the error spectrum, so that its power is minimized in the band of interest. In the second model, denoted filtered randomized sampling, a pre-filter, post-filter, and the sampling process can be designed to further frequency-shape the error to improve performance. The project develops design techniques using parametric binary process models to optimize the performance of randomized non-uniform sampling. In addition, a detailed second-order error analysis, including performance bounds and results from simulation, is presented for each type of sampling.

The second part of this project develops randomization as a method to improve the performance of multiplier-less FIR filters. Static multiplier-less filters, even when carefully designed, result in frequency distortion as compared to a desired continuous-valued filter. Replacing each static tap with a binary random process is shown to mitigate this distortion, converting the error into uncorrelated shapeable noise. As with randomized sampling, in certain applications this form of error may be preferable.

This project presents a FIR Direct Form I randomized multiplier-less filter structure denoted binary randomized filtering (BRF). In its most general form, BRF incorporates over-sampling combined with a tapped delay-line that changes in time according to a binary vector process. The time and tap correlation of the binary vector process can be designed to improve the error performance. The project develops design techniques using parametric binary vector process models to do so. In addition, a detailed second-order error analysis, including performance bounds, error scaling with over-sampling, and results from simulation, is presented for the various forms of BRF.

3. Fast Discriminative Training of GMMs

Sponsors
Texas Instruments, Inc. Leadership University Program
BAE Systems, Inc. P.O. No. 112991
MIT Lincoln Laboratory Signal Processing Research P.O. No. 3118144
MIT Lincoln Laboratory Signal Processing Research P.O. No. 7000032361

Project Staff
Zahi Karam, Dr. William Campbell (Lincoln Labs), Professor Alan V. Oppenheim

Gaussian mixture models (GMMs) are used in many classification tasks: speaker recognition, language identification, etc. Typically, a GMM is trained, using maximum-likelihood (ML) training, to model each of the classes. New data is then classified based on which of the class-dependent GMM is most likely to generate it.

ML training ignores the ultimate goal of minimizing classification error and instead focuses on modeling the individual classes. This focus yields GMMs that are poorly suited for the classification task. Discriminative training techniques, such as maximum mutual information (MMI) and minimum classification error (MCE), have been developed to train better suited GMMs by trading-off modeling the individual classes and minimizing classification error. The utility of these techniques, however, can in some cases be diminished by their computation complexity. Our focus is on fast discriminative training techniques for GMMs. Our approach uses support vector machines (SVMs) to modify ML trained GMMs so as to improve their discriminative power, thus rendering them better suited for classification tasks.

4. Acoustic Vector-Sensor Array Performance

Sponsors
MIT Lincoln Laboratory, Lincoln Scholars Program

Project Staff
Jon Paul Kitchens, Professor Arthur Baggeroer, Dr. Stephen Kogon.

Classical hydrophones measure pressure only, but acoustic vector-sensors also measure particle velocity. Velocity measurements can increase array gain and resolve ambiguities, but make vector-sensor arrays more difficult to analyze and susceptible to additional modeling errors. Most introductions to array processing characterize classical hydrophone arrays through performance measures that prove useful in both theory and practice.

This research derives a new set of useful performance measures for acoustic vector-sensor arrays. It characterizes the vector-sensor array beampattern with and without modeling errors, or “mismatch.” The resulting beampattern expressions relate vector-sensor arrays to pressure-sensor arrays. This research also develops a hybrid Cramér-Rao bound for direction-of-arrival estimation under mismatch. The results are analyzed, compared to Monte-Carlo simulations, and explored for insight in the published thesis. The tools developed should provide better design and performance prediction before vector-sensor array construction as well as improved analysis afterward.

5. NonUniform Sampling

Sponsors
Texas Instruments, Inc. Leadership University Program

Project Staff
Shay Maymon, Professor Alan V. Oppenheim

Our work explores the NonUniform Sampling process and reconstruction. It represents the nonuniform grid of samples as random perturbations around a uniform grid with nominal sampling interval. A simpler second order equivalent system to the nonuniform sampling process is suggested. This model contains pre-filtering before uniform sampling with white noise added at its output. The samples grid can be designed so that aliasing is avoided at the output of the sampling process. This will be referred to as "Anti-Aliasing Distribution" because of the similarity with the Anti-Aliasing Filter Uniform preceeding sampling.

Uniform sampling guarantees perfect reconstruction if the sampling rate is above the Nyquist rate. However, if the Nyquist condition does not hold, we cannot recover the original signal in general. However, if the signal is nonuniformly sampled, this can be taken into advantage. We have shown that the spectrum of the discrete time signal after nonuniform sampling contains information about the original continuous time signal. This information can be used not only to determine if the signal is undersampled or not, but also to recover the original signal under some conditions.

A Key problem with nonuniform sampling is the reconstruction process. Roughly speaking, the continuous time signal can be perfectly recovered from its nonuniform samples as long as the nominal sampling rate is greater than the Nyquist rate. However, the reconstruction method is very complicated and it is not time invariant.

We suggest simpler sub-optimal reconstruction methods, analyze and compare their performance in terms of the Mean Square Error (MSE) criterion. One method uses information about the sample locations for the reconstruction while the other ignores it. It is shown that for some cases we would prefer to ignore this information in order to have better performance. The performance of both methods depends on the signal spectrum as well as on the perturbations probability density function. One can think on how to design the Input signal spectrum in order to match the nonuniform grid so that to minimize the MSE. Another scenario is to choose a nonuniform grid so that for a given spectrum of the input signal the MSE will be minimized.

None of the methods suggested above is optimal for all kind of signals. These methods can be viewed as special cases of a more generalized reconstruction method which we refer to as "Randomized Reconstruction". In this method, we locate the samples in a nonuniform grid which can be related to the original NonUniform Sampling grid but may be different and optimize over this grid to get minimum MSE.

6. Signatures of Walking Humans Using Passive and Active Acoustic Sensors

Sponsors
Army Research Laboratory (ARL) Collaborative Technology Alliance Contract RP 6891

Georgia Institute of Technology Award No. E-21-6RT-G2, MURI under ARO Prime Award No. W911NF-04-1-0190

Project Staff
Melanie B. Rudoy, Dr. Charles E. Rohrs

This research aims to characterize the signature of a walking human target using passive acoustic and active ultrasound sensor data. The ability to detect moving human targets is crucial in many applications, ranging from intrusion detection to automotive safety systems to situational awareness enhancement in military applications. A sensor fusion framework was developed that combined data from two independent sensor modalities, and was used to compute local estimates of the acoustic energy of the footsteps and the velocity of the subject's torso and limbs. A time-varying vector autoregression (TV-VAR) was then used to model the evolution of these signals, which captures the physical correlations between them, creating a natural data fusion across different sensor modalities. The human signature was defined as a subset of the parameters from the TV-VAR model, and the quality of this feature set was evaluated using a support vector machine framework to classify multiple test subjects for both detection and discrimination applications.

7. Multistage Mean-Variance Portfolio Selection in Cointegrated Vector Autoregressive Systems

Sponsors
Texas Instruments, Inc. Leadership University Program
Georgia Institute of Technology Award No. E-21-6RT-G2, MURI under ARO Prime Award No. W911NF-04-1-0190

Project Staff
Melanie B. Rudoy, Dr. Charles E. Rohrs

This research is concerned with the application of stochastic signal processing and control theory to financial decision making. Specifically, the problem of multistage portfolio selection within a restricted universe of financial assets is considered. It is assumed that the prices (or log-prices) of the underlying assets are well modeled by a particular type of linear system, known as a cointegrated vector autoregressive process. Here, the random process modeling each component of the vector process is nonstationary due to the presence of one or more poles at unity (integrators), resulting in a variance that grows linearly with time. However, it is possible to find a linear combination of the signals that is wide sense stationary. Constraining the price dynamics to follow such a model in turn induces a probability distribution on the asset returns, which can then be exploited to achieve a higher average rate of return on the investment portfolio. A set of explicit solutions to the cointegrated asset allocation problem under varying assumptions are derived, and resulting portfolio weight vectors are characterized in relation to the underlying geometry of the cointegration system.

8. Sound Wave Propagation around Underwater Seamounts

Sponsors
Office of Naval Research, ONR contract No. N00014-04-1-0124

Project Staff
Joseph Sikora III, Hyun Joe Kim, Professor Arthur B. Baggeroer, Edward Scheer (WHOI), Keith Von Der Heydt (WHOI)

In the ocean, low frequency acoustic waves propagate with low attenuation and cylindrical spreading loss over long range, making them an effective tool for underwater source localization, tomography, and communications. Underwater mountains, or *seamounts*, are ubiquitous throughout the world's oceans. Seamounts can absorb and scatter acoustic energy, presenting many interesting acoustic modeling challenges. The goal of the subject research is to measure the scatter field of a large, conical seamount at long-range, and reconcile observations with 2-D and 3-D range-dependent acoustic models, for the purpose of adapting to, and utilizing highly range-dependent bathymetry.

The Basin Acoustic Seamount Scattering Experiment (BASSEX) experiment was conducted to measure the scatter field of the seamounts in the Kermit-Roosevelt Seamount Complex during September and October of 2004. Current results from the BASSEX experiment show a strong correlation between observed seamount scatter fields and simulated data generated with the RAM acoustic modeler. The seamounts absorbed and reflected high order acoustic modes, forming strong convergence zones in the forward scatter field. Convergence zones in the forward scatter field contained both refracted and reflected acoustic energy, and appeared in the upper water column with 50 km periodicity in range. Refracted acoustic energy strength, arrival time, and arrival angle were accurately predicted using both the RAY and RAM acoustic models. Reflected acoustic ray arrival patterns have "open-ocean" ray group structure, with some random intensity fluctuation and time delay, correlating well with RAM simulations. Further analysis will include generalizing the effects of different seamount characteristics and environmental parameters on the scatter field, and will investigate side scatter acoustic data.

Robust adaptive beamforming is used to process hydrophone array data gathered in the BASSEX experiment. The ambient noise field in the ocean and noise from the tow ship lower the data's signal to noise ratio. Non-stationarity in the observed noise field caused by array fluctuations and data acquisition system malfunctions motivate a time-varying Capon adaptive beamformer, and strong acoustic harmonics from ship operations motivate a frequency and steering angle dependent white noise gain constraint.

In an effort to process snap-shot deficient data sets, the novel physically constrained maximum likelihood (PCML) beamformer was further developed and applied. To improve computational efficiency, orthonormal array manifold vectors are used to model and determine the maximum likelihood spectral covariance matrix. Figure 1(a) shows the array gain of a 10 sensor array using the Capon (A_o), conventional (A_c), and PCML (A_{PCML}) beamformers, for a 20 dB plane wave interference at wavenumber u_I, where BW_{NN} is null-to-null beamwidth. The PCML beamformer gain shows improved sidelobe control and resolution compared with the conventional beamformer. Figure 1(b) shows the array gain of the adaptive beamformers using a snap-shot deficient spectral covariance matrix estimate. The figure demonstrates that the PCML beamfomer has higher array gain in the sidelobe region than the Capon and conventional beamformer using an underdetermined spectral covariance matrix.

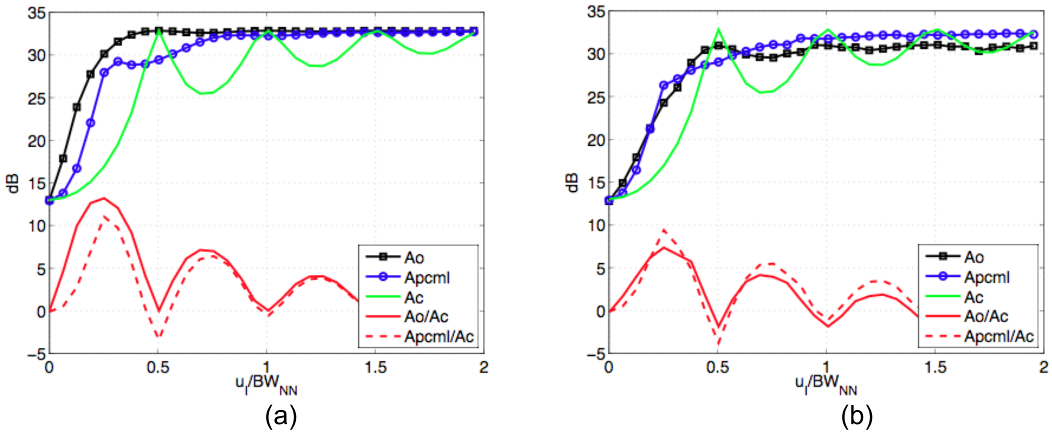

Figure 1 - Array beamformer gain using: (a) the actual, full-rank spectral covariance matrix, (b) the snap-shot deficient, spectral covariance matrix estimate, averaged over 100 samples.

9. Bilinear Sampling of Continuous-Time Signals for Matched Filtering Applications

Sponsors
Texas Instruments, Inc. Leadership University Program
BAE Systems, Inc. P.O. No. 112991
MIT Lincoln Laboratory Signal Processing Research P.O. No. 3118144
MIT Lincoln Laboratory Signal Processing Research P.O. No. 7000032361

Project Staff
Archana Venkataraman, Professor Alan V. Oppenheim

Matched filtering is an important technique in signal processing. Among the many matched filtering applications is detecting the presence of a desired signal based on analyzing a noise-corrupted received signal. Under certain conditions, the optimal decision rule reduces to comparing the inner product of the desired and received signals with some threshold value. One commonly used method for computing the inner product is by Nyquist sampling of the two continuous-time signals and taking the inner product of the resulting discrete-time sequences. However, Nyquist sampling requires that the signals be appropriately band-limited. Consequently, it may not be possible to represent potentially important high-frequency information. This work explores an alternative method of sampling which uses the bilinear transform to map the jω-axis in the continuous-time S-plane onto the unit circle in the discrete-time Z-plane. Since the entire frequency range is mapped from one domain to another, this representation eliminates the constraint on the signal bandwidth. We characterize the tradeoffs between Nyquist sampling and the bilinear transform as it applies to computing the inner product of two continuous-time signals. Simulation results indicate that for several classes of signals and for a fixed number of discrete-time coefficients, the use of bilinear sampling to compute the inner product achieves a higher probability of detection than using Nyquist sampling.

10. Design of Sparse Finite Impulse Response Filters

Sponsors
BAE Systems, Inc. P.O. No. 112991
MIT Lincoln Laboratory Signal Processing Research P.O. No. 3118144
MIT Lincoln Laboratory Signal Processing Research P.O. No. 7000032361

Texas Instruments, Inc. Leadership University Program

Project Staff
Dennis Wei, Professor Alan V. Oppenheim

Our recent work focuses on the design of sparse finite impulse response (FIR) filters, i.e., filters with few non-zero coefficients. Sparse filters are often desirable because of their reduced complexity due to the elimination of arithmetic operations or circuit components corresponding to zero-valued coefficients. In the analogous context of linear sensor arrays, sparse designs permit reductions in the number of array elements, yielding savings in hardware, processing, and power.

The exact problem of determining the sparsest filter subject to a set of specifications is computationally difficult. Accordingly, we have developed an approximate approach called successive thinning that can yield reasonably sparse filters in an efficient manner. The method begins with a non-sparse design and successively constrains more and more coefficients to zero values, with the goal of removing as many non-zero coefficients as possible while preserving feasibility. A number of heuristic rules can be used effectively to select coefficients to constrain to zero. Preliminary simulations show that successive thinning can achieve significant levels of sparsity with low design complexity.

We have also begun exploring an alternative approach to designing sparse filters based on the class of functions that are the extension of the p-norms for $0 < p < 1$. This approach builds upon previous work in the Digital Signal Processing Group[5] investigating the use of the 1-norm as an approximate but computationally efficient measure of sparsity. Theoretical analysis and algorithm development along these lines is ongoing.

Publications

Journal Articles, Published

A.V. Oppenheim, P. Boufounos and V.K. Goyal, "Causal Compensation for Erasures in Frame Representations," *IEEE Transactions on Signal Processing*, 56(3): 1053-82 (2008).

Meeting Papers

T. A. Baran and A.V. Oppenheim, "Design and Implementation of Discrete-Time Filters For Efficient Rate-Conversion Systems," *Proceedings of the 41st Annual Asilomar Conference on Signals, Systems, and Computers*, Asilomar, California, November 4-7, 2007.

Z. Karam and A.V. Oppenheim, "Computation of the One-Dimensional Unwrapped Phase," *Proceedings of the 2007 15th International Conference on Digital Signal Processing (DSP 2007)*, Cardiff, Wales, UK, July 1-4, 2007.

Z. N. Karam and W. M. Campbell, "A New Kernel for SVM MLLR based Speaker Recognition," *Proceedings of Interspeech*, Antwerp, Belgium, August 27-31, 2007.

Z. N. Karam and W. M. Campbell, "A Multi-Class MLLR Kernel for SVM Speaker Recognition," *Proceedings of the IEEE ICASSP*, Las Vegas, Nevada, March 30 - April 4, 2008.

---

[1] T. A. Baran and A.V. Oppenheim, "Design and Implementation of Discrete-Time Filters For Efficient Rate-Conversion Systems," Proceedings of the 41st Annual Asilomar Conference on Signals, Systems, and Computers, Asilomar, California, November 4-7, 2007.

M.B. Rudoy, C.E. Rohrs, and J. Chen, "Signatures of Walking Humans from Passive and Active Acoustic Data using Time-Varying Vector Autoregressions," *Proceedings of the 41st Annual Asilomar Conference on Signals, Systems, and Computers*, Asilomar, California, November 4-7, 2007.

A. Venkataraman and A. V. Oppenheim, "Signal Approximation Using the Bilinear Transform," *Proceedings of the IEEE ICASSP*, Las Vegas, Nevada, March 30 - April 4, 2008.

D. Wei and A.V. Oppenheim, “Sampling Based on Local Bandwidth”, *Proceedings of the 41st Annual Asilomar Conference on Signals, Systems, and Computers*, Asilomar, California, November 4-7, 2007.

Theses

S.R. Dey, *Randomized Sampling and Multiplier-Less Filtering*, Ph.D diss., Department of Electrical Engineering and Computer Science, MIT, 2008.

J. P. Kitchens, *Acoustic Vector-Sensor Array Performance*, S. M. thesis, Department of Electrical Engineering and Computer Science, MIT, 2008.

A. Venkataraman, *Signal Approximation using the Bilinear Transform*, M. Eng. thesis, Department of Electrical Engineering and Computer Science, MIT, 2007.