Digital Signal Processing Research Program

Academic and Research Staff
Professor Alan V. Oppenheim, Professor Arthur B. Baggeroer

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

Graduate Students
Petros Boufounos, Sourav Dey, Zahi Karam, Alaa Kharbouch, Joonsung Lee, Maya Said, Charles Sestok, Joseph Sikora III, Daniel Turek

Technical and Support Staff
Alecia Batson, Angela Glass, Eric Strattman

Introduction

Our current focus is on the development of signal processing algorithms with the view that signal processing is a technology which spans a host of application areas. The commitment in our current research is to the algorithms in general rather than to specific applications. Consistently, our group has successfully looked to application areas such as speech and image processing, sensor networks, communications, and radar and sonar, to name a few.

Much of our research is involved in the development of algorithms in traditional areas such as signal modelling, quantization, parameter estimation, sampling and signal representations. We also successfully explore unconventional directions such as algorithms based on fractal signals, chaotic behavior in nonlinear dynamical systems, quantum mechanics etc.

Another approach we often take in developing new algorithms is to look to nature for inspiration and as a metaphor for new signal processing directions. Our prior work on chaos, solitons and fractals falls in this category to a certain extent. More recently, studying quantum mechanics as a parallel has led us to the development of a variety of new algorithmic frameworks, which we describe as Quantum Signal Processing (QSP). In a similar vein, we are studying signal processing in cell biology and its potential as an analogy for new signal processing algorithms.

[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.

1. Frames, Geometric Algebra and Signal Processing

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Project Staff
Petros Boufounos, Professor Alan V. Oppenheim

The purpose of this research is to explore new mathematical tools and their application in Signal Processing. Specifically we are investigating frame expansions, and geometric (clifford) algebras.

Frame expansions are a very convenient way to generate redundant signal representations, thus creating a representation robust to noise and degradation. We are looking into issues of efficient signal reconstruction using such a representation. During the course of this research we focused on compensating modes of error using frame expansions. Specifically, we are investigating the compensation of quantization and channel loss errors. Frame representations prove to be quite effective in dealing with such issues in a linear fashion. Our goal is to find simple, cost-effective algorithms to compensate for such errors.

Furthermore, we are investigating the application of geometric algebras in signal processing. The algebraic structure of these tools provides a very convenient way to analyze and think about signal processing algorithms. Our objective is to exploit this structure to improve our understanding of existing algorithms, and obtain intuition into creating new ones. We believe that these mathematical tools have the potential to improve significantly our understanding of signal processing methods and create new signal processing paradigms.

2. Non-Uniform Sampling Grid Conversion for Lowpass Reconstruction: the ``Missing Pixel'' Problem

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Project Staff
Sourav Dey, Andrew Russell, Professor Alan V. Oppenheim

Sampling of a continuous-time bandlimited signal to obtain a discrete-time representation is an integral part of a variety of techniques for signal processing and storage. With uniform sampling, the sampling grid is uniform and the reconstruction is accomplished by lowpass filtering a uniform impulse train of samples. Under certain well-known conditions reconstruction from samples on a non-uniform grid can be achieved, although not in general with simple lowpass filtering.

In some situations, it is useful to convert from samples on a uniform grid to a representation on a non-uniform grid while keeping the reconstruction process through a lowpass filter. For example, consider a system designed for lowpass reconstruction from uniform samples in which, perhaps because of a faulty digital-to-analog (D/A) converter, the impulse timing is non-uniform or specific samples are forced to zero. For example, the latter problem might occur in a flat-panel display with defective pixel LEDs. Such displays inherently rely on a form of lowpass filtering accomplished by viewing the display from an appropriate distance. With defective LEDs, and without additional compensation, the perceived output is degraded.

Under certain conditions, it may be possible to compensate for the missing samples, by adjusting the other sample values, and still theoretically achieve exact reconstruction at the output of a lowpass filter. We informally refer to this as the "missing pixel" problem. This is suggestive of a defective flat-panel display, however, the development and theory are considerably more general than that specific example. It is important to note that this problem is not one of data recovery. It is assumed that the correct values of all the uniformly spaced samples are known. It is the conversion process preceding the lowpass filter that forces particular values to zero. In anticipation of the faulty conversion process, a compensated sequence is generated from the samples on the uniform grid.

We have developed a number of compensation techniques to convert from samples on a uniform grid to a representation on a non-uniform grid with particular samples missing. The ideal solution is a perfect, infinite-length compensation signal. For practical implementation, we have developed two algorithms to compute the optimal, finite-length solution: Constrained Minimization (CM), a closed-form calculation, and Iterative Minimization (IM), an iterative POCS algorithm. Despite calculating the optimal solution, both algorithms were found to be computationally expensive and ill-conditioned over a large range of parameters.

As an alternative, we developed a compensation strategy using discrete prolate spheroidal sequences. Our solution, termed DPAX, is shown to be a tight, first-order approximation of the optimal, finite-length solution in the DPSS basis. In addition, DPAX is less complex and better conditioned than either CM or IM. Also, while the DPAX and IM solutions formally require an ideal LPF, as with many useful signal processing algorithms, these empirically appear to be robust and effective without stringent requirements on the LPF characteristics.

3. A Novel Method for Phase Unwrapping

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Project Staff:
Zahi Karam, Professor Alan V. Oppenheim

This research focuses on developing new methods to unwrap the principal value of the phase of the Fourier transform of finite-length signals. The suggested method is a symbiosis between two existing methods: exploiting.

The first existing method is based on adaptive integration of the phase derivative. The difficulty with this method is caused by zeros near the unit circle in the z-transform of the signal. The second uses polynomial factoring to factor the z-transform of the signal. The zeros are then used to calculate the unwrapped phase. Emerging methods in polynomial factoring that exploit the FFT have proven to efficiently factor polynomials with zeros clustered near the unit circle. We are proposing a two step process: first use polynomial factoring to remove the phase contribution of the zeros that are clustered near the unit circle. Second use adaptive integration of the phase derivative to unwrap the rest of the phase that is the product of the remaining zeros located away from the unit circle.

4. The Bacterial Chemotaxis Network and Gradient Search

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Project Staff
Alaa Kharbouch, Maya Said, Professor Alan V. Oppeneheim

This project focuses on bacterial chemotaxis, the process by which cells adjust their movement in response to chemical stimulants. Escherichia coli bias their swimming based on information extracted from their environment about the concentrations of different substances. Mathematical models and systems approaches can be used to describe the locomotory response of E. coli to it environment and the incoming concentration information.

We have been examining the biochemical network which controls its swimming behavior, and studying its properties as a signal processor. This ties in with how the E.coli navigates its environment in a way that maximizes/minimizes its exposure to certain chemicals, as it senses concentration gradients. We have been interested the nature and effectiveness of its own version of a gradient search algorithm.

5. Non-linear Equalization Using Quadratic Filters

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Texas Instruments, Inc. Leadership University Consortium

Project Staff
Joonsung Lee, Dr. Gil Raz, Professor Alan V. Oppenheim

We have been exploring the use of nonlinear techniques for equalization. In particular, we are focusing on non-linear channels and other systems with additive noise that is dependent on the input.

We are exploring two approaches to this problem; the first involves using a quadratic equalization filter. The idea comes from linearization of nonlinear system using high order expansion. Until now, we assume that the prior information of input and noise is given. Simulation shows that this quadratic equalizer outperforms linear equalization in some cases. We are seeking a way to extend this idea for “blind” input. We are focused on “Adaptive & Blind Quadratic Equalizer”. In addition to designing the nonlinear equalizer, we are exploring dithering techniques to enhance the equalization. For example, going through quantization, the signal loses a lot of information. Dithering is widely used in quantization. Our objective is applying the dither to any kind of nonlinear systems to enhance the equalization.

6. Biological Signal Processing

Project Staff
Maya R. Said, Professor Alan V. Oppenheim, Douglas A. Lauffenburger

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Signal processing is an integral part of cell biology. The associated biological algorithms are implemented by signaling pathways that cell biologists are just beginning to understand and characterize. One of our objectives in the context of biological signal processing is to understand and model these biological algorithms. Another objective is to emulate these processes, i.e. to use them as metaphors in other engineered contexts such as ad-hoc wireless networks, distributed sensor networks, and general algorithm development with the possibility of developing new, efficient, robust signal processing systems. Toward these ends, our research in biological signal processing focuses on developing new frameworks for modeling, at different levels of abstractions, the information processing in biological cells in order to understand the algorithms implemented by the signaling pathways that perform the processing and exploiting the results to develop a new generation of algorithms for engineered distributed networks.

Modeling

Our mathematical models span different levels of abstraction. At the highest level, the dynamical properties of the components of the signaling network are ignored and the focus is on the network topology. In this regime, we focus on nodes interconnectivity while ignoring the nodes identities and introduce ideas from random graph theory to examine and analyze the distribution and properties of the network graph. We then add resolution to our models by examining nodes dynamics and how different nodes interact. This leads to a low-level model where we examine the dynamics of cellular signal processing using a new framework that we developed based on interacting Markov chains. The results of our analyses are then used along with experimental data obtained through collaborations with investigators in the Biology and Biological Engineering Departments to understand how the topologies and dynamics of biological signaling networks confer given system properties and characteristics.

Evolution of Biological Networks

These two complementary modeling approaches allow us to also examine the evolution of biological signaling networks. At one end, the high-level model provides a framework in which hypotheses about the process of network evolution can be formulated. At the other end, the low-level model offers, through comparative analyses of conserved pathways an understanding of how complexity and network topology leads to functionality and refinement of the system responses.

Biological Inspired Distributed Signal Processing Algorithms

Using the tools developed to understand and model biological networks, we also use these networks to explore a new generation of networks where nodes are no longer biological components but engineered systems such as sensors and processors. This leads to the formulation of new algorithms and topologies for signal processing on distributed networks.

7. Data Selection Techniques for Binary Hypothesis Testing

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Project Staff
Charles Sestok, Professor Alan V. Oppenheim

Traditionally, statistical signal processing algorithms are developed from probabilistic models for data. The design of the algorithms and their ultimate performance depend upon these assumed models.

In certain situations, collecting or processing all available measurements may be inefficient or prohibitively costly. A potential technique to cope with such situations is data selection, where a subset of the measurements that can be collected and processed in a cost-effective manner is used as input to the signal processing algorithm. Careful evaluation of the selection procedure is important, since the probabilistic description of distinct data subsets can vary significantly. An algorithm designed for the probabilistic description of a poorly chosen data subset can lose much of the potential performance available to a well-chosen subset.

Our research considers algorithms for data selection combined with binary hypothesis testing. We develop models for data selection in several cases, considering both random and deterministic approaches. Our considerations are divided into two classes, depending upon the amount of information available about the competing hypotheses. In the first class, the target signal is precisely known, and data selection is done deterministically. In the second class, the target signal belongs to a large class of random signals, selection is performed randomly, and semi-parametric detectors are developed.

8. Sound Wave Propagation Around a Seamount

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Project Staff
Joseph Sikora III, Professor A.B. Baggeroer

Experiments show that current acoustic modeling techniques are not complete in the case of down-sloping ocean bottoms. Regions of low transmission loss appear in areas where acoustic models predict shadow zones to be. Experimentation needs to be conducted to explore this phenomenon. Also, experimentation needs to be carried out to test whether acoustic models accurately predict how sound propagates around a seamount.

This project involves using current acoustic models to predict how sound propagates around a seamount. This fall we will explore this problem by conducting experiments out in the Pacific Ocean using a large hydrophone array and different types of sources. We are currently preparing for this experiment by running models that will help us determine exactly where and when we need to conduct our experiment. Once we conduct the experiment we will determine if the current acoustic models are complete, and if not, explore ways of improving them.

9. Design of Efficient Digital Interpolation Filters for Integer Upsampling

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Project Staff
Daniel Turek, Professor Alan V. Oppenheim

Digital interpolation systems can be implemented in a variety of ways. When interpolating for an integer upsampling factor, the most basic system implementation would cascade an expander unit with an interpolation low pass filter (LPF). More complex implementations consist of expander and LPF cascades, with the correct choices of intermediate LPFs. Further, there is leeway in the design of the interpolation filters.

This research explores how digital interpolation systems can be efficiently designed. Efficiency is measured in the number of multiplications required for each output sample. The following factors are studied for their effect on system efficiency: decomposing the system into multiple cascaded stages, using recursive and non-recursive interpolation filters, and using linear phase and minimum phase interpolation filters.