Digital Signal Processing Research Program

RLE Group
Digital Signal Processing Group

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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 have come from some unconventional directions, such as algorithms based on fractal signals, chaotic behavior in nonlinear dynamical systems and quantum mechanics, 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. Falling into this category to a certain extent, is our previous work on fractals, chaos, and solitons. Recently, we developed new algorithmic frameworks which we describe as Quantum Signal Processing (QSP), which use quantum mechanics for its inspiration. We are also studying signal processing in cell biology for improved modeling of cell biology and for its potential for inspiring new signal processing algorithms.

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1. Design and Implementation of Efficient Sampling Rate Conversion Systems

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Sampling rate conversion plays a central role in practical realizations for many discrete-time systems. The process pervades applications involving oversampled analog-to-digital and digital-to-analog converters, and it is also the enabling component for a wide array of multi-rate signal processing algorithms. This project examines techniques for designing and implementing efficient sampling rate conversion systems.

As an example, filters with symmetric impulse responses can in general be implemented using a “folding” technique which takes advantage of symmetry to reduce the number of multiplications per output sample. A technique has been developed for exploiting this and similar efficiencies for filters in sampling rate conversion systems, while still retaining the additional computational benefits associated with polyphase implementations. We have shown that in general, the proposed technique yields a lower number of multiplications per output sample than an equivalent filter implemented using polyphase alone, and in many cases, this gain allows for the implementation of high-order, linear-phase FIR filters at a significantly lower computational cost than low-order IIR filters with equivalent magnitude responses. We are also exploring other ways of improving computational efficiency in polyphase implementation.

2. Acoustic Modeling of Buildings for Acoustic Source Localization

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Project Staff
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This research focuses on developing models to describe acoustic signals propagating through buildings. The study of single room acoustics is a mature field, whereas there has been much less research performed to understand how acoustic signals travel between neighboring rooms, down hallways, and between adjacent floors.

We are using system identification theory to measure impulse responses inside buildings. In addition, we are looking at numerical simulations as well as advanced geometric methods based on ray tracing as methods of obtaining impulse responses. With the impulse response data we are building a model to describe the propagation of acoustic signals in buildings. With a thorough understanding of the propagation of acoustic signals, we can begin to develop algorithms to locate and track acoustic sources in a building.

3. Projective compensation of errors on redundant signal expansions

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Frame expansions are a very convenient way to generate redundant signal representations, thus creating a representation robust to noise and degradation. This project develops efficient signal reconstruction using such a representation. During the course of this research we focus on compensating modes of error using frame expansions. Specifically, we investigate the compensation of quantization, additive noise and erasure errors. Frame representations prove to be quite effective in treating such issues using linear, low-complexity systems. Our goal is to find simple, cost-effective algorithms to compensate for such errors.

The most significant contribution this year is the generalization of Sigma-Delta quantization methods to arbitrary frame expansions. We introduced the viewpoint of Sigma-Delta quantization as a sequence of error compensations using projections. Specifically, we demonstrated that quantizing a coefficient and projecting the quantization error to the space defined by the subsequent coefficients always minimizes the incremental error of quantization. Using this viewpoint, it is easy to extend Sigma-Delta noise shaping to arbitrary frame expansions. Furthermore, this viewpoint offers valuable insight in the design and the operation of existing Sigma-Delta analog-to-digital and digital-to-analog converters.

In addition to Sigma-Delta noise shaping, we also examined the problem of erasure errors and additive noise. Similarly to the quantization problem, both types of errors can be compensated for using projections. However, the details of the implementation differ, since the systems transmitting the expansion coefficients usually have no knowledge of the error. Thus the methods we developed pre-compensate for the errors at the transmitter, and undo the compensation at the receiver if the error does not occur. The resulting structures have very low computational complexity and low input-output delay, making them applicable to high-throughput applications.

4. Data Dependent Sampling

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In this project, we are exploring data-dependent sampling (DDS) as a method to improve analog-to-digital (A/D) conversion. An analog-to-digital converter (ADC) has two basic, limiting parameters: dynamic range and bandwidth. A larger dynamic range requires wider quantization levels, which leads to greater distortion in the digital representation. The larger the bandwidth of the input signal, the faster we need to sample. Our goal is to design DDS techniques that reduce the dynamic range and bandwidth constraints on an ADC while still achieving a faithful digital representation of an analog signal.

Specifically, our goal is to develop a smart ADC that can sample adaptively, based on the characteristics of the data.

To this end, we are exploring several techniques. One corresponds to sampling based on zero-crossings. A second is the use of subspace-based sampling architecture.

In addition to noise rejection, we are also exploring DDS as a new method of doing waveform coding. Current waveform coding techniques use uniform sampling and then use various methods at the quantizer to get a better bit-rate vs. distortion tradeoff. We are exploring methods to improve on this based on non-uniform sampling.

5. Computation of the One-Dimensional Unwrapped Phase

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Homomorphic signal processing via the complex cepstrum has been applied, with considerable success, to many areas of digital signal processing. For the complex cepstrum to exist and be uniquely defined, its computation requires obtaining an unambiguous continuous phase curve (unwrapped phase) of the Discrete Time Fourier Transform (DTFT) of the input signal.

In this project we have studied existing methods for the computation of the unwrapped phase, analyzing their strength and weaknesses. We propose a composite algorithm that uses the existing ones, combining their strengths while avoiding their weaknesses. The core of the proposed method is based on polynomial factoring: recent advancements in this field allow polynomial factoring to be a viable engine for the proposed algorithm.

Our aim is to develop a method that performs better than existing ones by obtaining the correct answer for most one-dimensional mixed phase sequences whose zero locations are not constrained to a certain region of the Z-plane.

6. A Bacterial Algorithm for Surface mapping Using a Markov Modulated Markov Chain Model of Bacterial Chemotaxis

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Texas Instruments, Inc. Leadership University Consortium

Project Staff
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Our research has focused 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 its environment and the incoming concentration information.

We have been examining aspects of the biochemical network which controls its swimming behavior, and our modeling efforts have led to a discrete Markov Modulated Markov Chain (3MC) model of the chemotaxis network. Our research includes the development of a surface mapping algorithm using virtual bacterial agents based on the 3MC model. This Bacterial Algorithm for Surface Mapping (B.A.S.M) simulates and tracks the motion of the bacterial agents as they navigate a one- or two-dimensional surface (representing a spatially varying concentration of a desired substance). A surface flattening algorithm also allows the E.coli agents to “consume” or reduce the value of the function at the locations they visit, instead of just passively sampling the surface.

7. Blind Identification of Acoustic Signal

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Project Staff
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We have been investigating how to identify intruders that would breach into certain secure locations by detecting and identifying the sound generated by them. The microphones or geophones can be attached to the outside of the adjacent walls of that location and be used to detect footsteps. We have been doing experiments to identify acoustic models that can simulate how sound propagates through the walls. We assume that the channels are LTI (Linear Time Invariant) filters. In other words, each measured signal on each microphone is the output of an LTI filter acting on the original sound. We will do experiments to see the diversity of the filter which might depend on the locations of the microphones such as distance from the microphone to the nearest wall and the size of the room.

We have developed an algorithm to identify the audio signal without knowing the channels and the audio source. This problem is typically referred to as "Blind Identification over FIR Channels". Our algorithm depends on the diversity (z-transforms of the impulse responses of the channels have no common zeros) of the channels. Specifically, we need at least two microphones. We can determine the subspace of the original sound without knowing the channels. We can determine the original input signal by intersecting two subspaces. The basis of the intersection space is the original audio signal. In a noiseless case, our method generates the original audio signal exactly up to a constant factor.

8. Signal Processing in Biological Cells: Proteins, Networks, and Models

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This work introduces systematic engineering principles to model, at different levels of abstraction the information processing in biological cells in order to understand the algorithms implemented by the signaling pathways that perform the processing. An example of how to emulate one of these algorithms in other signal processing contexts is also considered.

At a high modeling level, the focus is on the network topology rather than the dynamical properties of the components of the signaling network. In this regime, we examine and analyze the distribution and properties of the network graph. Specifically, we present a global network investigation of the genotype/phenotype data-set recently developed for the yeast Saccharomyces cerevisiae from exposure to DNA damaging agents, enabling explicit study of how protein-protein interaction network characteristics may be associated with phenotypic functional effects. The properties of several functional yeast networks are also compared and a simple method to combine gene expression data with network information is proposed to better predict pathophysiological behavior.
At a low level of modeling, the work introduces a new framework for modeling cellular signal processing based on interacting Markov chains. This framework provides a unified way to simultaneously capture the stochasticity of signaling networks in individual cells while computing a deterministic solution which provides average behavior. The use of this framework is demonstrated on two classical signaling networks: the mitogen activated protein kinase cascade and the bacterial chemotaxis pathway.

The prospects of using cell biology as a metaphor for signal processing are also considered in a preliminary way by presenting a surface mapping algorithm based on bacterial chemotaxis.

9. Simultaneous Tracking and Sensor Calibration in Urban Environments

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The primary goal of this research is to track a human being moving inside of a building using a network of non-overlapping sensors. We are able to simultaneously recover the calibration parameters of the sensors and target trajectory using MAP (maximum a posteriori) estimation. The MAP algorithm searches the space of all possible solutions and determines the one that maximizes the posterior probability. The location and rotation of each sensor in the global coordinate system is unknown, and each measurement is recorded in the local coordinate system of the sensor. Knowledge of the target's dynamics is encoded in the model to compensate for missing measurements.

The initial solution developed at MIT minimized the likelihood function over the set of all measurements and all time steps in the path. In order to increase the speed of the algorithm and to make the optimization computationally tractable, we introduced two improvements. The first is to estimate the unobserved path segments between sensors as independent problems. The effect of missing measurements is summarized by the computation of the a priori probability of the target state at the time it enters a new sensor's field of view, given the state of the target when it last exited at the previous sensor's field of view. Estimation of the path between sensors can be computed independently for each entry/exit pair, given initial and final position and velocity data and total length of time for which the target was out of range.

The second improvement is to treat the measurements as constraints. Since the measurement noise is negligible, we assume it is zero and express each measurement as a function of the target's position and sensor parameters, where this function is the mapping from global to local coordinates. The MAP estimate now amounts to finding the track and sensor parameters that will maximize the prior probabilities subject to the measurement constraints. We do not need to estimate the path within a given sensor, since that path is given by the measurements recorded. As a result, we only need to estimate the global coordinates of the target as it enters and exits the FOV (field of view) of each sensor.

The practical implication of this research is that now we can think of MAP estimation as an online smoothing algorithm, with the entire path and sensor calibration recomputed each time a new measurement arrives.

10. Sound Wave Propagation Around a Seamount

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Acoustic modelers, which use normal mode and parabolic equation methods, can estimate how sound propagates in the ocean between a source and receiver. Parabolic modelers make an approximation to the Helmholtz equation, the frequency domain wave equation, which reduces computation time. Normal mode modelers, using the Helmholtz equation, breaks the pressure field into range, and depth dependent, orthogonal modes, allowing for an accurate solution, at the expense of computation time. The ocean, being a range-dependent environment, makes the task of computing pressure more complicated and computationally intensive. The programs RAM (Range-dependent Acoustic Modeler), developed by The Office of Naval Research, and C-SNAP (Coupled SACLANTCEN normal mode propagation loss model), developed by SACLANTCEN, use the parabolic equation and normal mode methods, respectively, to compute the pressure field in a range-dependent environment.

The way sound travels around seamounts, in the ocean, is of particular interest, as it is a highly range-dependent environment. Intuition should suggest that seamounts block sound waves propagating at high elevation angles, and pass sound waves propagation at low elevation angles and traveling close to the surface. The makeup and the geometry of the seamount, however, makes it difficult to predict how sound wave travel around the seamount, due to incomplete sediment models and approximations made about the environment. Results from RAM and C-SNAP, in this situation, have not been consistent. The shadow of a seamount is important for performance predictions of sonars and systems for monitoring low yield nuclear test explosions.

Our research explores this phenomenon by examining data taken from the North Pacific Acoustic Laboratory's, or NPAL's, September - October 2004 SPICEX, LOAPEX, and BASSEX experiments. During these experiments, data was gathered from a hydrophone array as it was pulled behind the Kermit seamount, in the Pacific Ocean. The array was used to listen to m-sequence sources, positioned on the opposite side of the seamount. A number of data files are available from this cruise which allows for a quantitative look at the effect of the seamount on sound propagation. By applying matched filters to the m-sequences, it should be possible to correlate this data to predictions made with the acoustic modelers, previously mentioned, making it possible to evaluate of the accuracy of these two coding schemes.

Publications

Journal Articles

M. R. Said, T. J. Begley, A. V. Oppenheim, D. A. Lauffenburger, L. D. Samson, "Global network analysis of phenotypic effects: Protein networks and toxicity modulation in Saccharomyces cerevisiae", *Proc. Natl. Acad. Sci. USA*.101 (52): 18006-18011 (2004).

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S.R. Dey, A.I. Russell, A. V. Oppenheim, "Pre-Compensation for Anticipated Erasures in LTI Interpolation Systems", *IEEE Transactions in Signal Processing*, forthcoming.

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M.R. Said, T.J. Begley, A.V. Oppenheim, D.A. Lauffenburger, L.D. Samson, "Global Network Analysis of Phenotypic Effects: Protein Networks and S.cerevisiae Damage Recovery", 44th American Society for Cell Biology Annual Meeting, (Washington, DC), Dec. 2004.

Theses & Dissertations

M.R. Said, *Signal Processing in Biological Cells: Proteins, Networks, and Models*, Sc.D. diss., Department of Electrical Engineering and Computer Science, MIT, 2004.