Digital Signal Processing

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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, 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. 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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MIT Lincoln Laboratory Signal Processing Research P.O. No. 3041940
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BAE Systems, Inc. P.O. No. 112991
Texas Instruments, Inc. Leadership University Consortium

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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.

Recent focus has been directed toward design methods for anti-aliasing and reconstruction filters. The investigation of this specific problem is motivated by the fact that the Parks-McClellan algorithm, in addition to a number of constraint-based design techniques, results in filters which are optimal for a given order, but which are not guaranteed to be optimal in the same sense for a given required number of multipliers. In drawing from our earlier research into filter structures for rate conversion systems, we have proposed techniques for designing filters which, when implemented using the given structures, require fewer multipliers than Parks-McClellan designs implemented in direct form. We are also exploring computationally-efficient, near-optimal relaxations of these design methods.

2. 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.

This year significant progress was made on the compensation of erasures in the frame representation coefficients. It was shown that it is possible to mitigate or eliminate the effect of coefficient erasures in frame representations by compensating for the error using the remaining non-erased coefficients. The optimal compensation takes the form of a projection. The resulting systems can be designed to be causal and have very low implementation complexity, making them particularly suitable for high data-rate applications.

The applications of erasure compensation are numerous. Erasures are often encountered in faulty D/A and A/D converters and internet packet transmissions. Furthermore, they can be useful in sensor networks, in the case the sensor is not able to communicate with the central authority, either due to lack of power or due to the lack of a reliable channel. Erasures can also be introduced intentionally during the sampling stage to generate sparse representations.

3. Noise-Shaping using Randomized Sampling

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As a continuation of our ongoing work on data-dependent sampling, we have developed a technique to do noise-shaping using randomized sampling.

In traditional periodic sampling, if a signal is sampled below its Nyquist rate there is aliasing. Aliasing is a high-frequency distortion that is perceptually disturbing in many applications, manifesting itself as moire patterns in images and popping artifacts in audio signals. While an anti-aliasing filter prevents aliasing, it imposes a severe limitation on the range of frequencies that can be faithfully recovered. This makes anti-aliasing unsuitable when there is critical information content at the high-frequencies. In addition, there are certain contexts where anti-aliasing is impossible because of the nature of the problem. For example, in ray-traced computer graphics, one cannot anti-alias the image because it is not in analog form to begin with, instead it is a 3-D mathematical model that must be probed directly onto a discrete grid.

Randomized sampling can avoid aliasing without the use of an anti-aliasing filter. We are developing randomized sampling techniques that can be used to shape the reconstruction error spectrum so that its perceptual effect is minimized. The methodology is akin to noise-shaping in sigma-delta ADC's, but different in that the effect is due to correlations in the sampling pattern, rather than a feedback loop.

Our technique could have potential applications in certain contexts where there are limitations on the ADC. For example, in wide-band surveillance, the maximum sampling rate of an ADC may be below the desired Nyquist rate. In such a context, randomized sampling can be used to algorithmically extend the effective bandwidth of the ADC. In another example, sampling in sensor networks consumes a disproportionate part of the power in each sensor node. In this context, randomized sampling can be used to reduce the rate of sampling, and thus power consumption, while still achieving a faithful representation of the signal.

Extensions to randomized matched filtering and linear operations on the randomized sample stream are also being explored.

4. Computation of the One-Dimensional Unwrapped Phase

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Homomorphic signal processing using the complex cepstrum has been applied, with considerable success, to many areas of digital signal processing, most notably speech, seismic and EEG data processing. The computation of the complex cepstrum requires obtaining the unwrapped phase, which is the continuous and periodic instance of the phase of the discrete-time Fourier transform (DTFT) of the input signal. However, reliably computing samples of the unwrapped phase of a given mixed-phase signal is an open problem: existing algorithms are not reliable and in many cases fail.

In this project we presented two composite algorithms that use the existing ones, combining their strengths while avoiding their weaknesses. The core of the presented methods is based on recent advances in polynomial factoring. Our analysis of these methods, through numerous experiments, has demonstrated their superiority over the existing in terms of reliability and accuracy.

5. A Bacterial Algorithm for Surface Mapping Using a Markov Modulated Markov Chain Model of Bacterial Chemotaxis

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Bacterial chemotaxis is the locomotory response of bacteria to chemical stimuli. E.coli movement can be described as a biased random walk, and it is known that the general biological or evolutionary function is to increase exposure to some substances and reduce exposure to others. We introduced an algorithm for surface mapping, which tracks the motion of a bacteria-like software agent (based on a low-level model of the biochemical network responsible for chemotaxis) on a surface or objective function. Toward that end, a discrete Markov modulated Markov chains model of the chemotaxis pathway is described and used. Results from simulations using one- and two-dimensional test surfaces show that the software agents, referred to as bacterial agents, and the surface mapping algorithm can produce an estimate which shares some broad characteristics with the surface and uncovers some features of it. We also demonstrate that the bacterial agent, when given the ability to reduce the value of the surface at locations it visits (analogous to consuming a substance on a concentration surface), is more effective in reducing the surface integral within a certain period of time when compared to a bacterial agent lacking the ability to sense surface information or respond to it.

6. Acoustic Signal Estimation using Multiple Blind Observations

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Project Staff
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It is a common problem to attempt to recover a signal from observations made by two or more sensors. Most approaches to this problem fuse the information from the sensors through an a posteriori probabilistic model. We have been investigating entirely different approach to this problem by using the results of blind signal and channel estimation in the context of data communication theory.

We have developed the Coordinated Recovery of Signals From Sensors (CROSS) algorithm and the Averaging Row Space Intersection (ARSI) algorithm. These algorithms produce inverse channel filters and signal estimation that directly account for additive noise in the system. We believe the algorithms may be useful in fusing different modalities of sensors (seismic, radar, etc.) as long as the LTI model holds.

7. Simultaneous Tracking and Sensor Calibration in Urban Environments

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Project Staff
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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 imaging sensors, i.e. cameras. This research focuses on two topics related to the simultaneous calibration of the camera network and the recovery of the trajectory of an object moving among those sensors. The non-overlapping fields of view of the cameras do not cover the entire scene, and therefore there are times for which no measurements are available. A Bayesian framework is imposed on the problem in order to compute the MAP (maximum a posteriori) estimate, the solution that maximizes the probability of the sensor network configuration and target trajectory given the measurement set.

The first area addressed by this research is model order reduction to decrease the number of unknown parameters in the motion and measurement models, thereby reducing the computational requirements of the optimization algorithm. We reduce the dimension of the search space, with no loss in accuracy, by treating the estimation of target's trajectory while it is outside the field of view of the network as an independent sub-problem that can be solved at a later time. Additionally, we can further reduce the dimension of the search space by taking advantage of the fact that the measurement noise is negligible compared to other sources of error in the problem, and subsequently the measurements can be treated as equality constraints in the optimization. The second procedure yields results that are not equal to the solution determined by the original algorithm, with the magnitude of the difference dependent on the covariance of the original measurement noise.

The second focus of this research is to develop techniques to improve the accuracy of the solution by taking further advantage of the missing measurement information. The original algorithm at times yields a solution that is infeasible with respect to knowledge of the times at which measurements occurred. The MAP estimate may place the trajectory of the target inside the field of view of one of the sensors at a time step for which no measurement is available. We created a graphical simulation tool that enabled us to develop and test three techniques to treat the missing measurements as additional constraints. First we reformulate the problem as a mixed-integer nonlinear programming problem where the integer variables encode the information regarding where the target is permitted to be in the global map when no measurements are available. Next we develop an algorithm to systematically explore only feasible subsets of the search space, adding and removing constraints as needed according to a simple set of ad-hoc rules. Finally, we examine the tradeoffs involved in using circular constraints to approximate the actual sensor boundaries.

8. Sound Wave Propagation Around a Seamount

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Project Staff
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Our research involves testing the predictive capabilities of numerical models of sound propagation in long, range-dependent ocean waveguides using experimental data. In particular, we focus on how acoustic models predict the scatter field of underwater mountains (seamounts). This research will improve our knowledge of underwater acoustics, and has direct applications in underwater communications, tomography, military applications, and monitoring for low yield underwater nuclear explosions.

The NPAL 2004 BASSEX experiment was conducted to measure the scatter field of a typical seamount. The Kermit-Roosevelt seamount complex is a pair of seamounts in the central Pacific Ocean. They were chosen as representative of typical ocean seamounts and because they are relatively isolated. Acoustic point sources were moored approximately 600 km south of the seamounts. A hydrophone array was towed around the seamounts to listen to the sources and measure the scatter field of the seamount. These data were processed using conventional and adaptive beamforming techniques. The results are being compared with numerical models including ray tracers, C-SNAP (coupled-mode), and RAM (parabolic approximation). We are attempting to compare the beam outputs of the towed arrays as function of angle, travel time and amplitude to the predictions of these codes, which represent the state of the art in numerical modeling of acoustic propagation.

9. Discrete Representation of a Continuous-Time Signal Using Bilinear Sampling

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This work explores a previously proposed novel discrete representation of continuous-time signals using the bilinear transform. The effect of this bilinear sampling technique is a nonlinear warping of the frequency axis between the continuous-time and discrete-time signal representations. The system can be implemented as a Laguerre network consisting of two low-pass filters followed by a cascade of all-pass filters. The goal of this project is to develop a systematic characterization of the sensitivity/robustness of this system to real-world issues. These include additive noise at the input, component noise in the analog domain, quantization in the digital domain, and truncation of the series representation. Two preliminary investigations were conducted in order to gain a better understanding of the Laguerre network. First, the time-domain impulse response of the system was plotted using MATLAB for different cascade lengths. Second, the effect of internal/component noise was analyzed separately for each type of filter in the network.

10. Sampling Based on Local Bandwidth

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Traditionally, continuous-time signals are often sampled uniformly without regard to local bandwidth, a term that refers to how quickly or how slowly a signal varies on a local basis. At an intuitive level however, it may be more sensible to employ a sampling strategy in which the sampling rate adapts to the local bandwidth, being higher in regions of higher local bandwidth and vice versa. Sampling based on local bandwidth could offer a more efficient representation of a signal as compared with uniform sampling, given equal average sampling rates.

Two formal definitions of local bandwidth are developed. In the first definition, we consider a locally bandlimited signal to be the output of a time-varying lowpass filter. We show that locally bandlimited signals in general cannot be exactly represented by samples specified by the local bandwidth in a natural way. The error in re-synthesizing locally bandlimited signals from their samples depends on measures of the variation in local bandwidth, such as its dynamic range. There exists however a special case where a sampling representation based on local bandwidth is exact for a class of locally bandlimited self-similar signals.

In the second definition, we consider a locally bandlimited signal to be the result of subjecting a globally bandlimited signal to a transformation of the time variable, known informally as a "time-warping." Under the time-warping definition, an exact sampling representation based on local bandwidth does exist for locally bandlimited signals. We formulate an optimization problem in which we seek the time-warping that leads to minimum-error sampling and reconstruction of a signal based on its local bandwidth. Methods for determining the optimal time-warping are currently under investigation. They include the use of the calculus of variations, an iterative algorithm based on the zero-crossings of the signal to be sampled, and piecewise linear approximation and a related gradient-search algorithm. The concept of time-warping can also be used to compensate for the non-uniformity of a known sampling grid, in which the time-warping is chosen to yield uniform, Nyquist rate samples of a globally bandlimited signal.

11. Generating Sets of Quasi-Orthogonal Radar Waveforms Using Chaos Theory

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With the development of faster A/D converters and waveform generators, it is now practical to use high-bandwidth, arbitrary waveforms in radar applications. Large sets of mutually-orthogonal, high-bandwidth waveforms can be generated so that multiple radars can simultaneously operate in the same frequency band. Each individual radar receiver can process its own return as well as the orthogonal returns from the other radars, which opens the possibility for developing algorithms that combine data from multiple radar channels. Due to the random nature of chaotic signals, this project addresses the development of a procedure for generating large sets (>50) of quasi-orthogonal radar waveforms using deterministic chaos.

Deterministic chaos is defined as a bounded, aperiodic flow with a sensitive dependence on initial conditions. There are many different types of chaotic systems, but in this project, we will generate waveforms from the well-studied Lorenz system. Each waveform from the Lorenz system can be fully characterized by three parameters (σ, b, and r) and a set of initial conditions, (x_o, y_o, z_o). The parameter values combined with the method for selecting initial conditions greatly affect the autocorrelation properties and attractor shape for a particular waveform. This project analyzes how to determine both the parameter values and the method for selecting the initial conditions when generating chaotic waveforms, based on how effective these chaotic waveforms are as quasi-orthogonal radar waveforms.

Publications

Journal Articles, Published

S.R. Dey, A.I. Russell, and A. V. Oppenheim, "Pre-Compensation for Anticipated Erasures in LTI Interpolation Systems", *IEEE Transactions in Signal Processing*, 54(1): 325-35 (2006).

Journal Articles, Accepted for Publication

P. T. Boufounos, and A. V. Oppenheim, "Quantization Noise Shaping on Arbitrary Frame Expansions." *EURASIP Journal on Applied Signal Processing (JASP), Special Issue on Frames and Overcomplete Representations in Signal Processing, Communications, and Information Theory,* forthcoming.

Meeting Papers

P. T. Boufounos, and A. V. Oppenheim, "Compensation of Coefficient Erasures in Frame Representations." *Proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing, 2006. (ICASSP '06)*. Toulouse, France, May 14-19, 2006.

Theses

R.E. Bland, *Acoustic and Seismic Signal Processing for Footstep Detection*, M. Eng. thesis, Department of Electrical Engineering and Computer Science, MIT, 2006.

P. T. Boufounos, *Quantization and Erasures in Frame Representations,* D.Sc. diss., MIT, 2006.

Z.N. Karam, *Computation of the One-Dimensional Unwrapped Phase*, S.M. thesis, MIT, 2006.

A.A. Kharbouch, *A Bacterial Algorithm for Surface Mapping Using a Markov Modulated Markov Chain Model of Bacterial Chemotaxis*, S.M. thesis, MIT, 2006.

J. Lee, *Acoustic Signal Estimation using Multiple Blind Observations*, S.M. thesis, MIT, 2006.

M.B. Rudoy, *Simultaneous Sensor Calibration and Path Estimation*, S.M. thesis, MIT, 2006.

J.J. Sikora, *Sound Propagation around Underwater Seamounts*, S.M. thesis, MIT, 2005.