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 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. Falling into this category to a certain extent, is our previous work on fractals, chaos, and solitons.

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1. Design and Implementation of Efficient Sampling Rate Conversion Systems

Sponsors
MIT Lincoln Laboratory Signal Processing Research P.O. No. 3077828
Army Research Laboratory (ARL) Collaborative Technology Alliance Contract RP6891
BAE Systems, Inc. P.O. No. 112991
Texas Instruments, Inc. Leadership University Program
Bose Corporation

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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 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 investigating efficient implementations for these systems. Drawing on earlier research into combining folded and polyphase structures, we have arrived at a rate-conversion structure which requires one multiplier for each unique value taken on by the impulse response of the filter. This structure therefore takes advantage of any recurring value in the impulse response and is applicable to time-symmetric filters. All multipliers are implemented at the slower of the input and the output sampling rate, and the number of required compressor blocks is the same as that of a comparable polyphase structure. We are also exploring flow graph theorems which make correspondences between rate-conversion structures, a topic which may be applicable to the design of interleaved A/D conversion systems.

2. Data-Dependent Randomized Sampling and Filtering

Sponsors
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National Defense Science and Engineering Graduate Fellowship (NDSEG)
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Our work explores the use of data-dependent randomization to decrease the complexity of sampling and filtering, two of the most basic signal processing techniques. Though certain randomization techniques have been explored in the literature, they largely consider randomization in continuous-time that is independent of the input signal. By contrast, in our work we propose a discrete-time framework for data-dependent randomization. Using prior information about the input power spectrum, we propose to tune randomized sampling and filtering so that the resulting error is minimized. The techniques developed can be used to achieve a better complexity versus accuracy tradeoff in signal processing systems, particularly ones where there is a constraint on sampling rate or filter multiplies.

We have developed a model of discrete-time randomized sampling as a method to mitigate the effects of aliasing when sampling below the Nyquist rate. We have constructed two distinct randomized sampling architectures, simple randomized sampling and filtered randomized sampling, along with characterizations of the resulting sampling error. We also considered the design of non-white binary processes for use in randomized sampling. The work on these topics is largely summarized in two papers, [1], [2].

In addition to sampling, we have been exploring the randomization of filter coefficients in LTI filters as a method to reduce coefficient quantization error. To this end, we have developed a method of binary randomized filtering, where each filter coefficient is replaced with a binary process, as a low-complexity approximate filtering technique. The design of binary vector processes for use in binary randomized filtering has been considered and we are currently characterizing the limits and scaling of this design. Simulation shows binary filtering to be an effective low-complexity filtering technique.

Applications of these randomized sampling and filtering techniques to matched filtering and compressive sensing are also being explored.

Bibiliography

[1] Dey, Sourav and Alan Oppenheim. “Frequency-Shaped Randomized Sampling” *Proceedings of ICASSP*, Honolulu, HI, April 15-20, 2007.

[2] Boufounous, Petros. “Generating Binary Processes with All-Pole Spectra” *Proceedings of ICASSP*, Honolulu, HI, April 15-20, 2007.

3. A New Kernel for SVM MLLR based Speaker Recognition

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Speaker recognition using support vector machines (SVMs) with features derived from generative models has been shown to perform well. Typically, a universal background model (UBM) is adapted to each utterance yielding a set of features that are used in an SVM.

In this project we consider the case where the UBM is a Gaussian mixture model (GMM), and maximum likelihood linear regression (MLLR) adaptation is used to adapt the means of the UBM. We examine two possible SVM feature expansions that arise in this context: the first, a GMM supervector constructed by stacking the means of the adapted GMM and the second consists of the elements of the MLLR transform. We examine several kernels associated with these expansions. We show that both expansions are equivalent given an appropriate choice of kernels. Experiments performed on the NIST SRE 2006 corpus clearly highlight that our choice of kernels, which are motivated by distance metrics between GMMs, outperform ad-hoc ones. We also apply SVM nuisance attribute projection (NAP) to the kernels as a form of channel compensation and show that, with a proper choice of kernel, we achieve results comparable to existing SVM based recognizers.

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

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Bacterial chemotaxis refers to the locomotory response of bacteria to chemical stimuli, where the general biological function is to increase exposure to some substances while reducing exposure to others. We introduce an algorithm for surface mapping based on a model of the biological signaling network responsible for bacterial chemotaxis. The algorithm tracks the motion of a bacteria-like software agent, referred to as a bacterial agent, on an objective function. Results from simulations using one- and two-dimensional test functions show that the surface mapping algorithm produces an informative estimate of the surface, revealing some of its key characteristics. We also developed a modification of the algorithm in which the software agent is given the ability to reduce the value of the surface at locations it visits (analogous to a bacterium consuming a substance as it moves in its environment) and show that it is more effective in reducing the surface integral within a certain period of time than a bacterial agent lacking the ability to sense surface information or respond to it.

5. Array processing / Passive SONAR research

Sponsors
MIT Lincoln Laboratory, Lincoln Scholars Program

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This research involves analyzing the mean-squared error performance of acoustic vector sensor arrays under modeling errors. When used to estimate direction-of-arrival, these novel arrays have several theoretical benefits over arrays of omni-directional microphones. In practice, however, their performance may be limited by incorrect model parameters or “mismatch.” To understand the practical limits of vector sensor arrays, we seek to model and bound their mean-squared error under mismatch. These tighter performance bounds should allow more realistic design and performance prediction before construction, as well as improved analysis afterward.

Additional research concerns multiple-target tracking using bearing-only measurements. These measurements, provided by many passive SONAR systems, produce tracks that cross and fade frequently. Track association based on a single epoch is problematic because crossing or fading events may persist, while alternatives to single-epoch tracking seem computationally infeasible. Our research proposes two tracking stages, each using single-epoch association. This scheme requires only modest computation and shows improved association through fades and crossings.

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

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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. Video monitoring and machine vision methods are among the best of current approaches, but require careful calibration and are expensive, whereas low-cost, low-energy and rapidly deployable systems are often desired. It was previously shown that passive acoustic and seismic sensors can be used to characterize the signature of a human footstep. This project seeks to develop more robust signatures by combining data from both passive and active acoustic sensors. Systems based on the fusing of these two independent sensor modalities can then be used for the detection of walking subjects and the discrimination of different people.

One approach considered the development of a sensor fusion framework, in which three key signals of interest were extracted from the two data sources, corresponding to total footstep acoustic power, average velocity of the walker's torso, and average velocity of the target's legs. A non-stationary vector autoregressive (VAR) process with time-varying coefficients was then used to describe the evolution for these three signals over time, and successfully captured the physical correlations between them, creating a natural data fusion across different sensor modalities. A set of features derived from the parameters of the VAR model was studied, which in turn were used to train a detection system based on a linear support vector classifier using real data.

7. Sound Wave Propagation around Underwater Seamounts

Sponsors

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

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Our research involves testing the accuracy of numerical models of sound propagation in long, range-dependent ocean waveguides using experimental data. In particular, we are focused 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 enforcing the comprehensive test ban treaty.

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, 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. This data was 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).

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

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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 $j\omega$-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.

9. Sampling Based on Local Bandwidth

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We are investigating the sampling of continuous-time signals according to local bandwidth, i.e., the local rate of variation of the signal. Many classes of signals, such as frequency-modulated signals, natural images, etc., can be suitably described in terms of local bandwidth. Intuitively, such signals should be sampled non-uniformly at a rate adapted to the local bandwidth, with a higher rate where the signal varies more quickly, and a lower rate where it varies more slowly. Sampling based on local bandwidth could offer a more efficient representation for signals that are locally bandlimited in some sense and are not globally bandlimited to a sufficiently low frequency.

To formalize the intuitive concept of local bandwidth, a model based on a class of time-varying lowpass filters has been considered. A sampling and reconstruction method was developed according to the properties of signals generated by such filters. However, the method does not result in perfect reconstruction for many signals belonging to this class, and it is therefore unsatisfying to interpret these signals as being locally bandlimited. The method does have the property that the reconstruction error decreases as the variation in the cut-off frequency becomes more gradual, as shown by an analytical bound and numerical simulations.

We have also examined an alternative model based on the time-warping of globally bandlimited signals. Under the alternative model, a signal is defined to be locally bandlimited if it can be mapped into a globally bandlimited signal through an invertible transformation of the time axis, referred to as a time-warping. The model is attractive since any locally bandlimited signal can be perfectly reconstructed from a set of non-uniform samples taken according to local bandwidth. A sampling strategy has been formulated in which a time-warping is sought to minimize the ratio of the energy of a signal above a given maximum frequency to its total energy. The determination of the optimal time-warping remains an open problem under investigation.

10. 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

A.V. Oppenheim, "One Plus One Could Equal Three (and Other Favorite Cliches)," *IEEE Signal Processing Magazine* 23(6): 10-12 (2006).

Meeting Papers

P.T. Boufounos, "Generating Binary Processes with All-Pole Spectra", *Proceedings of ICASSP*, Honolulu, HI, April 15-20, 2007.

A.A. Kharbouch, M.R. Said, A.V. Oppenheim, “A Bacterial Algorithm for Surface Mapping Using A Markov Modulated Markov Chain Model of Bacterial Chemotaxis”, *Proceedings of ICASSP*, Honolulu, HI, April 15-20, 2007.

S.R. Dey, A.V. Oppenheim, “Frequency-Shaped Randomized Sampling”, *Proceedings of ICASSP*, Honolulu, HI, April 15-20, 2007.

S. Kraut, J. P. Kitchens, P. Bhadra, and B. Payne, “Two-stage Tracker for Improved Association with Bearing-only Measurements,” paper presented at the Adaptive Sensor Array Processing Workshop, Lexington, MA, June 5 – 6, 2007.

Theses

T.A. Baran, *Design and Implementation of Discrete-Time Filters for Efficient Sampling Rate Conversion*, S.M. thesis, Department of Electrical Engineering and Computer Science, MIT, 2007.

D. Wei, *Sampling Based on Local Bandwidth*, M.Eng. thesis, Department of Electrical Engineering and Computer Science, MIT, 2007.

M. Willsey, *Quasi-Orthogonal Wideband Radar Waveforms Based on Chaotic Systems*, M.Eng. thesis, Department of Electrical Engineering and Computer Science, MIT, 2006.