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.

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1. Conservation in Signal Processing Systems

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Texas Instruments, Inc. Leadership University Program

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In electrical network theory as well as various branches of physics, the concept of conservation, e.g. conservation of energy, plays a central role in gaining insight into the behavior of systems. The transfer of a conserved quantity may be used to characterize interaction among systems that are linear or nonlinear, deterministic or stochastic, and which may or may not contain memory. Moreover, conservation in physical systems leads to a number of useful properties related to stability, robustness and optimality. This research involves the development of a framework for representing signal processing systems in terms of conservation principles, facilitating a better understanding of the analysis and design of linear and nonlinear systems. Recent focus has been directed toward designing robust filter architectures for low-power applications and signal processing algorithms for solving optimization problems.

2. Multi-dimensional Processing of One-dimensional Signals

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One-dimensional signals are traditionally processed in one dimension. There are potential advantages, however in mapping one dimensional signals to higher dimensions, and after processing in the higher dimensional space, remapping the result back to a single dimension. One example is in the interpretation of the Volterra series expansion for non-linear systems. In this interpretation, the non-linear Volterra expansion is interpreted in terms of linear processing in higher dimensions followed by projection of the result to a one dimensional signal. In our research, we are focusing on developing Volterra operators that, once applied on a high dimensional counterpart of a signal, corresponds to a desired operation on the original signal in one dimension, which is difficult or impossible to implement by processing in only one dimension.

3. Graph-Embedding for Speaker Recognition

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Popular methods for speaker classification perform speaker comparison in a high-dimensional space; however, recent work has shown that most of the speaker variability is captured by a low-dimensional subspace of that space. Our work examines whether additional structure, such as a non-linear manifold capturing speaker variability, exists within the high-dimensional space. We achieve this using graph embedding and visualization techniques which allow for uncovering structure within datasets as well as motivate algorithms that leverage this structure to yield improved speaker comparison.

4. Acoustic Vector-Sensor Array Processing

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Existing theory yields useful performance criteria and processing techniques for acoustic pressure-sensor arrays. Acoustic vector-sensor arrays, which measure particle velocity and pressure, offer significant potential but require fundamental changes to algorithms and performance assessment.

This research develops new analysis and processing techniques for acoustic vector-sensor arrays. First, the research establishes performance metrics suitable for vector-sensor processing. Two novel performance bounds define optimality and explore the limits of vector-sensor capabilities. Second, the research designs non-adaptive array weights that perform well when interference is weak. Obtained using convex optimization, these weights substantially improve conventional processing and remain robust to modeling errors. Third, the research develops subspace techniques that enable near-optimal adaptive processing. Subspace processing reduces the problem dimension, improving convergence or shortening training time.

5. Reconstruction from Non-uniform Samples

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Exact reconstruction of a band-limited signal from its non-uniform samples involves the use of Lagrange interpolation, which is impractical to implement as it is computationally difficult. This research developed approximate reconstruction methods based on time-warping to obtain reconstruction of band-limited signals from non-uniform samples. A review of non-uniform sampling theorems is presented followed by an alternative interpretation of the Lagrange interpolation kernel by decomposing the kernel into its constituent components. A discussion of time-warping and its use in the context of non-uniform sampling is made. This includes an alternative interpolation known as delay-modulation, which we show to be a simpler representation for a specific case of non-uniform sampling where the sample instants are deviations from a uniform grid. Based on some essential characteristics of the Lagrange kernel, a framework using a modulated time-warped sine function is formed to obtain various approximations to the Lagrange kernel. We also formulated a vector space representation of non-uniform sampling and interpolation and incorporates warped sinc functions to obtain faster convergence in iterative algorithms for reconstructions of band-limited signals from non-uniform samples.

6. Quantization and Compensation in Sampled Interleaved Multi-Channel Systems

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This work considers the environment of interleaved, multi-channel measurements as arises for example in time-interleaved A/D converters and in distributed sensor networks. Such systems take the form of either uniform or recurrent nonuniform sampling, depending on the timing offset between the channels. Quantization in each channel results in an effective overall signal to noise ratio in the reconstructed output which is dependent on the quantizer step sizes, the timing offsets between the channels and the oversampling ratio. Appropriate choice of these parameters together with the design of appropriate compensation filtering is discussed.

It is shown for the multi-channel case that with equal quantizer step size in each channel, the resulting overall SQNR is maximized when the timing offsets are such that the interleaving is equivalent to uniform samples. For the case corresponding to recurrent nonuniform sampling, the SQNR is reduced relative to the uniform case. However, when the quantizer step size is not constrained to be the same in each channel, higher overall SQNR can often be achieved by appropriate choice of the compensation filtering and the timing offset between the channels.

7. Verification and Implementation of Sampled Interleaved Multi-Channel Systems

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Time interleaved systems are often employed to achieve a large oversampling ratio and mitigate the effects of quantization noise in analog-to-digital (A/D) conversion. Depending on phase offset between channels, the interleaved grid can correspond to uniform or recurrent non-uniform sampling. This research considers the effects of quantization in interleaved, oversampled, multi-channel measurements. Linear time-invariant (LTI) filters at the end of each channel are designed to compensate for non-uniform channel time offsets and quantization noise. We represent the effect of a linear quantizer in each channel with an additive white noise model. In the absence of quantization error, the output corresponds to uniform samples of the input at the Nyquist rate, and thus can be perfectly reconstructed. Theoretical analysis suggests that appropriate choice of quantizer step size and timing offsets in each channel may achieve a better signal to quantization noise ratio (SQNR) than equal step sizes and uniform sampling.

We implement the interleaved multi-channel system and optimal reconstruction filters in software to verify these results. This process involves several operations, including calculating the optimal reconstruction filter frequency response and performing filtering in the frequency domain. The model has three objectives. First, we verify that perfect reconstruction is achieved for a bandlimited signal with no additive noise source. Second, the theoretical predictions are verified using an additive white noise source to represent quantization. Finally, we test the predictions with an ideal quantizer in place of the additive noise source.

8. Design of Discrete-Time Filters for Computational Efficiency

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This research addresses the design of discrete-time filters with the explicit objective of reducing computational complexity, both as an end in itself and as a means toward improving other cost metrics such as power consumption and hardware usage. The current focus is on the design of sparse filters, i.e., filters with few non-zero coefficients relative to conventional designs. Our work is also applicable to designing linear sensor arrays where sparse designs allow for the elimination of array elements.

Recently we have been investigating the design of sparse filters under a quadratic constraint, a general framework that has applications in signal detection and estimation in addition to frequency response approximation. The problem of sparse filter design involves combinatorial optimization and is in general difficult to solve. We have discovered efficient design algorithms for cases in which the matrix in the quadratic constraint has special structure. For the general case, we have developed efficiently solvable relaxations of the problem that yield lower bounds on the true minimum number of non-zero coefficients. The quality of the lower bounds resulting from these relaxations has been studied both theoretically and numerically. A relaxation based on the substitution of a diagonal matrix has been found to be particularly useful. Our relaxations can improve the efficiency of algorithms for solving the problem exactly, and can also provide guarantees on the deviation from optimality for designs produced by heuristic algorithms.

We are looking to extend our work to measures of complexity based on the number of bits in the binary representation of filter coefficients. Both the total number of bits and the number of non-zero bits will be considered.

Publications

Journal Articles, Published

T. Baran, D. Wei and A. V. Oppenheim, "Linear Programming Algorithms for Sparse Filter Design," IEEE Transactions on Signal Processing, March 2010.

Journal Articles, Accepted

M.S. Willsey, K.M. Cuomo, A.V. Oppenheim, “Quasi-Orthogonal Wideband Radar Waveforms Based on Chaotic Systems,” IEEE Transactions on Aerospace and Electronic Systems.

M.S. Willsey, K.M. Cuomo, A.V. Oppenheim, "Selecting the Lorenz Parameters for Wideband Radar Waveform Generation," International Journal of Bifurcation and Chaos.

Journal Articles, Submitted

S. Maymon, A. V. Oppenheim, "Sinc Interpolation of Nonuniform Samples," IEEE Transactions on Signal Processing.

Meeting Papers, Published

T. Baran, N. Malyska and T.F. Quatieri, "Preserving the Character of Perturbations in Scaled Pitch Contours," Proceedings of the IEEE ICASSP (Dallas, TX), March 14 - March 19, 2010.

W.M. Campbell, Z.N. Karam, “A Framework for Discriminative SVM/GMM Systems for Language Recognition,” Proceedings of INTERSPEECH09 (Brighton, UK), September 6-10, 2009.

W.M. Campbell, Z.N. Karam, D.E. Sturim “Speaker Comparison with Inner Product Discriminant Functions,” Twenty-Third Annual Conference on Neural Information Processing Systems (NIPS) (Vancouver, B.C., Canada), December 7 – 11, 2009.

Z.N. Karam, W.M. Campbell, “Variability Compensated Support Vector Machines; Applied to Speaker Verification,” Proceedings of INTERSPEECH09 (Brighton, UK), September 6-10, 2009.

S. Maymon, A.V. Oppenheim, "Quantization and Compensation in Sampled Interleaved Multi-Channel Systems," Proceedings of the IEEE ICASSP (Dallas, TX), March 14 - March 19, 2010.

S. Maymon, A.V. Oppenheim, "Randomized Sinc Interpolation of Nonuniform Samples," Proceedings of the 17th European Signal Processing Conference (EUSIPCO 2009) (Glasgow, Scotland), August 24 - 28, 2009.

D.E. Sturim, W.M. Campbell, Z.N. Karam, D.A. Reynolds, F. Richardson, “The MIT-LL 2008 Speaker Recognition System,” Proceedings of INTERSPEECH09 (Brighton, UK), September 6 – 10, 2009.

D. Wei, “Non-Convex Optimization for the Design of Sparse FIR Filters,” IEEE Workshop on Statistical Signal Processing (SSP) (Cardiff, Wales, UK), August 31 – September 3, 2009.

D. Wei, A.V. Oppenheim, "Sparsity Maximization under a Quadratic Constraint with Applications in Filter Design," in Proceedings of the IEEE ICASSP (Dallas, TX), March 14 - March 19, 2010.

Meeting Papers, Accepted

Z. N. Karam, W. M. Campbell, “Graph□Embedding for Speaker Recognition,” INTERSPEECH10.

W.M. Campbell, Z. N. Karam, “Simple and Efficient Speaker Comparison using Approximate KL Divergence,” INTERSPEECH10.

Theses

J.P. Kitchens, "Acoustic Vector-Sensor Array Processing," Doctoral Thesis, June 2010.

K.S.J. Leow, "Reconstruction from Non-uniform Samples," Master's Thesis, February 2010.