Signal Transformation and Information Representation

RLE Group
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Introduction

Tools for Practical Source Coding

The primary focus of our work is the analysis and design of building blocks for practical compression systems. We tend to work at a level of abstraction where our parts fit in many applications, but we also sometimes follow through to final applications. Being practical means that we emphasize structured signal transformations and scalar and lattice quantization. Beyond just compression, we are interested in whole communication systems, including channel coding, networking, and congestion control.

Oversampling

Though it is not obvious on the surface, the power of oversampled representations is central to the digitization that surrounds us in this digital age. For scientific processing but also for most communication and storage, acquired signals are quantized to discrete values in the process of analog-to-digital conversion (ADC). ADC is made orders of magnitude cheaper by having very coarse (e.g., one bit) discretization of a highly oversampled version of a signal; it is much cheaper to run fast than to be accurate in analog electronics. The ubiquity of these techniques in audio processing is evidenced by the obscure "1-bit DAC" imprint on CD players, yet the full power of oversampled representations for higher-dimensional signals remains to be exploited.

Nonlinearities

For reasons of both computational complexity and mathematical elegance, linear transformations are central to the theory and practice of signal processing. But there are many nonlinear operations that are not too difficult to describe or implement that provide very valuable properties. Examples include sorting, as in the Burrows-Wheeler Transform or permutation coding; thresholding, which is prominent in denoising; and pseudolinear integer-to-integer transforms, which are promising for conventional lossy source coding and multiple description coding. We are interested in developing tools based on tractable nonlinearities.

Technology and Pedagogy

The goal in any engineering research should be to aid good engineering, specifically the design of objects and processes for the betterment of the human condition. While we strive to advance technology, at the same time we embrace the additional opportunities that come from being at an

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educational institution. We make some of our contribution by illuminating topics we find important to non-specialists. And we take the time to work beyond the point of having mathematical proof to also have clear, intuitive, and visual demonstrations.

1. Integer-to-integer Transform Coding

Project Staff
Demba Ba, Professor Vivek Goyal

Transform coding is the most commonly used technique for lossy compression of multimedia content. To transform code a vector, one first computes its transform coefficients and then quantizes them. It is well known that under a variety of weak conditions, the Karhunen-Loeve transform (KLT) is the optimal transformation for coding of a Gaussian source. It can also be shown that, at high rate, the performance of the KLT can be matched by a coding technique known as integer-to-integer (i-to-i) transform coding. In i-to-i transform coding, the components of the vector source are first scalar quantized and then transformed using an i-to-i approximation to some properly chosen nonorthogonal linear transform. The effect of the i-to-i transform is to reduce scalar entropy.

Parent/child pairs of wavelet transform coefficients are approximately uncorrelated. If they were jointly Gaussian, the wisdom of KLT coding would tell us that applying another transform would not lead to any gain in source coding efficiency. However, the said pairs show highly non-Gaussian joint statistics. Therefore, although uncorrelated, they are not independent. This means that some gain in performance could be achieved by removing the underlying dependency between parent/child pairs.

Considering all of the above, we evaluated the amount of mutual information present between parent and child under the said non-Gaussian joint statistics. Moreover, we attempted to find the best nonorthogonal linear transform that produces independent coefficients. While the theory of i-to-i transform coding establishes a general advantage over conventional transform coding for non-Gaussian sources, for the case of wavelet-domain image representations, it was found that very little mutual information was present between parent and child.

2. A Scalable Model of Stochastic Signals Based on Sampling of Signals with a Finite Rate of Innovation.

Project Staff
Julius Kusuma, Professor Vivek Goyal

The sampling theory of Shannon, Nyquist and Kotenikov states that it is possible to represent signals from certain bandlimited classes by using uniform sampling. This framework is often taken for granted: there exist rich classes of signals which are not bandlimited in this sense, yet can be represented without error by using uniform sampling at a rate close to its rate of innovation. A theory of sampling based on rate of innovation rather than Fourier bandwidth has recently been developed. It shows that it is possible to reconstruct certain nonbandlimited parametric signals—those with finite rate of innovation (FRI) from uniform samples.

The ongoing research addresses shortcomings in the nascent theory of sampling based on FRI and applications of this theory. FRI sampling lacks the L_2 approximation properties of conventional sampling. Furthermore, since it creates nonlinear mappings between parameters and observations and the only proposed reconstruction algorithms rely on a polynomial root finding step, the noise sensitivity and effects of sampling rate variation are not well understood.

We seek to develop new approximation properties based on multi-scale sampling, including an understanding of the analogue of aliasing, and estimation techniques including convex relaxations and methods for including probabilistic prior information. We have promising preliminary results in the treatment of finite-dimensional discrete-time stochastic processes, and multi-scale sampling schemes based on integrals and splines.

References:

M. Vetterli, P. Marziliano and T. Blu, "Sampling signals with finite rate of innovation", *IEEE Trans. on Signal Processing* 50(6): 1417-1428 (2002).

3. Low Sampling-rate Receivers for Bandwidth-expanding Communication Systems

Sponsors
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Project Staff
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We consider the problem of low-sampling rate high-resolution channel estimation and timing recovery for digital ultra-wide bandwidth (UWB) receivers. We extend some of our recent results in sampling of certain classes of parametric non-bandlimited signals and develop a frequency domain method for channel estimation and synchronization in ultra-wideband systems that uses sub-Nyquist uniform sampling and well-studied computational procedures. In particular, we show that it is possible to obtain high-resolution estimates of all relevant channel parameters by sampling a received signal below the Nyquist rate. Our approach leads to faster acquisition compared to current digital solutions, allows for slower A/D converters, and potentially reduces power consumption of digital UWB receivers significantly. Moreover, the proposed method can be used for identification of more realistic channel models, where different propagation paths undergo different frequency-selective fading.

References:

I. Maravic, M. Vetterli, K. Ramchandran, "High-resolution acquisition methods for wideband communication systems", *Proc. of ICASSP* (2003).
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J.Kusuma, A.Ridolfi and M. Vetterli, "Sampling of communication systems with bandwidth expansion", *Proc. ICC* (2002).
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4. Permutation Codes

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Permutation codes are lossy fixed-length source codes that have a computationally simple encoding scheme, have good operational rate distortion performance, and often produce equiprobable codewords (no entropy coding required). Overcomplete frame expansions of signals followed by scalar quantization show marked reduction in distortion with expansion redundancy. Given the individual advantages of permutation codes and frame expansions, we investigated the properties of joining the two. We established conditions on frames where classical pseudoinverse reconstruction of permutation-coded frame coefficients is a consistent reconstruction scheme. We also demonstrated rate distortion improvement by this scheme over permutation coding alone.

We have also been investigating the use of permutation source codes as joint source channel codes for memoryless sources and the binary symmetric channel, focusing on the index assignment problem. The best method of index assignment would be an exhaustive search over all index assignments, however this is quite computationally difficult. The number of possible permutation codewords for a given block length is the product of the multinomial coefficients of all integer partitions of the block length. As a computationally simpler index assignment method, we have been looking at graph theoretic methods based on the adjacency graphs of the partition regions.

5. Information Transmission

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In an end-to-end information transmission system with channel input costs and a fidelity criterion, the three most basic parameters are average cost, average end-to-end distortion, and number of channel symbols per source symbol. Shannon's joint source channel coding theorem provides conditions when source information may be transmitted over a noisy channel and achieve the fidelity requirement under the cost constraint. The theorem's proof also suggests a separation of source coding and channel coding, where the source code introduces all distortion and the channel code converts the channel into a "noise-free bit pipe," however this separation may require asymptotically long block lengths. If there is a delay requirement on the information, e.g. if the information is used for control, this is objectionable. Since we are concerned with end-to-end distortion, an alternative architecture that has no coding and all distortion introduced by the channel noise may also be optimal. This requires the source and the channel to be matched. We are investigating the properties of such matched systems as well as matching under alternative notions of fidelity.

Some work on neural information storage and retrieval is partially related to this project.

6. Audio Compression Techniques Using Sparse Approximation

Sponsor

National Defense Science and Engineering Graduate Fellowship (NDSEG), High Performance Computing Modernization Program (HPCMP), contract # F49620-02-C-0041.

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In this project we are investigating the use of sparse approximation to compress audio signals. Since sparse approximation is nonlinear, this approach differs from modern audio compression techniques (such as MPEG Layer-III) that are based on linear transforms.

Two approaches to compressing audio data were developed. In the first, small segments of time-domain audio data are sparsely approximated. In the second, perceptually relevant frequency domain data of an audio signal (i.e., magnitude and phase data) are sparsely approximated using a Perceptual Matching Pursuit (PMP) algorithm. Each approach then quantizes and entropy codes the data that is obtained from approximating the signal. Experiments showed that the former approach can compress audio data by a factor of 6.75:1 while maintaining low distortion of the reconstructed signal. Another approach currently under development uses the MPEG Layer-1 compression algorithm's analysis filterbank to produce spectral coefficients and then compresses these coefficients using sparse approximation methods.

None of the techniques beat the 10.7:1 compression ratio of the MPEG Layer-III (MP3) algorithm, but the results suggest that it may be possible to meet or beat the performance of MP3 if more sophisticated sparse approximation techniques are developed.

Publications

Journal Articles, Submitted for Publication

A.K. Fletcher, S. Rangan, V.K. Goyal, and K. Ramchandran, "Denoising by Sparse Approximation: Error Bounds Based on Rate-Distortion Theory," submitted to *EURASIP J. Appl. Signal Proc*.

L.R. Varshney and D.B. Chklovskii, "Unreliable Synapses and Information Storage," submitted to *Neuron*.

Meeting Papers, Presented

L.R. Varshney and D.B. Chklovskii, "Reliability and Information Storage Capacity of Synapses," paper presented at 2005 Cold Spring Harbor Laboratory Meeting on Learning & Memory, Cold Spring Harbor, New York, April 20-24, 2005.

Meeting Papers, Published

A.K. Fletcher, S. Rangan, V.K. Goyal, and K. Ramchandran, "Robust Predictive Quantization: A New Analysis and Optimization Framework," *Proceedings of the IEEE International Symposium on Information Theory*, Chicago, Illinois, June 27-July 2, 2004

A.K. Fletcher, S. Rangan, V.K. Goyal, and K. Ramchandran, "Optimized Filtering and Reconstruction in Predictive Quantization with Loses," *Proceedings of the IEEE International Conference on Image Processing*, Singapore, October 24-27, 2004.

A.K. Fletcher, S. Rangan, V.K. Goyal, and K. Ramchandran, "Analysis of Denoising by Sparse Approximation with Random Frame Asymptotics," *IEEE International Symposium on Information Theory*, forthcoming.