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Editorial

Adaptive Partial-Update and Sparse System Identification

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System identification is an important task in many application areas including, for example, telecommunications, control engineering, sensing, and acoustics. It would be widely accepted that the science for identification of stationary and dynamic systems is mature. However, several new applications have recently become of heightened interest for which system identification needs to be performed on high-order moving average systems that are either sparse in the time domain or need to be estimated using sparse computation due to complexity constraints. In this special issue, we have brought together a collection of articles on recent work in this field giving specific consideration to (a) algorithms for identification of sparse systems and (b) algorithms that exploit sparseness in the coefficient update domain. The distinction between these two types of sparseness is important, as we hope will become clear to the reader in the main body of the special issue.

A driving force behind the development of algorithms for sparse system identification in telecommunications has been echo cancellation in packet switched telephone networks. The increasing popularity of packet-switched telephony has led to a need for the integration of older analog systems with, for example, IP or ATM networks. Network gateways enable the interconnection of such networks and provide echo cancellation. In such systems, the hybrid echo response is delayed by an unknown bulk delay due to propagation through the network. The overall effect is, therefore, that an "active" region associated with the true hybrid echo response occurs with an unknown delay within an overall response window that has to be sufficiently long to accommodate the worst case bulk delay. In the context of network echo cancellation the direct application of well-known algorithms, such as normalized least-mean-square (NLMS), to sparse system identification gives unsatisfactory performance when the echo response is sparse. This is because the adaptive algorithm has to operate on a long filter and the coefficient noise for near-zero-valued coefficients in the inactive regions is relatively large. To address this problem, the concept of proportionate updating was introduced.

An important consideration for adaptive filters is the computational complexity that increases with the number of coefficients to be updated per sampling period. A straightforward approach to complexity reduction is to update only a small number of filter coefficients at every iteration. This approach is termed partial-update adaptive filtering. Two key questions arise in the context of partial updating. Firstly, consideration must be given as to how to choose which coefficients to update. Secondly, the performance and complexity of the partial update approach must be compared with the standard full update algorithms in order to assess the cost-to-benefit ratio for the partial update schemes. Usually, a compromise has to be made between affordable complexity and desired convergence speed.

We have grouped the papers in this special issue into four areas. The first area is sparse system identification and comprises three papers. In "Set-membership proportionate affine projection algorithms," Stefan Werner et al. develop affine projection algorithms with proportionate update and set membership filtering. Proportionate updates facilitate fast convergence for sparse systems, and set membership filtering reduces the update complexity. The second paper in this area is "Wavelet-based MPNLMS adaptive algorithm for network echo cancellation" by H. Deng and M. Doroslovački, which develops a wavelet-domain μ-law proportionate NLMS algorithm for identification and cancelling of sparse telephone network echoes. This work exploits the whitening and good time-frequency localisation properties of the wavelet transform to speed up the convergence for coloured input signals and to retain sparseness of echo response in the wavelet transform domain. In "A low delay and fast converging improved proportionate algorithm for sparse system identification," Andy W. H. Khong et al. propose a multidelay filter (MDF) implementation for improved proportionate NLMS for sparse system identification, inheriting the beneficial properties of both; namely, fast convergence and computational efficiency coupled with low bulk delay. As the authors show, the MDF implementation is nontrivial and requires time-domain coefficient updating.

The second area of papers is partial-update active noise control. In the first paper in this area "Analysis of transient and steady-state behavior of a multichannel filtered-x partial-error affine projection algorithm," A. Carini and S. L. Sicuranza apply partial-error complexity reduction to filtered-x affine projection algorithm for multichannel active noise control, and provide a comprehensive analysis of the transient and steady-state behaviour of the adaptive algorithm drawing on energy conservation. In "Step size bound of the sequential partial update LMS algorithm with periodic input signals" Pedro Ramos et al. show that for periodic input signals the sequential partial update LMS and filtered-x LMS algorithms can achieve the same convergence performance as their full-update counterparts by increasing the step-size appropriately. This essentially avoids any convergence penalty associated with sequential updating.

The third area focuses on general partial update algorithms. In the first paper in this area, "Detection guided fast affine projection channel estimator for speech applications," Yan Wu Jennifer et al. consider detection guided identification of active taps in a long acoustic echo channel in order to shorten the actual channel and integrate it into the fast affine projection algorithm to attain faster convergence. The proposed algorithm is well suited for highly correlated input signals such as speech signals. In "Efficient multichannel NLMS implementation for acoustic echo cancellation," Fredric Lindstrom et al. propose a multichannel acoustic echo cancellation algorithm based on normalized least-mean-square with partial updates favouring filters with largest misadjustment.

The final area is devoted to blind source separation. In "Time domain convolutive blind source separation employing selective-tap adaptive algorithms," Q. Pan and T. Aboulnasr propose time-domain convolutive blind source separation algorithms employing M-max and exclusive maximum selective-tap techniques. The resulting algorithms have reduced complexity and improved convergence performance thanks to partial updating and reduced interchannel coherence. In the final paper "Underdetermined blind audio source separation using modal decomposition," Abdeljalil Aïssa-El-Bey et al. present a novel blind source separation algorithm for audio signals using modal decomposition. In addition to instantaneous mixing, the authors consider convolutive mixing and exploit the sparseness of audio signals to identify the channel responses before applying modal decomposition.

In summary, we can say that sparseness in the context of adaptive filtering presents both challenges and opportunities. Standard adaptive algorithms suffer a degradation in performance when the system to be identified is sparse. This has created the need for new algorithms for sparse adaptive filtering—a challenge that has been well met to date for the particular applications addressed. When sparseness exists, or can be safely assumed, in input signals, this can be exploited to achieve both computational savings in partial update schemes and, in certain specific cases, performance improvements. There remain several open research questions in this context and we look forward to an ongoing research effort in the scientific community and opportunities for algorithm deployment in real-time applications.

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