Radioelectronics and Communications Systems
Vol. 49, No. 1, pp. 10–17, 2006

Izvestiya VUZ. Radioelektronika
Vol. 49, No. 1, pp. 15–26, 2006
UDC 621.513.6

PARALLEL LADDER ALGORITHMS BASED ON A PRIORI ERRORS OF PREDICTION AND SIMULATION

V. I. Djigan

GUP NPTs "ELVIS", Moscow, Russia

The paper considers parallel computations in ladder fast RLS-algorithms with a sliding window and/or dynamic regularization of the correlation matrix of adaptive filter. Computational procedures of the algorithms using a priori error of linear prediction and simulation are presented, and assessment of arithmetic complexity of the algorithms is carried out. The algorithms can be employed for processing nonstationary signals in applications not requiring calculation of weight coefficients of adaptive filters in the explicit form, where the sought output signal represents the filter error signal.

Among the large variety of algorithms employed in adaptive filters, the recursive algorithms using the least squares criterion (Recursive Least Squares — RLS) [1] are considered more effective (from the viewpoint of their transient process duration and the level of residual error in steady-state conditions) than the simplest gradient algorithm, using the least mean square criterion (LMS) [2], or its modifications. All RLS algorithms are based on calculation of weight coefficients in Wiener's filter. The complexity of these algorithms equals $O(N^2)$ arithmetic operations per iteration, or $O(N)$ operations in their efficient (fast) versions. Because of this, RLS-algorithms are less popular from the computational viewpoint as compared with the LMS-algorithm whose arithmetic complexity is $2N < 7N$, where $7N$ is complexity of the simplest fast RLS-algorithms: Fast a Posteriori Error Sequential Technique (FAEST) and Fast Transversal Filter (FTF), while N is the number of weight coefficients of the adaptive filter. At the same time, RLS-algorithms find application in adaptive filters, since the arithmetic complexity is no obstacle to implementation of these filters. The high-speed digital signal processors make it possible to realize efficiently very sophisticated algorithms.

When processing nonstationary signals, the arithmetic complexity of RLS-algorithms increases in double measure because of using the sliding window to estimate the correlation matrix of input signals of the adaptive filter [3] or regularization of this matrix [4]. It means that the complexity may increase by a factor of four if both techniques are used simultaneously. The above is also true for the ladder-type RLS-algorithms [5], where each iteration, recursive in time, includes N iterations to be performed when changing the order of the adaptive filter from $n = 1$ to $n = N$. For the ladder filters we could not find in literature any information about application of the sliding window and of regularization.

The parallel ladder RLS-algorithms. In the present work we consider several types of ladder adaptive RLS-algorithms with the sliding window and/or regularization, with the use of parallel computations [6–8]. When using the floating-point arithmetic, and at the appropriate initialization, these algorithms are mathematically equivalent to other known RLS-algorithms. By the equivalence is meant the same functioning during each iteration in the case when the adaptive filters under comparison have identical parameters and process identical input $x(k)$ and output $d(k)$ signals, where k is the index of discrete time. In our work we deal with algorithms for single-channel adaptive filters with complex weight coefficients $h_n(k)$ (see Fig. 1).

REFERENCES

1. V. I. Dzhigan, Uspekhi Sovremennoy Radioelektroniki, No. 11, pp. 48–77, 2004.

2. B. Widrow, IEEE Signal Processing Magazine, Vol. 22, No. 1, pp. 100–106, 2005.

3. V. I. Djigan, Unified approach to the fast time recursive least square adaptive filtering algorithms development, Proceedings of the 3rd International Conference on Antennas, Radio Communication Systems & Means (ICARSM-97) (Voronezh, 26–29 May 1997), Voronezh, Vol. 3, pp. 33–42. 1997.

4. S. L. Gay, Dynamically regularized fast RLS with application to echo cancellation, Proceedings of the International Conference on Acoustic Speech and Signal Processing (ICASSP-96), pp. 957–960, Atlanta (USA), May 7–9, 1996.

5. S. Haykin, Adaptive Filter Theory (4th edition), Prentice Hall, 2001.

6. K. Zhao, F. Ling, H. Lev-Ari, and J. G. Proakis, IEEE Trans. Signal Processing, Vol. 42, No. 8, pp. 1961–1972, 1994.

7. V. I. Djigan, Diversity of RLS for parallel implementation, Proceedings of the 5th World Scientific and Engineering Academy and Society (WSEAS) International Conference on Information Science, Communications and Applications (ISCA-2005) (May 12–15, 2005, Cancun), pp. 1–6 (CD), Cancun, Mexico, 2005.

8. V. I. Dzhigan, Izv. VUZ. Radioelektronika, Vol. 48, No. 12, pp. 61–70, 2005.

9. G. Zelniker and F. J. Taylor, Advanced Digital Signal Processing: Theory and Applications, Marcel Dekker, Inc., New York, 1994.

10. A. A. Giordano and F. M. Hsu, Least Square Estimation with Application to Digital Signal Processing, John Wiley and Sons, Inc., Canada, Toronto, 1985.

11. F. Ling, IEEE Trans. Signal Processing, Vol. 39, No. 7, pp. 1541–1551, 1991.

12. V. I. Djigan, Engineering solutions in adaptive signal processing, Proceedings of the 6th International Conference on Prospective Technologies in the Mass Media (April 20–22, 2005, Vladimir, Russia), pp. 250–253, Vladimir, 2005.

1 September 2005