A Brief Survey of Time- and Frequency-Domain Adaptive Filters

Andreas Spanias
SenSIP Center, School of ECEE
Arizona State University
Tempe, AZ 85287, USA
spanias@asu.edu

***Abstract*—This tutorial session paper provides a brief survey of adaptive signal processing methods and applications. We give the update equations of FIR and IIR adaptive filters. We provide brief descriptions of LMS, RLS, and block time and frequency domain algorithms. The tutorial begins with an introduction to adaptive methods and continues with a discussion on least squares and gradient techniques. The session covers various adaptive filter structures and provides information and references on several applications including system identification, linear prediction, noise cancellation, and adaptive arrays.**

Keywords—Adaptive Filters, Linear Prediction, LMS, RLS, System Identification, Noise and Echo Cancellation, Tutorial in Adaptive Signal Processing.

BRIEF SURVEY

This section provides a brief survey of adaptive algorithms for filtering applications. Discrete-time adaptive signal processing (ASP) algorithms [1-5] and more specifically Least Mean Square (LMS) [1] and recursive least square adaptive filtering methods have found several applications including: noise [6-10] and echo cancellation [11-20,55,160], channel equalization [21-28,122], active noise control [9,60], adaptive arrays [29-33], biomedical signal analysis [34,35,61,116,209], adaptive prediction [36-41], and system identification [42-45,158,161]. We begin our tutorial with the adaptive linear combiner [1] shown in Fig. 1. The error is the difference between the desired signal and the filter output. Mean square error minimization gives the Wiener solution [1], i.e., $\mathbf{b}=\mathbf{R}_{xx}^{-1}\mathbf{r}_{dx}$, where $\mathbf{b}$ is the weight vector, $\mathbf{R}_{xx}$ is the autocorrelation matrix and $\mathbf{r}_{dx}$ is the crosscorrelation vector.

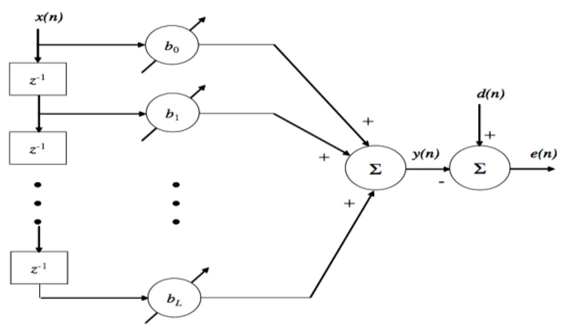


Fig. 1. The FIR Adaptive Filter.

The sequential LMS equation for adapting $\mathbf{b}$ is given by

$$\mathbf{b}(n+1)= \mathbf{b}(n)+2\mu\mathbf{x}(n)e(n)$$

where μ is the step size or convergence factor. The FIR coefficient vector converges in the mean to the Wiener solution if $0<\mu<1/\lambda_{max}$. where λ_{max} is the largest eigenvalue of the auto correlation matrix. Under a sequence of assumptions, it can be shown that misadjustment is approximately: $M \approx \mu$ trace($\mathbf{R}_{xx}$) [1]. The step size in many applications is typically normalized [45]. Optimal step sizes can also be derived for LMS algorithms [46, 181]. Block LMS (BLMS) adaptive algorithms [47] process data one block at a time and the gradient is computed from a block of error samples. The error vector and update equations for the BLMS are given by:

$$\mathbf{e}(i)=\mathbf{d}(i)-\mathbf{x}_B(i)\mathbf{b}(i); \quad \mathbf{b}(i+1)= \mathbf{b}(i)+\frac{2\mu}{N}\mathbf{x}_B^T(i)\mathbf{e}(i).$$

Block algorithms with optimal step size were proposed in [48]. All the sequential algorithms with a single step size have slow convergence when $\mathbf{R}_{xx}$ has large eigenvalue spread. Recursive Least Squares (RLS) type algorithms [2,3,50,51] generally converge more rapidly than LMS filters but also have considerably higher complexity. An RLS algorithm can be formulated using the matrix inversion lemma. The RLS updates filter coefficients using the following equation:

$$\mathbf{b}(n+1)= \mathbf{b}(n)+\mathbf{k}(n+1)\hat{e}(n+1)$$

where the gain $\mathbf{k}(n+1)$ is given by

$$\mathbf{k}(n+1)=\frac{\beta^{-1}\hat{\mathbf{R}}_{xx}^{-1}(n)\mathbf{x}(n+1)}{1+\beta^{-1}\mathbf{x}^T(n+1)\hat{\mathbf{R}}_{xx}^{-1}(n)\mathbf{x}(n+1)}.$$

Modified and fast RLS algorithms have been proposed in [11,52,53,55, 176]. Fast RLS for nonlinear Volterra filters have been proposed in [56-59]. RLS can be viewed as a special case of a Kalman filter [2,195]. Lattice structures for adaptive FIR filters and for linear prediction systems have been proposed in [63-66,145,184-187] with demonstrated advantages in fixed-point implementations [66]. Frequency-domain adaptive algorithms [67] use the FFT for fast convolution, Fig. 2.

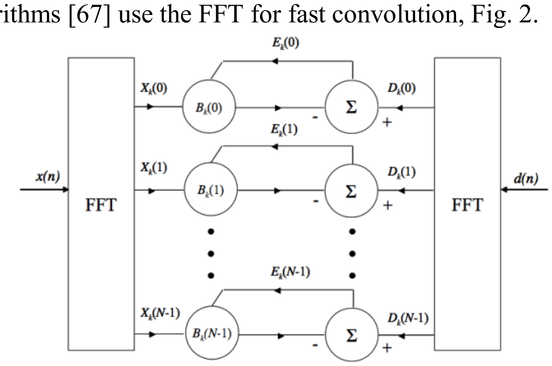


Fig. 2. The Frequency-Domain Adaptive Filter.

The update equation in this case is based on the complex LMS algorithm [68]. Computation of the output of the filter and the gradient is based on circular convolutions [70]. Analysis of frequency domain LMS algorithms was presented in [70]. FFT-based adaptive FIR algorithms that use the overlap and save method for output and gradient computation are presented in [71,72] and with optimal step size in [43,44]. Unconstrained FFT-based adaptive FIR algorithms are given in [73]. Relations of adaptive multirate filters and their frequency domain counterparts are established in [74]. Adaptive frequency-domain algorithms for multichannel noise reduction have been presented in [169]. IIR frequency domain adaptive filters are derived in [75]. Relations between the LMS and the running DFT have been derived in [144]. IIR adaptive filters are more difficult to analyze and implement because of the inherent nonlinearity of the error w.r.t. feedback parameters [78-91,172,177,189]. Simple adaptive IIR filters use the equation error (Fig. 3) model [77]. Although the equation error sequence is linear w.r.t. the feedback parameters, the algorithm converges to a biased parameter vector.

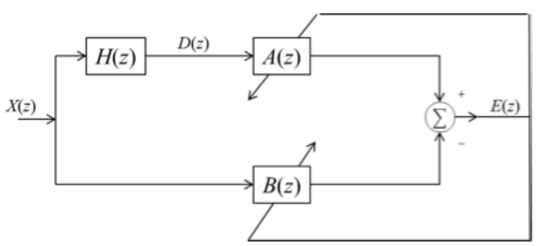


Fig. 3. The Equation Error IIR Adaptive Filter.

The error, update, data vector, and coefficient vector of the equation error IIR adaptive filter are given by

$$e(n) = d(n) + \mathbf{g}^T(n)\mathbf{c}(n), \quad \mathbf{c}(n+1) = \mathbf{c}(n) + 2\mu e(n)\mathbf{g}(n).$$

$$\mathbf{g}(n) = [-x(n) \quad -x(n-1) \quad \cdots \quad -x(n-L) \quad d(n-1) \quad d(n-2) \quad \cdots \quad d(n-M)]^T,$$

$$\mathbf{c}(n) = [b_0(n) \quad b_1(n) \quad \cdots \quad b_L(n) \quad a_1(n) \quad a_2(n) \quad \cdots \quad a_M(n)]^T$$

Algorithms based on the output error model (Fig. 4) have also been proposed in several papers. One of the simplest ones is the Simple Hyperstable Adaptive Recursive Filter (SHARF) [84].

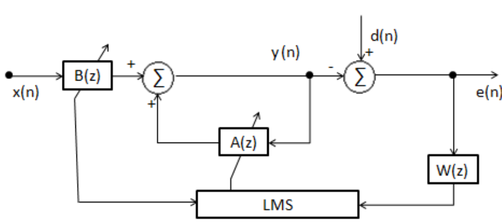


Fig. 4. SHARF Output Error IIR Adaptive Filtering.

Some more recent work in IIR adaptive filtering suggests using genetic algorithms [83]. All-pole models for filter synthesis can be obtained by making $B(z)=0$ in the structure of Figure 3. It can be shown easily that by minimizing the MSE with $B(z)=0$, *in* the structure of Figure 3, the system essentially becomes equivalent to linear prediction (LP) [37,38]. Adaptive Linear prediction algorithms have been proposed for spectral analysis [92,93] and other applications. Adaptive linear prediction has found applications in speech and audio coding [94-97] and more specifically ADPCM [98-100]. RLS "batch" type algorithms for linear prediction, and more specifically Levinson-Durbin type algorithms [37] are at the core of most LPC and CELP speech codecs [39,124]. Applications of LP to genomics [125], simulation environments for LP [126,127], applications to EMG [209] and extensions to pole-zero prediction [40] also appeared in the literature. Reference [37] is a comprehensive tutorial for LP.

Differential coding also uses a variant of LMS called the sign error LMS which also found applications in other quantized filter and control implementations. To that end, there are several variants of LMS algorithms [101,210]. Publication [101] and references therein describe several LMS variants including the normalized LMS, the leaky LMS, the LMS with dead-zones, the sign-sign LMS, and the median LMS. Other LMS type algorithms and fixed point effects have been reported in [130-134,161,168,208]. LMS algorithms with bounded error constraints have been proposed in [42,110,111]. LMS algorithms have been used in hard drive equalization [112,123] and hard drive control [113]. Diffusion and consensus based adaptive techniques have been proposed in [135-139, 146-150, 182]. In addition to diffusion techniques, adaptive methods using kernel mapping, have also been developed. [151]. Learning systems and adaptive gradient and RLS methods have also been used in neural networks [152-154]. The x-filtered LMS is used in several control applications and has been applied in active noise cancellation [7,9,80,114,156,157,162]. LMS algorithms for 2-dimensional processing are described in [102-106]. Filter bank and sub-band adaptive algorithms [107-109,117-121,140-143,155,159] have been developed and deployed in audio and image/video applications. Adaptive filters with sparse impulse responses (Fig. 5) have been useful in echo cancellation and other applications [115,128,129,202].

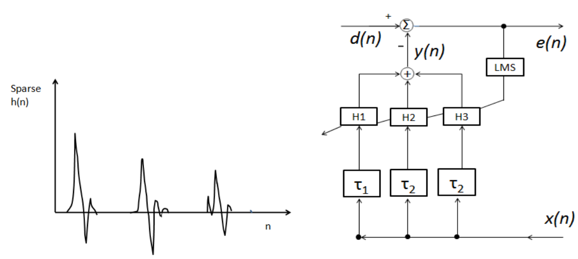


Fig. 5. Sparse adaptive signal processing; a) sparse impulse response, b) simple sparse adaptive filter.

.

Sub-band adaptive filters can be implemented using multirate structures (Fig. 6) which allow separate step-sizes and customized adaptation strategies for each sub-band.

ADAPTIVE SIGNAL PROCESSING APPLICATIONS

As mentioned before there are several applications of adaptive algorithms including adaptive noise and echo cancellation [205,207], channel equalization [21,188], adaptive line enhancement [178,179,180], active noise control [156], sensor arrays [170,171], estimation of head related transfer functions [166], hearing aids [163-165], sound processing and effects [167,173,174,175], machine learning [183], network applications [148] and neural nets [154]. Array processing using LMS and RLS algorithms has been used in several areas including adaptive antennas, radar, and linear and circular microphone sensor arrays. The weights of the array [170] control the beamwidth, sidelobes and directivity.

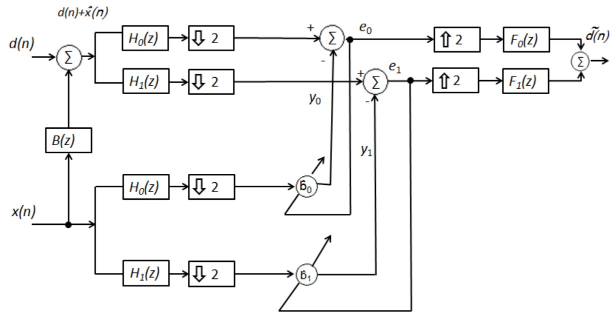
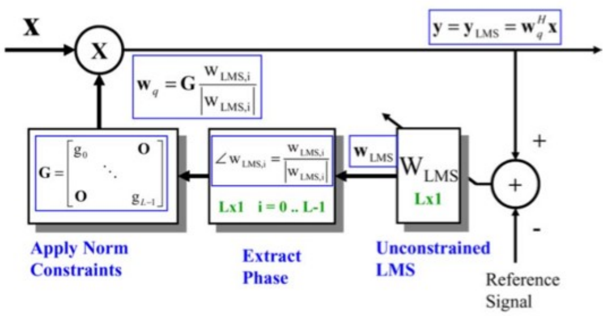


Fig. 6. Simple two sub-band adaptive algorithm.

An LMS algorithm with window constrains has been used to adapt the beam directivity while keeping low level sidelobes [192]. This is achieved by forcing constrains on the LMS array parameters (Fig. 7).



Fig, 7. LMS beamforming with window constraints for beam steering with suppressed sidelobes [192].

Adaptive arrays have also been used for acoustic signal localization [175], MIMO antenna systems [204], and GPS [203]. Microphone arrays [171,196-201] have also been used in multichannel adaptive echo cancellers, active noise control systems (Fig. 8), and smart internet of things (IoT) devices. In particular, adaptive circular microphone arrays [200, 201] are embedded in modern IoT speakers.

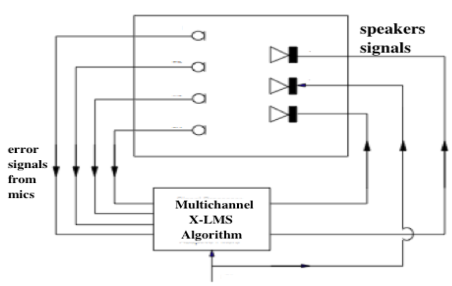


Fig. 8. Experimental Platform for adaptive multichannel active noise control [194].

Adaptive algorithms have also been used in subspace Wiener filters [193] and in orthogonal frequency-division multiplexing (OFDM) systems for channel estimation. In OFDM each channel is assumed constant for each symbol. The received signal is a combination of the transmitted signals. Additive noise is assumed and frequency-domain channel equalization is part of the system. The channel estimates can be obtained with the LMS or RLS algorithms [204]. DFTs and IDFTs are implemented efficiently with FFT algorithms [204].

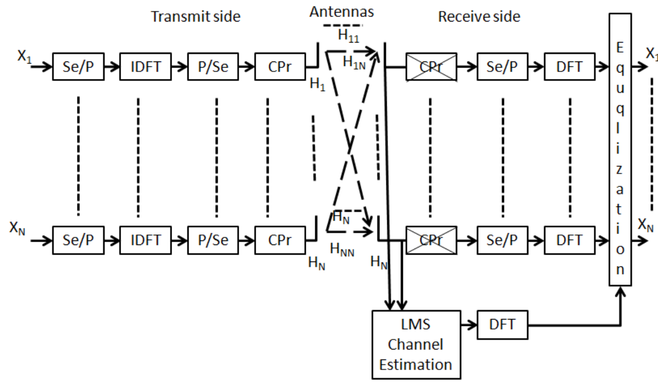


Fig. 9. OFDM with LMS Channel Estimation [204].

A multichannel echo canceller with M speakers and N microphones is shown in Fig. 10 [206]. There are MN echo acoustic paths to be estimated. DFTs are used for decoupling and the transformations **G** are used to decompose MIMO modes into single input/single output (SISO) modes. Efficient methods for estimating the echo paths are presented in [206].

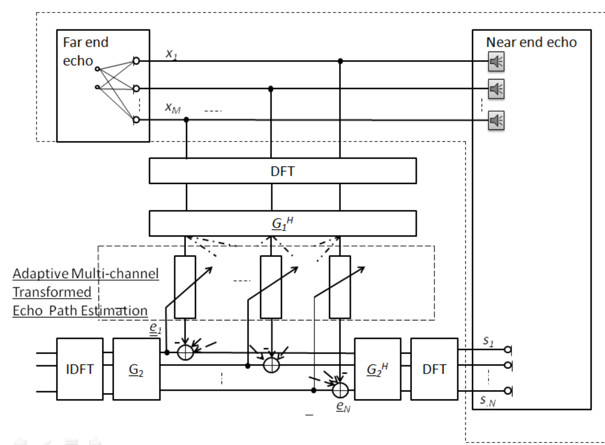


Fig. 10. MIMO Adaptive Echo Canceller [206].

Conclusion

This adaptive signal processing tutorial summary described briefly some of the basic methods and applications of adaptive filtering. The paper gave short overviews of algorithms and provided an extensive list of references.

Acknowledgment

Portions of this paper form the base of a STEM module on adaptive filters created as a product of the NSF IUSE award 1525716. We also acknowledge several industry partners of the SenSIP Center (sensip.asu.edu).

REFERENCES

[1] B. Widrow, S. Stearns, Adaptive Signal Processing, Prentice Hall, 1985.

[2] S. Haykin, Adaptive Filter Theory, Prentice Hall, 3rd Ed., 1997.

[3] Ali Sayed, Adaptive Filters, Wiley, 2011.

[4] Paulo Diniz, Adaptive Filtering, 4th Ed., Springer, 2013.

[5] T. Adali and S. Haykin, Adaptive Signal Processing: Next Generation Solutions, 2010.

[6] B. Widrow et al, Adaptive noise cancelling: Principles and applications, *Proc. IEEE,* Vol. 63(12), Jan. 1976.

[7] C. N. Hansen, Understanding Active Noise Cancellation, CRC, 2001.

[8] W. Mikhael, P. Hill, Performance evaluation of a real-time TMS32010-based adaptive noise canceller (ANC), *IEEE Trans.ASSP,* p. 411, 1988.

[9] Q. Shen+ and A. Spanias, "Adaptive Active Sound Reduction, *Noise Control Engineering Journal*, J44 (6), pp. 281-293, Nov. 1996.

[10] M. Deisher and A.S. Spanias, "Practical Considerations in the Implementation Frequency-Domain Adaptive Noise Cancellation," *IEEE Trans. CAS*, Part II, Vol. 41(2) pp. 164-168, Feb. 1994.

[11] T. Petillon, A. Gilloire, and S. Theodoridis, "Complexity reduction in fast RLS transversal adaptive filters with application to acoustic echo cancellation", *IEEE ICASSP-92* ,vol. 4, pp. 37-40, March 1992.

[12] G. Egelmeers, P. Sommen, and J. de Boer, "Realization of an acoustic echo canceller on a single DSP", *8th EUSIPCO*, pp. 1-4, Sept. 1996.

[13] A. Sugiyama et al, "A versatile echo-and-noise control suite for adverse environments with a user-control capability",*Digest of Technical Papers International Conference on Consumer Electronics*, pp. 1-2, Jan. 2009

[14] M.J. Smith, C.Cowan, P.F. Adams, "Nonlinear echo cancellers based on transpose distributed arithmetic", *IEEE Tran. CAS,* pp. 6-18, Jan. 198J.

[15] Benesty, T. Gaensler, D.R. Morgan, M.M.Sondhi, S. L. Gay, "Advances in Network and Acoustic Echo Cancellation," Springer, Berlin, 2001.

[16] D. Falconer, "Adaptive Reference Echo Cancellation," IEEE Trans. on Communications, Vol. COM-30 (9), Sept. 1982.

[17] D. L. Duttweiler, "A Twelve-Channel Digital Echo Canceller," *IEEE Trans. on Communications*, pp. 647-653 (May 1978).

[18] Eberhard Hänsler Gerhard Schmidt, Acoustic Echo and Noise Control: A Practical Approach, Wiley, Feb 2005.

[19] J. Benesty C. Paleologu T. Gänsler S. Ciochină, A Perspective on Stereophonic Acoustic Echo Cancellation, Springer, 2011.

[20] S. L Gay, Subband echo canceller including real time allocation among the subbands, US Patent 5,001,701.

[21] J. Lin, F. Ling, and J. G. Proakis, "An optimal windowed recursive least squares algorithm for fading channel estimation", *IEEE ISCAS*, vol. 1, pp. 734-737, May 1993.

[22] E. Eleftheriou and D. Falconer, "Adaptive Equalization Techniques for HF Channels," *IEEE Select Areas in Communications*, p.248, Feb. 1987.

[23] R.W.Lucky, Techniques for Adaptive Equalization of Digital Communication Systems*, Bell Tech. J.,* Vol. 45, pp. 255–286, Feb. 1966

[24] Treichler, J.R. and Agee, B.G., A new approach to multipath correction of constant modulus signals, *IEEE Trans. ASSP* -31:349-372, 1983.

[25] Qureshi,S.,Adaptive equalization, *Proc. IEEE,* p. 1349–87, Sept.,1985.

[26] G. Giannakis, S. Halford, "Blind fractionally spaced equalization of noisy FIR channels: Direct and adaptive solutions," *IEEE Trans. Signal Processing,* vol. 45, pp. 2277-2292, Sept. 1997.

[27] Hatzinakos, D.and Nikias,C.L., Blind equalization using a tricepstrum based algorithm,I EEE Trans. Commun., 39: 669–682, May, 1991.

[28] V. Atti, A.S. Spanias, K. Tsakalis. C. Panayiotou, V. Berisha+, L. Iasemidis, *"*Gradient projection-based channel equalization under sustained fading*," EURASIP Signal Processing*, 88, p.236, Feb. 2008.

[29] J. Griffiths, "Adaptive array processing. A tutorial," Proc. IEE Communications, Radar and Signal Processing, Vol 130, Feb. 1983.

[30] J. Foutz, A. Spanias, M. Banavar, Narrowband Direction of Arrival Estimation for Antenna Arrays, Synthesis Lectures on Antennas, Morgan & Claypool Publishers, ISBN-13: 978-1598296501, Aug. 2008.

[31] J. Foutz+, A. Spanias, S. Bellofiore and C Balanis, " Adaptive Eigen-Projection Beamforming Algorithms for 1-D and 2-D Antenna Arrays, *IEEE Antennas and Propagation Letters*, Vol. 83, pp. 1929-1935, 2003.

[32] S. Bellofiore, C. Balanis, J. Foutz, A, Spanias, "Smart Antennas Systems for Mobile Communications Networks: Part 1: Overview of the Antenna Design," *IEEE Ant. and Prop. Mag.,* pp. 145-154, June 2002.

[33] S. Bellofiore, C. Balanis, J. Foutz, A,S, Spanias, "Smart Antennas Systems for Mobile Communications Networks: Part 2: Algorithms," *IEEE Antennas and Propagation Magazine*, pp. 106 -114, Aug. 2002.

[34] Thakor, N.V.; Zhu, Yi-Sheng, "Applications of adaptive filtering to ECG analysis: noise cancellation and arrhythmia detection". *IEEE Transactions on Biomedical Engineering,* 38 (8): 785–794., 1991.

[35] Ferrara, E., Widrow, B., "Fetal electrocardiogram enhancement by time-sequenced adaptive filtering", IEEE Trans.Biom. Eng. p 458, 1982.

[36]A. Spanias, Digital Signal Processing; An Interactive Approach – 2nd Edition, ISBN 978-1-4675-9892-7, *Lulu Press*, May 2014.

[37] J. Makhoul, "Linear Prediction: A Tutorial Review," *Proc. IEEE*, pp. 561-580, April 1975.

[38] P.P. Vaidyanathan, Theory of Linear Prediction, Morgan & Claypool, Ed. J. Mura, 2008.

[39]. Ramamurthy and A. Spanias, MATLAB® Software for the Code Excited Linear Prediction Algorithm: The Federal Standard-1016, Morgan and Claypool Publishers, vol. 2, no. 1, ISBN 1608453847, Jan 2010.

[40] W.B. Mikhael, A.S. Spanias, G. Kang L. Fransen,"A Two-Stage Pole-Zero Predictor," *IEEE Trans. CAS*, Vol. 33, pp. 352-354, March 1986.

[41] A.S. Spanias, *IEEE Transactions on Signal Processing*, "A Block Time and Frequency Modified Covariance Algorithms for Spectral Analysis," vol. 41, No. 11, pp. 3138-3153, Nov. 1993.

[42] K. Tsakalis, M. Deisher+, and A. Spanias, "System Identification Based on Bounded Error Constraints, *IEEE Trans. on Signal Processing*, Vol. 43, No. 12, pp. 3071-3075, Dec. 1995.

[43] W.B. Mikhael and A.S. Spanias, "A Fast Frequency-Domain Adaptive Algorithm," *Proc. of the IEEE*, Vol. 76, No. 1, pp. 80-82, Jan. 1988.

[44] W.B. Mikhael and A.S. Spanias, "Comparison of Several Frequency-Domain LMS Algorithms," *IEEE Trans.on CAS*, pp. 586-588, May 1987.

[45] S. Douglas, "A family of normalized LMS algorithms," *IEEE SP Letters*, pp. 49-51, March 1994.

[46] Mikhael, W.,F. Wu, L. G. Kazovsky, G. Kang, and L. Fransen, "Adaptive filters with individual adaptation of parameters", *IEEE Trans. CAS,* vol. 33, 7, pp. 677-686, July 1986.

[47] G. Clark, S. Mitra, and S. Parker, "Block Implementation of Adaptive Digital Filters," *IEEE Trans. on ASSP-29,* pp. 744-752, June 1981.

[48] W. Mikhael and F. Wu, "Fast Algorithms for Block FIR Adaptive Digital Filtering," *IEEE Trans. on CAS* -34, pp. 1152-1160, Oct. 1987.

[49] M. Honig and D.G. Messerschmitt, Adaptive Filters: Structures, Algorithms, and Applications, Kluwer Academic Press, 1984.

[50] N. Kalouptsidis and S. Theodoridis, Adaptive System Identification and Signal Processing Algorithms," Prentice Hall 1993.

[51] A. Sayed and T. Kailath, A State Space Approach to RLS, *IEEE Signal Processing Magazine*, 11(3), pp. 18-60, July 1994.

[52] J, Cioffi, T Kailath, "Fast, RLS transversal filters for adaptive filtering," *IEEE Transactions on ASSP*, vol. 32(2), pp. 304-337, Apr. 1984.

[53] H. Kassa, T. Chernet, E. Yohannes, D. Hailemariam, Y.Astatke, F. Moazzami, W. Zegeye, "Investigation of RLS beamforming algorithm in terms of energy efficiency," *Annual CISS*, pp. 82-86, Mar. 2016.

[54] S. Gay, "Dynamically Regularized Fast RLS with application to Echo Cancellation:, *IEEE ICASSP 1996*, p. 957, Atlanta, May 1996.

[55] J. Lee and V.J. Mathews, "A fast recursive least squares adaptive second order Volterra filter and its performance analysis," *IEEE Trans. SP*, pp. 1087-1102, March 1993.

[56] Griffith D W, Arce G R, "A Partially Decoupled RLS Algorithm for Volterra Filters", *IEEE Trans SP*, Vol. 47, No. 2, pp. 579-582, 1999.

[57] T. Ogunfunmi, Adaptive Nonlinear System Identification: The Volterra and Wiener Model Approaches. Springer-Verlag New York, Inc., 2006.

[58] V.J. Mathews, "Adaptive polynomial filters," *IEEE Signal Processing Magazine*, vol. 8, no. 3, pp. 10–26, July 1991.

[59] L. Tan, J. Jiang, "Adaptive Volterra filters for active control of nonlinear noise processes," *IEEE Trans. SP*, vol 49(8), pp. 1667-1676, Aug. 2001.

[60] S.J. Elliott, I.M. Stothers, and P.A. Nelson, "A Multiple Error LMS Algorithm and its Application to the Active Control of Noise and Vibration," *IEEE Trans. ASSP-35 (10)*, pp. 1423- 1434, Oct. 1987.

[61] N. Razzaq, S. Sheikh, M. Salman, and T. Zaidi, “An Intelligent Adaptive Filter for Elimination of Power Line Interference from High Resolution Electrocardiogram”, *IEEE Access,* vol. 4, pp. 1676-1688, 2016.
[62] LJ Griffiths, “A continuously-adaptive filter implemented as a lattice structure”, *IEEE ICASSP-77*, pp. 683-686, May 1977.
[63] B. Friedlander, “Lattice filters for adaptive processing”, *Proceedings of the IEEE*, vol. 70, No. 8, pp. 829-867, Aug. 1982
[64] R. Zhu, F. Yang, and J. Yang, “An RLS-Based Lattice-Form Complex Adaptive Notch Filter”, *IEEE SP Let.,* vol. 23/2, pp. 217-221, Feb. 2016.
[65] P. Regalia, M. Bellanger, "On the duality between fast QR methods and lattice methods in LS adaptive filtering," *IEEE Trans. SP*, p.879, 1991.
[66] C. Paleologu,S. Ciochina, A. Enescu, C. Vladeanu, "Gradient Adaptive Lattice Algorithm Suitable for Fixed Point Implementation," Proc. IEEE ICDT 2008, pp. 41-46, Bucharest, June 1008.
[67] M. Dentino, J. McCool, B. Widrow, "Adaptive filtering in the frequency domain," Proc. of the IEEE, Vol. 66(12), pp. 1658-1659, 1978.
[68] B. Widrow, J. McCool, M. Ball, "The complex LMS algorithm,"Proc. of the IEEE, Vol. 63(4), pp. 719-720, 1975.
[69] C. Cowan and P. Grant, Adaptive Filters, Prentice Hall, 1985.
[70] N. J. Bershad; P. L. Feintuch, "Analysis of the frequency domain adaptive filter," *Proc. of the IEEE,* 1658 - 1659, Vol 67(12), 1979.
[71] E. Ferrrara, "Fast implementations of LMS adaptive filters," *IEEE Transactions ASSP,* Vol. 28(4), pp. 474-475, Aug. 1980.
[72] G. Clark; S. Parker; S. Mitra, "A unified approach to time- and frequency-domain realization of FIR adaptive digital filters," *IEEE Transactions ASSP*, Vol. 31(5), pp. 1073-1083, 1983.
[73] D Mansour, and A. H Gray Jr, “Unconstrained frequency-domain adaptive filter”, *IEEE Trans. ASSP*, vol. 30(5), pp. 726-734, Oct. 1982.
[74] J.J Shynk, “Frequency-domain and multirate adaptive filtering”, *IEEE Signal Processing Magazine*, 1992.
[75] W.B. Mikhael, A.S. Spanias, F.H. Wu, "Fast Frequency-Domain Implementation of a Block IIR Filter with Applications," *Proc. ISCAS-*88, pp. 285-288, Espoo, Finland, June 1988.
[76] J.J Shynk, “Adaptive IIR filtering”, *IEEE ASSP Mag.*, vol. 6/2, Apr. 1989.
[77] W.A. Gardner, "Cascade configurations for recursive-like adaptive noise cancellation," *Proc. of the IEEE*, Vol. 69(7), pp. 846-847, July 1981.
[78] P. Regalia, Adaptive IIR Filtering in Signal Processing and Control, CRC, Marcel Dekker, 1995.
[79] A. Carini, V. J. Mathews and G. L. Sicuranza, “Sufficient Stability Bounds for Slowly-Varying Discrete-Time Recursive Linear Filters and Their Applications in Adaptive IIR Filtering,” *IEEE Trans. Signal Proc.*, Vol. 46(9), pp. 2561-2567, Sept. 1999.
[80] R.Jones, B.L. Olsen, B.R. Mace, "Comparison of convergence characteristics of adaptive IIR and FIR filters for active noise control in a duct," *Elsevier Applied Acoustics*, pp. 729-738, Vol. 68 July 2007.
[81] M, Scarpiniti, D. Comminiello, R. Parisi, A. Uncini, "Nonlinear system identification using IIR Spline Adaptive Filters," *Elsevier, Signal Processing 108*, pp. 30–35, 2015.
[82] S. P. Harris and E. Ifeachor, “Automating IIR Filter Design by Genetic Algorithm”, *IEE conference on Genetic Algorithm in Engineering: Innovations and Applications* , pp.271-275, Sept. 1995.
[83] D.M.Etter, M.J. Hicks and K.M.Cho , “Recursive Adaptive Filter using an Adaptive Algorithm”, pp.635-638, *Proc. IEEE ICASSP*, 1982.
[84] M.G. Larimore, J.R. Treichler and C.R. Johnson Jr, "SHARF: An algorithm for adapting IIR digital filters," *IEEE Trans. ASSP,* Vol. 28(4), pp.428-440, Aug 1980.
[85] S. D. Stearns , G. R. Elliott and N. Ahmed, "On adaptive recursive filtering", *Proc. 10th Asilomar Conf. Cir., Syst., Comput*., pp. 5-10, 1976.
[86] S. A. White, "An adaptive recursive filter", *Proc. 9th Asilomar Conf. Circuits, Syst., Comput.,* pp. 21-25, 1975.
[87] M. G. Larimore , J. R. Treichler and C. R. Johnson Jr., "Multipath cancellation by adaptive recursive filtering", *Proc. 12th Asilomar Conf. Circuits, Syst., Comput.,* pp. 219-223, 1978.
[88] P. L. Feintuch, "An adaptive recursive LMS filter", *Proc. IEEE,* vol. 64, pp. 1622-1624, 1976.
[89] D. Parikh , N. Ahmed and S. D. Stearns, "An adaptive lattice algorithm for recursive filters", *IEEE Trans. ASSP-28*, pp. 110-111, 1980.
[90] P. A. Thompson, "A constrained recursive adaptive filter for enhancement of narrowband signals in white noise", *Proc. 12th Asilomar Conf. Circuits, Syst., Comput.,* pp. 214-218, 1978.
[91] C. R. Johnson Jr., "A convergence proof for a hyperstable adaptive recursive filter", *IEEE Trans. Inform. Theory*, IT-25, pp. 745-749, 1979.
[92] L.J. Griffiths, “Rapid Measurements of Digital Instantaneous Frequency,” *IEEE Trans. on Trans. ASSP,* Vol-23, pp. 207-222, Apr. 1975.
[93] S.L Marple, Digital Spectral Analysis with Applications, Prent. Hall, 1987.
[94] T. Painter and A. Spanias, “Perceptual Coding of Digital Audio,” *Proc. of the IEEE,* vol. 88, pp. 451-513, Apr. 2000.
[95] A. Spanias, T. Painter, V. Atti, Audio Signal Processing and Coding, Wiley, 2007.
[96] A. Spanias, “Speech Coding: A Tutorial Review,” *Proc. IEEE*, Vol. 82, No. 10, pp. 1441-1582, October 1994.
[97] Jayant, N.,Noll, P. Digital coding of waveforms, Prentice-Hall, NJ,1984.
[98] J.C. Pesquet, G. Tziritas,O. Macchi "Modified LMS algorithms for robust ADPCM," *Proc. ICASSP 1990*, pp.1405-1408, Albuquerque, April 1990.
[99] S. Korse, H. Kluger, M. Pawig, P. Vary, “LPC with backward adaption and Noise Shaping,” Speech Communication, 2014.
[100] ITU Adaptive Differential Pulse Code Modulation (ADPCM), G.726.
[101] W.A. Sethares, The LMS Family, In Adaptive System Identification and Signal Processing Algorithms, N. Kalouptsidis, S. Theodoridis (Eds.), Prentice Hall, 1993.
[102] A. Venetsanopoulos, and K. Plataniotis, “Adaptive filters for color image processing: A survey,” *10th Europ. Sig. Proc. Conf.* pp. 1-4, Sep. 2000.
[103] B. Liu and L. Bruton, “The 2D Complex LMS Algorithm Applied to the 2-D FFT,” *IEEE Trans. CAS II*, Vol. 40, pp. 337-339, May 1993.
[104] A. C. Tan and S. Chen, "Two-Dimensional Adaptive LMS IIR Filter." IEEE ISCAS '93, May 1993.
[105] M. Muneyasu, E. Uemoto and T. Hinamoto. "A Novel 2-D Adaptive Filter Based On The 1-D RLS Algorithm." IEEE ISCAS '97. Jun 1997
[106] Jenkins, K., and R. Faust, "A constrained 2-D adaptive digital filter with reduced computational complexity," *IEEE ISCAS-88*, Finland 1988.
[107] S. Ohnoa, H. Sakaib,"Convergence behavior of the LMS in subband adaptive filtering," Signal Processing, Vol. 81, pp.1053–1059, May 2001.
[108] C. Smith, D. R. Campbell, "The two-dimensional LMS algorithm and its application to sub-band filtering," Fifth International Conference on Image Processing and its Applications, Jul. 1995.
[109] A. Gilloire and M. Vetterli, “Adaptive filtering in subbands with critical sampling: Analysis, experiments and application to acoustic echo cancellation,” *IEEE Trans. Signal Processing*, 40, p. 1862, Aug. 1992.
[110] D. Joachim. J. Deller, "Multiweight optimization in OBE algorithms," *IEEE Trans. Sig. Pro.,* Vol. 54,pp. 678-690, Feb. 2006.
[111] J. R. Deller, M. Nayeri, S.F. Odeh, "Least-square identification with error bounds for real-time signal processing and control,” *Proc. of the IEEE,* Volume: 81(6), pp. 815 - 849, June 1993.
[112] Sizhong Chen and Tong Zhang,"Self-Timed Dynamically Pipelined Adaptive Signal Processing System: A Case Study of DLMS Equalizer for Read Channel," *IEEE Trans. CAS I*, pp. 1338 – 1347, July 2005.
[113] M. Kawafuku et al," Rejection of repeatable runout for HDDs using adaptive filter," *8th IEEE International Workshop on Advanced Motion Control (AMC04),* pp. 305-310, March 2004.
[114] A. Deb, A. Kar, M. Chandra, “Advanced linear adaptive filtering methods for active noise control: A technical survey”, *International Conference on Advanced Comp. and Comm. Systems*, Jan. 2015.
[115] S. Ciochina, C. Paleologu, J/ Benesty, and S. L. Grant, “An optimized proportionate adaptive algorithm for sparse system identification”, *Asilomar Conf. Sign., Sys. and Comp.,* p. 1546, Nov. 2015.
[116] M. Olivari, F. Nieuwenhuizen, H. Bülthoff, and L. Pollini, “Identifying Time-Varying Neuromuscular Response: A Recursive Least-Squares Algorithm with Pseudoinverse”, *IEEE International Conference on Systems, Man, and Cybernetics (SMC),* pp. 3079-3085, Oct. 2015.
[117] Benesty, T Gänsler, and P Eneroth, “Multi-channel sound, acoustic echo cancellation, and multi-channel time-domain adaptive filtering”, *Acoustic signal processing for telecommunication*, vol. 551, pp. 101-120, 2000.
[118] A Gilloire, “Experiments with sub-band acoustic echo cancellers for teleconferencing”, *IEEE Transactions on ASSP*, Apr. 1987.
[119] A Gilloire, M Vetterli, “Adaptive filtering in sub-bands”, *IEEE International Conference on ASSP*, Apr. 1988.
[120] BW Gillespie, HS Malvar, and, D. A. F. Florencio “Speech dereverberation via maximum-kurtosis subband adaptive filtering”, *IEEE ICASSP*, pp. 3701-3704, May 2001.
[121] Mariane R. Petraglia, Rogerio G. Alves, and Paulo S. R. Diniz, “New Structures for Adaptive Filtering in Subbands with Critical Sampling”, *IEEE Trans. on Signal Proc.,* vol. 48, No. 12, pp.3316-3327, Dec. 2000.
[122] E Cetin, I Kale, and R Morling, “On various low-hardware-complexity LMS algorithms for adaptive I/Q correction in quadrature receivers”, *IEEE Proceedings of the 2004 ISCAS),* pp.461-464, May 2004.
[123] JEC Brown, PJ Hurst, and L Der, “A 35 Mb/s mixed-signal decision-feedback equalizer for disk-drive applications”, *IEEE Journal of Solid-State Circuits*, vol. 31, No. 9, pp. 1258-1266, Sep. 1996.
[124] S Ahmadi, A.S Spanias, Cepstrum-based pitch detection using a new statistical V/UV classification algorithm,” *IEEE Transactions on Speech and Audio Processing,* 7 (3), pp. 333-338, 1999.

[125] N Chakravarthy, A Spanias, LD Iasemidis, K Tsakalis, "Autoregressive modeling and feature analysis of DNA sequences, *EURASIP Journal on Applied Signal Processing,* pp. , 13-28, 2004.

[126] A Spanias, V Atti, Interactive online undergraduate laboratories using J-DSP," *IEEE Transactions on Education* 48 (4), pp. 735-749, 2005.

[127] A. Spanias and T. Painter, "An Educational Software Tool for the Study of Speech Coding Algorithms in a DSP Class," Special Issue on DSP Education, *IEEE Trans. on Education*, Vol. 39, pp. 143-152, May 1996.

[128] Markus et al, "Affine projection algorithms for sparse system identification", *IEEE ICASSP-2014,* , pp. 5666-5670, May 2014.

[129] Y,Chen, Y. Gu, and A. Hero III, "Sparse LMS for system identification," *Proc. IEEE ICASSP-2009*, Taiwan, Mar 2009.

[130] Farhang-Boroujeny,"Adaptive Filters:Theory Applications,"Wiley, 2013.

[131] JJ Shynk and S Roy, "The LMS algorithm with momentum updating", *IEEE International Symposium on CAS,* pp. 2651-2654, June 1988.

[132] J. Cioffi, "Limited precision effects in adaptive filtering," *IEEE Trans. CAS-34,* p. 821, July 1987.

[133] S. Douglas, and D. Mandic, "Stability analysis of the least-mean-magnitude-phase algorithm", *IEEE ICASSP*, pp. 4935-4939, March 2016.

[134] Zengli Yang, Yahong Rosa Zheng, and Steven L. Grant, "Proportionate Affine Projection Sign Algorithms for Network Echo Cancellation", *IEEE Transactions on ASLP,* vol. 19, No. 8, pp. 2273-2284, Nov. 2011.

[135] S. Chouvardas, and M. Draief ," A diffusion kernel LMS algorithm for nonlinear adaptive networks", *IEEE ICASSP*, pp. 4164-4168, 2016.

[136] S. Kanna, and D. Mandic, "Steady-State Behavior of General Complex-Valued Diffusion LMS Strategies", *IEEE SPL,* vol.23, p. 722, May 2016.

[137] Jie Chen, Shang Kee Ting, Cédric Richard, , Ali H. Sayed, "Group diffusion LMS", *IEEE ICASSP-2016*, pp. 4925-4929, March 2016.

[138] I. D. Schizas, G. Mateos, and G. B. Giannakis, "Distributed LMS for consensus-based in-network adaptive processing*," IEEE Transactions on Signal Processing*, vol. 57, no. 6, pp. 2365–2382, June 2009

[139] G. Mateos, I. Schizas, and G. Giannakis, "Performance analysis of the consensus-based distributed LMS algorithm*," EURASIP J. Adv. Signal Process*., pp. 1–19, 2009.

[140] Yasukawa, H. and Shimada, S, "Acoustic echo canceler with high speech quality,". *Proceedings IEEE ICASSP'87*, Dallas, pp. 2125–2128, 1987.

[141] Yasukawa, H. and Shimada, S., "An acoustic echo canceler using subband sampling and decorrelation methods," *IEEE Transactions on Signal Processing*, 41 (2), 926–930., 2003.

[142] Petraglia, M, .Mitra, S.,"Adaptive FIR filter structure based on the generalized subband decomposition,"*IEEE Tran. CAS*40, 354–362, 1993.

[143] Morgan, D.R. and Thi, J.C. "A delayless subband adaptive filter architecture," *IEEE Transactions on Sign. Proc.*, 4 (8), 1819–1830, 1995

[144] Widrow, B., Baudrenghien, P., Vetterli, M., and Titchener, P.F., "Fundamental relations between the LMS algorithm and the DFT." *IEEE Transactions on Circuits and Systems,* CAS-34, 814–820, 1987.

[145] Ling, F., Monalakis, D.G., and Proakis, J.G. ,"Numerically robust least-squares lattice ladder algorithms with direct updating of the reflection coefficients," *IEEE Transactions ASSP-34*, 837–845, 1986.

[146] C. G. Lopes and A. H. Sayed, "Diffusion least-mean-squares over adaptive networks: formulation and performance analysis," *IEEE Transactions on SP*, vol. 56, no. 7, pp. 3122 – 3136, Jul. 2008.

[147] F. S. Cattivelli and A. Sayed, "Diffusion LMS Strategies for distributed estimation," *IEEE Trans. Sign. Pro.*, vol. 58, pp. 1035–1048, Mar. 2010.

[148] X. Zhao, S.-Y. Tu, and A. H. Sayed, " Diffusion adaptation over networks under imperfect information exchange and non-stationary data," *IEEE Transactions on SP*, vol. 60, no. 7, pp. 3460 – 3475, Jul. 2012.

[149] A. H. Sayed, " Diffusion Adaptation over Networks, " in *Academic Press Library in Signal Processing*, vol. 3. Academic Press, Elsevier, 2014, pp. 323 – 454, also available as arXiv:1205.4220 [cs.MA].

[150] J. Lee, C. Tepedelenlioglu, M. Banavar, and A. Spanias, "Nonlinear diffusion adaptation with bounded transmission over distributed networks," *IEEE ICC*, pp. 6707 – 6711, June 2015.

[151] W. Liu, J. C. Principe, and S. Haykin, *Kernel adaptive filtering: a comprehensive introduction*, Wiley, 2010.

[152] Stubberud, P., Bruce, J.W.: LMS Algorithm for Training Single Layer Globally Re- cursive Neural Networks. In: *IEEE International Conference on Neural Networks Conf. Proc.*, vol.3, pp. 2214–2217, 1998.

[153] Xu Y, Wong KW, Leung CS., ".Generalized RLS approach to the training of neural networks," "*IEEE Trans Neural Nets,* (1):19-34, 2006

[154] P. Lewis ; J.Hwang , "Recursive least-squares learning algorithms for neural networks ," *Proc. SPIE 1348, Advanced Signal Processing Algorithms, Architectures, and Implementations*, 28 (November 1, 1990.

[155] De Courville, M. and Duhamel, P., "Adaptive filtering in subbands using a weighted criterion." *Proc. IEEE ICASSP, Detroit,*, pp. 985–988., 1995

[156] Kuo, S.M. and Morgan, D.R., Active Noise Control Systems, Algorithms and DSP Implementations, John Wiley & Sons, Inc., New York, 1996.

[157] Kuo, S.M. and Morgan, D.R. (1999) Active noise control: a tutorial review. The Proceedings of IEEE, 87 (6),943 – 973, 1999.

[158] Garayannis, G., Manolakis, D., and Kalouptsidis, N, "A fast sequential algorithm for least squares filtering and prediction." *IEEE Transactions on ASSP*-31, 1394–1402, 1983.

[159] Glendis, G.O. and Kalouptsids, N, " Fast adaptive algorithms for multichannel filtering and system identification,". *IEEE Transactions on Signal Processing*, 40 (10), 2433–2459, 1992.

[160] Cioffi, J., "A fast echo canceller initialization method for the CCITT V.32 modem.," *IEEE Trans. Communications*, 38, 629–638, 1990.

[161] Cioffi, J.M., "The fast adaptive rotors RLS algorithm. *IEEE Transactions on ASSP-38,* 631–651, 1990.

[162] S. Krishna V. Ganguly, I. Panahi, "Real-time Active Noise Control of Multi-tones and MRI Acoustic Noise in fMRI bore with Signal Decomposition and Parallel Hybrid RLS-NLMS Adaptive Algorithms", *10th IEEE Dallas Circuits and Systems Conference(DCAS),* 2014

[163] B. Cornelis, K. Leuven, M. Moonen, J. Wouters, "A QRD-RLS based frequency domain multichannel wiener filter algorithm for noise reduction in hearing Aids," EUSIPCO, pp. 1953 - 1957, 2010.

[164] P. C. Loizou, Speech enhancement: Theory and Practice, CRC , 2007.

[165] T. van den Bogaert, S. Doclo, M. Moonen and J. Wouters, "The effect of multi-microphone noise reduction systems on sound source localization in binaural hearing aids", JASA, vol. 124, no. 1, pp. 484-497, 2008.

[166] G. Enzner, "Analysis and Optimal Control of LMS-Type Adaptive Filtering for Continuous-Azimuth Acquisition of Head Related Impulse Responses", *Proc. of the IEEE ICASSP -2008*, pp. 393-396, 2008.

[167] K I. Nordstrom; G. Tzanetakis; P. Driessen, "Transforming Perceived Vocal Effort and Breathiness Using Adaptive Pre-Emphasis Linear Prediction," *IEEE Trans. ASLP*, Vol 16,: , 6pp. 1087 - 1096, 2008.

[168] J. O. Smith,B. Friedlander, ``Global convergence of the constant modulus algorithm", Proceedings ASSP-85, Tampa, pp. 30, 1985.

[169] M. Souden, J. Benesty and S. Affes, "On optimal frequency-domain multichannel linear filtering for noise reduction", *IEEE Trans. Audio, Speech and Language Processing,* vol. 18, no. 2, pp. 260-276, 2010

[170] H. Krim, M. Viberg, "Two Decades of Array Signal Processing Research" , *IEEE Signal Processing Magazine*, pp. 67-91, July 1996.

[171] M. Brandstein, D. Ward , *Microphone Arrays: Signal Processing Techniques and Applications*, Springer-Verlag, 2001.

[172] Fan, H., W. Jenkins, "An Investigation of Adaptive IIR Echo Canceller: Advantages and Problems," *IEEE Trans. ASSP*, 36, 1819, 1988.

[173] S. Bharitkar and C. Kyriakakis, "Optimization of the bass management filter parameters for multichannel audio applications," 39th Asilomar Conf. on SSC, pp. 324-327, Nov. 2005.

[174] P. Cook, "Reverberation Cancellation in Musical Signals Using Adaptive Filters," Music Dept. Tech. Rep. STAN- M-50, Stanford, 1988.

[175] S. Benton, A. Spanias, K. Tu,H. Thornburg, G. Qian, T. Rikakis, "Adaptive Beamforming and Particle Filtering," Session 520-110,. *SPPRA Conference*, Innsbruck February 2006.

[176] Peters, S.D. and A. Antoniou, A parallel adaptation algorithm for recursive-least-squares adaptive filters in nonstationary environments, *IEEE Transactions Signal Processing*, 43(11): 2484-2495, 1995.

[177] Rao, B.D., Adaptive IIR Filtering using Cascade Structures. *27th Asilomar Conf. on Sign., Syst. and Comp,* pp: 194-198, 1993.

[178] Reddy, V.U., B. Egardt and T. Kailath, 1981. "Optimized Lattice-Form Adaptive Line Enhancer for a Sinusoidal Signal in Broad-band Noise," *IEEE Trans/ on CAS*, 28(6): 542-550., 1981.

[179] Watanabe, E., Y. Okamura and A. Nishihara, "A Realization of Fourth-Order Enhancement using Second-Order Adaptive Line Enhancer," *IEICE Tran/s*, J77-A(11): 1605-1607, 1994.

[180] McCool, J.M., B. Widrow, J.R. Zeidler, R.H. Hearn D.M. Chabries, Adaptive line enhancer. U. S Patent 4: 238-746, 1980.

[181] P. Sristi, W.-S. Lu,, A. Antoniou, "A new variable-step-size LMS algorithm and it's application on subband adaptive filtering for echo cancellation,² IEEE *ISCAS 2001,* p. 721, Sydney, May 2001.

[182] R. Arablouei, K. Doğançay, S. Werner, and Y.-F. Huang, "Adaptive distributed estimation based on recursive least-squares and partial diffusion," *IEEE Trans. Signal Process.*, 62, pp. 3510-3522, Jul. 2014.

[183] Vítor H. Nascimento and Magno T. M. Silva, "Adaptive Filters," Vol 1, Signal Processing Theory and Machine Learning, Chennai: Academic Press, Rama Chellappa, Sergios Theodoridis, editors, Academic Press Library in Signal Processing. p. 619-761, Elsevier, 2014.

[184] B.Friedlander, "Lattice filters for adaptive processing," *IEEE Trans. Signal Process*.70(8), pp.829–867, 1982.

[185] V.J. Mathews, Z. Xie, "Fixed-point error analysis of stochastic gradient adaptive lattice filters, " *IEEE Trans. ASSP*, 1990.

[186] R. North, J.Zeidler, W. Ku, T. Albert, A floating-point arithmetic error analysis of direct and indirect coefficient updating techniques for adaptive lattice filters, *IEEE Trans. Sig. Proc,*. 41, pp. 1809–1823, 1993.

[187] C. Samson, V. Reddy, Fixed point error analysis of the normalized ladder algorithm, IEEE Trans. ASSP 31 (5), pp. 1177–1191, 1983.

[188] C.B.Papadias, and D.T.M.Slock, "Normalized sliding window constant modulus and decision-direct algorithms: a link between blind equalization and classical adaptive filtering, "*IEEE Trans. SP.* 45, pp. 231–235, 1997

[189] S.Netto,P.Diniz,, P.Agathoklis, Adaptive IIR algorithms for system ID: a general framework, *IEEE Tran. Educ*. P 54, 1995.

[190] P. Djurić and M. F. Bugallo, "Particle filtering," Adaptive Signal Processing: Next Generation Solutions, First Edition. By Simon Haykin and Tulay Adali (ed), Wiley-IEEE Press, 2009.

[191] L. Deng, H Attias, .L Lee, and A. Acero, Adaptive Kalman smoothing for tracking vocal tract resonances using a continuous-valued hidden dynamic model," *IEEE Tran. ASLP*, vol. 15(1), pp. 13-23,, Jan. 2007.

[192] S. Miller and A. Spanias, "Algorithms for Quotient Control in Beamforming", *IEEE Antennas and Wireless Propagation Letters* , Volume 6, pp. 651 - 654 , December 2007.

[193] L.L. Scharf,, E.K.P. Chong, M.D. Zoltowski, J.S. Goldstein, I.S. Reed, " Subspace expansion and the equivalence of conjugate direction and multistage Wiener filters, *IEEE Trans. Signal Process*. 56 (10) , 2008

[194] Q. Shen and A.S. Spanias, "A Multichannel Block Adaptive Algorithm For Active Noise Control," 1992 *International Congr. on Noise Control Engineering (Inter-Noise 92)*, pp. 353-356, Toronto, July 20-22, 1992.

[195] An Introduction to Kalman Filtering with MATLAB Examples, N. Kovvali, M. Banavar, A. Spanias Synthesis Lectures on Signal Processing, , Morgan & Claypool Publishers, Editor Jose Mura, vol. 6, no. 2, pp. 1-81, ISBN 13: 9781627051392, September 2013.

[196] D. Florencio; H. Malvar, "Multichannel filtering for optimum noise reduction in microphone arrays, *IEEE Proc. ICASSP,*, pp. 197-200, 2001

[197] I. Tashev and H. Malvar, "A New Beamformer Design Algorithm For Microphone Arrays," *Proc.ICASSP*-2005, 2005.

[198] A. Karbasi and A. Sugiyama, A new DOA estimation method using a circular microphone array, *15th EUSIPCO*, pp. 778-782, 2007.

[199] K. Kumatani et al, "Microphone Array Processing for Distant Speech Recognition: Towards Real-World Deployment," 2015.

[200] Benesty, J. Chen, I. Cohen, Design of Circular Differential Microphone Arrays, Springer 2015.

[201] A. Griffin, D, Pavlidi; M, Puigt; A, Mouchtaris, "Real-time multiple speaker DOA estimation in a circular microphone array based on Matching Pursuit," *Proc. EUSIPCO,* pp. 2303 - 2307, 2012.

[202] J. Thiagarajan, K. Ramamurthy, P. Turaga, A. Spanias, Image Understanding Using Sparse Representations,, Synthesis Lectures on Image, Video, and Multimedia Processing, Morgan & Claypool Publishers, ISBN 978-1627053594, 118 pages, Ed. Al Bovik, April 2014.

[203] Steve Miller, Xue (Sophia) Zhang, Andreas Spanias, Multipath Effects in GPS Receivers, Synthesis Lectures on Communications, Morgan & Claypool Publishers, ISBN 978-1627059312, 70 pages, Ed. William Tranter, Vol. 8, No. 1 , Pages 1-70, December 2015.

[204] S. Sarnin, S. Alam, S. Sulong, N. Ya'acob"Adaptive channel estimation and tracking for indoor MIMO OFDM mobile communication systems,"Proc. ICEESE 2014, pp. 42-47, Dec. 2014

[205] J. Benesty, Y. Huang, T. Gänsler, M. Rupp, "Adaptive Algorithms For MIMO Acoustic Echo Cancellation,", Kluwer, pp 119-147, 2004.

[206] K. Helwani, H. Buchner, S Spors,"Source-domain adaptive filtering for MIMO systems with application to AEC," *Proc. IEEE ICASSP,* 2010.

[207] Gupta, T.; Suppappola, S., Spanias, A.; "Nonlinear acoustic echo control using an accelerometer," *IEEE ICASSP 2009*. 1313-1316, 2009.

[208] Poularikas, A.D. and Z.M. Ramadan, Adaptive filtering primer with MATLAB, Taylor & Francis, 2006.

[209] A. Elia, C. Pattichis, W. Fincham, A. Spanias, and L. Middleton, "Autoregressive Spectral Modeling of Motor Unit Action Potentials: Preliminary Findings*," Proc. IEEE Intern. Conf. of Engineering in Medicine and Biology*, pp. 1465-1466, Paris-France, Oct. 1992.

[210] C. Kotropoulos and I. Pitas, "Constrained adaptive LMS L-filters," *Signal Processing* 26, 335-358 Elsevier, 1992.