

CHAPTER 7. REFERENCES

- [1] B. S. Atal and S. Hanauer, *Speech Analysis and Synthesis by Linear Prediction of Speech Wave*, Journal of the Acoustical Society of America, vol. 50, pp. 637-655, August 1971.
- [2] A. A. (Louis) Beex, *ARMA Covariance Realization from Noisy Data*, ICASSP 86, Tokyo, pp. 2747-2750, 1986.
- [3] A. A. (Louis) Beex and V. E. DeBrunner, *The Effect of Identifier Structure on Parameter Convergence*, IEEE Transactions on Signal Processing, vol. 40, no. 11, pp. 2819-2822, November 1992.
- [4] A. A. (Louis) Beex and G. Zakaria, *Direct Computation of Line Spectral Frequency Using Cascade Recursive Least Squares with Subsection Adaptation*, Proceedings of the 33rd Annual Asilomar Conference on Signals, Systems, and Computers, TA.7-6, Pacific Grove, CA, March 24-27, 1999.
- [5] A. Benveniste and G. Ruget, *A Measure of the Tracking Capability of Recursive Stochastic Algorithms with Constant Gains*, IEEE Transactions on Automatic Control, vol. AC-27, no. 3, pp. 639-649, June 1982.
- [6] K. Berberidis and S. Theodoridis, *A new Fast Block Adaptive Algorithm*, IEEE Transactions on Signal Processing, vol. 47, no. 1, pp. 75-87, January 1999.
- [7] U. Bhaskar, K. Swaminathan, S. Nandkumar, and G. Zakaria, *Quantization of SEW and REW components for 3.6 kbit/s Coding Based on PWI*, IEEE Workshop on Speech Coding, pp. 99-100, Haikko Manor, Porvo , Finland, June 1999.
- [8] G. E. Bottomley and S. T. Alexander, *A Novel Approach for Stabilizing Recursive Least Squares Filters*, IEEE Transactions on Signal Processing, vol. 39, no. 8, pp. 1770-1779, August 1991.
- [9] M. C. Campi, *Exponentially Weighted Least Squares Identification of Time-Varying Systems with White Disturbance*, IEEE Transactions on Signal Processing, vol. 42, no. 11, pp. 2906-2914, November 1994.

- [10] CCITT Series P Recommendation, Telephone Transmission Quality, Recommendation P.81 Modulated Noise Reference Unit (MNRU), Volume V – Rec. P.81, pp. 198-203, Melbourne, Australia, 1988.
- [11] M. M. Chansarkar and U. B. Desai, *A Robust Recursive Least Squares Algorithm*, IEEE Transactions on Signal Processing, vol. 45, no. 7, pp. 1726-1735, July 1997.
- [12] J. Chen, M. J. Melchner, R. V. Cox, and D. O. Bowker, *Real-time Implementation and Performance of a 16 kb/s Low-Delay CELP Speech Coder*, ICASSP '90, pp. 181-184, Albuquerque, New Mexico, 1990.
- [13] P. C. Ching and K.C. Ho, *An Efficient Adaptive LSP Method for Speech Encoding*, IEEE Region 10 Conference on Computer and Communication Systems, Hongkong, September 1990.
- [14] N. I. Cho and S. U. Lee, *On the Adaptive Lattice Notch Filter for the Detection of Sinusoids*, IEEE Transactions on Circuits and Systems-II, vol. 40, no. 7, July 1993.
- [15] J. S. Collura, A. McCree, and T. E. Tremain, *Perceptually Based Distortion Measurements for Spectrum Quantization*, 1995 IEEE Workshop on Speech Coding for Telecommunications, pp. 49-50, September 1995.
- [16] V. E. DeBrunner and A. A. (Louis) Beex, *Sensitivity Analysis of Digital Filter Structures*, in Linear Algebra in Signals, Systems, and Control, B. N. Datta et. al. editors, SIAM, Philadelphia, 1988.
- [17] _____, *Sensitivity of Structures for the Identification of Linear Systems from Impulse Response Data*, ICASSP '89, Glasgow, Scotland, pp. 2214-2217, May 1989.
- [18] _____, *An Informational Approach to the Convergence of Output Error Adaptive IIR Filter Structures*, ICASSP '90, pp. 1261-1264, Albuquerque, New Mexico, 1990.
- [19] J. R. Deller Jr., J. G. Proakis, and J. H. L. Hansen, Discrete-time Processing of Speech Signals, Macmillan, New York, 1993.

- [20] A. Dembo and J. Salz, *On the Least Squares Tap Adjustment Algorithm in Adaptive Digital Echo Cancellers*, IEEE Transactions on Communications, vol. 38, no. 5, May 1990.
- [21] H. K. Dunn, *The Calculation of Vowel Resonances, and an Electrical Vocal Tract, Speech Analysis*, R. W. Schafer and J. D. Markel editors, pp. 31-44, IEEE Press, New York, 1979.
- [22] J. S. Erkelens and P. M. T. Broersen, *On the Statistical Properties of Line Spectrum Pairs*, ICASSP 95, pp. 768-771, 1995.
- [23] E. Eweda, *Comparison of RLS, LMS, and Sign Algorithms for Tracking Randomly Time-Varying Channels*, IEEE Transactions on Signal Processing, vol. 42, no. 11, pp. 2937-2944, November 1994.
- [24] E. Eweda, *Convergence of the RLS and LMS Adaptive Filters*, Transactions on Circuits and Systems, vol. CAS-34, no. 7, pp. 799-803, July 1987.
- [25] H. Eves, Elementary Matrix Theory, New York, Dover Publications, Inc., 1966.
- [26] B. Fisher and N. J. Bershad, *ALE Behavior for Two Sinusoidal Signal Models*, IEEE Transactions on Acoustics, Speech, and Signal Processing, vol. ASP-33, no. 3, pp. 658-665, June 1985.
- [27] A. Gersho, *Advances in Speech and Audio Compression*, Proceedings of the IEEE, vol. 82, no. 6, pp. 900-918, June 1994.
- [28] A. Gersho, S. Wang, and K. Zeger, Vector Quantization Techniques in Speech Coding, Advances in Speech Signal Processing, S. Furui and M. Sondhi editors, pp. 49-84, Marcel Dekker, Inc., New York, 1992.
- [29] G. Glentis, K. Berberidis, and S. Theodoridis, *Efficient Least Squares Adaptive Algorithms for FIR Transversal Filtering*, IEEE Signal Processing Magazine, pp. 13-41, July 1999.
- [30] M. C. Hall and P. M. Hughes, *A Simplified Adaptive IIR Cascade Structure*, International Conference on Digital Processing of Signals in Communications, Publ. No. 62, pp. 33-38, April 1985.

- [31] P. Händel, and A. Nehorai, *Tracking Analysis of an Adaptive Notch Filter with Constrained Poles and Zeros*, IEEE Transactions on Signal Processing, vol. 42, no. 2, pp. 281-291, February 1994.
- [32] S. Haykin, Adaptive Filter Theory, Englewood Cliffs, Prentice Hall, 1991.
- [33] P. Hedelin, P. Knagenhjelm, and M. Skoglund, *Vector Quantization for Speech Transmission*, Speech Coding and Synthesis, W. B. Kleijn and K. K. Paliwal editors, pp. 311-345, Elsevier Science B. V., 1995.
- [34] J. Homer, R. R. Bitmead, and I. Mareels, *Quantifying the Effects of Dimension on the Convergence Rate of the LMS Adaptive Filter Estimator*, IEEE Transactions on Signal Processing, vol. 46, no. 10, pp. 2611-2615, October 1998.
- [35] N. E. Hubing and S. T. Alexander, *Statistical Analysis of Initialization Method for RLS Adaptive Filters*, IEEE Transactions on Signal Processing, vol. 39, no. 8, pp. 1793-1804, August 1991.
- [36] D. R. Hush, N. Ahmed, R. David, and S. D. Stearns, *An Adaptive IIR Structure for Sinusoidal Enhancement, Frequency Estimation, and Detection*, IEEE Transactions on Acoustics, Speech, and Signal Processing, vol. ASSP-34, no. 6, pp. 1380-1389, December 1986.
- [37] F. Itakura, *Line Spectrum Representation of Linear Predictor Coefficients of Speech Signals*, 89th Meeting of the Acoustical Society of America, pp. S35, April 1975.
- [38] F. Itakura and S. Saito, *A Statistical Method for Estimation of Speech Spectral Density and Formant Frequencies*, Speech Analysis, R. W. Schafer and J. D. Markel editors, pp. 295-302, IEEE Press, New York, 1979.
- [39] ITU-T, Coding of Speech at 8 Kbps Using Conjugate-Structure Algebraic-Code-Excited Linear-Predictive (CS-ACELP) Coding, Draft Recommendation G.729, June 1995.
- [40] L. B. Jackson and S. L. Wood, *Linear Prediction in Cascade Form*, IEEE Transactions on ASSP, vol. ASSP-26, pp. 519-528, December 1978.
- [41] B. Juang, D. Y. Wong, and A. H. Gray, Jr., *Distortion Performance of Vector Quantization for LPC Voice Coding*, IEEE Transactions on Acoustics, Speech, and Signal Processing, vol. ASSP-30, no. 2, pp. 294-303, April 1982.

- [42] P. Kabal and R. P. Ramachandran, *The Computation of Line Spectral Frequencies Using Chebyshev Polynomials*, IEEE Transactions on ASSP, vol. ASSP-34, no.6, pp. 1419-1426, December 1986.
- [43] S. M. Kay, Modern Spectral Estimation, Prentice Hall, 1988.
- [44] W. B. Kleijn and J. Haagen, *Waveform Interpolation for Coding and Synthesis*, Speech Coding and Synthesis, W. B. Kleijn and K. K. Paliwal editors, pp. 175-207, Elsevier, 1995.
- [45] R. V. R. Kumar and R. N. Pal, *Tracking of Bandpass Signals Using Center-Frequency Adaptive Filters*, IEEE Transactions on Acoustics, Speech, and Signal Processing, vol. 38, no. 10, pp. 1710-1721, October 1990.
- [46] T. Kwan and K. Martin, *Adaptive Detection and Enhancement of Multiple Sinusoids Using a Cascade IIR Filter*, IEEE Transactions on Circuits and Systems, vol. 36, no. 7, pp. 937-947, July 1989.
- [47] W. P. LeBlanc, B. Battacharya, S. A. Mahmoud, and V. Cuperman, *Efficient Search and Design Procedures for Robust Multi-Stage VQ of LPC Parameters for 4 kb/s Speech Coding*, IEEE Transactions on Speech and Audio Processing, vol.1, no. 4, pp. 373-385, October 1993.
- [48] A. P. Liavas and P. A. Regalia, *On the Numerical Stability and Accuracy of the Conventional Recursive Least Squares Algorithm*, IEEE Transactions on Signal Processing, vol. 47, no. 1, pp. 88-96, January 1999.
- [49] F. Ling and J. G. Proakis, *Nonstationary Learning Characteristics of Least Squares Adaptive Estimation Algorithms*, ICASSP '84, pp. 3.7.1-3.7.4, March 1984.
- [50] K. Mayyas and T. Aboulnasr, *On the Transient Error Surfaces of Output Error IIR Adaptive Filtering*, IEEE Transactions on Signal Processing, vol. 46, no. 3, pp. 766-770, March 1998.
- [51] G. A. Mian and G. Riccardi, *A Localization Property of Line Spectrum Frequency*, IEEE Transactions on Speech and Audio Processing, vol. 2, no. 4, October 1994.

- [52] M. Montazeri and P. Duhamel, *A Set of Algorithms Linking NLMS and Block RLS Algorithms*, IEEE Transactions on Signal Processing, vol. 43, no. 2, pp. 444-453, February 1995.
- [53] G. Moustakides, *Study of the Transient Phase of the Forgetting Factor RLS*, IEEE Transactions on Signal Processing, vol. 45, no. 10, pp. 2468-2476, October 1997.
- [54] G. Moustakides and S. Theodoridis, *Fast Newton Transversal Algorithms- A New Class of Adaptive Estimation Algorithms*, IEEE Transactions on Signal Processing, vol. 39, no. 10, pp. 2184-2193, October 1991.
- [55] M. S. Mueller, *On The Rapid Initial Convergence of Least Squares Equalizer Adjustment Algorithms*, The Bell System Technical Journal, vol. 60, no. 10, pp. 2345-2358, December 1981.
- [56] M. Nayeri and K. Jenkins, *Alternate Realizations to Adaptive IIR Filters and Properties of Their Performance Surfaces*, IEEE Transactions on Circuits and Systems, vol. 36, no. 4, April 1989.
- [57] A. Nehorai and B. Porat, *Adaptive Comb Filtering for Harmonic Signal Enhancement*, IEEE Transactions on Acoustics, Speech, and Signal Processing, vol. ASSP-34, no. 5, pp. 1124-1138, October 1986.
- [58] A. Nehorai, *A Minimal Parameter Adaptive Notch Filter with Constrained Poles and Zeros*, IEEE Transactions on Acoustics, Speech, and Signal Processing, vol. ASSP-33, no. 4, pp. 983-996, August 1985.
- [59] M. Niedzwiecki and L. Guo, *Nonasymptotic Results for Finite-Memory WLS Filters*, IEEE Transactions on Automatic Control, vol. 36, no. 2, pp. 198-206, February 1991.
- [60] M. Niedzwiecki, *First-Order Tracking Properties of Weighted Least Squares Estimators*, IEEE Transactions on Automatic Control, vol. 33, no. 1, pp. 94-98, January 1988.
- [61] K. K. Paliwal and W. B. Kleijn, *Quantization of LPC Parameters*, Speech Coding and Synthesis, W. B. Kleijn and K. K. Paliwal editors, pp. 433-466, Elsevier Science B.V., 1995.

- [62] K. K. Paliwal and B. S. Atal, *Vector Quantization of LPC Parameters in the presence of Channel Errors* Advances in Speech Signal Processing, S. Furui and M. Sondhi editors, pp. 191-201, Marcel Dekker, Inc., NewYork, 1992.
- [63] T. Petillion, A. Gilloire, and S. Theodoridis, *The Fast Newton Transversal Filter: An Efficient Scheme for Acoustic Echo Cancellation in Mobile Radio*, IEEE Transactions on Signal Processing, vol. 42, no. 3, pp. 509-517, March 1994.
- [64] M. R. Petraglia, S. K. Mitra, and J. Szczupak, *Adaptive Sinusoid Detection Using IIR Notch Filters and Multirate Techniques*, IEEE Transactions on Circuits and Systems-II: Analog and Digital Signal Processing, vol. 41, no.11, pp. 709-716, November 1994.
- [65] P. Prandoni and M. Vetterli, *An FIR Cascade Structure for Adaptive Linear Prediction*, IEEE Transactions on Signal Processing, vol. 46, no. 9, pp. 2566-2571, September 1998.
- [66] R. P. Ramachandran et.al., *A Two Codebook Format for Robust Quantization of Line Spectra Frequencies*, IEEE Transactions on Speech and Audio Processing, vol.3, no.3, pp. 157-167, May 1995.
- [67] D. Rao, *Adaptive IIR Filtering Using Cascade Structures*, 1993 International Symposium on Circuits and Systems, pp. 194-198, Chicago, Illinois, 1993.
- [68] D. V. B. Rao and S. Kung, *Adaptive Notch Filtering for the Retrieval of Sinusoidal in Noise*, IEEE Transactions on Acoustics, Speech, and Signal Processing, vol. ASSP-32, no. 4, pp. 791-802, August 1984.
- [69] J. M. T. Romano, M. Belanger, L.C. Coradine, Least Squares Adaptive Filter in Cascade Form for Line Pair Spectrum Modeling, Signal Processing V: Theories and Applications, L.Toores, E. Masgrau, and M. A. Lagunas editors, Elsevier Science Publishers B. V., 1990.
- [70] A. A. Rontogiannis and S. Theodoridis, *New Fast Inverse QR Least Squares Adaptive Algorithms*, IEEE Transactions on Signal Processing, vol. SP-46, pp. 2113-2121, 1998.
- [71] J. Rothweiler, *A Rootfinding Algorithm for LSF*, ICASSP '99, Phoenix, Arizona, 1999.
- [72] M. R. Schroeder and B. S. Atal, *Code-excited Linear Prediction (CELP): High-Quality Speech at Very Low Bit Rates*, Proceedings of ICASSP '85, pp. 25.1.1-25.1.4, 1985.

- [73] Y. Shoham, Cascaded Likelihood Vector Coding of the LPC Information, Proceedings of ICASSP '89, pp. 160-163, May 1989.
- [74] F. K. Soong and B. Juang, *Line Spectrum Pair (LSP) and Speech Data Compression*, ICASSP '84, pp. 1.10.1-1.10.4, March 1984.
- [75] G. W. Stewart, Introduction to Matrix Computation, Academic Press Inc., New York, 1973.
- [76] P. Stoica and A. Nehorai, *Performance Analysis of an Adaptive Notch Filter with Constrained Poles and Zeros*, IEEE Transactions on Acoustics, Speech, and Signal Processing, vol. 36, no. 6, pp. 911-918, June 1988.
- [77] K. Swaminathan et. al, *A Robust Low Rate Voice Coding for Wireless Communications*, IEEE Workshop on Speech Coding, pp. 75-76, September 1997.
- [78] Y. H. Tam, P. C. Ching, and Y. T Chan, *Adaptive Recursive Filters in Cascade Form*, IEE Proc. F on Communications, Radar and Signal Processing, vol. 134, no. 3, pp. 245-52, June 1987.
- [79] S. Theodoridis, G. Moustakides, and K. Berberidis, *A Fast Newton Multichannel Algorithm for Decision Feedback Equalization*, IEEE Transactions on Signal Processing, vol. 43, pp. 327-331, January 1995.
- [80] F. F. Tzeng, *Analysis-by Synthesis linear Predictive Speech Coding at 2.4 kbit/s**, ICASSP '89, pp. 34.4.1-34.4.5, May 1989.
- [81] B. E. Usevitch and W. K. Jenkins, *A Cascade Implementation of a New IIR Adaptive Digital Filter with Global Convergence and Improved Convergence Rates*, ISCAS '89, pp. 2140-2142, 1989.
- [82] L. Welling and H. Ney, *Formant Estimation for Speech Recognition*, IEEE Transactions on Speech and Audio Processing, vol. 6, no.1, pp.36-48, January 1998.
- [83] G. Zakaria, Switching Adaptive Filter Structure for Improved Performance, M. S. Thesis, the Bradley Dept. of Electrical Engineering, Virginia Tech, Blacksburg, Virginia, 1993.

- [84] G. Zakaria and A. A. (Louis) Beex, *Relative Convergence of the Cascade RLS with Subsection Adaptation*, Proceedings of the 33rd Annual Asilomar Conference on Signals, Systems, and Computers, TA.7-6, Pacific Grove, CA, March 24-27, 1999.
- [85] G. Zakaria and A. A. (Louis) Beex, *Cascade Recursive Least Squares with Subsection Adaptation for AR Parameter Estimation*, ICASSP '98, pp. II.953-956, Seattle, Washington, May 1998.