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改進的向量空間可適性濾波器用於聲學回聲消除

Acoustic Echo Cancellation Using an Improved Vector-Space-Based Adaptive Filtering Algorithm

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摘要

在回聲消除系統的應用中，濾波器係數是否能有效的快速更新是相當重要的關鍵，而收斂的效果也是影響回聲是否能消除乾淨的重要因素。因此向量空間可適性濾波器被提出，其結合機械學習向量空間的想法，引進可適性演算法中，達到有效的快速收斂的目標，但其運算複雜度也相對提高。然而為對應到現實生活中應用，運算複雜度將會是較為重要的考量。因此本篇提出改進的向量空間可適性濾波器(Improved Vector-space Adaptive Filter)與改進的向量空間仿射投影符號演算法(Improved Vector-space Affine Projection Sign Algorithm)，藉由重新設計向量空間以及濾波器係數合成的架構，將運算的矩陣維度降低，並運用組合演算法的想法，對仿射投影符號演算法與改進的向量空間仿射投影符號演算法進行組合，達到在任何環境下皆能快速且穩定收斂的目標且比起向量空間可適性濾波器有著更低的運算複雜度和更好的收斂速度與收斂效果，提升在現實生活中應用地可行性。

Abstract

To eliminate acoustic echo, the convergence rate and low residual echo are very important to adaptive echo cancelers. Meanwhile, an affordable computational complexity has to be considered as well. In this paper, we proposed the improved vector space adaptive filter (IVAF)and Improved Vector-space Affine Projection Sign Algorithm (IVAPSA). The proposed can be divided into two phases: offline and online. In the offline phase, IVAF constructs a vector space to incorporate the prior knowledge of adaptive filter coefficients from a wide range of different channel characteristics. Then, in the online phase, the IVAF combines the conventional APSA and IVAPSA algorithms, where IVAPSA computes the filter coefficients based on the vector space obtained in the offline phase. By leveraging the constructed vector space, the proposed IVAF is able to fast converge and achieve a better echo return loss enhancement performance. Moreover, the computational complexity is less than a comparable work.

關鍵詞：回聲消除系統,可適性濾波器,向量空間可適性濾波器,機器學習,組合演算法,仿射投影符號演算法

Keywords: Acoustic echo cancellation, Adaptive Filter, Vector-space Adaptive Filter, Machine Learning, Combined Algorithm, Affine Projection Sign Algorithm.

參考文獻 [References]

[1] P. Ahgren, “Acoustic echo cancellation and doubletalk detection using estimated loudspeaker impulse responses,” *IEEE Trans. Speech Audio Process.*, vol. 13, pp. 1231–1237, 2005.

[2] C. Faller and C. Tournery, “Robust acoustic echo control using a simple echo path model,” in *ICASSP*, 2006, pp. 281–284.

[3] E. Hansler and G. Schmidt, *Topics in Acoustic Echo and Noise Control*. Germany: Springer-Verlag, 2006.

[4] T. S. Wada and B. Juang, “Enhancement of residual echo for robust acoustic echo cancellation,” *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 20, pp. 175–189, 2012.

[5] S. Haykin, *Adaptive Filter Theory*. USA: Prentice-Hall, 2003.

[6] T. S. Wada and B. Juang, "Acoustic echo cancellation based on independent component analysis and integrated residual echo enhancement," in *WASPAA*, 2009, pp. 205 – 208.

[7] T. v. Waterschoot and M. Moonen, "Fifty years of acoustic feedback control: State of the art and future challenges," in *IEEE*, 2011, pp. 288 – 327.

[8] B. Widrow and S. D. Stearns, *Adaptive Signal Processing*. USA: Prentice-Hall, 1985.

[9] Y.-R. Chien and S.-I. Chu, "A fast converging partial update lmsalgorithmwith randomcombining strategy," *Circuits, Syst. Signal Process.*, vol. 33, pp. 1883 – 1898, 2014.

[10] E. Soria, J. Calpe, J. Chambers, M. Martinez, G. Camps, and J. D. M. Guerrero, "A novel approach to introducing adaptive filters based on the lms algorithm and its variants," *IEEE transactions on education*, vol. 47, no. 1, pp. 127 – 133, 2004.

[11] A. Tandon, M. O. Ahmad, and M. Swamy, "An efficient, low-complexity, normalized lms algorithm for echo cancellation," in *Circuits and Systems, 2004. NEWCAS 2004. The 2nd Annual IEEE Northeast Workshop on*. IEEE, 2004, pp. 161 – 164.

[12] A. Feuer and E. Weinstein, "Convergence analysis of lms filters with uncorrelated gaussian data," *IEEE Trans. Acoust. Speech Signal Process.*, vol. 33, pp. 222 – 230, 1985.

[13] Y.-R. Chien and W.-J. Zeng, "Switching-based variable step-size approach for partial update lms algorithms," *Electronics Letters*, vol. 49, pp. 1801 – 1803, 2013.

[14] H. C. Huang and J. Lee, "A new variable step-size nlms algorithm and its performance analysis," *IEEE Transactions on Signal Processing*, vol. 60, no. 4, pp. 2055 – 2060, April 2012.

[15] L. Liao and A. W. Khong, "Sparseness-controlled affine projection algorithm for echo cancelation," in *APSIPA*, 2010, pp. 355 – 361.

[16] H.-C. Shin, A. H. Sayed, and W.-J. Song, "Variable step-size nlms and affine

projection algorithms," *IEEE signal processing letters*, vol. 11, no. 2, pp. 132–135, 2004.

[17] K. Y. Hwang and W. J. Song, "An affine projection adaptive filtering algorithmwith selective regressors," *IEEE Trans. Circuits Syst. II: Exp. Briefs*, vol. 54, pp. 43–46, 2007.

[18] J. M. Gil-Cacho, T. van Waterschoot, M. Moonen, and S. H. Jensen, "Nonlinear acoustic echo cancellation based on a parallel-cascade kernel affine projection algorithm," in *Acoustics, Speech and Signal Processing (ICASSP), 2012 IEEE International Conference on*. IEEE, 2012, pp. 33–36.

[19] T. Shao, Y. R. Zheng, and J. Benesty, "An affine projection sign algorithm robust against impulsive interferences," *IEEE Signal Processing Letters*, vol. 17, no. 4, pp. 327–330, April 2010.

[20] J. Yoo, J. Shin, and P. Park, "Variable step-size affine projection sign algorithm," *IEEE Transactions on Circuits and Systems II: Express Briefs*, vol. 61, no. 4, pp. 274–278, April 2014.

[21] J. Shin, J. Yoo, and P. Park, "Variable step-size affine projection sign algorithm," *Electronics Letters*, vol. 48, no. 9, pp. 483–485, April 2012.

[22] Y. Tsao, S. H. Fang, and Y. Shiao, "Acoustic echo cancellation using a vector-space-based adaptive filtering algorithm," *IEEE Signal Processing Letters*, vol. 22, pp. 351–355, March 2015.

[23] R. Kuhn, J.-C. Junqua, P. Nguyen, and N. Niedzielski, "Rapid speaker adaptation in eigenvoice space," *IEEE Transactions on Speech and Audio Processing*, vol. 8, no. 6, pp. 695–707, 2000.

[24] Y. Tsao and C.-H. Lee, "An ensemble speaker and speaking environment modeling approach to robust speech recognition," *IEEE Trans. Audio, Speech, Lang. Process.*, vol. 17, pp. 1025–1037, 2009.

[25] P. N. Belhumeur, J. P. Hespanha, and D. J. Kriegman, "Eigenfaces vs. fisherfaces:

Recognition using class specific linear projection,” *IEEE Transactions on pattern analysis and machine intelligence*, vol. 19, no. 7, pp. 711–720, 1997.

[26] J. Li-You, “A study of convex combined adaptive filtering algorithms,” 2015.

[27] M. Rages and K. C. Ho, “Limits on echo return loss enhancement on a voice coded speech signal,” in *The 2002 45th Midwest Symposium on Circuits and Systems*, vol. 2, Aug 2002, pp. 152–155.

[28] S. Sukhumalwong and C. Benjangkaprasert, “Adaptive echo cancellation using variable step-size algorithm lattice filters,” in *TENCON 2006 - 2006 IEEE Region 10 Conference*, Nov 2006, pp. 1–4.

[29] E. A. Habets, “Room impulse response generator techn. univ. eindhoven,” Techn. Univ. Eindhoven, Tech. Rep., 2006.

[30] H.-G. Hirsch and D. Pearce, “The aurora experimental framework for the performance evaluation of speech recognition systems under noisy conditions,” *ASR-2000.*, pp. 181–188, May 2000.

[31] D. Macho, L. Mauuary, B. Noé, Y. M. Cheng, D. Ealey, D. Jouvet, H. Kelleher, D. Pearce, and F. Saadoun, “Evaluation of a noise-robust dsr front-end on aurora databases.” in *INTERSPEECH*, 2002, pp. 17–20.

[32] N. Parihar, J. Picone, D. Pearce, and H. G. Hirsch, “Performance analysis of the aurora large vocabulary baseline system,” in *Signal Processing Conference, 2004 12th European*, 2004, pp. 553–556.