

# A Study of Noise Compensation Methods for Speech Recognition under Noisy Environments

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## ABSTRACT

When a speech recognition system in quiet environment is moved to a noisy environment, the recognition rate drops drastically. The compensation of noise effect becomes an important task for noisy speech recognition. In this study, we investigate the behavior of speech cepstral vector due to additive noise. We find that the cepstral vector deviates as the level of additive noise increases. In the case of white noise, the direction of cepstral vector deviation is approximately opposite to the direction of the cepstral vector of the clean speech. As power level of the white noise increases, the cepstral vector of the noisy speech will converge to the zero vector. However, for other types of noise, the change of cepstral vector is approximately at the direction of the difference vector of the noise cepstral vector and clean speech cepstral vector. Base on this behavior, we include a feature deviation vector into the reference model to compensate for the noise effect. The deviation vector is calculated according to the difference value of the cepstral vector of a few noisy speech and the corresponding model state cepstral mean vector. During the pattern matching phase, an optimally scaled deviation vector is added to the state mean vector of the clean speech model so that the clean speech model is adapted to the noisy environment. Experimental results show that the proposed method is effective for white noise and color noises.

Keywords : speech recognition ; environment adaptation ; additive noise ; feature deviation vector

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## REFERENCES

- [1]李立民, 雜訊環境下語音辨認之研究, 國立清華大學電機工程研究所博士論文, 1995。
- [2]謝秀琴, 數位語音訊號基本原理, 全華科技, 台北, 1996。
- [3]Guan, C., Chen, Y. and Wu, B., " Direct modification on LPC coefficients with application to speech enhancement and improving the performance of speech recognition in noise, " IEEE int. conf. Acoust., Speech Signal Processing, Vol. , pp.107-110, 1993.
- [4]Gales, M.J.F. and Young, S., " An improved approach to the hidden Markov model decomposition of speech and noise, " IEEE Int. Conf. Acoust., Speech, Signal Processing., pp.1-233-236, 1992.
- [5]Gales, M.J.F. and Young, S.J., " Cepstral parameter compensation for HMM recognition in noise, " Speech Communication, Vol.12, No.3,

pp. 231-239, 1993.

- [6]Hwang, T.-H., Lee, L.-M. and Wang, H.-C., " Feature adaptation using deviation vector for robust speech recognition in noisy environment, " Proceedings of IEEE international conference on Acoust., Speech Signal Processing, pp.1227-1230, 1997.
- [7]Hwang, T.-H., Lee, L.-M. and Wang, H.-C., " Cepstral behaviors due to additive noise and a compensation scheme for noisy speech recognition, " IEE Proc.-Vis. Image Signal Process., Vol.145, No.5, pp.316-321, 1998.
- [8]Juang, B.H., Levinson, S.E. and Sondhi, M.M., " Maximum likelihood estimation for multivariate mixture observations of Markov chains, " IEEE Trans. Inform., vol. IT-32, No.2, pp. 307-309, 1986.
- [9]Lee, L.-M., Chen, J.-K. and Wang, H.-C., " Nonlinear cepstral equalization method for noisy speech recognition, " IEE Proc.-Vis. Image Signal Process., Vol.141, No.6, pp.397-402, 1994.
- [10]Lee, L.-M. and Wang, H.-C., " An extended Levinson-Durbin algorithm for the analysis of noisy autoregressive process, " IEEE Signal Process. Letters, Vol.3, No.1, pp.13-15, 1996.
- [11]Lee, C.-H. and Huo Q., " On adaptive Decision Rules and Decision Parameter Adaptation for Automatic Speech Recognition, " Proceedings of IEEE, Vol.88, No.8, pp.1241-1269, August, 2000.
- [12]Mansour, D. and Juang, B.H., " A family of distortion measures based upon projection operation for robust speech recognition, " IEEE Trans. Acoust. Speech Signal Process., No. 37, pp. 1659-1671, 1989.
- [13]Moreno, P.J., Raj, B., and Stern , R.M., " A vector Taylor series approach for environment-independent speech recognition, " Proceedings of IEEE international conference on Acoust, Speech Signal Processing , pp.733-736, 1996.
- [14]Nadas, A. , Nahamoo, D. and Picheny, M.A., " Speech recognition using noise adaptive prototypes, " IEEE Trans. Acoust., Speech, Signal Processing, Vol-37, No.10, pp.1495-1503, 1989.
- [15]Quatieri, T.F., Discrete-Time Speech Signal Processing, Prentice Hall, New Jersey, 2002.
- [16]Rabiner, L.R., " A Tutorial on Hidden Markov Models and Selected Applications in Speech Recognition, " Proceedings of IEEE , Vol.77, No.2, pp.257-286, 1989.
- [17]Rabiner, L.R., Lee, C.H., Juang, B.H., Wilpon, J.G., " HMM clustering for connected word recognition, " IEEE Int. Conf. Acoust., Speech Signal Processing, Vol.1, pp.405-408, 1989.
- [18]Rabiner, L.R. and Juang, B.H., Fundamentals of Speech Recognition, Prentice Hall, New Jersey, 1993.
- [19]Varga, A.P. and Moore, R.K. , " Hidden Markov model decomposition of speech and noise, " IEEE Int. Conf. Acoust., Speech, Signal Processing, pp.845-848, 1991.