Research Progress Report

Junfeng Li

Graduate School of Information Science

Japan Advanced Institute of Science and Technology

1-1 Asahidai, Tatsunokuchi, Nomigun, Ishikawa, 923-1292, Japan

Email: junfeng@jaist.ac.jp URL: www.jaist.ac.jp/∼junfeng

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1 Research purpose

Recently, noise reduction algorithms are in great demand for the increased speech applications, eg., speech recognizer and celluar telephony. In comparison of the single-channel algorithms, the multi-channel algorithms has reflected great superiority in reducing noise and enhancing speech, due to its spatial filtering of suppressing the interfering signals arriving from directions other than the specified look-direction.

In this research, we focus on constructing a multi-channel noise reduction system which is used as a front-end processor for speech recognizer in adverse environments, eg., vehicular condition. Due to the practical requirements in car condition, the suggested algorithm should fulfil the following requirements: (a) high ability in dealing with various kinds of noises in practical environments; (b) small physical array size; (c) low computational cost (low energy cost); (d) real-time processing. Considering the above requirements, the purpose of our research is to design and develop a high-performance noise reduction system in various kinds of noise conditions.

2 Research results in 2004

In this research, we proposed a novel noise reduction system, using a hybrid noise estimation technique and post-filtering, to suppress both localized noises and non-localized noise simultaneously in arbitrary noise environments. To estimate localized noises, we presented a hybrid noise estimation technique which combines a multi-channel estimation approach we previously proposed and a soft-decision single-channel estimation approach.

Final estimation accuracy for localized noises was significantly improved by incorporating a *robust and accurate speech absence probability* (RA-SAP) estimator, which considers the strong correlation of SAPs between adjacent frequency bins and consecutive frames and makes full use of the high estimation accuracy of the multi-channel approach. The estimated spectra of localized noises were reduced from those of noisy observations by spectral subtraction. Non-localized noise was further reduced by a multi-channel post-filter which was based on the *optimally-modified log-spectral amplitude* (OM-LSA) estimator. With the assumption of a diffuse noise field, we proposed an estimator for the *a priori* SAP based on the coherence characteristic of the noise field at spectral subtraction output, high coherence at low frequencies and low coherence at high frequencies, improving the spectral enhancement of the desired speech signal. Experimental results demonstrated the effectiveness and superiorities of the proposed noise estimation/reduction methods in terms of objective and subjective measures in various noise conditions.

3 Publication

1. Junfeng Li, Xugang Lu and Masato Akagi, "A Noise Reduction System in Arbitrary Noise Environments and Its Applications to Speech Enhancement and Speech Recognition", In ICASSP2005 - IEEE Conference on Acoustics, Speech, and Signal Processing, Philadelphia, USA, Mar. 2005.

2. Junfeng Li and Masato Akagi, "Suppressing Localized and Non-Localized Noises in Arbitrary Noise Environments", In 2005 Joint Workshop on Hands-Free Speech Communication and Microphone Arrays, New Jersey, USA, Mar. 2005.

3. Junfeng Li and Masato Akagi, "Noise Reduction Using Hybrid Noise Estimation Techniques and Post-Filtering", In ICSLP 2004 - the 8th International Conference on Spoken Language Processing, pp. IV 2705-2708, Jeju Island, Korea, Oct. 2004.