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Noise reduction using paired-microphones

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Abstract

There is a big demand for noise reduction in proportion to make speech processing techniques practicable. The remarkable instances are automatic speech recognizer(ASR), hearing aids, and so on. In more and more aging society, these instruments for the social welfare will be important and indispensable.

Study on noise reduction has a long history, and a great variety of methods have been proposed. However, almost all methods can not reduce non-stationary noises. On the other hand, a method proposed in this thesis can reduce every signal added to an objective signal in the time domain. This method can reduce non-stationary noises that former methods are hard to deal with, for example a signal that generates during the section an objective signal exists and correlates closely with it.

In this thesis, the author proposes an algorithm of noise reduction based on the concept 'paired-microphones' to reduce non-stationary noises. This method combines the spatial filtering that extracts the signal comes from a specific direction and the frequency filtering that is the Spectral Subtraction method. Two directions of an objective signal and the most dominant noise signal is estimated first, and spatial filters are analytically designed using them. The recent methods using spatial filters adopt adaptive beamforming techniques because they can not estimate signal directions in noisy environments, but this method does not have to use adaptive filters owing to develop a very precise direction finder. Then, an objective signal is obtained by subtracting estimated noise component from the received signal in each short term frame. This method can reduce sudden noises that adaptive filters can not follow, for example a shock sound when doors open or shut, because it does not need any constraints for noise signals.

Experimental results show that proposed method is superior to every other method as to reduce non-stationary noises vary from moment to moment. On another evaluation supposing a front-end of ASRs or hearing aids, it is confirmed that proposed method can reduce non-stationary noises considerably on each purpose. Therefore, this noise reduction algorithm will be effective as a front-end of ASRs and instruments in need of decreasing the distortions on auditory impressions.

Key Words: **Noise reduction, Non-stationary noises, Paired-microphone, Subtractive beamformer, Analytical beamforming**

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