single-channel noise reduction method of speech by noise subtraction and blind source separation.

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In this thesis, a time-domain noise reduction system using single-microphone two-stage method to improve SNR of noise-contaminated speech is proposed. Initially, we reduce noise level that gives the first noise-reduced speech using time-domain noise subtraction (NS) method which is equivalent to spectral subtraction method. For NS, noise is estimated based on the harmonic properties of noisy speech in a frame basis without voice activity detection. The thesis introduces the Degree of Noise (DON) estimation process to tune the noise which results better noise estimation accuracy.

In the next stage, we implement the blind source separation (BSS) technique modeled by FIR filters to further reduce noise and noise residuals in the first noise reduced speech. We find that the BSS (alone) shows poor performance but by controlling noise using the NS, the BSS improves the performance of this stage. The noise dominant signal for BSS is estimated by taking the first noise characteristics into consideration.

In the next phase, further improvement of the algorithms is presented using the cascade of weighted NS and BSS (WNS+BSS). The weight for WNS is calculated from the square of the complement of the DON. We propose iteration number of the LMS algorithms that makes the system efficient and computationally inexpensive based on DON. The method is tested with five types of noise. For speech degraded by white noise, noise reduction is of about 11.3dB at 0dB and 5.7dB at 10dB input SNR, whereas for crowd noise, it is about 8.1dB at 0dB and 4.3dB at 10dB input SNR.