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An improvement of speech intelligibility using small-scale microphone array

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There are many noises in real environments. For example sound pressure level is 74 dB in a crowd of people, 82 dB in a train and at subway station. The Environment Agency reported that it is desirable that sound pressure level are less 45 dB indoors and 55 dB outdoors for people to talk articulately 1 m away. One of the approarch to improve an intelligibility is a noise reduction method.

Boll proposed spectral subtraction method with single microphone. Though processing of this algorithm is very simple, it assume noises to be stationary. However, there are a lot of unstationary noises in real environments. Recently, many noise reduction method with microphone array are proposed. These method can use arrival directions for making beamformer. However, delay and sum microphone array need large-scale and huge-number of microphones for sharping beam widths. Mizumachi and Kago et. al. proposed a noise reduction method for enhancing the target signal by subtracting estimated noise from a noisy signal using adaptive beamformer with small number of microphones. This method is robust for not only stationary noises but also unstationary, and multi-noises. The aim of this paper is improvement of intelligibility for noisy signals using noise reduction method with small-scale microphone array proposed by Kago et. al. However, this method is not proposed for hearing aids but automatic speech recognizer. Therefore, considerations for hearing aids is not carried out. Thus, this paper research problems of noise spectrum estimation at method proposed by Kago et. al. and improvement of intelligibility of speech signal.

First formant of Japanese vowels uttered male exist at below frequency of 1 kHz. It is important for consonant intelligibility to reduce noises at frequencies of 5∼6 kHz. However, the noise reduction method proposed Kago et. al. can not estimate noise spectrum at lower frequencies of 1 kHz. Because noises remain at lower frequencies on noise reduced speech signal, vowel intelligibility become lower. A complement function of beamformer do not perform well at frequencies 5∼6 kHz. One of the reasons to decrease consonant intelligibility is that noises remain at frequencies of 5 ∼6 kHz. In this paper, $\delta/2$ instead of delay operater τ is substituted into equation about $g_{cr}(t)$. Thus, the estimation accuracy of noise spectrum is improved by decreasing mismatch between estimated arrival directions and beamformer $g_{cr}(t)$. An amplitude of beamformer $g_{lr}(t)$ is also compensated with threshold, so an approximation accuracy of noise spectrum at lower frequency is modefied. An error at noise spectral estimation become lower and thresholds ε_1 and ε_2 are determined in order to perform the complement function of beamformer. As a result of improvement of noise spectrum estimation, the estimation of noise spectral become more accurate.

Computational simulation results show effectiveness for improvement of estimation accuracy and intelligibility using proposed method. There are improvement of noise reduction performance at stationary, unstationary, and multi-speech noises. We verificate effectiveness using word intelligibility experiments. Experimental results show that score of proposed method are higher than Kago et. al. one at high word intelligibility and low word intelligibility.