Abstract

Noise is a most problem of the applications that related to audio. There are a lot of incoming signals at microphones at input audio process, generally contaminated by noise. As a result, the microphone signal should be cleaned with digital signal processing tools before it is stored, analyzed, transmitted, or played. So it is considered necessary to perform signal processing incoming sound to reduce the existing noise.

One of noise reduction method developed at this time is ANC (Active Noise Control). ANC works by means of adaptive filters to reduce noise. In principle, the ANC reduce low frequency noise and create a quiet zone with the aim of improving the quality of the sound.

Based on previous studies Marco Jennifer Patrick [12] FXLMS is better than LMS algorithm at noise reduction on appeal because the works as a stochastic gradient-based algorithms using gradient vector of the weight of the filter tap weights converge to an optimal solution by passing the maximum step calculation problem fixed the iteration.

In this thesis would implemented FXLMS algorithm (Filtered-X LMS) where the data test is a combination between clean speech and background noise. Result of this thesis is increase the SNR and PSNR which is result of ANC filter using algorithms FXLMS and LMS.

Keyword: *noise reduction, filtered-X Least Mean Square. LMS*