

ABSTRACT

Acoustic echo is kind of noise that created by the reflection of sound waves by the wall space and other things that exist in the room. In hands-free telephony system teleconference system, and the acoustic echo signal can be noise or interference and degrade the quality of the sound when the signal interfere the ongoing conversation. For that need a specific method to solve this problem.

Acoustic Echo Cancellation (AEC) is a method that can be implemented to cancel this unwanted echo. AEC was built using adaptive filters, and the echo is modeled as a result of the convolution of the input signal and room impulse response. With the adaptive NLMS algorithm, adaptive filter coefficients will be adapted in order to get the Mean Square Error (MSE) minimum. In this thesis the AEC system will be simulated (using Matlab) to obtain the best parameters and implement the system on the TMS320C6455 DSK (using Code Composer Studio)

From the results of testing and analysis, Best value of filter tap is 2048, where the use of filter tap gives the best performance parameters of the system, whereas if we count the computation time as a performance parameter, then the optimum filter tap is 32. The optimum initial step size value is 0.4. By using 2048 filter tap and 0.4 initial step size, AEC simulation process produces the average performance which 88,75729236 % Echo Cancelled Rate, 7,859466763 dB ERLE, and 0,010740763 MSE, while the implementation of TMS 320C6455 DSK get the average 50,13333 % of echo cancelled rate

Key words: acoustic echo, adaptive filters, NLMS, MSE, ERLE.