SIMULATION OF LMS ALGORITHM FOR ACTIVE NOISE CANCELLATION

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ABSTRACT

Active noise control is used to reduce noise. An active noise control is designed to minimize this noise signal. An active noise control is used adaptive filter to reduce noise. Adaptive filter is a digital filter combine with adaptive algorithm. For this project, the LMS (least mean square) is used to adjust the coefficient of the digital filter in order to minimize the difference between the desired output d(k) and its filter output y(k). MATLAB simulation will be used to simulate the adaptive filter to provide silent noise waveform. Then, Xilinx ISE will be used to design the adaptive filter by using Verilog programming. The external sources are the mp3 music data as external source.

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CHAPTER 1

INTRODUCTION

1.1 INTRODUCTION

This chapter explains the overview of ANC system. Also, there are sections consist of background, problem statements, objectives and scope of project. In addition, the thesis outlines is also included.

1.2 Background

Active noise control (ANC) is a method for reducing noise signal [1]. ANC is used Least Mean Squares (LMS) adaptive filter to reduce the noise. LMS adaptive filter is a digital filter combined with an LMS adaptive algorithm, which is used to modify the coefficients of the filter in order to minimize a function of the difference between the desired output *d(k)* and its filter output *y(k)* [2, 3].LMS adaptive filters have been used in a wide range of digital signal processing (DSP) application because of its easy to compute and implement [4]. Fig. 1.1 shows a block diagram of ANC system, where *k* is the number of samples [2, 5]. The reference signal *x(k)* is a signal that same signal with the noise signal n(k). The block "filter" represents the digital filter used to processed to produce an digital filter output *y(k),* from *x(k)* . This filter will equalize output filter *y(k)* to desired input *d(k)*. The desired input *d(k)* is denoted as the input sound signal *s(k)* with the unwanted noise environment *n(k)* [5]. The process of equalization will take into account when at each iteration of the error signal *e(k)* is feedback into the LMS algorithm filter, where the filter characteristics are altered by adjusting the FIR coefficient *w(k)* accordingly [3].