ANALYSIS OF LMS AND NLMS NOISE CANCELLATION OF SPEECH SIGNAL USING MATLAB SIMULINK

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In the name of Allah, the Most Gracious and the Most Merciful.

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ABSTRACT

The LMS adaptive filter has several parameters which can affect their performance. From among these parameters, most papers handle the step size parameter for controlling the performance. This paper presents a study of LMS and NLMS adaptive filter for noise cancellation on speech signal. This analysis is based on the results from simulation of MATLAB Simulink and their outputs from both models are compared in order to investigate the suitable type of adaptive filter algorithm. It is found that the NLMS is a variant of the LMS algorithm which is more robust when implemented in finite-precision hardware and well understood convergence behaviour compare to the other adaptive algorithm.

Keywords-LMS algorithm; NLMS algorithm; step size; MATLAB Simulink;

TABLE OF CONTENTS

CHAPTER 1

INTRODUCTION

1.1. INTRODUCTION

Signal noise cancellation is a system or technique to suppress the noise that interferes in the telecommunication system. The interference noise is filtered from the signal through the filtering process and always come out the clear signal wanted at the end. Signal can be define as anytime varying physical phenomenon that can convey data information or act as channel or medium to transfer the information through it. The some examples of signal are human voice or speech, electrocardiogram, sign language, image and video. While noise are also known as unwanted signal that will interfere and disturb the wanted signal or information to reach the expected output.

Speech is a very basic way for humans to convey information from one another to one another with some kind of frequency bandwidth. The speech can convey information with the emotion of a human voice while sound is an essential form of human communication. However, unwanted sounds, or noise, can degrade the quality of speech signal due to this the communication process will be affected.

Signal noise cancellation is more related to adaptive filter which also known as a computational device that attempts to model the relationship between two signals in real time. The four applications of the adaptive filter classified are adaptive identification, adaptive inverse, adaptive prediction, and active noise cancellation due to the architecture of the adaptive filter algorithm implementation [1].

The major part of designing the noise cancellation is adaptive filter which can be analogical designs, digital design or mixed design. The digital filter gives an answer of greater precision and overcome the offset problem of analogical design. Adaptive filtering process consists of two major steps which are filtering process and adaptation process. Filtering process produces an output signal (response) from the input signal, and adaptation process will adjusts the coefficients of the filter in a way in order to