Noise Cancellation Using Least Mean Squares Adaptive Filter

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Introduction

There are many areas where adaptive filtering is useful, including echocardiograms, acoustic echo cancellation, feedback suppression, noise cancellation, signal prediction, and many others. This project looks at the area of adaptive filtering used for noise cancellation.

Overview

In a basic sense, adaptive filtering is the creation and use of a filter, the parameters of which are changed based on an optimization algorithm.

The least mean squares method (or LMS) is one of the most widely used adaptive filter algorithms, and also is claimed to be the most computationally simple. This is the method analyzed in this project.

LMS Algorithm

Least Mean Squares (LMS) is one of the most common algorithms used to implement adaptive filtering. LMS changes its filter coefficients based on the desired signal by finding the least mean square of the error signal. (where error signal is estimated by taking the difference between the desired signal and the actual signal) The “desired signal” reference can be a form of the desired input (without noise), or a delayed version of the noisy input. In the latter case, the algorithm tries to find the common noise in both the signal and the filter output, and estimates the error between the two. Figure 1 shows two stages, in practice a higher filter order can be used for better results.

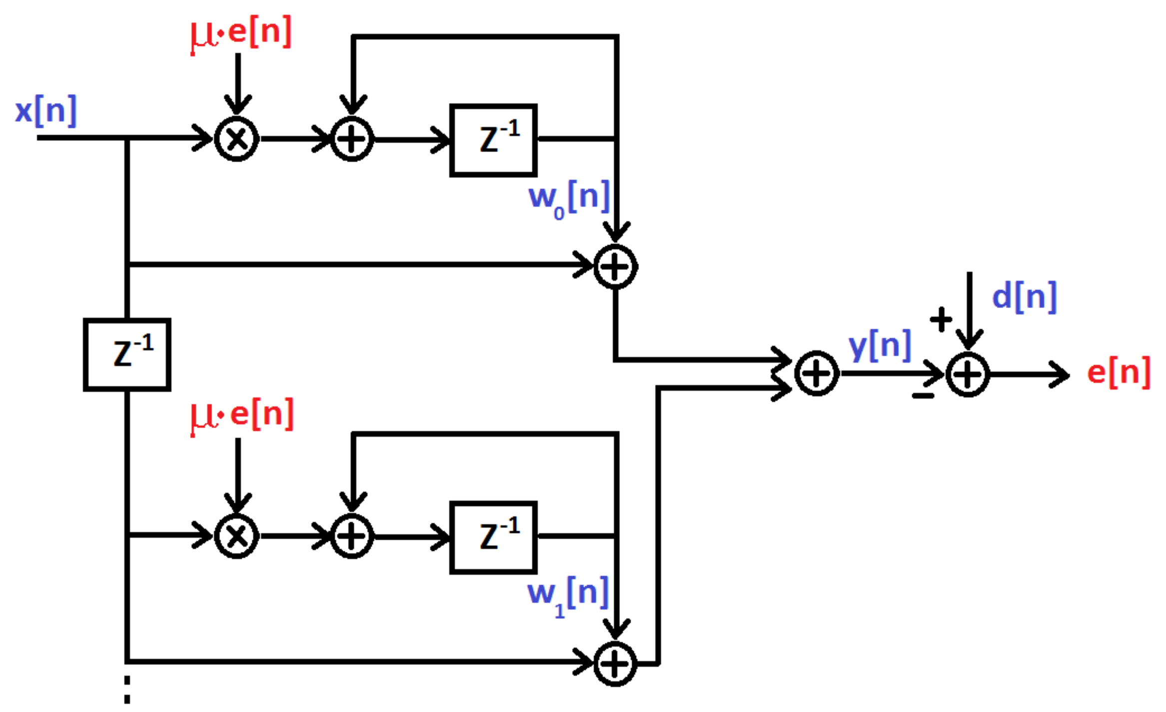

Figure 1: Simple block diagram of a second order LMS filter

Objectives

It was desired to explore the area of adaptive filtering with relation to audio signal processing, specifically the topic of active noise cancellation. As described, the method of choice was the LMS adaptive filter approach.

Implementation

A simulation of the LMS algorithm was done in MATLAB in order to become familiar with the algorithm parameters and operations. With this understanding, a real-time implementation was created using the OMAP-L138 board to apply the LMS filter, and the DSK 6713 was used as a microphone preamplifier/processor. The input to the filter (***x[n]***) was an audio signal and the room noise picked up by the microphone. The desired signal (***d[n]***) was the pure music signal (with no noise). The LMS filter implementation on the OMAP-L138 was order 65.

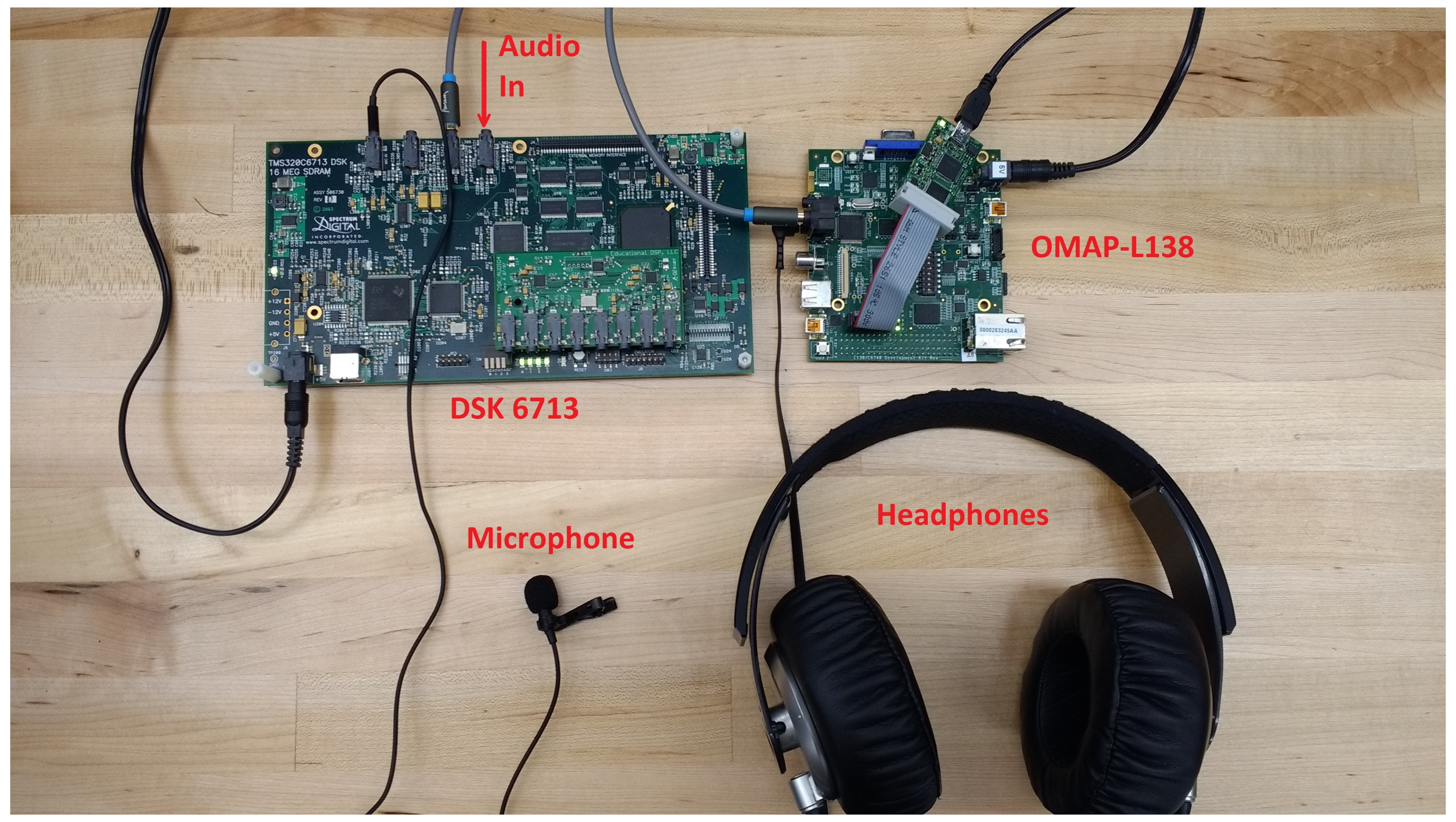

Figure 2: Diagram of our implementation

Results

In figure 3, the first plot shows the noisy input signal. The second plot is the filtered output, showing the desired sine signal isolated from the noise. The final plot is the error signal, showing how over time the filter removes much of the noise, without removing the desired sine wave.

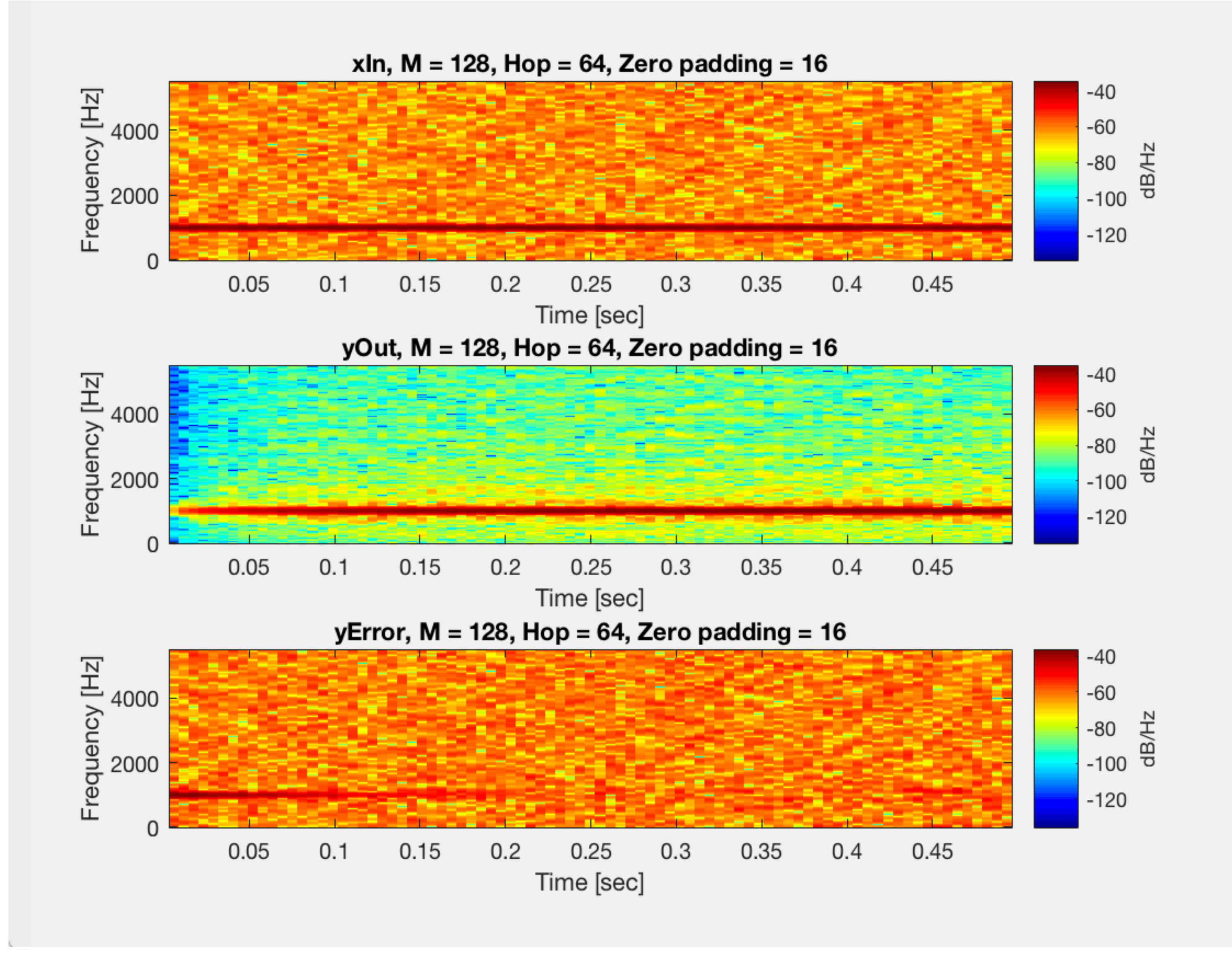

Figure 3: The results of the MATLAB simulation

The real-time implementation of the LMS adaptive filter was able to remove, with acceptable success, much of the noise added to the input signal. When implementing a microphone for measuring room noise, the system was able to remove some of the constant, wideband noise elements interfering with the music played through the headphones.

Future Work

The system implemented here is best suited for constant, wide bandwidth noise (that has little to no abrupt changes), such as noise from a plane or car engine, noise from air vents, or any constant “room noise.” The limitations come when there is some narrow band noise, such as people talking, sharp sounds like clapping, and other narrow band non-periodic sounds. While this system was able to demonstrate noise cancellation for constant noise, a more practical design would include a wider range of capabilities. This can be achieved by implementing other adaptive filter methods such as NLMS, RLS, LPC, etc.

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