Active Noise Cancellation System

Functional Description and Complete System Block Diagram

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Introduction

Traditional digital filters require a certain amount of information about the signal being filtered in order to be designed to function properly. In situations where this information cannot be obtained, or when the signal's characteristics change over time, traditional digital filters cannot be optimally implemented. To overcome these obstacles, adaptive filters are implemented that will converge to the optimum filter for the signal. This is done by recursively changing the filter coefficients in order to achieve the desired signal for the circumstance.

There are four basic implementation architectures for adaptive filters. They are interference cancellation, system identification, prediction, and inverse system identification. Active noise cancellation utilizes the interference cancellation.

Goal

The goal of the project is to design, simulate, and implement an adaptive digital filter to perform active noise cancellation.

- 1. Learn the basics of adaptive digital filters
- 2. Simulate an adaptive digital filter using MATLAB
- 3. Design an adaptive digital filter using the Xilinx block set in simulink
- 4. Perform hardware co-simulation of the designed filter on a DSP/FPGA board

High Level Block Diagram

The following is a high-level block diagram of an interference cancelling adaptive filter.



Figure 1 Basic Active Noise Cancellation System

The reference signal contains interference similar to the interference signal. The interference is then input into the adaptive filter. The resulting output is then subtracted from the reference signal. The error signal is used to compute the filter coefficients to minimize the power of the error signal. By minimizing the power of the error signal, the signal to noise ratio is increased.

System Level Flow Chart

The following is a flowchart for the MATLAB simulation of the active noise cancellation design.



Figure 2 Adaptive filter Flowchart

The filter coefficients are stored in a matrix, which are initialized to zero. The output of the filter is then subtracted from the desired signal resulting in the error signal. The error signal is then multiplied by a vector of previous input values and added to the filter coefficient matrix. This method of updating the filter coefficients is commonly used in adaptive filter design, and can be readily adapted to hardware implementation.

Summary

Adaptive digital filters are used when there is insufficient information about the signal to be filtered to adequately design a traditional digital filter. Adaptive filters work by recursively changing the filter coefficients to minimize the power of the error signal. This will result in the increasing of the signal to noise ratio of the error signal, reducing the interference in the signal. This will be applied to actively cancel noise in a target environment.