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적응필터를 이용한 능동소음제어시스템에 관한 연구

Active Noise Control System with Adaptive Filter and Spectral Inversion Using TMS320C6713

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1. 서 론

ANC has been applied in the system which is various with the strong point it will be able to reduce the sound arresting of the band width which is various.

The purpose of this paper is that ANC relation techniques about under embodying regarding the real-time ANC applications possibility by using DSP implementation. Active noise control (ANC) is achieved by introducing a canceling anti-noise wave through an appropriate array of secondary sources. These secondary sources are interconnected through an electronic system using specific signal processing algorithm, DSP, for the particular cancellation scheme. It has application to a wide variety of problems in manufacturing, industrial operations, and consumer products [1].

Recently, research report an active noise control announcement is become accomplished. In a domestic research report first, the research which applies LMS algorithms for the noise control of PCB industrial sites being reported. Applying Filtered-X LMS algorithms is proposed Active Noise Barrier where it uses the noise control technique.

This paper describes the digital signal processing system which is designed for ANC of noise experiment with DSP. It advanced the research regarding the phase inverter for an unwanted sound control system depicted in this paper and uses DSP the possibility of the real-time embodiment which it presented. In this paper, we suggest the solution of signal processing for ANC time delay with spectral inversion and adaptive filter using of DSP C6000 system.

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2. 본 론

2.1 THEORIES OF ACTIVE NOISE CONTROL

(1) Adaptive Filter

An adaptive digital filter consists of desired signal processing and an adaptive algorithm for adjusting the filter. A general form of adaptive filter is shown in figure 1, where d(n) is a desired response, y(n) is the actual output of a programmable digital filter driven by a reference input signal x(n), and the error e(n) is the difference between d(n) and y(n). The function of the adaptive algorithm is to adjust the digital filter coefficients to minimize the mean-square value of e(n). Therefore, the filter weights are updated so that the error is progressively minimized on a sample-by-sample basis [1].

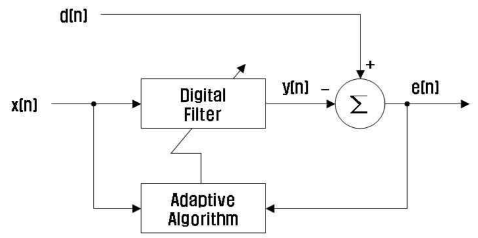

Fig. 1 Block diagram of adaptive filter.

(2) Spectral Inversion

Spectral inversion is the reversal of the orientation of the signal bandwidth. In this paper, DSP embodiment theories embodied Spectral Inversion methods. Spectral inversion means that when the input signal which is low frequency territory is confronted in high frequency territory of output signal, or opposition occurs in case. Figure 2 shows that example of spectral inversion.

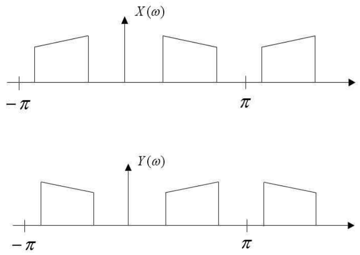

Fig 2. Spectral Inversion

3.1 SIMULATION AND EXPERIMENT

(1) Theorical Experiment by MATLAB

We experiment an adaptive filter using Matlab. We chose a wave file which is noise of an aisle of high speed train. Figure 3 shows the origin of signal and output signal by adaptive filter in Matlab.

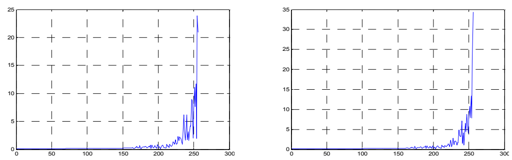
Fig 3. Simulation in Matlab.

(2) Experiment by TMS320C6713

For real-time DSP experiments, the input signals used MP3 files and to the equipment which is used in experiment TCM320AC36 Codec chip of Texas Instrument Company. it convert the analog signal to digital signal and it has the analog amplifier, band pass filter, AD/DA converter and digital input output device to be had built-in. AD/DA converters accomplished 13 bit linear conversion and the sampling speed until of the maximum 16Khz the possible outdoors selected 16Kbps from the experiment which it sees. The picture 4 with input signal (a) output signal (b) shows DFT Graph.

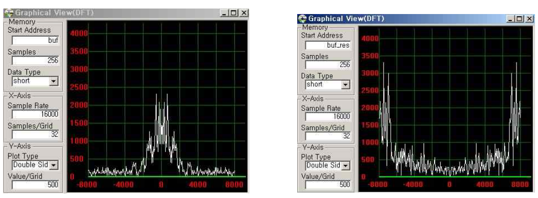
Fig 4. Input signal and output signal by DFT

(3) Verification of Experiment

It did real-time DSP experiments from III.C paragraph. Because it is not the data verification against an input signal and a spectrum reversal signal data of input signal and output signal about under extracting, it drew a graph with separate way. Figure 5 shows the original signal and out signal graphs.

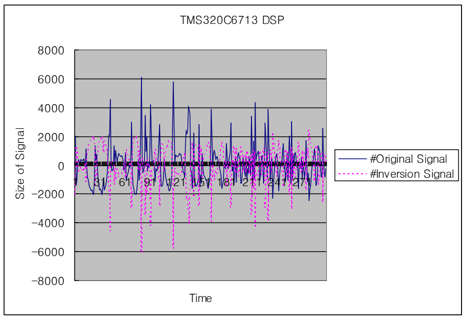

Fig 5. Data Verification

4. 결 론

To improve the performance of the active noise control system with DSP system using spectral inversion and adaptive filter by Matlab are proposed. The adaptive filter is obtained by the approximation method. Combining DSP system and adaptive filter, it is proved that the proposed control scheme improves the performance as well as the well estimation via computer simulation.

We will discuss an active noise control based on DSP system completely in a nearly future.

후 기

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