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A LOW COST ACTIVE NOISE CONTROLLER

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1. INTRODUCTION

The active noise control, to reduce acoustic noise with acoustic noise in opposite phase, is used more and more in practical applications and can be used for educational purposes in acoustic concepts. This paper shows how to develop and implement a simple inexpensive active noise controller to make acoustic experiments, with this fascinating technique, in different noisy environments.

The system consists of only a few elements, usually used in a multimedia computer so the implementation cost is very low. These components are: a microphone, two powered multimedia speakers and a digital signal processor. In order to reduce the complexity and cost of the system, the digital signal processor is a DSP starter kit of *Texas Instruments* [1], although other DSP starter kits can be used with similar features [2].

The active controller is a "single input single output" (SISO) system which reduces an undesired acoustic noise in any enclosure. In the experiment, the noise is produced by a multimedia soundcard [3]. A sound file (noise.wav file) is continuously played with the media player of *Windows*. The block diagram of the active noise control experiment is shown in figure 1.

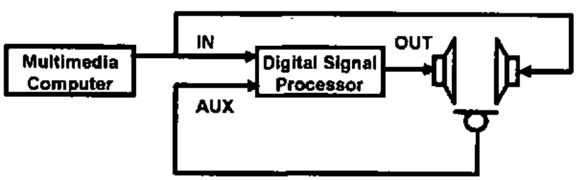

Figure 1. Block diagram of the basic active noise controller

An adaptive algorithm has been developed on the DSP Starter Kit. This algorithm is based on the Filtered-X LMS [4] with an on-line estimation of the cancellation path (speaker-path-microphone) as it is proposed by Fan and Vemouri [5]. The DSK is programmed with an assembler and is loaded into the DSK with RS232 interface. Once the program is tested with a debugger, a high level software application for *Windows* permits to the user modify easily the parameters of active noise controller, that is: filter taps, sampling rate, convergence factors and leaky factors.

2. ADAPTIVE ALGORITHM

The adaptive algorithm used is based on the Filtered-X LMS [4], the classical adaptive algorithm in the design of an active noise control. The system works correctly with a dynamic estimation of the cancellation path (speaker-path-microphone, H(z)), that is, the delay between the output signal of the digital signal processor -DSK- and the microphone input signal to the DSK. If this delay is unknown the algorithm diverges. For estimating this delay there are some methods [6]. Some of them use an uncorrelated noise added to the output signal to identify this transfer function with a LMS algorithm [7]. These methods get unbiased estimates but add a noise in the system and the total performance gets worse. Other techniques try to get the transfer function estimate directly with the output signal [8]. The main disadvantage is that the estimate is biased and it is not possible to reach the minimum signal in the microphone input, that is, the maximum attenuation of the acoustic noise. Both of the techniques can be on-line or off-line estimation. The off-line estimation is made before the Filterd-X LMS algorithm begins to work. The on-line estimation is desirable because of possible cancellation error path alterations. The estimation used in this system is on-line and directly with the output signal of the digital signal processor [5], figure 2.

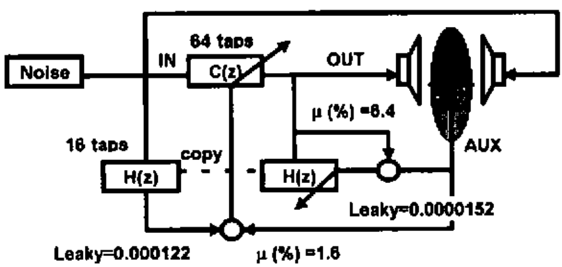

Figure 2. Filtered-X LMS Algorithm with on-line path estimation

The adaptive algorithm modifies automatically its scalar convergence factor μ to guarantee the convergence. This factor is increased when the

RMS level of the input signal (IN) is greater than RMS level of the error signal (AUX), and decreased otherwise. The μ adjustment is made even when the error signal is minimized because of adaptation [6]. So, the time constant of adaptation is now invariant, independent of the RMS level difference.

3. STARTER KIT TMS320C26

The TMS320C26 DSP Starter Kit (DSK) is an ideal low-cost tool for first-time users interested in evaluating a DSP platform [1]. The DSK allows users the user to experiment with and use a DSP for real-time signal processing, that is, to write and run real-time source code, evaluate that code, and debug their system.

The DSK development kit is designed to be operated from a tipical computer.The DSK circuit board module includes: TMS320C26 DSP processor, TLC32040 Analog Interface Circuit (Codec), RS-232 interface chips and AC/DC power supply circuitry. The digital processor has 1.5K words of on-chip RAM and the time cycle is 100 nsg. The codec permits two analog inputs and analog output, necessary to implement the SISO active noise control. The maximum sampling rate is 19200 Hz, enough for the application, and the samples have a resolution of 14 bits. The DSK software is a debugger interface, an assembler, a loader and a test program.

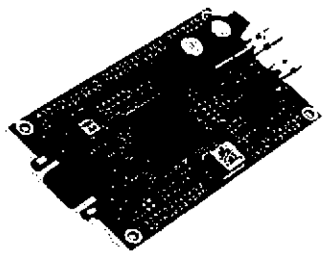

Figure 3. DSK Starter Kit TMS320C26. Texas Instruments.

4. DEVELOPMENT OF THE PROGRAM ON THE DSK

The algorithms have been programmed with the assembler of the DSK and tested with the debugger with good results. The final application is made on a high level software for *Windows*, where the user can modify continually all the parameters of the active noise controller. The processing is made once the two samples in input IN and AUX are caught as it is shown in figure 2. The DC is removed from the samples. The RMS levels of both inputs are calculated. These values serve to calculate the appropriated convergence factor, sample by sample. The cancellation path H(z) is estimated with the LMS algorithm (the reference is the OUT signal and the residual is the AUX input signal). The H(z) coefficients are copied to a FIR filter in the Filtered-X LMS algorithm that processes the input signal IN (acoustic noise reference) to produce the output sample OUT. The error signal caught by microphone makes the algorithm to converge to a minimum.

5. RESULTS AND CONCLUSIONS

The system has been tested with multiple periodic signals and the results have been excellent in all the cases. The attenuation is more than 30 dB in the fundamental frequency of the signals. This attenuation is presented in all the enclosure when both speakers are very close (by 25 cms). The figure 4 shows the attenuation produced when the acoustic noise is a tone and square signal. The sampling rate is 5000 Hz. The Filtered-X LMS has 64 taps with a FIR filter of 16 taps (estimation path). The convergence factors are 1.6 % and 6.4 % of the maximum possible values (figure 2) including leaky factors to guarantee the stability of the adaptive filters, making null coefficients when the acoustic noise level disappears.

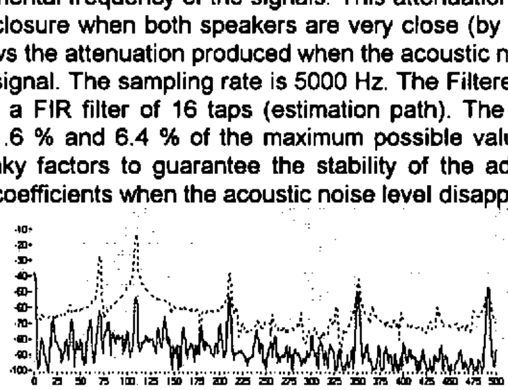

Figure 4. Cancelled spectrum produced by the ANC (dB vs Hz).

The results show a fast convergence and a robust stability independent of the RMS levels of both inputs. The cost paid is a little residual increasing. The sytem is perfect for eduactional and first-time experiments on acoustic noise control, because of simplicity and besides is inexpensive.

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