Development of an Active Noise Controller in the DSP Starter Kit

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Contents

Figures

Development of an Active Noise Controller in the DSP Starter Kit

Abstract

This paper shows the implementation of a low cost active noise control (ANC) on the DSP Starter Kit TMS320C26 [1]. The ANC system uses a filtered-x LMS adaptive algorithm [2] with on/off-line path cancellation estimates [3,4,5]. The system consists of an error microphone plus a signal amplifier, two powered speakers (primary and secondary source) and the controller (the DSP Starter Kit). The total performance is limited by the DSK features but it is possible to reach 30 dB of acoustic attenuation in narrowband acoustic noises. The fixed-point DSP controller has been programmed with an assembler and a debugger supplied by *Texas Instruments,* and the final application is built on the *Windows* operating system environment where the user can easily modify the parameters of the active noise controller (filters taps, convergence factors, leaky factors) with a friendly control panel.

This document was an entry in the 1997 DSP Solutions Challenge, an annual contest organized by TI to encourage students from around the world to find innovative ways to use DSPs. For more information on the TI DSP Solutions Challenge, see TI's World Wide Web site at www.ti.com.

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Introduction

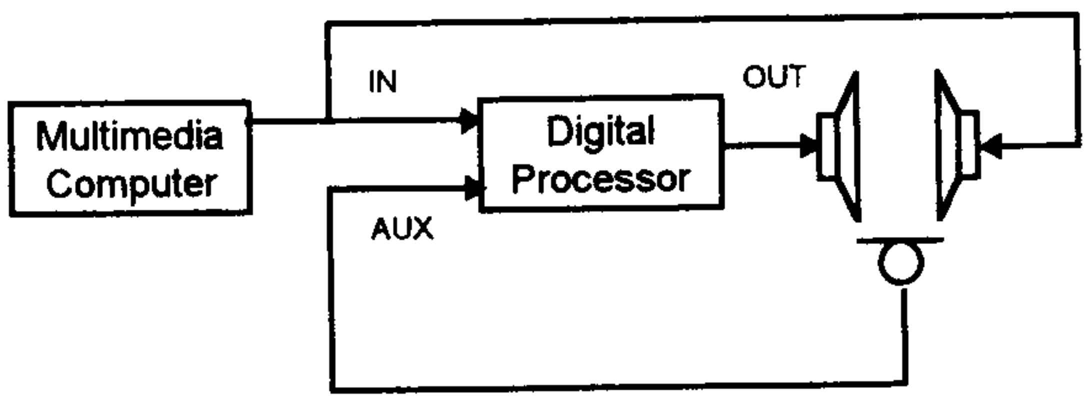
What is active noise control? [6] Active control is sound field modification, particularly sound field cancellation, by electro-acoustical means. In its simplest form, a control system drives a speaker to produce a sound field that is an exact mirror-image the offending sound (the "disturbance"). The speaker thus "cancels" the disturbance, and the net result is no sound at all. In practice, of course, active control is somewhat more complicated. The name differentiates "active control" from traditional "passive" methods for controlling unwanted sound and vibration. Passive noise control treatments include "insulation", silencers, vibration mounts, damping treatments, absorptive treatments such as ceiling tiles, and conventional mufflers like the ones used on today's automobiles. Passive techniques work best at middle and high frequencies, and are important to nearly all products in today's increasingly noise-sensitive world. But passive treatments can be bulky and heavy when used for low frequencies. The size and mass of passive treatments usually depend on the acoustic wavelength, making them thicker and more massive for lower frequencies. The light weight and small size of active systems can be a critically important benefit. In control systems parlance, the main four parts of an active control system are:

- The *plant* is the physical system to be controlled; typical examples are a headphone and the air inside it, or air traveling through an air-conditioning duct.
- *Sensors* are the microphones. accelerometers, or other devices that sense the disturbance and monitor how well the control system is performing.
- *Actuators* are the devices that physically do the work of altering the plant response; usually they are electromechanical devices such as speakers or vibration generators.
- The *controller* is a signal processor (usually digital) that tells the actuators what to do, the controller bases its commands on sensor signals and, usually, on some knowledge of how the plant responds to the actuators. Analog controllers may also be used, although they are somewhat less flexible and thus more difficult to use.

Active noise control configuration

The ANC developer kit to implement is working in an acoustic configuration as shown in figure 1.

Figure 1. Block diagram of the basic active noise controller



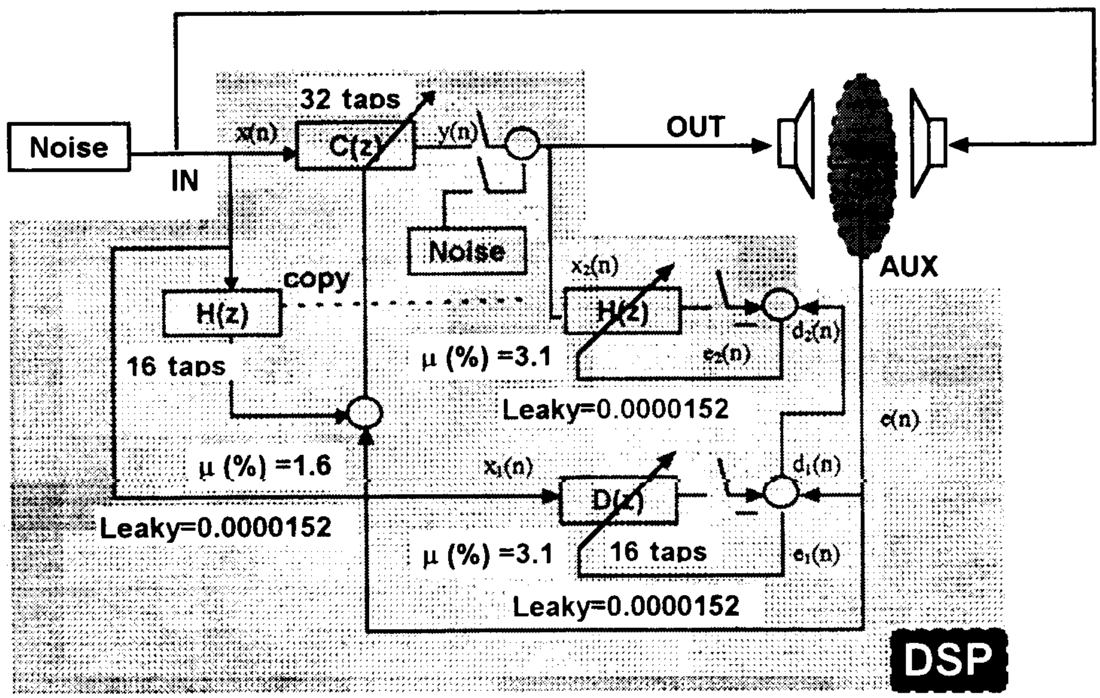
A personal computer, equipped with a low cost soundcard like any *Soundblaster* [7] or compatible card, and with an audio software like *Cool-Edit* [8], can generate any kind of noise signal in a loop mode. The *Cool-Edit* software generates many types of periodic signals and three different random signals. The maximum frequency of these signals is 500 Hz. The attenuation of higher frequencies is irrelevant because the silent zone is only one tenth of the acoustic wavelength [9]. For an acoustic wave of 500 Hz the effective silent zone is only 7 cm.

One of the two channels of a stereo file, noise generated by the soundcard, is applied to one powered speaker to produce an undesired acoustic noise to cancel (figure 1). The other channel is applied to one input of the DSP starter kit, the active noise controller. The DSP produces an output signal that is also applied to another powered speaker, trying to cancel the noise. A microphone catches both added noises. This signal makes the DSP to work properly in order to reach the maximum attenuation.

Adaptive algorithms.

The active controller is based on the classical filtered- X LMS adaptive algorithm [2], working with other adaptive FIR algorithms that keep a robust convergence to the minimum residual signal at the error microphone. The complete block diagram is shown in figure 2.

Figure 2. Adaptive algorithms implemented in the DSP



As shown in the figure, the user can choose any of the typical configurations used in a *single input single output* active noise canceller. The algorithm LMS applied to the FIR H(z) estimates the electrical-acoustic error path, the transfer function, and this estimate is used by the filtered-X LMS adaptive algorithm applied to C(z). The estimation is necessary to keep a working stability in the whole process [2]. Such estimates can be configured to be on/off line with the cancelling process made by the adaptive C(z) FIR filter. If off line estimation is chosen, an auxiliary random noise is generated by the DSP before C(z) begins to have the optimum coefficients [3]. Once H(z) has its minimum residual, the *unbiased* H(z) coefficients are copied to the filtered-X LMS algorithm. Then, the active control attenuation begins to work.

On the other hand, if on line estimation is chosen, the LMS algorithm applied to H(z) works together [4] with the filtered-X LMS algorithm applied to C(z). In this case, the *biased* coefficients are updated and copied to the filtered-X LMS algorithm in each iteration. There is a third LMS algorithm applied to another FIR filter, D(z). This algorithm tries to remove any bias in the H(z) estimation [5], making the total performance more robust to instabilities.

In every adaptive filter the filter taps, the percentage of the maximum convergence factor μ and a noisy leaky factor are previously chosen by the user. The maximum convergence factor is calculated and updated each iteration. First, the average power of the input signal to any adaptive FIR filter, p(n), is calculated with a simple recursive IIR algorithm, and then μ factor is also calculated:

$$p(n) = \alpha x^2(n) + (1-\alpha)p(n-1)$$

$$\mu = \frac{1}{taps \cdot p(n)} \cdot \frac{\mu(\%)}{100}$$

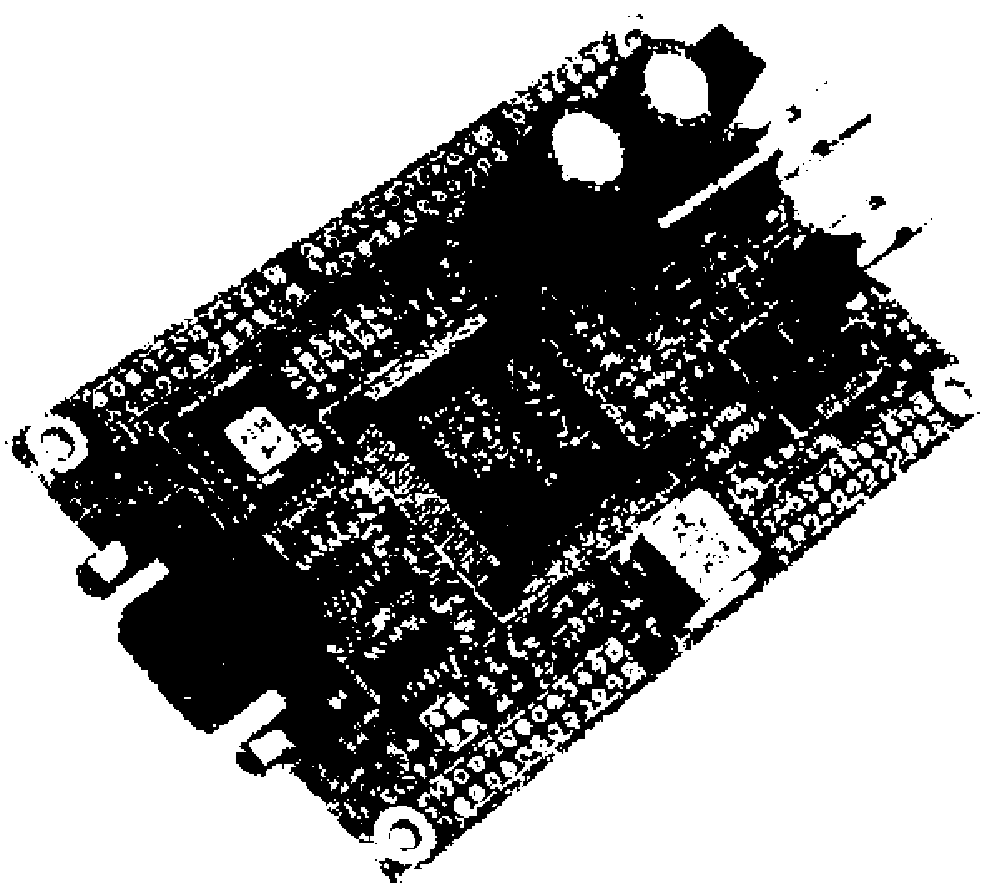
Instabilities can occur when the error signal, caught by the microphone, is higher than the reference signal input. In this case, the algorithm multiplies the error signal by a multiplier to make both RMS values similar at the beginning of the adaptation process [10].

DSP Starter Kit

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 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 typical 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 has two analog inputs and one 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 TMS320C26



Development of the ANC program (DSP code)

All the active noise controller is written in an ASCII editor. This file is assembled, creating an object file that is loaded with a RS-232 interface to the DSP. The structure of the program is shown in the next figure. The main program runs in a loop from *BEGINNING* to *BEGIN TO BEGIN.* Previously an include file (setup.asm) initializes all the parameters, variables and memory used. The memory configuration is of great importance because there are many algorithms working together.

Figure 4. Programming structure of the active noise controller

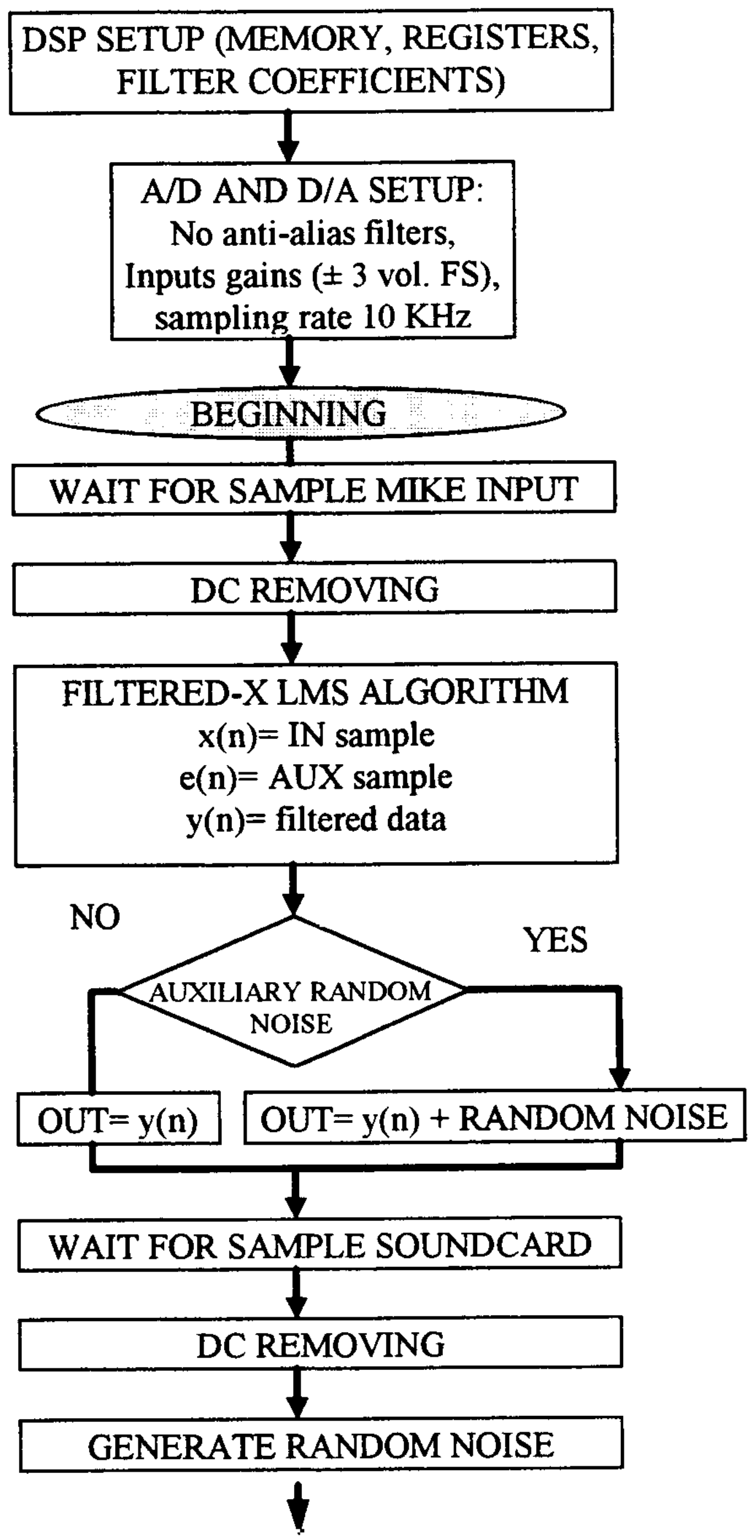
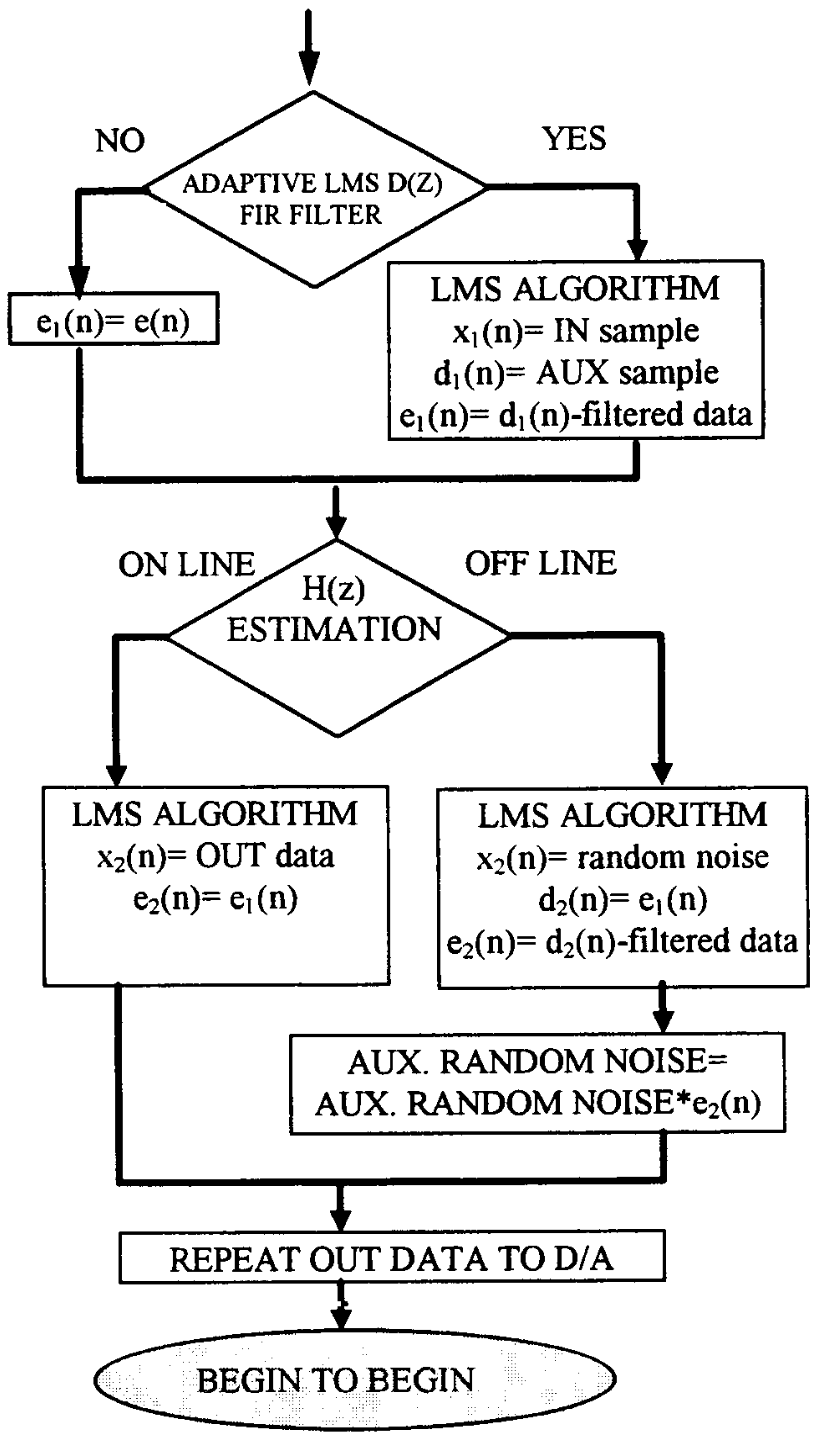


Figure 4. Programming structure of the active noise controller (continued)

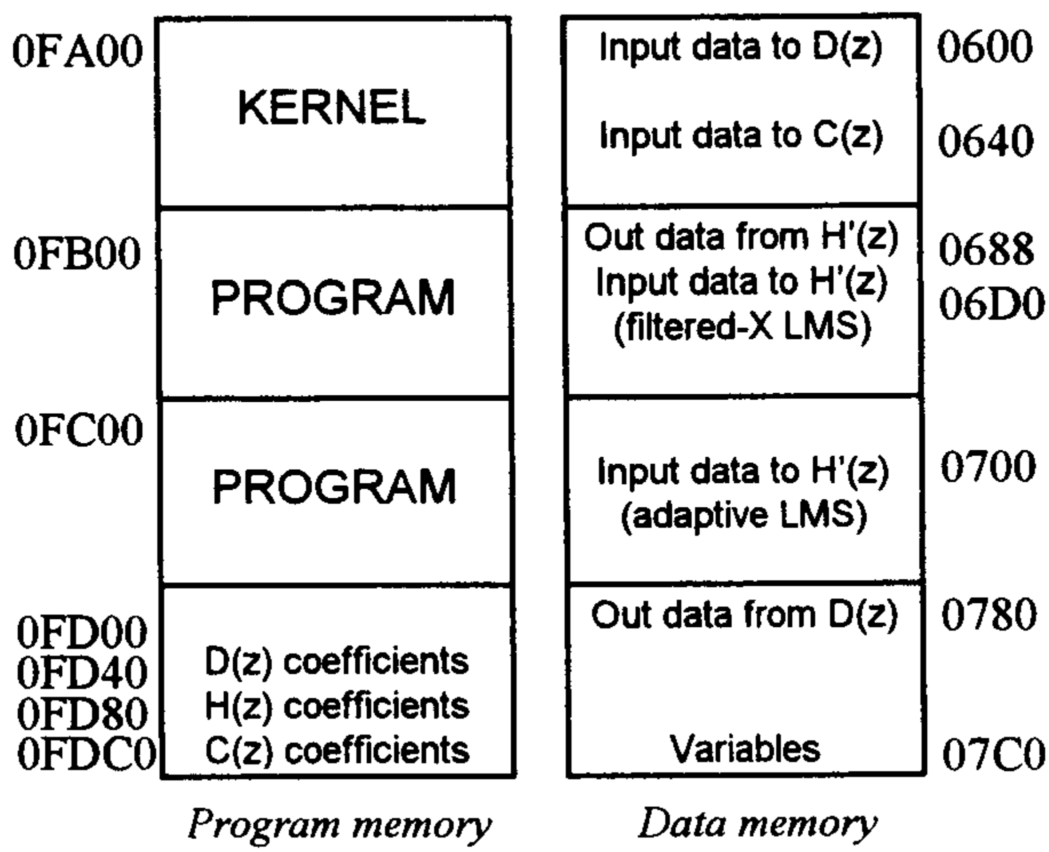


Programming Details

Initialization

A file called setup.asm is included in the main program in order to ensure that the active noise controller works properly every time it is executed. The AIC (analog converter) works with: sampling 10Khz, ± 3 volt full scale and no anti-alias filter (we need the two inputs of the AIC). The data memory is initialized with the value 0000h and the adaptive filters coefficients start with the value 00400h.

Figure 5. Memory configuration



Main program

The main program makes the whole processing with every two input samples. It waits for each input sample with an idle instruction. Then, with the two input samples (inputs IN and AUX) a processed sample is left at the DXR (data transmit register) twice. This causes the effective sampling rate to be divided by two. Every output data left in the DXR register has the two LSB bits with the value "11" so a secondary transmission will follow after a delay of four shift clock cycles. This permits switching the input data, IN and AUX. One part of the processing is made once the IN data is collected from DRR (data receive register) and the rest when the AUX data is present.

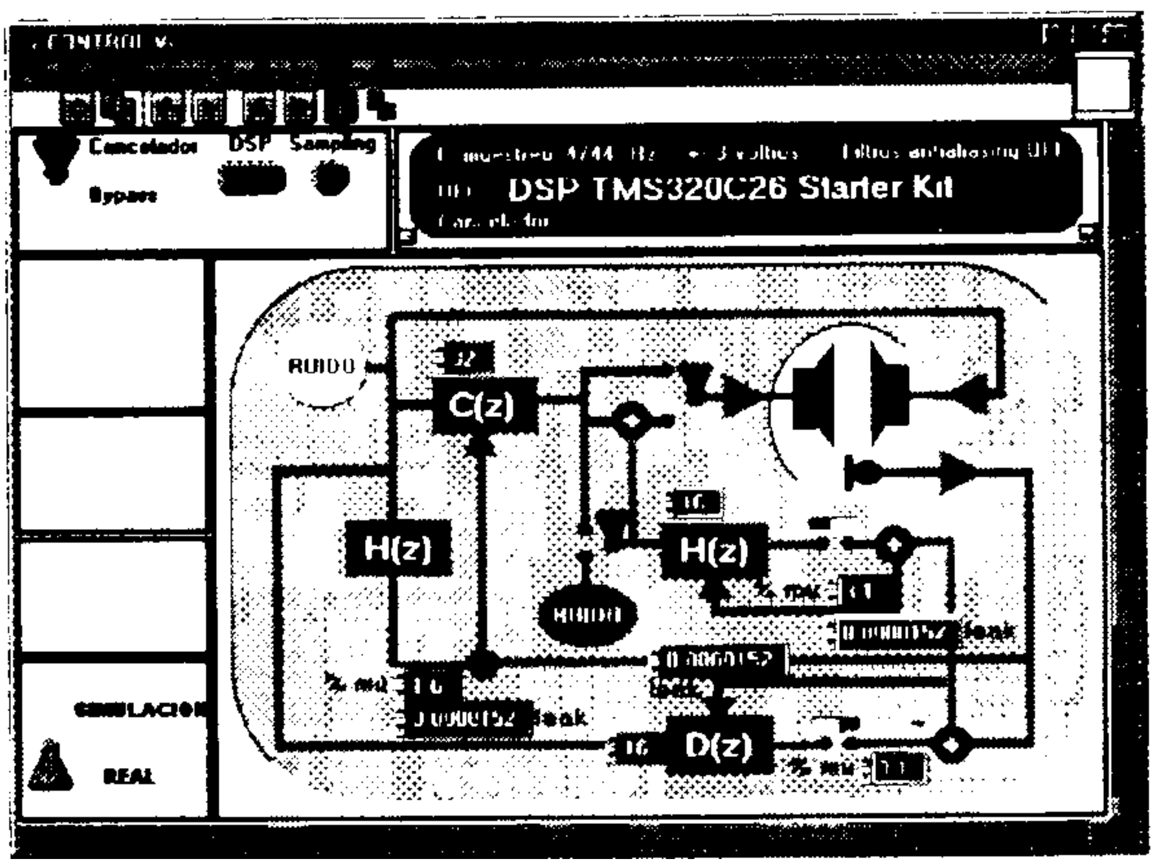
Adaptive algorithms

These algorithms work as shown in figure 2. They are loaded as subroutines in the first part of the program memory so the FIR coefficients can be read as data memory (switch from CONF1 to CONF2). The input data of the algorithms are DC cleaned previously and then, the RMS value of the two inputs are estimated in each iteration so the best convergence factor can be calculated in each iteration too. The stability of the adaptive algorithms is improved [10].

User interface

Once the program has been debugged, a graphic application, built for running alone in Windows operating system, permits to the user modify the parameters of the control. These parameters are updated directly in the object file. Then, the file is loaded to DSP for running the new active noise control configuration. The user interface has been developed with the *LabView* Software of *National Instruments.* This software is very easy to use and has a complete set of high level functions in a diagram block programming [11]. Other software can be used like *Visual Basic* or *C++* with similar results. The aspect of the graphic user interface is shown in the next figure.

Figure 6. Graphic control panel of the application



Details of the Windows application

The final application is merely a user control panel that modifies the DSK file to be loaded into the DSP. For example, each switch ON/OFF in the control panel is made with a NOP operation. The variables are also modified directly in the DSK file. Once the file has been modified, it is loaded into the DSK.

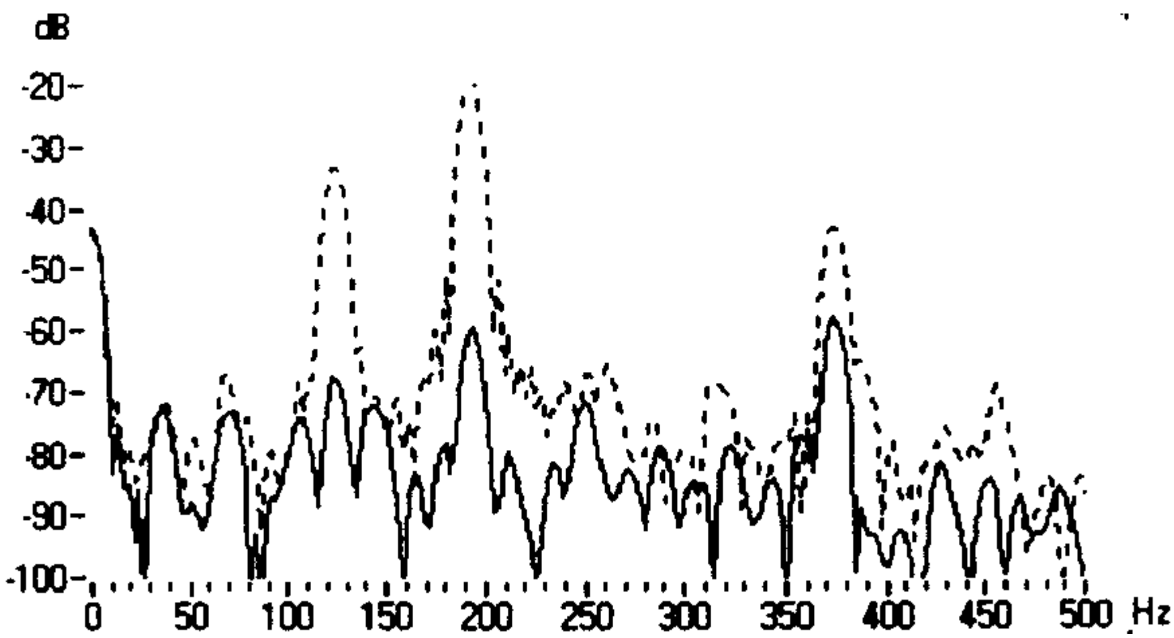
Experiments and results

The active noise canceller has been tested in a small room with different types of noisy signals produced by fans, engines, cars, helicopters, and so on. These signals have been recorded previously in the *Windows* audio file format *(noise.wav)* and then played in loop mode with the *Cool-Edit* audio software [8]. Other typical signals like random and periodic noise, generated directly by the audio software, have also been tested. The values of the control parameters depends on the speakers separation, the microphone position and the input signal levels in the DSP.

One experiment tested had the next configuration: the two speakers were very close, about 25 cms, and the microphone was closer to the speaker tied to the DSP output than that tied to the soundcard output. The attenuation produced is shown in the next figure. This attenuation was perceived in all the room because the noisy sources were very close [9]. When the speaker separation was increased the attenuation was still produced but this time only in the microphone environment, and the attenuation sensation perceived in the room disappeared.

Anther important aspect to take into account is the input signal levels. Both signal levels have to be similar and below 3 volts. Without this requirement, instabilities can occur in the adaptation process of the adaptive algorithms, and the active noise controller does not work [10].

Figure 6. Comparative spectrums: Noise vs cancelled noise by the ANC



Conclusions and future trends

This paper shows how it is possible to develop a simple low cost kit for making active noise control experiments. It is only necessary to have a multimedia computer and a DSP Starter Kit. The final user application is a graphic control panel, where the user can repeatedly modify all the parameters used in the adaptive algorithms until the optimum values for the experiment are found. The application is very useful for educational purposes, students can experiment themselves with the acoustic phenomena.

The DSK features (1.5 Kwords, sampling rate and mono input/output channel, band-pass anti-alias filter, 14 bits) limit the performance of the active noise controller but even with these limitations the results are good. Some of them can be solved with other DSK Starter kits based on DSP TMS320C50 and recently on TMS320C30 too.

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