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Implementation of Active Noise Filter for Real-time Noise Reduction Using the TMS320C5402 DSP Kit

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Abstract: Noise detection and reduction has become a primary concern in the fields where high speed and reliable communication is the essential criterion. It is therefore necessary to develop and build certain real time components, which can be made to work under the conditions of real-time environment where reliability and speed should be very high. In this field of Real-Time Digital Signal Processing numerous processors have been made by various companies like Texas Instrumentation, Motorola, Intel, etc. These devices offer an agreeable solution for this application. A comparable study has been made between some of the kits namely TMS320C5402 made by Texas Instrumentation. For the purpose of analysis a fundamental Active Noise filter design has been used as a test program and results are obtained from real time working conditions.

Key words: Active noise control, FIR filter, LMS algorithm, SNR, error microphone, TMS320C5402, Matlab 6.5, C code

INTRODUCTION

Adaptive Signal Processing is a method to cancel the undesired ambient noise by adding a secondary sound wave with the same amplitude and the reverse phase to the original signal. A computer processing electrically produces such secondary sound. This technique is effectively performed for the low or middle frequency sound waves (Elliott and Nelson, 1993). An adaptive filter is composed of the Finite Impulse Response (FIR) filter as a digital filter and the Least Mean Square (LMS) algorithm as adaptive control algorithm, these criterion apparently imposes certain restrictions on the canceling system. Firstly, for highly effective noise canceling system the noise source must be nearly stationary in relation to the speaker emitting the anti-noise waveform. Second, the noise source should be located in close proximity to the noise filter. Acoustic delay is another important issue that must be dealt with in a noise cancellation system. Physically there is always a distance between the source, the anti-noise generator and the residue noise detector. These physical distances provide noise propagation delays, which in turn cause different phase shifts, depending on the relative location of objects.

PROBLEM STATEMENT

Our aim is to achieve noise reduction for signals transmitted through the wireless medium. In such a communication, all the noise is added in the channel. The noise is highly random. Here there is no source for obtaining a correlated noise at the receiving end. (Morinushi, 1991). Only the received signal can tell the story of the noise added to it. Hence somehow, only if it is possible extract the noise from the received signal, through some means, then the above-mentioned adaptive techniques to enhance the signal to noise ratio of the received signal (Takahashi and Hamada, 1991). A method to obtain a correlated noise from the received signal.

MATHEMATICAL MODELING OF ADAPTIVE NOISE FILTER

For the sake of simplicity both noise and anti-noise waveforms within the same vicinity are assumed. The basic mathematics involved in building an Adaptive Noise Filter is listed below:

$$e(n) = x(n)+y(n-d) \quad (1)$$

$$w(n+d) = w(n)-mu*e(n) \quad (2)$$

$$y(n) = w(n+d) \quad (3)$$

e (n): corresponds to the error generated because of the combined effect of the noise signal and the antinoise waveform.
w(n): corresponds to the correction factor that corresponds to the proportion of error has to be applied to the existing anti-noise waveform (Fig. 1).

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y(n): is the anti-noise waveform that is applied (Tohma, 1991).

During this process of noise reduction there is a delay factor that has to be taken into account. The delay corresponds to the sum of propagation delay and processing delay. Propagation delay is the time taken for the signal to reach the processor from the external transducer, which is the micro-phone (Takahashi and Abe, 1993). Similarly a time delay has to be taken into account of the processing speed. The total delay serves as a Fig. 2 of merit for the system.

The process should be optimized such that the noise signal is reduced to a considerable extent without creating any additional disturbance in the system (Eriksson, 1993; Widrow and Stearns, 1985). The delay corresponding to the propagation of the sound is fixed for a given setup and is solely dependant on the velocity of sound in that medium, which is a constant. The factor that can be altered is the processing delay. The basic constraint of the Active Noise Filter is its inability to be used as a noise reduction system for high frequency components. In the likelihood's of this system being used for filtering high frequency components, the delay time is significant to that of the signal and can introduce more amount of noise.

During simulation of the circuit points of singularities were created when delay time became significantly larger than time period of the frequency component. This will lead to creation of additional noise and the system fails to perform.

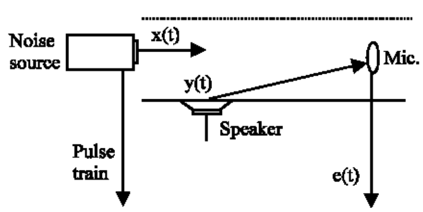

Fig. 1: Simple active noise detection model

In this critical period the algorithm for ANF has to be processed and sent to the antinoise generator. The processing delay mentioned above is dependant of the processor and the optimization of the algorithm. Keeping the same algorithm, the choice of processor solely determines the delay time and correspondingly fixes an upper limit on the maximum frequency that could be detected and reduced by the setup (Haykin, 1984).

IMPLEMENTATION OF ACTIVE NOISE FILTER USING TMS320C5402 PROCESSOR

The complexity of an adaptive filter is usually measured in terms of its multiplication rate and storage equipment. The data flow and handling considerations are also major factors due to parallel hardware multiplier, pipeline architecture and the size limitation of the fast on-chip memory.

Implementation should be made more efficient by taking advantage of these attributes in the DSP's architecture. The TMS3220C25 can execute an instruction in as little as 8i0ns and the processors architecture makes it possible to execute more than one operation per instruction cycle (Casali and Robinson, 1994). In order to produce the fastest filtering routine, all data buffer memories and filter coefficients are stored in data random access memory.

The two models, which were used to test the filter, are Floating Point Arithmetic and Fixed Point Arithmetic. A brief comparison about the two is given below:

- In terms of economy the Floating Point Arithmetic processor is more expensive than the Fixed Point Arithmetic. The amount of heat generated also is significant in the case of the former

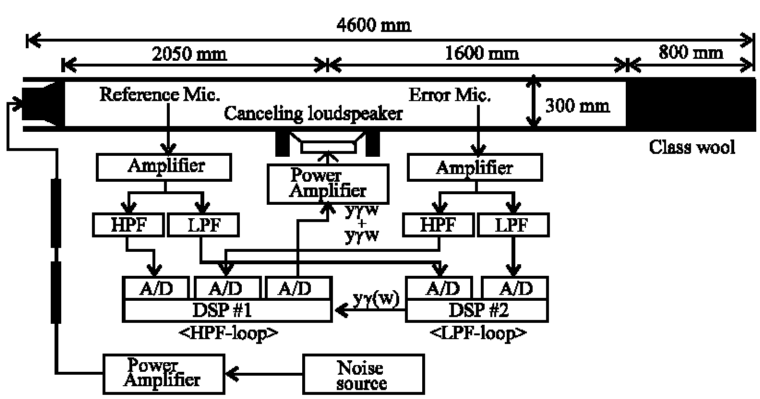

Fig. 2: Critical issues in the design of an Active Noise Filter

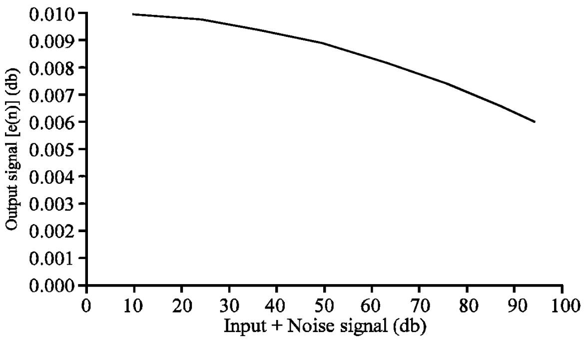

Fig. 3: Simulation result for Error signal

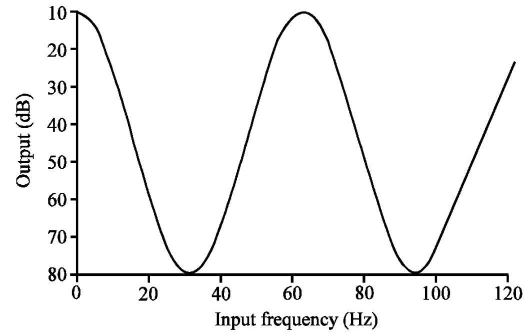

Fig. 4: Simulation result for Adaptive Filter using a delay time of two counts

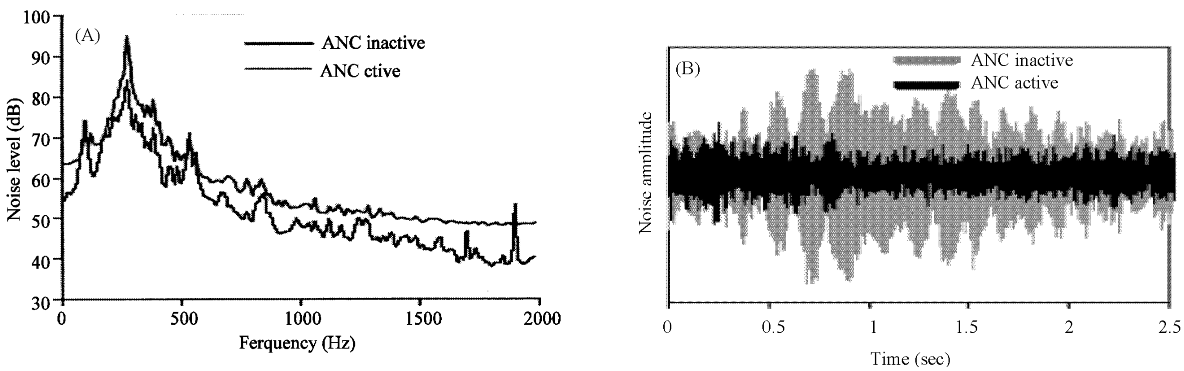

Fig. 5a, b: Noise signal in time domain and noise power spectrum at an error microphone

- In terms of precisions the Floating Point Arithmetic processor uses the IEEE 754 standard of representation of floating point numbers in terms of mantissa and exponent (Nishiyama *et al*., 1979). All arithmetic operations retain their precession during the operation.

The C Code that was implemented in the processor is put as an Appendix. The code is relatively larger and optimization should be done very carefully. For the sake of simplicity Code Conversion Studio was used and the code is placed in Appendix.

IMPLEMENTATION OF ANF USING LMS ALGORITHM IN MATLAB6.5

The algorithm for the simulation of the Active Noise Filter was originally done in Matlab 6.5 (Fig. 3-5). The code was rewritten in standard C format for easy conversion to assembly language. (Sherratt *et al*., 1999). The Matlab source code that was originally used to test the system is listed in Table 1.

Table 1: Total noise reduction in the Error Microphone

Error Microphone	Measure 1	Measure 2
Mic. 1	7.13 dB	8.61 dB
Mic. 2	6.41 dB	6.94 dB
Mic. 3	8.82 dB	9.48 dB
Mic. 4	8.50 dB	9.30 dB

CONCLUSIONS

The Active Noise Filter was embedded as a programming the DSP kit TMS320C5402 and the results were found to be similar to the results generated by Matlab 6.5. The real time application of the Active Noise filter was successful. Modern day DSP Kits like TMS320C5402, TMS 320C6211 are fast enough to provide the speed and reliability of real-time Noise filtering. The significance of this DSP Kit shows the major advancements in the field of signal processing which will have a dramatic impact on Real-Time processing is done Compact devices embedded with TMS processors installed can be used as effective communication equipments. Finally the DSP processor kit was more accurate as compare to Matlab6.5 software for real-time process.

APPENDIX

I: "C" code for LMS algorithm

```
#include<stdio.h>
#include<math.h>
void main()
{
int I=2;
float mu=1.0;
float
i,x[100],y[100],a[100],b[100];
int s=0, L,j;
for(I=0;i<=1;i=I+0.1)
{
x[s]=I;
s=s+1;
y[s]=sin(x[s]);
}
L=sizeof(y)/4;
for(j=0;j<=L;j++)
{
printf("%f\t",y[j]);
}
for(I;I<100;I++)
{
a[I]=0;
b[I]=0;
}
for(I=1;I<=95;I++)
{
x[I]=y[I]-a[I];
b[I+1]=b[I]+mu*x[I];
a[I]=b[I+1];
}
printf("\nThe error \n");
for(I=1;I<=95;I++)
{
printf("%f\t" ,x[I]);
}
printf("\n%weights");
for(I=1;I<=95;I++)
{
printf("%f \t",b[I]);
}
printf("\nThe actual output \n");
for(I=1;I<=95;I++)
{
printf("%f\t" ,a[I]);
}
}
```

PART OF THE ASM CODE GENERATED FOR TMS320C5402 KIT

```
0000:0000 0001 ADD 1h,A
0000:0001 0001 ADD 1h,A
0000:0002 0001 ADD 1h,A
0000:0003 0001 ADD 1h,A
0000:0004 0001 ADD 1h,A
0000:0005 0001 ADD 1h,A
0000:0006 0001 ADD 1h,A
0000:0007 0001 ADD 1h,A
0000:0008 0001 ADD 1h,A
0000:0009 0001 ADD 1h,A
0000:000A 0001 ADD 1h,A
0000:000B 0001 ADD 1h,A
0000:000C 0001 ADD 1h,A
0000:000D 0001 ADD 1h,A
0000:000E 0001 ADD 1h,A
0000:000F 0001 ADD 1h,A
0000:0010 0001 ADD 1h,A
0000:0011 0001 ADD 1h,A
0000:0012 0001 ADD 1h,A
0000:0013 0001 ADD 1h,A
0000:0014 0001 ADD 1h,A
0000:0015 0001 ADD 1h,A
0000:0016 0001 ADD 1h,A
0000:0017 0001 ADD 1h,A
0000:0018 0001 ADD 1h,A
0000:0019 0001 ADD 1h,A
0000:001A 0001 ADD 1h,A
0000:001B 0001 ADD 1h,A
0000:001C 0001 ADD 1h,A
0000:001D 0001 ADD 1h,A
0000:001E 0001 ADD 1h,A
0000:001F 0001 ADD 1h,A
0000:0020 0001 ADD 1h,A
0000:0021 0001 ADD 1h,A
0000:0022 0001 ADD 1h,A
0000:0023 0001 ADD 1h,A
0000:0024 0001 ADD 1h,A
0000:0025 0001 ADD 1h,A
0000:0026 0001 ADD 1h,A
0000:0027 0001 ADD 1h,A
0000:0028 0001 ADD 1h,A
0000:0029 0001 ADD 1h,A
0000:002A 0001 ADD 1h,A
0000:002B 0001 ADD 1h,A
0000:002C 0001 ADD 1h,A
0000:002D 0001 ADD 1h,A
0000:002E 0001 ADD 1h,A
0000:002F 0001 ADD 1h,A
0000:0030 0001 ADD 1h,A
```

```
0000:0031 0001 ADD 1h,A
0000:0032 0001 ADD 1h,A
0000:0033 0001 ADD 1h,A
0000:0034 0001 ADD 1h,A
0000:0035 0001 ADD 1h,A
0000:0036 0001 ADD 1h,A
0000:0037 0001 ADD 1h,A
0000:0038 0001 ADD 1h,A
0000:0039 0001 ADD 1h,A
0000:003A 0001 ADD 1h,A
0000:003B 0001 ADD 1h,A
0000:003C 0001 ADD 1h,A
0000:003D 0001 ADD 1h,A
0000:003E 0001 ADD 1h,A
0000:003F 0001 ADD 1h,A
0000:0040 0001 ADD 1h,A
0000:0041 0001 ADD 1h,A
0000:0042 0001 ADD 1h,A
0000:0043 0001 ADD 1h,A
0000:0044 0001 ADD 1h,A
0000:0045 0001 ADD 1h,A
0000:0046 0001 ADD 1h,A
0000:0047 0001 ADD 1h,A
0000:0048 0001 ADD 1h,A
0000:0049 0001 ADD 1h,A
0000:004A 0001 ADD 1h,A
0000:004B 0001 ADD 1h,A
0000:004C 0001 ADD 1h,A
0000:004D 0001 ADD 1h,A
0000:004E 0001 ADD 1h,A
0000:004F 0001 ADD 1h,A
0000:0050 0001 ADD 1h,A
0000:0051 0001 ADD 1h,A
0000:0052 0001 ADD 1h,A
0000:0053 0001 ADD 1h,A
0000:0054 0001 ADD 1h,A
0000:0055 0001 ADD 1h,A
0000:0056 0001 ADD 1h,A
0000:0057 0001 ADD 1h,A
0000:0058 0001 ADD 1h,A
0000:0059 0001 ADD 1h,A
0000:005A 0001 ADD 1h,A
```

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