MULTI-CHANNEL ACTIVE NOISE CANCELLATION USING THE DSP56001

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ABSTRACT

This paper reports on the performance of a portable active noise cancellation system based around a PC hosted 20MHz Motorola DSP56001 processor with a 4 channel analogue I/O board connected to the real world via standard consumer audio components. The system will perform active noise cancellation over the frequency range 65-500Hz. Quantitative results are presented for the cancellation of single tone noise, and of narrowband noise, and a measure of the ANC power spectrum is calculated for various parameters of the filtered-X LMS algorithm in different acoustic environments. Qualitative results based on human hearing perception of the attenuation of various narrowband and real world noise sources are also discussed.

1. INTRODUCTION

Active noise cancellation (ANC) is the acoustic reduction of noise by introducing a corresponding anti-phase sound wave. At a known area in space destructive interference of the noise source with the anti-phase sound wave is made to occur, resulting in a *quiet* zone. In recent years ANC has become increasingly popular due to the availability of inexpensive DSP devices and the development of robust DSP algorithms [5], [3], [1]. Also TV and popular press exposure [4] and the requirement for companies to introduce hi-tech attributes to further promote their products has lead to working systems to reduce the noise produced from rotating machinery in cars, factories, aircraft, air conditioners, and medical equipment.

2. ACTIVE NOISE DSP

Based around the well known single channel (non-active) adaptive LMS noise canceller [5], a number of architectures have been developed for ANC. Because the error signal in ANC is created by the sum of two acoustic waveforms rather than a sum of two signals within hardware/software, the LMS algorithm cannot be applied directly for ANC due to the transfer functions following the adaptive filter i.e. those of the amplifier, loudspeaker, acoustic path between loudspeaker and microphone, etc. To cope with such a situation, a modified version of the LMS algorithm, called the filtered-X algorithm was developed firstly for adaptive control and subsequently for ANC. The filtered-X algorithm requires that the combined transfer function of the above elements be evaluated and the resulting model used to filter the $\mathbf{X}$ input to the noise cancelling LMS weight update calculation [5], [2]:

$$\mathbf{W}(k+1) = \mathbf{W}(k) - 2\mu\epsilon(k)\mathbf{F}(k) \qquad (1)$$

where:

$\mathbf{F}(k) = [\, \mathbf{C}^T\mathbf{X}(k) \;\; \mathbf{C}^T\mathbf{X}(k-1) \;\ldots\; \mathbf{C}^T\mathbf{X}(k-N+1)\,]^T$

$\mathbf{C} = [\, c_0 \;\; c_1 \;\ldots\; c_{J-1}\,]^T$

$\mathbf{X}(k-i) = [\, x(k-i) \;\; x(k-i+1) \;\ldots\; x(k-i-J+1)\,]^T$

$\mathbf{F}(k)$ is known as the filtered-X vector, and $\mathbf{C}$ represents the J weights of the combined transfer function model, $\mathbf{C}(z)$. With a single error microphone, noise attentuation can be achieved within an approximate diameter of 1/10th of the noise wavelength from the microphone. To obtain noise reduction over a wider area, multiple error microphones and secondary sources must be used. A multichannel extension of the filtered-X algorithm was developed by Elliott et al [3], such that:

$$\mathbf{W}_m(k+1) = \mathbf{W}_m(k) - 2\mu \sum_{l=1}^{L} \epsilon_l(k)\mathbf{F}_{m,l}(k) \qquad (2)$$

$m : 1 \ldots M$; M = number of secondary sources
$l : 1 \ldots L$; L = number of error sensors
$\mathbf{W}_m(k)$ = weight vector of adaptive FIR controlling output of secondary source m
$\epsilon_l(k)$ = output of error sensor l
$\mathbf{F}_{m,l}(k) = [\, \mathbf{C}_{m,l}\mathbf{X}(k) \;\ldots\; \mathbf{C}_{m,l}\mathbf{X}(k-N+1)\,]^T$
= noise reference filtered by model of system from source m to error sensor l
$\mathbf{C}_{m,l} = [\, c_0 \;\; c_1 \;\ldots\; c_{J-1}\,]^T_{m,l}$
= weight vector of model of system from secondary source m to error sensor l

Figure 1 shows the architecture of the multichannel filtered-X algorithm and overall set-up for the system developed in this paper. The FIR filter models $\mathbf{C}_{m,l}$ are found using LMS system identification [3], [5] prior to the overall active noise control program being run. If the combined transfer function changes significantly during ANC then the models should be recalculated.

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3. ANC PERFORMANCE

In this section we report on the performance of the ANC system for various parameter settings. and when used in two differing acoustic environments: (1) a large office area of dimension of 7 × 6 metres; and (2) a stone walled corridor, of dimension 15 × 4 metres and a reverberation time of almost 10 seconds. In both cases the microphones were about 0.5 metres apart and the secondary noise sources were within 2 metres of the error microphones and within 1 metre of the primary noise source.

Figure 2(a) shows the 3-D spectrogram (normalised to the range 0-50dB) of uncancelled noise at one of the error microphones in the *office* when tones ranging from 65 to 500Hz were generated by a loudspeaker noise source connected to a Yamaha synthesizer outputing a constant peak to peak voltage into the first stage audio amplifier. The shape of the response is clearly due to the combined transfer function of the microphones, amplifiers, loudspeaker, room acoustics etc, and the trace of the peaks gives the (linear) frequency response. (Note that around the 200Hz frequencies there are sizeable non-linearities due to loadspeaker resonance.) Comparing the frequency response read from the 3-D spectrogram matches well with the combined transfer function, $\mathbf{C}(z)$, generated by the initial LMS system identification phase.

3.1 Sampling Rate

Identical tests were carried out for cancellation of tonal noise at sampling rates of 1000Hz, 2000Hz and 4000Hz. The frequency range of interest is 65-500Hz, and therefore 1000Hz is only just above the Nyquist rate. Test results clearly showed that $f_s = 2000$ performed significantly better than $f_s = 1000$. Surprisingly the $f_s = 4000$Hz system did not perform appreciably better, and therefore the remainder of the tests were performed with $f_s = 2000$Hz in order to allow longer FIR filter lengths to be used in the system. For the $f_s = 2000$Hz system 3rd order Butterworth filters with a cut-off of 723Hz were used for anti-aliasing and reconstruction. Filters cutting off above this frequency, and those of a lower order impaired the noise cancellation process by allowing the quantisation steps through (resulting in unwanted, interfering harmonics).

3.2 Filtered-X Algorithm Parameters

Figure 2(b) shows the office environment 3-D spectrogram at one of the microphones after ANC at each of the sweep frequencies for system identification model lengths of 80 taps, and noise cancelling filters of length 80 taps (denoted as test *Off-80id-80canc*). The *ANC-power spectrum* (ANC-PS) of the system can be demonstrated by calculating the dB noise reduction at the individual tones (subtracting cancelled from uncancelled peaks). Figure 4 shows the ANC-PS for various values of system identification and noise cancelling filter length. Note that in all tone tests, adequate time was given to allow the system to adapt to a minimum.

From Figure 4 it is clear that the long noise cancelling filter lengths do not significantly improve the system performance. This is not surprising for tone tests, as in theory a 2 tap filter can be used to cancel a single sine wave. However, when the (system identification) combined transfer function FIR length was reduced the system began to fail. The linear

ANC Spec.	Single Tone (dB reduction)				Multiple Tone (db Reduction)			
Freq. (Hz)	100	200	300	400	100+200+300+400			
80id-80canc	18	15	16	23	21	15	16	26
80id-60canc	15	9	9	16	20	13	30	31
80id-40canc	15	16	4	13	21	7	14	14
40id-80canc	15	5	7	14	22	13	8	32
40id-60canc	15	8	12	9	26	14	12	32
40id-40canc	10	8	12	7	13	6	12	12

Table 1: *Single and Multiple Tone ANC Comparison.*

ANC-PS of *Off-40id-20canc* has poor performance at very low frequencies, and test *Off-20id-80canc* actually enhances the noise at frequencies below 100Hz, however above 100Hz it shows the same characteristic ANC-PS as higher filter order implementations. When the system model lengths were reduced below 20 taps, and the response began to approach the transport delay of the system (i.e. delay primarily due to the acoustic transmission from secondary source to microphone) complete failure occurred (instability).

Tests were also performed in the corridor using 80 taps for the system identification and 80 taps for the noise cancellation *Cor1-80id-80sys*) as show in Figure 3. ANC in this reverberant environment although possible, often went unstable unless the filtered-X LMS step size was kept very small. In general it was very difficult to get stable performànce below 80 tap system identification filter lengths. *Cor2-80id-80sys* shows the ANC-PS for a step size reduced by an order of magnitude over *Cor1-80id-80sys*. Although the ANC was improved the system still did not perform well below 80 taps. The shape of the ANC-PS for office and corridor are *similar*, suggesting that the system identification frequency models have more to do with the transfer functions of the amplifiers, loudspeaker etc, than with the actual room acoustics.

3.3 Multiple-Tone Performance

Table 1 shows the results of various FIR lengths for single tones of 100Hz, 200Hz, 300Hz and 400Hz. and also for the composite signal consisting of the 4 tones (with the same magnitude) played simultaneously. In almost every case the tones in the composite signal are reduced by a **larger** amount than the single tones. The performace is shown to be non-linear, unpredictable, and unfortunately inconclusive.

3.4 Real Signal ANC

The system discussed on this paper has also been demonstrated to work with recorded electrical transformer noise, vacuum cleaner noise, and car engine noise. Using a sound pressure level meter directly at the error microphones, the transformer noise was reduced by about 15dB, the car engine noise by 12dB, and the vacuum cleaner noise by 10db. All tests took place in a static *office* environment. Figure 5 shows the transformer noise power spectrum before and after ANC.

3.5 Qualitative Hearing Tests

The testbed system is set up such that a *quiet chair* is placed near the error microphones to allow qualitative testing by the human ear. Testing with sine wave tones does not provide a particularly good qualitative test. The perceived loudness of a single tone with the same SPL level or spectral density as a signal of bandwidth 1/6 of an octave is much lower. Hearing perception tests were performed with the tone sweeps discussed above, and when expected a reduction in loudness was always perceived, however it was very difficult to quantify the reductions between tones of differing frequency. More impressive tests were performed by setting up the noise source to a be narrow band of tones, with an uncancelled SPL of about 115dB (uncomfortable) at the error microphones. Two multiple tone (MT) sets were used: MT-Test-A: *98Hz + 110Hz + 123Hz + 130Hz* MT-Test-B: *98Hz + 246Hz + 294Hz + 390Hz*

The ANC system reduced MT-Test-A by a measured 18dB which was perceived as a considerable reduction at the human ear. (A subjective attribute of normal hearing is that a reduction of 10dB in a sound will roughly sound half as loud.) This level of performance was achieved using noise cancelling FIR filter lengths of only 14 taps; increasing the number of taps up to 128 provided no increase in performance. MT-Test-B gave a reduction of around 26dB again using only 14 taps. Going below 14 taps severly degraded the system performance in both cases.

Tests were performed where two speakers were speaking within 1 metre of the error microphones. During the qualitative testing with MT-Test-A noise sources no significant degradation in the noise reduction was perceived. The *quiet chair* listener did however perceive the speech with a very small echo and *electric* sound, however not enough to interfere with intelligibility.

Using MT-Test-A and MT-Test-B signals, tests were performed for an empty *office*, and then repeated (without changing the combined transfer function models $\mathbf{C}(z)$) when 10 people all stood within 2 metres of either the noise source or the error microphones. The performace of the ANC was not significantly degraded and no great difference was perceived by the quiet chair listener.

4. CONCLUSION AND FUTURE WORK

This paper has reported on the performance of an active noise canceller designed around the DSP56001 processor, and interfacing to the acoustic world with standard *consumer* audio components. The system can provide very impressive reduction in perceived noise levels, however it is unpredicatble and frequently non-linear. A number of the practicalities of the ANC system design have been discussed and their importance in the construction of a working system has been demonstrated. The current focus of research is in performing a practical comparison of the filtered-X LMS with RLS and filtered-U algorithms for ANC, and also looking into techniques for feedback suppression from secondary sources to primary microphone.

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5. REFERENCES

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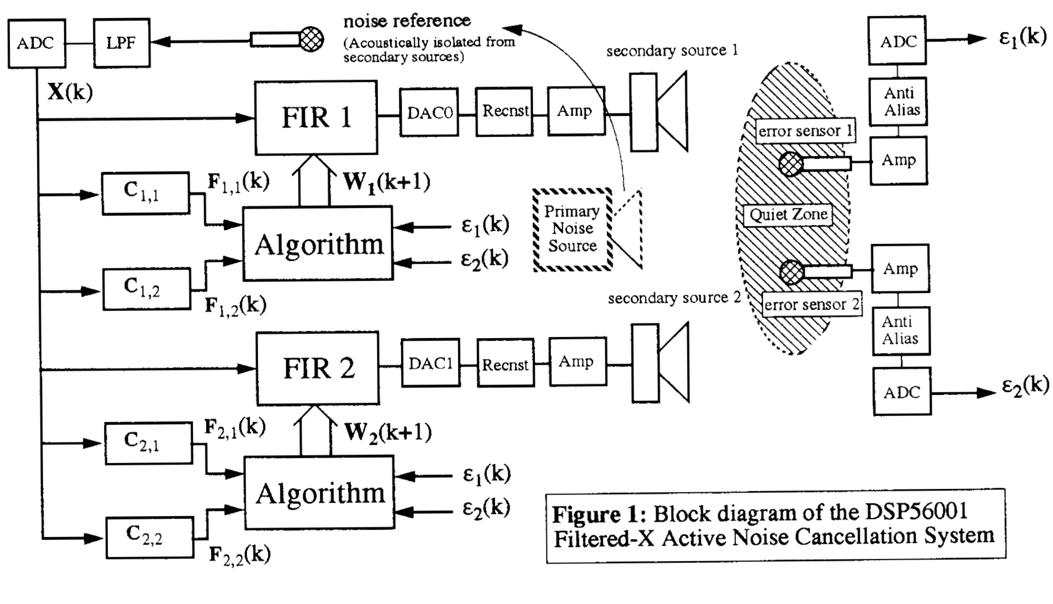

Figure 1: Block diagram of the DSP56001 Filtered-X Active Noise Cancellation System

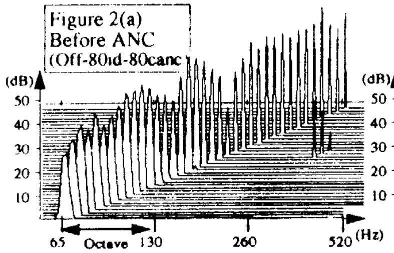

Figure 2(a)
Before ANC
(Off-80id-80canc)

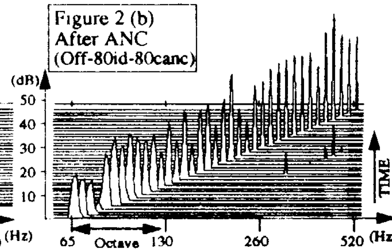

Figure 2 (b)
After ANC
(Off-80id-80canc)

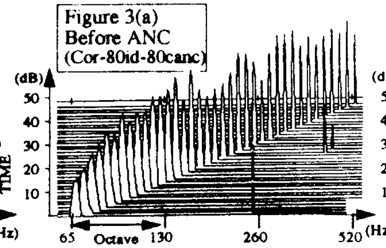

Figure 3(a)
Before ANC
(Cor-80id-80canc)

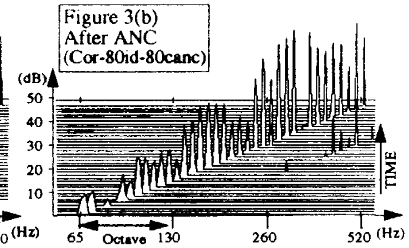

Figure 3(b)
After ANC
(Cor-80id-80canc)

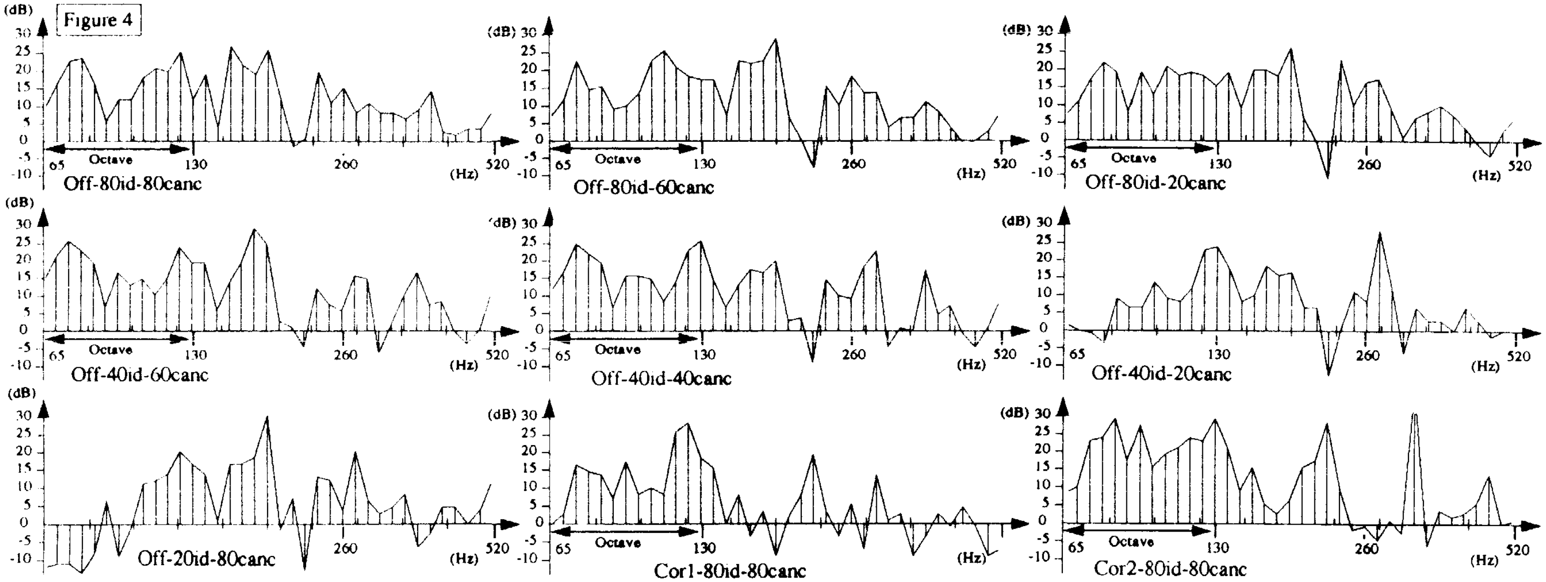

Figure 4

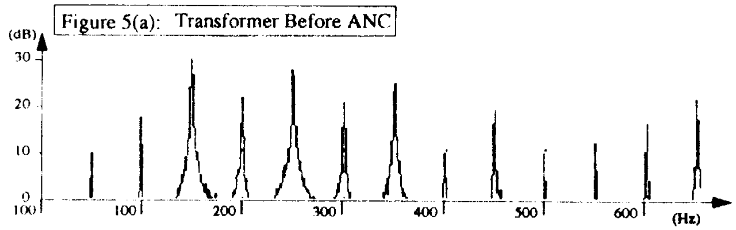

Figure 5(a): Transformer Before ANC

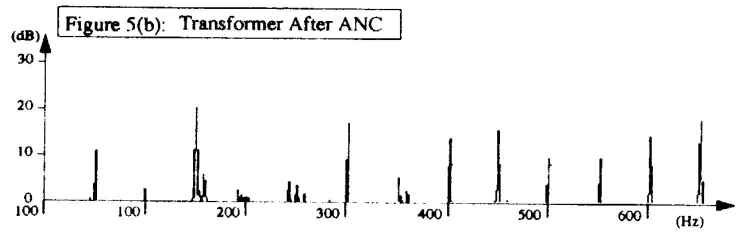

Figure 5(b): Transformer After ANC