ACTIVE NOISE CONTROL FOR HEARING PROTECTION USING A LOW POWER FIXED POINT DIGITAL SIGNAL PROCESSOR

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ABSTRACT

This contribution presents a fixed point implementation for acoustical active noise control in hearing defenders. The well known filtered-x least mean squares structure is conformed to fixed point arithmetic and evaluated in real time. The measured performance of the implementation is $20dB$ to $30dB$ attenuation of broad band noise and ca $60dB$ for sinusoidal interference. The implementation uses a low power fixed point digital signal processor and is well suited for industry application.

1. INTRODUCTION

Sound and noise emitted at a working site is often at levels hazardous to human hearing. Enduring noise might cause irreversible damage to the hearing and impair the overall health of working site personnel. Workers are thereby required (occasionally by legislation) to use hearing protection. Traditional passive circumaural ear defenders give good attenuation in the upper frequency range but for lower frequencies (below $500Hz$) the passive protection is inadequate (see Fig. 6) and active methods are necessary [1, 2]. In Active Noise Control (ANC) signals are actively emitted such that the sound pressure level is minimized within the ear cups. Hence, an ANC implementation require sensors measuring the reference signal and the error signal (the signal to be minimized). One or more supplementary exciters are also required for producing the noise canceling signal. Analog control systems are predominant today since they introduce minimal delays in the control loop as well as they are often highly cost effective. However, analog solutions tend to be static and reconfigurable digital solutions based on Digital Signal Processors (DSP) are desired. A basic setup of ANC using a DSP in an ear defender is illustrated in Fig. 1. DSP of today can use either floating point or fixed point arithmetic. An implementation using fixed point arithmetic require more attention as opposed to the floating point counterpart. However, the fixed point approach is likely to be more oriented towards industry since it is often more cost effective and less power consuming. In [3] an ANC implementation using a floating point processor is presented. Digital initiatives

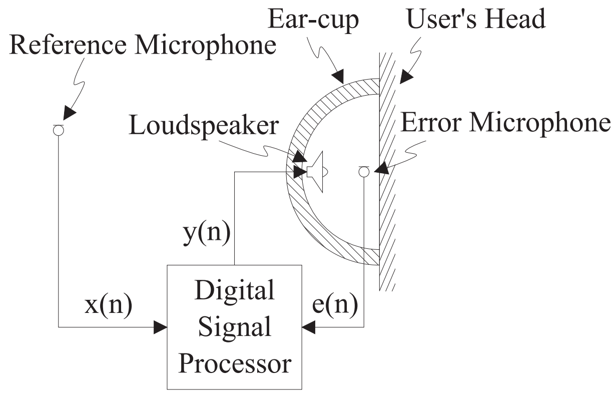

Figure 1: *Acoustical Active Noise Control (ANC) employing a Digital Signal Processor (DSP) for application in circumaural ear defender.*

for ANC are in many cases based on the popular Least Mean Squares (LMS) algorithm invented by Widrow et al. in 1956 [4]. Variants and applications of the original LMS structure exists in a bundle today [5, 6, 7], where the Filtered-X LMS (FXLMS) is a commonly used algorithm. The FXLMS compensates for the delays introduced in the physical medium between the exciter(s) and the error signal sensor(s).

This contribution presents and evaluates a fixed point acoustical ANC implementation for hearing protection in ear defenders using a low power digital signal processor.

The outline is as follows. Active noise control is theoretically outlined in Section 2 and a corresponding fixed point ANC formulation is given in Section 3. The experiment setup is given in Section 4 and Section 5 highlights important measured results. Finally, Section 6 summarizes this contribution and introduces future work.

2. ACTIVE CONTROL OF NOISE USING FXLMS

The basic setup illustrated in Fig. 1 is presented in detail in Fig. 2 where the primary path, $H(\omega)$, constitutes a transfer function from the reference microphone to the error

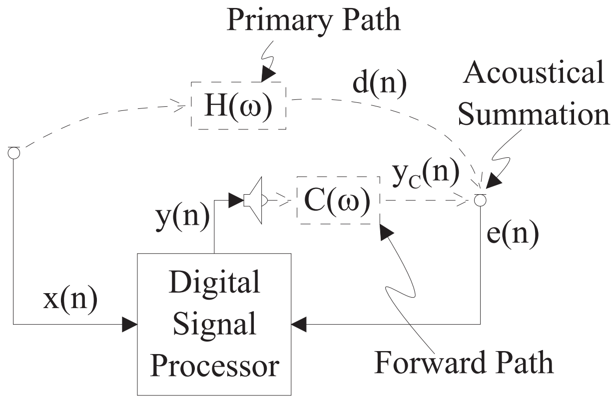

Figure 2: *Active control of noise in ear-muffs introducing a primary path, $H(\omega)$, and a forward path, $C(\omega)$.*

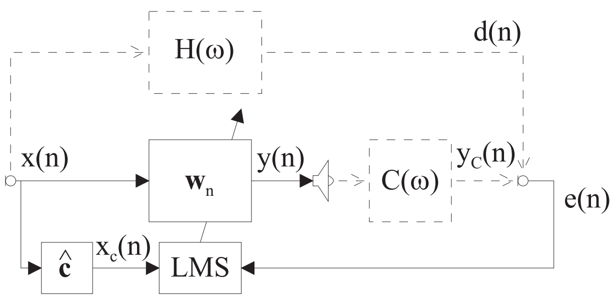

Figure 3: *Model of active noise control using the Filtered-X Least Mean Squares (FXLMS) structure where a forward path estimate, $\widehat{\mathbf{c}}$, compensates for the forward path.*

microphone. Similarly, the forward path, $C(\omega)$, denotes a transfer function from the input of the loudspeaker to the error microphone. The model assumes a linear summation of two acoustical signals, the noise signal and the anti-noise signal, in the error microphone. Due to the damping of the ear-muffs we assume that the path from the loudspeaker to the external reference microphone can be neglected. The FXLMS approach is illustrated in Fig. 3 where a forward path estimate, $\widehat{C}(\omega)$ with impulse response $\widehat{c}(n)$, is introduced to compensate for the delays introduced by the forward path. Derivation of the FXLMS structure leads to the following expressions

$$y(n) = \sum_{k=0}^{K-1} w_n(k)x(n-k) = \mathbf{w}_n^T \mathbf{x}_n \qquad (1)$$

$$x_c(n) = \sum_{m=0}^{M-1} \widehat{c}(m)x(n-m) = \widehat{\mathbf{c}}^T \mathbf{x}_n', \qquad (2)$$

where the adaptive filter coefficients and forward path estimate are denoted

$$\mathbf{w}_n = (w_n(0), \ldots, w_n(K-1))^T \qquad (3)$$

$$\widehat{\mathbf{c}} = (\widehat{c}(0), \ldots, \widehat{c}(M-1))^T . \qquad (4)$$

The vectors $\mathbf{x}_n$ and $\mathbf{x}_n'$ contain present and former reference signal values and are of size K and M respectively. The adaptive filter coefficient update for the FXLMS structure is given according to

$$\mathbf{w}_{n+1} = \mathbf{w}_n + \mu e(n)\mathbf{x}_{c,n}, \qquad (5)$$

where $\mathbf{x}_{c,n} = (x_c(n), \ldots, x_c(n-(K-1)))^T$ and μ is the algorithm step size. The forward path estimate, $\widehat{\mathbf{c}}$, is estimated prior to active noise control by using a supplementary system identification LMS structure. It is worth mentioning that the robustness to forward path estimate errors is high in the FXLMS structure since the adaptive filter, $\mathbf{w}_n$, and the forward path estimate, $\widehat{\mathbf{c}}$, interacts.

3. FIXED POINT FXLMS IMPLEMENTATION

The proposed implementation uses a fixed point processor for implementing the FXLMS structure. The processor uses 16 bit arithmetic for calculations but has extended capabilities of up to 48 bit Multiply and accumulate and 32 bit Multiplications.

Two main parts are identified in the FXLMS structure: Filtering and Adaptation. The adaptation part is the more sensitive of the both, since to low values could lead to stalling in the algorithm, i.e. the filter coefficient update halts, and this is undesirable. To minimize the risk of stalling the adaptive filter coefficients are stored in an extended 32 bit format according to

$$\mathbf{w}_n = 2^{16}\mathbf{w}_n^{hi} + \mathbf{w}_n^{lo}, \qquad (6)$$

where each component of $\mathbf{w}_n^{hi}$ and $\mathbf{w}_n^{lo}$ is of 16 bits. The full 32 bits are used in the adaptation phase but only the higher 16 bits are used in the filtering phase. The filtering part using the extended coefficient format can be expressed as

$$y(n) = \left\lfloor 2^{-8} \sum_{k=0}^{K-1} w_n^{hi}(k)x(n-k) \right\rfloor = \left\lfloor 2^{-8} {\mathbf{w}_n^{hi}}^T \mathbf{x}_n \right\rfloor, \qquad (7)$$

where the floor-operator $\lfloor\ \rfloor$ in conjunction with the constant, 2^{-8}, extracts the middle word, i.e. bit 8 to bit 23, of a 32 bit number. By using the floor-operator the high word of the filter coefficients can virtually be expressed as $2^{-8}\mathbf{w}_n^{hi}$, hence the filter coefficients are in the interval

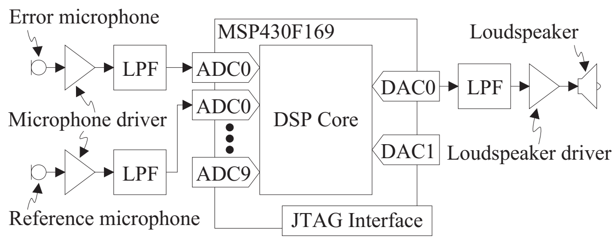

Figure 4: *Two channel in-two channel out prototype circuit board. (LPF denotes low pass filter.)*

$w_n^{hi}(k) \in [-(2^7 - 2^{-8}), 2^7]$ with the resolution 2^{-8}. The adaptive filter coefficient update is according to

$$\mathbf{w}_{n+1} \quad = \quad \mathbf{w}_n + \mu_q e(n)\mathbf{x}_{c,n}, \tag{8}$$

where the fixed point adaptive filter step size is $\mu_q = \lfloor 2^8 \mu \rfloor$ hence the actual step size used is $\mu = 2^{-8}\mu_q$.

4. EXPERIMENTAL SETUP

For evaluation of the fixed point FXLMS implementation, a Printed Circuit Board (PCB) is manufactured. A block diagram of the PCB is given in Fig. 4. The PCB supports two input channels and two output channels. A micro controller from Texas Instruments (MSP430F169) is situated on the PCB. The maximum clock rate of the processor is $8MHz$ with an average execution rate of about two Million Instructions Per Second (MIPS). The micro controller has eight 12 bit Analog-to-Digital Converters (ADC) of type Successive Approximation Register (SAR) and dual 12 bit Digital-to-Analog Converters (DAC) on board. Properly configured 10th order low pass filters from Linear Technology (LTC1569-6) are situated prior to ADC and after DAC for aliasing reduction and signal reconstruction. The input and output signal dynamic range is ± 1.25 V. Microphones are powered by external microphone amplifiers, likewise is the anti-noise loudspeaker using an external driver. The microphones and loudspeaker are positioned in the ear-cup according to Fig. 5. The passive attenuation performance of the ear-muff is illustrated in Fig. 6 where the attenuation is greater than $20dB$ for frequencies above $500Hz$. Hence, the sampling frequency of the experiment is set to $F_s = 1000Hz$. The maximum number of filter coefficients for the adaptive filter and the forward path estimate is 48 in the current platform.

5. RESULTS

From theory we know that after convergence of the FXLMS algorithm the filter chain: Adaptive filter and forward path should preferably equal the primary path with opposite sign, i.e. $W_{opt}(\omega)C(\omega) = -H(\omega)$, where $W_{opt}(\omega)$ is the fourier transform of the adaptive filter coefficients after convergence. In Fig. 7 the filter chain is presented after convergence. Power spectral density of the error microphone signal is presented in Fig. 8 with and without the control algorithm applied in a broad band noise case. Corresponding power spectrum is given in Fig. 9 for suppression of tonal interference (sinusoidal at $200Hz$).

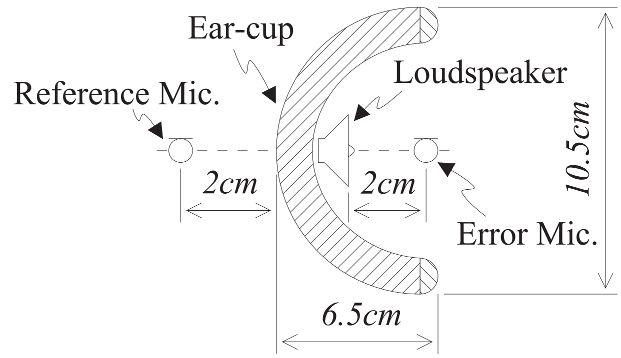

Figure 5: *The reference microphone, loudspeaker, and error microphone positioned in the ear defender.*

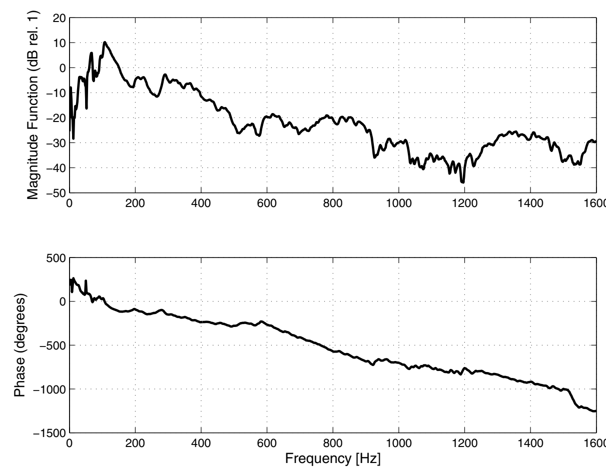

Figure 6: *Transfer function estimate of the primary path, $H(\omega)$, from reference to error microphone.*

6. SUMMARY AND FUTURE WORK

This contribution shows the derivation of the FXLMS structure for implementation on a fixed point processor running on a few MIPS only. The implementation is evaluated online and the performance of the fixed point FXLMS is $20dB$ to $30dB$ in the frequency interval $100Hz$ to $375Hz$ for broad band noise and ca $60dB$ for tonal interference (sinusoidal at $200Hz$).
The implementation is suitable for use in active hearing

protection since the processor used is of low power yet with high performance.

Future work include applying the algorithm in other than acoustical ANC. One application for ANC is active control of noise and vibrations in a lathe [8]. In adaptive feedback control of tool vibration this fixed point FXLMS controller is likely to be suitable. It is also interesting to further elaborate on combinations of analog and digital control where main features of the two domains are utilized.

7. REFERENCES

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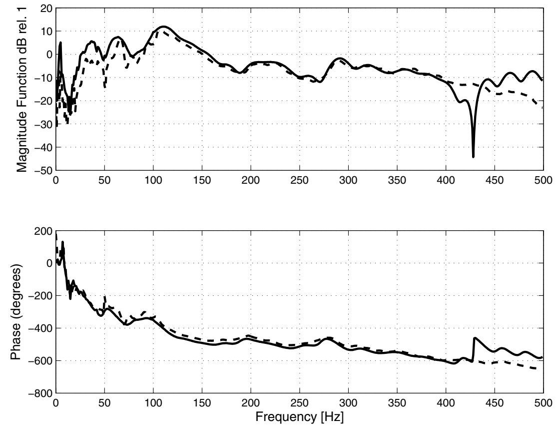

Figure 7: *Filter chain of the converged adaptive filter and the forward path (solid), i.e. $W_{opt}(\omega)C(\omega)$, and the primary path (dashed), $-H(\omega)$.*

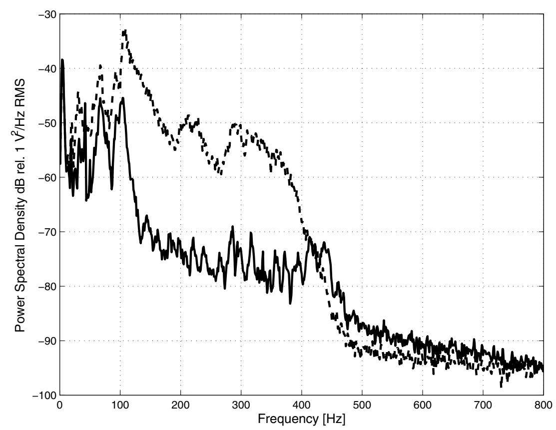

Figure 8: *Power spectral density of the error signal without (dashed) and with (solid) active noise control. Broad band noise.*

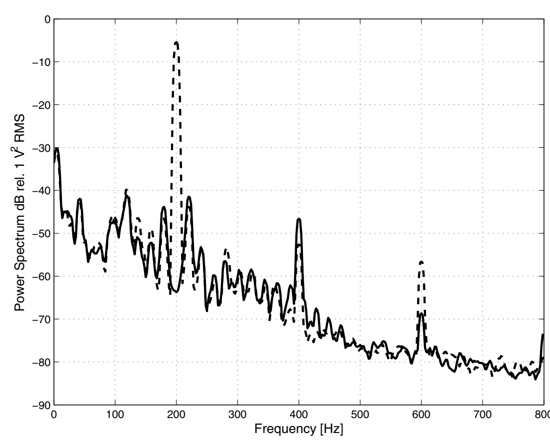

Figure 9: *Power spectrum of the error signal without (dashed) and with (solid) active noise control. Tonal interference sinusoidal at 200Hz.*