Block frequency domain implementation of feedback active noise control algorithm

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***Abstract*—Active noise control is a means of controlling the acoustic noise in real-time by generating antinoise. Therefore, computational complexity of an ANC algorithm plays a vital role in its implementation. The feedback ANC computes an extra secondary path estimate compared to the feedforward ANC and hence has more computational burden compared to the feedforward ANC. In this paper, a new frequency domain ANC algorithm is proposed that is computationally efficient compared to the standard time domain feedback ANC algorithm. The algorithm is made delayless and is suitable for real-time implementation.**

Keywords— Active noise control; Feedback ANC; Fast Fourier Transform

I. INTRODUCTION

Active noise control [1-4] is a real-time algorithm. Therefore, computational complexity of an ANC algorithm plays a vital role in its implementation. In feedback ANC [5-6], there is no reference microphone. The reference input is estimated from the error microphone signal. Therefore, the feedback ANC computes an extra secondary path estimate compared to the feedforward ANC. Therefore, the feedback ANC has more computational burden compared to the feedforward ANC. When the length of the secondary path estimate filter is large, this complexity is significant. This paper is devoted to the design of a new ANC algorithm that is computationally efficient compared to the standard time domain feedback ANC algorithm. Frequency domain algorithms are computationally efficient than the time domain convolution and correlations as the former exploits the Fast Fourier Transform (FFT) [7-9,12]. In a feedback ANC algorithm there are three linear convolution and one cross-correlation operations. All these four operations can be implemented in frequency domain, but if it is done so, it will have delay of one block length which would restrict the ANC for its real-time implementation. Therefore, two of the linear convolutions are computed in time domain to make the ANC delay-less [10,11]. In this process, the secondary path filtering to estimate the reference signal and the ANC adaptive filter used to generate the antinoise are implemented in time domain. The secondary path filtering of the reference signal to compute the error gradient and the cross-correlation operation between the filtered reference signal and the error signal is computed in block frequency domain.

The paper is organized as follows. Section II presents the proposed delay-less algorithm. Section III deals with the computational complexity analysis. Section IV. deals with the exhaustive computer simulation. Section V. concludes the paper.

II. PROPOSED DELAY-LESS BLOCK FREQUENCY DOMAIN FEEDBACK ANC ALGORITHM

In the proposed delay-less block ANC as shown in the Fig. 1, the error signal $e(n)$ is measured as follows.

$$e(n) = d(n) + y(n)\,, \tag{1}$$

where $d(n)$ is the noise at the cancellation point and $y(n)$ is the output of the secondary path filter with $s(n)$ as the impulse response and is expressed as follows.

$$y(n) = s(n) * [x_{est}(n-1) * w(n)]\,, \tag{2}$$

where, $w(n)$ is the impulse response of the ANC adaptive filter $W(z)$, and $x_{est}(n)$ is the estimated reference input which is computed from the error signal as follows

$$x_{est}(n) = e(n) - \hat{y}(n)\,, \tag{3}$$

where, $\hat{y}(n)$ is the output of the secondary path estimate with input as $[x_{est}(n-1) * w(n)]$, and is expressed as follows.

$$\hat{y}(n) = \hat{s}(n) * [x_{est}(n-1) * w(n)]\,, \tag{4}$$

where, $\hat{s}(n)$ is the impulse response of the secondary path estimate. In the conventional time domain feedback ANC, the weights of the ANC adaptive filter are updated as follows.

$$w(n+1) = w(n) - \mu X' e(n)\,, \tag{5}$$

where

$$X' = [x'(n), x'(n-1), \ldots, x'(n-N+1)]$$

and

$$x'(n) = x_{est}(n) * \hat{s}(n)\,. \tag{6}$$

and N is the length of the adaptive filter with impulse response $w(n)$.

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In the proposed block frequency domain feedback ANC algorithm, (5) and (6) are implemented in frequency domain and (2.2) and (2.4) are computed in time domain. This makes the ANC delay-less [8,9] in its operation. However, the weights of the ANC are updated after every N^{th} sample time. In this algorithm, the filter length of the ANC filter as well as all the estimated acoustic path transfer functions are assumed to be $N = 2^m$, where m is a definite positive integer.

Here the two signal vectors $\mathbf{x}(n)$ and $\mathbf{e}(n)$ are defined as follows

$$\mathbf{x}_{est}(n) = [x_{est}(n)\ x_{est}(n-1)\ \ldots x_{est}(n-N+1)] \text{ and}$$
$$\mathbf{e}(n) = [e(n)\ \mathrm{e}(n-1)\ \ldots\ \mathrm{e}(n-N+1)].$$

The impulse response vector of the secondary path estimate $\hat{S}(z)$ is converted into the frequency domain offline by appending a zero vector of length N and then taking their Fourier transforms which is expressed as follows.

$$\hat{S} = F[\hat{s}\quad 0_{\mathrm{N}}], \tag{7}$$

where, 0_{N} is a zero vector of size $1 \times N$ and $F[.]$ represents the fast Fourier transform operation.

To perform the convolution operation $x_{est}(n) * \hat{s}(n)$, the past and present blocks of vector $\mathbf{x}_{est}(n)$ of length N are to be concatenated and then the Fourier transform is computed as follows.

$$X_{est}(k) = F[\mathbf{x}_{est}(nk-N)\quad \mathbf{x}_{est}(nk)], \tag{8}$$

where, $\mathbf{x}_{est}(nk-N)$ and $\mathbf{x}_{est}(nk)$ are the past and present vectors of size $1 \times N$ each of the k^{th} block and $X_{est}(k)$ is the k^{th} block frequency domain estimated input signal vector of length $2N$.

Then the N point filtered output of the estimated reference input $\mathbf{x}_{est}(n)$ through the filter $\hat{s}(n)$ is computed for the k^{th} block as follows.

$$\mathbf{x}'(k) = \text{last } N \text{ points } \left\{F^{-1}[X_{est}(k) \otimes \hat{S}]\right\}, \tag{9}$$

where $\otimes$and $F^{-1}[.]$ represent the point by point multiplication and inverse fast Fourier transform operation respectively. The N point filtered output of the estimated reference signal is again concatenated with its previous block as

$$X'_{est}(k) = F[\mathbf{x}'(nk-N)\quad \mathbf{x}'(\mathrm{nk})], \tag{10}$$

The frequency domain error signal vector is computed by concatenating present and the past N samples each and taking the $2N$ -point fast Fourier transform

$$E(k) = F[\mathbf{e}(nk-N)\quad \mathbf{e}(\mathrm{nk})], \tag{11}$$

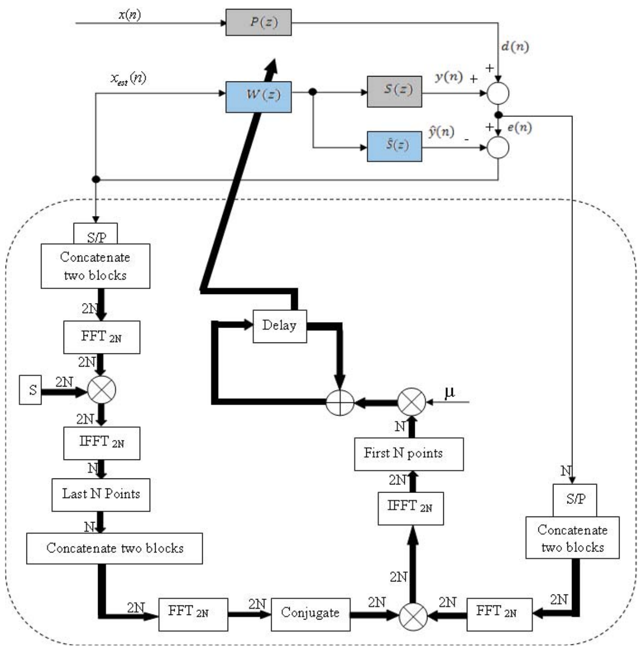


Fig. 1. Delay-less block frequency domain feedback ANC

The change in the time domain weights can be computed in frequency domain as

$$\Delta\mathbf{w}(k) =$$
$$\text{First } N-\text{points} \left\{F^{-1}[E(k) \otimes X'_{est}(k)^H]\right\} \tag{12}$$

where,

$X'_{est}(k)^H$ =conjugate of $X'_{est}(k)$.

The change in the weight $\Delta\mathbf{w}(k)$ is computed at every N samples and added to the ANC adaptive filter co-efficients

$$\mathbf{w}(k+1) = \mathbf{w}(k) - \nabla\mathbf{w}(k) \tag{13}$$

This algorithm is a delay less algorithm, where the ANC adapts at every N^{th} samples. The block diagram of the proposed algorithm is shown in Fig.1.

III. Computational Complexity Analysis

Computational complexity reduction is the major advantage of the proposed algorithm. Therefore, in this section, computational complexity analysis of the time domain and proposed frequency domain feedback ANC algorithms are presented. In this analysis it is assumed that the length of the ANC filter and the secondary path estimate filter are same and is equal to N. In this study, only multiplication count is considered. Equivalent number of additions are also involved which can be analyzed in a similar fashion.

In the time domain feedback ANC algorithms there are three N point convolution operations and one N point correlation operation. Each convolution and correlation takes N^2 multiplications for generating N samples. Therefore, there are $4N^2$ multiplications required for time domain feedback ANC to generate N samples of antinoise.

In the proposed frequency domain algorithm, two convolution operations are evaluated in time domain and hence involve $2N^2$ multiplications. As shown in Fig. 1, there are five $2N$ point FFTs and two $2N$ point complex multiplications are involved for the weight update part of the algorithm. The multiplication by μ is ignored as it can be done using shift operation by choosing its value a negative power of 2. Therefore, total multiplication count is $(2N^2 + 20N\log_2(2N) + 4N)$. To compare the computational complexity for various values of *N*, in both the algorithm, multiplication count required for generating N samples are plotted in Fig. 2. This shows that the proposed algorithm is computationally efficient for higher order of filter length (*N*).

IV. Simulations and Results

In this section, simulation experiments are conducted to check the effectiveness of the proposed model over the conventional one. For all the experiments, the primary path transfer function is considered to be $P(z) = 0.1 + 0.5z^{-1} + 0.3z^{-2} + 0.5z^{-3} - 0.2z^{-4} + 0.7z^{-5} - 0.3z^{-6}$. The secondary path taken was of order 128. The step size taken is 0.001. The order of the ANC adaptive filter is considered to be $N = 128$.

A. Experiment 1- Error convergence with block frequency domain feedback ANC

In this experiment, a multi-tone noise is taken as the primary noise with harmonics of 400 Hz (at 1200 Hz, 2000 Hz)and sampled at 8000 Hz and the error convergence is tested as shown in the Fig.3 (a). It is found that the proposed algorithm provides error convergence as that obtained with the conventional time domain feedback ANC. It is also observed that the unlike the conventional time domain feedback ANC, in the proposed block frequency domain feedback ANC , the ANC weights are getting updated after 128 samples (i.e. block length of the samples) as shown in the Fig. 3 (b).

B. Experiment 2- Weight updating in Block frequency domain feedback ANC

In this experiment, the simulation parameters are taken to be same as that taken in the previous Experiment except the frequency of the primary noise is changed to the harmonics of 200 Hz (at 600 Hz ,1000 Hz) after 10000 samples. Then the error convergence is plotted to ensure the stability of the proposed algorithm as shown in the Fig. 4 (a). It is found that even if the frequency of the noise signal is changed, the proposed algorithm is also able to adapt to these changes. The corresponding weight updating is shown in the Fig.4 (b).

V. Conclusions

In this paper, a block frequency domain feedback ANC algorithm is approached to reduce the computational complexity of the conventional time domain algorithms. The results established that the proposed algorithm reduces the computational complexity significantly and makes the ANC delay less.

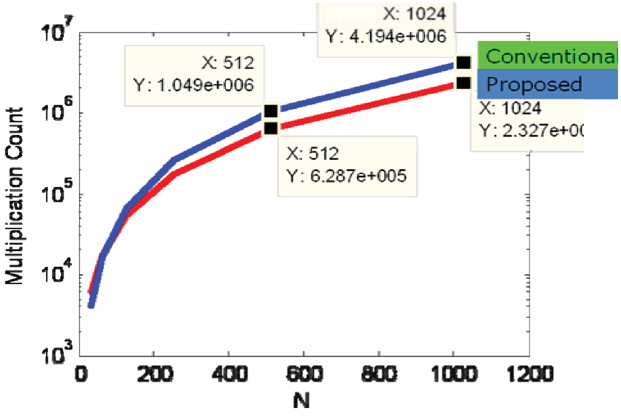
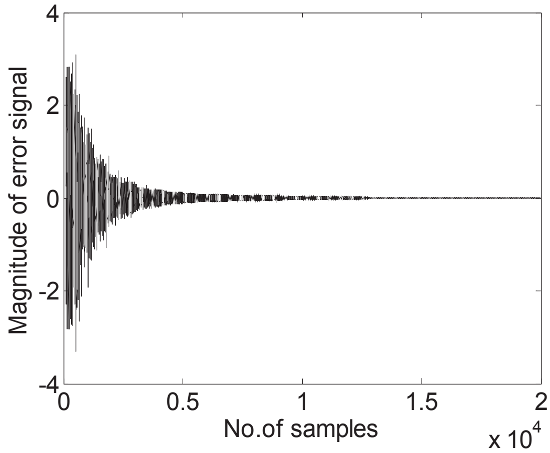
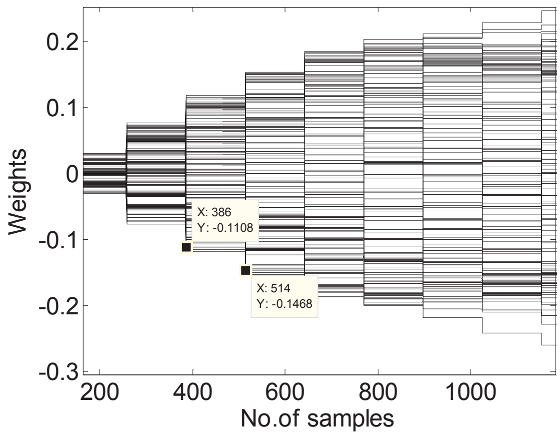


Fig. 2. Comparison of computational complexity



(a)



(b)

Fig. 3. Results of expermENT 1 (a) error convergence, (b) weight updating (zoomed) in block frequency domain feedback ANC.

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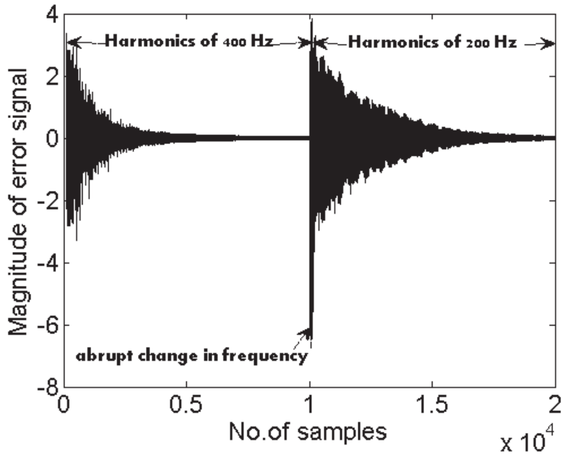
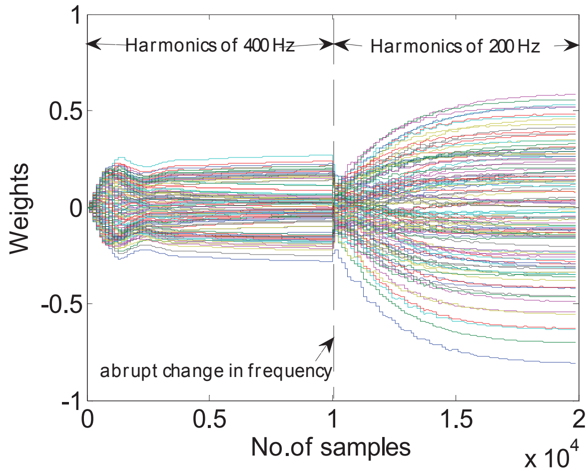


(a)



(b)

Fig. 4. Results of experment 2 (a) error convergence, (b) weight updating (zoomed) in block frequency domain feedback ANC.