ONLINE ACOUSTIC FEEDBACK PATH MODELING IN MULTI-CHANNEL ACTIVE NOISE CONTROL SYSTEMS USING VARIABLE STEP-SIZE ALGORITHM

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ABSTRACT

In this paper we have investigated the effects of acoustic feedback on multi-channel ANC system. The presence of strong acoustic feedback degrades the convergence speed of the ANC filters, and in worst case the multi-channel ANC system may become unstable. The acoustic feedback path can be modeled offline, but in most applications it is time varying, hence it needs to be modeled online when ANC is in operation. Here, we propose a *Variable Step Size* based method for multi-channel ANC systems with online acoustic feedback path modeling. The computer simulations are carried out, which demonstrate the effectiveness of the proposed method, in comparison with the existing methods for $1 \times 2 \times 2$ ANC system.

Index Terms— Adaptive filtering, multi-channel active noise control, acoustic feedback, online feedback-path modeling.

1. INTRODUCTION

Active noise control (ANC) is achieved by introducing a canceling "antinoise" signal through an appropriate array of secondary sources. These secondary sources are interconnected through electronic system using a specific signal processing algorithm for the particular cancellation scheme. The most commonly used ANC system is a single channel ANC system in narrow ducts. But to mitigate the effects of multi-modal noise in enclosures and large duct system, there is a need to use multi-channel ANC systems. A structure for $J \times K \times M$ multichannel ANC systems with J reference inputs, K secondary sources,and M error sensors is shown in Fig. 1.

In comparison to single-channel ANC systems, the complexity of multiple-channel ANC in a multi-dimensional space with many inputs and outputs is significantly higher. The multiple-error FxLMS (MeFxLMS) algorithm proposed in [1] requires compensation for the transfer function of each secondary path $S_{km}(z)$ between $y_k(n)$ and $e_m(n)$, where $k = 1, 2, \ldots, K$ and $m = 1, 2, \ldots, M$. Furthermore,the performance of the MeFxLMS algorithm is degraded due to the acoustic feedback by $J \times K$ feedback paths $F_{jk}(z)$ between $y_k(n)$, $k = 1, 2, \ldots, K$ and $x_j(n)$, $j = 1, 2, \ldots, J$. The secondary and feedback paths may be time varying and need to be identified during online operation [2].

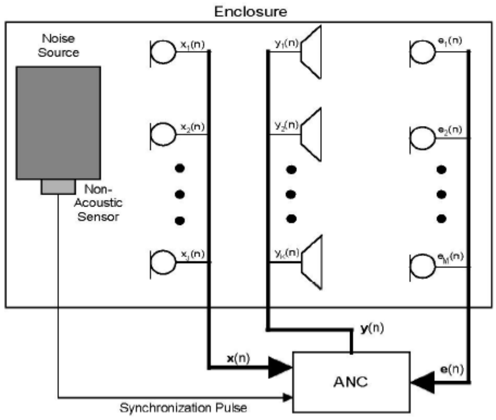


Fig. 1. *Structure of a multi-channel acoustic ANC system.*

This "antinoise" signal propagates downstream towards the error microphones and interferes destructively with the unwanted noise. However it also propagates upstream to the reference microphones and, hence corrupts the reference signal. This is called *acoustic feedback* [1]. For sake of simplicity, we assume that the secondary paths are exactly identified, and investigate various methods for online feedback path modeling (FBPM). In particular, we consider $1 \times 2 \times 2$ ANC system with only one reference signal.

The simplest approach for solving the feedback problem is to use a separate feedback path neutralization (FBPN) filter with in the controller, this method is called *acoustic feedback neutralization*. This electrical model of the feedback path is driven by the antinoise signal and its output is subtracted from the reference sensor signal. The FBPM filter, $\hat{F}_k(z)$, may be obtained offline and kept fixed during the operation of the ANC system. In many practical cases, however, $F_k(z)$ may be time varying. For these cases, online modeling of $F_k(z)$ is needed to ensure the convergence and stability of the MeFxLMS algorithm for multi-channel ANC systems [1].

The rest of paper is organized as follows. In Section 2, the existing methods for acoustic feedback path modeling in

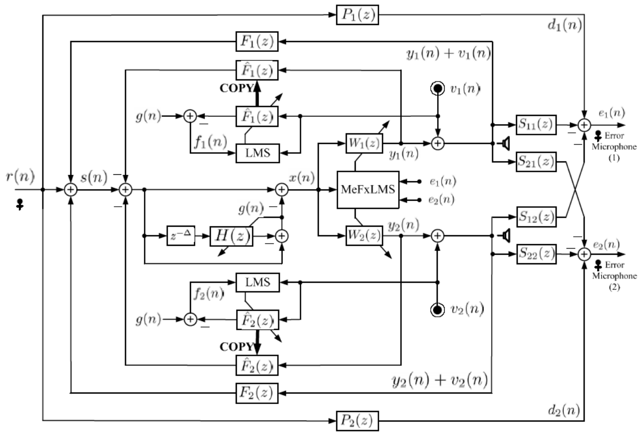


Fig. 2. *Kuo's method extension for online feedback path modeling in $1 \times 2 \times 2$ multi-channel ANC system [3].*

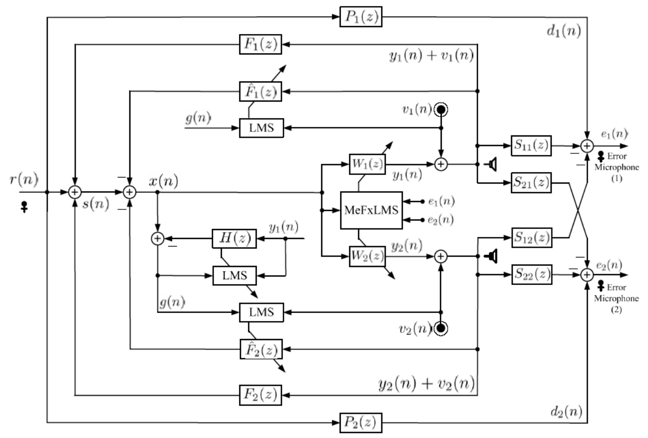


Fig. 3. *Akhtars's method extension for online feedback path modeling in $1 \times 2 \times 2$ multi-channel ANC system [3].*

multi-channel ANC system are summarized. The proposed method making use of variable step-size to improve the convergence speed of modeling filters is presented in Section 3. The simulation experiments conducted on existing methods for online modeling of acoustic feedback path in ANC systems are presented in Section 4 and finally paper is concluded in Section 5.

2. SUMMARY OF EXISTING METHODS

Initially the algorithms for acoustic feedback path modeling have been formulated for the single-channel ANC system as it offers simplicity and straightforward implementation. Therefore for the multi-channel ANC system, we will go through the extension made by the authors of the respective methods to $1 \times 2 \times 2$ ANC system.

The extension of Kuo's method [4] is shown in Fig. 2. Here $r(n)$ is the reference signal being picked by the reference microphone; $P_1(z)$ and $P_2(z)$ represent primary paths between noise source and two error microphones $e_1(n)$ and $e_2(n)$, respectively; $W_1(z)$ and $W_2(z)$ are two adaptive filters generating the canceling signals $y_1(n)$ and $y_2(n)$, respectively for two error microphones; $F_1(z)$ and $F_2(z)$ are two feedback paths between secondary loudspeakers and reference microphone; $S_{kj}(z)$ represent secondary path between k^{th} microphone $e_k(n)$, $k = 1, 2$ and j^{th} canceling signal $y_j(n), j = 1, 2$; $v_1(n)$ and $v_2(n)$ are the internally generated random noise signals that are uncorrelated with $d_1(n)$, $d_2(n)$, $y_1(n)$, and $y_2(n)$ [4].

In Kuo's method [4], the noise source is assumed to be predictable. The feedback path modeling (FBPM) is achieved by using additive-random noise based adaptive filters, $\hat{F}_1(z)$ and $\hat{F}_2(z)$, modeling two feedback paths, $F_1(z)$ and $F_2(z)$, respectively. After adaptation, the weights of FBPM filters are copied to feedback path neutralization (FBPN) filters $\hat{F}_1(z)$ and $\hat{F}_2(z)$, being excited by the outputs of two adaptive filters $W_1(z)$ and $W_2(z)$, respectively. A discrimination filter $H(z)$, based on a decorrelation delay, is used to predict the predictable component in the corrupted reference signal. The error signal of $H(z)$, $g(n)$ is used to generated reference signal $x(n)$ for the multiple-error FxLMS (MeFxLMS) algorithm for adaptation of control filters. The same error signal is used in adaptation of FBPM filters [4].

There are few problems with this method such as the performance depends on the proper choice of the decorrelation delay. In addition to that, it assumes predictable noise source, and for unpredictable noise sources the discrimination filter will not work, and hence the performance of the ANC system will be degraded.

To overcome the problems with the Kuo's method, the Akhtar's method [3] is shown in Fig. 3. It is evident that the structure of the Akhtar's method is somewhat simpler than the Kuo's method. As in Kuo's method, only additive random noise signal is used for online feedback path modeling. But here, the sum signals $[y_1(n) + v_1(n)]$ and $[y_2(n) + v_2(n)]$ are propagated by the canceling loudspeakers downstream to generate the error signals, $e_1(n)$ and $e_2(n)$ respectively. The sum signals are also passed through feedback path modeling and neutralization (FBPMN) filters, whose output is used to generate reference signal for the adaptation of control filters using MeFxLMS algorithm [3].

The generated reference signal $x(n)$ is also used as a desired response for the adaptive noise cancelation (ADNC) filter $H(z)$. The ADNC filter $H(z)$ is excited by $y_1(n)$, and its output converges to that part in $x(n)$ which is correlated with $y_1(n)$. The coefficients of $H(z)$ are updated by LMS algorithm. As $g(n)$ contains error signals for both FBPMN filters, thus gradient in LMS equations for $\hat{F}_1(z)$ and $\hat{F}_2(z)$ is computed using $g(n)$ as error signal and the random signal $v_1(n)$ and $v_2(n)$ as input signal, resulting in the LMS update equations for $\hat{F}_1(z)$ and $\hat{F}_2(z)$, respectively [3].

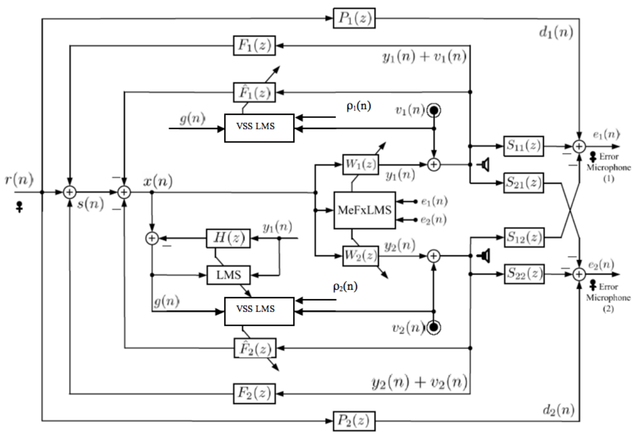


Fig. 4. *Proposed method for online feedback path modeling in $1 \times 2 \times 2$ multi-channel ANC system.*

3. PROPOSED METHOD

A block diagram of the proposed $1 \times 2 \times 2$ ANC system is shown in Fig. 4. As the multi-channel ANC system comprises of sets of control and feedback path modeling filters, a following phenomena is observed:

- Due to large disturbance in the beginning of operation of multi-channel ANC, as the canceling signals are zero; the convergence of the modeling filters is degraded. In worst case, the filters might become unstable.
- As $n \to \infty$, the canceling signals should converge to the desired signals $d_1(n)$ and $d_2(n)$ respectively and thus ideally the error signals should be zero.

These observations lead to a conclusion that in ANC systems, the proper selection of step-size for the adaptation is very important. A small value will reduces the chance of instability but the ANC system will overall take more time for convergence; on the other hand the higher value can make the ANC system unstable but can improve convergence. In [5], it was argued that for online modeling of secondary path $S(z)$. This was called as *Variable Step-Size algorithm*, it was initially designed for single-channel ANC system and extension of the same algorithm to $1 \times 2 \times 2$ ANC system was presented in [6].

Here the same concept has been applied to the adaptation of the FBPM filters, as the error signal for the online FBPM filter adaptation is corrupted by a disturbance that is decreasing in nature as well. This allows us to initially use a small step size, and when the ANC system starts converging we gradually increase the step size. The procedure to vary the step size is explained below. We define a ratio $\rho_k(n)$ as

$$\rho_k(n) = \frac{P_g(n)}{P_{e_k}(n)}, \tag{1}$$

where $P_g(n)$ and $P_{e_k}(n)$ are, respectively, the power of error signals $g(n)$ and $e_k(n)$.

These powers can be estimated by using low-pass estimators of the form

$$P_\gamma(n) = \lambda P_\lambda(n-1) + (1-\lambda)\gamma^2(n), \tag{2}$$

where γ is an arbitrary signal and λ is the forgetting factor $(0.9 < \lambda < 1)$.

The $P_g(n) = P_{[r(n)]} + P_{[y_{f1}(n)-\hat{y}_{f1}(n)]} + P_{[y_{f2}(n)-\hat{y}_{f2}(n)]} + P_{[v_{f1}(n)-\hat{v}_{f1}(n)]} + P_{[v_{f2}(n)-\hat{v}_{f2}(n)]} + P_{[u(n)]}$. Similarly, $P_{e_k}(n)$ can be expressed as:

$$P_{e_k}(n) = P_{[d_k(n)]} + P_{[y'_{k1}(n)-y'_{k2}(n)]} + P_{[v'_{k1}(n)-v'_{k2}(n)]} \tag{3}$$

At $n = 0$, when the ANC system is started, the canceling signals are zero, hence $\rho_k(n)$ at $n = 0$ will be given as, $\rho_k(0) \approx \frac{P_{[r(n)]} + P_{[v_{f1}(n)-\hat{v}_{f1}(n)]} + P_{[v_{f2}(n)-\hat{v}_{f2}(n)]}}{P_{[d_k(n)]} + P_{[v'_{k1}(n)-v'_{k2}(n)]}}$. Since the modeling signals are generated from a low-level random noise signal, and $d_k(n)$ is filtered version of $r(n)$, hence $\rho_k(0) \approx 1$. When the ANC system converges, then $P_g(n) \to zero$. This makes the numerator of the (1) converges to zero but the denominator is non-zero due to presence of the term $P_{[v'_{k1}(n)-v'_{k2}(n)]}$. Thus when the ANC system converges, $\rho_k(n)$ approaches 0.

Initially when $\rho_k(0) \approx 1$, we use a small step size and later we subsequently increase the step size in accordance with a decrease in $\rho_k(n)$. Thus the step size for the online FBPM filters $\hat{F}_1(z)$ and $\hat{F}_2(z)$ is calculated as:

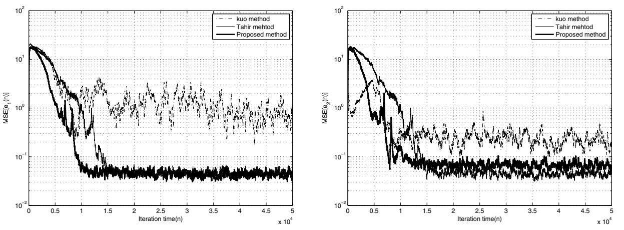
$$\mu_{f_k}(n) = \rho_k(n)\mu_{min} + [1 - \rho_k(n)]\mu_{max}, \tag{4}$$

where μ_{min} and μ_{max} are experimentally determined values for lower and upper bounds of the step size.

Now the online FBPM filters are adapted by using the variable step size computed from using (1), (2) and (4). The control filters are updated using the MeFxLMS algorithm. The structure of the proposed method is similar to that of Akhtar's method [3], except for the differences of a few more computations required in the variable-step-size LMS (VSS LMS) algorithms used for online FBPM modeling filters.

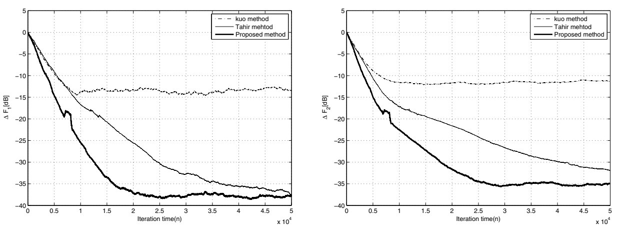
4. SIMULATION RESULTS

This section presents the simulation experiments performed to verify the effectiveness of the proposed method in comparison with the Kuos method and Akhtar's method for $1 \times 2 \times 2$ multi-channel ANC system. For acoustic paths, the experimental data provided by [1] is used. Using this data, primary, secondary and feedback paths are modeled as FIR filters of tap-weight lengths 48, 16, and 32, respectively. The tap-weight lengths of adaptive filters $W_1(z)$ and $W_2(z)$ are $L = 32$, of the FBPM filters $\hat{F}_1(z)$ and $\hat{F}_2(z)$ are $N = 32$, and that of $H(z)$ is 32.



(a) (b)

Fig. 5. *Mean square error curves for residual noise at error microphones (a) $e_1(n)$ and (b) $e_2(n)$ for tonal noise of* 300 *Hz.*



(a) (b)

Fig. 6. *Relative modeling error curves of (a)$F_1(z)$ and (b) $F_2(z)$ for tonal noise of* 300 *Hz.*

The reference signal is a tonal of 300 Hz. A sampling frequency of 4 kHz is used and a zero-mean white noise is added to the reference signal with SNR of 30 dB. The modeling excitation signal $v_1(n)$ is a zero-mean white Gaussian noise of variance 0.05. The modeling signal $v_2(n)$ is generated by delaying $v_1(n)$ by 32 samples. The simulations are carried out for 10 iterations.

The step size parameters are adjusted for fast and stable convergence, and, by trial-and-error, are found to be: $\mu_\omega = 2.5 \times 10^{-6}$, $\mu_h = \mu_d = 5 \times 10^{-4}$ for all methods and $\mu_f = 5 \times 10^{-3}$, $\mu_{min} = 4.5 \times 10^{-3}$ and $\mu_{max} = 7.5 \times 10^{-3}$ for existing and Proposed method respectively. The decorrelation delay Δ in Kuo's method is 30.

The simulation results of mean-square error for residual noise at error microphones are compared for all the methods. In addition to that, in order to demonstrate the performance of the algorithms in terms of online feedback-path modeling, the curves for relative modeling error ΔF being defined as $\Delta F(n) = (\| \mathbf{f}(n) - \hat{\mathbf{f}}(n) \| / \| \mathbf{f}(n) \|)$ are plotted for each algorithm.

The results representing the mean square error versus iteration n are shown in in Fig.5(a) and (b). We see that, the Kuo's method, Akhtar's method and Proposed method can stabilize the multi-channel ANC system, with Proposed method giving better performance. Fig. 6(a) and (b) show the relative feedback modeling errors for $F_1(z)$ and $F_2(z)$. It is evident that proposed method is better able to remove the acoustic feedback, than the Kuo's method and converges much faster than Akhtar's method.

5. CONCLUSIONS

In this paper, we have proposed a new method for $1 \times 2 \times 2$ ANC system with online feedback path modeling. The proposed method uses variable step-size algorithm in adapting the feedback path modeling filters. This method improves the convergence of the feedback path modeling filters, and hence overall performance of the multi-channel ANC system. The computer simulations demonstrate that the proposed method gives improved performance. At present, the complete analysis and validation of the proposed method is accomplished. The development of algorithm for a general $J \times K \times M$ multichannel ANC systems is a task of future work.

6. REFERENCES

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