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論 文 内 容 要 旨

Abstracts: **This thesis is study of active noise control (ANC) systems with the viewpoint of adaptive signal processing. We have proposed new signal processing methods to improve the performance of active noise control systems. We combine the concepts of the FxLMS algorithm and adaptive filtering with averaging, and propose a new algorithm (FxAFA filtered-x adaptive filtering with averaging algorithm) for ANC systems. The proposed algorithm outperforms the conventional FxLMS based ANC systems under the situation of large measurement noise. The effectiveness of the proposed algorithm in ANC systems with online secondary path modeling is also demonstrated. This thesis also proposes a new variable step size (VSS) LMS algorithm for online secondary path modeling in ANC systems. Here step size is varied on the basis of power of desired response of the modeling filter. The proposed VSS LMS algorithm, in contrast to the existing VSS algorithms, uses a small step size initially and later its value is increased in accordance with the decrease in the residual noise.**

I. INTRODUCTION

Active noise control (ANC) [1] is based on the simple principle of destructive interference of propagating acoustic waves. The most popular adaptation algorithm used for ANC applications is the FxLMS algorithm, which is a modified version of the LMS algorithm [2]. The schematic diagram for a single-channel feedforward ANC system using the FxLMS algorithm is shown in Fig. 1(a). Here, $P(z)$ is primary acoustic path between the reference noise source and the error microphone. The reference noise signal $x(n)$ is filtered through $S(z)$ and appears as a primary noise signal at the error microphone. The objective of the adaptive filter $W(z)$ is to generate an appropriate antinoise signal $y(n)$ propagated by the secondary loudspeaker. This antinoise signal combines with the primary noise signal to create a zone of silence in the vicinity of the error microphone. The error microphone measures the residual noise $e(n)$, which is used by $W(z)$ for its adaptation to minimize the sound pressure at error microphone. Here S^(z) accounts for the model of the secondary path $S(z)$ between the output of the controller and the output of the error microphone. The filtering of the reference signal $x(n)$ through is demanded S^(z) by the fact that the output of the adaptive filter $y(n)$ is filtered through $S(z)$ [2].

We have considered following problems in our study.

1) The convergence speed of the FxLMS algorithm is slow.
2) The performance of FxLMS based ANC system is degraded when there is a large measurement noise in the reference and error signals.
3) In practical ANC systems the secondary path, following the adaptive filter, may be time varying. So it is necessary to model its characteristics during the online operation of ANC system.

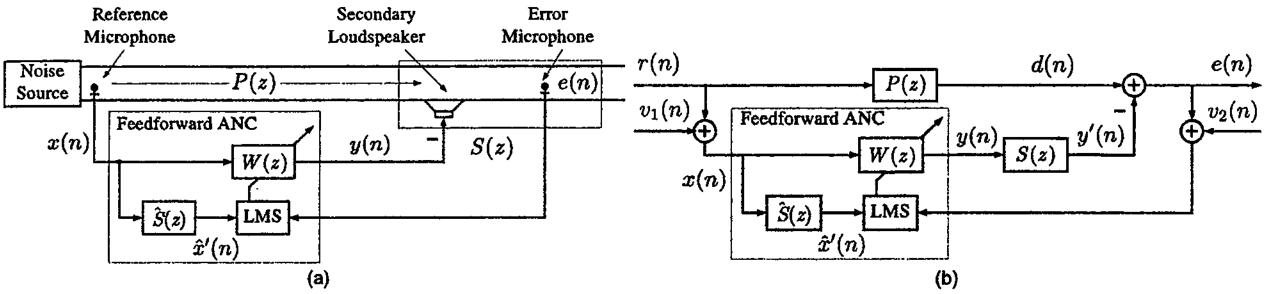

Fig. 1. FxLMS algorithm based single-channel feedforward ANC system. (a) Schematic and (b) block diagram.

To solve first two problems we have proposed a new filtered-x algorithm based on the concept of adaptive filtering with averaging (AFA). As compared with FxLMS algorithm, the proposed FxAFA algorithm gives somewhat faster convergence and achieves better performance in the presence of measurement noise. We have also proposed two new methods for ANC systems with online secondary path modeling. The main feature of these methods is that these can achieve improved performance by using two adaptive filters only. This is in contrast to existing methods, which use three adaptive filters.

II. FILTERED-X ADAPTIVE FILTERING WITH AVERAGING (FXAFA) ALGORITHM

Fig. 1(b) shows the block diagram of the feedforward ANC system of Fig. 1(a). Here, $v_1(n)$ and $v_2(n)$ are measurement noise signals associated with the reference and error microphones, respectively. We assume that 1) $v_1(n)$ and $v_2(n)$ are zero mean white Gaussian noise signals and 2) are uncorrelated with each other and with the reference and error signals as well. Here, 1) comes from the fact that $v_1(n)$ and $v_2(n)$ are produced by the turbulent air flow (random in nature) over the microphones [3], and 2) is evident from the nature of the system [Fig. 1(a)]. The FxLMS update equation for the coefficients of $W(z)$ is given as:

$$w(n+1) \quad = \quad w(n) + \mu\{e(n) + v_2(n)\}\hat{x}'(n)$$

We see that presence of the measurement noise $v_2(n)$ is frustrating the convergence of the FxLMS algorithm. It can be shown that for residual noise signal $e(n)$ to converge to zero, the control filter $W(z)$ should converge to the optimal transfer function:

$$W^{o}(z) = \frac{P(z)}{S(z)\{1 + V_1(z)/R(z)\}}$$

This equation shows that the optimal solution is independent of the measurement noise $v_2(n)$ associated with the error microphone. However the optimal solution for ideal case [$P(z)/\ S(z)$ for $v_1(n)=v_2(n)=0$] is distorted by the reference input measurement noise, $v_1(n)$.

Since we have assumed that the both $v_1(n)$ and $v_2(n)$ are white Gaussian noise signals, we can use *averaging* to remove their effects. We incorporate averaging with both the iterates, $w(n)$, and the correction term (the observation vector), $\mu\{e(n)+\ v_2(n)\}\ x'(n)$, and propose the filtered-x adaptive filtering with averaging (FxAFA) algorithm [4], as summarized below:

$$w(n+1) = \overline{w}(n) + \overline{g}(n)$$

where

$$\overline{w}(n) = \frac{1}{n}\sum_{k=1}^{n} w(k)$$

$$\overline{g}(n) = \frac{1}{n^{\gamma}}\sum_{k=1}^{n} \mu\{e(k) + v_2(k)\}\hat{x}'(k); 1/2 < \gamma < 1$$

Further details about the convergence properties of the algorithm, the comments on the choice of parameter γ, and simulation results can be found in [5, Chapter 2].

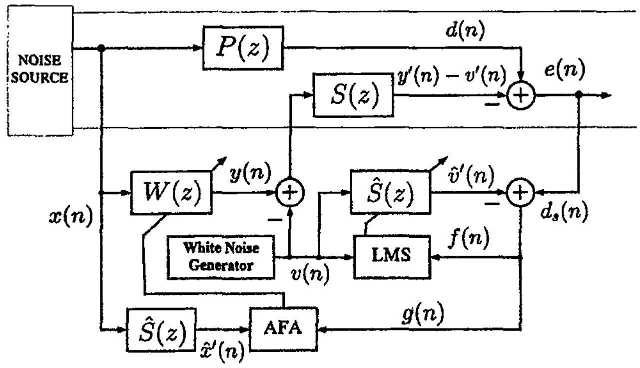

Fig. 2. Proposed-1 method for ANC systems with online secondary path modeling.

III. FxAFA Algorithm Based ANC System With Online Secondary Path Modeling (Proposed–1 Method)

Consider block diagram of Fig. 2, which is Proposed-1 method for ANC systems with online secondary path modeling. The main features of this method are summarized below.

1. It comprises two adaptive filters, $W(z)$ for noise control, and $S^\wedge(z)$ for secondary path modeling.
2. Both $W(z)$ and $S^\wedge(z)$ are updated using same error signal:
$$g(n) = f(n) = e(n) - \hat{v}'(n).$$
3. The $W(z)$ is adapted using FxAFA algorithm.

The main idea in this method is that if the control filter $W(z)$ is efficient in reducing the residual noise, it in turn improves the performance the modeling filter $S^\wedge(z)$ [6].

The detailed description and analysis of the method is presented in [5, Chapter 3]. The simulation results presented in [5, Chapter 5] show that this method achieves best noise reduction performance among the existing methods. The main limitation is its poor tracking properties, which is due to averaging based control filter.

IV. Modified FxLMS Algorithm Based ANC System With Online Secondary Path Modeling (Proposed–2 Method)

Consider block diagram of Fig. 3, which is Proposed-2 method for ANC systems with online secondary path modeling. The main features of this method are summarized below.

1. It comprises two adaptive filters, $W(z)$ for noise control, and $S^\wedge(z)$ for secondary path modeling.
2. The control filter $W(z)$ is updated using modified-FxLMS algorithm.
3. A new variable step size (VSS) LMS algorithm is proposed to adapt the modeling filter $S^\wedge(z)$.

The main idea in this method is to investigate the application of modified FxLMS (MFxLMS) algorithm in ANC systems with online secondary path modeling. Since convergence properties of the MFxLMS algorithm are known to be better than the FxLMS algorithm, the noise control filter $W(z)$ in the Proposed-2 method is adapted using MFxLMS algorithm. Furthermore, we propose a new variable step size (VSS) based LMS algorithm for secondary path modeling filter $S^\wedge(z)$. The step size is varied in accordance with the power

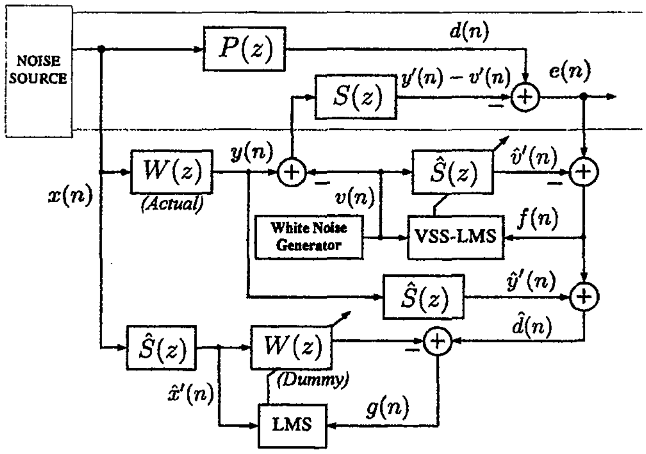

Fig. 3. Proposed-2 method for ANC systems with online secondary path modeling.

of the residual error signal [the desired response for the modeling filter]. It is found that the desired response for the modeling filter is corrupted by a noise, which is decreasing in nature, (ideally) converging to zero. Hence a small step size is used initially and later its value is increased according to decrease in the residual error signal $e(n)$ [7].

The details of this method can be found in [5, Chapter 4]. The simulation results presented in [5, Chapter 5] shows that this method achieves best secondary path modeling performance among the existing methods. Furthermore, it shows excellent tracking performance in non-stationary environment, and hence appears a best method for ANC systems with online secondary path modeling.

V. Concluding Remarks

Here we propose a new algorithm for ANC systems. This algorithm combines the concept of adaptive filtering with averaging (AFA) with the filtered-x LMS algorithm and is called FxAFA algorithm. It shows better convergence performance than the FxLMS algorithm, especially in the presence of a large measurement noise in error and reference microphones. We also propose two new methods for ANC systems with online secondary path modeling. The main feature of these methods is that these can achieve improved performance than the existing methods, without requiring third adaptive filter.

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論文審査結果の要旨

Active Noise Control（ANC）は，機械が発生する騒音やそれらが複合した環境における騒音に対して，同振幅で逆位相の音を放射し，音波の干渉により騒音を低減する技術である．これまでのANCでは，適応フィルタの個数を増やすことで雑音の除去性能の向上や適応処理の妨げとなる付加雑音の除去を行ってきたが，本研究では，入力信号の統計的性質を検討して，適応アルゴリズムの提案とフィルタ構造の改良（適応フィルタの個数の削減）を行っている．本研究では，これまでに，従来のFiltered-x Least Mean Square(FxLMS)アルゴリズムと比較して，高速に収束しかつ誤差の小さいアルゴリズムとしてFiltered-x Adaptive Filtering with Averaging (FxAFA) を提案している．さらに，2次経路のオンライン推定において，対象とする信号の性質を考慮して，これまで3個の適応フィルタを使用してシステムを構成していたが，これを2個で同等の性質をもつシステムを構成できることを示した．本論文は，これらをまとめたものであり，全編6章よりなる．

第1章は，ANCの基礎としてその構造や適応アルゴリズムについて述べている．さらに，オンライン2次経路推定について，その構造や適応アルゴリズム，従来法について述べている．

第2章は，Filtered-x Adaptive Filtering with Averaging (FxAFA) によりANCを実現する方法について提案している．ここでは，適応アルゴリズムの検討，計算量の評価，収束性能の実験的な検討，観測雑音の特性への影響の評価を行っている．

第3章は，FxAFAを用いた2次経路のオンライン推定を行うANCを提案している．ここでは，その適応アルゴリズムと構造を提案し，FxAFAの必要性，計算量について評価している．

第4章は，Modified-FxLMSアルゴリズムを用いた2次経路のオンライン推定を行うANCを提案している．ここでは，その適応アルゴリズムと構造を提案し，提案手法の収束特性の解析，適応アルゴリズムへの可変ステップサイズ導入の有効性について評価している．

第5章は，第3章と第4章で述べた手法について，計算機実験による評価・検討を行っている．入力信号やシステムの変動に対して，提案手法がANCとして性能が高いことを実験結果より確認した．

第6章は，結言である．

以上，要するに本論文は，Active Noise Controlにおいて，その適応アルゴリズムとフィルタ構造を新たに提案し，これを実装するとともに，計算機実験による評価・検討を行ったものであり，ディジタル信号処理および電子工学の発展に寄与するところが少なくない．

よって，本論文は博士(工学)の学位論文として合格と認める．