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NEW COEFFICIENT UPDATE METHOD FOR FEEDBACK CONTROL FILTERS IN ACTIVE NOISE CONTROL SYSTEM

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ABSTRACT
A feedforward active noise control system includes the secondary path filter for compensation of acoustic coupling between loudspeaker and reference microphone. In ordinary systems, the filter is designed to be a pre-set type, without any system identification for it, because the adaptation of its filter taps without any additional noise is difficult. This paper introduces a new identification method for the secondary path filter. It forms two independent equations by using the estimation error resulted by the two sets of different coefficients of the noise control filter. A numerical method is also described as a practical technique to solve the equations.

1 - INTRODUCTION

The feedforward type of active noise control system consists of three filters, the noise control filter, secondary path filter and the feedback control filter [1]. The adaptive filtering technique is applied only to the noise control filter in ordinary systems. The coefficients of the other filters are fixed to the values estimated by extra noise signal fed previously to starting the noise control. However, the continuous refreshment of these two filters are also necessary to avoid instability of the system and considerable worst case where the system falls into uncontrollable [2]. The conventional way of refreshing the coefficients of these filters, feeding the extra noise to the secondary source [3], makes a problem. The authors already presented a method to refresh the coefficients of the secondary path filter without feeding any extra noise [4]

This paper proposes a new refreshment way named "simultaneous equations method" for the coefficients of the feedback control filter without feeding any extra noise.

2 - PRINCIPLE

The available signal for identification calculation is only the output of the noise detection microphone, while this output involves two unknowns related to the feedback path and the primary source. The method given here composes two equations relating to the two different estimation errors involved in the noise control filter coefficients at the different point of time, and applies the linear prediction analysis to the signal in order to form two equations. The linear prediction filter coefficients consequently yield two equations corresponding to the noise control filter coefficients with different estimation errors. These simultaneous equations are solvable, and one of the solutions gives the impulse response of the feedback path.

The feedback path from the secondary source to the noise detection sensor forms the closed loop shown in Fig. 1.

For this loop, the input signal of the noise control filter is expressed as

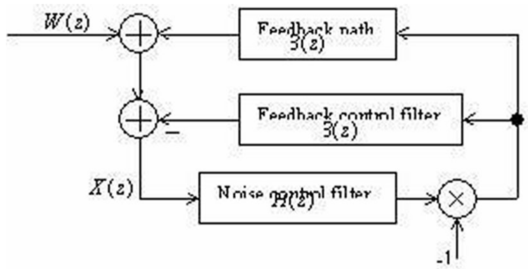

Figure 1: Feedback loop structured by the feedback path.

$$X\left(z\right)=\frac{W\left(z\right)}{1+H\left(z\right)\left\{B\left(z\right)-\hat{B}\left(z\right)\right\}} \tag{1}$$

by using $H(z)$, $B(z)$, $\hat{B}\left(z\right)$ and $W(z)$ which are the transfer functions of the noise control filter, feedback path, feedback control filter and the primary noise, respectively. We see only $X(z)$ is available for updating the coefficients of the feedback control filter. This input signal, however, has two unknowns, namely, $B(z)$ and $W(z)$.

Here, we suppose that the primary noise is synthesized through the following process by dividing $U(z)$, the linearly unpredicted and $P(z)$ the linearly predicted components:

$$W\left(z\right)=\frac{U\left(z\right)}{1-P\left(z\right)}\ ,\ P\left(z\right)=\sum_{k=1}^{K}p\left(k\right)z^{-k} \tag{2}$$

Then, the input signal is rewritten as

$$X\left(z\right)=\frac{U\left(z\right)}{1-P\left(z\right)}\cdot\frac{1}{1+H\left(z\right)\left\{B\left(z\right)-\hat{B}\left(z\right)\right\}} \tag{3}$$

Next, we apply the linear prediction analysis to calculate two unknowns, $B(z)$ and $P(z)$. The filtered-x algorithm [5] gives the coefficients with different estimation errors to the noise control filter at each sample time. We can get two coefficient sets with the different estimation errors by suspending the estimation process at two different times. Assume the transfer function for as $H_1(z)$ and $H_2(z)$ for two instances, we denote the linear prediction filter formulae

$$S_1\left(z\right)=P\left(z\right)-H_1\left(z\right)\left\{B\left(z\right)-\hat{B}\left(z\right)\right\}+P\left(z\right)H_1\left(z\right)\left\{B\left(z\right)-\hat{B}\left(z\right)\right\} \tag{4}$$

$$S_2\left(z\right)=P\left(z\right)-H_2\left(z\right)\left\{B\left(z\right)-\hat{B}\left(z\right)\right\}+P\left(z\right)H_2\left(z\right)\left\{B\left(z\right)-\hat{B}\left(z\right)\right\} \tag{5}$$

The simultaneous equations can be solved by the elimination technique. One of the unknowns, $P(z)$, however, is unnecessary for estimating the coefficients of the feedback control filter. In addition, since $H_2\left\{1-S_1\left(z\right)\right\}-H_1\left(z\right)\left\{1-S_2\left(z\right)\right\}\neq 0$ when $H_2\left(z\right)\neq H_1\left(z\right)$, we can get the difference by eliminating $P(z)$ as,

$$B\left(z\right)-\hat{B}\left(z\right)=\frac{S_1\left(z\right)-S_2\left(z\right)}{H_2\left\{1-S_1\left(z\right)\right\}-H_1\left(z\right)\left\{1-S_2\left(z\right)\right\}} \tag{6}$$

we denote this factor as $\hat{D}\left(z\right)$. This equation means the principle of the simultaneous equations method proposed in this paper.

3 - IDENTIFICATION TECHNIQUE

In practical use, the difference $\hat{D}\left(z\right)$ must be expressed as filter coefficients. The adaptive system shown in Fig. 2 updates the coefficients of the adaptive filter $\hat{D}\left(z\right)$, by assuming all the filters are the finite impulse response (FIR) type.

The adaptive filter coefficient vector $\hat{\mathbf{D}}$ can be estimated by white noise. After the error became minimum, $\hat{\mathbf{D}}$ consequently gives the difference between the impulse response sample vector and the feedback control filter coefficient vector,

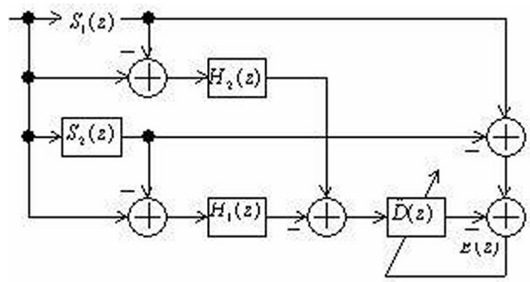

Figure 2: System identification technique to solve the simultaneous equations.

$$\mathbf{B} = \left[\ b\,(1) \quad b\,(2) \quad \ldots \quad b\,(I)\ \right]\ ,\ \hat{\mathbf{B}} = \left[\ \hat{b}\,(1) \quad \hat{b}\,(2) \quad \ldots \quad \hat{b}\,(I)\ \right] \tag{13}$$

where the number of elements of these vectors is supposed to be equal to I. The coefficients of the feedback control filter are thus updated by adding $\hat{\mathbf{D}}$ to $\hat{\mathbf{B}}$.

4 - ESTIMATION PROCEDURE

The method given here can be applied also to the estimation at the initial stage previous to starting the active noise control, when the noise control filter coefficients are not estimated yet.
Figure 3 is the procedure of repeatedly updating the coefficients of the feedback control filter from the initial stage without feeding the extra noise to the secondary source. Each stage is:

- Fix $\mathbf{H}_1 = \left[\ h \quad 0 \quad \ldots \quad 0\ \right]$ to the noise control filter, where h must be small so as not to cause howling.
- Fix $\hat{\mathbf{B}} = 0$ to the feedback control filter, because the coefficients of the feedback control filter optimally modeling the feedback path are not estimated yet. At least, in this stage, fixing $\hat{\mathbf{B}} = 0$ is safe.
- Estimate $\mathbf{S}_1$ corresponding to $\mathbf{H}_1$ by applying the linear prediction analysis to $X(z)$.
- Replace the coefficient vector of the noise control filter with $\mathbf{H}_2 = \left[\ 0 \quad 0 \quad \ldots \quad h\ \right]^T$.
- Estimate $\mathbf{S}_2$ corresponding to $\mathbf{H}_2$ by applying the linear prediction analysis to $X(z)$. After this linear prediction analysis, the coefficients of all the filters in the adaptive system shown in Fig. 2 become known.
- Feed white noise to the adaptive system and then estimate $\hat{\mathbf{D}}$.
- Add $\hat{\mathbf{D}}$ to $\hat{\mathbf{B}}$. This sum $\tilde{\mathbf{B}}$ gives the coefficient vector of the feedback control filter canceling the feedback path enough, thereby the estimation of the noise control filter coefficients become possible. In practical system, however, the feedback path may change after the estimation of $\tilde{\mathbf{B}}$. The feedback control filter coefficient vector needs to be repeatedly estimated.
- Replace $\mathbf{H}_1$ and $\mathbf{S}_1$ with $\mathbf{H}_2$ and $\mathbf{S}_2$ therefore:
- Resume the process of estimating the coefficients of the noise control filter, for example, by the filtered-x algorithm. This estimation process gives new $\mathbf{H}_2$.
- Return to (5), thereby the feedback control filter coefficients are repeatedly refreshed.

Figure 4 shows the convergence properties of $\tilde{\mathbf{B}}$ where h=0.2, N=192. In this example, the estimation procedure is repeated and gives the following three convergence properties:

- The first estimation in the initial stage.
- The second estimation: the noise control filter coefficients $\mathbf{H}_1$ and $\mathbf{H}_2$ are estimated with estimation error of -20 dB as the result of the estimation of $\hat{\mathbf{B}}$.
- The third estimation.

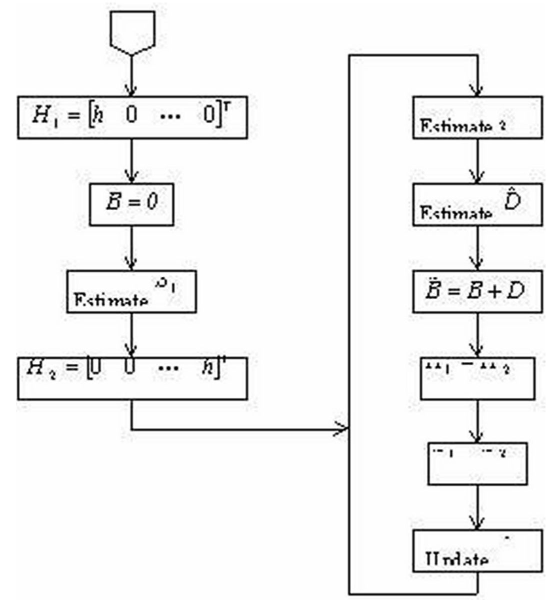

Figure 3: Procedure to repeatedly update the coefficients of the feedback control filter.

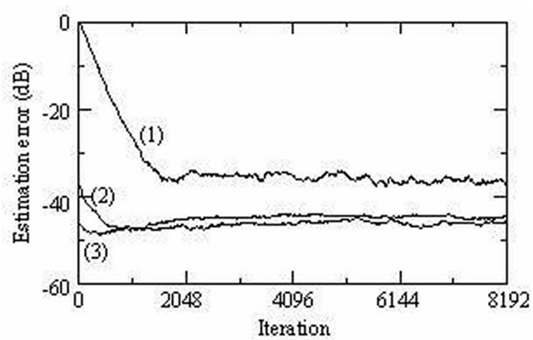

Figure 4: Repetition of the estimation: (1) the first estimation in the initial stage, (2) the second estimation and (3) the third estimation.

This example shows that the simultaneous equation method is available for estimating the coefficients of the feedback control filter in the initial stage and can keep the estimation error low.

5 - SUMMARY

This paper proposed the simultaneous equation method to ceaselessly update the coefficients of the feedback control filter without feeding the extra noise to the secondary source and has presented the system identification technique to solve the simultaneous equations. According to the simulation results, the method and technique estimate the coefficients of the feedback control filter automatically from the initial stage.

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