Duct 잡음 제거를 위한 적응 능동 소음 제거법

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Adaptive active control for duct noise attenuation

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Abstract

The adaptive active noise control algorithm for duct noise attenuation is considered.

A new algorithm to estimate the secondary path transfer function using multiple models is presented. The computational burden of the proposed algorithm is much smaller than the existing methods, so it could be applied to the multi-channel cases.

Computer simulations were done to show the effectiveness of the proposed algorithm in a duct case.

I. Introduction

Acoustic noise has many negative effects on humans and animals. Passive noise control, which uses sound absorbent materials mounted on and/or around the sources of noise, is effective at high frequencies (greater than 500 Hz, say), but it becomes either ineffective or tend to be very expensive or bulky at low frequencies. Active noise control(ANC) can be applied to overcome these difficulties[1].

There have been a considerable amount of effort devoted to duct noise problem. Since the acoustic tripole and dipole have geometry related limitations, the current trends is to use the acoustic monopole.

In practical situations, the characteristics of the system to be controlled as well as the primary noise characteristics can be changed with time. In this case, an algorithm which simultaneously performs identification and control must be used.

Since these approaches require knowledge of the secondary path transfer function, some adaptive algorithms which simultaneously estimate the transfer function of a secondary path have been presented[2]. Such techniques are difficult to be applied to the multiple sensor multiple speaker cases because there are too many parameters to be estimated in each step.

We present a new algorithm for system identification in active control using multiple models. Multiple models were used for identification of the secondary path transfer function and the IIR structure was used for the control filter. Since this approach requires only a small amount of computation, it may also be used in the multiple channel case.

2. Adaptive ANC Problems

Active sound attenuation systems where an adaptive digital filter is used as the feedforward controller is shown in fig. 1.

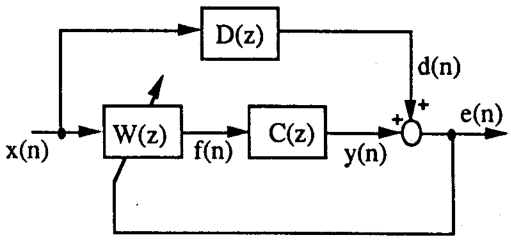


Fig. 1 Block diagram of the ANC system

The ideal control filter which, in the absence of measurement noise and any causality constraints, drives e(n) to zero, is given by

$$W_R(z) = -\frac{D(z)}{C(z)} \qquad (1)$$

The usual adaptive filter algorithm, such as the least mean square(LMS) algorithm, cannot be applied because of the presence of the secondary path transfer function C(z) between the input to the loudspeaker and output from the microphone.

In order to use the LMS algorithm for this case, we must change the order of the control filter and the secondary path transfer function. If the plant is linear and if the plant and the control filter are time-invariant or slowly time-varying compared to the combined memory times or time constants of the adaptive filter and the plant, the order can be changed as shown in fig. 2.

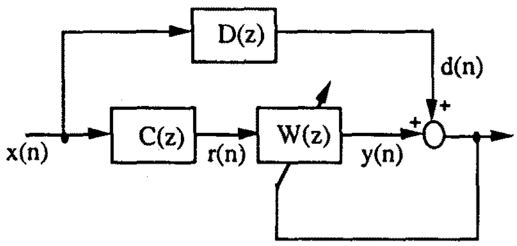


Fig. 2 Modified block diagram of the ANC

If C(z) is known, then we can apply the filtered-x LMS algorithm.

The filter coefficient vector $\underline{w}(n)$ can now be updated using the equation

$$\underline{w}(n+1) = \underline{w}(n) - 2\mu e(n)\underline{r}(n) \quad (2)$$

where μ is the convergence parameter that controls the rate of convergence, e(n) is the error signal and $\underline{r}(n)$ is the input signal vector.

If C(z) is unknown or time-varying, $\hat{C}(z)$ must be identified to apply the filtered-x LMS algorithm. Elliott et al.[3] used off-line estimation of C(z), i.e. it can be estimated before installation for a particular setting and fixed thereafter.

In many practical applications, C(z) could be changed with time, for example temperature and flow changes in the duct result in sound velocity changed in the system. In this case, the adaptive active control algorithm which estimates the parameters of C(z) and adjusts the filter parameters simultaneously should be applied.

3. The multiple-model adaptive system approach

We present the multiple-model adaptive algorithm to identify the secondary path transfer function[4].

Most approaches of MMAC use multiple controllers designed to control each of the assumed models of the plant. But, in this research, multiple-models are used in the identification routine to estimate the secondary path transfer function as shown in Fig. 3., i.e. we estimate the secondary path transfer function using the multiple-model techniques and then adjust the single control filter coefficients using the LMS algorithm.

The output of i-th model is

$$y_i(n) = \hat{C}_i(z)\, v(n) \quad (3)$$

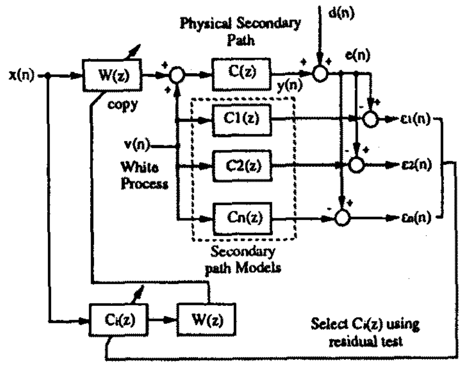


Fig. 3. Multiple-model adaptive system

where v(n) is an identification process which is usually a white process.

The residual error of the i-th model is

$$e_i(n) = e(n) + y_i(n) \quad (4)$$

where e(n) is the output of the error microphone.

The power of the i-th residual error is

$$PE_i = \sum_{j=n-D}^{n} e_i^T(j)e_i(j) \quad (5)$$

where D is a memory length.

The procedure for MMAC is as follows:

(i) Construct n models for expected variations of the second path, for example change in the sound propagation speed due to the temperature change in a duct.

(ii) Calculate PE_i for each model.

(iii) Select the model which has the minimum residual power.

(iv) Use this model to generate the filtered reference signals needed to adjust the IIR filter coefficients in the controller using the RLMS algorithm.

In this approach, the most important problems are how to select models and how to choose the memory length D.

4. Computer simulations

We selected a simple duct as a simulation model. We assume that there is no acoustic feedback. This algorithm, however, can be easily extended to when the acoustic feedback exists. A representation of a simple duct is given in figure 4.

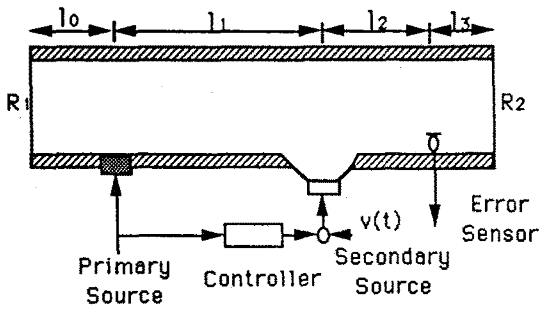


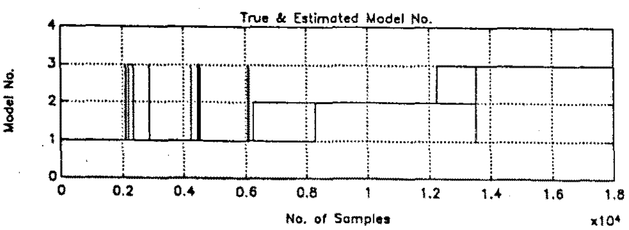
Fig. 4 Representation of a simple duct

The primary source is located a distance of l_0 from one end of the uniform duct, and the secondary source is located a distance of l_1 from the primary source. A sensor which detect the error is located a distance of l_2 from the secondary source. The left and right ends of the duct have complex pressure reflection coefficients R_1 and R_2 respectively, and v(t) is an identification signal.

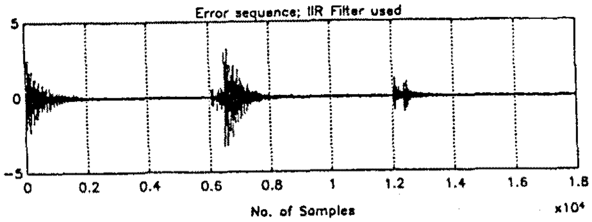
This duct considered as the acoustic and electroacoustic system which has two electrical inputs; V_p and V_s that used to drive the primary and secondary source, and one electrical output, V_e, that from the error sensor.

We have done computer simulations in case of the simultanous identification and control. The speed of sound changes from 330(m/s) to 350(m/s). 3 Secondary path models which are correct for sound speeds of 330(m/s), 340(m/s) and 350(m/s) were used. And the memory length D set to 1000 samples.

The results are shown in figure 5. It shows that the proposed algorithm converges very well.



(a) True and estimated model number



(b) Error sequence

Fig. 5 True and estimated model numbers and errors for simultanously identification and control

5. Conclusions

In this research, the adaptive active noise control algorithm for duct noise attenuation was considered. The new identification algorithm for the secondary path transfer function was presented using the multiple model approach.

Computer simulations was done to show effectiveness of the proposed algorithm in a duct case.

This algorithm could be used for adaptive multi-channel active noise control problems

References

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