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ACTIVE NOISE CONTROL BASED ON ADAPTIVE PREDICTION

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1. INTRODUCTION

In this paper, we propose an new active noise control system using adaptive prediction algorithm[1]. Adaptive prediction type active noise control system (we refer as AP-ANC) has many attractive advantages. For example, the AP-ANC system can carry out noise reduction only from the error sensor informations without using input reference signals. This advantage makes it possible to apply the AP-ANC system to the case of primary noises being distributed everywhere in the sound field, or the case of the arrangements of input sensors are restricted to install with the physical constrains. For these sound field, it seems to be difficult to apply the ANC system based on the feed-forward (we refer as FF-ANC) strategies. However, due to the theoretical limits of predicting future waveform of noise signals, we have to investigate the effective algorithms for the prediction and also consider the configurations as the practical control system. The BOP (Block Orthogonal Projection) algorithm is applied to update an adaptive predictor in our system. We also discussed an new configuration of ANC system that incorporated with FF-ANC and AP-ANC to expand the control frequency band and to achieve the successful noise reductions.
On the other hand, an auxiliary sensing morphines is introduced to AP-ANC system in order to improve the effect of the control. This system can be considered as a cooperative working system of dual AP-ANC sharing the same error signal. The effectiveness of auxiliary sensing microphones is also discussed in this paper.

2. OUTLINE OF THE SYSTEM

Fig.1. shows a block diagram of a simple AP-ANC system. AP-ANC system

can be divided into two process, such as a control and learning process. In the learning process shown in the lower part of Fig. 1., an instantaneous value of the noise signal at a error microphone can be predicted by the time history of the past noise signal. This process suffices to update a filter coefficient for the control process. Therefore, only information from an error microphone is used to control the noise in the AP-ANC system. In contrast with this simple AP-ANC, AP-ANC with an auxiliary sensing microphone (we refer as ASM-AP-ANC) uses two different information from the two microphones for predicting the future noise signal[3][4]. The AP-ANC system using an auxiliary sensing microphone is shown in Fig.2. It is worth to put emphasis on the algorithm of the filtered-x update processes using the same transfer function $\hat{C}_e$ between the secondary source and error microphone, that is, two W filters minimized the one error microphone signal but use the two input signals for prediction. The algorithm for the ASM-AP-ANC is formulated as follows.

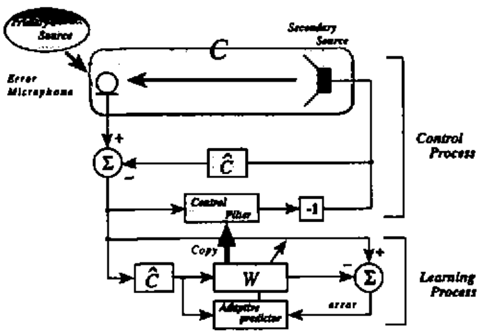

Fig. 1. A Block diagram of Adaptive Prediction ANC

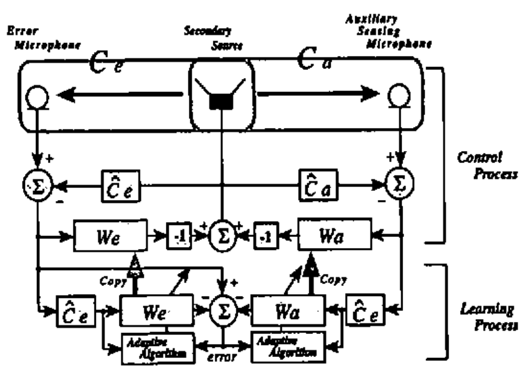

Fig. 2. A Block Diagram of APANC using Auxiliary Sensing Microphone

3. FORMULATION

3.1. Input signal

We denote that the secondary source output signal is $s(n)$, secondary path of auxiliary microphone side is $\mathbf{C}_a$, secondary path of error microphone side is $\mathbf{C}_e$, the tap lengths of $\hat{C}_a$ and $\hat{C}_e$ are J_a and J_e, and $\hat{C}_a$ and $\hat{C}_e$ denote the estimated impulse responses of the secondary path of $\mathbf{C}_a$ and $\mathbf{C}_e$, respectively.

The impulse responses of the secondary paths are represented as the vector forms.

$$\hat{C}_a = \left[C_a^1, \cdots, C_a^{J_a}\right] \tag{1}$$

$$\hat{C}_e = \left[C_e^1, \cdots, C_e^{J_e}\right] \tag{2}$$

The sequences of the secondary source outputs are written as

$$S_a(n) = \left[s(n), \cdots s(n - J_a + 1)\right]^T \tag{3}$$

$$S_e(n) = \left[s(n), \cdots s(n - J_e + 1)\right]^T \tag{4}$$

The input signal for auxiliary sensing microphone is denoted as ε_a, and input signal of the error microphone is $\varepsilon_e(n)$. The input signal used for the learning process and for the control process are represented as follows.

$$d_a(n) = \varepsilon_a(n) - \hat{C}_a S_a(n) \tag{5}$$

$$d_e(n) = \varepsilon_e(n) - \hat{C}_e S_e(n) \tag{6}$$

If we define I_a and I_e are the tap length of W_a, W_e, respectively, the time series of input signal can be written as

$$D(n) = [D_a(n), D_e(n)] = \left[d_a(n), \cdots, d_a(n - I_a + 1), d_e(n), \cdots, d_e(n - I_e + 1)\right] \tag{7}$$

3.2. Learning process

The coefficient of the adaptive filter is denoted by the equation (8).

$$W(n) = \left[W_a^1(n), \cdots, W_a^{I_a}(n), W_e^1(n), \cdots, W_e^{I_e}(n)\right] \tag{8}$$

The input signals to the adaptive filters are obtained by convolution of the input signal and the impulse response of secondary path.

$$x_a(n) = \hat{C}_a D_a(n) \tag{9}$$

$$x_e(n) = \hat{C}_e D_e(n) \tag{10}$$

Then the input signal sequence to adaptive filter is written in next equation.

$$X(n) = \left[x_a(n), \cdots, x_a(n - J_a + 1), x_e(n), \cdots, x_e(n - J_e + 1)\right]^T \tag{11}$$

Therefore, the error signal used in the learning process is written as

$$e(n) = d_e(n) - W(n)X(n) \tag{12}$$

Here, the second order (p=2) of the projection algorithm are introduced for updating the filter coefficients of the predicting filter.

$$\begin{gathered} W(n+1) = W(n) + \mu\{a_1(n)x(n) + a_2(n)x(n-1)\} \\ \begin{bmatrix} a_1(n) \\ a_2(n) \end{bmatrix} = \begin{bmatrix} x(n)^T x(n) & x(n)^T x(n-1) \\ x(n-1)^T x(n) & x(n-1)^T x(n-1) \end{bmatrix}^{-1} \begin{bmatrix} e(n) \\ (1-\mu)e(n-1) \end{bmatrix} \end{gathered} \tag{13}$$

where μ denotes a step size parameter.

3.3. Control process

The secondary source output $s(n)$ can be obtained by the convolution of the input signal and the coefficients of the predicting filter that obtained in the learning process described above. Therefore, the secondary source output can be shown in equation (14).

$$s(n) = -W(n)D(n) \qquad (14)$$

4. SIMULATION

In this chapter, we examined on the effectiveness of ASM-AP-ANC by comparing with FF-ANC and AP-ANC system in the control of the narrow band noise source.

4.1. Conditions of the computer simulations

The FIR type of transfer functions were used in all conditions. The sampling frequency was 2000[Hz], the order of projection algorithm p was 2, and the step size parameter μ was set as 0.1. Adaptive iterations were made up to 2000. Time delay of the secondary path C_e and C_a are z^{-3} and z^{-10}, respectively. We also assumed that these secondary path transfer functions were perfectly modeled.
The total tap length of the filter in ASM-AP-ANC is bounded to 100 taps in all simulations. For example, if W filter for error microphones side I_e is set as 80 tap filter, then W filter for auxiliary microphone side I_a should be set as 20 tap filter.
A band limited white noise signal, obtained by the band pass filter having a center frequency of 200[Hz], was used as a primary noise source.
The simulations were carried out in the several conditions of the auxiliary sensing microphone positions and error microphone positions as shown in Fig. 3. to Fig 5. In order to simulate the worse conditions of sensing, the additional simulations were also made by adding a narrow band white noise signal (175-225 [Hz]) to the error microphone.

4.2. Results

Fig. 3., 4 and 5 show the mean square error (MSE). Fig. 3. shows the result of the arrangement where the auxiliary sensing microphone located in the upstream of the noise field. Fig. 4. presents the case where the error microphone was replaced instead of the auxiliary sensing microphone. Fig. 5. shows the case that the auxiliary sensing microphone was located in the upstream of the noise field and also an additive the noise was applied to the error microphone.
Fig. 6. and 7 show the frequency characteristics of the results corresponding to Fig. 3. and 5 respectively. In Fig. 3., we can observe that the AP-ANC achieved small noise attenuation as increasing the frequency band width. Though the limited improvement of noise reduction compared to that of FF-ANC, the results of ASM-AP-ANC demonstrates its possibility of

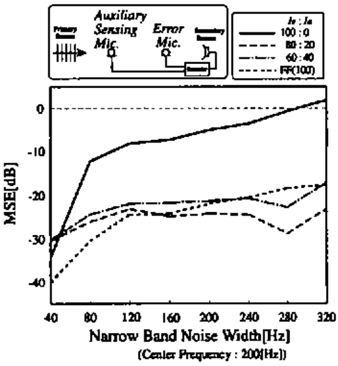

Fig. 3. Mean square error at control point (Auxiliary sensing microphone in the upstream of the primary noise field)

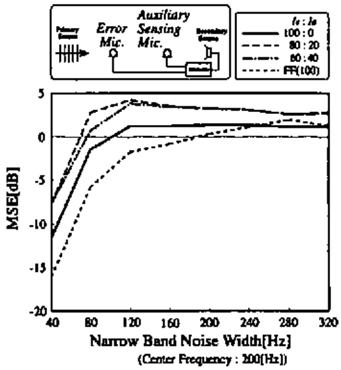

Fig. 4. Mean square error at control point (Error microphone in the upstream of the primary noise field)

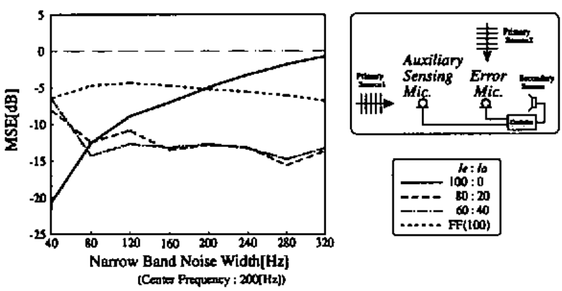

Fig. 5. Mean square error at control point (Auxiliary sensing microphone in the upstream of the primary noise field. Additional primary noise source(175-225[Hz]) is mixed at the error microphone)

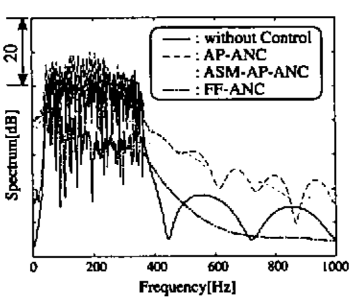

Fig. 6. Frequency spectrum at control point (Corresponding to the Fig. 3.)

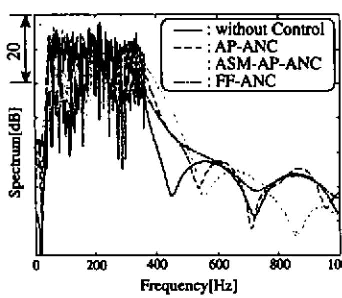

Fig. 7. Frequency spectrum at control point (Corresponding to the Fig. 5.)

improvement for the wide band control of noise.

On the other hand, Fig. 4. shows that the effect of control is remarkably affected on the arrangement of the auxiliary microphone position. Comparing Fig. 3. with Fig. 4., we can see that the noise reduction obtained by the single microphone AP-ANC in Fig. 4. is worse than that of Fig.3. This may due to the long term prediction requirement depended on the distance between the error microphone and secondary source.

As shown in Fig. 5., the use of auxiliary sensing microphone provide the effective control in the wide frequency range, while FF-ANC does not enable to control in the worse sensing condition of the reference signal.

In Fig. 7., we can see the uncontrolled frequency range between 175 and 225[Hz] by the FF-ANC, and this follows that these frequency contents were not sensed in the feed-forward strategy. However, in the case of ASM-AP-ANC, it may not be a serious problem whether sensing is completed in all frequency range that is expected to be controlled.

5. CONCLUSION

In this paper, we considered the possible applicability of ANC based on adaptive prediction strategy, that is, typical adaptive prediction type ANC with a single microphone and adaptive prediction type ANC using auxiliary sensing microphone (ASM-AP-ANC). As the result, we showed that the ASM-AP-ANC achieved remarkable improvement of noise reduction than that of FF-ANC if complete sensing was not expected.

Moreover, as far as our computer simulations, we shown that the ASM-AP-ANC performs broad band active noise control than that of AP-ANC without auxiliary microphone.

In order to develop actual ANC system such as ear-defender system, further investigations will be needed on the hybrid control of FF-ANC and AP-ANC. Especially, we consider that the theoretical studies on cooperation of the different control strategies will be one of very important issues.

References

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