Real-Time Implementation of The Active Adaptive Noise Controller Considering the Reflected Noise

(반사 소음을 고려한 능동 적응 소음 제어기의 실시간 구현)

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ABSTRACT

Real-time implementations of the active adaptive noise controller are proposed and tested. There are three problems in active noise control such as real-time processing, an acoustic feedback of secondary signal and a time-delay of control system elements.

For real-time processing, the DSP56001 (Digital Signal Processor) was used. To avoid acoustic feedback, the secondary signal (control signal) was excluded from prediction. And for compensation of time delay, the ahead prediction was applied. As the primary noise (source noise) is reflected in space, the reflected noise should be controlled for perfect noise control. But in this case, the controller might be unstable. For solving the problem, it is proposed that the source noise and the reflected noise are predicted separately. Some experimental results show the stability and effectiveness of the proposed controller.

요 약

실시간 능동소음 제어기를 제안하고 이를 구현하였다. 2차 음원을 이용한 능동소음 제어기는 저주파의 소음을 감쇠시키는데 있어서 종래의 수동적 방법 보다 탁월한 효과를 보여주고 있으나, 이의 구현에는 제어기 요소의 시간지연, 음향궤환 및 실시간 처리 등의 문제가 발생하게 된다.

본 논문에서는 제어 시스템의 요소들이 갖는 시간지연을 보상하는 방법으로 지연된 시간을 고려한 선형예측을 사용하는 적응 필터를 이용하고, 2차음의 궤환으로 인한 안정도의 파괴문제는 궤환되는 신호를 예측에서 제외시키는 방법으로 해결하였다. 또한 하드웨어적으로 연산속도가 매우 빠른 신호처리기(DSP56001)를 이용하고 소프트웨어적으로 Pipe-Lining 기법을 사용하여 실시간처리 문제를 해결하였다.

한편, 임의 공간에서 반사되는 소음과 소음원에서 방출되는 소음과의 상호관계는 공간의 기하학적 배치에 따라 항상 변하게 되므로, 소음의 예측만으로는 완전한 소음 감쇠를 기대할 수 없다. 따라서 반사소음도 소음과 함께 예측되어야 한다. 그러나 소음과 반사소음을 함께 예측하면 제어 초기에 많은 추정 오차를 갖게되고, 제어기가 발산하는 현상이 발생한다.

본 논문에서는 반사소음의 제어와 잘못된 예측에서 오는 제어기의 발산을 방지하기위하여, 2개의 마이크를 이용하여 소음원에서 발생되는 소음과 반사소음을 분리하여 검출, 예측하는 방법을 제시하였다.

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I. Introduction

As the industry and economy grow, we are more concerned about environments, specially about noise.

This problem has been studied in two classes. The one is how to lessen the noise from the primary source (passive control), the other is how to cancel the noise using secondary source (active control). Active control has been actively studied in America, Japan and Europe since 1970[10].

Active methods are the best at low-frequency noise, which complements more conventional passive methods since these tend to work best at higher frequency[10].

The basic ideas of active noise control are as follows. The controller detects source noise by a microphone, changes its phase(180°) and generates the control signal through the speaker. It is desirable to make the controller adaptive. Because the frequency or spatial distribution of the primary noise changes with time, and the controller is required to track these changes. A more difficult adaptive task has to be performed when the response of the system to be controlled to a given secondary excitation also varies with time. In this case, an algorithm must be able to perform idenification and control simultaneously. So far, the active noise control in a duct is mainly studied[9]. In this paper, the control problem in a free space was studied.

In section 2, the basic concepts of active adaptive noise control are introduced. The problems of the active control in a free space and its solutions are persented in Section 3, 4. The real-time controller is implemented in section 5, and conclusions are presented in section 6.

II. Basic Concepts of Active Adaptive Noise Control.

In general, noise signals are nearly periodic, so it can be written as sum of sinusoidal signals.

$$s(t)=b_0+\sum_{n=1}^{\infty} a_n \sin(nwt+\theta_n) \qquad (2.1)$$

If we know the factors (b_0, a_n, w, θ_n), noise signal $s(t)$ can be canceled by the control signal $s_c(t)$ which has 180° phase shift. But in many cases, the factors are time-varying, so controller must contain predictor as shown in Fig. 2.2.

Let the noise signal $s(t)$ be approximated by AR model,

$$s(t)=\sum_{n=1}^{N} a_n s(t-n)+e(t) \qquad (2.2)$$

Estimating the coeficient vector a_n by adaptive filter, we can make the control signal $s_c(t)$ as (2.3), (2.4).

$$\hat{s}(t)=\sum_{n=1}^{N} \hat{a}_n s(t-n) \qquad (2.3)$$

$$s_c(t)=-\hat{s}(t) \qquad (2.4)$$

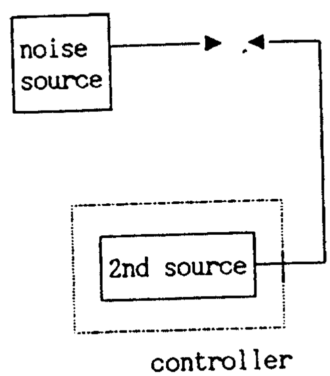


Fig. 2.1. Time-Invariant Case

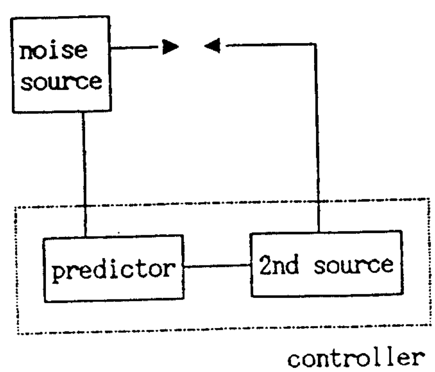


Fig. 2.2. Time-varying Case

Ⅲ. Problems on Active Noise Control and Solutions.

There are problems on active noise control such as

- Real-Time Processing
- Time Delay of Control System Elements
- Acoustic Feedback of Secondary Signal

3.1. Real-Time Processing

It is well known that

$$2f_o < f_s \qquad (3.1)$$

f_o : maximum frequncy of signal

f_s : sampling frequency

In processing low-frequency noise signal (lower than 1KHz), sampling frequency f_s should be much higher than 2 KHz. So sampling interval T_s is,

$$T_s < 1/(2\ \mathrm{KHz}) = 0.5\times10^{-3}\ \mathrm{sec} \qquad (3.2)$$

The prediction should be finished in the sampling interval. But general processors (80286 or 80386 based personal computer) can not perfome the computations (prediction) within that interval.

For real-time processing, we adopt the DSP chip and apply the pipe-lining technique[5].

- Pipe-Lining.

perfrom the calculation and fetch the next data in single instruction cycle.

(ex)

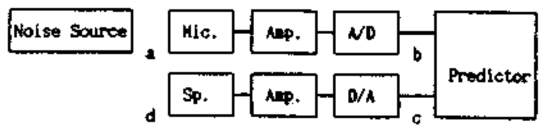
mac X0, Y0, A	X:(R0)+,X0 Y:(R4)-,Y0
multiply data X0, Y0 and accumulate it to A	fetch next data

- DSP56001.

DSP56001 caculates about 10^7 instructions in a second.

3.2. Time Delay of the Control System Elements.

During noise signals are transmitted to the predictor, the control system elements (microphone, amplifier, A/D conveter, etc) have time delay (T_d) as shown in Fig. 3.1.



where, Sampling Interval : d_{bc}
Time Delay of Mic, Amp, A/D : d_{ab}
Time Delay of D/A, Amp, Sp : d_{cd}
Total Time Delay : $T_d = d_{ab} + d_{bc} + d_{cd}$

Fig. 3.1. Time-Delay of Control System Elements

Though the controller predicts precise s(t) and generates the control signal $s_c(t)$, noise signals can not be canceled because of the time dela T_d.
at time to

$$s_c(t_o) = -s(t_o - T_d) \fallingdotseq -s(t_o) \qquad (3.3)$$

For solving this problem, it is necessary that the ahead prediction considering the delay time is applied.

- Ahead Prediction

The ahead step can be found as follows,

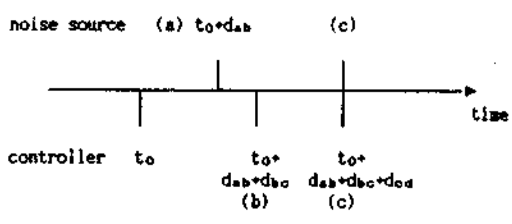


Fig. 3.2 n-step Ahead Prediction

(a) (t_o+d_{ab}) ; noise source generates $s(t_o+d_{ab})$, predictor receives $s(t_o)$.
(b) $(t_o+d_{ab}+d_{bc})$; predictor estimates $s(t_o+d_{ab}+d_{bc}+d_{cd})$
(c) $(t_o+d_{ab}+d_{bc}+d_{cd})$; noise source generates $s(t_o+d_{ab}+d_{bc}+d_{cd})$, controller generates $s_c(t_o+d_{ab}+d_{bc}+d_{cd})$.

Hence, suitable steps of ahead prediction is

$$\# \text{ of ahed steps} = \frac{d_{ab}+d_{bc}+d_{cd}}{d_{bc}}$$

$$= T_d / d_{bc} \tag{3.4}$$

3.3. Acoustic Feedback of Secondary Signal.

This problem is related to the reflected noise, so it is considered in next section.

Ⅳ. Reflected Noise.

A general active noise controller is shown in Fig. 4.1.

The microphone is located very close to the primary source in order to take the source noise only. Hence, the controller predicts and controls the source noise only.

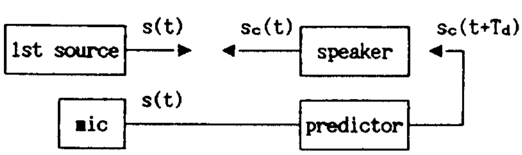


Fig. 4.1. Control Scheme for Source Noise only

Even though s(t) is precisely estimated, the noise can not be perfectly canceled for the effects of reflected noise signals $(s_{r1}(t), s_{r2}(t))$.

Hence, the reflected noise signals should be considered.

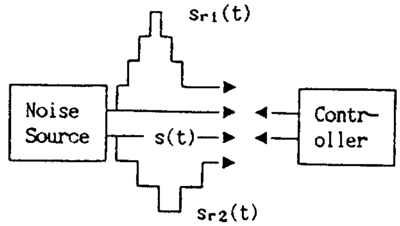


Fig. 4.2. Effects of Reflected Noise.

4.1. Consideration of reflected noise.

The microphone takes both noise and the secondary signal (control signal), as shown in Fig. 4.3. Therefore, the secondary signal returns to the predictor, which disturbs next prediction (Acoustic feedback).

The error signal detected by the microphone is,

$$e(t)=s(t)+s_r(t)+s_c'(t) \tag{4.1}$$

To avoid the acoustic feedback, $s_c(t)$ should be

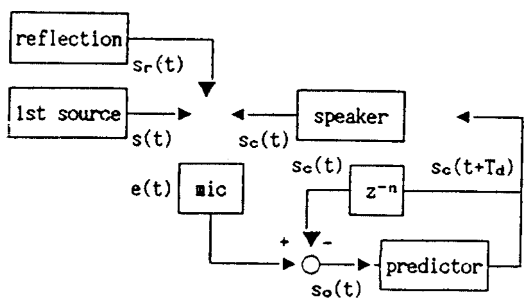


Fig. 4.3. Control Scheme for Source Noise and Reflected Noise

excluded.

Actually, $s_c'(t)$, detcted by the microphone, is slightly different from $s_c(t)$. But if the microphone were close enough to speaker, then

$$s_c(t)=s_c'(t) \tag{4.2}$$

Hence,

$$s_c(t)=e(t)-s_c(t) \tag{4.3}$$
$$=s(t)+s_r(t)$$

where the reflected noise $s_r(t)$ is,

$$s_r(t)=s_{r1}(t)+s_{r2}(t)+\cdots \quad (4.4)$$

The reflected noise $s_{r1}(t)$, $s_{2r}(t)$ differs from source noise in phase respectively.

$$s_{r1}(t)=s(t-t_1) \quad (4.5)$$
$$s_{r2}(t)=s(t-t_2)$$
$$\vdots$$

This method has much error in initial ststes, specially at ahead prediction. (See following simulations) Beacause of the poor prediction, the reflected noise is generated by not only source noise but also control error.

$$s_r(t)=s_{r1}(t)+s_{r2}(t)+\cdots \quad (4.6)$$
$$+e_{r1}(t)+e_{r2}(t)+\cdots$$

where,

$$e_{r1}(t)=e(t-t_1) \quad (4.7)$$
$$e_{r2}(t)=e(t-t_2)$$

Thus, the control error $e(t)$ can not converge.

4.2. Prediction Source Noise and Reflected Noise Separately.

For stable noise controller, it is proposed that the source noise and the refelcted noise are estimated separately.

In Fig. 4.4,

$$e(t)=s(t)+s_r(t)+s_c(t) \quad (4.6)$$

$$e'(t)=e(t)-s(t) \quad (4.7)$$
$$=s_r(t)+s_c(t)$$

To avoid acoustic feedback,

$$s_r'(t)=e'(t)-s_c(t)=s_r(t) \quad (4.8)$$

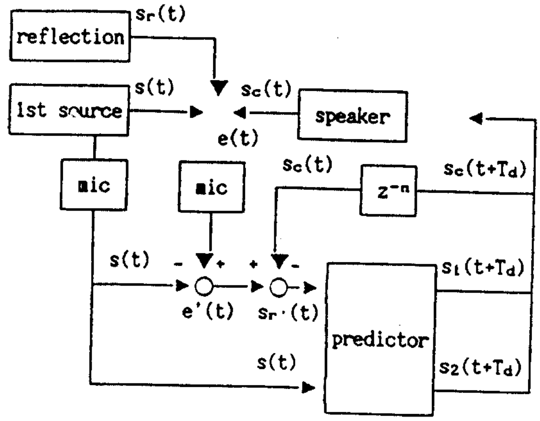


Fig. 4.4. Control Scheme for Separate Prediction

Hence, the reflected noise $s_r(t)$ can be obtained from (4.8). $s(t)$ and $s_r(t)$ can be predicted as (4.9), (4.10) respectively.

$$s_1(t)=\sum_{i=1}^{2m} a_i s_0(t-i) \quad (4.9)$$

$$s_2(t)=\sum_{j=1}^{2m} b_j s(t-j) \quad (4.10)$$

then, the control signal is,

$$s_c(t+T_d)=s_1(t+T_d)+s_2(t+T_d) \quad (4.11)$$

If $s_r(t)$ is badly estimated or there are sudden disturbances, then the controller generates control signal $s_c(t+T_d)$ using $s_2(t+T_d)$ only. Then, over-all controller can be stable.

$$\text{if } s_1(t+T_d)>\alpha \quad (4.12)$$
$$s_c(t+T_d)=s_2(t+T_d)$$

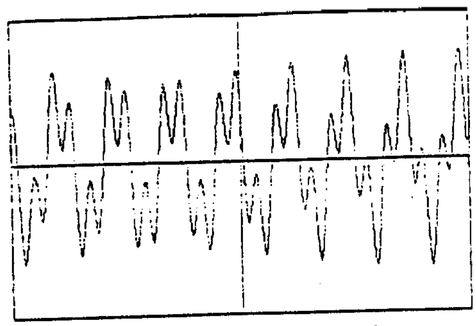
where, α is maximum error bound

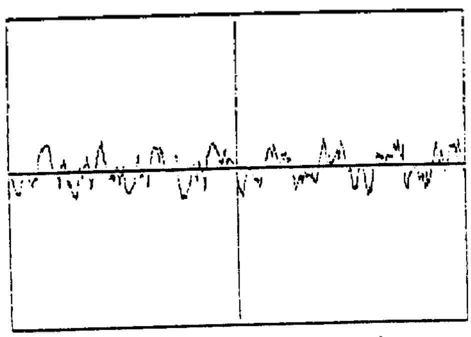
⊙ Simulations.

Source Noise ;

$s(t)=4.12*\sin\ (2*pi*0.32*t)\ +2.94*\sin\ (2*pi*0.99*t)$

Reflected Noise ;


original signal


reflected signal

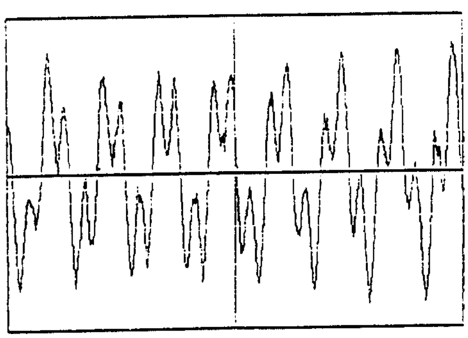
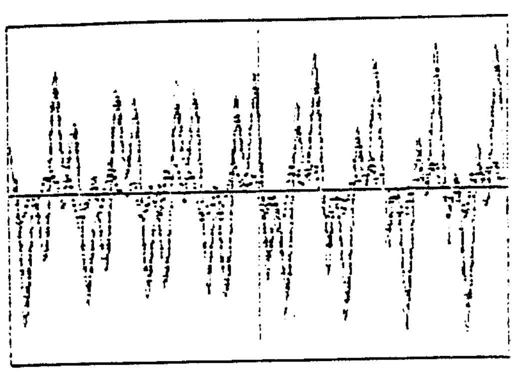

original+reflected signal

Fig. sample signals.


original signal
estimated signal
error signal

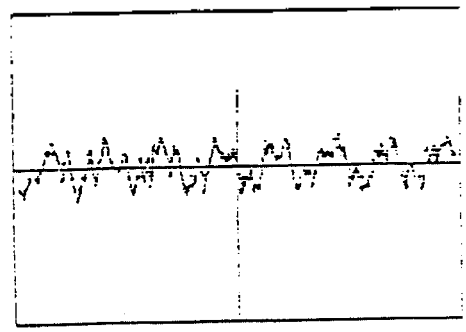
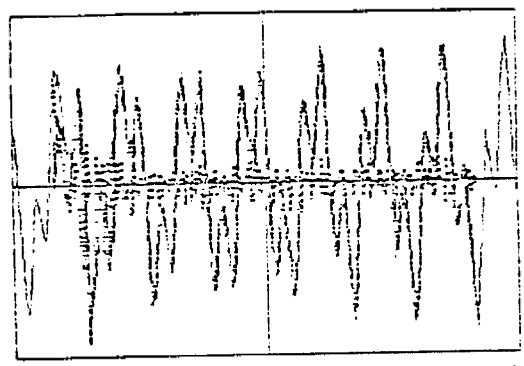

error signal

Fig. (a) Prediction Source Noise only.


original+reflected signal
estimated signal
error signal

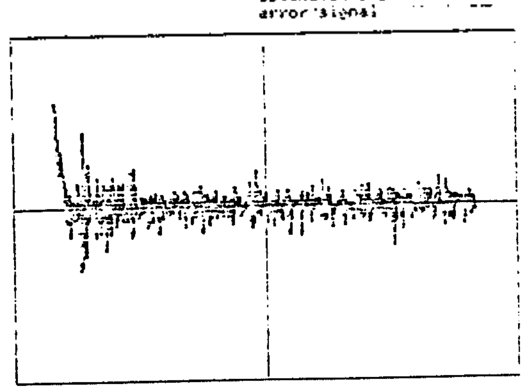
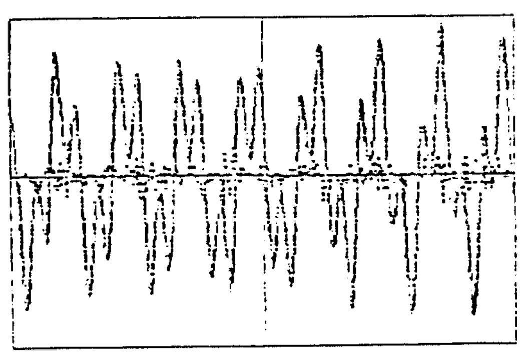

error signal

Fig. (b) Prediction Source Noise and Reflected Noise


original+reflected signal
estimated signal
error signal

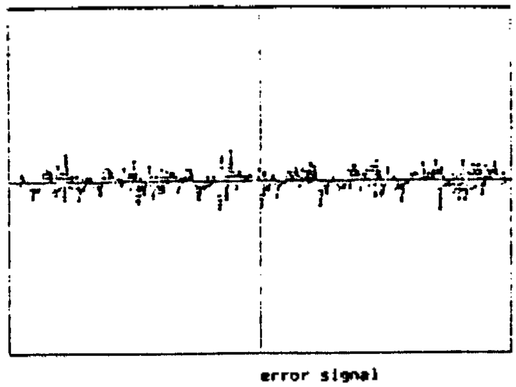


Fig. (c) Prediction Source Noise and Reflected Noise separately.

$$r(t)=\sum_{i=1}^{3}(a_i+0.412)*\sin\ (2*pi*(0.32-b_i)+c$$
$$+\sum_{j=1}^{3}(a_j+0.294)*\sin(2*pi*(0.99-b_{ji}))$$
$$+c_j))+w(t)$$

where, $w(t)$ is (0, 0.1) White Noise

$0<a_i,b_i<0.1$ $0<a_j,b_j<0.1$

$0<c_i<10$ $0<c_j<10$

# of ahead step : 16 step

V. Implementation of Active Noise Controller.

5.1. Specification

- Primary Noise

sinusoidal signal (120 Hz)

- A / D, D / A Conveter

56ADC16[6].

resoultion : 16 bit

containig a reconstruction filter and an anti-aliasing filter

- Processor

DSP56001 ;

programmable real-time digital signal processor (10 MIPS)

- Interfacing

IBM PC (AT)

- System Constants

Sampling time : 0.25 msec

Delay Time : about 4 msec

# of Ahead Step : 16 step

- Algorithm

Reculsive Least Squares

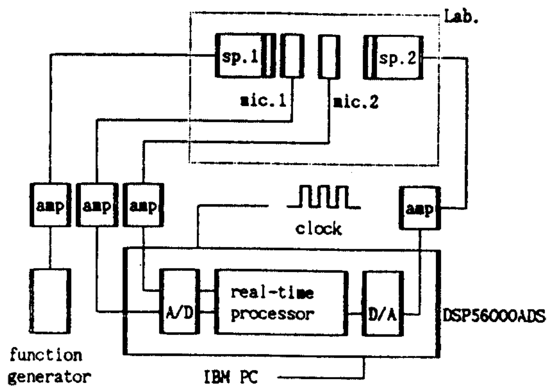


Fig. 5.1. Block diagram of the Active Adaptive Noise Control Experiment

5.2. Experimental Results.

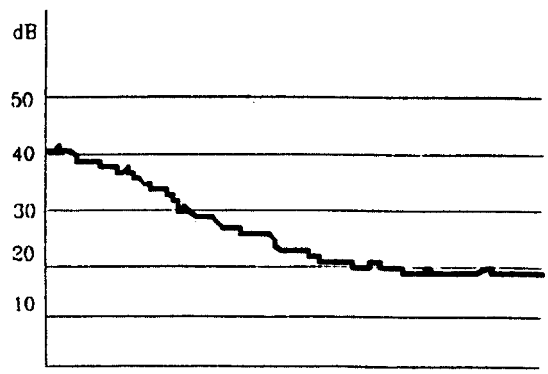


Fig. 5.2. Prediction Source Noise only.

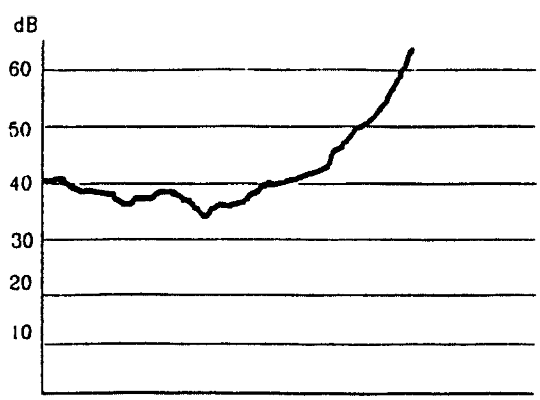


Fig. 5.3. Prediction Source Noise and Reflected Noise.

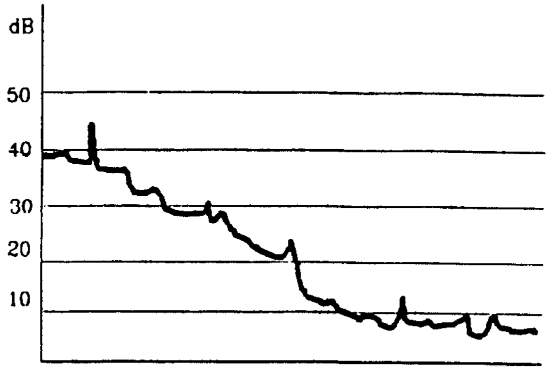


Fig. 5.4. Prediction Source Noise and Reflected Noise separately.

Ⅵ. Conclusion

The active adaptive noise controller is introduced and implemented. Using the ahead prediction and the pipe-lining technique and the DSP56001, the controller can be real-time implemented. The experimental results show that the separate prediction method improves efficiency by canceling the reflected noise, which agree well with computer simulations. Applying this method, 40 dB noise can be attenuated below 20 dB, and proposed controller is assured the stability.

For the modifications to noise control of broad space, a study for parellal processing is needed.

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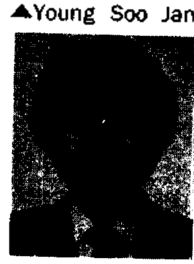

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