Digital Active Noise Control System Used Inverse model

역모델을 이용한 디지탈 능동 소음제어시스템

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ABSTRACT

The problem of active noise control has been analysed using a adaptive signal processing technique. In this methods, the adaptive signal processor or model predicts the primary sound wave travelling along the acoustic plant and generates the secondary source 180° out of phase which attempts to attenuate the undesired noise by destructive interference.

In the solutions presented here, acoustic propagation delay is considered as a part of the model which used the FIR filter. The effects of error path and auxiliary path transfer function are analyzed and a new on-line technique for error path modeling, adaptive delayed inverse modeling is presented.

In this study, using these new concepts, our system can more reduce the noise level in duct to 5dB-15dB than only using LMS algorithm system.

요 약

공조덕트계의 騷音은 사무실, 스튜디오, 반도체 공장등 정숙한 공간을 필요로 하는 장소에 있어서 큰 문제가 되어왔다. 지금가지 덕트계의 消音은 흡음재를 이용한 騷音除去方式이 주류를 이루고 있으나 이러한 흡음형 消音器의 消音性能은 500Hz 이상의 중고음 영역에 있어서 유효하나 그 이하의 주파수에서는 消音效果가 떨어져 많은 消音器를 직렬로 사용하여야만 한다.

이러한 흡음형 消音器의 단점을 극복하기 위해 저주파의 소음에 대해 원음의 위상과 180° 차이가 나고 동일한 크기의 2차음을 스피커로 부가하여 相互의 干涉效果로 소음을 제거하는 소위 能動騷音制御技術이 제기되고 있다. 본 연구에서는 제어스피커와 오차마이크로폰 사이에 존재하는 보조경로의 전달함수를 적응모델링하고 이를 보상하므로써 能動騷音制御 시스템의 성능을 향상시킬수 있는 역 모델링 기법에 대하여 연구 하였으며 MOTOROLA DSP칩 DSP56001을 이용하여 실시간 구현하였다.

구현된 능동소음 제어기는 LMS 알고리즘만을 이용한 시스템보다 5dB-15dB의 성능개선이 있음을 확인 하였다.

I. Introduction

Acoustic noise control has become a major problem for our society. The increased use of large industrial equipment such as engine, blowers and compressors after results in very high sound levels.

In addition, the growth of high density housing increases the exposure of the population to noise

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from a variety of sources. Cost constraints have resulted in a tendency to use lighter structures for buildings and transportation equipment. This may also result in an increased noise problem.

The traditional approach to acoustic nosie control has been to use passive techniques such as enclosures, barriers, and silences to attenuate the noise emitted.

In this paper, another approach is treated, which is known as active noise control. In particular, the application of adaptive digital signal processing techniques to this area could made the active noise control system to be powerful one and we will shown that.

Chapter 2, contains a review of the basic technology.

This includes principles and adaptive processing technique of the active noise control system.

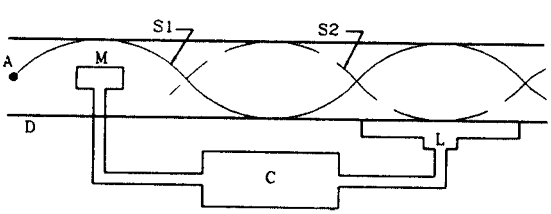
In Chapter 3, A new approach to the active noise control system in which uses inverse model and filtered-x LMS alogrithm is introduced. This approach could make the adaptive active noise controller effective and stable.

In chapter 4, the experimental system for the implementation of the new adaptive active noise controller and the result of the experiments are contained and followed the conclusion of this paper in Chapter5.

Ⅱ. The active noise control system

2.1. The principle of active noise control system

In an effort to overcome the problems of large size and ineffectiveness at low frequencies of passive silencers, the concept of active sound control was devised. The basic idea is illustrated in figure 2.1.



A: Noise Source M: Microphone C: Control Box D: Duct
L: Speaker S1: Noise S2: Secondary Source

Fig 2-1. Active Sound Attenuation concept

A microphone(M) is used to sense the undesired noise(A) and reintroduce an inverted signal through the transducer(L) that will cancel the undesired noise. It is the rapid transmission of electrical signals compare to the acoustic wave velocity that makes such a system possible.

This system is simple and low cost, but can be used only enabled relatively narrow range to be attenuated.

This is due to the limitations of non adaptive systems, simple delay in the controllers, inadequacies of the tranducers, and acoustical interactions between the secondary sound generator and input microphone.

In the recent time, digital active noise control system is studied which contains model identification part as well as adaptive controller part.

Figure 2-2 is the concept of adaptive digital active noise control system.

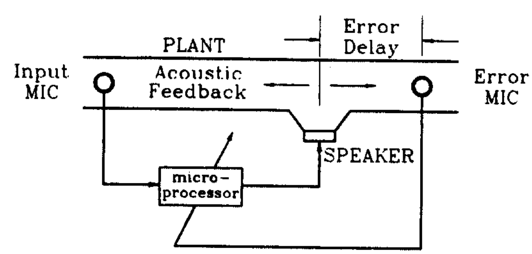


Fig 2-2. Schematic diagram of active sound control system

2.2. Adaptive signal processing

This digital controller in an active noise control system could be considered as an adaptive signal processor. Adaptive signal processor may be used for purposes of prediction, identification of filtering.

In digital active noise control system, the goal is to use the processor as an adaptive controller that will generate the proper signal to control the secondary sound source(loud speaker L) in such a manner as to reduce the undesired noise.

The adapation mechanism may be use the least mean square (LMS) algorithm, lattice algorithm, least squares(LS) algorithm, as will as mamy others. The LMS algorithm is relatively simple adaptive filter described by Widrow that can be used to model an unknown system.

This algorithm uses an all zero transversal filter with coefficients that are adaptively varied to minimize the mean square error.

The error is defined as the difference between the plant output and the model output as shown in figure 2-3.

The mean square error is a quadratic function of the coefficients. Through the use of an estimate of the gradient of this mean square error function based on instantaneous values of the error and input, an expression is obtained for updating the coefficients on a recursive basis.

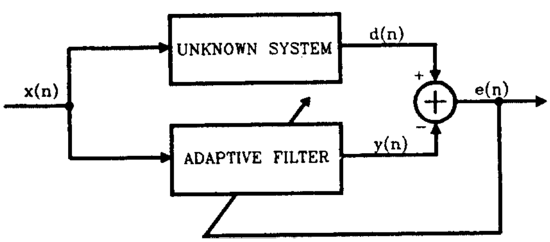


Fig 2-3. Principle of adaptive filter

The brief introduction is followed.

Let X(k) and Y(k) be the input and output respectively and W(i) be i-th filter coefficient, then,

$$Y(k) = \sum_{i=p}^{n-1} X(k-i)W(i) \tag{2-1}$$

If we define the vector $\mathbf{X}(k)$, $\mathbf{W}$ as follow

$$\mathbf{X}(k) = [X(k), X(k-1), \cdots, X(k-N+1)]^T$$
$$\mathbf{W} = [W(0), W(1), \cdots, W(N-1)]^T$$

Then the equation (2-1) can be described in the following vector type.

$$Y(k) = \mathbf{X}(k)^T \mathbf{W} \tag{2-2}$$

Now, the mean square error could be selected as the performance measure such that,

$$J = E\{e^2(k)\} \tag{2-3}$$

where E means expection value and e(k) is the error between output Y(k) and desired value d (k).

From equation (2-1)~(2-3), we can get,

$$J = D - 2\,\mathbf{W}^T\underline{r} + \mathbf{W}^T R \mathbf{W} \tag{2-4}$$

where

$$D = E\{|d(k)|^2\}$$
$$R = E\{\mathbf{X}(k)\mathbf{X}^T(k)\}$$
$$\underline{r} = E\{d(k)\ \mathbf{X}^T(k)\}$$

The signal d(k),X(k) could be assumed to be wide-sense stationary in many application fieid, and the coefficients which minimize the performance measure J of equation (2-4) could be computed recursively.

$$e(k) = d(k) - y(k) \tag{2-5}$$

$$\mathbf{W}(k+1) = \mathbf{W}(k) + 2\lambda e(k)X(k) \tag{2-6}$$

With equation (2-1), the above two equations consists of an adaptive algorithm using stochastic gradient method, called least mean square(LMS) algorithm.

Ⅲ. Adaptive controller using inverse model.

3.1. Instruction

The digital adaptive noise control system is depicted schematically in the figure 3-1.

In this figure, P.E.F and S is the transfer of forward path, error path, feedback path of sound and speaker, respectively.

A and B are a part of the controller.

The input and output signal of the controllerM can be descreibed as.

$$Y_1 = U_1 - S\,F\,U_2 \tag{3-1}$$

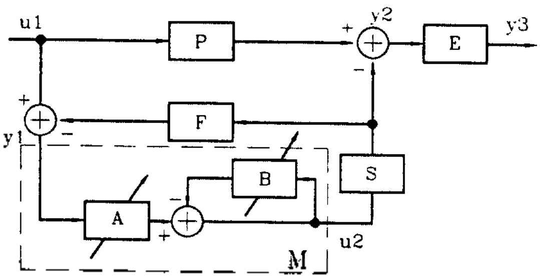


Fig 3-1. Active Noise Control system configuration with acoustic feedback error path acoustics and non-ideal source

$$U_2 = M\, Y_1 \qquad (3\text{-}2)$$

and with the equation (3-2), from the equation (3-1) we get

$$Y_1 = d_1 / (1+SFM) \qquad (3\text{-}3)$$

And canceled noise Y_2 and sensed residual Y_3 are

$$Y_2 = PU_1 - SU_2 \qquad (3\text{-}4)$$

$$Y_3 = EY_2 \qquad (3\text{-}5)$$

After some algebraic treatment, we get the transfer function form U_1 to Y_3 as follow.

$$Y_3 / U_1 = (P - (SM / (1+SFM)))E \qquad (3\text{-}6)$$

We want to make the controller such that it make residual error Y_3 zero although there exist nonzero U_1.

This can be achieved by take

$$P = SM / (1+SFM) \qquad (3\text{-}7a)$$

or $\quad M = P / (S(1-PF)) \qquad (3\text{-}7b)$

The transfer function of the controller shown in figure 3-1 is

$$M = A / (1-B) \qquad (3\text{-}8)$$

and we now could say that if we choose A and B as (P/S) and PE respectably, the residual error will be zero.

In addition, if we want to make the controller which make the residual error be zero, we should know A and B or make M equal to (A/(1−B)).

3.2. Inverse model and filtered-x LMS algorithm

In this section, we consider or form of modeling, inverse modeling, to model the transfer function M.

The inverse model of a system having an unknown transfer function is itself a system having a transfer function which is in some sense a best fit to the reciprocal of the unknown transfer function.

Figure 3-2 shows the type of adaptive inverse modeling. The unknown system, called plant in control system, to be modelled is seen with input s_k. Noise n_k is added to its output and the noisy output x_k is available as an input to the adaptive filter, the inverse model.

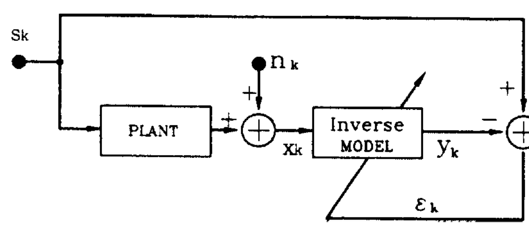


Fig 3-2. Type of adaptive inverse modeling

Upon convergence, the adaptive filter output is a best least squares match to the plant input, and the transfer function from plant input S(k) to filter output Y(k) is unity.

The ability to form an invese with low mean square error will generally limmited by three factors :

1. The presence of plant noise n_k causes the output of the adaptive inverse filter to be noisy, raising the mean square error.

In the presence of noise, the transform of t. converged adaptive impulse response, a least squares solution, will not generally be the reciprocal of that of the plant.

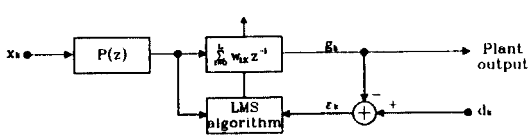
2. The plant is generally a causal system, and the signal s_k will be delayed as it goes through a physical plant.

Such conditions would require the inverse to be a predictor, a task that can only be performed approximately by a causal adaptive filter in a statistical sense.

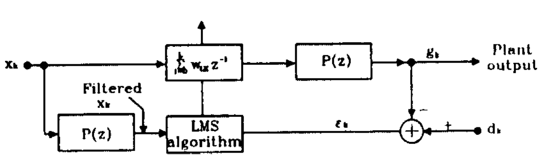
To overcome this problem, a delayed inverse is used in many applications.

3. The adaptive filter, when realized as an adaptive transversal filter, has a finite impulse response. Such an impulse response can only approximate an infinite impulse response when the latter is required to realize the optimal inverse.

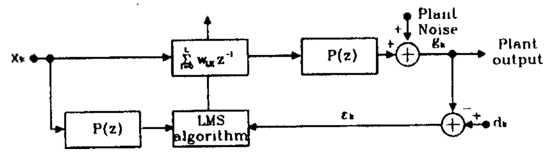
The problem of the plant noise, the first limitation of the above, has motivated the development of a new algorithm, the "filtered-x" LMS algorithm, which allows adaptation of the inverse filter placed forward of the plant in the cascade sequence. Assuming commutability of the plant and the adaptive filter and assuming that the adaptive coefficient of the system of figure 3-3(a) are initialized these cofficients would undergo identical chance and would be suitable for the adaptation problem of figure 3-2. With this new algorithm, figure 3-3(c), where there is a plant noise, show that the input of the adaptive filter does not contain the plant noise. The plant noise will of course cause an additional misadjustment in the filter coefficient, but not directly and the effect is small compare to that of figure 3-2 system.



(a)



(b)



(c)

Fig 3-3. Development of the filtered_x LMS algorithm

3.3. Active noise controller with inverse model and filtered-x LMS.

In the previous sections we considered the digital active noise control system, inverse modeling, and filter-x algorithm.

Our active noise control system consists of all as these as shown in figure 3-4.

The adaptive filter c is used to model the path, speaker s, error path E and sensor microphone M2. Another filter M is the inverse model, which is resiprocal to the transfer function C. This the digital processor of our digital active noise control system contains three FIR filter and two LMS adaptive algorithm.

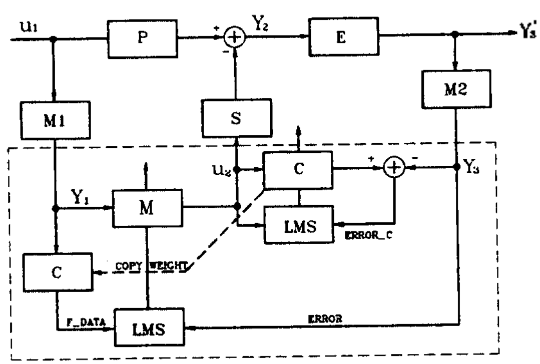


Fig 3-4. Block diagram of active Noise Control system using adaptive inverse model

Ⅳ. Design and experiment of the active noise control system using the DSP chip

4.1. System hardware configuration

Figure 4-1 show schematic diagram of the active noise control system proposed in this paper. Analog signal taken from the microphone are preamplified, filtered by antialiasing filters, and then connected to digital signal by A/D converters. The active noise control system, stored in the program memory and the two data memorys of the digital signal processor(DSP) board, processor input digital signal to produce output for generate cancelling waveforms. Output digital signal is converted to analog signal, and filtered by reconstruction filters. This signal is amplified enough to drive the control speaker. Noise waveform generated from the control speaker cancels noise from the source speaker.

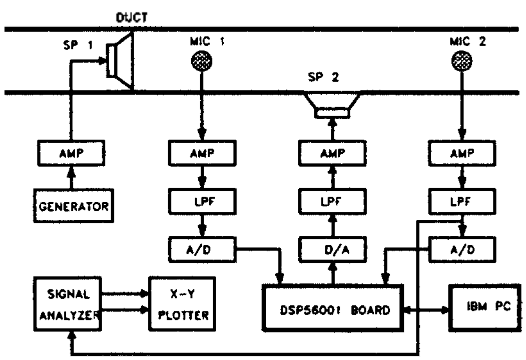


Fig 4-1. Schematic diagram of Active Noise Control system

The residual noise detected by the error microphone is preamplified, low-pass filtered, and A/D converted. Signal analyser computer frequency components of the error signal. The specification of the implemented active noise controller are as follows :

1)A/D converter(AD568), D/A converter (AD767), A/D converter(AD568) by Analog Devices Provides 12-bit resolution, 4[μsec] conversion time, and 72dB dynamic range.

2)Motrola's DSP56001 provides 24-bit data, two 56bit adders, and 144dB dynamic range. If consists of program memory, X-and Y-data memories, 8 adder registers, and modifier registers. The modifier register form circular buffer to update two parameter. The DSP chip prepetches two data from X-and Y-data memory, and executed 24-bit data multiplication and 56-bit addition in one instrution cycle. DSP 56001 process 13.5 million instruction per seconds(13.5 MIPS), and computer 1024-point compex FFT in 2.45 msec.

3)Acoustic duct

In experiment, we used a cylindlical acryl duct represented in Fig 4.2

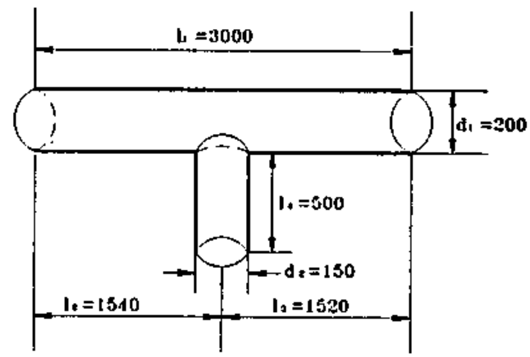


Fig 4-2. The duct size using in this experiment

4)Microphone

Condenser microphone with 12 mm diameter.

5)Speaker

Woofer with 130 mm diameter.

4.2. System softwere configuration

We used filtered-x LMS algorithm described in

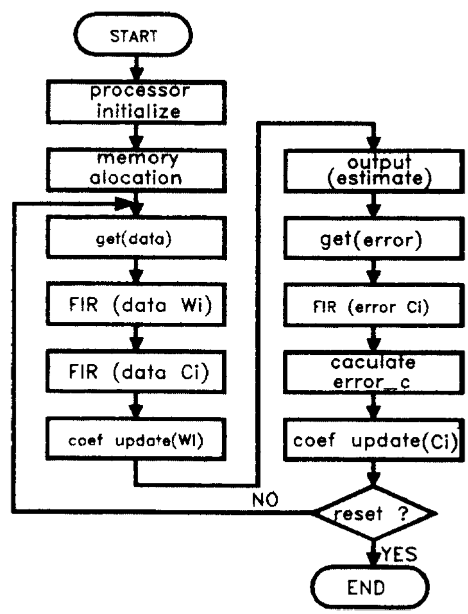


Fig 4-3. Flow chart of Filterd_x LMS algorithm using delay adaptive inverse model

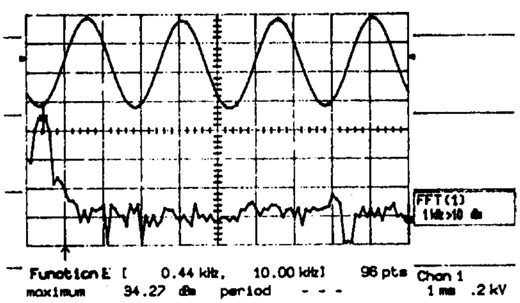
section 3.4 for modeling noise and generating control signal for the adaptive filter. Fig 4.3 show the flow chart of the filtered-x LMS algorithm using adaptive delayed inverse model.

4.3. Experiment result

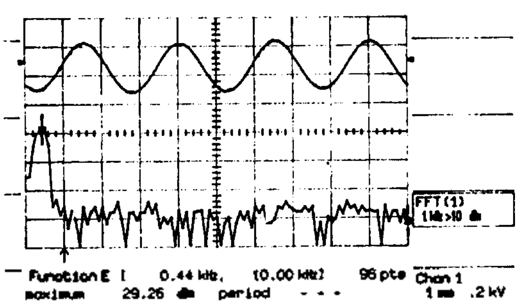
In experiment, we measured the performance of the adaptive delayed inverse model for auxiliary and error paths. We obtained electrical signal directly from the first speaker, the simulated noise sources. Therefore acoustic feedback did not occur. To measure system perpormance, we compared the system① used the LMS algorithm and the system② used filtered-X LMS algorithm with adaptive inverse model for the single frequence noise source at 240, 280, 340, 380 and 440Hz, respectively. Table 4-1 show the experimental equipment implemented through this study. Figure 4-4 show convergence characteristic of the system① and the system②. Table 4-1 show that the system② achived 5~15 dB improved attenuation effects over the system②.

Table 4-1. Attenuation effect of controllers

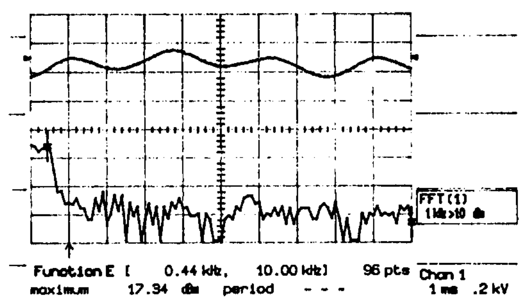
	Amplitude (max [dB])			Attenuation effect	
Frequency of the input signal [Hz]	input(A)	residual noise of system①(B)	residual noise of system②(C)	A−B	A−C
240	34.95	36.37	24.20	−1.42	10.75
280	34.82	19.76	14.30	15.06	20.52
340	35.44	24.58	14.98	10.86	20.46
380	32.86	14.76	10.00	18.10	22.86
440	34.27	29.26	17.34	5.01	16.93



(a)noise signal and spectrum in 440 Hz



(b)residual and spectrum in system①



(c)residual and spectrum in system②

Fig 4-4. Characteristic ANC system in 440 Hz

V. Conclusion

This paper proposed the system using the adaptive delayed inverse model and the filtered-x LMS algorithm as a solution to the problem of error path of the active noise controller in a duct and auxiliary path including error microphone. We also implemented the real time control system using the DSP processor. Experiment results show 5-15 dB attenuation effects over the LMS system.

When the residual noise becomes too small, as the adaptive filter reduces noise, the system, sometimes becomes unstable because of the small value of the auxiliary filter C that estimates the transfer function of the erro path. To overcome this problem, seperate signals are applied to the input of the auxiliary filter C.

In this experiment, we constructed feedback signal directly from the noise source speaker to accurately measure the inverse modeling effect, but in order to apply to the actual system, the acoustic feedback problem should be considered in the algorithm. When noise is stochastic as in fan noise, multichannel processing dividing input signal to fixed frequency bands would produce bigger attenuation effect.

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