THE SIMULATION AND IMPLEMENTATION OF AN ACTIVE NOISE CONTROL SYSTEM IN A LABORATORY DUCT

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Abstract

In this paper a single-channel feed-forward ANC system has been primarily simulated in an experimental duct to reduce low frequency acoustic noise. Moreover, the robustness of the FXLMS adaptive algorithm to modeling error has been emphasized by this simulation. In the implementation phase, acoustic feedback often caused instability, and the problem was overcome using a heuristic change in system setup. At last three adaptive algorithms, such as FXLMS with non-acoustical sensor, FXLMS with acoustical feedback neutralization, and FURLMS were implemented. The results show that tonal acoustic noise can be attenuated by ANC system over 20dB that is apparently recognizable in laboratory ambient.

1 Introduction

One of the most successful applications of Active Noise Control Systems has been in ducts so far. It is the basic idea due to the wide-range applications of ducts in HVAC systems. Fan noise is one of the common disturbances that is excessively produced in electronic devices such as transformers, computers as well as air conditioning systems and industrial exhausts [3]. This acoustic noise can harmfully affect the human life physically and psychologically. As the traditional methods of sound absorbers for noise attenuation loose their performance in low frequencies, Active Noise Control technique has been suggested as an ideal alternative. Although the first idea about ANC goes back to 1936 [4], practical implementations started in 80’s decade thanks to the quick improvement of DSP chips. This method is also widely used for reducing low frequency complex vibrations in mechanical structures.

In an ANC system, a digital controller processes acoustic noise and issues a signal with the same magnitude and the opposite phase in order to cancel the main noise signal. Consequently due to the acoustic interference between the anti-noise signal and the main signal, the acoustic noise will be possibility eliminated. The simplest realization of ANC system has been in narrow ducts. If duct length is larger than its dimensions, the sound propagation can be supposed one dimensional in the case of plane waves and uniform distribution. The operation of ANC system is based on superposition principal in linear systems that is approximately true here [1].

In this paper a single-channel feed-forward ANC system has been implemented using three different adaptive algorithms, and the experimental results are compared.

2 Single-channel ANC system

A single-channel ANC system in a duct is illustrated in Fig.1, known as "feed-forward" method [3]. The ANC system should be able to estimate the channel acoustic path transfer function. As the features of acoustic source and environment are usually time-varying, an adaptive filter estimates the response from the reference sensor to the error sensor. Therefore digital filters accompanied by adaptive algorithms -usually LMS family- are used.

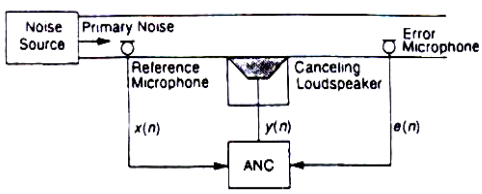

Fig1. Single-channel feed-forward ANC system in duct

The anti-noise signal made by the adaptive filter should pass through several compartments such as the D/A converter, reconstruction filter, amplifier and speaker. As this signal will be sampled by the error microphone, it has to pass again through acoustic path from speaker to error microphone, anti-aliasing filter, pre-amplifier and A/D, to be received by controller. So the transfer function from speaker to error microphone should be estimated as well. This is referred to as the "secondary path" $S(z)$, shown in Fig.2. In this figure, $P(z)$ denotes the acoustic path transfer function from reference microphone to error microphone, whereas $F(z)$ refers to the acoustic feedback path transfer function from secondary speaker to the reference microphone [3].

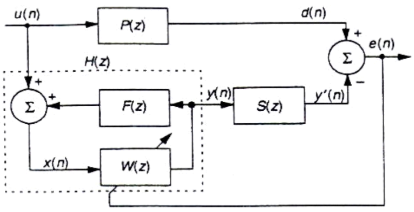

Fig2. ANC system with acoustic feedback and secondary path

3 Linear adaptive filters

Linear adaptive filters are usually categorized to finite-impulse-response (FIR) or infinite-impulse-response (IIR). In this paper both types have been used. The modeling filters for both secondary path and acoustic feedback path, as well as ANC controller for FXLMS algorithm are FIR, whereas ANC controller for FURLMS algorithm is chosen IIR.

4 The filtered-X LMS adaptive algorithm

The simple and famous LMS algorithm can not be stable in ANC system; because according to Fig.2 the filter output signal $y(n)$ encounters phase shift while passing through $S(z)$. The solution seems to apply a similar phase shift to the reference signal before gradient estimation [2]. The block diagram of ANC system using FXLMS algorithm is depicted in Fig.3. By using $\hat{S}(z)$ which models $S(z)$ to make filtered reference signal $\mathbf{x}'(n)$, the mentioned solution is realized. It has been shown [6] that this algorithm is robust to estimation error in $\hat{S}(z)$; so this filter is usually realized in terms of relatively few coefficients. The filter coefficients are updated using the gradient method:

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu \mathbf{x}'(n) e(n) \qquad (1)$$

In the case of acoustic feedback path, Fig.3 will be replaced by Fig.4, referred to as the "feedback neutralization" [1]. In this block diagram, $\hat{F}(z)$ models acoustic path transfer function.

Since the acoustic signal is fed back one sample later from speaker to reference microphone, a unit delay element should be considered.

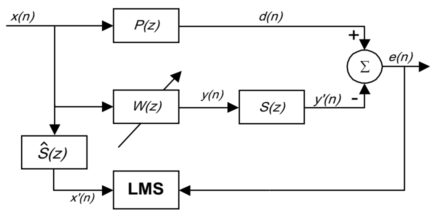

Fig3. ANC system with FXLMS algorithm

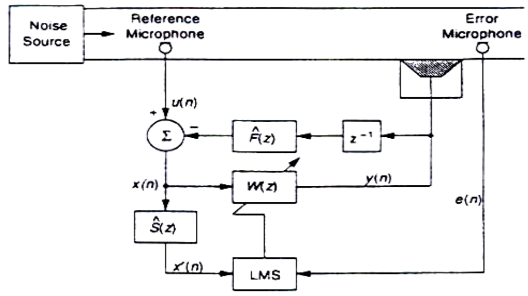

Fig4. ANC system with acoustic feedback neutralization

5 The filtered-U recursive LMS algorithm

An application area in which adaptive IIR controllers have been very successful is the active control of plane sound waves in uniform ducts, such as in air conditioning systems [1].

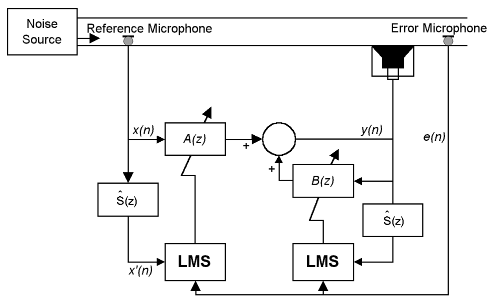

Fig5. ANC system with FURLMS algorithm

This is because an IIR filter can accurately model required plant response for ANC controller implicitly without compensating acoustic feedback effect. When individual responses of secondary path, feedback path and plant response are combined together to calculate the frequency response of the optimum controller, due to the sharing of common poles and the special form of the zeros of the individual responses, the resulting controller has a much simpler form than that of the individual responses in the duct. Thus the structure of a simple IIR controller is very well suited to the problem of controlling plane sound waves in ducts with acoustic feedback. The block diagram of a single-channel ANC system with FURLMS algorithm has been illustrated in Fig.5. The forward and feedback compartments of IIR filter coefficients will be updated as the following:

$$\mathbf{a}(n+1) = \mathbf{a}(n) + 2\mu.\mathbf{x}'(n)e(n) \qquad (2)$$

$$\mathbf{b}(n+1) = \mathbf{b}(n) + 2\mu.\hat{\mathbf{y}}'(n-1)e(n) \qquad (3)$$

6 Physical system description

Fig.6 illustrates the primary experimental system setup in laboratory. The channel is made of PVC with 4.5inch in diameter and 2.5m in length. The noise speaker has been located at the beginning of duct while the secondary speaker is fixed at the end of a short branch orthogonal to the main channel. The reference and error microphones have been located on the surface of the duct. The Data Acquisition Card was PCL-1800.

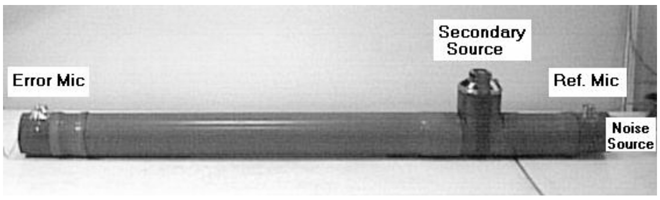

Fig6. The primary system setup in DSP Laboratory

7 System identification for secondary path modeling

The identification signal was chosen a white noise with uniform distribution and variance 0.09. This signal, produced in computer, reaches to speaker via D/A, reconstruction filter and an amplifier, and then propagates in duct. This sound is received by error microphone after passing through duct, pre-amplifier and anti-aliasing filter. Then it comes again back to computer via A/D. The input/output data were practically produced for secondary path modeling in this way. These two data records as input signal $x(n)$ and output signal $y(n)$ were given to LMS algorithm to estimate secondary path transfer function.

8 Simulation of single-channel ANC system

In many engineering applications, the simulation is considered as one step before the implementation. In the simulation stage, input/output data as well as estimated transfer function were achieved from real setup and then the FXLMS algorithm run without relating to hardware and equipment (just within PC and so that the acoustic feedback effect was ignored in this step). While in implementation phase, the corresponding relation has been established and the acoustic disturbance was practically reduced. By the method described in previous section, an FIR model of order 128 was achieved for secondary path. The filter coefficients which describe the impulse response have been shown in Fig.7. As illustrated in this figure, the values of the first 12 coefficients are close to zero. Assuming sound speed to be 340m/s, the sound delay for traveling secondary path distance (2.15m) is roughly 6.3ms. Since the sampling frequency is 2KHz, this delay is seen in the first 12 coefficients of the designed filter.

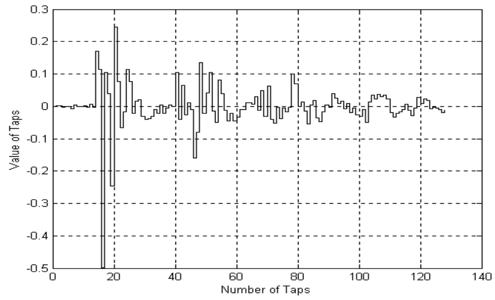

Fig7. The impulse response of secondary path modeling filter

As in this paper the robustness of FXLMS algorithm is investigated by simulation of ANC system the required transfer functions are $S(z)$ and $\hat{S}(z)$. $S(z)$ has been determined before, and an FIR filter of order 64 was estimated as an inaccurate model for $\hat{S}(z)$. As mentioned earlier, due to the robustness of the FXLMS algorithm [6], this estimation does not cause instability.

So far $S(z)$ and $\hat{S}(z)$ have been prepared and it is just necessary for ANC system simulation to obtain input signal $x(n)$ and output signal $y(n)$. In this step the speaker at the beginning of the duct issued a tonal sinusoidal signal (300Hz) and the error microphone received this sound at the end of the duct. This input/output data -produced from real setup- was applied to FXLMS algorithm. The power of normalized voltage received by error microphone during convergence process has been shown in Fig.8. It can be seen that ANC system has significantly reduced the power of the acoustic noise 17dB in error microphone location.

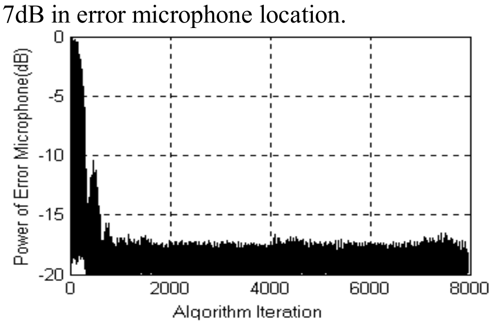

Fig8. The received power by error microphone with algorithm iteration

In Fig.9 the power spectrum density of main noise and error microphone signal have been compared. The horizontal axis has been normalized in terms of Niquist frequency. Since the sampling frequency of the main noise is 6KHz, the peak in 0.1 is equal to 300Hz and this shows the noise power spectrum density has been cut down 17dB.

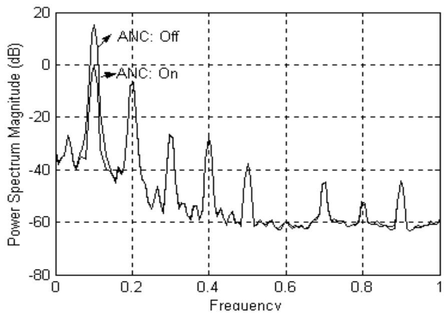

Fig9. The power spectrum density of main noise and error microphone signal

9 Implementation of single-channel ANC system

During the implementation phase, a major problem was encountered: acoustic feedback often caused instability in FXLMS algorithm. The first step had to be the reduction of this corrupting signal. So we changed the experimental setup as recommended in [5]. Fig.10 illustrates the new system setup in DSP lab. The secondary speaker is fixed at the end of a short branch mounted at $45°$ on the main body of the duct. In this stage the sampling rate was 10KHz and the acoustic noise was selected sinusoidal tonal noise at frequency 400Hz.

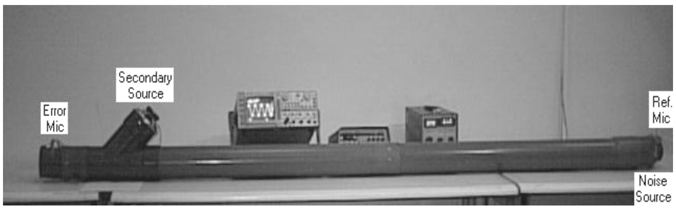

Fig10. New system setup in laboratory

9.1 ANC system realization through non-acoustical sensor

In this section the most common method for attenuating tonal noise has been implemented. As in the case of tonal noise the only essential information is the frequency of main noise; the feed-forward ANC system can be implemented without reference sensor. The most important advantage of this method is the omission of acoustic feedback [3]. This means that the frequency components from noise speaker -or reference signal- are directly given to ANC controller (here PC). Fig.11 shows the power spectrum density of received signal in error microphone before and after applying ANC system. According to this figure, PSD of primary noise has been reduced 21dB at the error microphone location.

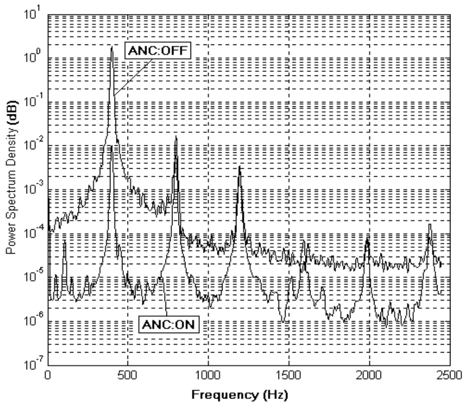

Fig11. PSD of error microphone signal, before and after ANC system with non acoustical sensor and FXLMS algorithm

9.2 ANC system realization with acoustic feedback neutralization

Although by using the new setup, the power of acoustic feedback signal was clearly decreased, the remaining acoustic feedback could still cause occasional instability. To overcome this problem, we realized the single-channel feed-forward ANC system with FXLMS algorithm and feedback neutralization, depicted in Fig.4. Hence the transfer function of this acoustic feedback path, $\hat{F}(z)$, should be modeled as an FIR filter by using the modeling method described in section 7. In addition to step size, a "leakage factor" was considered in coefficient updating equation to achieve more algorithm robustness to modeling error [3]; as:

$$\mathbf{w}(n+1) = \alpha\mathbf{w}(n) + \mu\mathbf{x}'(n)e(n) \tag{4}$$

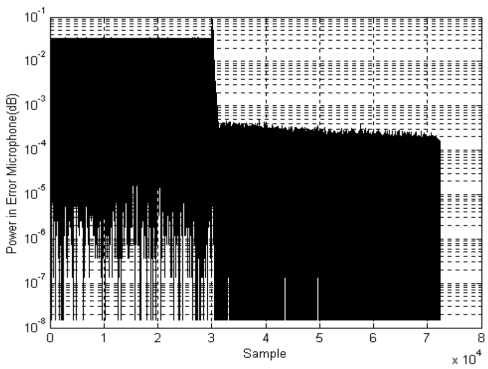

Fig12. Noise power reduction in error microphone during FXLMS algorithm convergence with acoustic feedback neutralization

Fig.12 shows the power reduction of the received signal in error microphone during quick algorithm convergence. This figure states that after applying ANC system the power of primary noise has been reduced 21dB at the error microphone location.

9.3 ANC system realization with FURLMS algorithm

In previous section three FIR filters were used: $\hat{S}(z)$ that models the secondary path, $\hat{F}(z)$ that models the acoustic feedback path and $W(z)$ that models the plant as ANC controller, whereas in FURLMS algorithm, $\hat{S}(z)$ remains unchanged, while an IIR filter is used instead of two FIR filters. Fig.13 shows the power reduction of received signal in error microphone during algorithm convergence. This figure states that after applying ANC system the power of primary noise has been reduced 24dB at the error microphone location.

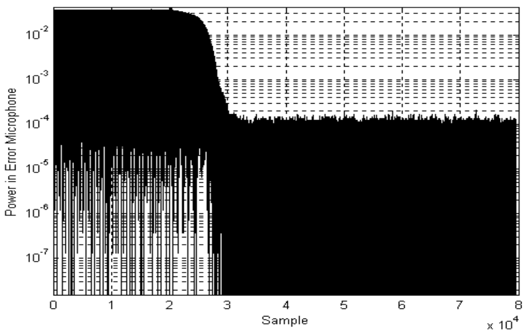

Fig13. Noise power reduction in error microphone during FURLMS algorithm convergence

10 Conclusion

In this paper, a single-channel feed-forward ANC system was simulated and implemented in an experimental duct. The results verify that the FXLMS adaptive algorithm is robust to secondary path modeling error. Although the exact model was not applied to algorithm, due to the adaptive feature of ANC system, the tonal sinusoidal noise was reduced up to 17dB. However, in implementation stage the acoustic feedback caused considerable instability. Then by applying a heuristic change in the setup, the instability problem was effectively removed. Consequently by using FXLMS algorithm with feedback neutralization, the tonal acoustic noise was reduced 21dB. Furthermore the FURLMS adaptive algorithm was implemented as well. It was shown in comparison, that the FURLMS algorithm could have more noise reduction while being absolutely stable. Although FIR filters are more common in engineering applications, the IIR one is preferable in this special case as recommended in [1, 3]. That is because the acoustic feedback has considerable effect in duct, and using FURLMS algorithm does not need implicitly require any acoustic feedback compensating. The implementation of this system is on progress on TMS processor in our DSP Lab.

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