ACTIVE CONTROL OR PASSIVE CONTROL? (OUTLINE OF ACTIVE NOISE CONTROL AND IT'S APPLICATION TO PRACTICAL PROBLEMS)

KEN'ITI KIDO

Chiba Institute of Technology (Tudanuma 2-17-1, Narashino 275, JAPAN)

This paper overviews the history, the principle and some key technologies of the active noise control and points out that the passive control should be incorporated with the active control to improve the annoyance of noise. That is, the suppression of higher frequency noise by the passive method should be jointly used with the active control. There are feed forward method and feed back method in the active control system, but this paper concentrates the feed forward method only as the stability is important. This paper concentrates the technology using FIR filter, and the renewal algorithm of the coefficients of FIR adaptive filter is shown in unified form. Next, the relation between the arrangement of additional sound source and the reduction of noise radiation is given according to the numerical computation and the control of power transformer noise is briefly explained as a practical example carried out by the author. Finally, the change in A-weighted noise level and the sound quality in hearing by the ANC is discussed.

1. Introduction

The active control of sound and vibration has much progressed in recent years by the progress in digital signal processing technology. Even now, the researches on the active noise and vibration control are very active. Since the first international symposium on the active control of sound and vibration held in Tokyo [1] in 1991, there have been many conferences on the active control. Another symposium "ACTIVE 95" will be held in the beginning of July 1995 in Newport Beach, California, U.S.A. And there are various publications on the active control [2, 3, 4, 5]. There are some industrial products using the active control technology [6, 7]. And someone expects that the active control will grow up to the main technology in the noise control and the conventional passive technology will go out of use.

This paper overviews the short history of the progress of active control and the principal technology of active noise control and discusses the role of active control in the noise control engineering.

2. The outset of active control

The first ideas of active control of sound and vibration concern with the rolling of ships. Thonycroft proposed to move a weight on the deck [8] in 1891 and Motora proposed to steer the side fins [9] for the same purpose. The Motora's method was practically used for a 3.600 ton class big ship in 1922 and this method is now widely used after some improvements [10]. The methods using weight or water in tank are now used not only for the control of ship but also for the supression of the quake of highrise buildings [11, 12, 13]. Some different methods are developed and practically used in which the cabin is supported by the oil pressurized pistons which are controlled to keep the cabin at rest [14]. The similar technologies are used in the vehicles on land [15].

While the proposal of the first idea of the active control of acoustic noise [16] was in 1930's and the first trial [17] of the active noise control was carried out in 1940's. And it was 1960's that the active noise control had become a practical research object of a number of researchers [18, 19, 20].

3. Principle of the active noise control

As described in the previous chapter, the active control of vibration has already been practically used in various field. But the active control of noise is not yet so widely used. One of the most crucial reason is considered to be the frequency range to be controlled. In case of ship rolling or building vibration, the frequency range is usually lower than a few Hz, while the audible frequency range is higher than 20 Hz. Usually, the frequency range of noise to be controlled is between 20 or 100 Hz and 10.000 Hz.

Figure 1 illustrates the principle of active noise control. The noise can be diminished by the superposition of the sound wave having the same wave form. How to make the sound of such the wave form and how to extend the controlled space are the main problems in the active noise control.





In the view point of controlled space, the active noise control is roughly divided into three categories. One of them is to reduce the total noise radiation, another one is to reduce noise level of a specified direction and the remaining one is to reduce noise level at a specified space.

The first one can be achieved by setting the additional source very close to the primary noise source if the dimension of the noise source is much smaller than the wave length of the objective noise. The ANC of duct noise belongs to the first category as the cross sectional size of the duct should be smaller than the wave length of the controlled noise. The second one can be achieved even if the size of the noise source is not so small. The author succeeded in the active control of power transformer noise by the directivity synthesis [19]. The third one can be used to reduce the noise level at the ears of a specified person.

In the other view point, the ANC technology can be divided into two categories. The first one is the feed forward method and another one is the feed back method. The signal to drive the additional source, that is called here the control signal, is synthesized using the wave form of primary noise source in the feed forward method as shown in Fig. 2 (a) and is synthesized using the resultant sound in the feed back method as shown in Fig. 2 (b).



Fig. 2. Block diagram of active noise control system (a) feed forward system (b) feed back system.

It is anticipated that the feed back system can be unstable as there is a closed loop and the feed forward system is stable. But, there can be a closed loop which causes instability of the system even in the feed forward system as long as the sound radiated by the additional source propagates to the reference microphone whose output signal is used to synthesize the control signal.

The wave form coherent to the primary noise source is necessary for the synthesis of the control signal. When the source signal is picked up by the reference microphone set close to the primary noise source to synthesize the control signal, the sound radiated by the additional source propagates to the reference microphone through the noise transmission path. The acoustic feed back through the additional sound source to the reference microphone is inevitable in such the system.

Some method to diminish the acoustic feed back is proposed. Figure 3 shows two typical methods. The first method uses the difference of the outputs of two microphones set at symmetrical points to the additional sound source to synthesize the control signal and the microphone at the downstream is used as the monitor microphone (error microphone) as shown in Fig. 3 (a). The second method uses an electric circuit (Wave synthesizer 1 in Fig.) which is equivalent to the transfer function from the driving point of the additional sound source to the output terminal of the reference microphone as shown in Fig. 3 (b). The stability of the system is achieved by such the methods.



Fig. 3. Two methods to diminish the acoustical feedback from the additional source to the reference microphone.

4. Adaptive control of filter coefficients

An FIR filter is usually used to synthesize the control signal. The schematic diagram of FIR filter is shown in Fig. 4, where the triangles indicate the multiplication with the filter coefficient w(i) and the direction of signal flow. The filter coefficients are adaptively controlled so as to minimize the output of the monitor microphone.

The original noise output transmitted through the noise transmission path is expressed as follows:

$$y(n) = \sum_{i=0}^{N-1} h(i) \cdot x(n-i)$$
(4.1)

where y(n) — the output noise at n, x(n) — the source signal at n, h(n) — the impulse response of noise transmission path.



Fig. 4. Active noise control system with FIR filter.

The noise output can be made zero subtracting the output signal z(n) from the output noise y(n), if the coefficients of the filter w(i) are made the same as h(i).

The filter coefficients to synthesize z(n) should be adaptively controlled to minimize y(n) - z(n), because the environment of sound field, the temperature, doors or moving man etc. changes always. And the theory of the system identification is effectively used in the active noise control.

The output of the FIR filter is expressed by the following equation:

$$z(n) = \sum_{i=0}^{N-1} w_n(i) \cdot x(n-i), \qquad (4.2)$$

where $w_n(i)$: *i*-th coefficient of filter after *n*-th renewal.

The well known renewal algorithm [21, 22] of the filter coefficients is the LMS algorithm shown in the following:

$$w_{n+1}(i) = w_n(i) + \mu \cdot e(n) \cdot x(n-i), \tag{4.3}$$

where, e(n) = y(n) - z(n) — the error signal, μ — the step size parameter.

Some alternatives of the LMS algorithm have been proposed which are expressed by the following equations:

$$w_{n+1}(i) = w_n(i) + \mu \cdot e(n) \cdot \text{sign}[x(n-i)], \qquad (4.4)$$

$$w_{n+1}(i) = w_n(i) + v \cdot \mu \cdot e(n) \cdot \operatorname{sign}[x(n-i)], \qquad (4.5)$$

$$w_{n+1}(i) = w_n(i) + v \cdot \mu \cdot e(n) \cdot x(n-i)^m \cdot \operatorname{sign}[x(n-i)], \qquad (4.6)$$

where,

v=1 for $x(n-i) \ge$ thershold, v=0 for x(n-i) < threshold.

Eq. (4.6) involve Eqs. (4.4) and (4.5) as the special cases.

Using the following delayed LMS algorithms, the computation time can be shortened to 2/3 of the normal LMS algorithm [23].

$$w_{n+1}(i) = w_n(i) + \mu \cdot e(n-1) \cdot x(n-i-1).$$
(4.7)

5. Decrease in the total noise radiation

It is considered that the power of radiated noise can be decreased putting the additional sound source close to the primary noise source as those sound sources compose a dipole. As for the dipole source, it is well known that the sound pressure decreases rapidly with the increase in the distance from the sound source. But the



Fig. 5. The attenuation of radiated power by setting additional sound sources for the arrangements shown in the top of each figure, where the volume velocity of the additional sound source (a) is the same as that of the primary source except for the sign.

noise source and the additional sound source can not compose a perfect dipole as the dimension of noise source is not always much smaller than the wave length. The shapes and the arrangement of the primary and additional sources have a big effect on the total noise radiation.

We investigated the effect of the arrangement of primary and additional sound sources on the sound power radiation by numerical computation [24]. Figures 5 (a) through (c) show the changes in the radiated sound power for various combinations of primary and additional sound sources as the functions of frequency. The geometry of the primary and the additional sound source used in the numerical computation is illustrated in each figure.

In all the cases, the decrease is limited to the low frequency range. In the low frequency range, the attenuation is proportional to the reciprocal of frequency, that is, the slope is -6 dB/octave, when only one additional source is used, and is proportional to the reciprocal of squared frequency, that is, -12 dB/octave, when two or more additional source sencircle the primary source. The biggest attenuation is achieved when the additional source is at the center of the primary source.

An arrangement of additional source shown in Fig. 6 is consisted from those results for duct noise suppression system. Using such the arrangement, the closed loop gain through the additional source to the reference microphone can be neglected.



Fig. 6. Schematic diagram of active noise suppression system for duct noise where the additional sound source is set at the center of duct end.

6. Noise reduction by directivity synthesis

We have succeeded the noise control of transformer noise [19, 25] by use of the directivity synthesis. The transformer noise is composed of discrete frequency component and the random component. The former component is caused by the magneto-strictive vibration of iron core and the latter by the cooling fan. The former one is composed of even numbered harmonics of power frequency and stable both in frequency and amplitude. This type of steady sound can be frequently an object of

complaint as it has a special sound quality. The control of directivity pattern by the analogue computers was planned as the sound is very stable.

Figure 7 shows the outline of the system. The directivity pattern is controlled by use of the additional loudspeakers surrounding the power transformer. The loudspeakers are driven by the signals which is synthesized from the wave form of the power line. Therefore, there is no feedback loop in the system and the system is completely stable.

The amplitude and the phase of each frequency component are controlled so as to make the power sum of the output of monitor microphones set on the field minimum by an adaptive controller. The experimental system is composed of three controllers, three microphones and three loudspeakers. The amplitudes and the phases of the frequency components up to 300 Hz are effectively controlled.



Fig. 7. Outline of transformer noise control system by use of adaptive directivity control.



Fig. 8. Transient change in the noise level of the transformer.

Figure 8 shows the transient characteristics of the system. The system reaches the final state in 20 seconds after switch is turned on. The largest attenuation in sound pressure level is observed in 100 Hz component. Figure 9 shows the directive distribution of the sound pressure in 100 Hz component. Figure 9 shows the directive distribution of the sound pressure level on a semicircle of 10 meter radius. Using three monitor microphones, the sound pressure level within 30 degrees decreases more than 10 dB. But the sound pressure increases in some direction. More loudspeakers and monitor microphones will be necessary to decrease sound pressure level in a wider angle. The loudspeaker interval should be less than a quarter wavelength. It may give a practical limitation of the use of this system for the higher frequency range.





7. Changes in A-weighted noise level and sound quality

The ANC system suppresses the noise in the low frequency range but the higher frequency noise remains unchanged. Therefore, much decrease in the A-weighted sound level by the ANC can not always be expected.

Figure 10 shows some examples of the spectrum of power transformer noise in which the attenuation of the low frequency components is assumed according to our former experiments. The attenuation assumed here in 100 Hz component is 26 dB, that in 200 Hz component is 16 dB and that in 300 Hz component is 6 dB. Those values are considered to be practically maximum attenuation only by the active control method in the field.

The attenuation of sound pressure level in the flat weighting is remarkable for all the noise, but the attenuation in A-weighted level is not so. In the case of the transformer (2), the attenuation is only 2.4 dB because the transformer noise has much higher frequency components. The attenuation in the transformer (3) is more than 10 dB in A-weighted level, because the transformer is well furnished to suppress higher frequency noise.

We have carried out simple hearing test using those noise. According to the results, this type of noise suppression will get good evaluation in the noise (3), that is, the hearing evaluation was better after the attenuation than before the attenuation.

But, in the noise (2), the result was opposite. In the noise (1), we could not get clear result.

More close investigation will be necessary to make clear the relation between the change in the spectrum and the hearing evaluation. There can be another factor in the annoyance of noise than the loudness. But, it may be sure that the attenuation in the higher frequency components is important.

ATTENUATION FOR EACH FREQUENCY COMPONENT: 10 Hz: 26 dB 200 Hz: 16 dB 300 Hz: 6 dB 400 Hz: 0 dB

BEFORE ATTENUATION: FL = 75.1 dB AL = 57.3 dBAFL = 63.1 dB AL = 51.5 dBA FL = 72.2 dB AL = 54 dBAAFTER ATTENUATION: FL=55.9 dB AL=50.9 dBA FL=53.5 dB AL=49.1 dBA FL = 50 dB AL = 43.3 dBAATTENUATION: -19.2dB -6.4 dBA -9.6 dB -2.4 dBA -22.2 dB-10.2 dBA (2) (3)(1)dB dB dB 30-80 80 SP 60 60 40 40 42 22 20 17 .1 .5 1kHz . 1 5 .5 1kHz . 1 1kHz

Fig. 10. Examples of the transformer noise with the decrease of low frequency components.

8. Conclusion

By surveying the history of the active control of sound and vibration, we can see the active control technology has quite a long history and is well applied to suppress the rolling of ships and the quake of high-rise buildings. But the active control is not yet so popular in audible noise control. The main reason is in the frequency range to be controlled. We have to process complicated wave form of higher frequency in the noise control engineering. Those problems are solved by the use of FIR adaptive digital filters, where a number of computation is necessary. The difficulty in the computation speed is now become out of problem by the progress of the digital signal processors, but there remains some other difficulties in the practical applications. How to diminish the feed back from the additional sound source to the reference microphone is one of the key points. This paper introduces typical two methods for reducing the feed back. Then, the algorithms of the adaptive control of the filter coefficients are shown in a unified form. As the another key problem, the relation between the arrangement of additional sound source and the reduction of noise radiation is given according to the numerical computation. Next, the control of power transformer noise by the adaptive synthesis of directivity is briefly explained as a practical example carried out by the author. Finally, the change in A-weighted noise level and the sound quality in hearing by the ANC is discussed. As the ANC is effective only in the low frequency range, much decrease in A-weighted sound level can not be expected only by the ANC. And in some case, the annoyance of the noise can increase by the decrease in the very low frequency noise. It is important to decrease the noise of higher frequency range by the passive noise control method in the application of the ANC.

References

- Proc. International Symposium on Active Control of Sound and Vibration, April 9-11, 1991, Acoustical Society of Japan, Tokyo, Japan.
- [2] P.A. NELSON and S.J. ELLIOT, Active control of sound, Academic Press, London 1991.
- [3] S.M. KUO and D. VIJAYAN, Adaptive algorithms and experimental verification of feedback active noise control system, Noise Control Engineering Journal, 42, 2, p. 37-46 (1994).
- [4] T.J. SUTTON, S.J. ELLIOT, A.M. MCDONALD and J. SAUNDERS, Active control of road noise inside vehicles, Noise Control Engineering Journal, 42, 4, p. 137-147 (1994).
- [5] M. NISHIMURA and T. ARAI, Active control of noise radiated from duct end, J. Acoust. Soc. Jpn. (J), 45, p. 672-680 (1989).
- [6] S. SUZUKI, K. Nagayama, T. Hayashi, S. Saruta, H. Tamura, Y. Sekiguchi and K. Nakanishi, A basic study on an active noise control system for compressor noise in a refrigerator, Proc. International Symposium on Active Control of Sound and Vibration, Acoustical Society of Japan, Tokyo, Japan, p. 255-260 (1991).
- [7] S. HASEGAWA, T. Tabata, A. Kinoshita and H. Hyodo, The development of an active noise control system for automobiles, SAE Technical Paper 922086, ISSN 0148-7191, p. 1-9 (1992).
- [8] J.I. THORNYCROFT, Studying vessel at sea, TINA, 33, (1892).
- [9] S. MOTORA, On motora's ship rolling suppression system, Bull. Ship Build. Assoc., 32, (1923).
- [10] S. WATANABE, Method for the reduction of ship rolling (2), Proc. Symposium Ship Stabilization, Jpn. Soc. Ship Build., 156-179 (1969).
- [11] T. KOBORI, Active vibration control for architectural structure, Proc. International Symposium on Active Control of Sound and vibration, Tokyo, p. 87-98, 1991.
- [12] K. YOSHIDA and T. WATANABE, Active vibration control of high-rise buildings. Proc. Internat. Symposium on Active Control of Sound and Vibration, Tokyo, p. 185-194, 1991.
- [13] M. IZUMI, Control of structural vibration Past, present and future, Proc. Internat. Symposium of Active Control of Sound and vibration, Tokyo, p. 195-200 (1991).
- [14] I. TAKAHASHI, Development of ship with no pitching nor rolling cabin, Jour. Ship Build. Society of Japan, 733 (1990).
- [15] M. NAGAI, Vibration control of vehicle by active suspension, Jour. INCE/J, 12, 3, p. 197-201 (1988).
- [16] P. LUEG, Verfahren zur Daempfung von Shallschwingungen, Germ. Pat. No. 655508, Field: Jan. 27, 1933, Patented: Dec. '37. US Equivalent Process of Silencing Sound Oscilations, US Patent No. 2,043,416. Field: Mar. 8, '34, Patented June 9, '36.
- [17] H.F. OLSON and E.G. MAY, *Electronic sound absorber*, J. Acoust. Soc. Am., 25, p. 1130-1136 (1953).
- [18] J. OKDA, T. NIMURA and K. KIDO, Internal impedance of a system of electroacoustic transducers. J. IECE Jpn., 44, p. 330-334 (1961).

- [19] S. ONODA and K. KIDO, Automatic control of stationary noise by means of directivity synthesis, Proc. 6th ICA, F15-3, p. F185-188, 1968.
- [20] M.J.M. JESSEL and G.A. MANGIANTE, Active sound absorbers in an air duct, J. Sound and Vibration, 23, 283-390 (1991).
- [21] C.F.N. COWAN and P.M. GRANT, Adaptive filters, Prentice-Hall; Englewood Cliff, New Jersey 1985.
- [22] B. WIDROW and S.D. STEARNS, Adaptive signal processing, Prentice-Hall; Englewood Cliff, New Jersey 1985.
- [23] S. ISE, Hardware and software techniques for adaptive filter, J. Acoust. Soc. Jpn., 48-7, p. 501-505 (1992).
- [24] H. KANAI, M. ABE and K. KIDO, A new method to arrange an additional sound source in active noise control, Acustica, 70, p. 258-264 (1990).
- [25] K. KIDO and S. ONODA, Automatic control of acoustic noise emitted from power transformer by synthesizing directivity, Sci. Rep. Res. Inst. Tohoku Univ. (RITU), Ser. B, Part I, Rep. Res. Inst. Electr. Commun., 23, p. 97-110 (1972).