acta phys. slov. 42 (1992), No. 3, 175–181

A NON RECURSIVE ADAPTIVE SYSTEM FOR ACTIVE NOISE REDUCTION[1)]

GILLESPIE, A. F. R.[2)], London

This paper describes an adaptive system for the active attenuation of low frequency random noise in a section of an acoustic waveguide. It shows how a combination of two adaptive non-recursive filters can be used to overcome the effect of upstream radiation on the operation of active monopole attenuators. Such filters have important advantages over recursive types. Experimental results are presented showing the significant attenuations obtained on band limited random noise in the frequency range 30–500 Hz.

I. INTRODUCTION

The principle underlying active noise attenuation is that an unwanted noise can be attenuated by combining it with a secondary signal of the necessary phase and amplitude characteristics, such that interfere destructively. Although this principle could theoretically be applied to an arbitrary acoustic signal, in practice it is as yet only possible to apply it with any degree of success to the one-dimensional enclosures. The theoretical limitations underlying the restricted applications of active noise control are discussed in [1]. Even in the applications mentioned above, apart from physical limitations, there is the difficulty of designing a control system to cope with the variations that commonly occur in practise in both the characteristics of the noise and in other parameters such as air temperature and flow rate. Due to these inherent variations prior attempts at active noise control using non adaptive methods have met with limited practical success. Systems are now being used met with limited practical success. Systems are now being used based on time-varying or adaptive controllers/filters. In this paper we discuss one such digital system applied to the reduction of plane wave random noise in a length of air conditioning duct and demonstrate its effectiveness in dealing with band limited random noise. Before doing so a brief analysis of the basic monopole attenuator is given to illustrate clearly what is required of any control system aiming to provide significant attenuation.

---

1) Contribution presented at the 12th Conference on the Utilization of Ultrasonic Methods for Studying the Properties of Condensed Matter, August 29th-September 1st, 1990, Žilina, CSFR

2) South Bank Polytechnic, IoEE, 103 Borough Road, LONDON, SE1OAA, Great Britain

II. THE MONOPOLE ATTENUATOR

A diagram showing the basic monopole attenuator system structure is shown in Fig. 1. The primary noise that we wish to attenuate is detected by a microphone (2) placed upstream of a secondary acoustic source, in this case a loudspeaker (4). The signal from the microphone is then processed by the controller (3) with the aim of producing an output from the loudspeaker such that the acoustic pressure downstream of the loudspeaker is reduced, ideally to zero. For solely plane wave propagation and no end reflections this would mean zero end radiation. The second microphone (5) is only required for adaptive systems and provides an estimate of the system performance {error function} which is used to continuously control the adaptive process.

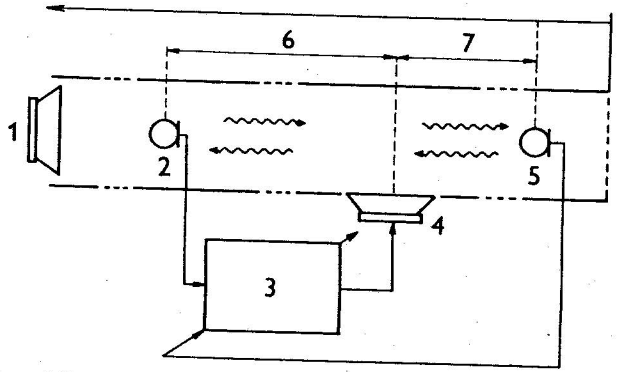

Fig. 1. Scheme of Basic Monopole Attenuator. 1. Noise Producing Source. 2. Pick-up Microphone. 3. Control System. 4. Secondary Loudspeaker 5. Error Microphone 6. Acoustic Path 7. Error Path

An idealized analysis of the required static (digital) controller transfer function to achieve zero pressure can be obtained by considering the system in control terms as shown by Fig. 2. For zero pressure the output from the loudspeaker $y'(n)$ must be

$$y'(n) = -n'(n). \tag{1}$$

Solving for the required controller function $W(z)$ to satisfy Eq. (1) we get that

$$W(z) = \frac{-P(z)}{L(z)\{1 + P(z)P(z)\}}, \tag{2}$$

where $L(z)$ is the response of the speaker, amplifier etc., $P(z)$ is the response of the acoustic path. Two aspects of the required function are relevant to this paper. Firstly, it depends on the characteristics of the acoustic path and the electroacoustic elements of the system {lumped together under $L(z)$} both of which

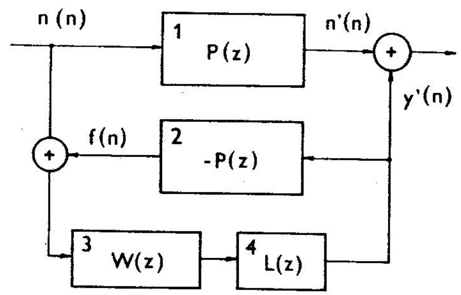

Fig. 2. Block Diagram of Monopole Attenuator. 1. Acoustic Path $P(z)$, 2. Feedback Path $-P(z)$, 3. Controller $W(z)$, 4. Loudspeaker.

will vary in practice, thus necessitating an adaptive control system. Secondly, the controller function apparently needs to be of the pole-zero, Infinite Impulse Response {IIR } or recursive type. This has led to recent systems for adaptive noise attenuators focusing on the use of adaptive IIR filters [2]. Such filters have inherent problems regarding stability and, in the adaptive case, convergence, although such difficulties can often be overcome [3]. However, form Fig. 2 it can be seen that the need for the controller to be pole-zero in form is due to the propagation of the secondary wave upstream to the pick-up microphone. Early attempts to overcome this disturbing effect and avoid the use of IIR controllers have considered the use of directional microphones or directional loudspeaker arrays, but the function of such devices is usually highly frequency dependent, thus limiting their usefulness in attenuating broad-band noise [4]. Another approach, the one used in the system described in this paper, is to use a second adaptive filter to attempt to model the upstream propagation and then remove its effect upon the signal from the primary microphone. This results in a controller that requires only Finite Impulse Response {FIR }, all zero, control elements which have important advantages in terms of stability and transient performance.

III. THE ADAPTIVE CONTROL SYSTEM

Adaptive filters are based on the estimation of a set of parameters such that they minimize a specific, usually quadratic, error function. In this application the primary error function to be minimized is derived from a microphone, placed in the noise field inside the duct. The method of parameter estimation is constrained by the need to produce an updated output at every sample, since the output of the filter is used to attenuate a real acoustic signal. The present system is shown in block form in Fig. 3 with the two adaptive elements W_1, W_2 based on the

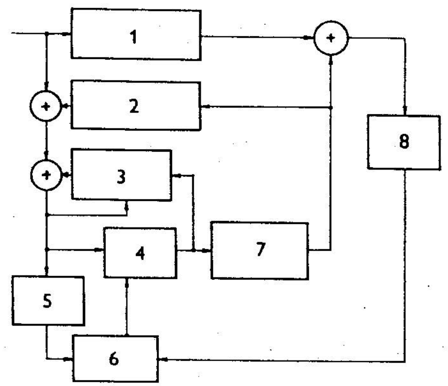

Fig. 3. Block Diagram of Experimental Adaptive System. 1. Forward Acoustic Path $P(z)$. 2. Feedback Acoustic Path $-P(z)$. 3. Interference Cancelling Filter W_2. 4. Noise Cancelling Filter W_1. 5. Error Path Model. 6. Adaptive Update. 7. Loudspeaker $L(z)$. 8. Error Path.

simplest and most extensively applied adaptive algorithm, the Least Mean Square {LMS} algorithm developed by Widrow [5]. Its properties regarding convergence and transient performance have been extensively investigated. It assumes that a given error function can be minimized by a FIR controller defined by the discrete convolution.

$$y(n) = \sum_{k=0}^{N-1} w_k x(n-k), \tag{3}$$

where $x(n)$ is the discrete input signal; $y(n)$ is the discrete output signal; w_k represents the kth filter coefficient; N is the number of coefficients in the filter. The adaptive filter W_1 is used as before to minimize the pressure at the error microphone while the second adaptive filter W_2 is used to minimize the effect of the upstream propagation. The configurations of W_1, W_2 are commonly termed system identification and interference cancelling respectively. A full description of these configurations is given in [5].

In the case of W_1 the error function results from the superposition of the output of the controller and the primary noise and thus the instantaneous error signal $e(n)$ is a measure of the instantaneous pressure at the point of superposition and thus $e(n) = n'(n) + y'(n)$ or:

$$e(n) = n'(n) + \sum_{k=0}^{N-1} w_k n(n-k), \tag{4}$$

where $n'(n)$ represents the primary noise after allowing for the acoustic path and $y'(n)$ represents the output from the loudspeaker.

The error function $\varepsilon(n)$ to be minized is a measure of the average pressure, i.e. $\varepsilon(n) = E\{e^2(n)\}$. In the case of W_2 the error function equates with removing the feedback signal $f(n)$ from the primary noise $n(n)$. Since $f(n)$ and $n(n)$ will be uncorrelated, in the case of random noise this can be done successfully, for a fuller discussion of this see [6]. The error path model is required for successful convergence of the adaptive controller W_1 but for this investigation it was set by hand to a static time delay that ensured convergence.

The advantages of the above system can be seen when solving for the static transfer functions of the filters required, $W_1(z)$ and $W_2(z)$. To eliminate the feedback signal $f(n)$ the required transfer function $W_2(z)$ can be found to be

$$W_2(z) = -P(z)L(Z). \tag{5}$$

Using this we now get that the controller function required to minimize the pressure at the error microphone, i.e. $e(n) = 0$, is given by

$$W_1(z) = -L^{-1}(z)P(z). \tag{6}$$

The above equations show that using this approach only FIR filters need to be implemented.

IV. EXPERIMENTAL RESULTS

An experimental test rig was set up to investigate the operation of a prototype attenuator based on the methods outlined above. This primary loudspeaker Fig. 1 (4) was driven by bandlimited noise in the frequency range 40-500 Hz. Both adaptive filters used contained 128 coefficients and the sampling frequency used was 3200 Hz. The digital adaptive controller was based on the Motorola 56001 fixed point processor with 16 bit A/D and A/D convertors. The pressure spectra at the error microphone before and after the system was applied are shown in Fig. 4. The error microphone was positioned centrally at the end cross section of the duct, approximately 60 cm downstream of the secondary loudspeaker. The levels of attenuation are clearly demonstrated by Fig. 5, which shows reductions of 15 dB over the range 80 - 480 Hz with overall average reduction of 20 db. With the coefficients of W_2 fixed the system was also tested with sinusoidal signals and gave comparable reductions to those obtained on broad band noise.

V. CONCLUSIONS

The experimental results presented demonstrate the effectiveness of the system in significantly reducing lowfrequency random noise, while at the same time retaining the inherent stability of non recursive filtering. For a fully self adaptive

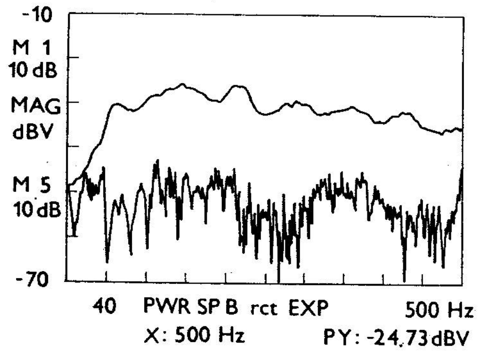

Fig. 4. Spectra at output cross section when control point inside the duct.

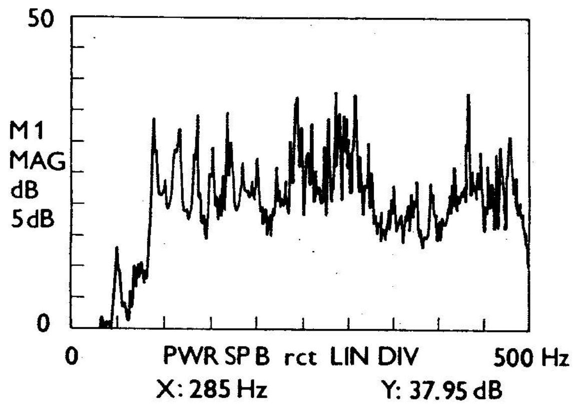

Fig. 5.

system accurate on-line modelling of the error path under realistic assumptions, imperfect cancellation and appreciable reflections will have to be implemented. This is discussed more fully in a companion paper, but no inherent difficulties are apparent at present.

REFERENCES

[1] Flowcs Williams, J. E.: Proc. Royal Society of London *A395* (1984), 63.

[2] Eriksson, L.J., Allie, M. C., Greiner, R. A.: IEEE Trans. on ASSP *35* (1987), 433.

[3] Stearns, S.: IEEE Trans. on ASSP. *20* (1981), 763.

[4] Eghtesadi, K., Leventhall, H.: Acoustics Letters *4* (1981).

[5] Widrow, B. W., Stearns, S. D.: *Adaptive Signal Processing* (1985), Prentice-Hall, Englewood Cliffs, NJ.

[6] Windrow, B.W., et al.: Proc. IEEE. (19875), 162.

Received December 6th, 1990
Accepted for publication April 3rd, 1991

НОВАЯ НЕРЕКУРСИВНАЯ АДАПТИВНАЯ СИСТЕМА ДЛЯ АКТИВНОГО ПОДАВЛЕНИЯ ШУМОВ

В работе показано примемемие адаптивной системы активного подавления низкочастотных случайных шумов в части акустического волновода. Показано, что с применением комбинации двух адаптивных не рекурсивных фильтров можно преодолеть эффект встречного излучения на монополные ослабители. Такие фильтры имеют значительное преимущество в сравнении с рекурсивными. Приведены экспериментальные результаты которые показывают на значимое ослабение в полосе ограниченного случайного шума в диапазоне частот 30–500 Гц.