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ACOUSTIC ASPECTS OF ONE–DIMENSIONAL SYSTEMS FOR ACTIVE NOISE CONTROL[1]

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This paper gives a brief description of the general principles of active attenuation techniques to the reduction of airborne noise in one-dimensional systems, i.e. acoustical waveguides such as ducts and pipes. Some aspects concerning the influence of the acoustic environment on the so-called basic system identification problem are considered. The considerations and experimental results presented might be useful for the development of acoustic plant modelling principles in the framework of a complete active attenuation system functionning under non-ideal conditions.

I. INTRODUCTION

The essential principle of active attenuation is the combination of a secondary acoustic wave with the original wave, such that they interfere destructively. According to this quite old basic concept described in many sources (see, for instance, [1]) the complex electroacoustic system should contain at least an input microphone to detect the primary noise and a processing system to generate the electrical signal required for driving a cancelling loudspeaker. Effective noise control requires the loudspeaker output signal to be a precise inverse of the primary sound waveform. Early systems used fixed filters for this aim [2].

However, usually in real applications the primary acoustic noise source contains a broad band of frequencies, and electroacoustic transducers are non-ideal, inside the duct there is often significant air flow generating pseudo-sound and finally there is the problem of the influence of the sound generated by the cancelling loudspeaker on the input microphone. These problems make it very difficult to produce a proper model of the complete system and, as a result, to provide effective attenuation. In addition, in most applications the characteristics of the sound to be attenuated vary with time due to environmental changes, changes in the source or the equipment etc., making it impossible to achieve the desired effect without using adaptive filters.

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Let us consider the difficulties associated with the simplest active attenuation application, from the acoustic point of view, related to the use of active techniques, the reduction of plane waves in ducts.

II. THE ONE-DIMENSIONAL SYSTEM

A typical monopole duct control system is shown in Fig. 1. The primary noise, generated by a sound source –1 (in most systems it is usually a fan) propagates along the duct –2. A pick-up microphone –3 is placed a distance upstream of a secondary cancelling noise source –4 and is used to detect the sound pressure of the primary wave. This distance is usually termed the acoustic path –5. The detected noise is processed by the system –6, which generates a cancelling signal, which is then radiated inside the duct by an electroacoustic transducer (loudspeaker) –4. A second microphone placed at a distance (this distance called the error path –7) from the cancelling source is used to give a measure between the desired system performance and the actual performance. The signal from this microphone is then used to adjust the digital controller with the aim of minimizing the sound pressure at the error microphone.

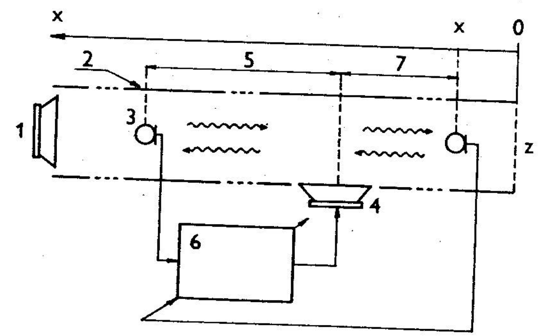

Fig. 1. Schematic of monopole attenuator.

A simplified block diagram of the adaptive attenuation system is shown in Fig. 2, where: block 1 represents the real acoustic path of the primary signal; block 2 – the acoustic error path. Block 3 – a model of the error path and block 4 – the digital controller forming the electronic part of the system – 5. The separation of the blocks 3 and 4 inside the system –5 has been done to stress problems of interest. In reality they form part of an integrated control system.

The detected noise $n(t)$ is processed by the system to produce an output from the cancelling noise source equivalent to $n'(t)$, where $n'(t)$ is the signal considering the acoustic path between microphone and cancelling source. For ideal cancellation

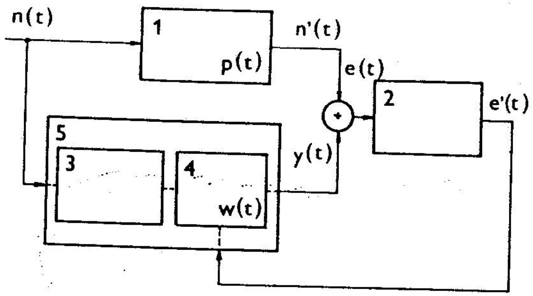

Fig. 2. Block diagram of the system.

to occur it is necessary that $y(t) = -n'(t)$. The signal $y(t) = n(t) * w(t) * l(t)$ is the output acoustic signal from the system, where: $l(t)$ is the impulse response of the cancelling noise source and the corresponding electronic circuits; $w(t)$ – impulse response of the controller. Thus $w(t) * l(t) = -p(t)$, where $p(t)$ is the impulse response of the acoustic path.

It is clear that the required controller transfer function depends on the acoustic transfer function between the pick-up microphone and cancelling source and on the electroacoustic transducers characteristics. Since the acoustic transfer function depends on many environmental parameters (the air flow rate, thermodynamic parameters etc.) to provide effective attenuation the system must adjust to these variations as well as to any changes in the electroacoustical equipment.

To overcome the effect of noise variations requires an adaptive digital controller. The non-zero error acoustic signal $e(t)$ is detected by an error microphone after passing through the error path. The output from this microphone $e'(t)$ is used to adjust the digital controller and minimize the sound pressure at the point where the error microphone is placed. This is equivalent to minimizing the expected value of the square of the error signal, a typical least squares problem [3].

Omitting the details of the electronics used, described in general features throughout literature, let us consider the problems connected with the acoustic part of the system.

III. GENERAL QUESTIONS OF ACOUSTIC PATH MODELLING

For real application of the adaptive attenuation technique at least two main problems associated with the modelling of the primary acoustic path and error path are known at present. Problems of pseudo-sound generated due to the presence of air-flow in real duct systems are the subject of a different consideration and will not be described in this paper.

The first problem is that the output acoustic signal influences the input of the system due to the back propagation of acoustic waves inside the duct towards the pick-up microphone (along axis X in Fig. 1), termed the feedback path and frequently discussed in literature [4]. Various efforts have been made to design a directional cancelling loudspeaker or set of loudspeakers [5] as well as directional pick-up microphones. However, such electroacoustic transducers are highly frequency dependent and the use of such approaches introduces considerable complexity into the design of the required electroacoustic equipment, especially at low frequencies. An approach which takes into account the acoustic feedback in terms of an overall adaptive model seems to be more convenient. At least two possible solutions to the problem in question are known [6]. One of them is the representation of the acoustic feedback as one part of a single overall model and results in poles of the transfer function of the model. These poles can theoretically be cancelled using a pole-zero or infinite impulse response adaptive filter. A second approach is to use an additional filter to attempt to model the upstream propagation and then remove its effect from the signal produced by the pick-up microphone.

A similar approach to the second method outlined above using two adaptive filters has been utilized in the experimental prototype of an adaptive attenuation system for this work. The implemented, non-recursive adaptive system, which is described in detail in a previous paper, shows clear reductions at all tested frequencies and appears to be more reliable in terms of stability under the conditions of an existing standing wave pattern influencing the output signal from the pick-up microphone (Fig. 1).

The second problem is connected with the necessity to model the acoustic error path between the cancelling source and the error microphone in the framework of the adaptive system with the aim to provide adjustment of the system to possible changes in the acoustic error path, transfer function of microphone etc. A possible solution of this problem using on-line modelling and requiring an independent noise source to get the composite transfer function of the cancelling loudspeaker, acoustic error path and error microphone are known [7]. Nevertheless, such models do not take into acount reflections of the sound waves from the duct elements downstream of the cancelling source and error microphone nor possible impedance changes in this area. This results from the assumption that the model of the whole system can be evaluated under perfect cancellation resulting in the zero error signal in the area downstream of the cancelling source thus eliminating the possibility of reflections.

However, there are at least two points that have to be kept in mind when dealing with real acoustic systems usually containing such elements as bends, junctions and definitely producing some acoustic output which will cause reflections. Firstly the error control signal in real applications is detected at one point and even under ideal conditions the system could guarantee the expected attenuation only in this point but not in the whole downstream area. Secondly it is impossible in practice to achieve perfect cancellation. It means that in most practical applications the error microphone functions in a standing wave field.

IV. SYSTEM COMPLICATIONS DUE TO THE PRESENCE OF STANDING WAVES

If there is residual noise propagated in the downstream area towards the end of the duct along the direction opposite to the axis X in Fig. 1 and the presence of some passive construction elements causing reflections and back radiation, the sound field in this area can be described by the potential $\tilde{\Psi}$:

$$\tilde{\Psi}(x,\omega) = \tilde{A}(\omega)e^{-jkx} + \tilde{B}(\omega)e^{+jkx}, \qquad (1)$$

where: $\tilde{A}$ and $\tilde{B}$ – are complex coefficients representing the propagating and reflected waves respectively; x in the spatial coordinate (Fig. 1); $k = \omega/c$ are wave numbers; c is sound speed.

Equation (1) is the general solution of the Helmholtz equation resulting from the Fourier transform of the homogeneous wave equation. This can be used for describing the sound field inside a spatial area of interest in which the presence of sound sources is not supposed. The primary field parameters following from (1) are

$$\begin{aligned} \tilde{p}(x,\omega) &= j\omega\rho\left[\tilde{A}(\omega)e^{-jkx} + \tilde{B}(\omega)e^{+jkx}\right], \\ \tilde{v}(x,\omega) &= -jk[\tilde{A}(\omega)e^{+ikx} - \tilde{B}(\omega)e^{-jkx}], \end{aligned} \qquad (2)$$

where: $\tilde{p}$ is sound pressure; $\tilde{v}$ - particle velocity; ρ -medium density.

The active intensity Ia is the correlation function of the sound pressure and the particle velocity and can be represented by $Ia = 1/2Re(\tilde{p}\tilde{v}^*)$, where Re means the real part; $*$ is the complex conjugate. It follows from (2) that:

$$Ia = \frac{1}{2}k\omega\rho(A^2(\omega) - B^2(\omega)) = \text{const}\,(x). \qquad (3)$$

The value of active intensity responsible for the transfer of sound energy can be considered as a measure of the quality of an active attenuation system. According to the definition of acoustic impedance $\tilde{Z}(x,\omega) = \tilde{p}/\tilde{v}$ the active intensity can be expressed from (2), (3) as

$$Ia = \frac{1}{2}p^2(x,\omega)\left[\frac{Re(\tilde{Z}(x,\omega))}{Z^2(x,\omega)}\right]. \qquad (4)$$

It is easy to see from (4) that the reduction of waves propagating along the duct depends not only on the reduction of the sound pressure at the point x where the error microphone is placed, but also on the spatially dependant "load" impedance as and changes in that impedance. The sound pressure alone at one spatial point cannot serve as the measure of quality of the system performance except for the practically impossible case when $Z = \rho c$.

V. EXPERIMENTAL VERIFICATION AND RESULTS

The experimental prototype of the adaptive active attenuation system used for the examination of dependence of the system performance upon the "load" impedance is shown in Fig. 1 with its block-scheme in Fig. 2 and is described in general features above. Here it has to be stressed that the implementation of the electronic part of the system (block 5 in Fig. 2) was such that it allowed us to get similar attenuation levels at different error microphone placements by precise manual adjustment of the model of the error path (block 3).

The experiment was carried out at a frequency range below the lowest cut-off frequency of the duct and was from 40 Hz up to 500 Hz. The error microphone was placed at various points of the duct in the downstream area and attenuation levels at these points were adjusted to the optimal values allowed by the system. As the measure of the system performance for the different control points the levels of attenuation at the point of the output cross-section ($\tilde{X} = 0$ in Fig. 1) of the duct have been used. It is apparent from equation (4) that this attenuation level is directly connected with reduction of sound power radiated outside the duct and that this is the final aim of attenuation systems. The acoustic output of the system was loaded into a typical room.

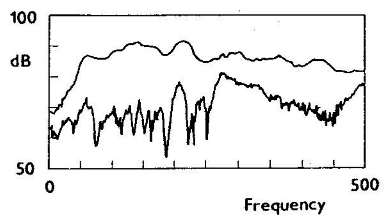

Fig. 3. Spectra at output cros section when control point at output cross section.

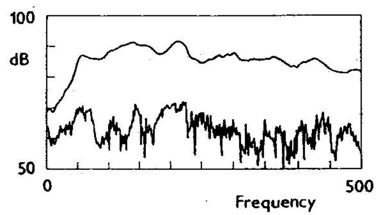

Fig. 4.

Representative results are shown in Figs. 3 and 4. Fig. 3 clearly shows the noise reduction at the output point when the error microphone control point was at the output cross-section. The overall reduction level for this case is 21.8 dB. Fig. 4

shows the reduction at the output point when the system has been adjusted to reduce the sound pressure at one of the error points inside the duct. It is necessary to note the difference in the reduction levels at the various error points, for a given control point, for all cases was less than 1 dB, while at the same time the difference in the overall attenuation levels at the output has achieved 8 dB as it is in the case shown in Fig. 4. It should be stressed that this difference appeared due to reduced attenuation in the frequency ranges which corresponded to the distance between the output cross-section and previously estimated reflection coefficients of the open end of the duct. It is easy to see in Fig. 4 the significantly worse reduction at the frequencies near 300 Hz.

The more detailed representation of the experimantal results will be given in another paper. In the conclusion of this part it is interesting to note that the differences in the sound pressure levels inside the room serving as the acoustic load, for the same quality of system performance in the terms of sound pressure reduction at one point but for different error points, could be easily distinguished by the personal.

V. CONCLUSIONS

The results presented clearly show the effect of standing waves on the performance of the system with regard to the reduction in the pressure spectra at the end cross-section and thus the reduction in radiated power. It has been clearly demonstrated that the system is effective in controlling low frequency noise but that for a totally self adaptive system accurate modelling of the error path must be undertaken that taking into account these effects.

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АКУСТИХЕСКИЕ АСПЕКТЫ ОДНОМЕРНЫХ СИСТЕМ ПРИ АКТИВНОМ УПРАВЛЕНИИ ШУМАМИ

В работе приведено основное описание общих принципов техники активного ослабления в применении подавления воздушных шумов в одномерных системах, акустических волноводах в форме канав и труб. Рассмотрены некоторые аспекты обязанные с влиянием акустического окружения на так называемую проблему определения базовой системы. Показанные предложения и экспериментальные результаты можно применить при развитии принципов моделирования акустики ландшафта при поддержке полной системы активного затухания работающей в неидеальных условиях.