inter-noise 83

CONTROL OF SOURCES FOR ACTIVE SOUND PROPAGATION IN A DUCT

V. Martin and A. Roure

Laboratoire de Mécanique et d'Acoustique du C.N.R.S.,
31, chemin Joseph Aiguier
13277 MARSEILLE - France

I. INTRODUCTION

Of all the problems to be dealt with in active sound absorption the most essential, when the purpose is the absorption over a broad bandwidth, is the control of the sources. Therefore it is necessary to improve the transfer function of an electroacoustic source in such a way that the relation between the voltage on the source and the pressure radiated from it, in the duct, is a perfect delay. This approach is of interest for plane waves in the duct i.e. for sufficiently low frequencies so that only the mode without nodal lines appears.

Figure 1 shows the synoptic schema of an electroacoustic system for active acoustic absorption(A.A.A.) [1]

- a control microphone M1
- a numerical filter with adaptable coefficients F (see II)
- a directional system of two loudspeakers (HPD) with the analog network to compensate the modified radiation in Ω (network C)
- a test microphone M2
- a processor to manage the whole system.

Remark : Since the idea is to radiate by the sources a signal which is exactly the same - but opposite - to the one given by the microphone M1, the importance of the control of the sources becomes apparent.

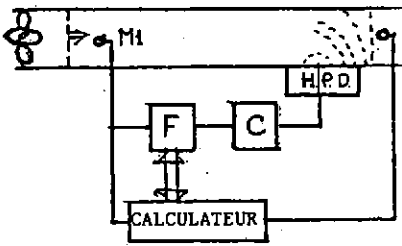
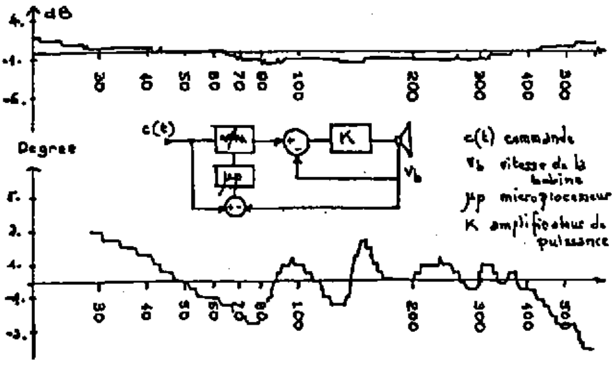


Figure 1

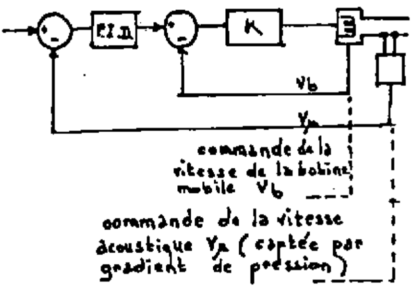
II. CONTROL OF SOURCES

The source we work on is an ordinary loudspeaker. On first approximation the system is resonant and therefore modifies the input signal. We aim to correct the transfer function to gain a follow up system.

Given that the response of the transductor depends on the acoustic impedance, a feedback control is needed. The theory shows that the pressure in the duct is proportional to the velocity of the source on the wall. Thus the velocity is, as an output signal, always compared with the input i.e. with the voltage on the source. The resonance is damped but not sufficiently due to the stability. However the new response is easy to improve with a network in front of the loop ; and moreover it is sufficient to adapt a resistance in the network in order to compensate the change of the acoustic impedance. This parameter is automatically adapted by a microprocessor. The response of the system using this technique is given below [2].



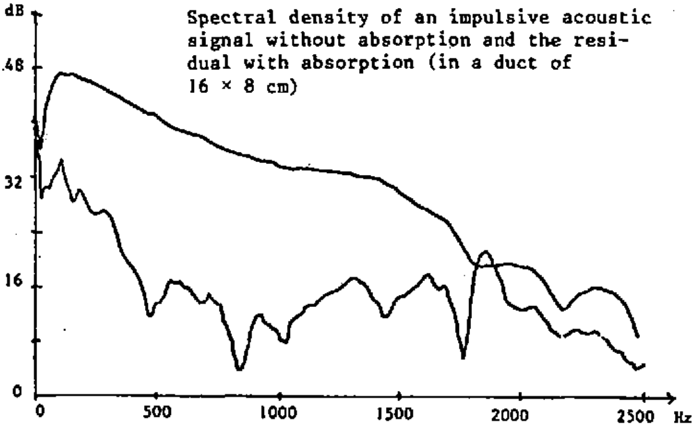
The output signal is the velocity of the driver coil. In fact the velocity of the membrane must be controlled. It is not easy to avoid the convolution between these two velocities. § 3 shows how a new technique replacing the analog network in front of the loop is more suitable. An alternative method is to control other sources such as acoustic drivers [3].



For the time being the control is performed with two loops : a feedback from the velocity of the coil, a feedback from the velocity of the air (measured by the difference between two pressures). A program optimizes the parameter values of the analog network.

III. NUMERICAL CORRECTION

It appears that, whatever care is taken in the control of the sources, this control cannot be perfect. Moreover § 1 explains that the pressure radiated in only one direction must be compensated. The compensating network, although suitable, is also insufficient on account of the precision required. All these reasons lead us to envisage a numerical filter to improve the whole system. In this way, a programmable transversal filter makes the convolution of the input signal, in real time. The process takes place with 128 programmable points of an impulse response determined by a calculator in the following way : without the disturbing noise and flow in the duct the transfer function H (ω) - between the pressure at the test microphone and the voltage at the input of F which impulse response is initialized by a Dirac - is measured. The time dependent-signal is short enough to avoid the reflection at the ends of the duct. With the inverse Fourier Transform of 1/H (ω) the calculator provides the coefficients of the filter. This technique with certain precautions due to the sampling and the I.F.T. enables us to obtain an attenuation from 15 to 30 dB over a wide frequency range (80 - 2000 Hz).



Spectral density of an impulsive acoustic signal without absorption and the residual with absorption (in a duct of 16 × 8 cm)

IV. CONCLUSION

To sum up, the mixed control (analog and digital) of the sources is very successfull in absorbing noises in ducts (for the time being only plane waves propagate). At the moment an improvement is being studied to dispense with the preliminary measure described in § 3. The coefficients of the filter will be calculated from the residual signal at the test microphone. Thus the system would be self adapted.

BIBLIOGRAPHIE

[1] M. BERENGIER, A. ROURE - Broad-band active sound absorption in a duct carrying uniformly flowing fluid - J.S.V. 68 (3), p. 437-449 (1980)

[2] V. MARTIN - Commande en vitesse d'une source électro acoustique en vue de l'application à l'absorption acoustique active - Thèse de docteur-ingénieur, Université d'Aix-Marseille II, 18/3/81

[3] A. MASLOUHI - Réalisation d'une source sonore ponctuelle asservie en vitesse - note interne LMA, 10/82.

[4] A. ROURE, B. NAYROLES - Asservissement de sources sonores - Communication dans le cadre du GRECO 45, 5/5/82