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ACTIVE CONTROL OF THE ACOUSTIC REFLECTION COEFFICIENT AT LOW FREQUENCIES

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INTRODUCTION

Sound absorption at low frequencies is problematic if conventional methods are considered since both wedge-type and resonance absorbers become very bulky, and the latter are effective in narrow frequency intervals only. On the other hand, low frequency noise can be quite disturbing, and in room acoustics a short reverberation time at low frequencies is necessary if good speech intelligibility is required.

In principle, active control of the acoustic impedance of a wall offers an alternative to passive absorption especially for the low frequency range. An active wall lining for application in room acoustics consists of loudspeakers fed by the incident sound wave (via microphones and properly adjusted electronic networks) in such a way that the acoustic input impedance, and hence the acoustic reflection coefficient, is controlled to a specified value. A practical realization of this concept has been approached in several steps, beginning with one-dimensional model experiments.

ONE-DIMENSIONAL ARRANGEMENTS

A commercial impedance tube (Kundt tube) has been used for these experiments. The control loudspeaker is attached to the open end (Fig. 1), and in the simplest case one control microphone picking up the sound field is connected to the speaker through a filtering amplifier with adjustable gain and phase shift (described by a complex transfer function $V(\omega)$). By appropriate control setting, the apparent acoustic reflection coefficient of this active system can be adjusted to values below 5 % at low frequencies, but above 400 Hz the tendency to feedback instability prevents satisfactory

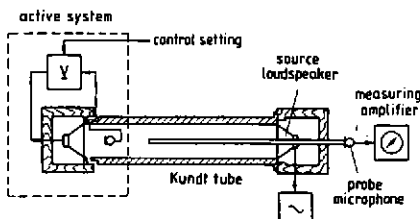


Fig. 1. Experimental arrangement with one microphone and hand control.

results (Fig. 2, broken line).

Better results are obtained with a two-microphone arrangement permitting the standing wave field to be separated into incident and reflected wave (Fig. 3). The microphones are a distance Δx apart, and the travel time of the wave, $\tau = \Delta x/c$, is compensated or doubled by delay units τ

(assuming plane wave propagation) so that, e.g.,

$$P_{2\tau}(t) = P_1(\Delta x, t - \tau) + P_r(\Delta x, t - \tau) = P_1(0, t) + P_r(0, t - 2\tau). \quad (1)$$

The contributions of the reflected wave to the microphone signals are cancelled in the difference signal

$$P_3 = P_{1\tau} - P_2 = P_1(\Delta x, t - 2\tau) - P_1(\Delta x, t) = P_1(0, t - \tau) - P_1(0, t + \tau). \quad (2)$$

Similarly, the incident wave is cancelled in

$$P_4 = P_{2\tau} - P_1 = P_r(0, t - 2\tau) - P_r(0, t) = \pm [P_1(0, t - 2\tau) - P_1(0, t)]. \quad (3)$$

The third delay unit transforms P_3 from Δx to $x = 0$ so that

$$P_5 = P_1(0, t - 2\tau) - P_1(0, t), \quad (4)$$

and the complex reflection coefficient is given by the ratio

$$\underline{r} = P_4/P_5 \quad (5)$$

(if sinusoidal signals are used).

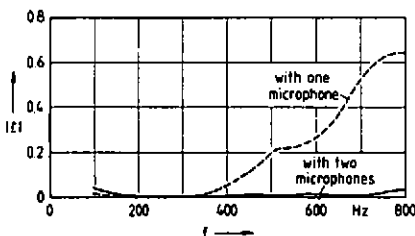


Fig. 2. Minimum obtainable reflection coefficient for sinusoidal sound. Complex gain V readjusted for each frequency of measurement.

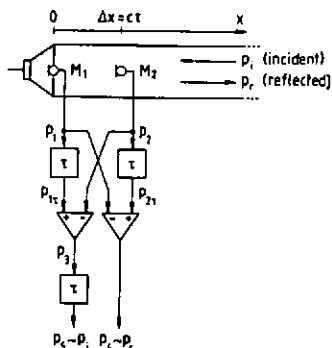


Fig. 3. Electronic separation of the standing wave field into incident and reflected wave.

The easy electronic measurement of the reflection coefficient instead of scanning the standing wave field speeds up the measurement with almost no loss of accuracy; but the main advantage of this device is a stabilization of the active system. In fact, using p_5 as feedback signal, no instability could be observed for control settings realizing reflection coefficients from $\underline{r} = 0$ to 1.5. The minimum reflection obtained is plotted as solid curve in Fig. 2.

For a broadband application, the open-loop gain of the active system should be flat in amplitude and phase. This can be achieved by an appropriately chosen amplifier response $V(\omega)$. In our experiments with a moving-coil woofer (membrane radius 9 cm), the filter required to compensate the frequency response of all other units could be approximated by a 270° phase shifter. In this arrangement, the magnitude of the reflection coefficient can simply be controlled by varying the amplifier gain. As an example, Fig. 4 shows the frequency dependence of the energy absorption coefficient for optimum non-reflecting adjustment; other values of α can be realized likewise. Similar results have been obtained with a specially developed electret loudspeaker the advantage of which is a much smaller depth of construction (about 1 cm).

An automatic adjustment of the amplifier setting by closed-loop feedback control has been investigated, too (for details see [1]). Such a device may be of interest for noise control in systems with varying open-loop gain such as ventilation ducts with fluctuating speed of air flow.

THE THREE-DIMENSIONAL CASE

Proceeding from the one-dimensional Kundt tube to the three-dimensional free-field, one encounters several difficulties: spatial energy spreading, oblique incidence, interaction between neighbouring active sources, and diffraction waves emerging from discontinuities in the wall. In order to develop an active wall

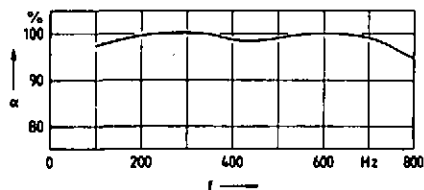


Fig. 4. Maximum energy absorption of an active broadband system with two microphones and constant-phase-shift filter.

lining composed of loudspeakers with control microphones in front, it is necessary to know the sound field produced by a single loudspeaker and by planar arrays on a rigid wall. Such sound fields have been studied for various speaker configurations and parameter combinations by computer

simulation. It could be shown by comparison with experimental results that the analytical far-field approximation for a piston radiator [2] describes the sound field of a single cone loudspeaker sufficiently well at distances of more than one membrane diameter, while the nearfield has to be calculated by solving numerically the Huygens-Rayleigh integral [2]. Fig. 5 shows as an example the calculated sound field of an array of 13×3 loudspeakers fed with equal time signals, but Hamming-weighted amplitudes (see symbols at the bottom of the figure) and linear phase shift producing main radiation at an angle of -45° . With constant-amplitude feeding, the side lobes are more pronounced. The 180° phase jumps in the direction of minimum radiation are caused by a sign change of the resultant when passing through the minimum. The funnel-shaped sink is supposed to indicate an eddy of the energy flux. Such irregularities complicate the derivation of the control signal from the sound field. The actual state of the continuing computations and experiments with a 3×3 loudspeaker array in an anechoic room will be outlined in the oral presentation at the conference.

REFERENCES

- [1] K. Karcher, "Aktiv beeinflussbare Wandimpedanz bei senkrechtem Schalleinfall." Dissertation Göttingen 1982.
- [2] H. Stenzel, O. Brosze, "Leitfaden zur Berechnung von Schallvorgängen." Springer-Verlag, 2. Auflage 1958.

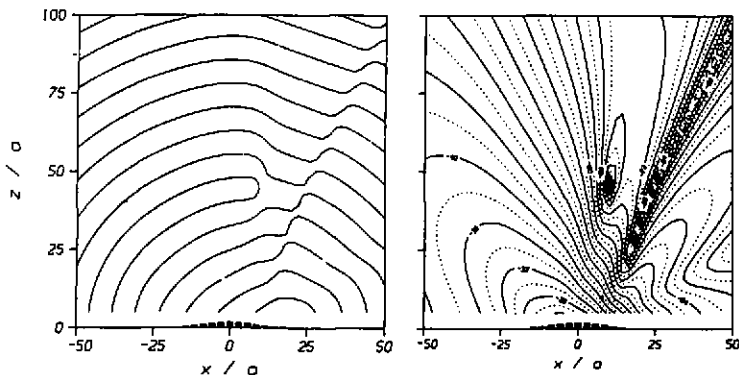


Fig. 5. Calculated contours of constant phase (left) and amplitude in front of a baffled 13×3 loudspeaker array, Speaker radius $a = 9$ cm, spacing $3a$, frequency 250 Hz. Feeding with linear phase shift and Hamming-weighted amplitude. Contour steps: π resp. 2 dB.