Practical Approach for Application of Impedance Control for Noise Attenuation

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Introduction

Active noise control (ANC) becomes more and more an important issue, because of increased restrictive regulations regarding noise emission, e.g. in employment protection. Instead of using passive noise reduction methods, which cannot always be adopted, active control applications can provide efficient sound pressure attenuation in the lower frequency range. The cancellation of a disturbing primary noise field by a destructively interfering secondary sound pressure at the area of interest is widely known and has been the subject of many publications.

For the global reduction of noise in enclosures the secondary noise sources should be located either in the vicinity of the primary source or at a maximum of the excited acoustic modes. For the application of the ANC method the error sensors should be distributed globally as well.

In the late 80's and early 90's several papers described the theoretical work on other control mechanisms like power absorption and power output minimization with global effects on the acoustic sound field, see references [1]-[4].

This paper deals with a practical approach for an ANC-system with a local allocation of actuators and sensors which is designed to absorb maximal sound power by changing the radiation impedance of a loudspeaker. The method is based on measuring active sound intensity at the centre axis of the loudspeaker membrane. While minimizing the sound intensity the actuator will absorb sound power. For the practical implementation the signals obtained from a phase corrected pair of sensors consisting of an accelerometer and a microphone serve as input for a control algorithm. This method is applied on a simplified one – dimensional test bed.

Theoretical Approach

The power W_{S} radiated from a single source can be defined as

$$W_{\mathrm{S}} = \frac{1}{2}\mathrm{Re}\left\{\underline{p}_{\mathrm{S}}\underline{q}_{\mathrm{S}}^{*}\right\} \quad , \tag{1}$$

with $\underline{q}_{\mathrm{S}}$ as the complex secondary source strength or volume velocity. For a piston source like an idealized loudspeaker $\underline{p}_{\mathrm{S}}$ is the averaged complex pressure over the radiating surface of the secondary source. In the presence of a primary source the resulting complex pressure is the sum of the primary pressure $\underline{p}_{\mathrm{SP}}$ at the surface of the secondary source and the pressure $\underline{p}_{\mathrm{SS}}$ caused by the secondary source itself.

$$\underline{p}_{\mathrm{S}} = \underline{p}_{\mathrm{SP}} + \underline{p}_{\mathrm{SS}} = \underline{p}_{\mathrm{SP}} + \underline{Z}'_{\mathrm{SS}}\underline{q}_{\mathrm{S}} \, . \tag{2}$$

Hereby $\underline{Z}'_{\mathrm{SS}}$ is the acoustic radiation impedance of the secondary source in absence of any other acoustic source. With (2) the radiated sound power can be written as:

$$W_{\mathrm{S}} = \frac{1}{2}\mathrm{Re}\left\{\left(\underline{p}_{\mathrm{SP}} + \underline{Z}'_{\mathrm{SS}}\underline{q}_{\mathrm{S}}\right)\underline{q}_{\mathrm{S}}^{*}\right\}. \tag{3}$$

NELSON et. al. stated that this equation is a hermitian quadratic form [2] with only one global minimum in respect to $\underline{q}_{\mathrm{S}}$. The optimal solution for the maximum power absorption can be calculated according to [1] with:

$$\underline{q}_{\mathrm{SA}} = -\frac{1}{2}\frac{\underline{p}_{\mathrm{SP}}}{\mathrm{Re}\left\{\underline{Z}'_{\mathrm{SS}}\right\}} \, . \tag{4}$$

Instead of using the radiated sound power the active sound intensity at the centre axis of the loudspeaker can be considered:

$$I_{\mathrm{S}} = \frac{1}{2}\mathrm{Re}\left\{\underline{p}_{\mathrm{S}}\underline{v}_{\mathrm{S}}^{*}\right\} = \frac{1}{2}\mathrm{Re}\left\{\left(\underline{p}_{\mathrm{SP}} + \underline{Z}_{\mathrm{SS}}\underline{v}_{\mathrm{S}}\right)\underline{v}_{\mathrm{S}}^{*}\right\}, \tag{5}$$

In this case the complex pressure $\underline{p}_{\mathrm{S}}$ is measured at a pressure sensor in front of the secondary source, $\underline{v}_{\mathrm{S}}$ is the velocity of the membrane and $\underline{Z}_{\mathrm{SS}}$ is the specific acoustic impedance from the secondary source to the location of the microphone. The optimal solution for the minimization of the active intensity to maximize absorbed power can be derived according to (4):

$$\underline{v}_{\mathrm{SA}} = -\frac{1}{2}\frac{\underline{p}_{\mathrm{SP}}}{\mathrm{Re}\left\{\underline{Z}_{\mathrm{SS}}\right\}} \, . \tag{6}$$

For tonal excitation $\underline{v}_{\mathrm{SA}}$ can be substituted by:

$$\underline{v}_{\mathrm{SA}} = \frac{1}{\mathrm{i}\omega}\underline{a}_{\mathrm{SA}} \, . \tag{7}$$

Hereby, $\underline{a}_{\mathrm{SA}}$ is the membrane acceleration and ω is the angular frequency. With (7) equation (6) can be written as:

$$\underline{a}_{\mathrm{SA}} = -\frac{\mathrm{i}\omega}{2}\frac{\underline{p}_{\mathrm{SP}}}{\mathrm{Re}\left\{\underline{Z}_{\mathrm{SS}}\right\}} \, . \tag{8}$$

By introducing a transfer function $\underline{H}_{SS}$ between $\underline{a}_S$ and $\underline{p}_{SS}$, which is defined by

$$\underline{p}_{SS} = \underline{H}_{SS}\underline{a}_S , \tag{9}$$

an equation for the optimal secondary acceleration due to maximum power absorption can be derived:

$$\underline{a}_{SA} = \frac{i}{2} \frac{\underline{p}_{SP}}{\mathrm{Im}\{\underline{H}_{SS}\}} . \tag{10}$$

Thus, it is possible to derive a control law depending on membrane acceleration and primary pressure, which can be used as a input signal for the controller.

Practical Approach

Implementation of control algorithm

Figure 1 shows the implementation of equation (10) as a block diagram. It is divided in the signal processing domain and the domain that represents the physics of the experimental setup. Hereby $\underline{L}$ represents the physical

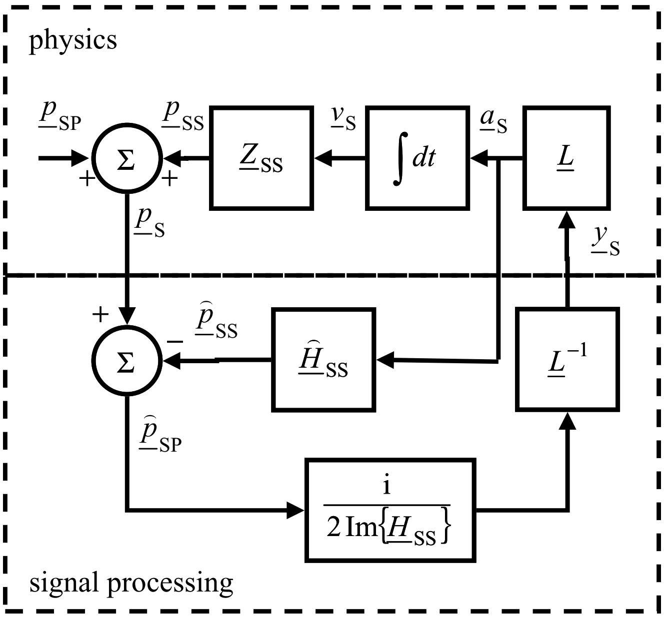

Figure 1: Block diagram of the control aproach

transfer function between $\underline{a}_S$ and the loudspeaker drive signal $\underline{y}_S$.

Two values are handed over to the signal processing domain. This is the measured pressure at the secondary loudspeaker $\underline{p}_S$ and the acceleration of the surface $\underline{a}_S$ of the secondary source. The estimated transfer function $\hat{\underline{H}}_{SS}$ can be used to calculate the secondary pressure $\hat{\underline{p}}_{SS}$ and in the next step to estimate primary pressure $\hat{\underline{p}}_{SP}$ at the secondary source. Equation (10) is used in order to calculate the gain for the primary pressure signal. The last step to achieve the proper output signal is a multiplication with $\underline{L}^{-1}$ the inverse of the physical transfer function of the loudspeaker, filter and amplifier in the secondary transfer path.

Experimental setup

The arrangement of experimental setup is shown in figure 2 and figure 3. Figure 2 shows the used hardware components and the measurement chain.

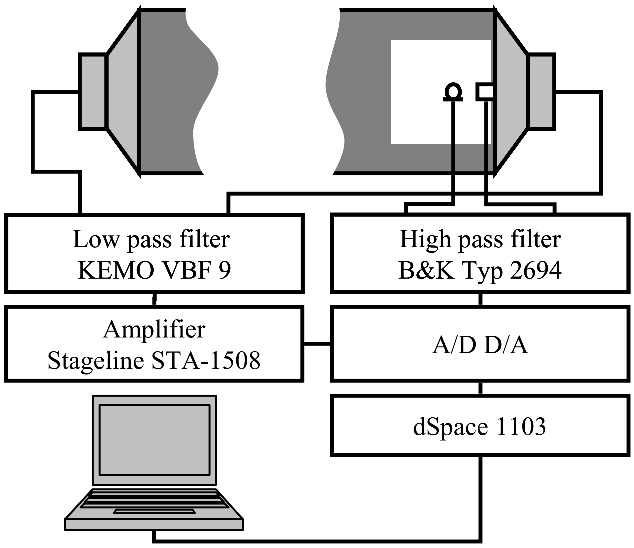

Figure 2: Hardware components and measurement chain

Figure 3 illustrates the hardware setup and the design of a wooden duct, which is used to conduct experiments in order to examine the control performance. In this one-dimensional case study the duct provides the acoustical medium between the primary and the secondary source. It was made from MDF-material consisting of squeezed ramming fibres with an overall length of 2m. For measuring the sound pressure at certain points within the duct between the primary and the secondary source several holes in a distance of 0.1m provide microphone access during experiments. The secondary as well as the primary loudspeaker were placed at either end of the duct.

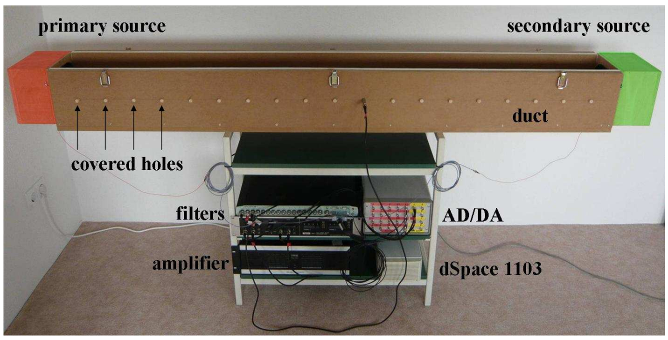

Figure 3: Experimental setup with duct

Two EIGHTEEN SOUND Type 6ND430 loudspeaker in wooden housings, two BRÜEL & KJÆR Type 4188 microphone units and two BRÜEL & KJÆR Type 4517 DeltaTron® accelerometer were installed in the experimental

Figure 4: Arrangement of the accelerometer and microphone at the secondary source

setup. The arrangement shown in figure 4, displays the allocation of the accelerometer mounted with glue on the painted loudspeaker membrane and a microphone close to the surface of the secondary loudspeaker.

Identification of the required transfer functions

A modelling to determine the required transfer function has to be made prior to run the experimental setup. At least the filters $\underline{\hat{H}}_{SS}$ and $\underline{L}^{-1}$ have to be identified.

Due to the different design of the accelerometer and pressure sensor a phase correction has to be implemented in order to measure the correct transfer functions. Figure 5 shows the approach for the phase correction. It is assumed, according to equation (5), that the acoustic intensity has no active component between close sources, which are out of phase. This provides a strategy to determine the phase difference between both signals of the microphone and the accelerometer at the secondary source.

For the phase correction and only for this attempt, it has to be assured that the drive signals at the primary and the secondary source have exactly the amplitude [1]

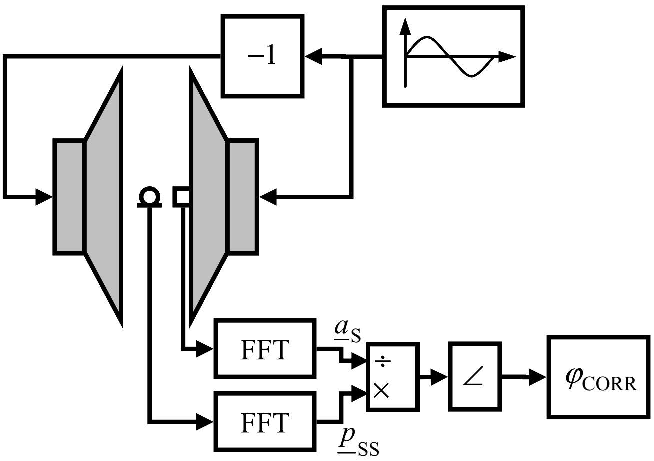

Figure 5: Block diagram for modelling of phase difference between accelerator and pressure sensor

$\underline{L}^{-1}$ represents the inverse of the physical transfer function

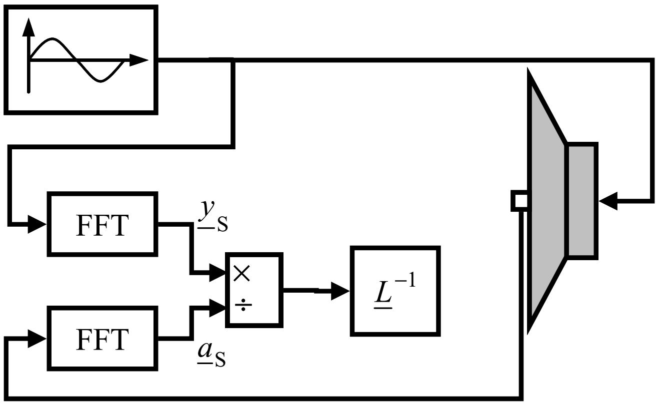

Figure 6: Block diagram for modelling of $\underline{L}^{-1}$

between the input signal of the experimental setup y_S and the acceleration $\underline{a}_S$ of the surface of the secondary source. The block diagram for the modelling is shown in figure (6).

The secondary loudspeaker is driven by a signal generator. The drive signal as well as the resulting acceleration signal are transformed in the frequency domain and $\underline{L}^{-1}$ is calculated by:

$$\underline{L}^{-1} = \frac{\underline{y}_S}{\underline{a}_S}. \tag{11}$$

In absence of any primary noise the estimated transfer function $\underline{\hat{H}}_{SS}$ can be easily calculated by dividing the pressure radiated by the secondary source $\underline{p}_{SS}$ by the acceleration on the surface of the secondary source $\underline{a}_S$. Figure 7 shows this method in a block diagram.

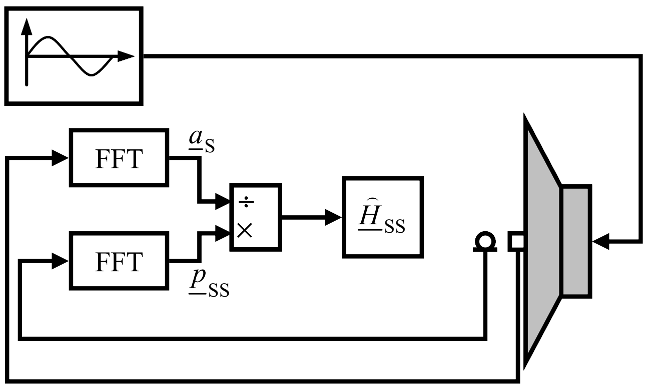

Figure 7: Block diagram for modelling $\underline{\hat{H}}_{SS}$

Results

The first result to be considered is that the decoupling of the input signals $\underline{p}_{SS}$ and $\underline{p}_{SP}$ was successfully and the controller was stable. It was possible to run the experimental setup without any feedback.

Measurements were taken at the first two resonances at (86Hz, 172Hz) and the antiresonance between (129Hz). Figure 8 shows the results. It can be seen that at the resonances attenuation is possible and at the antiresonances an increase of pressure level can occur. These results confirm the theoretical approach of NELSON et. al. in [1].

Furthermore it can be noticed that by decreasing frequency the degree of attenuation is increased. This effect is caused by the rather vague method of determination of the phase correction between the sensor pair.

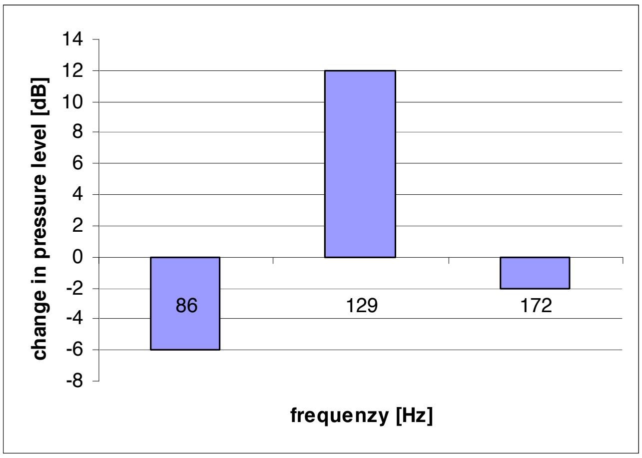

Figure 8: First results in change of pressure level at the given resonances (86Hz, 172Hz) and antiresonance (129Hz).

The results show that an active control of maximum power absorption is not appropriate for all frequencies in an enclosure.

Conclusion

This paper deals with the entirely local active power absorption method. A theoretical and practical approach is discussed. The results show that only at resonances the method of maximum power absorption can lead to a global attenuation.

To verify the first results further test series have to be made. Measurements using an intensity sensor can lead to better methods to improve calibration of the phase correction.

Next steps will lead to researches on varying the impedances of the secondary source to achieve other effects, e.g. free field impedance.

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