Considerations on the Realtime Realisation of a 2D-Feedforward-ANC-System Part 1: Aspects to the software

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Introduction

The purpose of the 2D-feedforward-system is a noise control inside a loudspeaker circle. Figure 1 is a drawing of the configuration, which is not proportional. A drawing with measures is given in [1]. The inner circle of the 12 loudspeakers is surrounded with an outer circle of 12 microphone pairs.

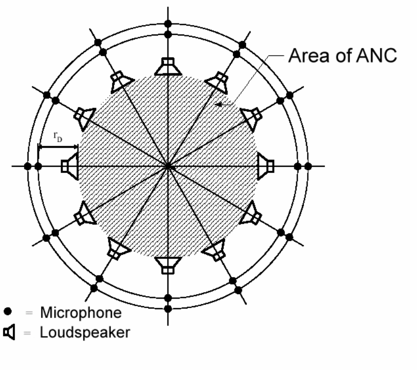

Figure 1: Configuration of the loudspeaker and microphone circle

Based on the Kirchhoff-Helmholtz integral but discretised the sound pressure inside the circle can be calculated from the microphone signals measuring the pressure and the pressure gradient on the circle. The discretisation is allowed so long as the distance of the neighbouring microphones are smaller than about the half of the shortest wave length. Equation (1) described this step in vector/matrix form, where the pressure of the primary field at chosen points (P_{AP}) inside the circle can be determined from the microphone pairs (M_1,M_2) via the transmission functions (TM_xA). The secondary field (P_{AS}) at the chosen points radiated by the loudspeakers can be determined from the source strength via the transmission function (T_{LA}) (eq.2). Under the condition of equation (3) the connection between the microphone signals and the source strength of the loudspeakers (eq.4) can be derived, so that both fields cancel each other. More details to the theory are to find in [2][3][4].

$$\overrightarrow{P_{AP}} = \overleftrightarrow{TM_1A} * \overrightarrow{M_1} + \overleftrightarrow{TM_2A} * \overrightarrow{M_2} \qquad (1)$$

$$\overrightarrow{P_{AS}} = \overleftrightarrow{TLA} * \overrightarrow{Q_L} \quad \Rightarrow \quad \overrightarrow{Q_L} = \overleftrightarrow{TLA}^{-1} * \overrightarrow{P_{AS}} \qquad (2)$$

With

$$\overrightarrow{P_{AP}} = -\overrightarrow{P_{AS}} \qquad (3)$$

$$\Rightarrow \quad \overrightarrow{Q_L} = -\overleftrightarrow{TLA}^{-1} \left[\overleftrightarrow{TM_1A} * \overrightarrow{M_1} + \overleftrightarrow{TM_2A} * \overrightarrow{M_2} \right] \qquad (4)$$

Till now the investigations were made by simulation or by experiments in the offline modus. Under these circumstances a processing delay doesn't matter. In this phase the calculation were made on the base off the FFT. But in a real time realization this solution has the disadvantage of blockwise processing causing an additional delay on principle. Therefore the further processing is done in the time domain and the conditions for this are investigated.

Time Domain Processing

The transfer functions in equation (4) are transformed in the time domain. Doing this a limitation in the frequency range is necessary, because the discretisation in space is restricts the frequency range to an equivalent of an half of the shortest wave length. The distance of neighbouring microphones results in an upper limit of about 350Hz. Figure 2 shows the secondary field of a 350Hz-wave in the inner circle. In the whole the field is produced properly, but first deviations –especially at the border- can be observed.

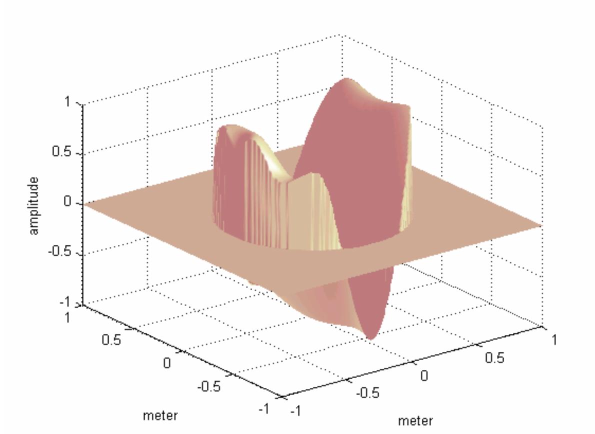

Figure 2: Secondary field of a 350Hz-wave inside the loudspeaker circle

Impulse Response and Damping Contour

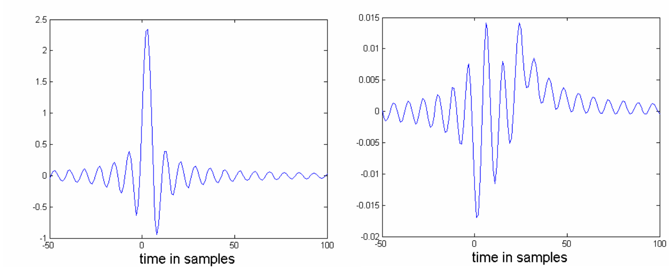

Figure 3: Impulse response between a microphone and the loudspeaker near by (left) respectively in opposite position (right); transfer function hard-limited to 350Hz, sampling frequency 2.8 kHz

Figure 3 shows two impulse responses, where the frequency range were hard-limited to 350Hz. Both have a sizeable part before zero time, which is not realizable on account of non-causality. What happens, if this part is simply cut off?

With the cut impulse responses the damping contour in the circle is calculated (figure 5). This contour indicates how strong the primary field is damped by the secondary field in the space. To see the effect of cutting, also the contour with the non-cut impulse responses are drawn (figure 4). In both cases a 200Hz-wave are assumed.

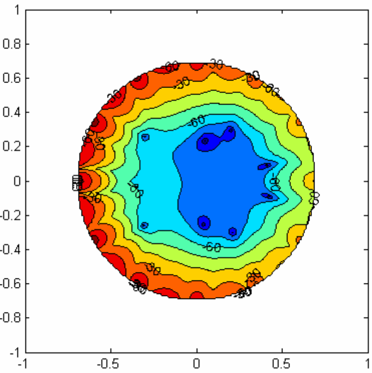

Figure 4: Damping contour inside the loudspeaker circle using a 200Hz-wave, non-cut impulse responses and a sampling frequency of 2.8 kHz

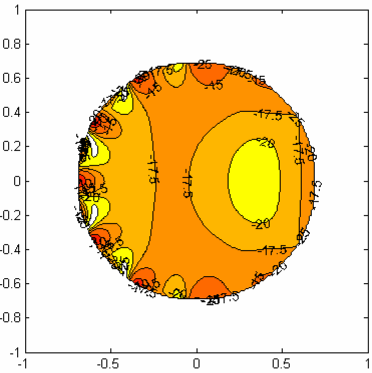

Figure 5: Damping contour inside the loudspeaker circle using a 200Hz-wave, impulse responses cut before zero time and a sampling frequency of 2.8 kHz

A comparison of figures 4 and 5 uncovers the remarkable reduction in the damping. With a soft limitation of the frequency range and a higher sampling frequency the part before zero time can strongly be reduced and with it the effect of cutting. Figure 6 shows the impulse responses using a soft limitation by a Hamming-edge and a sampling frequency of 11.2 kHz.

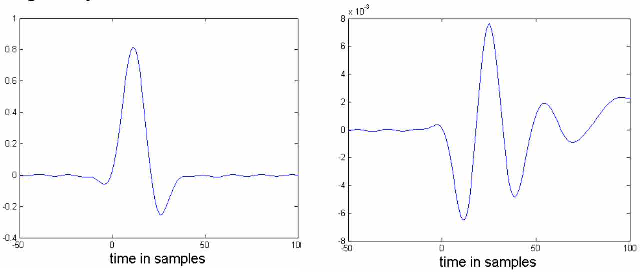

Figure 6: Impulse response between a microphone and the loudspeaker near by (left) respectively in opposite position (right); transfer function soft-limited to 350Hz, sampling frequency 11.8 kHz

After this manipulation the part before zero is hardly important. Cutting this the damping is decreased clearly less as figure 7 shows.

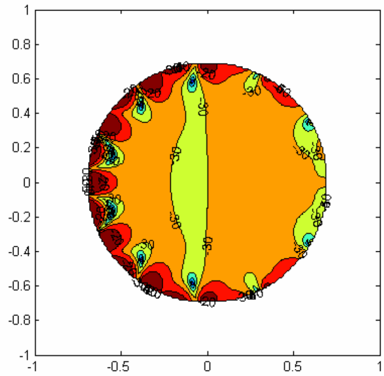

Figure 7: Damping contour inside the loudspeaker circle using a 200Hz-wave, transfer function soft-limited to 350Hz, impulse responses cut before zero time; sampling frequency 11.2 kHz

Comparing figure 5 and 7 the increasing of the damping is obvious. In figure 5 the damping is not higher than 17.5dB in a large area, whereas in figure 7 it is more than 30dB nearly in the whole area.

A further improvement in the damping gives an enlarging of the frequency range, but the limitation by the geometry of the set-up, by the number of microphones and loudspeakers has to be taken into account. Generally the choice of the system parameters has to be matched to get the best result. Also attention has to be paid to the limitation caused by inexactness of system components (mainly the analogue components) in order to avoid useless efforts.

For the real time realization the impulse responses in figure 6 finally gives an important information, which order is required for the FIR-filters used here. In the considerations to the hardware [1] a order of 75 is chosen on average in the case of a sampling frequency of 11.2 kHz. The transmission between opposite devices may have a more extended impulse response (fig. 6, right), but in relation to the other the amplitude is rather small, so that cutting a part of the end has only a little effect. Summarized it can be said:

- A causal system can be configured.
- A good benefit in damping is achievable under this condition.
- The necessary processing power is practicable.

References

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