УДК 621.382.24

DIGITAL SIGNAL PROCESSING IN AN ACTIVE NOISE CONTROL SYSTEM

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Submitted 15 December 2003

Keywords: digital signal processing, adaptive filter, noise, and active noise control.

Introduction

Noise is a great problem in all industrial countries. There is a high level on effort and expenditure to reduce the noise, but in many situations, the practical and/or the financial conditions make this impossible. Here the conventional solutions refuse. Therefore, an old idea was taken up, the active noise control (ANC).

The principle of active noise control is described in a patent in 1932 first time. It uses the destructive interference of two sound waves, a given noisy sound wave (primary wave) and an additionally generated sound wave, the secondary wave. For total cancellation of the primary wave the secondary wave has to have exactly the same amplitude, but the opposite phase. Clearly, this is to be done in the whole region, where the cancellation should work. If this condition is not fulfilled, instead of a cancellation only a damping is achieved, which decreases rapidly with decreasing accuracy of amplitude and phase. Figure 1 shows the achievable damping depending on the amplitude and phase error.

The demanded accuracy was not to reach or only with a plenty of means in the past. This was the reason, why ANC was realised only in few special applications, where the requirements were more easily to meet. These are realisations in the lower frequency range, in a smaller local range and with a simpler wave field form. So the most applications were and are to find with ducts of e.g. air condition systems and headphones for pilots especially in helicopters. These applications can even realised in rather simple analogous technique. When the digital signal processing was coming up, new technical possibilities were offered to get a higher accuracy and complexity. The developments of the 80's and early 90's in the automobile industry are to mention here at first. Mullet-channel systems were created to reduce the noise inside the cars and some times also outside the car. The systems worked well, but they are still too expensive, even if they are without an alternative passive solution.

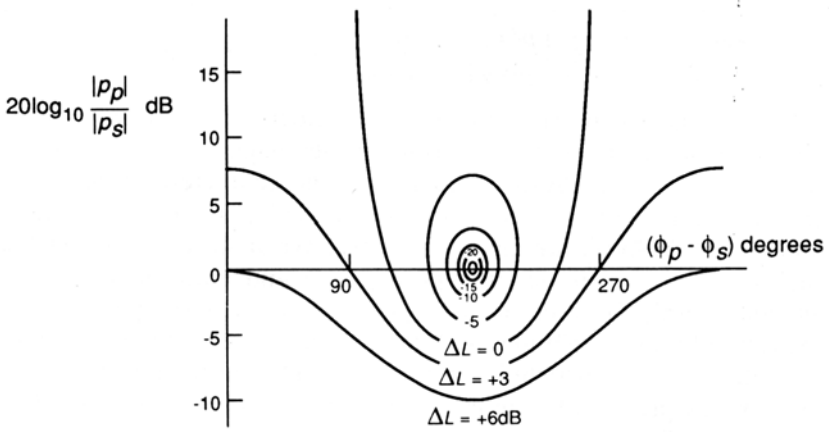


Fig. 1 Achievable damping ΔL depending on the amplitude ratio and the phase difference of the primary and secondary wave; from [1]

The last five years brought again a strong push in the development of devices for digital signal processing with the tendency: more power and less cost. So a breakthrough will be foreseeable, where ANC-solution in more and more (not all) situations are not only more effective than passive solutions, but even cheaper.

Feedforward and Feedback Systems

In ANC, two basic systems are known: the feedforward and the feedback system [1,2]. The first one is perhaps the rather intuitively found approach, which also the above mentioned patent and here figure 2a show.

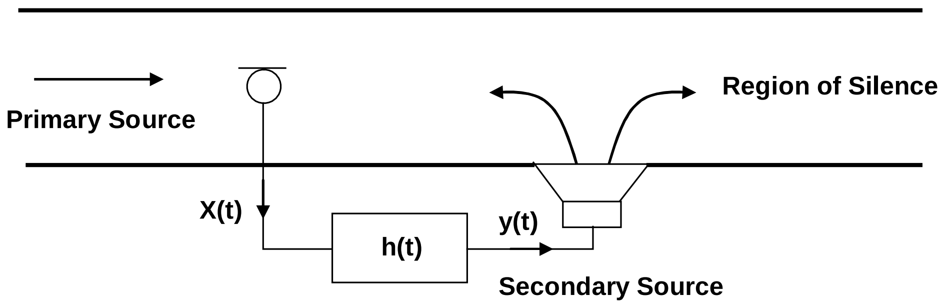


Fig. 2a. Feedforward System

A microphone picks up the pressure of the primary wave coming from left. In the time, in which the wave is travelling father to the right side the signal $x(t)$ is formed by a filter with the impulse response $h(t)$ to the signal $y(t)$ in such a way, that the loudspeaker exactly produces the sound pressure (with the opposite sign) of the corresponding part of the wave just in the moment, when this part arrives at the loudspeaker. Then both waves cancel each other and right of the loudspeaker a region of silence results.

In this arrangement, the filter has the task to model the transmission path of the wave between the microphone and the loudspeaker. Further, the filter has to equalise the transfer function of the microphone and the loudspeaker. In practice this task is generally not so easy, because the path is not stationary (e.g. changing sound speed) and the transfer function (especially of the loudspeaker) is often flawed with high variations in amplitude and phase. The possibility of equalisation is limited, because the filter has a limited time to react. For reacting on changes in the transmission path the filter is in general adaptively controlled by a second microphone positioned at the begin of the silent zone. A further additional measure is often necessary to avoid a feedback from the loudspeaker to the microphone (echo compensator) [2]. Beside these disadvantages, the feedforward system offers a great advantage. It can cancel or damp broadband and impulse noise, because it has a "look to the future".

The second basic solution, the feedback system, is not provided with such a feature. It is shown in figure 2b.

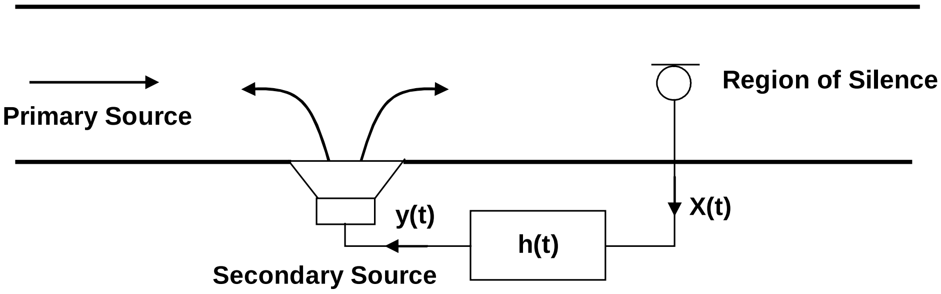


Fig. 2b. Feedback System

The filter generates the signal $y(t)$ with the aim of achieving silence around the microphone. Clearly, if there is a sudden change in the primary wave, the system does not react until this change arrives at the microphone, but that is too late for cancelling by the secondary source. Therefore, the feedback system can only cancel stationary or (more precisely) predictable primary waves. These are waves with small band characteristic (the smaller the better). Since the microphone signal should be near to zero the filter has a high amplification. This can lead to an unstable behaviour of the system. On the other side, the feedback system has the advantage to compensate variations in the transmission path and transfer functions of the microphone and the loudspeaker automatically.

In figure 2 an one-dimensional approach is introduced, for example a duct with a small diameter in relation to the shortest wave length. In the next section the idea for a two-dimensional approach is introduced.

2D-ANC System

Is it possible to build e virtual wall around a limited and closed area lying in noisy environs and create so an "island of silence"? Perhaps, this approach, developed at the University of Wuppertal [5,6,7], tries to go in this direction. The figure 3 shows a model of the 2D-system.

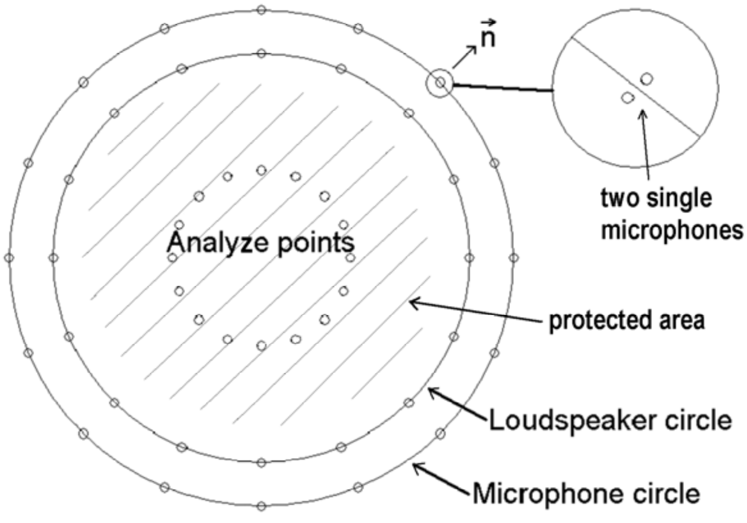


Fig. 3. Model of the 2D-ANC-System

The system is consisting of two concentric circles. On the outer circle, microphones are positioned in equal distances. In each position, two microphones are taken in a very small distance. The direction of the connection line between the two microphones is radial for each pair. The microphones have an omnidirectional characteristic.

The inner circle surrounds the area, which should be protected against the noise. On this circle the loudspeakers are positioned also equally spaced and with an omnidirectional characteristic.

The approach with two circles is necessary for two reasons: Firstly the system bases on the feedforward principle and it takes some time to calculate the loudspeaker signals from the microphone signals. The time length being available for this is not more than a sound wave needs to run over the distance of the difference from the outer to the inner circle. Secondly, the loudspeakers are handled as pointsources, which have an infinite sound pressure at their location. Therefore, they can not coincide with the locations of the microphones.

The points named analyze points in figure 3 are locations, in which the primary and secondary field should cancel each other by superposition. The task of cancellation can split in two parts:

Part I (Analysis): A primary wave coming from outside is picked up by the microphones on the circle. This allows determining the sound pressure at each point inside the circle.

Part II (Synthesis): Based on the knowledge of the sound pressure distribution inside the circle the driving signals of the loudspeakers can calculated so, that the loudspeakers generate a secondary sound field with an identical pressure distribution. After that only the polarity of the loudspeaker signals are to be changed.

The analysis falls back on Kirchhoff-integral:

$$P(\vec{r}_0,\omega)=\int_V G(\vec{r}_0,\vec{r},\omega)\Psi_{tot}(\vec{r},\omega)dV+\oint_S\left[G(\vec{r}_0,\vec{r},\omega)\vec{\nabla}P(\vec{r},\omega)-\vec{\nabla}G(\vec{r}_0,\vec{r},\omega)P(\vec{r},\omega)\right]\vec{n}dS \qquad (1)$$

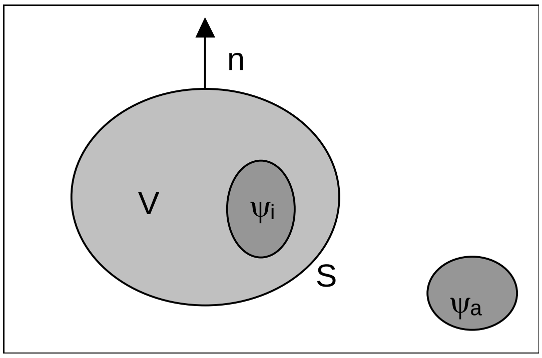


Fig. 4. Situation with a source distribution inside and outside a closed volume

Equation 1 corresponds with the situation in Figure 4. A distribution of source (Ψ_i) inside and (Ψ_a) outside a closed volume *V* causes a sound pressure $P(r_o,\omega)$ inside this volume at the location r_0. For calculating this pressure by integration, it is enough to know the distribution inside and the pressure and its gradient on the surface *S* of the volume. If there are not source inside ($\Psi_i = 0$), the first integral on the right side of eq.1 disappears. Then the pressure and the gradient of the pressure on the surface only determine the pressure inside. Equation 1 describes the 3-dimentional solution; therefore, *G* is the 3-dimentional Green function. ω is the angular frequency.

The 3-dimentional solution requires not only microphones completely around the volume (having a maximal distance from each other), but for a cancellation of waves from all direction also loudspeakers in the same manner. This would be the ideal, but rather impracticable solution. Therefore, the further considerations base on the assumption, that the wave field is constant in the height. That is realistic for a situation outside, where the traffic in the horizontal (nearly) generates such wave fields, which can described by a cylindrical wave. For the analysis of this wave, it is enough to pick up the sound pressure and its gradient on a horizontal circle surrounding the area of interest.

A further step is making a discrete local resolution. Equation 1 demands a locally continuous measurement of the pressure and its gradient. But similarly to sampling theorem of Shannon in the time domain it can be shown, that a sampling in a determined distance ensures a lossless information about the wave field, if the distance is smaller than a half of the minimum wavelength. That is the condition for positioning the microphones in figure 3. The pairs are needed to get the gradient. In all eq. 1 can be reduced to eq. 2:

$$P_{pri}(\vec{r},\omega) = \sum_{i=1}^{N_M} \left(\vec{\nabla} P(\vec{r}_i,\omega)\, G(\vec{r},\vec{r}_i,\omega) - P(\vec{r}_i,\omega)\, \vec{\nabla} G(\vec{r},\vec{r}_i,\omega) \right) \vec{n}\; S_i \;. \tag{2}$$

P_{pri} is now the pressure caused by the primary field inside the circle at r. G is now the 2-dimentional Green function, r_i the locations of the microphones on the circle and S_i the length of the section represented by the microphone i.

The loudspeakers are lying inside the microphone circle and therefore the first integral on the right side in eq.1 can be used and processed in a similar manner as the other part. The result of this is eq.3:

$$P_{sek}(\vec{r}) = \sum_{i=1}^{N_L} \left(G(\vec{r},\vec{r}_i)\Psi_i \right) \tag{3}$$

P_{sek} is the pressure caused by the loudspeakers. Also G is here the 2-dimentional Green function and Ψ_i is the distribution of the second sources.

Some words to the kind of the second sources: The approach assumes a field constant in the height. To fulfil this vertical line sources are needed at the positions of the loudspeakers. Different to the microphones, the third dimension has to be considered and cannot be neglected, because there is a propagation of the waves in the third dimension. Only a distribution of point sources on the horizontal circle meets not the condition to cancel for example a horizontal plane wave. Of course, unlimited lines sources are unrealisable, but fortunately, they are dispensable. It is enough to approximate the effect of the line sources in a particular section of the height, so that e.g. the listeners have theirs ear in a layer of a certain thickness.

The approximation can achieved by a short line array of point source and be improved by a special processing of the loudspeaker signals for each array [4,8].

Equation 2 and 3 can be written in matrix form:

$$\overline{P}_{pri} = \overline{\overline{G}}_{pri}\; \overline{\vec{\nabla} P_M \vec{n}} - \overline{\overline{\vec{\nabla} G_{pri} \vec{n}}}\; \overline{P_M} \tag{4a}$$

with

$$\overline{\overline{G_{pri}}} = \begin{bmatrix} G(\vec{r}_{A1},\vec{r}_{M1}) & \dots & G(\vec{r}_{A1},\vec{r}_{LN_M}) \\ \dots & & \dots \\ G(\vec{r}_{AN_A},\vec{r}_{M1}) & \dots & G(\vec{r}_{AN_A},\vec{r}_{LN_M}) \end{bmatrix}, \tag{4b}$$

$$\overline{\vec{\nabla} P_M \vec{n}} = \left[\vec{\nabla} P(\vec{r}_{M1})\vec{n} \quad \dots \quad \vec{\nabla} P(\vec{r}_{AN_M})\vec{n} \right]^T, \tag{4c}$$

$$\overline{\overline{\vec{\Delta} G_{pri} \vec{n}}} = \begin{bmatrix} \vec{\Delta} G(\vec{r}_{A1},\vec{r}_{M1})\vec{n} & \dots & \vec{\Delta} G(\vec{r}_{A1},\vec{r}_{MN_M})\vec{n} \\ \dots & & \dots \\ \vec{\Delta} G(\vec{r}_{AN_A},\vec{r}_{M1})\vec{n} & \dots & \vec{\Delta} G(\vec{r}_{AN_A},\vec{r}_{MN_M})\vec{n} \end{bmatrix}, \tag{4d}$$

$$\overline{P}_M = \left[P(\vec{r}_{M1}) \quad \dots \quad P(\vec{r}_{AN_M}) \right]^T \tag{4e}$$

and

$$\overline{P}_{sek} = \overline{\overline{G}}_{sek} \cdot \overline{\Psi}_{sek} \tag{5a}$$

with

$$\overline{P}_{sek} = \left[P_{sek}(\vec{r}_{A1}) \quad \ldots \quad P_{sek}(\vec{r}_{AN_L}) \right]^T , \tag{5b}$$

$$\overline{\overline{G}}_{sek} = \begin{bmatrix} G(\vec{r}_{A1},\vec{r}_{L1}) & \ldots & G(\vec{r}_{A1},\vec{r}_{LN_L}) \\ \ldots & & \ldots \\ G(\vec{r}_{AN_A},\vec{r}_{L1}) & \ldots & G(\vec{r}_{AN_A},\vec{r}_{LN_L}) \end{bmatrix}, \tag{5c}$$

$$\overline{\Psi}_{sek} = \left[\Psi_1 \quad \ldots \quad \Psi_{N_L} \right]^T . \tag{5d}$$

The superposition of the primary and the secondary field should result in a cancellation and leads the eq.6:

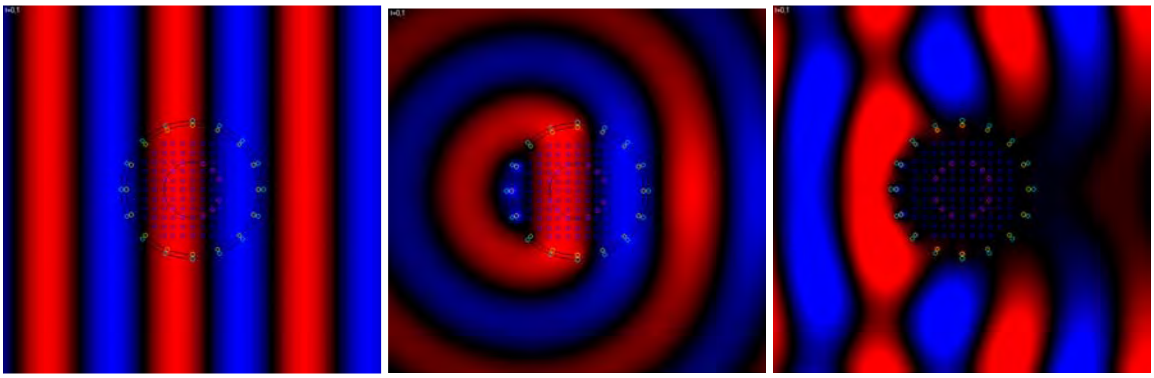
$$\overline{P}_{pri} = -\overline{P}_{sek} . \tag{6}$$

After entering eq.4a and eq.5a into eq.6 a and separation of Ψ_{sek} on the left side equation 7 gives the instruction for processing the microphone signals to the loudspeaker signals:

$$\overline{\Psi}_{sek} = -\overline{\overline{G}}_{sek}^{\,-1} \cdot \left[\overline{\overline{G}}_{pri} \cdot \overline{\vec{\nabla} P_M \vec{n}} - \overline{\overline{\vec{\nabla} G_{pri} \vec{n}}} \cdot \overline{P_M} \right]. \tag{7}$$

Results of Simulation and Measurement

The correctness of eq. 7 was tested in simulations and the practicability of the approach was investigated with experimental set-ups. The next figure 5 shows three pictures of a simulation.



a) *b)* *c)*

Fig. 5. Results of a simulation: *a*) primary field; *b*) secondary field; *c*) resulting field

Figure 5a shows a plane wave coming from left and forming the primary field. In the centre, the two circles are recognisable. Figure 5b shows the calculated secondary field caused by the secondary sources. The field is not restricted to the inside of the circle. Outside the circle, the wave front is curved, but inside it is straight. Superposing the primary and the opposite secondary field brings the field in figure 5c. Now the inside of the circle is nearly fieldfree. Outside the circle, the system has an effect like a wall: The wave is reflected and led around the circle. Theoretically, also an absorption of the primary wave is possible, but this requires a hardly realisable combination of a point source (monopole) and a dipole source.

In measurements with experimental set-ups, the ANC-function could successfully be tested too, although the general set up brought some problems. The figure 6 shows the attenuation measured in small set-up with an area of about 0.5m×0.5m: The height of the set-up was only 0.1m, so that the field up to about 800Hz was only varied in two dimensions. There were used 18 microphones and 9 loudspeakers.

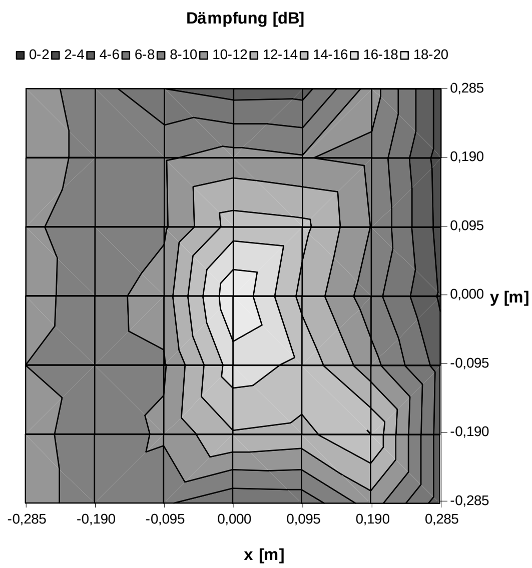


Fig. 6. Attenuation measured in an experimental setup [3]

Real-time Hardware

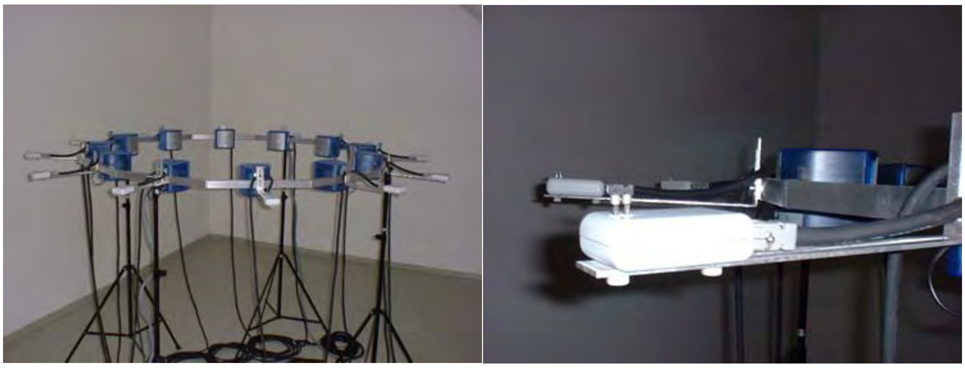
The measurements were done in an offline-modus: This means, in a first step the reproducible primary wave was recorded by in a computer on 18 tracks, in a second step the signal were processed by the computer and in a third step the primary wave and the calculated secondary wave are produced simultaneously.

Doing so causes some problems: Smallest changes e.g. in the positions of the devices or a shift in the sound speed let decrease the attenuation dramatically (see figure 1). This can only be avoided by a real-time system with the ability of adaptation to changing conditions. To this, also an automatic calibration belongs equalising the transfer function of the microphones and the loudspeakers.

Of course, a real-time system offers additionally better possibilities to work interactively in investigations and to demonstrate the system. Aiming at this a larger microphone and loudspeaker circle was built up first. Figure 7 shows the circle and some details of it.

This set-up can also be driven by a studio system, but this is not fast enough and works offline. Therefore, the missing link in the processing chain is hardware, fast enough for real-time processing.

According to eq.7, a particular transfer function between each microphone and loudspeaker is to realise. 12 loudspeakers and 24 microphones result in 288 transfer functions. In the off-line modus, these transfer functions are realised with the FFT, because the implementation of the processing steps from eq.4 to 7 is simpler to make in this way. For a real-time solutions this way is not practicable: the blockwise processing corresponds with a high delay caused by collecting the data for the blocks. Also former investigation gave the result, that the blockwise processing using the FFT requires a large overlapping of the blocks for avoiding artefacts caused by the circular convolution. For all these reasons, it is preferred to realise the transfer function with FIR filters. A mean filter order of 100 seems to be enough as some tests proved. The order depends on the used sample frequency and the value of 100 corresponds to a sample frequency of 32kHz. Certainly, the ANC should only work in a frequency range up 1kHz avoiding a smaller distance between the loudspeakers or microphones. Signal components of the primary wave with higher frequencies, which are not controllable by ANC, have to keep away from ANC processing, what is easier and better to do by digital means. This, however, requires the higher sampling frequency.



a) *b*)

Fig. 7. Experimental set-up: *a*) whole circle with microphones and loudspeakers; *b*) a carrier with a pair of microphone capsules

What are the demands?

In all there are $288\times100\times32000=922\times10^6$ multiplication to calculate each second. This number can not do by a single DSP. Therefore, a multi-processor system of four DSP's (type TI6713) [9] is planned. This processor type is available in a form of a starter kit [10], which offers other useful features: Over two multi-channel buffered serial port (McBSP) the cards can communicate with each other and exchange their data. In addition, an external ADC-module (analogue-digital-converter) just as a DAC-module can be connected via these ports. With a six-channel ADC (ADS8364) and a four-channel DAC (DAC7644) [9] one card can provide three pairs of microphones and three loudspeakers. The named types also bring the advantage to be faster than the built-in sigma-delta-types. The conversion takes only few microseconds instead of several sample periods. It is essential to waste no time, since the shortest distance between a microphone and a loudspeaker is only about 0.3m and so the propagation time over this distance is less than 1ms. For the same reason, all additional delay has to be avoided. All components in the processing chain should have a transfer function as flat as possible in the amplitude and (!) phase lowering the order and so the delay of the equalisation filter.

Conclusion

It should be shown that the digital signal processing gives the possibility to make not only new ideas realisable, but also some rather old ones. ANC is an outstanding example for this. Digital signal processing enables a greater accuracy and new kind of functions, e.g. phase linear filters or filters with determined amplitude an phase. Further DSP offers new ways in the realisation of adaptive algorithms. All this brings the potential for ANC being rather a powerful than a nice theoretical solution against noise in the future.

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