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

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Introduction

In the paper [1] a short view is given on the principle and the theory of the 2D-feedforward-ANC-system. That paper mainly deals with the algorithm and its criterions for real time realization.

It is shown, that the transfer functions from the microphones to the loudspeakers require a FIR-filter of order about 75 on average in the case of a sampling frequency of 11.2 kHz. The system consists of 24 microphones and 12 loudspeakers, where each microphone / loudspeaker combination has its particular transfer function. Therefore a processing power of 24x12x75x11200= 242-E6 multiplications per second is needed. Beyond this, the processing delay is a further criterion analysed in paper [1] to avoid a degradation in the system performance caused by non-realizable non-causal transfer functions.

In this paper a solution for the hardware will be considered, which meets the requirements of paper [1] on the whole.

Hardware

DSK 6713

Beside the touched on requirements the large number of connections to the system (24 microphones and 12 loudspeakers) is a basic task. For this and other reasons the system is built up with four DSK6713 boards of the Digital Spectrum Inc.. Figure 1 shows a block diagram .

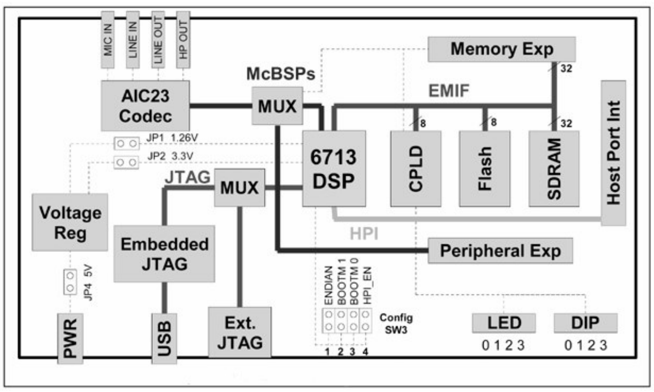

Figure 1: Block diagram of the DSK6713, Spectrum Digital Inc. [2]

The board is provided with expansion ports, which will be used to interconnect the boards and to plug in daughter cards with special analogue-digital and digital-analogue converters (ADC / DAC). The connection to the development system is done by USB or JTAG.

Some other important key features are:

- floating point DSP TMS320C6713, 225MHz,
- 8Mbyte of synchronous DRAM,
- 512 Kbyte of non-volatile Flash memory,
- configurable boot options.

More details are to find under [2][3]. With a operation frequency of 225MHz four boards are be able to calculate 900-E6 multiplications per second. That is nearly by a factor 4 more than needed, but it gives a plenty of reserve e.g. for increasing the sampling frequency.

Interconnection and Interfaces

For the data transmission to other processors and to peripheral function groups (e.g. ADC / DAC) the processor has so-called multi-channel serial ports, namely:

- multi-channel buffered serial ports (McBSPs),
- multi-channel audio serial ports(McASPs).

The two McBSPs are very useful to interconnect the boards via a simple 3-wire serial bus. A master and slaves can be configured and the data can be transferred in addressable time slots. The transfer rate of each port is 67 Mbit/s. This is more than enough for distributing 24 microphone signals in the system digitised with a resolution of 16 bit and a sampling frequency of 11.2kHz.

The two McASPs make available up to 16 high-rate serial audio channels each assigned to a separate pin. This permits an easy interfacing of a multi-channel ADC and/or multi-channel DAC. For connecting 24 microphones to the 4 boards a 6-channel ADC daughter card is plugged in each board, for the 12 loudspeakers each board is provided with a 4-channel DAC daughter card.

As ADC the type ADS8364 of Texas Instr. Inc., as DAC the type DAC7644 of Burr&Brown Inc. is chosen. The converter on board (AIC23) supports two only channels, but that is not the sole reason, why this converter is unusable. Operating as Sigma-Delta converter it is too slow. Paper [1] makes clear, that each additional micro second delay has to be avoided. The chosen types have separate converters for each channel converting the samples in a fixed time. The ADS8364 has a throughput of 4µs, the DAC7644 a settling time of 10µs. Both have a resolution of 16 bit.

Measurement

The transfer function of the chain ADC / DSP / DAC was measured with a sampling frequency of 43.5kHz to see also the behaviour in the higher frequency range. The result is shown in figure 2 in the form of magnitude (upper) and phase (lower).

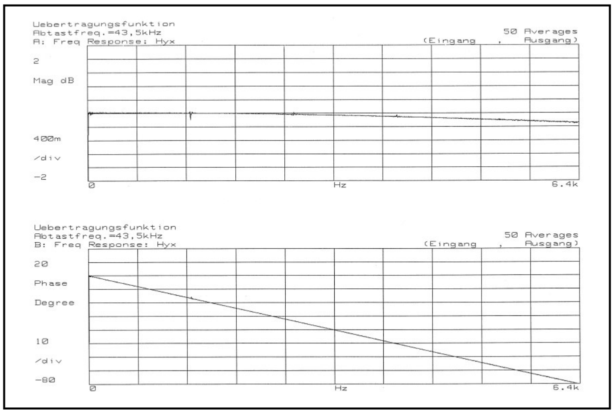

Figure 2: Transfer function of the chain ADC / DSP / DAC in the range of 0 to 6.4 kHz (linear)
Upper: magnitude with vertical scale of 2 dB/div
Lower: phase with vertical scale of 10 degree/div

The magnitude differs only at higher frequency a little from 0dB. This is caused by holding the sampling value at the output for one period (si-aperture). A correction is not necessary, if the upper limit of the operation range in frequency is much smaller than the sampling frequency. The small spikes in the curve come from the switching power supply of the board and should be suppressed.

The phase function is a straight line indicating a constant group delay time in the whole range. From the gradient the delay time can be calculated:

$$\tau_g = 80/(360*6.4\text{kHz}) = 34.5\ \mu\text{s} \qquad (1)$$

This value is the sum of one sample period and the settling time of the DAC.

More than these parts of the transmission chain the remaining links of the chain are interesting for the global behavior of the system, especially referring to the system delay. These are the microphone, the amplifier, the loudspeaker. Therefore the same measurement were done with these three devices in a line. Figure 3 shows the result.

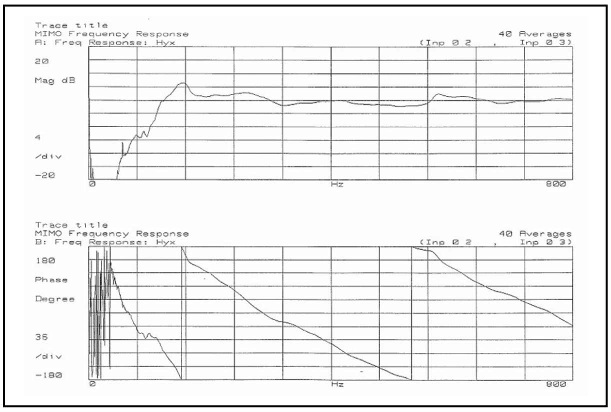

Figure 3: Transfer function of the chain microphone / amplifier / loudspeaker in the range of 0 to 800 Hz (linear)
Upper: magnitude with vertical scale of 4 dB/div
Lower: phase with vertical scale of 36 degree/div

The phase diagram in figure 3 leads to a group delay time of 2.63ms, but herein a propagation time is included for the distance of 68cm (figure 4) between a microphone in the centre of the circle and one of the loudspeakers.

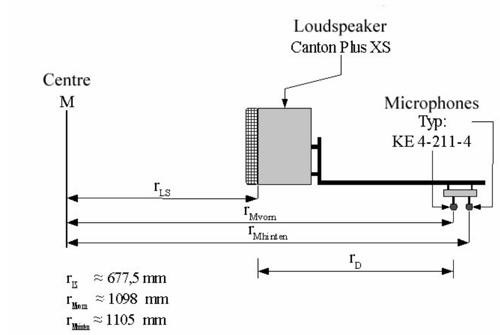

Figure 4: Dimensions of the experimental set-up (drawing not proportional)

This propagation time is 1,97ms, so a delay time of 660µs is left caused by the microphone, amplifier and mainly the loudspeaker. The sum of the measured delays (≈ 700µs) is equivalent 7.8 sample periods with a sampling frequency of 11.2kHz. This means, that the impulse responses in [1] figure 6 are shifted 7.8 samples left and the part before zero time is to cut. With a higher frequency range for the secondary field [1] the impulse response will be time compressed , so the part before zero time becomes smaller. But with a higher frequency range, also the distance between the loudspeakers (should be not larger than a half of the smallest wave length) and between the microphones has to become smaller. Here a balanced compromise has to be found with the aim to minimise the total or relevant error. The distance of about 40cm between the loudspeaker and microphone circle is equivalent a propagation time of 1.16ms leaving a reserve of 1160µs-700µs=460µs at best.

Clearly the loudspeaker (a common middle range type) ist the weakest link in the chain. At first it causes the largest part of the delay. Further the magnitude und phase diagram shows some deviation from a straight line requiring an equalization possibly. But this brings an additional delay . Therefore such corrections will be avoided in the a first try, especially as the deviations have just acceptable limits. Other characteristics like the lower limit of the frequency range (too high) or the directivity pattern at high frequencies are also in need of improvement.

Conclusion

The described hardware solution for the 2D-feedforward-ANC-system is able to operate in real time, where the number of operations per second is not problematical. The requirement of causality are more difficult to meet and demands a good matching of all system parameters for avoiding a larger deterioration of the ANC effect.

References

[1] M. Trimpop, D. Krahé: Considerations on real time realization of a 2D-feedworward-ANC-system; Part1: Aspects to the software, DAGA 2004, Strasbourg

[2] http://www.spectrumdigital.com

[3] http://www.ti.com