Modern digital to analogue conversion techniques

Lukas Funke
University of Paderborn
lfunke@mail.uni-paderborn.de

December, 23 2013

Abstract

Almost all modern audio equipment works with digital data. Either the data is stored in a compressed format like mp3, is synthesized digitally or post processed using audio software. However sound is perceived in an analogue manner through sound waves, generated by speakers, which are analogue components. This makes it necessary to convert digital data into analogue voltage. Hence the following paper will discuss some common digital to analogue converters.

1 Motivation

In order to achieve high quality audio in context of a digital signal processing system it is important to know all parts of the system. This holds for digital-to-analogue (D/A) converter, which acts as a bridge between the digital and the analogue world, as well. The following paper summarizes basic methods, up-to-date methods of D/A conversion as well as the terms and definitions that go along with this topic.

2 System context

Most of the digital signal processing systems encountered in science and engineering consist of the modules illustrated in figure 1.

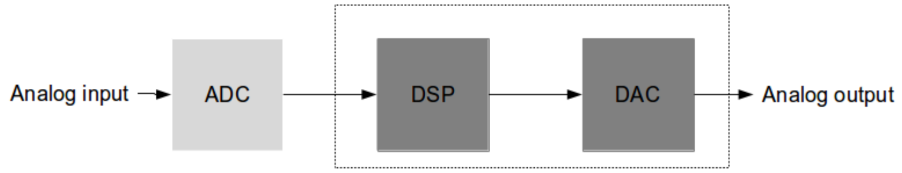

Figure 1: System context of a digital to analogue converter.

Because of the analogue nature of most real world problems one first has to transfers the analogue signals into the digital world. This is accomplished by the analogue-to-digital converter (ADC) component. The digital output signal is than processed by a digital circuit. Most of the times this circuit is a processor, because of the its reconfigurable, flexible nature. Depending on the problem and real time requirements of the system, sometimes special purpose architectures like FPGAs or ASICs are used. In context of the project group a mixture of both worlds is used, where the processor has a strong coupling with an FPGA.

The digital input of the DAC transferred into the analogue world by a component call digital-to-analogue converter (DAC). Not all of this components of figure 1 do necessarily exists in a digital processing system. Sometimes the analogue input or analogue output values are not necessary ot solve a given task. In the domain of digital audio, especially digital synthesizers, the system only consists of a processing component and a DAC. The analogue input is not required at all.

3 Terms and definitions

Before diving into the topic of digital to analogue converters it is useful to know some of the relevant vocabulary.

3.1 Errors

DACs are not perfect. Figure 2 shows the ideal and two common DAC error, the offset and the gain error.

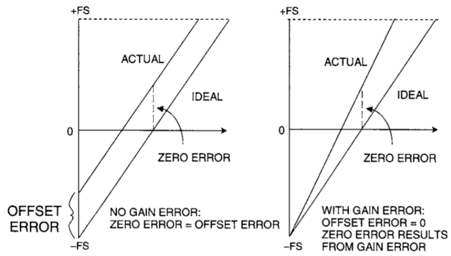

Figure 2: Ideal transfer function with offset and gain error [3]

3.1.1 Offset error

On offset error occurs if the DACs output is displaced by a constant factor over the whole transfer function. This error can usually be corrected by additional analogue components.

3.1.2 Gain error

A gain error occurred if the output drifts away from the ideal transfer function when increasing input value. This error can also be trimmed near full-scale (FS) level, but requires some sophisticated methods.

4 DAC architectures

Many different DAC architectures evolved of timer. In the following section some common architectures will discussed for completeness of the topic. But the focus lies mostly on a technique which is used in nearly all modern audio equipment: the sigma-delta conversion. After this section the reader should be able to understand the major advantages and drawbacks of the discussed methods.

In general a digital to analogue converter is an electronic components that converts a digital coded value into an analogue. In order to understand the function of a DAC it is not necessarily important to know where these digital values come from, but it should be noted that because there exists only a finite number of digital states in a given range (e.g. for a binary number with 4 digits there exists $2^4 = 16$ possible analogue output levels) the analogue output will have **quantization error**. Usually quantization errors are introduced at the stage of analogue to digital conversion, where the analogue input signal is sampled, where each sample has only a finite number of states. In case of an pure digital audio system, as currently used by the project group, quantization errors can the difference between the internal representation of the signal and its approximated analogue output value. Thus errors are mainly introduced by truncation of a sample.

4.1 Thermometer DAC

The thermometer dac architecture is shown in figure 3. It consists of 2^n equal resistor R and a switch that controls whether the resistor is connected to the output. The switches are controlled by an n-to-2^n decoder. While this architecture is simple, has high precision and is fast, the number of switches and resistors grows exponentially with the input size, which makes it very expensive in manufacturing. Nonetheless in some use cases where only low resolution is required, such as digital potentiometers, thermometer are still a viable option[3].

4.2 Binary weighted DAC

As seen in 4.1 a decoder and 2^n resistors were required to derive the analogue output signal. By choosing the resistor values as a power of two, the number of components can be reduced. However resistors with high precision are hard to manufacture and therefore only used in mid-scale DACs. Another disadvantage is that binary weighted DACs are not necessarily *monotonic*, which means, that an increase of the digital input value does not increase the output voltage directly [3].

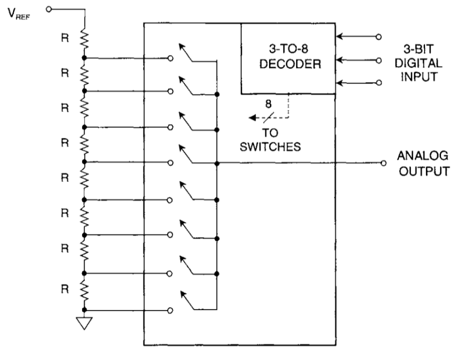

Figure 3: Thermometer DAC architecture[3].

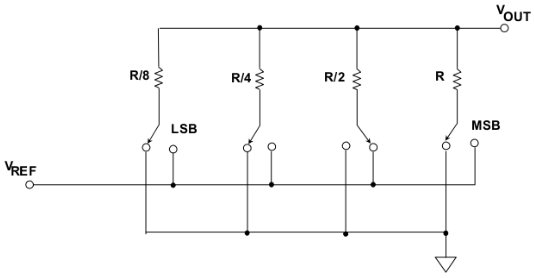

Figure 4: A binary weighted digital to analogue converter[3].

4.3 R-2R-Networks

An R-2R DACs uses an R-2R network as a building block, which consists of a series of resistors with a ration of 2:1.

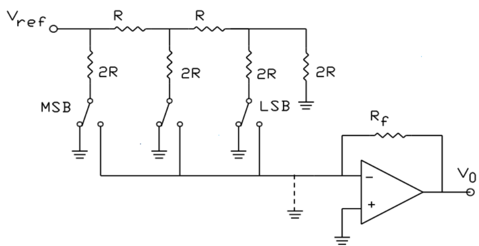

Figure 5: A DAC build from by an R-2R network.

The key principle of the R-2R ladder is that the current flowing at each node of the ladder is divided by a factor of two, according to the Kirchhoffs law [2].

The advantage of the R-2R DAC is that the number of resistors grows linear with increasing resolution. Only $2N$ resistors are required for an N input DAC.

4.4 $\Sigma\Delta$-Digital to analogue converter

To approach the topic of $\Sigma\Delta$-conversion it is somehow easier to discuss the key concepts with the analogue to digital conversion first. Digital to analogue conversion will be explained afterwards, because it is working similar.

A $\Sigma\Delta$ DACs are using the method of *sigma-delta modulation*. This modulation technique encodes a higher resolution signal to a lower resolution signal, introducing a quantization error, which the modulator keeps track of.

The error is corrected if it exceeds a certain threshold. Doing this, the *average* of encoded outgoing signal fits roughly the original signal. Although the quantization noise is not fully eliminated, it is shifted to higher frequency regions. This process is called noise shaping. Together with oversampling, sigma delta modulators achieve very good signal-to-noise ratio in a lower frequency regions. We will analyse this in more details in the following.

Figure 6 shows the architecture of a sigma-delta digital to analogue converter. It consists of a digital interpolation filter, a sigma-delta modulator, a 1-Bit DAC and an analogue low-pass filter.

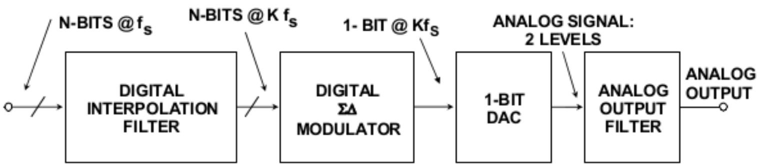

Figure 6: Architecture of a $\Sigma\Delta$-DAC

The converter accepts a PCM coded signal at a given sampling frequency f_s. The output is an analogue representation of this signal. Because the modulator is working with a much higher sampling frequency Kf_s the sampling frequency of the input signal have to be increased by a constant factor K. This process is referred to as *upsampling* or *interpolation*.

In the upsampling process, every sample of the original input is concatenated with $K-1$ zeros. Doing this, the output signal of the interpolation filter will have distortion in the frequency band above the Nyquist-frequency $f_{Nyquist} > \frac{f_s}{2}$ and thus have to be filtered using a digital low-pass filter.

Figure 7 shows the internals of a single bit, first-order sigma-delta ADC. The key concept and basic components does not change, regardless of the conversion direction.

The modulator consist of a summation node, an integrator, a single bit ADC and a single bit DAC that is feed back summation node. In case of a digital to analogue modulator the integrator is replaced by an accumulator and the comparator can be removed and instead the MSB can be feed back.

The modulator works as follows: assume that the modulator has just been initialized and there is a value at the input (analogue or digital, does not matter). The output of the sum is equal to the input value. This holds for the integrator as well. If the input of

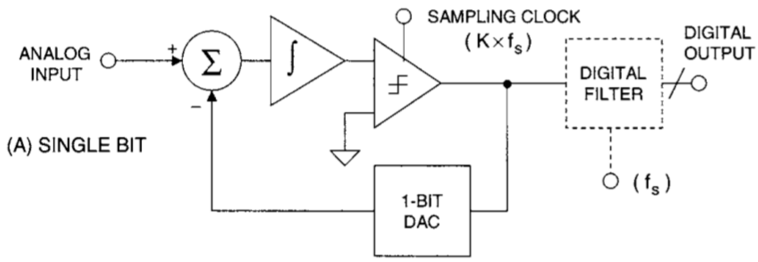

Figure 7: Single bit $\Sigma\Delta$ modulator

the comparator is greater than ground (or some arbitrary threshold) then the modulators output is a one, zero otherwise. In case of a digital to analogue modulator the "one" output is translated by the DAC into V_{ref}, a zero into $-V_{ref}$. Because the output signal is feed back and subtracted from the input signal, the summation node gives the quantization error, which the integrator keeps track of. This generated a bitstream that where the density of "ones" increases when the input signal is trends towards full scale and decreases if the input trends towards negative full scale. Figure 8 shows this effect. As stated before the average of the bitstream represents the input signal, which implicates a single value of the bitstream is meaningless.

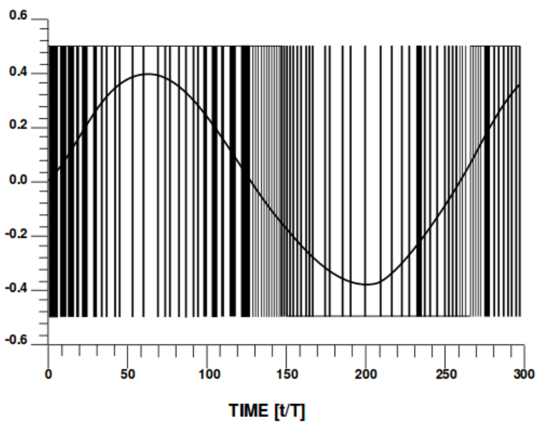

Figure 8: Output of a $\Sigma\Delta$-modulator [5]

The process described above repeats with at a high rate (usually 64x or 128x of the original sample frequency). The reason to do this oversampling is called noise shaping. The quantization noise introduced by the conversion is distributed equally over the whole band, up to the sampling frequency f_s. By increasing the sampling frequency the noise is distributed over the larger band, thus the noise in the band of interest decreases. By keeping track of the error, the noise can be further shifted out of the band of interest.

The output of the modulator has to be filtered by of low-pass filter with a cut-off at the original sample frequency to eliminate the noise in higher frequency region.

There are several possibilities to increase the accuracy to the modulator and to de-

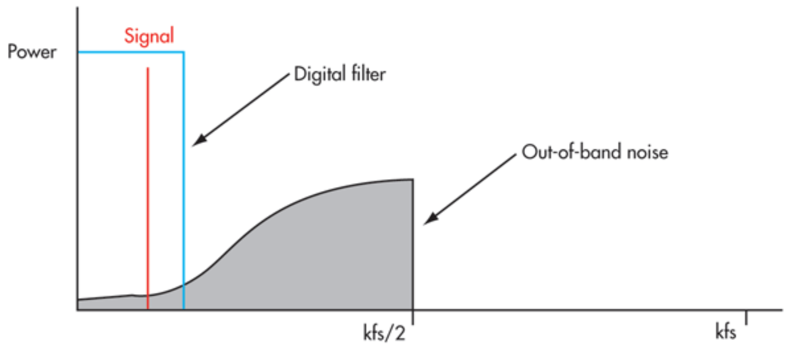

Figure 9: Noise shaping function of a ΣΔ-modulator [4]

crease the quantization noise. For example the number of stages can be increased. In audio applications it is common to use five or more stages [3]. It is also possible to increase the bit width of the outgoing DAC, which leads to an *multi bit* sigma-delta modulator.

5 ADAU1761

In order to achieve high quality audio, the data sheet of the project groups main platform will the investigated in this section.

Figure 10 shows a block diagram of the adau1761 audio codec internals.

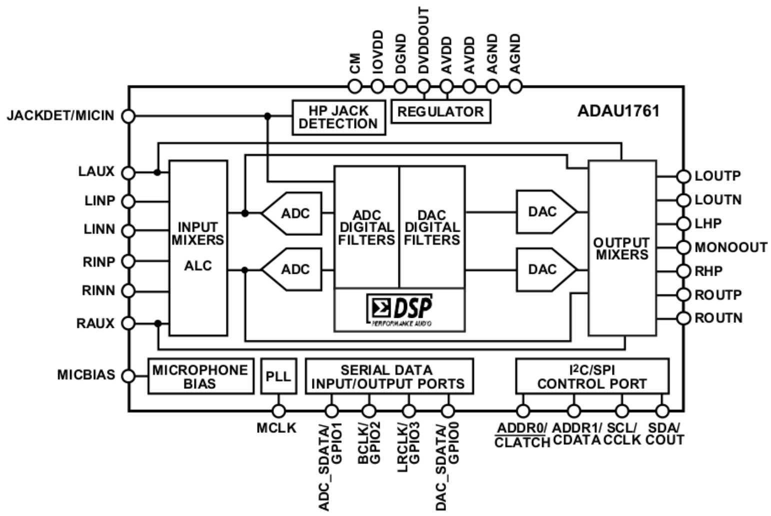

Figure 10: Block diagram of the Analog Devices adau1761 audio codec [1]

5.1 Theory of operation

The audio codec consists of four 24-bit sigma-delta modulators which were already discussed in 4.4, two in AD direction and two in DA direction. One converter for each direction is responsible for one stereo channel. There exist an input (record) mixer and output mixers (playback) to alter the signal path for each direction. A DSP core, embedded in the audio codec is capable to process the input or output directly inside the codec.

5.2 Interfaces

More interesting for the project group is how the audio codec is controlled, how data is feed to the codec and what control capabilities are offered.

In general data is supplied via an serial interface. The data has to be in the two's complement with the MSB first. Two channels are supported in stereo mode, which is the default mode, four or eight channels are supported in TDM mode. The protocol used to transfer the data is referred to as I2S [6]. The data to control the codec is written by the manufactured provided driver via I2C into the control register of the codec. Using the control registers the datapath can be altered (i.e. passing the input signal directly to the output), the DSP can be programmed and several control flags can be set (i.e. muting the output, which is interesting for pop/click suppression).

6 Conclusion

There exists several method to go from the digital to the analogue world. While all of them are suitable for audio applications, nowadays it is common to use $\Sigma\Delta$-modulation, because it can achieve high resolution and a good signal to noise ratio in the limited band used for audio. Thus the codec is well suitable for to achieve the projects group goals.

References

[1] Analog Devices. Adau1761 datasheet, December 2013.

[2] P. Horowitz and W. Hill. *The Art of Electronics*. Cambridge University Press, 2006.

[3] Walt Kester and inc Analog Devices. *Data conversion handbook*. Elsevier, Amsterdam, 2005.

[4] Arash Loloee. Understanding delta-sigma modulators, December 2013.

[5] Motorola. Principles of sigma-delta modulation for analog-to-digital converters, December 2013.

[6] Phillips. I2s protocol specification, December 2013.