Delta-Sigma Audio Conversion

ET 4801 Project Report

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Introduction

In the modern world, digital audio is ubiquitous. There are analog to digital (AD) and digital to analog (DA) converters in our laptops, phones, TVs, Bluetooth speakers, and more. Because the world around us is continuous and so is sound, systems and methods must be employed to capture and reproduce sound in a discretized digital realm. There are many methods for doing just that. One of the most popular methods for conversion in professional audio is the delta-sigma technique. As Paul Horowitz and Winfield Hill explain in The Art of Electronics, delta-sigma converters are “sometimes called a ‘1-bit DAC.’ That’s a seriously misleading name, though, because these things in fact deliver stunningly linear output signals of high resolution.” (888)

My major at MTSU is in computer science with minors in mathematics and electronics. In addition to my current studies, I have a previous degree from Belmont University in Audio Engineering Technology and work as a music producer. My main academic area of interest is in digital signal processing. AD and DA conversion represent a major intersection of my studies at MTSU and my personal interests.

This report documents my experience building both the Beis AD24QS and the Beis DA24QS. Two DIY converter kits that can be built to perform professional quality delta-sigma conversion.

Delta-Sigma Conversion

Baker describes delta-sigma converters as being roughly “three-quarters digital and one-quarter analog” (“Part 1” 13). The conversion process itself can be broken into two main sections: an oversampling modulator and a digital/decimation filter (Horowitz 924). Together, these components generate a high-resolution data-stream output (“Part 1” 13).

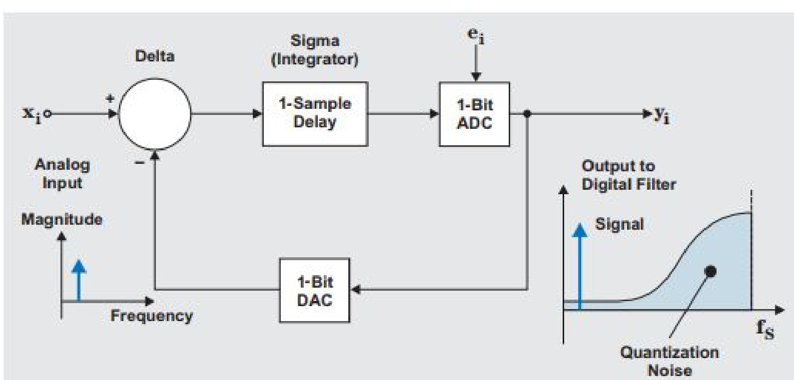
A unique characteristic of delta-sigma converters is that they feature two sampling rates: an input sampling rate and an output data rate. This is due in part to the operation of the modulator (“Part 1” 13). In digital audio, sampling rates are very important and help to indicate the highest frequency that can be recorded which is half the sample rate. This frequency is also known as the nyquist frequency. In the case of delta-sigma converters this sampling rate would be the output data rate. The output data rate is determined by the decimation filter. The input sampling rate is determined by the modulator.

Oversampling Modulator

The modulators operates by sampling the incoming analog signal at an oversampling rate (OSR) times higher than the output sample rate (Horowitz 925). This allows the modulator to create a single bit data stream that is fed to the decimation filter (“Part 1” 13). This data stream’s average value matches the input signal being sampled (Horowitz 925).

Being a bit stream though, this representation causes a lot of error, or quantization noise, in the output data-stream as the input signal is either represented as present or not at any given time step (Horowitz 925). As such, an integrator is used to shape the quantization noise out of the desired operating range (“Part 1” 14). In effect, the integrator acts as a single sample delay (“Part 1” 14). When the the ADC’s output is fed back to the input like in the figure below it acts as a high pass filter giving the desired noise shaping (“Part 1” 14).

Fig 1. First Order Delta-Sigma Modulator in the Frequency Domain



(“Part 1” 15)

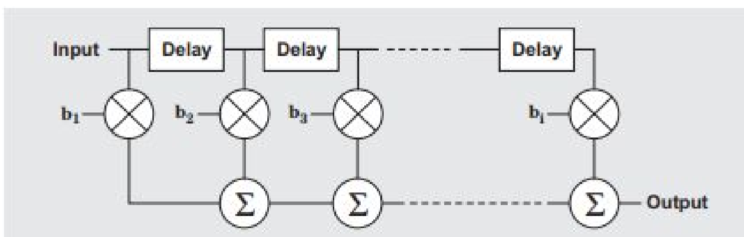
Baker explains “quantization noise for a first-order modulator starts low at zero hertz, rises rapidly, and then levels off at a maximum value at the modulator’s sampling frequency” (“Part 1” 15). Integrators can be chained together to give even better noise shaping performance. This causes the quantization noise to be moved to even higher frequencies (“Part 1” 15). In real world applications, it is not uncommon to see sixth-order modulators (“Part 1” 16).

Digital/Decimation Filter

The output of the modulator is an output data-stream with the quantization noise shifted to frequencies above the desired operating range. This stream is received by the second stage of the delta-sigma converter, the digital/decimation filter.

The first part of the second stage is a digital lowpass filter. The lowpass filter has the effect of attenuating the shifted quantization noise of the previous stage (“Part 2” 5). The most typical lowpass filter found in delta-sigma converters is an averaging filter, like the one pictured in figure 2 (“Part 2” 5).

Fig 2. First-Order, Low-Pass Averaging Filter



(“Part 2” 5)

The output of the filter stage still resembles that of the incoming data-stream, but with the quantization noise severely reduced (“Part 2” 6). It is the averaging filter that is also responsible for giving the 24 bit encoding of the data stream at a very fast clock rate (“Part 2” 6).

The decimation filter, as previously mentioned, is responsible for reducing the sample rate to the desired output rate. The decimation filter does so by throwing away a large number of samples in the data-stream. By doing so, the desired sample rate can be achieved (“Part 2” 6). It may seem that a lot of useful data is being lost; however, it is important to remember that a low pass filter was applied in the previous part of the digital/decimation filter process (“Part 2” 6). This means that the decimated signal actually contains the same information as the output of the digital filter while using significantly less data, illustrated by figure 3. The output of the decimation filter represents the completely digitized signal.

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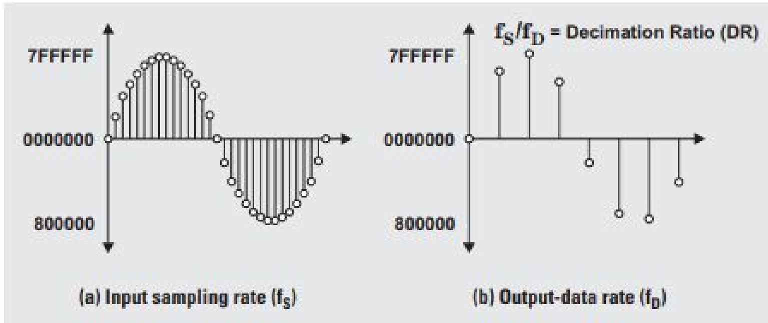
Fig 3. Digital/Decimation Filter Input and Output



(“Part 2” 6)

Beis AD24QS and DA24QS

Uwe Beis is a German electronics enthusiast that offers many do it yourself electronics kits on his website. For this project, I ordered the AD24QS and DA24QS from him to construct.

The AD24QS makes use of the Cirrus Logic CS5361. The CS5361 is a fully integrated chip for delta-sigma conversion. It features a fifth-order, multi-bit modulator followed by a digital/decimation filter. It boasts 24 bit resolution stereo conversion with very low total harmonic distortion (THD) and a dynamic range of 114 dB.

The DA24QS uses the Cirrus Logic CS4398. The CS4398 is also a delta-sigma converter and boast 24 bit resolution with low THD and a dynamic range of 120 dB.

Project Budget

Item	**Cost**
AD24QS	$164.93
DA24QS	$174.63
AD-IOA	$117.53
Total:	$457.09

Photos

Photo 1. Construction AD24QS in Progress

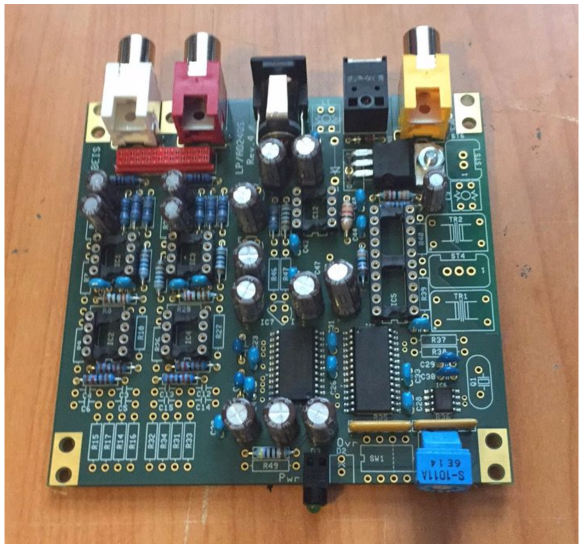


Photo 2. AD24QS and DA24QS Installed in Housing

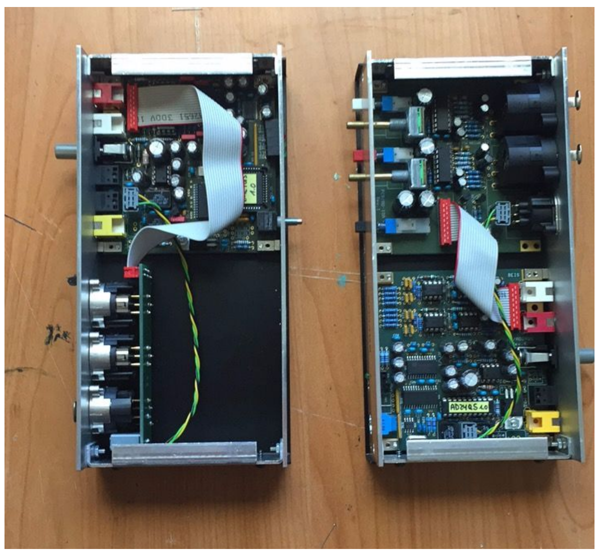


Photo 3. Complete and Operating AD24QS and DA24QS

THD

There are many useful parameters that can be collected around the performance of converters. One of which is that of Total Harmonic Distortion (THD). THD "is a measurement that tells you how much of the distortion of a voltage or current is due to harmonics in the signal" (Williams). In the case of audio conversion, the lowest THD possible is desirable.

A voltage or current that is sinusoidal at a single frequency features no harmonic distortion (Williams). A square wave represents the highest amount of harmonic distortion. In general, the farther a waveform is from being sinusoidal, the greater the harmonic distortion (Williams). In the real world, harmonic distortion is unavoidable but is something that can be minimized. Thus it tends to be a good measure of a converter's performance.

The mathematical definition of THD is as follows:

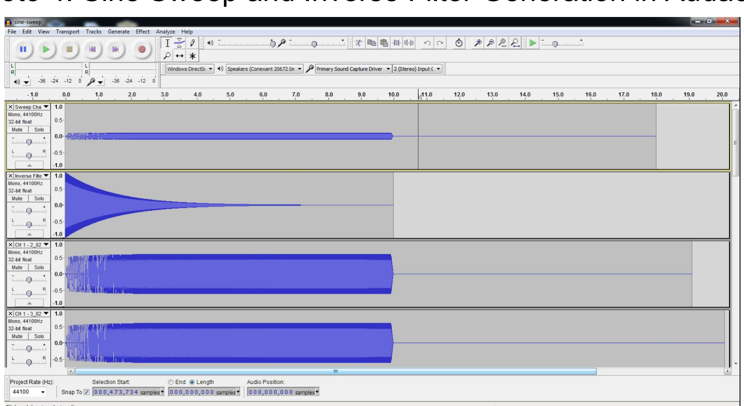
$$THD = \frac{\sqrt{\sum_{n=2}^{\infty} V_{n\_rms}^2}}{V_{fund\_rms}}$$

(Williams)

Collecting Data

With THD now defined, a method had to be used to collect data on the performance of these converters. In his paper "Simultaneous measurement of impulse response and distortion with a swept-sine technique," Angelo Farina describes a process by which using an exponential sine sweep and its inverse filter will separate the impulse response of a system from its harmonic distortion (Farina 1). Fortunately, he has made a plugin for Audacity, a free recording software. This plugin handles both the generation of the sine sweep and the inverse filter operation.

Photo 4. Sine Sweep and Inverse Filter Generation in Audacity



Using this approach, it became possible to treat the converters as a single system when measuring harmonic distortion. The converters were connected directly to each other using professional balanced cable for minimal noise. They were then digitally connected via AES/EBU to a MOTU HD124 recording interface so that they could be accurately clocked together to minimize any potential clock jitter. The impulse response was then played out the DA24QS and recorded by the AD24QS at a sample rate of 44.1 kHz. Assuming the distortion introduced by cabling is negligible, this approach gives an accurate depiction of the converter's performance alone. 10 second sine sweeps were recorded three times for each channel. They were then inverse filtered, using Farina's plugin to extract the impulse response and distortion contained in each sine sweep.

Photo 5. Setup for Impulse Collection

Analyzing Data

Following data collection, Python was used for analyzing the generated impulse responses. The Python script is attached at the end of this paper. Paiva et al. describe a method for taking the output of Farina's plugin and locating the harmonic distortion. The equations for which are shown here:

$$T = \left(\frac{2\pi M}{k-1} - \frac{\pi}{2}\right)\frac{\ln f_2/f_1}{2\pi f_1},$$

(Paiva 2)

$T\ =\ time\ in\ seconds\ of\ fundamental,\ M\ =\ Length\ of\ sine\ sweep\ (s),\ k\ =\ k^{th}\ harmonic$

$f_1\ =\ start\ frequency\ of\ sweep,\ f_2\ =\ end\ frequency\ of\ sweep$

With T, the offsets of the harmonics can then be calculated using the following equation:

$$\Delta n_k = \frac{T f_s \ln k}{\ln(f_2/f_1)}$$

(Paiva 2)

$\Delta n_k \; = \; sample\ offset\ from\ fundamental,\ f_s \; = \; sample\ rate$

Table 1. Harmonic Offsets for 10 Second Exponential Sine Sweep

Harmonic	Offset (sample)	Harmonic	Offset (sample)
2	7450.898974	16	29803.5959
3	11809.39547	17	30455.27269
4	14901.79795	18	31069.68992
5	17300.45166	19	31650.87875
6	19260.29444	20	32202.24961
7	20917.31791	21	32726.71338
8	22352.69692	22	33226.77447
9	23618.79094	23	33704.60314
10	24751.35063	24	34162.09239
11	25775.8755	25	34600.90332
12	26711.19342	26	35022.50147
13	27571.6025	27	35428.18641
14	28368.21688	28	35819.11586
15	29109.84713	29	36196.32561

Offsets are fractional due the discretized nature of digital audio, meaning that true peaks occur between samples. For the purposes of this project, the fractional offset was considered negligible and the maximum sample was used as a representation of the peak voltage. These peak voltages were then converted to RMS values and used to find the THD of the system using Williams equation from above. The THD of the system is displayed below:

Table 2. THD of AD24QS and DA24QS

Sweep	Channel 1	Channel 2
1	0.001679225197	0.001901753808
2	0.001542641613	0.001506403164
3	0.002207576775	0.001602943836
AVG	0.001809814528	0.001670366936

As can be seen in the table above, the AD24QS and DA24QS boast very low harmonic distortion. Channel one features 0.181% THD and channel two features 0.167% THD. From a distortion standpoint, these converters are truly of a professional caliber.

Conclusion

Delta-sigma converters are a part of everyday for most people living in the United States. They can be found in our car stereos, dvd players, phones, and just about any other device that play back high quality audio. They are also the common choice for professional grade audio equipment. The Beis AD24QS and Beis DA24QS are no exception when it comes to quality delta-sigma conversion. This project truly helped to further my understanding of digital audio in the modern age.

Works Cited

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