ABOUT SAVITR, AUDIO ANALOG TO DIGTAL CONVERTER

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Science Offers Clarity

While different digital signal processing markets require different features, they share a basic concept, the Nyquist Theorem about digitizing signals. It states that a signal contained within a limited frequency bandwidth can be digitized at a twice (or faster) rate of the highest signal frequency with absolutely no loss of detail. Assuming an audio bandwidth of 20KHz, a theoretical sample rate of 40KHz is sufficient for a perfect reconstruction of the original signal.

This concept, while not very intuitive, is correct. It tells us that sampling faster and creating larger data files yield no advantage, no additional detail. Yet, sampling at twice the bandwidth is not practical because it requires a “brick wall filter” to create the required limited bandwidth signal, removal of all the energy above half the sample rate. The solution is to use practical filters and sample a bit faster than twice the bandwidth.

A familiar example is the CD format aimed at providing the 20Hz-20KHz Hi-Fi bandwidth. The theoretical sample rate of 40KHz is increased to 44.1KHz. The analog filter needs to pass the signals up to 20KHz and block signals above 22.05KHz (Nyquist frequency is 44.1KHz/2 = 22.05KHz).

Background

I have designed converters since 1973. Those converters were designed for high-speed instrumentation, postal weighting scales, telecommunications and medical MRI. The two most basic parameters for conversion are signal bandwidth (the required sample rate) and conversion accuracy (the required number of bits).

The previous industries I worked for provided the designer with clearly specified goals and requirements. Then the products were tested thoroughly to insure meeting performance specifications. I can assure you that no one would deviate from such requirements for an MRI converter. The customers were technical people, and the communication with them was about technical details.

Physics always imposes a tradeoff between speed and accuracy. A weighing scale requires very low bandwidth, allowing operation at low sample-rate, which enables great accuracy. Digital video operates at higher sample-rate than digital audio, but provides fewer bits. More is not always better - Sampling faster than necessary results in both increased file size and decreased accuracy. For each given application, there is such a thing as optimal sample-rate. Too slow or too fast causes problems.

Due to my love of music, I was very excited to switch my full attention to the emerging digital audio market. I came in ready to apply my previous converter experience, but was surprised at how different the audio industry was from the previous markets I worked for. Most customers were not technical. Also, most products lacked a lot of written specifications.

To this day, a typical distortion (if published) is still specified at only 1KHz, which is far from sufficient to describe audio converter performance. That alone may explain much of the disregard for technical specification. Lacking comprehensive technical information leaves the door open to opinions, often influenced by “marketing-oriented misinformation”. Plenty of “alternate facts” are due to the difficulty to correctly corollate specific distortions to their real cause.

Given the above, I approach much of my work the old way, focusing on very detailed design specifications and the required set of extensive measurements.

<u>Analog and Digital Differences</u>

There are two sides to converter design: analog and digital.

The analog advantage is negligible latency, but achieving analog precision is difficult. The major drawback to analog design is lack of analog memory. Other issues are susceptibility to noise, limited component accuracy, temperature dependance, and signal routing issues.

Digital processing is the main cause of latency. But digital offers a lot of advantages, such as memory, precision, temperature independence, computation, very high immunity to noise and interference. It is also easier to work with digital signals.

Savitr AD Goals

Sound is analog; We cannot hear a digital signal. We use digital because it offers us UTILITY, it enables us to do things that cannot be done with analog. Conversion to and from digital is NOT ABOUT SOUND. The A/D changes the signal to a format enabling storage, processing and transmission of the audio. The D/A enables listening to the result.

The converter function is to convert. Sonic alterations can be done with analog gear, digital gear, and computer software. A converter is not an equalizer, mixer or compressor. My main goal was always clear: to achieve accurate conversion with minimal sonic alteration. Unfortunately, I could not base the Savitr A/D design on the existing Lavry gold A/D series.

The first reason was latency. The MKIII and MX are mastering A/D converters. The latency is high, which limits their use to studios and applications where latency is a non-issue. High latency is a real problem for overdubbing. My first new goal for Savitr was to have extremely low latency.

The next issue was to add 176.4Khz and 192KHz sampling. My previous A/Ds sampled up to 96KHz, but there were many customers that insisted on having 192KHz included. The market required it. This was a business decision, contrary to science and engineering principles. I refused any suggestion for 384KHz and 768KHz, which is a crock.

But there was more I wanted to accomplish. I wanted the AD to perform beyond the sensitivity of any golden ear. Sonic alterations should be below human perception.

I was very satisfied with matching the -126dBFS dynamic range of the previous Lavry gold Ads measured with no A weighting (-128dBFS with A weighting). Such low noise floor is 30dB better then a CD format (-96dBFS). A -126dB noise floor is comparable to that of a sensory deprivation tank.

The remaining and most important goal was to improve the distortion performance to be better than -112dBFS (.00025%) across the audible range. This is about a factor of 4 better than the gold MKIII and MX. It was the biggest challenge.

Added Savitr Features

There are two hardware configurations for the Savitr.

The first configuration offers a second output with independent format and sample rate. The second configuration offers multiple word clock outputs.

The Savitr (AUX) has a main output and a second output (the auxiliary output). The two outputs can operate independently, with different internal or external clock sources and independent format setting (AES or SPDIF). The auxiliary output may benefit studio work flow.

Example: simultaneous outputs. 1.) 96KHz internal clock with AES format for archive purpose and 2.) 44.1KHz external clock with SPDIF format for commercial release.

I expected to find out that many users don't need the additional auxiliary output. Removing the auxiliary output provides space for another configuration, the Savitr (MWC).

The Savitr (MWC) offers multiple word clocks outputs (5 clock outputs), instead of the one clock output of Savitr (AUX). It enables the Savitr AD be the master clock for up to 5 devices.

Clocks for Digital Audio

The ideal way to clock an A/D is to use an internal clock (for lowest jitter). Clock jitter can add distortions and noise to an A/D output. A/D jitter impact cannot be removed. It is part of the audio forever. D/A jitter can be improved, just get a better D/A.

Internal clock is best because the clock source is located inside the AD chassis and near the converter. External clock may contribute added jitter because it is located in another chassis, which involves a driver, a coax cable and a receiver. The Savitr (MWC) enables the A/D to operate with internal clock while serving as a master clock for a system.

I described my approach to Clock jitter and Clock Accuracy in the paper:

https://lavryengineering.com/wp-content/uploads/2022/01/Clock_Jitter_and_Clock_Accuracy_for_Digital_Audio.pdf

The bottom-line is that crystal-based technology is more than adequate to offer the needed timing precision, pitch accuracy, and jitter performance.

A 0.01% (100 part per million) clock time error would make an hour-long performance longer or shorter by only 0.36 seconds. A 3-minute performance would have an error of only 0.018 seconds. Crystal technology offers much better than 0.01% accuracy. Synchronized operation of all devices to within 0.01% results in imperceptible time duration error.

A 0.01% error will alter the pitch by around 0.2 cents. The interval between 2 piano notes is 595 cents. We don’t hear a 1 cent error. Also, a crystal with better than 0.01% yields even smaller pitch error, far below human perception.

The jitter of a good crystal oscillator is more than sufficient to offer 120dBFS dynamic range. The Savitr offers 126dBFS.

There is no need for atomic clocks for digital audio. The extra time accuracy is not beneficial. Better pitch accuracy is not needed and internal crystal operation offers less jitter. The Savitr (MWC) may eliminate the need for an external clock device.

I hope you enjoy using Savitr A/D and Quintessence D/A. They are the culmination of many years of experience.

Cheers!

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