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Converter Technologies Boost Audio

by Steven Green

A decade ago, the industry's best analog-to-digital converter chip sets generated 18-bit data, had 97 dB of dynamic range, had a maximum sample rate of 50 kHz and sold for upwards of $50. The situation for digital-to-analog converters was not much better. The general consensus at the time was that converters were the greatest barriers to achieving comparable performance to analog systems and the broad acceptance of digital audio.

My, how times have changed! Today's converters support 24-bit word lengths at sample rates up to 192 kHz, achieve dynamic range performance that challenges the limits of human hearing, are generally easier to design with, and sound better than their predecessors at prices that were unheard of a decade ago. But this begs the question of how did we get here, and what comes next?

Cirrus Logic (then Crystal Semiconductor) introduced integrated single-bit delta-sigma converters to the audio community in the late '80s. This shift from R-2R ladder architectures was a monumental advance in converter technology that increased dynamic range, lowered distortion and improved overall sound quality. Single-bit delta-sigma conversion was the preferred architecture throughout the '90s for professional, personal computers and consumer audio applications. Single-bit delta-sigma converters continue to play a significant role in professional audio and dominate consumer and PC applications. However, single-bit architectures became increasingly less attractive as the industry progressed higher up the performance ladder.

Beginning in the mid to late '90s, converter manufacturers began to investigate various multi-bit delta-sigma topologies for digital-to-analog converters in order to reach the performance levels of 120 dB dynamic range and distortion performance better than 105 dB. There are many advantages to multi-bit delta-sigma at the higher performance levels, including reduction of idle tones, lower out-of-band noise and improved insensitivity to clock jitter. However, the greatest challenge to multi-bit delta-sigma is dealing with the mismatch in the conversion elements that, if not dealt with properly, can lead to high levels of distortion.

There are several techniques that have been implemented over the past few years that have met with varying degrees of success. All of these architectures have included a circuit block, following the delta-sigma modulator, which performs what is commonly referred to as dynamic element matching (DEM). DEM randomizes the mismatch errors in such a way as to convert the error into noise and not distortion. The latest innovation from Cirrus Logic moves the mismatch-shaping function within the feedback loop of the delta-sigma modulator. Not only does this randomize the mismatch error, the errors are noise-shaped by the modulator to further lower the audio band noise and maximizes audio quality.

There has been a lot of discussion, including several AES papers and magazine articles over the past few years, discussing why 96 kHz and 192 kHz sample rates sound better if our hearing is limited to 20 kHz. Much of this discussion has focused on parameters associated with digital interpolation and decimation filters. One of the characteristics of digital filters is that they introduce a time delay or latency. This delay is generally referred to as "Group Delay" in the converter data sheets and specified in multiples of the sample rate. Latency is (fortunately for the music business) truly inaudible in all but one application.

Consider the situation for a moment where a performer has the live sound mix fed back via headphones or loudspeakers. It becomes very difficult, if not impossible, for an artist to continue to perform if there is as little as 50 milliseconds of delay in the monitor signal chain that creates the perception of an echo. These effects are well documented, and it is very likely that the majority of the population has experienced this phenomenon with a poor telephone connection. There is also anecdotal information that suggests delays of only a few milliseconds can adversely affect audio quality. A typical group delay specification for interpolation and decimation filters is within the range of 30 to 40 samples periods for each stage for a combined delay of 60 to 80 samples periods. At a 48 kHz sample rate, this combined delay (1.25 to 1.67 milliseconds) is sufficient to place the delayed signal within the range that may affect audio quality.

The latest generation of converters addresses this issue with latency characteristics that are roughly 75 percent lower than previous solutions. Manufacturers of equipment intended for live sound applications are very much aware of the implications of delay and take every opportunity to minimize system latency to avoid the perception of an echo and preserve audio quality. The latest developments in digital interpolation and decimation filters make this task easier to accomplish.

Cirrus announced the CS5381 analog-to-digital converter and the CS4398 digital-to-analog converter at the 114th AES Convention in March 2003. These products achieve 120 dB of dynamic range and are targeted for the most demanding of applications. Cirrus Logic has introduced 14 new products targeted for professional audio applications since the 2002 fall AES Convention using this advanced technology.

Long a leader in audio technologies and recently cited by Forward Concepts as the industry leader in digital audio ICs, Cirrus continues to invest in the technology developments that will drive the pro audio market and create opportunities across the board for manufacturers and the industry at large.

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