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THE ABCS OF DACS

A LOOK AT THE DESIGN OF MULTI-FORMAT, HIGH-PERFORMANCE DIGITAL-TO-ANALOG CONVERSION SYSTEMS.

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Since the introduction of recorded music, there has been a quest to create audio recording and playback technologies that re-create the experience of a live musical performance in our living rooms. This journey has taken us from the purely mechanical systems of the late 19th century to the multichannel digital audio systems of the 21st century. Today, consumers have available two different digital formats for high-resolution audio, DVD-Audio (DVD-A) and Super Audio CD (SACD). While both of these formats deliver truly impressive sound quality, there are challenges to designing digital-to-analog conversion (DAC) systems that support both formats, are backward compatible to CD, minimize systems costs, and achieve the high-performance capabilities of each format.

A typical block diagram for conventional PCM digital audio system is shown in fig. 1. Notice that during the recording process the analog input is sampled in the delta-sigma modulator at a 64x over-sampling rate, 64 x 44.1 kHz or 2.8224 MHz for CD. This data stream is then decimated to a multi-bit PCM word at a lower sample rate. On playback, the data is retrieved from the disc and a digital interpolation filter and a digital delta-sigma modulator process the multi-bit data. Notice that in this process the information returns to a 1-bit data stream at a 64x oversampling ratio prior to the digital-to-analog conversion. The fundamental difference between the two technologies is that the SACD process theoretically does not include decimation or interpolation, and preserves the 1-bit data stream throughout the process.

DVD-A is similar to the CD format in that both formats utilize linear Pulse Code Modulation (PCM) encoded data. As a result, manufacturers of integrated DACs have generally addressed the needs of the systems designer. SACD is unique in that it does not use PCM-encoded data. As a result, there are several interesting challenges to implementing a system that maintains the inherent audio quality of SACD, has the capability of playing the linear PCM formats, and minimize systems costs. It is in this area that manufacturers of integrated

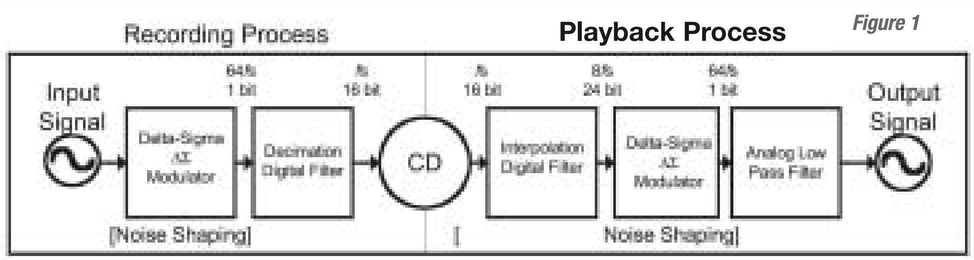


Figure 1

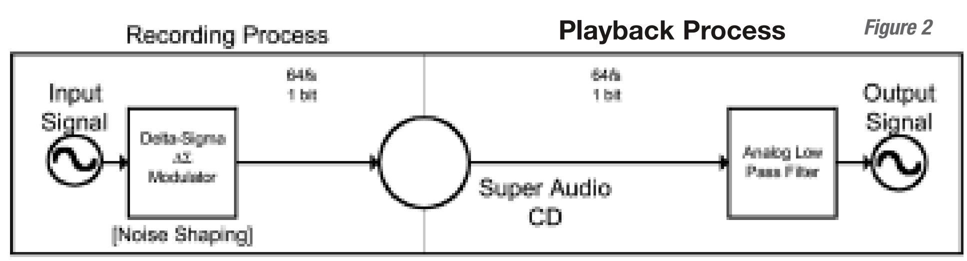


Figure 2

digital-to-analog converters have only recently addressed the needs of the systems designer.

Systems Requirements for Multi-Format Digital-to-Analog Conversion

There are several systems-related issues that are generally not addressed in today's integrated converters. For example, the majority of digital-to-analog converters require a plethora of external circuitry to support both DSD and PCM audio formats. In today's systems, the DSD and PCM data come from different decoders and, unfortunately in many of today's converters, the functionality of the converter pins changes between PCM and DSD modes. These differences require the use of external logic devices to reconfigure the converter pins for use in the selected mode and reroute the audio data. Not only does this external logic add cost, the additional design complexity can degrade audio performance. The preferred solution is to have dedicated serial audio inputs for both DSD and PCM operation, which requires the user to select between either the PCM and DSD inputs for the converter.

Not only are there systems issues on the digital side; there is another very common problem on the analog side. Most would expect that the loudness or volume of a converted full scale PCM signal would match that of a full-scale DSD signal. Unfortunately, this is generally not the case with the majority of the currently available converters. Converted DSD signal amplitudes are usually lower than those of PCM modes. These devices require the systems designer (and the end-user) to either accept this difference or assume the burden in cost and sonic degradation of the additional analog and control circuitry that allows the analog output stage to switch between gain settings.

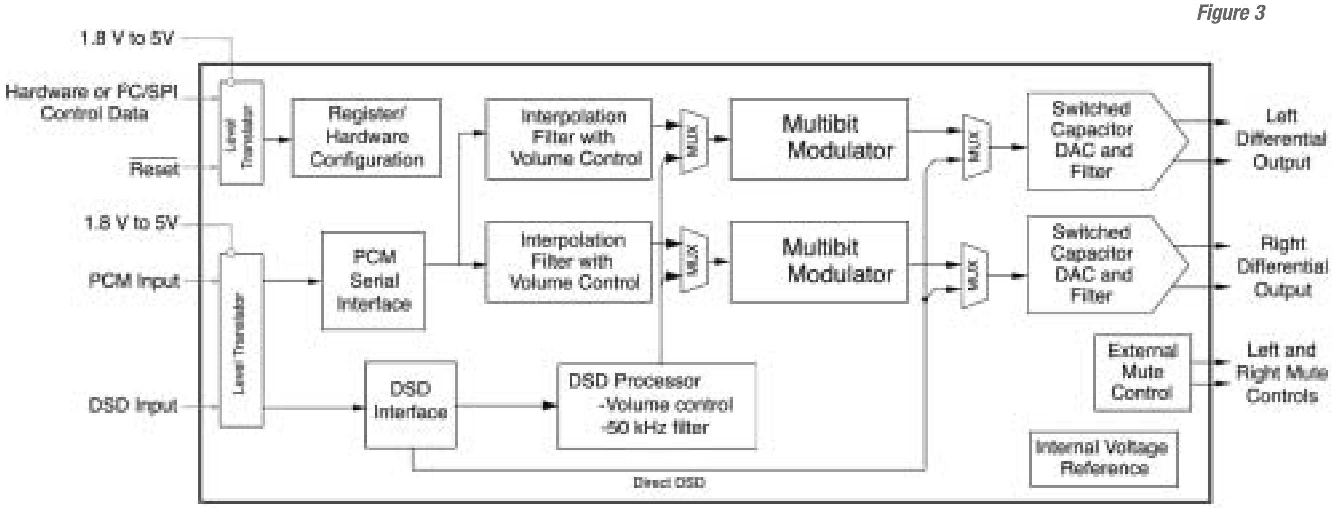
The majority of converters today include a digital volume control for PCM modes, and it is often the preferred solution since digital volume controls are more accurate, offer improved channel-to-channel tracking, are easier to control with digital logic, and are less expensive when compared to their analog counterparts. Unfortunately, as simple as this function appears to be, it is deceptively complicated for DSD and not commonly supported.

Requirements for Amplifier and Loudspeaker Protection

By their nature, analog-to-digital converters that utilize single-bit delta-sigma techniques generate a significant amount of quantization noise. These converters operate at high oversampling ratios, 2.8224 MHz for SACD, which distributes this noise over a wide bandwidth. In addition to the extended bandwidth, the quantization noise is shaped by the modulator in such a way as to minimize the noise that exists within what is generally considered to be the audio bandwidth. When operating in a PCM mode, the digital decimation filter in the A/D removes the noise that is beyond the audio bandwidth. However, for DSD this noise is transferred to the disc, and there are potential systems issues that can arise when this high-frequency noise is converted into the analog domain and applied to an audio amplifier and speaker.

These issues can include slew rate limiting in the amplifier stages as well as inter-modulation distortion and damage to the high-frequency drivers in loudspeakers. Philips Electronics and Sony Corp. recognize and address this issue in a document titled "Super Audio CD System Description." This document, commonly referred to in the audio industry as the Scarlet Book, is the defining document for Super Audio CD standards and systems requirements. Part 2, Annex E of this document recommends that, for protection of analog loudspeakers and amplifiers, a SACD player employ an analog low-pass filter with a cut-off

Figure 3



frequency of 50 kHz and a minimum 30 dB per octave slope (though this bandwidth recommendation is relaxed for wide-band audio gear).

Unfortunately, this recommended analog filter has several potential pitfalls that include the frequency and phase response variations that result from standard component tolerances, additional circuit complexity, potential performance degradation, and increased cost. The systems designer must also determine if this analog filter is to remain within the signal path when the system is operating in a high-sample rate PCM mode, such DVD-A at 192 kHz sample rate. If the decision is to remove the filter, an additional cost is incurred to provide the required switching circuitry and control logic, as well as a risk of degrading sonic quality with this additional circuitry in the signal path. The overwhelming majority of digital-to-analog converters that are capable of converting DSD require an external analog filter to meet the Scarlet Book recommendations. Fortunately there is another option. Since this noise originates in the DSD recording process and is present in the digital domain, it is feasible that the 50 kHz filter can be implemented digitally to meet the goal of protecting analog amplifiers and speakers.

Challenges of Processing DSD Data

As previously stated, DSD data is 1-bit data stream at 2,822,400 samples per second and there are difficulties associated in processing this data. The difficulty is related to the accuracy and speed required. For simple audio filters, the accuracy requirement varies with the square of the sampling speed and the computational speed varies directly with the sampling rate. With normal filter topologies, it rapidly becomes impractical to implement audio processing without first decreasing the sample rate. As a result, the standard processing technique has been to decimate the 1-bit data to multi-bit word at a lower sample rate that allows application of standard DSP techniques. Because of these issues, the majority of DSD converters currently available do not support any DSP functionality. The few that do, such as volume control and a 50 kHz low-pass filter, have implemented these features by decimation of the 1-bit data to a multi-bit word at a lower sample rate. Not only does this decimation process reduce the audio signal bandwidth, but it also violates the fundamental premise of Super Audio CD.

The latest investigations into these issues has led to the development of a novel and patented Direct Stream Digital processor. This proprietary DSD processor enables the implementation of a digital volume control, signal amplitude matching for DSD and PCM modes, and the recommended 50 kHz low-pass filter without the use of a decimation filter. This new architecture maintains the native DSD sample rate of 2.8224 MHz and preserves the sound quality and integrity of Super Audio CD.

The first action taken in the DSD processor is to translate the data stream of digital ones and zeroes to a similar data stream of 1's and –1's. This translation allows the data to be multiplied by a multi-bit gain coefficient in the second stage, where a gain coefficient of digital full scale equates to unity gain and decreasing the gain coefficient effectively attenuates the signal. This multi-bit data stream is then upsampled to match the rate of the digital delta-sigma modulator. The output from this stage is then applied to the multi-bit delta-sigma modulator. Historically, the signal transfer functions of delta-sigma modulators have been designed to maintain flat response over the audio frequency band of interest. However, in the case of DSD, this is no longer the desired response. The latest architectural development allows modification of the delta-sigma modulator signal transfer function to include the 50 kHz filter response that is recommended in the Super Audio CD System Description. This development preserves the integrity of the DSD data and takes advantage of a multi-bit delta-sigma conversion.

A Solution

The block diagram of the preferred solution is shown in fig. 3, as realized in the Cirrus Logic CS4398 DAC IC. This solution has independent inputs of DSD and PCM data, and independent level shifters for the serial audio interface and control ports. This solution requires the minimum in both analog and digital external components.

There are many commercially available recordings that demonstrate the viability of SACD and DVD-A as high-resolution audio formats. The ability of a player to fully deliver the performance capabilities of these formats in addition to standard CD, as well as AC-3 and other compressed audio formats, is truly a benefit to the consumer and pro alike. ■