By Joshua Israelsohn, Technical Editor

IF IT WEREN'T FOR AUDIO CONVERTERS, SOME OF OUR FAVORITE PRODUCTS WOULD JUST BE SPINNING THEIR WHEELS.

Illustration by Richard Borge

SOUND BYTES

SINCE HARVEY FLETCHER AND WA MUNSON in 1933 defined the first broadly adopted characterization of human hearing, research has refined our collective understanding of the auditory response (references 1 through 4). The accepted range of useful audible stimulus, however, has not varied significantly and remains essentially 20 Hz to 20 kHz and 0 to 125 dB SPL (sound-pressure level, referenced to 20 μPa at 1 kHz). Far more changeable have been the myriad ways we use sound to communicate information and entertainment, particularly since the advent of inexpensive digital technologies, such as high-density storage, low-cost communication, and high-speed signal processing. So, though we remain analog creatures in an analog universe, we increasingly depend on digitally encoded audible information and the converters that bridge the two domains.

Despite an essentially invariant concept of human auditory capability, variability in the set of desirable audio-converter traits is substantial and engenders many dozens of devices from a handful of the most active IC makers. Various applications use digital audio encoded in 8- to 24-bit samples at rates from 8 to 192k samples/sec (see **sidebar** "How much do we need?"). As much as applications dictate sample rate and resolution, such nameplate characteristics serve only as starting points for defining converters targeting particular market segments. Indeed, so common are certain sample-rate and resolution combinations that, by those measures, markets are blurred, not only among various consumer products, but also between consumer- and professional-audio devices.

HIT THE ROAD

With the exception of those for the uppermost market strata, audio converters for consumer applications no longer generally exist as stand-alone components. Rather, IC manufacturers integrate the converter function into audio subsystem chips that provide a host of functions targeting specific end products. These ICs rarely compete on issues of audio performance due to limitations of the signal sources or destinations; the fidelity requirements of, say, a mobile phone, portable MP3 player, or boom box are not problems that keep IC designers up at night. More important for these products are signal-processing features, such as volume, tone, and signal-dynamics controls; source selectors and mixers; and, in some applications, data-rate agility.

AT A GLANCE

▷ Audio-converter features and attributes do not stratify as clearly as the end products in which they find use.

▷ Integration is king in devices for most consumer applications. Converter performance is sufficient, and further significant improvements are unlikely in the near term.

▷ Plenty of room still exists for innovation, but these efforts in the consumer market will focus not on performance but on cost, size, power, and features.

▷ Requirements for the top-of-the-line consumer- and professional-audio markets are blurring, but distinctions remain. Economies of scale will drive many converters into both segments, though features attractive to one will go unused in the other.

Requirements that originate in handheld portable devices are finding their way into fixed-installation consumer equipment. These attributes include low power consumption, small packaging, and few support components. Thus, those companies that most efficiently integrate mixed-signal functions dominate much of the consumer end of the market. This notion holds true even when the function list includes nonaudio-specific items.

The WM9713L from Wolfson Microelectronics, for example, integrates dual codecs with a touchpanel interface for PDAs, smart phones, and handheld computers. A high-fidelity stereo codec provides dual microphone preamplifiers, a signal-source selector, an ADC, an automatic level control, tone control, a DAC, a phone/headphone/speaker-switch matrix, output amplifiers, and an AC'97 Revision 2.2-compatible interface. An on-chip PLL generates the sample clock for audio data rates to 48k samples/sec. A separate voice codec supports audio for telephone traffic through a separate PCM/I^2C interface. A resistive-touch-panel interface supports four- and five-wire touchpanels and measures XY-axis position and Z-axis pressure.

The 9713L's maximum THD (total harmonic distortion) at the line-level output is −74 dB at −3 dB full scale; the line output's minimum A-weighted SNR is 85 dB. The signal chain from the line-input port to the ADC yields −80-dB THD and 80-dB SNR. The headphone amplifier delivers a maximum of 45 mW; the speaker amplifiers drive 400 mW into 8Ω, bridge-tied loads.

Wolfson packs numerous other features into the 7×7-mm, QFN-48 package—enough to fill a well-organized data sheet longer than 100 pages. The features and flexibility are impressive. However, two standouts both center on the relatively mundane topic of power supply: On the positive, the $3.35 (1000) WM9713L operates with a digital supply voltage as small as 1.62V and an analog supply of only 1.8V. (Maximums are 3.6V except for the amplifier supply, which you can drive to 4.2V.) On the negative, despite experience with two previous generations of this type of product and no doubt substantial simulation during the design phase, the company gives no hint as to the device's power dissipation. This omission may well be a more complicated question than a single datum can fully answer: Clock rates, sample rates, operating modes, signal levels, and load impedances can all affect the total dissipation; that fact goes with the territory. Most portable products compete on a combination of features and per-charge operating time, and, though it is nice to know that the 9713L will keep ticking away long after many other functions have passed their low-line limits, it is reasonable to ask how much the dual codec hastens the battery's end of charge.

Be careful what you wish for, though. Texas Instruments devotes a full page in its TSC2101 data sheet to power-supply requirements. If you read it side by side with the part's block diagram and some 19th-century accounting technology, such as a pen and a scratchpad, you can quickly get a good idea of the dissipation for the IC's various operating modes.

Like the WM9713L, the TSC2101 integrates an audio codec with speaker and headphone amplifiers and a four-wire touchscreen controller for PDAs, smart phones, and MP3 players. The stereo DAC can play audio files at sample rates to 48k samples/sec. Its companion ADC is similarly rate-agile, but captures only monophonic inputs. The TSC2101 doesn't have a second codec dedicated to telephone traffic but fills out the QFN-48's pin count with a battery monitor and an on-chip temperature sensor.

The 2101 provides a stereo-headset interface, a cell-phone-headset interface, a monophonic, 8Ω speaker amplifier, and a 32Ω receiver driver. The 8Ω speaker amplifier can drive 400 mW and is rated for a maximum THD of −55 dB at 335 mW. The stereo-headphone amplifier supplies a 90-mW drive to 16Ω loads with −60-dB THD at 71 mW.

The $4.95 (1000) TSC2101's battery monitor measures 0.4 to 6V with an accuracy of ±15 mV over the −40 to +85°C operating-temperature range after a system-calibration step. The battery monitor, temperature monitor, and two auxiliary inputs share a multiplexed SAR (successive-approximation-register) ADC with the four-wire touchscreen port.

Texas Instruments also offers a low-power stereo codec, the TLV320AIC26, which trades in the touchscreen controller for more audio-signal-processing features and a 5×5-mm QFN-32 package. The stereo-DAC channel can reproduce 16-, 20-, 24-, and 32-bit/sample files. On-chip tone controls allow users to adjust bass, midrange, and treble, or you can configure the audio-processing block as an equalizer. Programmable power controls allow stereo-audio playback at 48k samples/sec with only 11-mW dissipation from a 3.3V supply.

You control the $3.25 (1000) AIC26 through a set of addressable registers through the codec's serial interface. On-chip state machines configure the various circuit blocks according to the registers contents. Exercise care to ensure that the configuration settings you invoke correspond correctly to the operating conditions your application imposes. For example, you can set the DAC's output to one of four voltage spans,

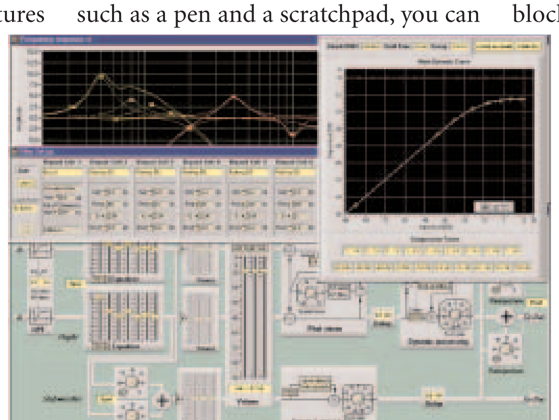

Figure 1 GUI-based development and configuration software gives you easy access to the sophisticated signal-processing capabilities of the Sigma-DSP product line from Analog Devices.

but be sure that the power-supply voltage meets the minimum requirements for the span you choose. The AIC26 provides two battery-monitor inputs that share a SAR ADC with the on-chip temperature sensor and an auxiliary input. The SAR ADC's output is available through an SPI interface.

The data-sheet conditions for measures of analog performance include a 2V p-p output span (the smallest of the four), a 48k-sample/sec data rate, and 3.3V and 1.8V supplies for the analog and digital sections, respectively. Under such conditions, the line output's THD is typically −95 dB with a minimum of 85-dB A-weighted SNR. Ironically, though the line output's THD carries only a typical specification, the headphone output offers a minimum of −55 dB at −1 dB full scale. This figure compares to a typical value of −91 dB under the same conditions. A spread of a few or even several decibels between a typical and a limit specification would cause not so much as an eyebrow to rise. But with 36 dB between the two, you may want to approach the typical specifications with caution if they appear on data-sheet items that are important to your application.

National Semiconductor's approach sheds the bell-and-whistle features in favor of further miniaturization: The company neatly tucks a stereo DAC, a voice codec, a speaker amplifier, stereo-headset amplifiers, a microphone preamp, PCM and I^2S data interfaces, and a control port into a 36-bump micro SMD package that measures less than 3.5 mm on a side. And, lest you imagine that this results in a 36-bump miniaturized Trump Tower rising preposterously above the mounting

HOW MUCH DO WE NEED?

Various segments of the audio industry have long been debating the desirability of digital-coding methods that use, alternately, greater resolutions and greater sample rates than 16 bits and 44.1k samples/sec. This performance baseline, which derives from the more-than-20-year-old Red Book CD standard, represents technical-capability limits the IC industry has long since passed. But increasing either resolution or sample rate demands greater storage per unit program time and greater real-time processing capability for both program creation and playback. As inexpensive as storage and processing resources have become, the argument for more of either must appeal to fidelity: the promise of a more lifelike listening experience.

I've spent considerable time listening to digital-audio equipment for both production and reproduction. As such, I wonder whether these parameters are those of interest or just those that are the most conveniently available for the largest segment of the population to debate.

The resolution of a digital-coding scheme limits that method's theoretical dynamic range; other issues further limit the attainable dynamic range. So, of the "more-bits-are-better-bits" camp, I ask: How much dynamic range is sufficient? Would a system that replicates, for example, any sound louder than a human breathing at nine feet to the sound of a jackhammer working at six feet unduly limit your experience? Well, that range is 90 dB, which the now-ancient CD Red Book coding more than amply accommodates (**Table A**).

Recognizing that nonlinear-recording media behaves gracelessly when it runs out of codes, it is imperative to prevent a digital-recording system from clipping: The artifacts that result splatter themselves all over the audible band in ways that are anything but subtle. Converter subsystems targeting the upper end of the consumer market often include dynamic processors—compressors or limiters—to prevent digital clipping. Converters for the professional market typically provide no dynamic processing. Purpose-made dynamic processors provide that function, many of which operate in the digital domain after the ADC has done its work. To prevent otherwise-unrestrained signals from driving the converter into the stops, studio personnel often use 20- and 24-bit converters with several decibels of overhead to accommodate unanticipated program peaks.

Professional equipment also has a legitimate need to accommodate the additional noise that various postproduction steps introduce after the initial recording complete. However, it is doubtful that audio reproduction can make full use of the dynamic range that even a 20-bit processor suggests: The threshold of hearing isn't decreasing; in fact, it tends to rise as we age. According to the Occupational Safety and Health Administration, the ill effects of prolonged exposure to loud sounds commence at about 90 dB sound-pressure level (**Table B**, **Reference A**).

The "faster-samples-are-better-samples" camp points out, for example, that converters that operate at faster sample rates rely less on interpolation and decimation filters. Converter manufacturers, however, do not separately specify on-chip digital-processing blocks, such as interpolators, so you would be hard-pressed to compare the effects of design choices. Instead, converters are properly specified for their "pin performance," which unfortunately does not entirely capture a device's audible traits (**Reference B**).

TABLE A—EXAMPLES OF SOUND INTENSITY

Sound source	Intensity (dB SPL)
Rocket launching	180
Jet engine at 3m	140
Thunderclap, air-raid siren at 1m	130
Threshold of pain	125
Jet takeoff at 60m, rock concert	120
Discotheque, accelerating motorcycle at 5m	110
Firecrackers, subway train, jackhammer at 2m	100
Heavy truck at 15m, city traffic, noisy factory	90
Alarm clock at 1m, hair dryer, vacuum cleaner	80
Noisy restaurant, business office	70
Conversational speech	60
Light traffic at 50m, average home, quiet restaurant	50
Living room, quiet office, residential area at night	40
Library, empty movie theater, soft whisper at 5m	30
Recording-studio ambient noise	20
Human breathing at 3m	10
Threshold of hearing	0

TABLE B—MAXIMUM EXPOSURE TO SOUND

Duration per day (hours)	Maximum noise level (dB SPL)
Eight	90
Six	92
Four	95
Three	97
Two	100
1.5	102
1	105
0.5	110
0.25 or less	115

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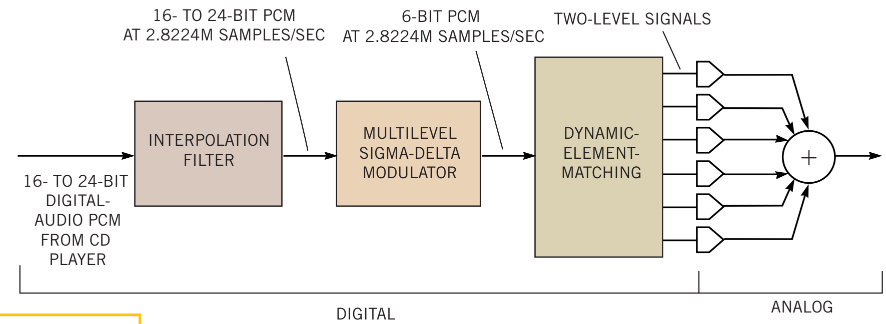

Figure 2 DACs that exploit multibit delta-sigma modulators use dynamic-element matching to reduce their dependence on device matching within the IC (courtesy Wolfson Microelectronics).

plane, the LM4930 audio subsystem stands only 0.6 mm high.

The \$3.95 (1000) IC operates with a 2.6 to 4.5V digital supply and a 2.6 to 5.5V analog supply. National Semiconductor specifies performance at two supply states—3.3V digital and 5V analog and 3V on both supplies. The LM4930's output span is 2.5V p-p, which corresponds to maximums of about 25 and 390 mW for 32Ω and bridged 8Ω loads, respectively. With 3V supplies, the stereo-headphone amplifier develops at least 15 mW before reaching the 0.5% THD+N (THD-plus-noise) mark at 1 kHz. For the monophonic-speaker amplifier, the numbers are 270 mW minimum and 2% THD+N under the same conditions as the headphone amplifier. Typical SNR values are 72 and 86 dB, respectively, for the mono-voice and stereo-music paths.

MAKE YOURSELF AT HOME

Digital-audio converters for home-entertainment equipment also demand high levels of integration. The voice codecs, battery monitors, and on-chip speaker amplifiers have gone, replaced by larger channel counts to accommodate surround-sound encoded audio, higher sample rates, and more sophisticated digital signal-processing capabilities.

The little ear-bud headsets and 1-in. speakers have also disappeared. The trend in this segment includes a variety of acoustic drivers from full-range speakers that can expose any warts in the signal-processing chain to single limited-range speakers that take advantage of signal-processing techniques to sound larger than life-size. So, although ICs for consumer applications compete primarily on price, power dissipation, size, and features, IC makers know that they needn't pitch their parts far along an OEM's product line before performance issues start tipping the balance for or against acceptance. As one IC product-line manager says, "Performance isn't enough to get a part on the print, but it's enough to drop one off."

Home-entertainment equipment is anything but a homogeneous category, particularly when you consider signal sources. Common digital-audio sources range from 16-bit, 44.1k-sample/sec red-book-standard CDs to 24-bit, 96k-sample/sec DVDs. Analog inputs include legacy sources, such as terrestrial broadcasting and Philips cassette players. They also include analog-derived-from-digital signals from set-top boxes, VCRs, TiVos, and CD and DVD players that lack digital outputs.

Despite the usual model turnover in the home-entertainment-equipment market, the technical requirements for digital-audio converters have recently changed little, and, to the extent that they have evolved, they've done so in fairly predictable ways. This trend has extended the useful life of many audio-codec designs. Your choices now reasonably include not only a vendor's latest and greatest, but also devices introduced during the last couple of years, particularly in cases in which the vendor maintains pin compatibility within an evolving product line. Thus, it is typical to see product revisions as IC makers invest further in their most successful digital-audio parts with design optimizations and process ports.

Home entertainment near the commodity end of the market has attracted differing approaches to digital-audio-subsystem design. Two products—from Analog Devices and Cirrus Logic—that occupy opposite ends of the feature scale exemplify these approaches. Despite their striking differences, both chips depend on a high degree of integration.

Analog Devices' AD1954 signal-processing DAC perhaps offers an extreme example of a digital-audio converter with on-chip signal-processing horsepower. At the heart of the 1954 is a DSP core designed from the ground up for audio signal processing functions. The DAC's front end multiplexes three serial-data and three clock sources. The DSP core provides extensive filtering, level detection, and dynamics control and feeds three DACs for 2.0 or 2.1 stereo systems. On-chip program and parameter memory allows you to configure the DSP on the fly, and the associated GUI-based software provides a ready means of configuring the chip's signal-processing flow and fix each function's parameter set (**Figure 1**). The GUI creates a file that you load into the DAC's program memory. You can also develop and load multiple parameter-RAM settings to facilitate various operating modes within your application.

Beyond the usual tone and volume controls, you can use the 1954's signal-processing capabilities to implement equalizers to compensate for speaker-cabinet or room deficiencies. You can invoke signal delays as long as 6 msec to adjust for suboptimal speaker locations. You can also broaden a stereo image in applications that fix the two speakers too close together, as is often the case in compact stereos, television sets, and video monitors.

The main channel's differential outputs minimally measure 108-dB A-weighted SNR; the subwoofer channel is worse by only 4 dB. The \$5.88 (10,000) DAC's THD is −03 dB for the main channels and −90 dB for the subwoofer. Outputs are within millivolts of 2.75V p-p on a 2.5V common-mode level. Analog Devices provides the AD1954 in MQFP-44 and TQFP-48 packages.

A high degree of integration does not always imply a feature-packed IC. It can also suggest a device that provides a basic function in an aggressively small package. Cirrus Logic makes this point with its CS434x family of 24-bit, 192-kHz stereo DACs, which fit into a TSSOP-10 package for so-called value-line DVD-based entertainment equipment, digital-television sets, and set-top boxes—devices that either require no volume and tone controls or implement them in the analog domain. The individual converters in this four-member family differ only in the data-interface format they support. A couple of bits in a control register usually manage this distinction. But the CS4344, 45, 46, and 48 have no pins for a control port. The DACs automatically detect audio sample rates of 2 kHz to 200k samples/sec—meaning, for practical purposes, the standard rates of 32 to 192k samples/sec—and accommodate resolutions of 16 to 24 bits.

The 434x DACs operate on single supplies. Cirrus specifies their performance at 5 and 3.3V with a 6-dB dynamic-range advantage for the 5V operation at resolutions greater than 16 bits. If you factor in the DACs' THD performance, the difference disappears. The minimum A-weighted SNR is 90 and 99 dB, respectively, for 16-bit and greater-than-16-bit resolutions at 5V and 90 and 93 dB at 3.3V. The maximum THD+N measures −87 dB for 16-bit audio and −89 dB for 18- through 24-bit words. The $1.20 (10,000) converters operate from −10 to +70°C. The CS4344, which takes I²S data, is also available for operation at −40 to +85°C.

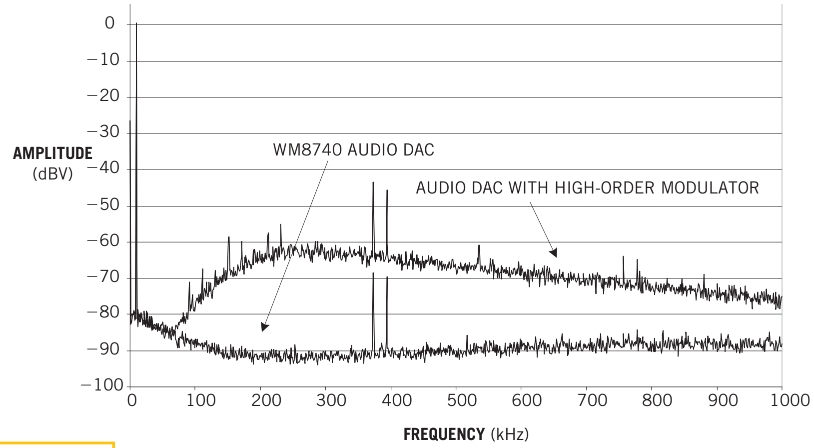

Figure 3 A DEM-based structure with unequally weighted elements reduces out-of-band noise in the Wolfson WM8740 DAC.

PUTTIN' ON THE BITZ

As you move your focus to upper-end consumer and professional equipment, market requirements begin to blur. Though both segments invite a healthy level of price competition, end users are willing to pay for a perceived performance advantage. Both have set 24-bit, 192k-sample/sec capability as a basic requirement. Though there are still applications for codecs in the consumer market, the professional segment with rare exception uses separate DACs and ADCs. Applications for this type of device include digital-audio consoles, workstations, and effects processors; broadcast equipment; and high-end audio/video receivers and surround-sound processors.

To the extent that IC makers continue to squeeze the specs for these applications, circuit-design innovation continues to focus on raw performance as a primary goal at the expense of functional density. For the professional end of the market, which is least price-sensitive, this trait may manifest itself in wide internal datapaths to prevent round-off errors within the digital filters or in highly linear converter stages.

During the course of a chip development, IC designers are concerned about linearity beyond the audio band. One of the advantages of the delta-sigma converter architecture is that it shapes its noise away from the base band into the region beyond the Nyquist frequency, at which, for the purposes of the audio experience, listeners care little. But nonlinearities in the supersonic spectral region can cause noise energy to fold back into the audio band. Subjectively, this complex behavior reduces acoustic transparency. Analytically, normal THD specs with 1-kHz excitation do not reveal this performance trait. The best datasheet evidence is found in the characteristic curves, but beware those that indicate steep declines in the upper audio range; they likely employ filters external to the converter, which, though perhaps justifiable on other grounds, can hide more than they reveal about this issue.

FOR MORE INFORMATION...

For more information on products such as those discussed in this article, contact the following manufacturers, and please let them know you read about their products in *EDN*.

AKM
www.akm.com

Analog Devices
www.analog.com

Cirrus Logic
www.cirrus.com

National Semiconductor
www.national.com

Texas Instruments
www.ti.com

Wolfson Microelectronics
www.wolfsonmicro.com

Virtually all converters in this class exploit multibit delta-sigma architectures (**Figure 2**). This concept has filtered down to some consumer-market devices as well, but not to the level of sophistication you find in the market's upper strata. As in the case of a single-bit DAC, a multibit DAC starts with an interpolation filter, which increases the effective sampling rate and suppresses out-of-band signal images. A DEM (dynamic-element-matching) array decomposes the signal from the multibit delta-sigma modulator into a number of individually delta-sigma-modulated signals (**Reference 5**). The DEM system determines how the individual equal-weight signals recombine, usually by selecting the least recently used elements. This approach reduces the extent to which individual DAC elements must match for a given performance (**Reference 6**).

Examples of high-performance audio converters include the four-channel PCM4104 DAC from the Burr-Brown group at Texas Instruments. The 4104 accepts 16-, 18-, 20-, and 24-bit audio data at rates as high as 200k samples/sec in the usual formats—left-justified, right-justified, and I^2S—and as a TDM (time-division-multiplexed) stream for simple interfacing to a DSP's serial port.

The quad converter can operate as a stand-alone device, or you can configure it under software control through a four-wire SPI. The software controls give you access to features such as individual channel mutes and output-phase reversal that digital mixing consoles require.

The DACs provide 5Ω differential outputs with 6.15V p-p spans that can drive 600Ω loads. TI limits the THD at 48k samples/sec to −94 dB with the usual 1k-sample/sec excitation at 0 dB full scale. The company extends the measurement bandwidth to 40 kHz for THD measurements at 96 and 192k samples/sec but provides typical values of only −100 and −97 dB, respectively. The typical unweighted SNR is 113 dB or better at all sample rates, but TI hasn't published SNR-limit specs for the $7.50 IC. The PCM4104 comes in a TQFP-48, operates on 5V analog and 3.3V digital supplies, and dissipates only 256 mW at 48k samples/sec and incrementally more at higher sample rates.

Cirrus Logic's CS4398 stereo DAC processes 16- to 24-bit digital audio in the usual PCM formats and supports data rates to 192k samples/sec. It also includes a DSD bit-stream input. Both PCM and DSD inputs includes logarithmic volume controls that you access through the control port. You can change individual channel volumes and mute states immediately, upon a zero-crossing, at a fixed ramp rate or at a fixed 1/8-dB/zero-crossing ramp rate. In addition to the software-controlled volume and mute mechanism, you can independently mute each channel under hardware control for applications that operate the converter as a freestanding device.

The $4.32 (10,000) CS4398's dynamic range is 114 dB A-weighted and 111 dB unweighted with 24-bit PCM data and 3 dB less operating from the DSD input. THD is limited to −99 dB with either 24-bit PCM or DSD inputs. The device has 50Ω differential outputs, which can drive 1-kΩ loads. Dissipation is 340 mW when the chip operates at 5V, dropping to 240 mW if you power the digital section from a 3.3V supply and 1 mW in its power-down mode.

Cirrus packages the 4398 in a TSSOP-28. Its companion ADC, the CS5381, has a fifth-order delta-sigma modulator and is available in TSSOP-24 and SOIC-24 packages for $14.95 (1000).

AKM offers a 24-bit, 192k-sample/sec delta-sigma ADC with differential inputs and a linear-phase decimation filter for high-end audio/video receivers, home-video recorders, and musical instruments. You can operate the AK5385 at rates ranging from 8 to 216k samples/sec. The architecture includes a highpass filter in the digital section that strips off input offsets and ensures a zero-centered data stream.

SNR plus distortion minimums are −92 dB at −1 dB full scale in a 20-kHz bandwidth at 48k samples/sec and −90 dB in a 40-kHz bandwidth at 96k samples/sec. Minimum A-weighted SNR measures 107 dB. AKM packages the $5 (5000) stereo ADC in VSOP-28.

Wolfson's WM8740 embodies an interesting variation on the usual multibit delta-sigma DAC architecture. A DEM block with 14 nonequally weighted elements decomposes 6-bit PCM words in several cascaded stages. The 8740's DEM array includes two unity elements and two 2×, two 4×, and eight 8× elements. Individual elements combine in like or dissimilar polarity based on the LSB at each stage. After acting on the LSB, each stage passes on the remainder to the stage that follows. At the end of the cascade, the final remainder feeds a nonweighted DEM block. The 8740's DEM blocks use random-number sequences in the element-selection process to prevent the occurrence of tones on the output that repetitive element-access patterns can generate in the presence of small non-idealities in individual elements.

The result of Wolfson's unequally weighted DEM block is a substantial reduction of out-of-band noise compared with conventional DACs based on multibit delta-sigma modulators (**Figure 3**). The DAC's in-band SNR is 110 dB in 20 kHz at 48k samples/sec. THD is −95 dB or better.

The $4.56 (1000) stereo DAC can operate either in a stand-alone hardware mode or under software control through its four-wire SPI interface. Hardware control pins allow you to configure the device for audio-data format, de-emphasis, and mute state. Software controls add attenuation in 256 0.5-dB steps and phase inversion. You can also configure the WM8740 as a single-channel device. In this differential-mono mode, the two outputs operate in counter phase. When you appropriately combine the two channels, they produce a monophonic channel with a 3-dB improvement in the typical SNR performance. In such a configuration, you would connect two of the SSOP-28 devices with their clock and audio-data inputs in common to implement a stereo pair.

Wolfson also reports new work in the area of digital-filter design and claims benefits to implementations that, strictly speaking, do not follow a linear phase response. Though many manufacturers in some manner characterize their digital filter's amplitude performance, few provide details with regard to phase response. Though IC designers acknowledge that a converter's phase response has audible consequences, no useful correlation between objective and subjective observations has yet to gain acceptance in the industry.□

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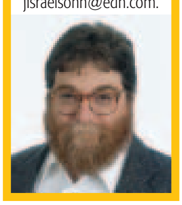

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