Sounding Off On Semiconductors

by Frank Wells

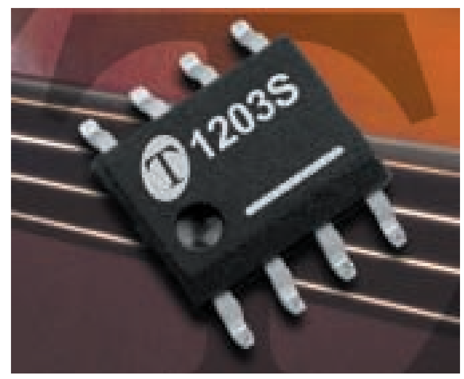
Without their semiconductor building blocks, modern audio electronics could not exist. For this special report, *Pro Sound News* talks with pro audio design engineers about the state of their art, and the chips of their trade.

In analog design, one school maintains a loyalty to discrete design, others to tried-and-true op-amp-based implementation, while others have embraced new building blocks that incorporate substantial portions of circuits within a single chip. Design consultant Marc Lindahl, owner of Bowery Engineering Associates (http://engineering.bowery.com), spoke highly of one such building block, THAT Corporation's (www.thatcorp.com) 4320, a "complete dynamic processing system with really great performance."

Lindahl also says THAT also has "an improved version of the line receiver" he's been employing. Other modular parts finding their way into signal paths include the SSM balanced driver/receivers, the SSM2019 and TI INA217 mic pres, and THAT's 2159 VCA. A low parts count and ease of troubleshooting are important reasons for selecting modular mic pres, according to Echo Digital Audio's (www.echoaudio.com) chief technology officer, Milo Street.

While relying mostly on previously implemented designs, "in order to improve purchasing power across our entire line of products," Alan T. Meyer, director of engineering for Alesis Studio Electronics (www.alesis.com), says that at Alesis, "we try and stay in touch with the major IC manufacturers in case new, low-cost solutions arise." The path of experimentation is followed by most designers, such as Michal Jurewicz, president of Mytek Digital (www.mytekdigital.com). "We are always on this quest" he says, "testing new chips in both the analog and digital domain. This coupled with new circuit solutions allows us to stay within our credo: 'Our new product must sound at least a notch better than the last generation.'"

The increase in digital processing means less analog circuitry is required in many designs. "We're using fewer and fewer analog components," says Tony Agnello, chief technical officer at Eventide (www.eventide.com) and chairman and president of Manifold Labs (www.manifoldlabs.com) and Princeton Digital (www.princetondigital.com), respectively. "The higher levels of integration suck so much functionality into a single chip that design has become a matter of connecting blocks. A key milestone happened nearly 20 years ago when the first sampling A-to-D converters were introduced, eliminating the need for complex analog anti-aliasing filters and other analog signal conditioning. The trend has continued." Agnello's cohort, Don Elwell, president of Manifold Labs, adds, "We mainly use balanced drivers and receivers. Our analog blocks for those interfaces are proven and thus are pretty cookie-cutter now and for us, the analog designs pretty much end at the converters."



Nathan O'Neill, chief technology officer at Symetrix (www.symetrixaudio.com), comments, "Our analog audio stages are still predominantly the same: using op-amps for front-end line level inputs, instrumentation amps for mic pre/line inputs (together with low-resistance switches), and digitally controlled gain/attenuation ICs for fine control of input level. We are also using a lot more switching regulators than in the past, preferring them to linear regulators because of efficiency, and also because we have improved methods of filtering any noise generated from such regulators." Among the new chips finding their way into contemporary designs are the Analog Devices (www.analog.com) ADG452 digitally controllable analog switch and the Texas Instruments (TI; www.ti.com) PGA2500 digitally controlled analog mic preamp. B.J. Buchalter, VP R&D at Metric Halo, (www.mhlabs.com), says, "The Analog Devices ADG452 analog switch has a really low Rds-On and full analog supply rails. As a result, it really enables excellent low-noise, low-distortion, digitally controlled analog solutions."

"The only truly modular chip we use is the Cirrus (www.cirrus.com) CS3310 [stereo digital volume control]," says O'Neill, "which TI have a pin-compatible cross, the PGA2311. We also use the +/-15V version, the TI PGA2310, which is often a better fit for our high-end designs. Mic preamps use the TI INA103 or INA162 instrumentation amp, in conjunction with some low-on-resistance switches allowing varying coarse gain settings." But O'Neill qualifies that while "the modular analog audio products offer ease of design for novice engineers, in general we haven't seen them as being price-competitive with roll-your-own designs."

Designers using very traditional approaches to circuit design, such as Kevin Burgin, design team member at Rupert Neve Designs (www.rupertneve.com), eschew many of the IC-based solutions to analog design. "We are always searching for new components," explains Burgin. "However, we frequently run into limitations set by the new components. Switching frequencies below 200 kHz interfere with our audio bandwidth. Rail voltages of +5 or +/- 3 volts (and getting smaller every day!) are of little use to us as we are looking for maximum headroom, and most analog ICs are 36 volts average maximum. Noise figures and distortion are of importance, also." Buchalter somewhat echoes that view as it relates to many modular components. "We have looked at a number of these integrated solutions for mic pres and differential driver/receivers, but we have not chosen to use them because of either sound quality (e.g., they don't match the 'sound' of our discrete solutions), or due to supply rail limitations that make them unsuitable for integration for our purposes." Jurewicz adds that "we believe that our (semi-discrete) approach is far more advanced than function chips offered when it comes to the main characteristic of our products: transparent and detailed high-end sound." Agreeing with that sentiment was John Hanson, VP of engineering and product development, MI group, Harman Music Group (www.harmanmusicgroup.com). "Most of the time, we are able to achieve as good or better results with our discrete designs," says Hanson, "and many times the discrete designs are less expensive."

Designing analog circuits and digital circuits require two different mindsets, says Meyer. "It's the difference between art (analog) and logic (digital). From my experience, a good analog design is completed by an engineer with an artistic mind, while a good digital design is completed by a system level designer with superb high-level vision." Jurewicz comments that "digital allows for easier integration of numerous features at no added production cost."

"It's much harder to simulate an analog design," adds Lindahl, "so you end up with more trial and error to get it right. The approach is more simulation- and modeling-oriented when you're working in the digital domain, just because you can, and it's a lot easier. Either way, if you don't use your ears, and if you're not willing to work on something until it sounds right, then, well, as they say, 'GIGO.'" Jurewicz says that Mytek also focuses on sound quality regardless of the domain. "In our approach, the focus is on sound quality, and the approach to design for that is similar in both analog and digital parts of projects."

Analog design can be more finicky, says Agnello. "Cross-talk and noise coupling are issues, but the primary issue is correctly handling grounding. The designer must minimize digital clock noise injected into the analog signal path." Elwell has a one-word description of analog design: "Magic! Seriously, the really good analog designers I've known have a knack for being able to use esoteric characteristics or second-order effects of devices to their advantage." O'Neill relates some additional issues: "In analog design, one must be sensitive to gain-staging, distortion, oscillation and noise floors, especially in the front-end stage of an ADC design. Also, PCB layout tends to be more critical with analog designs than digital designs, especially when you have both on the same circuit board."

Conversion between analog and digital has been greatly simplified by modern components, with even inexpensive modern devices employing circuits that outperform the conversion in many, if not most, early digital audio devices. Does our panel agree that good converter performance is now easy and inexpensive? "I think we've seen more that a certain performance level getting less expensive, rather than a significant performance improvement. Of course, compared to 10 or 15 years ago, sure, it's much, much easier to get excellent performance. But 'ultimate performance' still takes some work."

Jurewicz agrees: "In mid-quality analog/digital equipment, decent performance can be achieved 'by the book' (i.e., easily) and at moderate cost. This is reflected in price of equipment, which has plummeted threefold since the mid-'90s. However, top-notch high-end design still requires special selection of parts and a lot of design experience, and by no means can be described as easy."

Meyer says that while it is "easier for entry-level companies to produce converters with high-performance," cost is still about the same for the best-performing circuits, with a definite improvement in performance overall. O'Neill agrees, explaining that "the market expectation has increased in such a way that our designs still cost about the same to make, but now offer greatly improved performance."

"However," Meyer adds, "excellent measured performance doesn't always mean excellent-sounding performance. All the best converters measure well, so the true test that sets one converter apart from another is purely subjective."

Converter parts from AKM (www.akm.com) and Cirrus Logic topped those cited as currently employed by our panel. Meyer adds that for applications limited to 48 kHz or less, Wavefront Semiconductor (www.wavefront.com) parts are the primary choice at Alesis. This prompts an aside, based on Alesis' use of these converters and Wavefront's new DICE II digital I/O—does Alesis enjoy "special" benefits due to Alesis and

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Wavefront belonging to the Jack O'Donnell family of companies? Meyer responds that "Wavefront is now its own entity that must show a profit for each company it deals with, including Alesis. The only way this is possible is to charge Alesis the same price for parts as everyone else. However, there are some real benefits that might be missed by the casual observer. ICs can take years to develop, so Alesis helps guide the direction of new chip development. Likewise, Wavefront Semiconductor has the inside track as to what chips will be successful without having to spend excess time and money performing market research."

While converter selection is undoubtedly important for sonic character and for specs like latency, Buchalter cautions that "it is the complete conversion system that is important, not just the converter chip. The same chip can be used in a different system with an incredible difference in the quality of the overall system." The analog interface to the converter, clock and jitter management, power supply design and signal conditioning are all cited as being as important as the actual converter chip. Buchalter also reminds that while consumer devices use many D/A chips, helping drive the price down with volume, "On the A/D side of things, this does not hold true."

The latest rage in digital I/O and control communication is the use of conventional computer industry protocols, like USB, Ethernet and FireWire, for audio interface with DAWs, an area of some experience for our panelists. "Plugzilla uses USB for communicating to both the front and real panel subassemblies," Lindahl says, "Also, I'm working on some FireWire development boards, which I will make available to other audio companies for experimentation, development, etc." Lindahl plans Wavefront and Yamaha (www.yamaha.com) mLAN versions of the boards, and with other chipsets possible for future versions.

The Harman Music Group's products include USB and Ethernet capabilities, while Alesis released its first such USB and FireWire equipped products at the recent NAMM Convention. Buchalter reminds that Metric Halo was "one of the first companies to ship FireWire audio (almost four years ago). All of our digital audio hardware products are built upon our FireWire audio implementation." Symetrix implements CobraNet over Ethernet and, for its Lucid line, are also exploring FireWire, as are Mytek and Echo Digital.

For USB, the Cypress EZ-USB and EZ-USB-FX family of devices, Texas Instruments (TAS1020B) and Burr-Brown (PCM2902), and the Anchorchip USB microprocessor have been employed by our surveyed designers. Cirrus makes the CS18101x family of CobraNet chips, employed by Symetrix, who, O'Neill adds, "would like to see a gigabit implementation of CobraNet offering more channels and lower latency, as well as a very low-cost solution for 2-channel designs."

For FireWire, there's the Bridgeco interface, the new DICE II from Wavefront Semiconductor (which also supports most of the professional audio interface standards and includes two cross-point routers with independent clocking and jitter elimination) and Yamaha's mLAN chipset. Echo Digital employs TI's TSB43CB43 coupled with its TMS320C6713 DSP for FireWire, and Metric Halo rolled its own solution early on with TI Link and PHY and the Analog Devices SHARC DSP to handle the FireWire audio layer.

The new generation of interfaces are gladly received by designers, particularly, as Jurewicz says, "as the R&D required for such custom solutions is prohibitive for small audio companies." Street reminds that "The complexity of interface protocols such as FireWire and USB requires much more than just silicon. There is quite a bit of software and firmware support that must either be provided by the vendor or developed in-house."

FireWire seems to be generating the most excitement at present. Richard Foss, managing director of Networked Audio Solutions (www.networkedaudiosolutions.com), comments, "We are making extensive use of designs with FireWire digital I/O capabilities. The DICE II chip is a good example of such a chip, enabling the flexible encapsulation of audio samples into FireWire packets, and the controlled extraction of these samples. Our task is typically to write firmware that utilizes the capabilities of a chip such as the DICE II, to provide audio and MIDI connectivity between hosting devices. We believe that FireWire, with its supporting chips, has the potential to radically simplify and enhance the management of connections and control in the field of pro audio."

Meyer summarizes much of the enthusiasm for the DICE II, calling it "an extremely capable and affordable IC for products that require streaming audio over FireWire incorporating a medium-to-high number of channels, high sample rates (up to 192 kHz), with minimized barriers to entry." Lindahl adds praise for the DICE II tool set. "They use a GNU tool chain," he says, "which means it's all open source, debugged code, and a well known interface, debugger, etc. In general, I would like to see more vendors support open IDE platforms like Eclipse, for better tool-chain integration."

Both Meyer and O'Neill expressed a desire for an even lower cost, simplified version of the DICE II for low channel count applications where less versatility in other digital audio interfaces are required. The TCAT (TC Applied Technology) division of TC Electronic (www.tcelectronic.com), which developed the DICE II before turning it over to Wavefront for manufacture and marketing, has told *Pro Sound News* that it is working on such future implementations.

Once in the digital domain, the DSP engines chosen for use vary by designer, from the custom parts used in some applications by the Harman Music Group to off-the-shelf solutions. "I had used the Motorola 56301 in the Sonorus Studi/o, says Lindahl, "since it had a built-in PCI interface and plenty of horsepower. In the Powercore PCI, I used 56362s because they are cost-effective, and TC wanted to use the vast 56K DSP code base they had developed for their hardware boxes. Then again, on that board, I used an embedded PowerPC because of its powerful floating-point ability. So there, I used a RISC microprocessor for (simple) DSP. The big advantage for Motorola is the existing base of algorithms already coded." Agnello and Elwell agree that, simply based on its catalog of code, Motorola is their typical choice.

At Alesis, Meyer says the company turns to Wavefront for DSP for most products, while for large-scale DSP, "we stick with the major players, including Texas Instruments and Analog Devices. The big advantage of TI right now is its highly parallel architecture, coupled with a high-clock rate, memory interface and floating-point math. The TI really shines for intensive DSP operation."

Buchalter says that at Metric Halo, "Our key considerations have been the ease of integrating the DSP into the types of systems we want to build, and the ease of programming the DSP once it is integrated." After much evaluation,

AD SHARCs were chosen based on programming ease, predictable performance from a performance-oriented architecture and the ability to function as a microcontroller as well as processor. Buchalter also praises the company's vision for future products and features.

O'Neill says Symetrix also uses SHARCs and has "made a slow migration from 24-bit, fixed-point processors to 40-bit floating point processors over the last few years. Our real-world implementations, and tests of similar algorithms on different processors, show it's far easier to design with the actual single-precision, 40-bit floating-point processing available than tying up your processor with double-precision math on a 24-bit, fixed-point DSP. That's why, adds Lindahl, "anything with floating point is, of course, much easier to deal with from the algorithm point of view. That's why people are starting to use microprocessors like PowerPCs and Pentium-class processors for DSP."

Agnello agrees that "Running DSPs on host CPUs (Pentium, PowerPC) is now practical for the majority of audio apps, and it's happening everywhere (plug-ins, Garageband, etc)." Harman's Hanson adds, "Off-the-shelf DSPs continue to get more powerful and more affordable. Because of this, future products will be able to achieve better sound at lower prices."

While the audio IC market offers more solutions than ever, often with a high degree of integrated functionality, the parts and their applications become ever increasingly sophisticated. As Agnello says, "ICs have become systems and designing circuits has become systems design."

The full (and exhaustive) transcript of replies from the participating design engineers can be found at: www.prosoundnews.com/articles/Semiconductors05.shtml.

Device Drivers—A Critical Bridge

When interfacing audio gear with computer hardware, software device drivers can be as critical to performance as the devices they connect. Devendra Parakh, VP of engineering for software developer Singing Electrons (www.singingelectrons.com), created the first completely full duplex USB 1 driver, worked with Yamaha on mLAN drivers, has done drivers for the BridgeCo 1394 implementation, and is currently leading the driver development effort for the Dice II. He comments on the changes he has seen over the last six to eight years:

From the perspective of device drivers, a lot has changed from USB 1.1 audio to FireWire audio. When audio was introduced with USB, the goal was not professional audio. It was targeted for consumer audio, for streaming applications such as media players, DVD players, gaming and likes of such. And as such, the driver model that Microsoft came up with had little or no concern for things like latencies, synchronization, and external clock sources.

That is where Steinberg's ASIO driver model came into the picture and became a de facto standard for DAW use. ASIO was designed with professional audio applications in mind, and so it provided for determinate (and low) latencies, synchronization, and external clock sources, etc. And ASIO was (is!) available on both Windows and Mac (OS9) platforms.

On the Mac side, Apple introduced CoreAudio with OS X. CoreAudio overcomes most of the limitations of older audio driver models, and is suitable for both consumer and professional audio use. On the Windows side, RTAudio (to be introduced with Longhorn) promises to overcome the current limitations of WDM Audio architecture, and make it useful for professional audio use. Similarly, hardware manufacturers have improved a number of things over time.

With USB 1.1 silicon, there were two big concerns—one was to be able to implement isochronous transfers with low overhead (some applications did it with DMAs, some did it by using fast micro-controllers), while the other concern was to avoid clock drift. Most applications used some sort of a PLL or did SRC on the device side. Either technique was expensive in terms of hardware or firmware resources. Very few USB audio devices implemented the technique of slaving the host PC to the device's clock. In fact, most USB audio hardware designers thought (and I've worked with many that still think so!) that it was impossible to do so. We have implemented it successfully in three different devices—one for SMSC and two for Analog Devices.

With FireWire audio, things are quite different. The 61883-6 has been designed from ground up for professional audio use, and it provides for having very accurate presentation times (although the implementation is up to the silicon/device manufacturer).

With the drivers for FireWire audio devices, designers usually have to do more work than they had to for USB audio devices. With USB Audio, it's possible for drivers to simply pass down the audio sample data to the USB driver stack without much manipulation. However, with FireWire audio, the drivers have to repackage the samples, as well as add labels to each and every sample according to the audio format.

At the same time, with proper drivers, it's possible to have multiple FireWire devices connected to the PC and a single application sending and receiving audio from multiple devices, without worrying about clock drift, phase shift, or other timing related issues that crop up when using multiple devices. This opens a lot of possibilities that are not otherwise possible with other kinds of devices.