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Is SACD Doomed?

By Gary Galo, Regular Contributor

New York AES 2007: A Convention Notable for What Wasn't There

The Audio Engineering Society held its annual US convention at the Javits Convention Center in New York City over the long Columbus Day weekend, Oct. 5-8, 2007. Unlike many previous AES conventions, this one didn't promote any particular theme, with the program cover simply noting "Where Audio Comes Alive." As usual, the program was huge and diverse, way too much for one person to canvas thoroughly. Like most attendees, I focused on presentations and exhibits of particular interest to me and, I hope, readers of this magazine.

SACD SUPPORT?

In the September 1999 version of the New York AES, Sony introduced the Super Audio Compact Disc (SACD) with considerable fanfare. A special five-channel demonstration room was fitted with Sony's top-of-the-line SACD player, five Sony SS-M9ED loudspeakers (at 16 grand a pair), each powered by a Pass X600 class-A mono amplifier. In my guest editorial, "The High End at AES," I declared the demonstration to be ". . . without a doubt, the finest sound I have ever heard at an AES convention" (*Audio Electronics*, 2/2000, p. 8).

Sony also gave a special presentation in a large room filled with several hundred attendees. Panelists at this presentation included the well-respected recording engineer Tom Jung, President of DMP Records, who offered a most enthusiastic endorsement of the SACD and its Direct Stream Digital recording system. Jung noted that, in his opinion, the SACD system yields recordings closer to the direct feed from the mixing console than anything he had previously heard. At three subsequent New York AES conventions—2001, 2003, and 2005—Sony's flagship SACD players were featured prominently in their exhibit.

The October 2007 AES convention was notable for what *wasn't* there. Namely, any trace of Direct Stream Digital or the SACD in the Sony exhibit area (Sony didn't even have a demonstration room this year). Indeed, it was difficult to find anyone promoting high-resolution digital audio. I asked a Sony representative whether the company had given up on SACD. He said that while Sony had not dropped its support for the format, there has been no clear support in the industry for a replacement for the conventional CD. Sadly, the Sony rep is correct.

The industry has simply followed the demands of the marketplace and, with most music lovers delighted with the sound of Apple's iPod and other compressed digital formats, interest in DVD-Audio and SACD is confined to a fringe corner of the audio marketplace. Granted, the iPod doesn't need to contain compressed digital audio, but that's how the vast majority of Apple's enthusiastic customers use it. Indeed, papers on various coding schemes for Internet and portable audio occupy a significant place on AES programs.

The rejection of high-resolution digital audio by the mainstream audio community is due to one of two things. Either most people who listen to music can't hear the difference between the CD and high-resolution digital, or the difference is not significant enough to matter. Convenience is often more important than sound quality, which certainly accounts for the success of various portable music formats, from the analog cassette to MP3 players and the iPod ("I have 12,000 tunes on my iPod—why should I buy SACDs?").

The inability of many to perceive the differences between the CD and high resolution formats was documented in an article recently published in the *AES Journal*, "Audibility of a CD-Standard A/D/A Loop Inserted into a High-Resolution Audio Playback."[1] Authors E. Brad Meyer and David R. Moran, both members of the Boston Audio Society (as well as AES), conducted a series of double-blind listening tests over the course of a year using an A/B/X box. These tests compared the direct output of an SACD player to an A/D/A conversion at the 16-bit/44.1kHz CD standard. In other words, the analog output of the SACD player was fed through the CD-standard conversion, and an A/B/X box was used to switch between the direct output from the SACD player and the output from the CD-standard converters. Out of 554 trials, there were 276 correct answers—no better than "coin flip" results. To those of us who have recorded in both high-res and CD-standard digital, the conclusions seem ridiculous.

Several recording engineers and audio professionals I spoke with found the conclusions baffling—either the participants can't hear, or there's something seriously wrong with the test. I know a number of people who work in the audio field and seem to fit the "can't hear" category. One recently told me that he did not believe that the Sony PCM-F1 digital recorder had any audible shortcomings. In other words, digital audio today doesn't *sound* any better than it did in 1981.

In their conclusion, Meyer and Moran admit that "virtually all" of the SACDs and DVD-Audio recordings they heard sounded better than most CDs. But, they don't believe this can be attributed to the inherent superiority of high-res digital formats. From discussions they've had with engineers who work on commercial releases in high-res formats, they conclude that the extra care taken in the production of SACD and DVD-Audio releases accounts for their superior sound (I'd be willing to bet that much of this "extra care" involves things that A/B/X testing would fail to substantiate).

But, this doesn't account for the proliferation of dual-layer SACDs in which the CD layer is identical in production to the SACD layer, except for a conversion from DSD to 16-bit/44.1kHz PCM. Such SACD discs include the RCA Victor Living Stereo series and the Telarc SACD catalog (the Mercury SACDs are an exception, because the CD layers are Wilma Cozart Fine's older CD transfers, and are thus different from the newly transferred SACD layers).

On my own equipment, the differences between the SACD and CD layers are readily audible. Yet the conclusion of Meyer and Moran is that "Our test results indicate that all of these recordings could be released on conventional CDs with no audible difference." In their view, "Further claims that careful 16/44.1 encoding audibly degrades high resolution signals must be supported by properly controlled double-blind tests."

Apparently they believe that recording engineers such as Tom Jung are simply deluding themselves. Perhaps we also need a series of A/B/X tests to verify whether there are any visible differences between HDTV and NTSC.

DIGITAL DOMAIN

One manufacturer who was demonstrating DSD recordings was ATC Loudspeakers (www.lasvegasproaudio.com). Their SCM150ASL 3-way active loudspeaker was used in a 5-channel surround system with a full-range center channel. Digital hardware included DSD playback equipment from EMM Labs, a company run by renowned digital hardware designer Ed Meitner (www.emmlabs.com). The jazz recordings featured in this demonstration were made on a Sonoma DSD Recording and Editing Workstation, using Meitner-designed DSD converters.

PHOTO 1: Sonoma's DSD-Pure multi-track recording and editing system operates entirely in the DSD domain. Leading recording engineers and producers have proclaimed it the most transparent recording and editing system available. (Courtesy of Super Audio Center, LLC)

Unlike some editing systems used in the production of SACDs, Sonoma's "DSD Pure" system keeps the audio in the DSD domain at all times (**Photo 1**, www.superaudiocenter.com). Although the recordings were impressive enough, I don't think the playback system revealed their full potential.

In Sonoma's product literature, Tom Jung offers the following endorsement: "The Sonoma system, when optically interfaced with Ed Meitner's DSD converters, is the most musically accurate recording and editing system available today at any price." Sonoma boasts endorsements from several other renowned engineers and producers, including Telarc engineer Michael Bishop, who states: "I've recorded to analog tape for well over 30 years and to digital nearly as long. The Sonoma 32-track DSD workstation gives me the quality of analog—without the drawbacks—plus the convenience and speed of a DAW."

Two manufacturers were promoting Digital eXtreme Definition, or DXD, converters. DXD is a 24-bit PCM system operating at a sampling frequency of 352.8kHz, which claims to offer the resolution of DSD with the ease of editing of PCM systems. One of the disadvantages of native DSD editing is the limited signal processing capabilities compared to PCM editing systems. With the DXD system, you can make a recording in DSD format, convert it to DXD for editing on a DXD workstation, and then convert it back to DSD to make the finished SACD master. Or, you can record the original in DXD, edit it, and convert it to DSD for the finished SACD product.

Digital Audio Denmark (DAD, www.digitalaudio.dk) exhibited its AX24 multi-channel converter, noting that "24-bit DXD at 352.8kHz is the perfect work format when producing SACDs (**Photo 2**). It offers the editing and processing advantages known from PCM and converts smoothly to DSD." Digital Audio Denmark further notes that the DXD format is supported by Merging Technology's Pyramix 5.0 workstation.

Merging Technology's exhibit featured the Pyramix workstation (**Photo 3**), now in version 5.1, along with its own Sphynx 2 DXD multi-channel converters (www.merging.com). The

Sphynx 2 was jointly developed with Digital Audio Denmark, and appears to be virtually identical to the DAD AX24. On their website, Merging Technologies makes the following case for the Pyramix system in DSD/SACD production:

“For those who want to work in the DSD domain at 2.82MHz sampling rates for SACD production, Pyramix has a unique answer. . . It is the first system in the world to offer multi-track record/editing and mixing as well as mastering while maintaining all the real-time audio processing (effects) such as EQ, Dynamics, Reverb and Surround Sound in a DSD compatible quality level up to the final SACD master. Quite simply, to have only a mastering capability for DSD is not enough. Unless you can record, edit and process the signals in a true multi-track configuration, mastering makes little sense.

“Pyramix is the first system with enough processing power and resolution to handle in real time a complete digital mixing console capable of operating in 32 bit floating point at 352.8kHz, which is the minimum requirement to preserve all the intrinsic original quality of 1 bit 2.8MHz DSD signals. With real-time EQ, Dynamics, Reverb, etc., and with full SACD scarlet book specification for mastering, we can truly state that ‘Pyramix is the only commercially available system on the market that is capable of a complete multi-track source to master project for Super Audio CD production.’ ”

Regarding the DXD format, they note: “A new format, DXD (Digital eXtreme Definition for high quality and low noise recording and editing for SACD), has recently been acknowledged by Philips and Sony. DXD was initially developed for Merging’s Pyramix DSD workstation and recognized as one of the best formats for DSD source recording.”

The Sphynx 2/AX24 converter was used on an excellent SACD of Mozart Violin Concerti Nos. 3, 4, and 5, performed by Marianne Thorsen with the Trondheimsolistene led by principal cellist Øyvind Gimse, on the Norwegian 2L label (www.2l.no). Digital Audio Denmark handed out a sample copy of this recording containing two discs with identical programs, an SACD and a conventional CD, thus making it easy for the listener to compare the two formats without going into the SACD player’s setup menu. Levels between the two discs appear to have been precisely matched.

To my ears, the SACD is superior to the CD—more spacious, with a silky-smooth treble region that becomes slightly grainy on the CD. The CD also sounds as though it were recorded in a slightly drier venue than the SACD. A single-disc, dual-layer version of this recording is available from a number of Internet dealers, including ArkivMusic (www.arkivmusic.com/classical/album.jsp?album_id=146118). The SACD version makes a strong case for the transparency of the DXD conversion process, though I admittedly have no way of hearing the original, prior to DSD conversion.

The audio community is fortunate that there are still a few manufacturers committed to advancing the state-of-the-art in audio recording. Those of us who care about high-quality sound can only hope that room will remain in the marketplace for such companies. They face an uphill battle against those who claim that it makes no difference, and those who simply don’t care.

Keith O. Johnson, the recording engineer responsible for the material on the Reference Recordings label, gave a master class titled “The Art and Science of Making and Playing Great Recordings—The High Resolution Experience.” Johnson is also the co-designer of the HDCD system, which Reference Recordings is still promoting. They have not issued any SACD or DVD-Audio discs, but their website notes: “We record two-channel masters at 176.4kHz, 24-bits, and discrete five-channel masters

PHOTO 2: Digital Audio Denmark’s state-of-the-art AX24 D/A converter offers 24-bit DXD at 352.8kHz, as well as DSD. (Courtesy of Digital Audio Denmark)

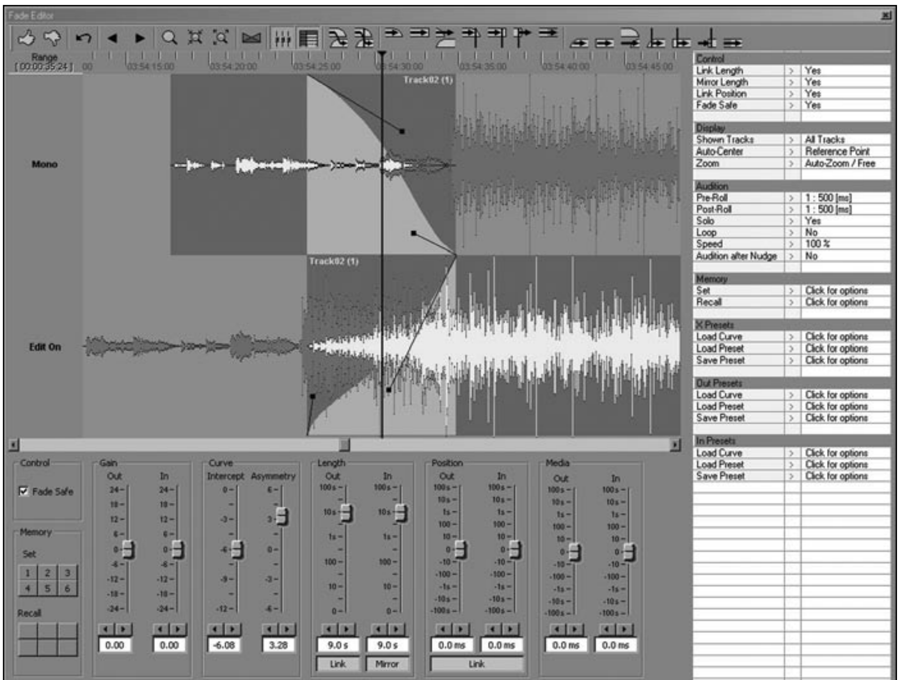

PHOTO 3: The Pyramix editing system operates in 24-bit/382kHz DXD mode, and offers exceptional flexibility and performance. All editing operations, including cross-fades, are 100% real-time—no rendering time is lost. The Pyramix system also includes a multi-track recording system with 32-bit floating/382kHz performance. (Courtesy of Merging Technologies)

at 88.2kHz, 24-bits for future release in high-resolution surround sound" (www.referencerecordings.com).

The large room provided for this 90-minute presentation contained a multichannel surround system with Parasound Halo-series electronics driving PMC BB5 loudspeakers that appeared to be quite capable of providing high-definition sound in a large venue (PMC is a UK-based company specializing in high-resolution professional loudspeakers; www.pmc-speakers.com). Unfortunately, Johnson's presentation was not well timed. He spent too much time at the beginning discussing the hearing mechanism, and left little time at the end for playback of recordings. This was unfortunate because his recording of the Rachmaninoff *Symphonic Dances*—what little of it we heard—was very impressive. His presentation would have been more meaningful if he had interspersed recorded examples throughout his talk, to illustrate the points he was making along the way.

THAT SEMINAR

THAT Corporation gave an excellent seminar titled "Analog Secrets Your Mother Never Told You," a presentation by four representatives of the company, each focusing on a specific product line and application. All of the presentations were excellent, but I was particularly interested in the one given by Gary Hebert, Chief Technology Officer (gkh@thatcorp.com), titled "Balanced Outputs." Hebert described THAT's OutSmarts® 1606 and 1646 balanced line drivers.

In 2000, at the 108th AES convention, Hebert presented the paper "An Improved Balanced, Floating Output Driver IC." As stated in the paper abstract, "The design and implementation of an improved balanced, floating output driver IC for professional audio applications is described. It is shown that when the most common existing designs are used to drive ground-referred loads, the grounded output is forced into current limiting whenever the active output clips. This results in large current spikes flowing into the ground of the receiving device. Techniques used to eliminate this problem as well as the overall performance of the resulting design are described." The OutSmarts devices are based on the concepts described in that paper, and are an improvement over conventional cross-coupled balanced line drivers.

Balanced line drivers are often connected to unbalanced loads, with pin 2 on the XLR connector used as signal, and pin 3—the "negative" audio leg—connected to ground pin 1. Traditional designs can lose control over output current if clipped when one output is grounded. Under such conditions, common-mode feedback is lost. Output current in the grounded leg increases to current limit, which can lead to distorted crosstalk. The OutSmarts common-mode feedback loop maintains control, with no current limiting. The OutSmarts devices are also less sensitive to PCB layouts.

Hebert's presentation focused on measurements that show how the 1606 and 1646 line drivers exhibit much better behavior than their competitors' devices when driven into clipping while feeding single-ended loads. Hebert demonstrated the superior performance of the OutSmarts line drivers using the test setup shown in **Fig. 1**. The test rig allows easy comparison of four balanced line drivers, a cross-coupled circuit using "discrete" 5534 op amps, an Analog Devices' SSM2142, a TI/Burr-Brown DRV134, and a THAT 1646. The SSM2142 and DRV134 are cross-coupled designs in a single, self-contained IC, and have been widely used in the pro-audio industry. All of the devices were driven into clipping into a 10k, single-ended load (one of the outputs is grounded while the other drives the 10k load).

The current sensors in **Fig. 1** monitor the output current of the ± outputs of the line driver, though the current sensor for the ungrounded output was not used for these tests. The test setup has two outputs that can be monitored on an oscilloscope: the voltage output of the active, non-grounded output, and the current output of the grounded output. In **Figs. 2-4** the left trace is the output voltage of the active side and the right is the current output of the grounded side. Scope calibration is 5V/div for the left trace and 20mA/div for the right.

Hebert notes: "The current into ground in the case of the 1646 is an inverted replica of the output voltage (**Fig. 2**). In the case of the other parts (**Figs. 3** and **4**), as soon as the output voltage clips, the output current driven into ground increases to the current limit of the device (50-60mA in the case of these devices). Depending on where the output pin is grounded, these currents may con-

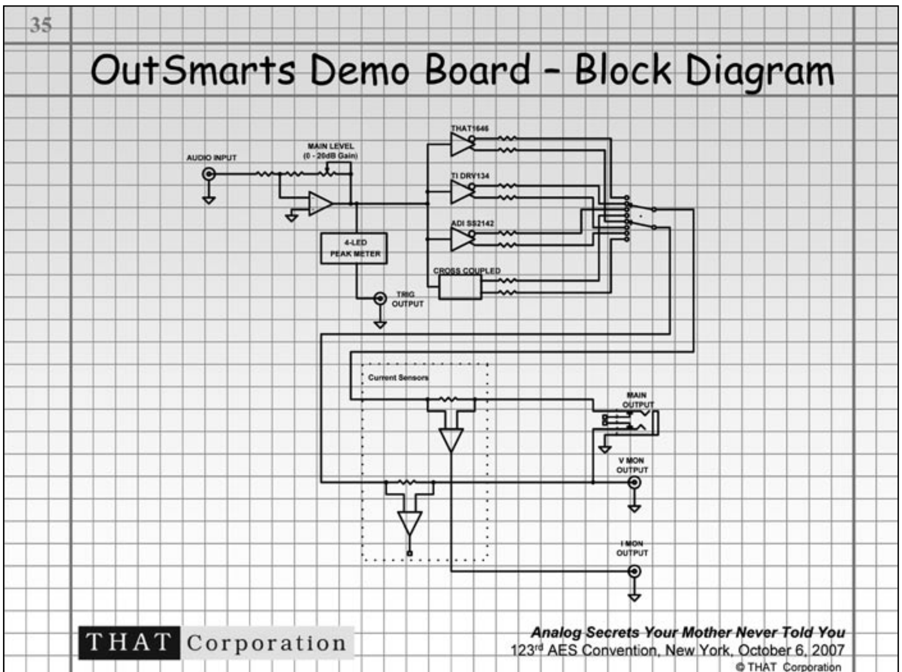

FIGURE 1: Demo board used by THAT Corporation to compare the performance of various balanced line drivers. The board's output allows monitoring of voltage in the active output and current in the grounded output. (Courtesy of THAT Corporation)

taminate other circuitry leading to crosstalk of a very nasty sort as they return to the supply pins of the device."

The performance of the "discrete" cross-coupled circuit using 5534 op amps shows instability similar to the SSM2142 (**Fig. 3**). Hebert offered the following comment for this report: "This behavior is not uncommon in op amps when they are in current limiting. Often there's a need to enclose more circuitry in the feedback loop created by the current limit circuitry for one polarity than the other. This makes the stability of such loops tricky. Since current limiting is not 'normal operation,' this is usually considered acceptable as long as the part is protected. In the case of the floating, balanced output driver driving a single-ended load, the cross-coupled op amp topology goes into current limit every time it's clipped. In audio, clipping can occur a lot in 'normal operation.'"

The OutSmarts line drivers were designed to imitate some of the more important characteristics of output transformers, including high common-mode output impedance (several kΩ) and low differential output impedance. Feedback minimizes common-mode output current (Iout+ = Iout-). The output appears across two output terminals even if one is grounded. In fact, THAT recommends grounding one output whenever unbalanced loads are driven. Doing so yields lower noise and distortion, without stability issues. Like the InGenius® line receivers, the OutSmarts devices use bipolar rather than JFET technology.

Hebert also reinforced a few points made by Bill Whitlock in his 2005 AES presentation *Grounding and System Interfacing*, which I reported on the Jan. 2007 issue of *aX*³. Most important is that common-mode rejection is determined by impedance matching, and has nothing to do with signal balance in the two conductors of a balanced line. Signal balance affects headroom, and may even affect crosstalk in multi-pair cables, but it does not affect common-mode rejection. Trimming of differential line drivers to obtain matched output impedances, and hence the best CMRR, is complex.

THAT's 1606 and 1646 devices have been internally trimmed, eliminating the need for user adjustments. When used with their InGenius differential input receivers, optimum CMRR is easily achieved, along with excellent signal symmetry. The OutSmarts line drivers are clearly a significant contribution to the analog audio IC market.

Other presentations given at THAT's seminar are also worth mentioning. Bob Moses, IC Program Manager for THAT (rwm@thatcorp.com), presented "New ICs," which include the 2162 dual VCA

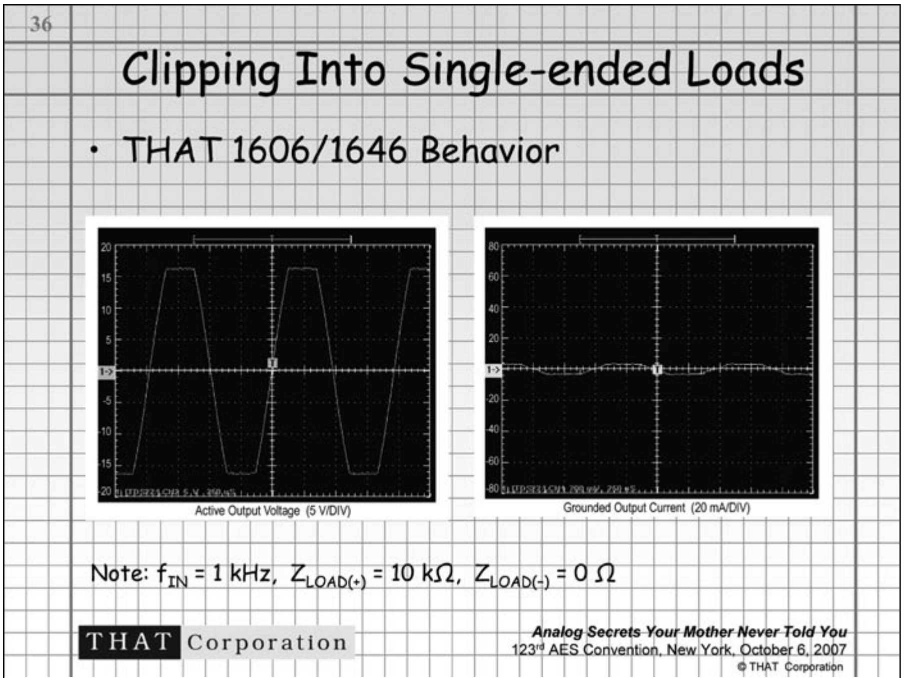

FIGURE 2: THAT 1606 and 1646 balanced line drivers clipping into grounded, single-ended 10k loads. The current into ground, shown on the right, is an inverted replica of the output voltage, and shows no signs of instability. (Courtesy of THAT Corporation)

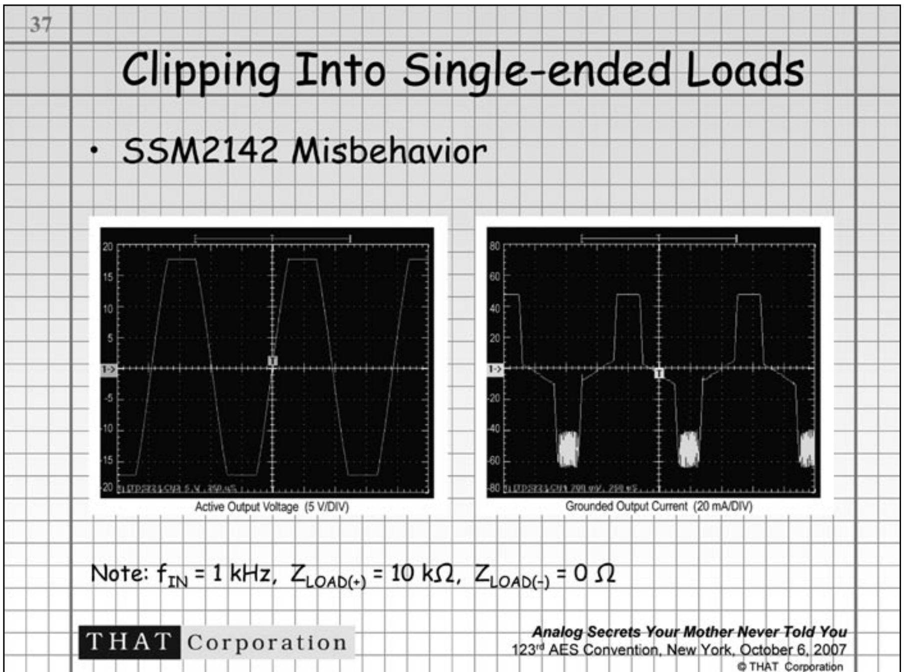

FIGURE 3: The well-known SSM2142 cross-coupled balanced line drivers. The output current driven into ground increases to the current limit of the device. (Courtesy of THAT Corporation)

and the 1280-series of dual line receivers. The 2162 devices are "pre-trimmed" and feature two completely independent VCAs in a QSOP-16 package. The 1280-series devices are manufactured in three gain versions, and are pin-compatible with the TI/Burr-Brown INA2134 and INA2137 devices. They are clearly intended for applications in which the cost of the InGenius products is prohibitive.

Rosalfonso "Ros" Bortoni, Applications Engineer (rb@thatcorp.com), presented "Mic Preamps," focusing on THAT's 1510-series of mike preamp ICs. Bortoni revealed a method for eliminating pops caused by fluctuating DC offsets when the gain is changed, and also showed an implementation with DC servo control. He also illustrated a method for implementing digital gain control using CMOS switches.

Leslie B. Tyler, President of THAT (lbt@thatcorp.com), presented "VCA/RMS & Log Math," which showed designs of Basic Voltage Controlled Amplifiers (VCAs), RMS Detectors, and Analog Engines® using THAT's 2181, 2252, and 4315 chips. He also discussed "Cool 'log math' " to simplify designs using these chips. The excellent Power Point presentation that accompanied this seminar is available under "Tradeshow News" on THAT's website, www.thatcorp.com. **Figures 2-4** in this report are taken from that slide show, and I thank Denise M. Waterhouse, Marketing Manager (dmw@thatcorp.com) for granting permission to reproduce them here. Thanks also to Gary Hebert for clarifying a few points while this article was in preparation.

THAT's seminar was one of the most worthwhile sessions I've attended at AES. Feel free to contact the various presenters at the e-mail address given—I have found the entire staff at THAT to be helpful and enthusiastic in answering questions about products and applications.

NATIONAL CHIPS

National Semiconductor also offered several audio design seminars. Unfortunately, I seemed to be tied up with other convention activities during the times they were given, but a visit to their booth was well worth my time. National seems to have increased their commitment to audio applications and high-performance audio op amps in recent years, and I urge anyone involved with analog design to visit their website (www.national.com), which has an enormous assortment of product data and application notes. In particular, National's LME 49000 series of bipolar-input op amps boasts THD of 0.00003% driving 600Ω loads. National notes that the LME 49870 op amp has been "optimized for superior audio fidelity" and features a slew rate of ±20V/µs, a gain-bandwidth product of 55MHz, and a low input noise density of 2.7nV/√Hz. The SOIC-packaged chip also can operate over a very wide range of supply voltages, from ±2.5 to ±22V.

The op amp can also be driven to within 1V of either supply rail when driving 2k loads or higher; 1.4V for 600Ω loads. Unlike many low-noise op amps suitable for high-gain applications, the LME49870 is unity-gain stable. A dual version, the LME49860, is also available in both SOIC and DIP packages. One suggested application is a passively-equalized RIAA phono preamp, shown on the front page of the 49860 datasheet (**Fig. 5**). The 49000 series op amps are well worth investigating.

Texas Instruments was promoting their own impressive lineup of high-performance ICs for professional audio

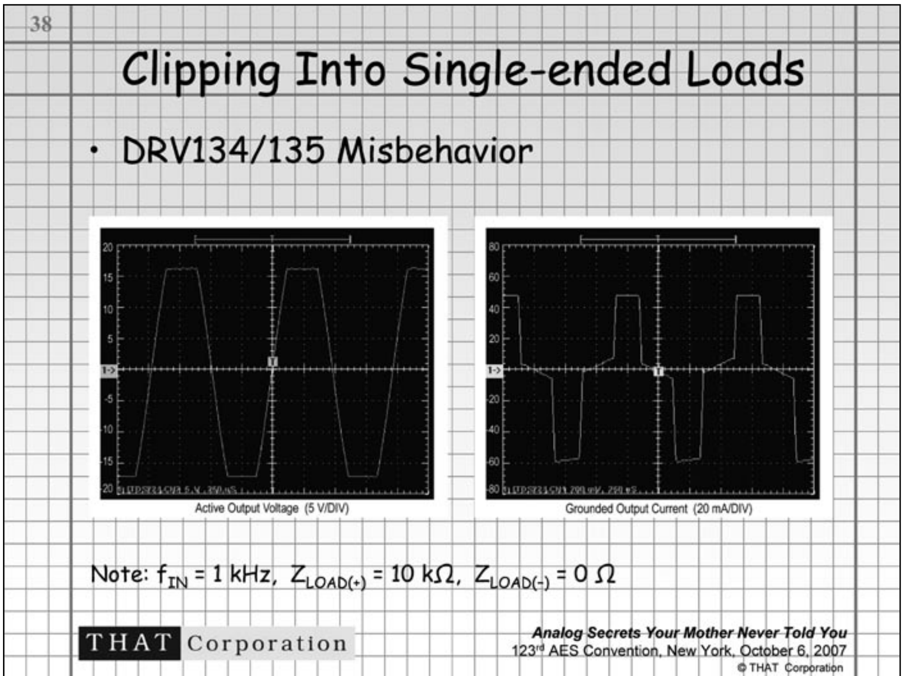

FIGURE 4: The DRV134/135 cross-coupled balanced line drivers. They exhibit the same ground-current problem as the SSM2142, though without the instability seen in Fig. 3. (Courtesy of THAT Corporation)

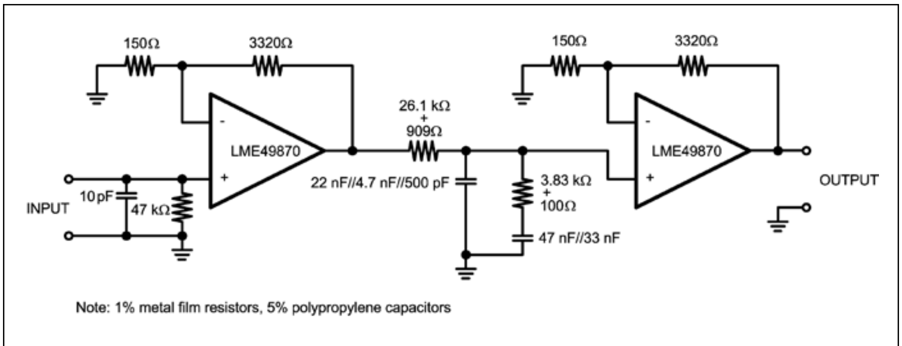

FIGURE 5: Passively-equalized RIAA phono preamplifier using National Semiconductor's high-performance LME49680 dual op amp. (Courtesy of National Semiconductor and THAT Corporation)

use, including those manufactured under the Burr-Brown name (www.ti.com). These included some of the best op amps available for audio applications, as well as A/D and D/A converter chips operating at 24-bit resolution and sampling frequencies of 192kHz, including the PCM1792/4. Their *Audio Solutions Guide*, which they were distributing in printed form at the convention, is available in .PDF form on their website, as are numerous other documents of interest to audio builders and designers. *aX*

REFERENCES

1. Meyer, E. Brad and David R. Moran. "Audibility of a CD-Standard A/D/A Loop Inserted into a High-Resolution Audio Playback," *Journal of the Audio Engineering Society*, Sept. 2007, pp. 775-779.

2. Herbert, Gary. "An Improved Balanced, Floating Output Driver IC." Presented at the 108th convention of the Audio Engineering Society, Paris, France, Feb. 19-22, 2000. AES Pre-print 5152, www.aes.org.

3. Galo, Gary. "Grounding and System Interfacing," *audioXpress*, Jan. 2007, pp. 26-33. (This article includes a summary of Bill Whitlock's Oct. 7, 2005, AES presentation "Audio System Grounding and Shielding—An Overview").