show report | By Gary Galo

New York AES 2009

The audio industry feels the economic pinch.

The 127th Convention of the Audio Engineering Society was held in New York City over the long Columbus Day weekend, Oct. 9-12, 2009. The state of the world economy was alarmingly evident at the 2009 convention. I usually arrive at the Javits Convention Center late Thursday afternoon to beat the crowds at the registration desks, which normally occupy a large space in the lobby. When I arrived for this convention, there was no sign of AES anywhere in the lobby—the registration desks had been moved into the exhibit area, occupying space normally reserved for the large, high-priced exhibits found at the heads of the exhibit aisles.

In recent years, there have been 11 aisles of exhibits at AES, with several demonstration rooms off the main exhibit area, and two corridors of demo rooms on the lower level of Javits. This year, there were eight shortened aisles of exhibits and no demonstration rooms. One manufacturer commented to me that no one was willing to spend the "tens of thousands" of dollars that it costs to rent a demo room for the four days. Even Sony Corporation, which normally has a large exhibit area at the head of one of the aisles, and has often rented a demo room, shared a small booth with one of their dealers.

Only one semiconductor manufacturer exhibited at this year's AES: THAT Corporation. Analog Devices and Cirrus Logic, regular exhibitors in the past, were absent, as were TI/Burr-Brown and National Semiconductor. Cirrus was listed in the Exhibitor Directory, but their booth was empty—they must have pulled the plug at the last minute. By my count, there were just under 320 exhibitors this year, compared to nearly 450 in 2007. The convention itself seemed to lack the usual focus—even the convention theme, "Making the Right Connections," was rather generic and left me with the impression that they couldn't think of anything better. Nonetheless, there were many very worthwhile exhibits, presentations, and workshops.

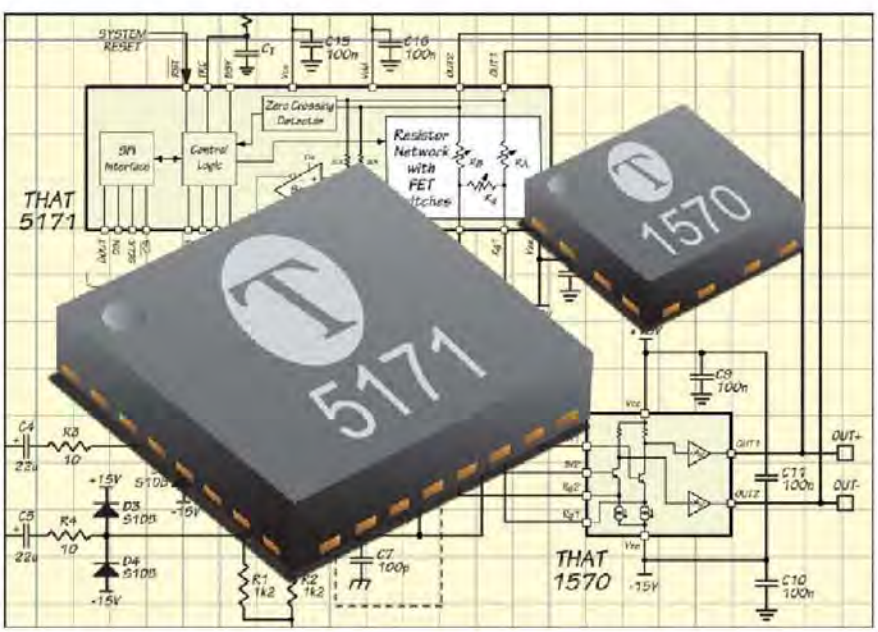

PHOTO 1: THAT 1570 Programmable Microphone Preamplifier and 1571 Digital Preamplifier Controller. These high-performance chips are manufactured in small QFN packages (Quad Flat No leads) to minimize PC board real estate. (Courtesy of THAT Corp.)

THAT SEMINAR

THAT Corporation (www.thatcorp.com) presented another excellent manufacturer's seminar, a sequel to their "Analog Secrets Your Mother Never Told You," which they featured at the 2007 and 2008 conventions. At this year's presentation, they discussed their new 1570 high-performance microphone preamplifier (**Photo 1** and **Fig. 1**). The 1570 employs a current-feedback amplifier and has a slew rate of 53V/µS.

Noise is an impressive 1nV/√Hz at 60dB gain, THD+N is 0.0008%, and bandwidth is 4.2MHz. Gain is adjustable up to 60dB, with three external resistors, but the 1570 can also be mated with their 1571 Digital Preamplifier Controller IC, which allows programmable gain of +5.6dB and +13.6 to 68.6dB in 1dB steps. Four general-purpose control outputs can be used to control a variety of peripheral functions, including muting, phantom power switches, signal routing switches, LED indicators, and so on.

The company offers this description of the 1570 on their website: "Designed from the ground up in THAT's complementary bipolar dielectric-isolation process and including laser-trimmed Si-Chrome thin-film resistors, the 1570 improves on existing integrated microphone preamps by offering more versatile gain configuration, lower noise at low gains, higher slew rate, and lower distortion." The 1570 has differential inputs and outputs and, unlike many IC-based mike preamps, it can operate on rail voltages as high as ±17V, giving it a dynamic range of 130dB.

Back in 2008, Sadie, manufacturer of one of the finest high-end digital audio editing systems, came close to closing its doors. Fortunately, Prism Sound rescued the company and Sadie is back, alive and well (www.prismsound.com; www.sadie.com). Sadie editing systems have typically run in the five-figure range, in part due to the need for proprietary computer hardware. This year, Sadie introduced their first editing system designed to operate on any Windows-platform computer, though Sadie recommends a Prism Sound Orpheus sound card (**Photo 2**).

The Sadie 6 system is available in four configurations: Radio Producer, Post Suite, Mastering Suite, and Sound Suite. The Sound Suite is the premium version, which includes all plug-ins offered by the company. The Sadie 6 software can also be used with their own proprietary hardware, such as the PCM-H8 Digital Audio Workstation.

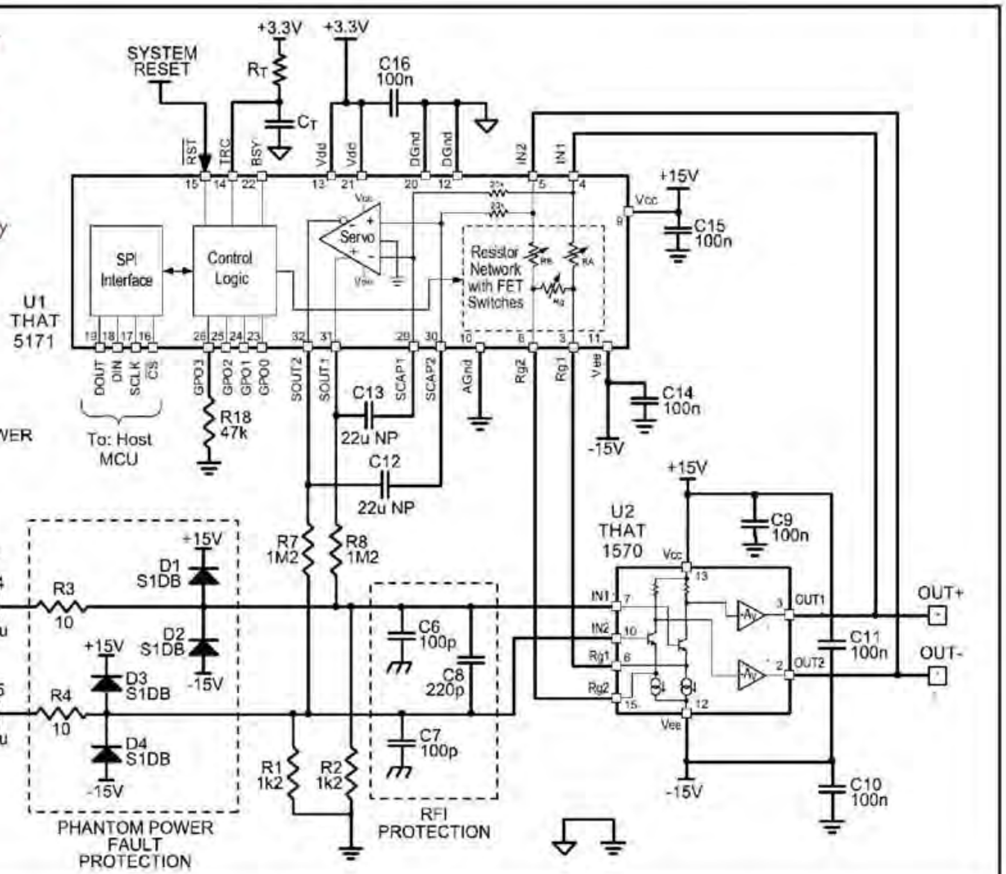

FIGURE 1: THAT 1570: THAT 1570 and 1571 in a typical microphone preamp configuration with phantom powering and differential outputs. (Preliminary drawing, Courtesy of THAT Corp.)

PHOTO 2: Now under the ownership of Prism Sound, Sadie has introduced Sadie 6, a digital audio recording and production system that runs on a standard Windows platform. The Sadie 6 software can also be used with their proprietary hardware, such as the PCM-H8 Digital Audio Workstation. (Courtesy of Prism Media Products Ltd.)

Though Prism Sound has gained a reputation for their outstanding A/D and D/A converters, they have also become a leader in the field of audio measurement. This year they were featuring their dScope III Audio Test System, including a new, low-cost version (**Photo 3**).

The test system comes in three versions: dScope Series III Digital + Analog Audio Test System, which is the original full version with extensive digital and analog capabilities; dScope Series IIIA+ Analogue-Plus Audio Test System for those who don't need digital audio capabilities, but do need the advanced features of the full dScope Series III; and dScope Series IIIA Analogue Audio Test System for those who don't need digital audio capabilities or the full power of dScope analysis. All standard audio measurements are supported, including amplitude, frequency, inter-channel phase, distortion, sweeps of up to four measurements simultaneously, up to 40 simultaneous FFT derived measurements, digital interface testing, limit checking, both scalar values and traces, real-time oscilloscope display, real-time frequency spectrum display, and filter shape display.

There is little new in audio recording these days, at least no significant improvements. The industry has not seen the need to move beyond 192kHz/24-bit PCM, or the DSD format used for SACDs. Digital eXtreme Definition (DXD), developed by Pyramix/Merging Technologies, is a PCM system that does 32-bit floating, 352.8kHz recording, but it's used primarily for mastering and editing of high-resolution recordings prior to conversion to DSD for release on SACDs (www.merging.com).

Even DXD has been around for at least two years, and was on exhibit at AES in 2007. Version 6.1, exhibited this year, offered workflow improvements over V. 6.0. Given the fact that the consumer has shown little interest in high-res digital, the industry really has had no incentive to take digital audio any further.

What's changing is physical size of high-performance audio recorders, and manufacturers are making high-resolution digital recording more portable than ever. Sony featured their PCM-M10 palm-sized professional digital recorder, which records linear PCM at sample rates up to 96kHz with up to 24-bit resolution (**Photo 4**). Linear PCM data is stored as WAV files, but the PCM-M10 can also record in MP3 format, for non-critical applications. It has 4GB of built-in flash memory, which can store up to six hours of CD-quality (44.1kHz/16-bit) audio.

It also has a dual MicroSD and Memory Stick Micro (M2) slot, for additional memory options. The recorder has a pair of built-in omnidirectional condenser microphones, and a built-in speaker. A stereo mini jack allows connection of external microphones, and there's also a stereo line-level input, and a line-level output. The PCM-M10

PHOTO 3: Prism's dScope III Audio Test System is available in three versions: Analogue, Analogue Plus, and Digital + Analogue. Prism's proprietary hardware interfaces with any Windows computer via a USB-2 connection. (Courtesy of Prism Media Products Ltd.).

PHOTO 4: Sony's PCM-M10 palm-sized professional digital recorder, which records linear PCM at sample rates up to 96kHz with up to 24-bit resolution. The recorder has 4GB flash memory, and a pair of built-in omnidirectional microphones. (Courtesy of Sony Corp.)

requires 3V to operate, which can be supplied by a pair of AA batteries, or the included external AC adaptor.

Tascam (www.tascom.com) was exhibiting their HS-P82 8-track digital field recorder (**Photo 5**). The unit records on dual Compact Flash cards, with no moving parts, and supports eight-track recording at up to 96kHz/24-bit, and four-track at 192kHz/24-bit. It will also record a stereo mixdown. Inputs include eight high-quality microphone preamps with +48V phantom powering.

The eight XLR input connectors can also function as line inputs, and stereo XLR line outputs are also included. Digital I/O includes eight AES/EBU digital inputs and outputs, via a DB-25 connector, and a stereo digital BNC output. Each AES/EBU digital input is equipped with a sample rate converter, allowing compatibility with virtually any standard digital audio datastream.

The recorder also has SMPTE time code and word sync in and out, via BNC connectors. A PS-2 keyboard connector facilitates naming of tracks, and a USB-2 connector allows high-speed data transfer to computers. The recorder stores data as Broadcast WAV files that include iXML metadata, allowing quick import into nearly any video or audio editing system. Several powering options are available, including AA (X10) or NP batteries, the supplied AC adapter, external DC input, or a V-mount adapter for Endura batteries. The HS-P82 is housed in a rugged, lightweight aluminum chassis. With a list price of $6500, it's certainly not cheap, but the HS-P82 is a breakthrough in high-performance multi-track portability.

HISTORICAL PRESENTATIONS

Among the historical presentations, Tom Fine's "Mercury Living Presence" traced the history of this record label, whose classical recordings are still highly regarded by audiophiles for their extraordinary realism, and by music lovers for the many outstanding performances captured by Mercury. Tom is the youngest son of the late Mercury engineer C. Robert Fine and producer Wilma Cozart Fine (sadly, Wilma passed away only two weeks before this convention). Tom's presentation was filled with photographs and illustrations documenting the development of the fabled Mercury sound from their single-microphone beginnings in the late 1940s, through the three-spaced omni stereophonic technique first captured on ½", three-track tape, and then on 3-track, 35mm magnetic film. He also discussed the CD reissues of the classic Mercury recordings for which his mother returned from retirement to perform the analog-to-digital transfers. Tom also played many recorded examples along the way.

In recent years, AES has been devoting one room to presentations requiring a good audio system, complete with PMC M2-series loudspeakers that are capable of filling a very large room with surprisingly good sound. Fortunately, Tom's presentation was held in this room, allowing the audience to

PHOTO 5: Tascam's HS-P82 8-track digital field recorder stores digital audio on dual Compact Flash cards, with no moving parts. It supports 8-track recording at up to 96kHz/24-bit, 4-track at 192kHz/24-bit, and will also record a stereo mix down. (Courtesy of Tascam).

really hear the examples he played. This was an excellent presentation on one of the most important chapters in what many of us regard as the Golden Age of classical recording.

Those who are active in the Association for Recorded Sound Collections (ARSC) are by now well familiar with the research of David Giovannoni (**Photo 6**). At the 2008 ARSC Conference, held at Stanford University in Palo Alto, Calif., Giovannoni and his colleagues at First Sounds reported on the research they've conducted on Edouard-Léon Scott de Martinville, the inventor of the Phonautograph. The Phonautograph was a recording device that etched lateral sound vibrations on paper coated with lamp black. The paper was wrapped around a drum that was hand-turned. The inventor had no means of playing back these Phonautograms, as they're called—the purpose was to provide a means of visual analysis of sound vibrations, a precursor to the modern-day oscilloscope.

Giovannoni and his colleagues, during the course of research at French Academy of Sciences in Paris, discovered that many of Scott's Phonautograms still existed and, by scanning and digitizing them, they could be played for the first time! Several digitally restored phonautograms were played at the ARSC conferences in 2008, the earliest dating from 1860, 17 years prior to Edison's invention of the phonograph. Giovannoni's AES presentation "Before Edison" offered attendees an update on their ongoing research, including the discovery of even earlier recordings. Anyone interested in sound recording history should visit their website www.firstsounds.org, where you can hear sound clips of some of these recordings.

Three faculty members from the University of Tokyo—Teruo Muraoka, Takahiro Miura and Tohru Ifukube—contributed to a poster session titled "Frequency Characteristics Measurements of 78-rpm Acoustic Record Players by the Pulse Train Method" (Mr. Muraoka was one of the developers of the CD-4 Quadraphonic LP record; his Pulse Train measurement method dates back to 1973, and was originally used for phase measurements on the disc recording and reproduction system). The authors


PHOTO 6: David Giovannoni examines one of Léon Scott de Martinville's 1860 phonautograms in the archives of the *Académie des Sciences* of the *Institut de France*, where it was deposited in 1861 (left); detail of an 1859 phonautogram made by Scott and included in his patent paperwork, preserved at the *Institut National de la Propriété Industrielle* (INPI), the French patent office (right). (Courtesy of First Sounds, www.firstsounds.org; photos by Isabelle Trocheris)

have been involved in research on restoration of seriously damaged audio signals using a technique known as Generalized Harmonic Analysis. The premise of the authors is that, in order to properly reproduce vintage recordings, it is important to know the frequency response characteristics of contemporary reproducing equipment. The flaw in this argument is that phonographs manufactured during the 78-rpm era, especially during the era of acoustical recording (up to c. 1925), were not capable of revealing all that was contained in the record grooves, and often the frequency response of an old phonograph has little to do with the most optimum playback characteristics for vintage records.

The authors claim that the Pulse Train method gives a more precise measurement of the frequency response characteristics of whatever is being measured, whether a loudspeaker or an acoustic phonograph. They made their own Pulse Train record by putting eight radial scratches on a 78-rpm shellac record. That record was used to measure the frequency response of a Victor "Credenza," an acoustic phonograph invented by Maxfield and Harrison at Bell Labs for the reproduction of early electrical recordings (the "Credenza" was the first phonograph with an exponential horn). The authors provide a frequency response graph for the Credenza that they claim is more accurate than the original graph published by Maxfield and Harrison in 1926.

There are many English language difficulties in their convention preprint, as well as in the handout they distributed, so I hope my summary is reasonably accurate. It is difficult to determine what they consider to be an optimum restoration of an old recording. Whether they are trying to reproduce the sound of an old phonograph or compensate for its deficiencies is unclear.

Mr. Muraoka generously gave me a CD he produced titled "Super Nikisch: Monumental Recordings 1913-1920." The disc contains recordings by the legendary conductor Arthur Nikisch, recordings with which I am very familiar from various reissues, as well as a vinyl 78-rpm pressing made from original metal parts of his 1914 *Oberon* Overture by Weber. The sonic results on the CD I received were very disturbing. Throughout, the music is accompanied by serious artifacts from the digital signal processing, including swirling sounds, comb filtering, plus strange-sounding pops and dropouts that are probably by-products of the overzealous removal of clicks and pops on the original records.

ELECTRONICS FOR RECORDING STUDENTS

Michael Stucker, a faculty member at Indiana University, chaired the workshop "Teaching Electronics to Recording Students—Why Bother?" Panelists included Brian Bender of Motherbrain (substituting for Eric Brengle of Swinghouse Studios, who was unable to attend), Eddie Ciletti from Tangible Technology, Dale Manquen from Manco, Walter Sear from Sear Sound, and Jensen Transformers' President Bill Whitlock. The panel focused on integrating electronics into an audio recording curriculum.

Several "basics" were deemed essential areas of study, including voltage, current (conventional and electron), power, AC and DC, series and parallel circuits, passive components (resistors, capacitors, inductors, transformers), active components (tubes, bipolar and FET transistors, op amps and other ICs), reading schematics, and safety grounds. More advanced theory would follow, with math background being important. Graphing skills are also important, for frequency response, transfer functions, and so on.

Many of the panelists emphasized the need for hands-on experience with electronics, which few young people have these days. Whitlock noted that this is not a plug-and-play world. There are many badly-designed products on the market, and students need to be educated in proper troubleshooting techniques when dealing with interfacing problems, including those caused by a frequent source of hum: the "pin 1" problem. In identifying prospective students for a recording engineering program, latent interests are important. Students should be interested in working with their hands; they should be inquisitive about technology, and interested in taking old equipment apart. Altogether, this was a very worthwhile session.

BLU-RAY AUDIO

Stefan Bock, of msm-studios in Germany, chaired the session "Blu-ray as a High Resolution Format for Stereo and Surround." Panelists included Gary Epstein from Dolby Labs, Tom McAndrew from DTS, Johannes Müller from msm-studios, and Ronald Prent from Galaxy Studios in Belgium. The company represented by Bock and Müller, msm-studios, is promoting the Pure Audio Blu-ray format for high-resolution stereo and surround audio. The Pure Audio format was designed to make Blu-ray audio playback as simple as a CD. Unlike DVD-Audio, which normally requires that you operate your television in order to play an audio-only disc, Pure Audio Blu-ray is an autonomous audio medium. Colored keys on the remote control select the desired program, whether stereo or surround, DTS, Dolby Digital or linear PCM.

During his presentation, Müller noted that SACD remains a niche market, requiring special players, and the audio market simply isn't big enough to have its own HD format. Part of Blu-ray's appeal is that it isn't just an audio format—it's also the high-end format for video, and offers a chance to finally produce a high-def audio standard for stereo and surround. As a demonstration, he played a recording of the Grieg Piano Concerto, with Percy Grainger's Duo-Art reproducing piano roll made in the early 1920s accompanied by the Kristiansland Symphony Orchestra conducted by Rolf Gupta. Several audio discs, including the Grieg, are currently available on the Pure Audio website (www.pureaudio-bluray.com).

Twice during this panel it was stated that DVD-Audio has a maximum sampling rate of 96kHz in stereo, which is not true. During Q&A I pointed out that DVD-Audio supports 192kHz sampling rates in stereo—the 96kHz limit is true only for surround (the only difference between Blu-ray and DVD-Audio is that Blu-ray allows 192kHz/24-bit in surround). For many audiophiles, the lack of consumer interest in high-resolution digital formats such as DVD-Audio and SACD is frustrating. Epstein acknowledged the difficulty of promoting Blu-ray as an audio-only format when DVD-Audio didn't make it. Bock and Müller really didn't have an answer to the question of overcoming consumer ambivalence toward high-quality sound.

Epstein and McAndrew each offered their thoughts on why the HD audio formats developed by their respective companies are superior to the competition. DTS is against linear PCM on Blu-ray discs, noting that DTS HD is a lossless CODEC (www.dts.com). The same is true of Dolby True HD, of course (www.dolby.com).

Prent expressed his preference for linear PCM, noting its superior sync. He played an excerpt from *Black Symphony*, featuring the Dutch band Within Temptation and the Metropole Orchestra. Everyone in the orchestra was miked, producing a decidedly film-like sound through the PMC M2-series loudspeakers. The demonstration switched between various audio formats—DTS, Dolby Digital, and linear PCM. Prent stated that linear PCM allows greater involvement with the concert.

GREAT PICTURE—MEDIOCRE SOUND

Michael Fremer, Senior Contributing Editor of *Stereophile* magazine, chaired a workshop titled "We Got the Picture. What Happened to the Sound?" Panelists included John Atkinson, long-time Editor of *Stereophile*, Steve Berkowitz of Sony/BMG, Greg Calbi of Sterling Sound, Alan Douches of West West Side Music, Bob Ludwig of Gateway Mastering & DVD, EvaAnna Manley of Manley Audio Labs, and Randy Ezratty. The panel focused on the irony in today's audio and video preferences. Consumers are enamored, and rightly so, of 1080p high-definition pictures, yet are also satisfied with audio resolutions lower than that of conventional CDs. The general public finds compressed digital standards, such as MP3, more than adequate for music listening.

Ludwig stressed the convenience factor—the ease of MP3 playback versus CDs, SACDs, and DVD-Audio discs is comparable to the cassette tape versus the LP years ago. Ludwig also pointed out that SACD won the high-res audio format war, with 360 releases in that format last year compared to 15 on DVD-Audio. I didn't check the figures, but at 360 releases SACD may have won the war, but it's nonetheless a fringe audiophile format, and DVD-Audio seems all but extinct. Mastering is now geared toward earbuds or earphones, and not high-quality loudspeakers. Consumers want free music and they're getting what they pay for.

Ludwig also recalled the old AR demonstration room in Grand Central Station, asking why we can't have something similar today. Fremer said that he often invites friends of his own kids into his listening room, to hear their favorite music on a high-resolution system from uncompressed sources. Their reactions have been one of amazement—"I had no idea music could sound like this!," or words to that effect. Perhaps an AR-style listening room in a prominent public location would help educate today's listeners and excite them about audio as much as they have become excited about video. Someone would need to foot the bill, of course. Fremer also pointed out that very few people listen to music to the exclusion of all else.

In my 2007 AES convention report, I discussed an article by two members of the Boston Audio Society—E. Brad Meyer and David R. Moran—who claim that double-blind, ABX testing fails to reveal any differences between CDs and high-resolution digital formats[1,2]. Their conclusion was that there are no audible differences. In that AES report, I concluded that either the participants can't hear, or there's something wrong with the test. Atkinson noted that ABX testing fails to reveal differences between high-res and 16-bit/44.1kHz digital, yet conventional listening consistently reveals those differences. One audience member expressed the view that double-blind testing changes perception vis-à-vis conventional listening.

Bravo to Berkowitz for stating that, yes, there are audible differences between digital audio at 44.1kHz/16-bit, and 96kHz, 24-bit. Ludwig mentioned the audible degradation introduced by jitter when a glass master is made faster than real-time. In his view, even 4× is sonically unacceptable. It was an altogether lively and worthwhile discussion.

AUDIO MYTHS

There's always been a strong contingent in AES who are skeptical, and in some cases downright hostile, to the idea that there are audible differences between amplifiers, cables, and high- versus low-resolution digital formats, and so on. That skepticism was reflected in the workshop "Audio Myths—Defining What Affects Audio Reproduction," chaired by Ethan Winer of RealTraps, with the panelists Jason Bradley of Intel Corporation, Poppy Crum of The Johns Hopkins University School of Medicine, and James Johnston of DTS.

Crum, a neuroscientist, discussed how the ear processes sound. Dr. Crum described the Haas Effect, the McGurk Effect, and auditory continuity (how the ear/brain fill in missing information, which is why music sounds continuous even when a record has clicks and pops). Crum is an acknowledged authority in her field, and while no rational attendee would take issue with anything she said, there was a definite disconnect between her presentation and basic premise of this session. I believe the other panelists were hoping that her portion of the presentation would lend credence to some of their own agenda items, but it was not to be, at least not from where I sat.

The panel defended double-blind testing, and Winer snickered at a variety of audiophile tweeks, including Brilliant Pebbles (www.machinadynamica.com), Acoustic Art resonators (www.synergisticresearch.com), high-end power cords, and LP and CD demagnetizers, some of which I also find hard to swallow. (Full disclosure: Acoustic Art resonators are designed for treating room acoustics, and Winer's company RealTraps [www.realtraps.com] also manufactures room treatment products.)

Winer's argument that there is much more distortion in loudspeakers than in A/D and D/A converters just doesn't hold water, because the types of distortions produced by each are very different, and the distortion in a good loudspeaker will not necessarily mask the distortions produced by digital converters.

Winer described four audio parameters:

- Frequency response
- Distortion, including THD, IM, and aliasing "birdies"
- Noise, including hiss, hum, buzz, and vinyl crackles
- Time-based errors, including wow, flutter, and jitter

He also stated that transparency is not the only goal, and that sometimes euphonic coloration is desirable. The latter is certainly true, as evidenced by the popularity of tube, zero-feedback and single-ended designs in both high-end consumer and pro audio gear. Winer also spent some time on artifact audibility, but the audio demos that he played were virtually useless in that room on the decidedly low-end audio system. WAV files used for those demos, along with an explanation, are available on his website (www.ethanwiner.com/audibility.html). Winer disagrees with Bob Ludwig, stating that jitter is inaudible, as is dither (though he still recommends that you use it!).

However you may view the never-ending debate on what is and isn't audible, one thing is fairly certain: AES conventions are much more balanced than they were, say, 25 years ago. The workshops and presentations offered at AES conventions over the past 20 years have increasingly reflected the diversity of viewpoints prevalent at AES. In my view, this has only made the organization stronger. *aX*

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2. 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.

THAT's Balanced Line Drivers and Receivers

Getting the most from these high-performance chips.

By Gary Galo

During the past year, I completed the design and construction of an archival phono preamplifier, which provides variable equalization along with several other operational necessities for playing vintage records (78-rpm discs, pre-RIAA LPs, lacquer transcription discs, and so on). When it came to selecting topologies for the balanced inputs and outputs, I ruled out transformers due to high cost, physical bulk, and weight. The most practical choice, in terms of size and cost, are the balanced line driver and receiver ICs made by several IC manufacturers.

After surveying the field, it became clear that the products from THAT Corporation offer superior performance to any of their competitors' chips. Like IC op amps, these devices are capable of varying degrees of performance depending on the quality of the passive components used in the associated circuitry, and power supplies. This article describes how to get the most from these chips. Anyone intending to use them should download the datasheets for these devices on THAT's website (www.thatcorp.com).

During the October 2005 convention of the Audio Engineering Society (AES), I attended a seminar given by Bill Whitlock, President of Jensen Transformers, on "Grounding and System Interfacing." I gave a report on this presentation in the January 2007 issue of *aX*.[1] Near the end of his presentation, Whitlock described the InGenius series of balanced line receiver ICs he designed for THAT Corporation. Whitlock developed a unique bootstrapping technique that raises the common-mode input impedance into the megohm region, while maintaining reasonable values for the input bias return resistors.

Raising the values of these resistors will degrade noise performance, but Whitlock's clever topology maintains low noise, keeping the resistor values low but the common-mode input impedance very high. The three chips in the InGenius line are the THAT 1200, 1203, and 1206, which are identical except for gain: 0dB for the 1200, -3dB for the 1203, and -6dB for the 1206.

THAT Corporation gave a seminar at the October 2007 AES convention covering—among other things—their OutSmarts line of balanced line drivers. I reported on this seminar in the June 2008 issue of *aX*.[2] The OutSmarts line, which includes THAT's 1606 and 1646 devices, incorporates a dual feedback loop design which prevents excessive ground currents typical of cross-coupled output stages when clipping into single-ended loads. In fact, when driving single-ended loads, the OutSmarts devices yield their lowest noise and distortion when one output leg is grounded.

THAT 1200 OFFSET

The maximum output offset voltage for the THAT 1200-series chips is specified as 10mV, and most samples I've measured tend to run between 4 and 7mV. This may be fine for some applications, but, in the two locations I use them in my archival phono preamplifier, they are followed by DC-coupled gain stages. The typical "pro" audio solution to DC offsets is to fill equipment with electrolytic coupling capacitors, which is simply unacceptable in a piece of high-end audio equipment.

I e-mailed Gary Hebert, Chief Technology Officer at THAT, and asked whether it was possible to null the 1200's offset with a DC-servo integrator. He said that it should be possible to do so, applying the servo output to the Reference pin (pin 1). He noted one caveat: If you follow the integrator with the usual resistive attenuator, you'll seriously degrade the common-mode rejection of the chip. As you'll see shortly, the 1200-series chips actually contain the resistive divider internally, but the resistor values are such that the integrator must have a considerably longer time constant in order to achieve the same -3dB point for the circuit as it would in a more conventional implementation.

Integrators configured as DC servos for audio amplifiers have received scant coverage in electronics text-books and periodicals. The best tutorial on their design was published by Walt Jung in Analog Devices' massive 1993 *Systems Applications Guide.*[3] **Figure 1** shows an op amp configured as a unity-gain inverting amplifier with DC-servo control of the circuit.

The unity-gain inverter consists of IC1, R1, and R2 (Gain = R2/R1). IC2 is configured as an integrator, with its time constant determined by R5 and C1 (T = RC). In this application, the integrator functions as a DC-servo amplifier, applying DC feedback to the non-inverting input of IC1 through the resistive divider (attenuator) R3 and R4.

Following Jung's guidelines, R3 is equal to R1, and R4 is ten times the value of R2. The -3dB point for the entire circuit should be about 1/100 of the lowest audible frequency—usually 20Hz—in order to minimize rolloff and phase shift within the audible spectrum. R5 and C1 are selected for reasonable values; R5 will normally be at least 500k and the R5/C1 combination should yield a -3dB point for the entire circuit of around 0.2Hz.

Following the math outlined in Jung's article, which I won't repeat here, the integrator time constant for **Fig. 1** is 0.15 seconds and the -3dB point for the entire circuit is 0.19Hz. The simulated frequency response for this circuit is shown in **Fig. 2**, which verifies that the calculations are correct.

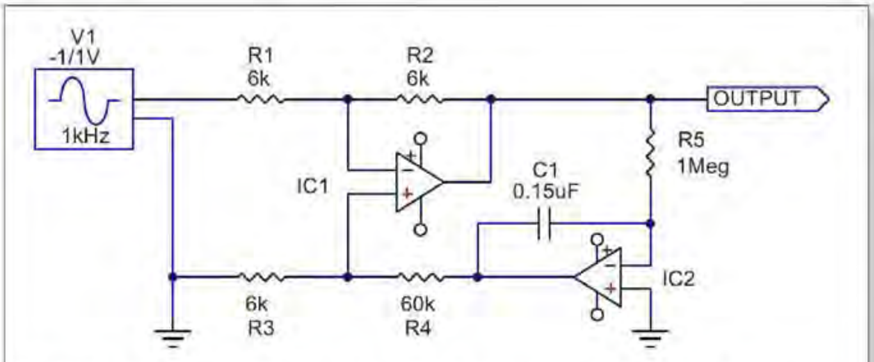

FIGURE 1: Unity-gain, inverting buffer with DC-servo control. R4 is normally ten times the value of R2. The integrator RC time constant is 0.15 seconds, producing a -3dB point of 0.19Hz for the entire circuit. The signal generator is for circuit simulation.

CIRCUIT SIMULATION

Figure 3 shows a simulation of the THAT 1200 balanced line receiver in a complete working circuit, including RFI protection at the input, as recommended in the manufacturer's datasheet, which also shows methods for ESD (electrostatic discharge) protection, if needed. Internally, the THAT 1200-series chips have diodes that

FIGURE 2: Simulated frequency response of the circuit shown in *Fig. 1*, confirming the -3dB point of 0.19Hz.

provide "modest protection against common low-voltage ESD incidents." The manufacturer's own testing has shown that repeated exposure to levels of ESD above 1kV through the non-inverting and/or inverting inputs can degrade the common-mode rejection and, ultimately, cause failure of the chip. If your application may expose the chip to high levels of ESD, I recommend using the protection. Note that the datasheet shows one method for the 1200 and another for the 1203 and 1206, because the 1203 and 1206 allow input signals that swing higher than the supply rails.

The circuit inside the dashed line is the THAT 1200 equivalent circuit, which is shown on the first page of the datasheet. The type of op amp used in the simulation isn't critical if all you need to do is verify that the circuit works with the servo connected, and check the low-frequency -3dB point. The generic "Op-Amp 5" model found in most Spice-based simulation programs will work fine.

When I asked Gary Hebert what op amp would be suitable for simulating the THAT 1200, he suggested a 5532. Bear in mind, however, that the performance of the THAT 1200 is an order of magnitude better than the dated 5532, with higher slew rate (12V/µS versus 9V/µS for the TI NE5532A), and wider bandwidth (22MHz versus 10MHz). Both chips will operate with rail voltages higher than the standard ±18V. The absolute maximum rating for the NE5532A is ±22V, compared to ±20V for the THAT 1200. So, the NE5532A (and the single NE5534A) will allow you to run your simulation at the full supply rail capability of the THAT 1200. A safety margin is always desirable for long-term reliability; the ±20V maximum rating ensures that long-term operation at ±18V will be no problem.

Note the output stage consisting of OA3 plus the resistors R101 through R104. R101 and R102 are the functional equivalents of R1 and R2 in **Fig. 1**. Normally, the reference pin (1) on the THAT 1200 is grounded, so OA3 functions as a unity-gain differential amplifier with a single-ended output.

With a DC-servo integrator connected as shown in **Fig. 3**, the reference pin is still effectively at signal ground. R103 and R104 become the functional equivalents of R3 and R4 in **Fig. 1**. But, R104 is internally set at 6k, the same value as R102. If these two resistors aren't the same value, the common-mode rejection of the chip will be severely compromised. So, R104 won't be ten times the value of R102.

In order to make the -3dB point for the entire circuit similar to **Fig. 1**, the integrator time constant will need to be considerably longer. C5 and R4, 1µF and 750k, yield an integrator time constant of 0.75sec, which produces a -3dB point for the entire circuit of 0.2122Hz. The simulated frequency response of this circuit is shown in **Fig. 4**, again verifying that the calculations are correct.

The THAT 1203 and 1206, with gains of -3dB and -6dB, respectively, may be desirable in applications where

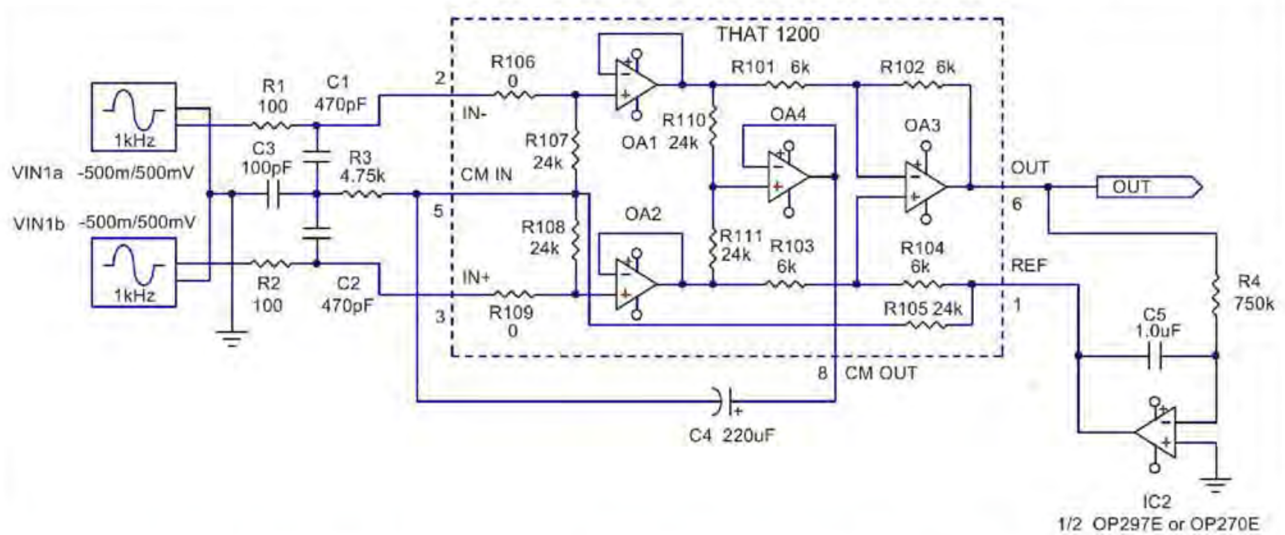

FIGURE 3: Simulation of the THAT 1200 balanced line receiver with input RFI protection and DC-servo control. R104 is internally set to the same value as R102. In order to achieve the -3dB point of 0.21Hz for the entire circuit, the integrator time constant must be set at 0.75 seconds.

higher input voltages are likely. In the 1203, R101 through R104 are the same values as in the 1200. But, in the 1206, R101 and R103 are 7k, with R102 and R104 set at 5k. This is no problem for the servo, since the math is a wash. The -3dB point for the circuit using a 1206 will be the same as it is with the 1203 and 1200.

So, you can swap 1200-series chips without having to re-calculate the integrator R/C values. In the 1203 and 1206, R106 and R109 are set at 7k, and R107 and R108 are 17k. These will obviously have no effect on the DC servo, but they do affect the overall gain of the device.

SERVO AMP CHOICES

The choice of op amp for the servo amplifier will depend on what's most important in your application—the lowest possible noise or the lowest possible output offset. Noise of the THAT 1200 is specified at an impressive -107dBu, so any servo amplifier has the potential of degrading its excellent noise performance. Gary Hebert suggested an OP27.

For a stereo pair, I prefer to use a dual op amp to conserve space, so I chose Analog Devices' OP270E. As noted in the datasheet, the OP270E offers comparable performance to the OP27E. With the OP270, I measured noise at -96dB relative to 2V RMS out.

Input offset voltage of the OP270E is extremely low—typically 10μV—but input bias current is 5nA. So, the output offset of the circuit will be somewhat affected by the source impedance, set by R4, which is 750k. Output offset of the THAT 1200 using the OP270E usually runs between 1 and 2mV, which may be sufficiently low for many applications. It wasn't for one of mine, however, so I decided to try the OP297E, which I have used in many other DC servos.

The OP297E has higher input offset voltage than that OP270E—25μV—but it also has ultra-low input bias current, typically 20pA. Output offset from the THAT 1200 using the OP297E is usually less than 0.2mV. Although the OP297E is noisier than the OP270E, noise performance of the THAT 1200 is degraded no more than 2dB compared to the OP270E. For applications where ultra-low output offset is necessary, the OP297E is the best choice. Digi-Key carries only the ceramic DIP packages for the OP297E and OP270E ("Z" suffix); the plastic DIP version ("P" suffix) are certainly fine, if you can find them.

THAT 1200 CONSTRUCTION

In choosing parts for these circuits, I avoided components containing steel, because I believe that non-permeable materials have a positive effect on the sound. **Figure 5** shows a single-channel THAT 1200 balanced line driver circuit with DC-servo control, input XLR connector, and power supply bypassing. The DH Labs D4 prototyping board sold by Old Colony is ideal for building a stereo pair using a dual op amp for the servo amplifier.

Photos 1 and **2** show a recommended layout for the circuits; the

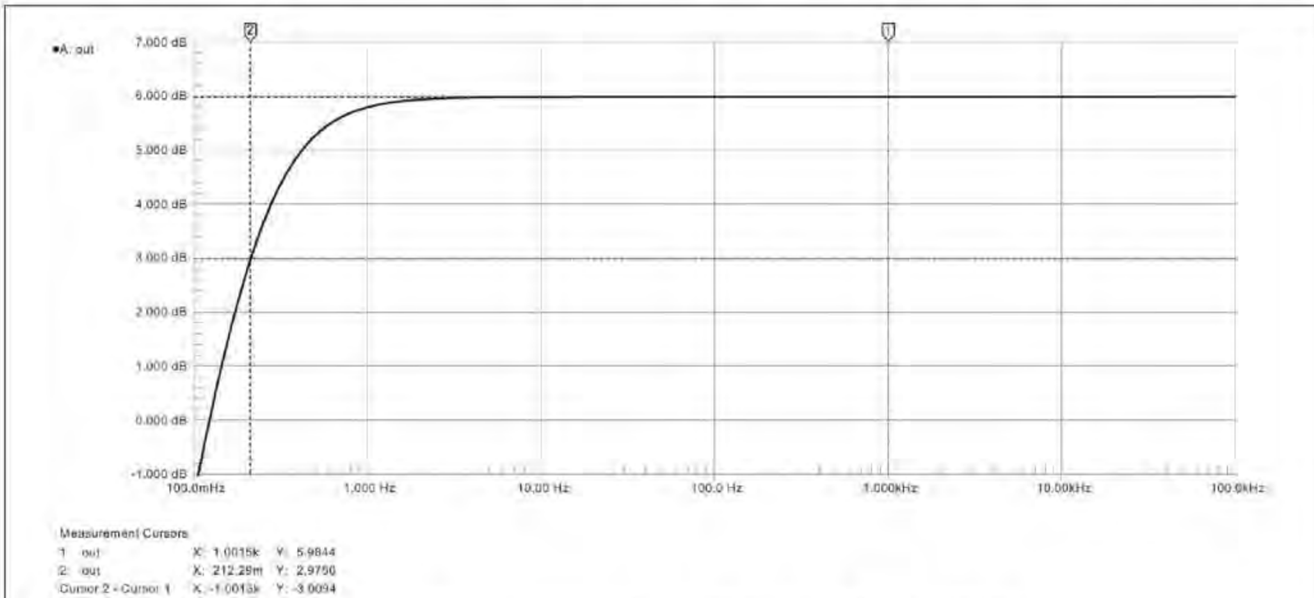

FIGURE 4: Simulated frequency response of the circuit shown in *Fig. 3*, confirming the -3dB point of 0.21Hz.

temporary leads are attached for testing. Mount the supply bypass capacitors on either end of the board. You'll need to cut traces under these capacitors to avoid shorting them out (**Photo 2**), and install jumpers to make connections to the supply rails and the center ground bus. I recommend Nichicon FG-series electrolytics bypassed with Wima or Roderstein polyesters.

I used Mil-Max machined-pin, gold-plated IC sockets for the THAT 1200 chips and the servo op amp, because I was still experimenting after my boards were built. Consider them optional. If you wish to experiment with different servo op amps, or the three 1200-series chips, I suggest soldering the ICs to an Aries gold-plated header. This gives a gold-on-gold connection when the chips are mated with the IC sockets.

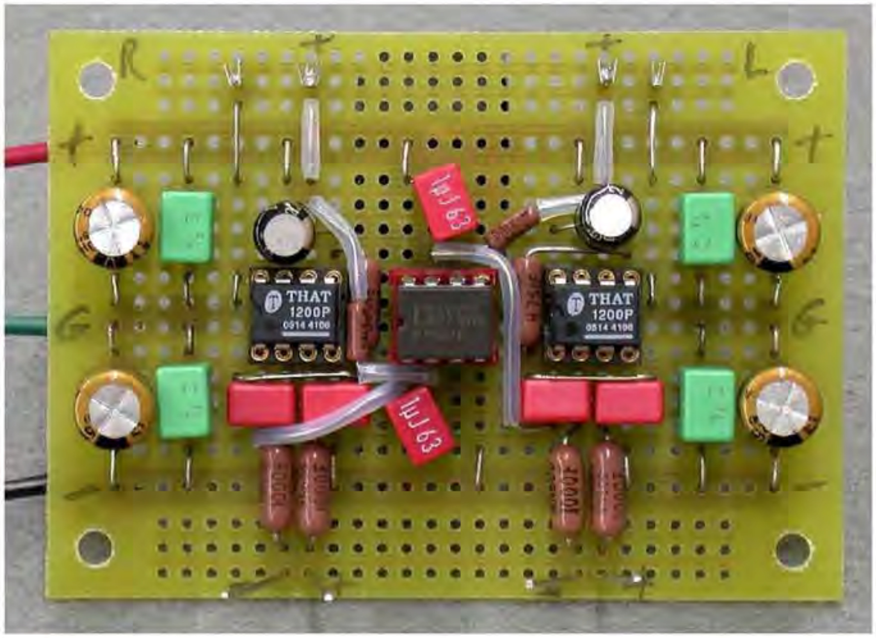

PHOTO 1: A stereo pair of THAT 1200 balanced line receivers with DC-servo control, built on DH Labs' D-4 Prototyping Board. The servo amplifier is a dual op amp mounted in the center of the board. Power supply bypass capacitors are located on each end of the board. Leads have been attached for testing.

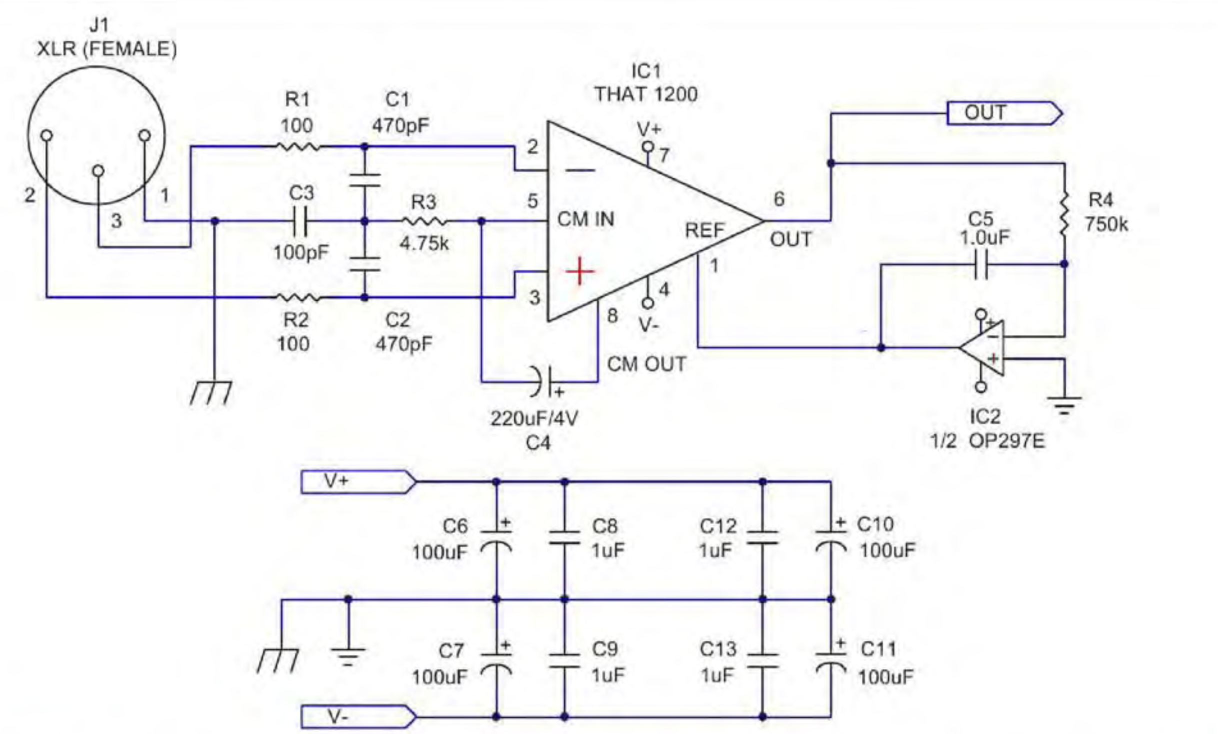

FIGURE 5: Complete circuit for the THAT 1200 balanced line receiver, including the input XLR connector and power supply bypassing. One channel is shown. Supply bypassing is sufficient for two channels.

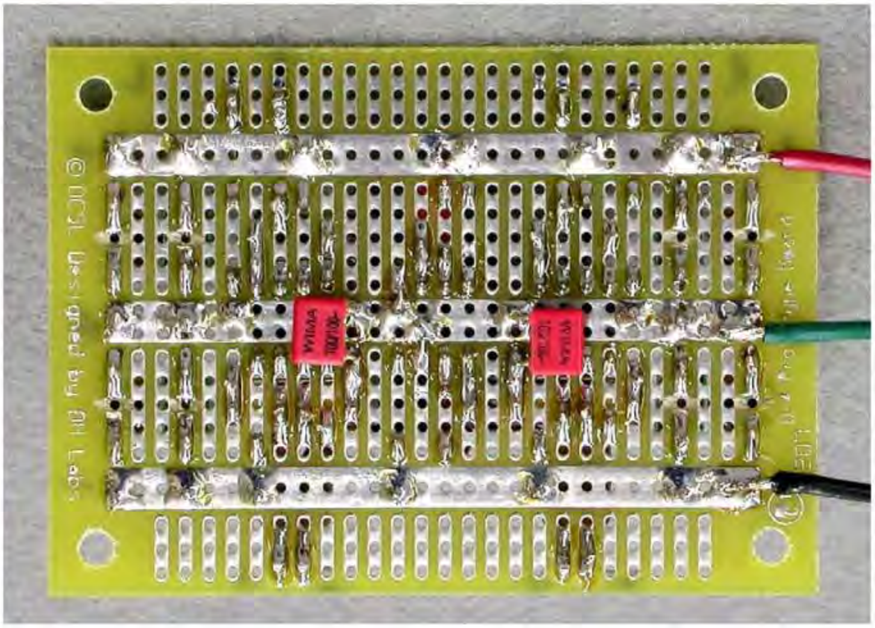

PHOTO 2: Bottom view of the THAT 1200 balanced line receiver board. The 100pF capacitors in the RFI filter, C3, are mounted on the bottom of the board.

Capacitors C1 through C3 are Wima FKP2-series polypropylenes. C3 is the only part that won't fit on the component side of the board. Mount it on the foil side, as shown in **Photo 2**. THAT Corp. specifies a 4V capacitor for bootstrap capacitor C4. The best type I've found is a Black Gate PK-series sold by Michael Percy Audio. Unfortunately, Rubycon no longer manufactures the Black Gate line, but Percy still has them in stock. When these are no longer available, I suggest substituting a Nichicon KZ-series. The smallest available is 25V, but it should still fit. Servo capacitor C5 is the same Wima or Roderstein polyester type used for power supply bypassing.

All resistors are Vishay/Dale RN60 metal film types except for R4, which

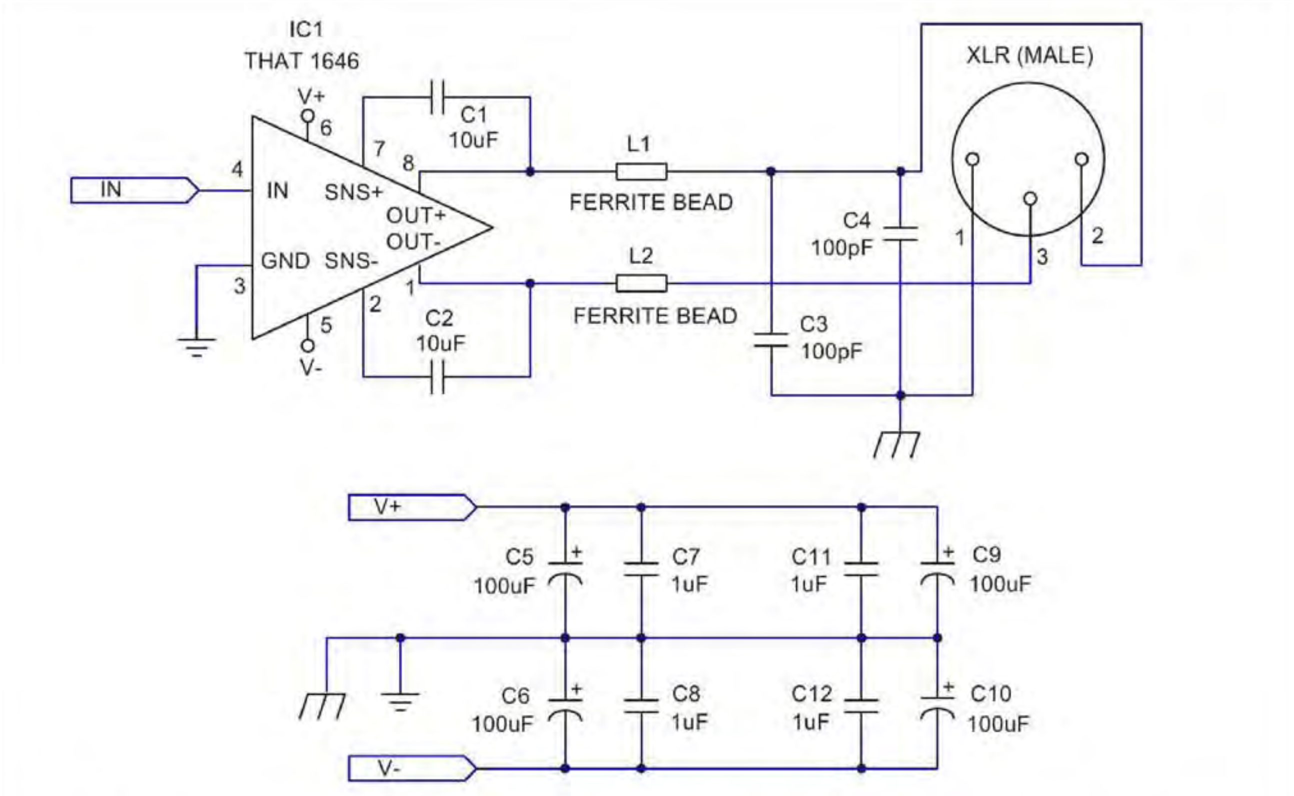

FIGURE 6: Complete circuit for the THAT 1646 balanced line driver, including the output XLR connector and power supply bypassing. One channel is shown. Supply bypassing is sufficient for two channels.

can be an RN55. I consider these to be good entry-level high-end audio resistors, particularly since all of the RN60 types still seem to be made with copper end caps (copper leads are specified, but the datasheet makes no mention of the end caps). Some of the RN55 types have been coming through with steel end caps, but the current through the 750k resistor is so low that the magnetic effects are of little consequence. Use sleeving over any leads that may touch others.

If your application demands more exotic resistors, by all means use them. The PR9372-series from Precision Resistive Products are designed specifically for audio applications and have gained an excellent reputation in high-end audio circles (www.prpinc.com/pdf/Audio_PR9372_Series.pdf). Parts Connexion carries a large assortment of these resistors at very reasonable prices. I prefer Neutrik XLR connectors with gold-plated contacts.

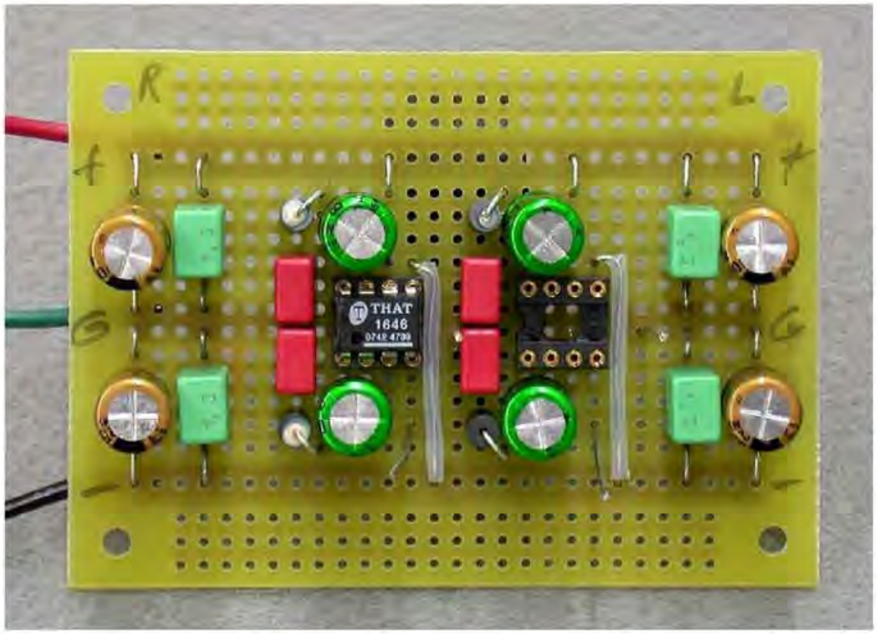

PHOTO 3: A stereo pair of THAT 1646 balanced line drivers built on DH Labs' D-4 prototyping board. For illustration, the 1646 chip has been removed from the machined-pin, gold-plated IC socket on the right. Power supply bypass capacitors are located on each end of the board. Leads have been attached for testing.

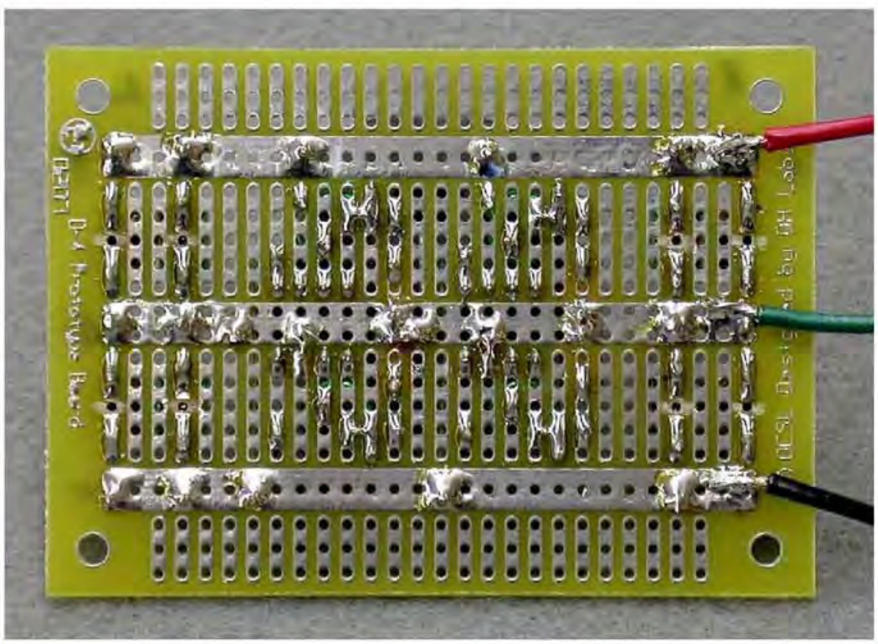

PHOTO 4: Bottom view of the THAT 1646 balanced line driver board. All parts fit on the component side of the board.

BALANCED LINE DRIVER

THAT's 1646 Balanced Line Driver is a very simple IC to implement, requiring very few external components. A complete schematic, including RFI protection and power supply bypassing, is shown in **Fig. 6**. A stereo pair will fit on the D4 prototyping board with room to spare; IC sockets are optional. Use the same types of capacitors for supply bypassing that I recommended for the 1200. Locate them on either end of the board, cutting the PC traces under the capacitors, and installing jumpers to make connections to the supply rails and ground bus (**Photos 3** and **4**, which also show leads attached for testing).

The common-mode offset protection capacitors C1 and C2 ensure very low output common-mode voltage offset—typically ±3.5mV (maximum ±15mV, though most samples I've used were within the typical range). Without these capacitors, typical offset will be ±50mV, and can be as high as ±250mV; even ±50mV is unaccept

PARTS LISTS

THAT 1200 Line Receiver Board

(Quantities are for a 2-channel stereo board)
(1) DH Labs D4 Prototype Board (Old Colony Sound Lab #PCBD-4)
(2) THAT 1200, 1203, or 1206 Balanced Line Receiver IC (IC1; Mouser #887-1200P08-U, 887-1203P08-U or 887-1206P08-U; see text)
(1) Analog Devices OP297E Dual Op Amp (IC2; Digi-Key OP297EZ-ND) – or – OP270EZ (Digi-Key OP270EZ-ND)
(3) Mill-Max 8-Pin Low-Profile IC Socket (Mouser 575-113308)
(3) Aries 8-Pin DIP Header (Mouser 535-08-600-11; Optional – see text)
(4) Wima 470pF/100V, 5% Polypropylene Capacitor (C1, C2; Mouser 505-FKP2470/100/5)
(2) Wima 100pF/100V, 5% Polypropylene Capacitor (C3; Mouser 505-FKP2100/100/5)
(2) Black Gate PK-Series 220μF/4V Electrolytic Capacitor or Nichicon KZ-Series 220μF/25V Electrolytic Capacitor (C4; M. Percy)
(4) 100Ω, 1% Vishay/Dale Metal Film Resistor (R1, R2; Mouser 71-RN60D-F-100)
(2) 4.75k, 1% Vishay/Dale Metal Film Resistor (R3; Mouser 71-RN60D-F-4.75k)
(2) 750k, 1% Vishay/Dale Metal Film Resistor (R4; Mouser 71-RN55D-F-750k)
(4) Nichicon 100μF /25V FG-series Electrolytic Capacitor (C6-7, C10-11); M. Percy)
(5) Wima – or – Roederstein 1.0μF /63V Polyester Capacitor (C5, C8-9, C12-13; Mouser #505-MKS21.0/63/5 or #75-MKT1817510064)
(2) Neutrik NC3FD-L-B-1 Female, Chassis-mount XLR Connector (Mouser 568-NC3FD-L-1-B)

THAT 1646 Line Driver Board

(Quantities are for a 2-channel stereo board)
(1) DH Labs D4 Prototype Board (Old Colony #PCBD-4)
(2) THAT 1646 Balanced Line Driver IC (IC1; Mouser 887-1646P08-U)
(2) Mill-Max 8-Pin Low-Profile IC Socket (Mouser 575-113308)
(2) Aries 8-Pin DIP Header (Mouser 535-08-600-11; Optional – see text)
(4) Nichicon ES-series 10μF/50V Non-Polar Electrolytic Capacitor (C1, C2; M. Percy)
(4) Wima 100pF/100V, 5% Polypropylene Capacitor (C3, C4; Mouser 505-FKP2100/100/5)
(4) Murata Ferrite Bead Inductor (L1, L2; Mouser 81-BL01RN1A1D2B)
(4) Nichicon 100μF/25V FG-series Electrolytic Capacitor (C5-6, C9-10); M. Percy)
(4) Wima or Roederstein 1.0μF /63V Polyester Capacitor (C7-8, C11-12; Mouser #505-MKS21.0/63/5 or #75-MKT1817510064)
(2) Neutrik NC3MD-L-B-1 Male, Chassis-mount XLR Connector (Mouser 568-NC3MD-L-1-B)

VENDORS

Digi-Key Corporation
701 Brooks Avenue South
Thief River Falls, MN 56701
800-344-4539 or 218-681-6674
www.digikey.com

Michael Percy Audio
11731 Stillwater Creek Road
Nevada City, CA 95959
530-470-8650
www.percyaudio.com
mpercy@pacbell.net

Mouser Electronics
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Mansfield, TX 76063
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Old Colony Sound Lab
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Peterborough, NH 03458-0876
888-924-9465 or 603-924-9464
www.audioxpress.com (click on "Shopping")
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905-631-5777 (FAX)
www.partsconnexion.com

ably high for most applications. Use very high quality non-polar electrolytic capacitors, such as the Nichicon ES series. A film capacitor would be ideal, but the physical size of a 10μF film cap, at any readily-available voltage, makes this an impractical choice. Use the Murata ferrite bead inductors and Wima FKP-2-series polypropylene capacitors for RFI protection, L1, L2, C3, and C4.

The 1646 incorporates a proprietary internal protection scheme which should be sufficient for the levels of ESD encountered in most normal field situations. It is possible, however, for any balanced output to be accidentally plugged into a microphone preamp with +48V phantom power. The 1646 datasheet shows an external scheme for protecting the chip from such hazards. If you think you need it, by all means use it.

I mentioned power supplies only in passing at the beginning of this article. Suffice to say, these chips will give their best performance with high-quality, regulated power supplies.

The THAT 1200 and 1646 balanced line receivers and drivers represent the state-of-the-art in active, IC-based balanced inputs and outputs. With a proper operating environment they offer excellent performance at very reasonable cost. *aX*

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3. Jung, Walt, "Audio Preamplifiers, Line Drivers and Line Receivers," *System Applications Guide*, Analog Devices, Inc., 1993, pp. 8-93 to 8-97.