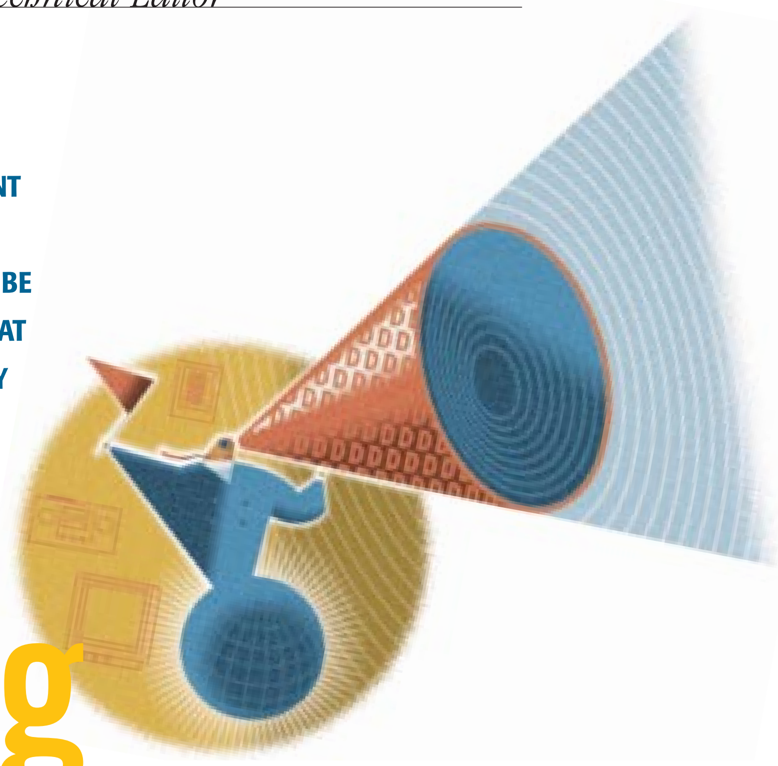
 By Joshua Israelsohn, Technical Editor

YOU NEEDN'T LOOK FURTHER THAN CLASS D AUDIO AMPLIFIERS TO FIND A TECHNOLOGY WHOSE PROPONENTS TEND TOWARD EXUBERANT ADVOCACY AT BEST AND BALD-FACED HYPE AT WORST. ALTHOUGH THESE ICs ARE NOT YET THE BE ALL AND END ALL OF AUDIO AMPLIFICATION THAT THEIR MAKERS WOULD HAVE YOU BELIEVE, THEY ARE REMARKABLY GOOD—CERTAINLY MORE LISTENABLE THAN THE CLAMOR ABOUT THEM.



Listening to Class D

If you've ever perceived the slightest gap between the marketing and the metal in an electronic product, you ain't seen nothin' yet, as the song goes, until you read the promotional materials that accompany audio products. You'd think that the engineering community would have dispensed with the hype long ago. Electronic audio amplifiers, starting with Lee De Forest's triode vacuum tube, are nearly a century old. Even the relatively newfangled Class D is no longer a novelty. From an instrumentation perspective, audio bandwidths aren't challenging, the amplitudes are generous, and the impedances look kindly upon probing. With other products, you would just compare data sheets and be done with it.

But the data sheets and supporting documentation available for audio amplifiers often say precious little that forecasts how the products will perform in their primary role: driving speakers in various environments for humans to enjoy. Lest this sound like an appeal to subjectivism, let me assure you that it is not (see **sidebar** "The trouble with subjectivists"). Unfortunately, data sheets typically don't reveal the kind of measurable detail that allows for meaningful comparisons. This state of affairs is not entirely the fault of amplifier manufacturers. They are not, after all, trying to drape their wares behind a veil of secrecy or false promises. However, no industry-wide consensus exists on what comprises a measurement set that correlates to the listening experience: a surprising and disappointing fact considering the conceptually simple task that audio amplifiers have to perform and how long they have been doing it. The problem is not

Image by Daniel Guidera

that we can hear things that are immeasurable—the fatally flawed claim of the subjectivists—but rather that we haven't a defined set of measurements that adequately predict what we hear. Though the fi is as hi as an elephant's eye, there's no "fidelimeter" with which to measure it.

Data sheets reliably contain harmonic-distortion measurements and noise measurements, often combined into a single number THD+N (total harmonic distortion plus noise) taken at 1 kHz at full rated power. Amplifier makers also report the frequency response with a tolerance. Beyond that, little consistency exists.

In analog amplifiers, harmonic distortion rises with frequency, partly because some of the generating mechanisms are excited more at high frequencies than lower and partly because the available loop gain falls with frequency. Single-frequency harmonic-distortion measurements make for easy—but not necessarily informative—numeric comparisons, because they don't indicate the rising distortion curve that can differ significantly from design to design.

A growing and welcome trend among amplifier manufacturers is to include a THD plot that sweeps a power range at a nominal frequency, usually 1 kHz, and a plot that sweeps a spectrum at a fixed power level. Be aware, however, that some manufacturers use brick-wall filters for the spectral plots, which hides their amplifiers' performance at high frequencies.

AT A GLANCE

- ▷ A good number of claims and comparisons–some true and others dubious–often accompany Class D-amplifier ICs. Read the numbers carefully before jumping to conclusions about any amplifier implementation.
- ▷ Class D amplifiers compress rather than clip when their power supplies reach their current limit, which can save speakers a lot of abuse.
- ▷ You can still cause Class D amplifiers to clip if you overdrive their analog inputs or, in the case of amplifiers that take digital-audio inputs, allow signal-processing algorithms to hit the code ceiling.
- ▷ Reference designs are great starting points, but they are not all things to all applications. Plan on working the details.

The argument for this practice is that the filter more closely approximates the range of human auditory perception than would an unencumbered measurement. In cases where a THD spectral sweep is the only evidence of an amplifier's nonlinearities, the information could reveal performance distinctions well within the traditional audio band. Filtering the data does not add to our understanding.

Those who would like to assess an amplifier's performance by objective means welcome another trend in harmonic-distortion measurement: FFT plots that reveal the relative amplitudes of the ordered harmonics. Still, few manufacturers are providing such detail, preferring the lumped specification, which they can quickly test during production and which results in a simple numeric comparison. But the human response to harmonic distortion is not constant with order (**Reference 1**). Even small amounts of higher order harmonics are so offensive that research suggests that you should weight your assessment of distortion products by $n^2/4$ (**Reference 2**). This approach would result in factors of unity, 2.25, 4, 6.25, and 9 for the second through sixth harmonics, respectively.

Class D output stages have distortion mechanisms of their own. One example is the distortion that the output devices' finite rise and fall times cause. The output transistor's limited switching dynamics cause the otherwise rectangular output of the PWM driver to appear trapezoidal. At extremes of duty cycle, the trapezoid wave rise- and fall-time "skirts" meet, and the waveform decays further to a triangle that never reaches its target amplitude. A number of Class D designs employ schemes to compensate for the effects of pulse-shape errors. But the compensation mechanisms don't measure the error in real time, as is the case in a feedback amplifier, so corrections are

THE TROUBLE WITH SUBJECTIVISTS

In his essay on the subject, Douglas Self traces subjectivism in the audio industry to 1977 (**Reference A**). The proponents of subjectivism dismiss objective measures of performance and prefer to evaluate audio devices based solely on their listening experience. They claim, among other things, that cables are directional–a remarkable trick considering that audio signals themselves are ac.

Yet one amplifier vendor sent along with his demo board a heavy-duty cable with gold-plated RCA plugs on either end, one marked "source" and the other marked "amplifier." The cable itself was marked with little arrows along its length. When I mentioned these markings to the director of engineering for that product line, he blushed and assured me that the company put no credence in claims of directional preference.

Another subjectivist claim is that tone controls cause audible deterioration even when set flat. As Self points out, subjectivists often blame phase shift for this deterioration, but a circuit with a flat response cannot contribute a phase shift unless it is an allpass network, and tone controls aren't. I find that common tone controls are largely useless for correcting deficiencies in an acoustic system of a speaker and a room because the acoustic deficiency and the tone control's center frequency are most likely not going to coincide. But that lack of coincidence does not suggest that sonic evil is associated with the presence of a tone control. Carefully adjusted one-sixth-octave-graphic- or parametric-equalizers, essentially tunable tone controls, successfully correct room responses in many carefully designed listening environments.

If the subjectivists get any traction at all, it may be because the design community depends heavily on harmonic-distortion measurements, which serve as poor predictors of an amplifier's subjective deficiencies. Harmonic-distortion measurements give experienced designers information that often leads to the root cause of the nonlinearity. Other distortion measures give less information to the designer but correlate better with subjective assessments of amplifier performance.

REFERENCE

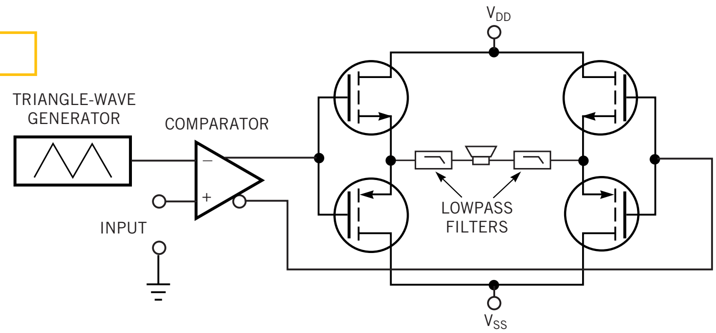
A. Self, Douglas, *Audio Power Amplifier Design Handbook*, Newnes, 1996, pg 5 to 21.

limited to the extent that a designer can predict the errors across the entire range of signal and operating conditions. Correction schemes succeed to varying degrees and within varying condition limits. As in the case of analog amplifiers, you wouldn't need to be too skeptical to wonder whether a single-point distortion specification adequately describes such a circuit's performance.

Comparisons among competing amplifier designs might also benefit from other objective measures, such as intermodulation distortion and phase response. Some Class D-amplifier data sheets mention intermodulation distortion, though often without specifying the test conditions. The data sheets do not specify phase response, however, for any of the amplifiers in this article. With no industry-accepted norm for specifying audio amplifiers, seekers of honest objective comparisons are more often than not left with unanswered questions. End users, even in professional-audio applications, such as recording-studio monitoring or concert-sound reinforcement, rarely have access to the instrumentation you need to answer these questions. With no other plausible course of action, they invariably resort to listening tests as the final arbiter.

Last, you need to determine how good is good enough. This criterion is a moving target as the recording quality of source materials and their media continue to improve. You can make that assessment only if you can "calibrate" your subjective experience to objective measurements.

Figure 1



A basic PWM modulator compares the amplitude of an incoming audio signal to a high-frequency triangle wave. The output pulse train—often mistaken for a digital signal—is in fact an analog representation of the original waveform.

IF IT QUACKS LIKE A DAC

A number of proponents of Class D amplifiers argue that an "all-digital amplifier" is desirable in and of itself, as if to equate digital with good and, by implication, analog with bad. Class D amplifiers avoid the errors that plague DACs, goes the argument. One company goes so far as to refer to its digital-input/Class D-output amplifier as a "digital-to-digital converter."

The biggest problem with this argument is that PWM, the output of all Class D amplifiers, is not a digital signal; it's analog. The fact that Class D outputs ideally switch between discrete voltages makes them no less analog. Perturbations in either the voltage or the timing of a PWM output can appear at the output, and no threshold exists below which this

SEVEN TIPS FOR LISTENING

In more than two decades of audio-production work, I've auditioned a lot of gear whose data sheets, like those for audio amplifiers, give at best an incomplete impression of the product's sonic performance. In that time, I've discovered a number of ways to ease the job of assessing gear and avoid the subjectivist's trap of generating utterly irreproducible observations. The goal is not to discern minuscule differences but rather to gain an understanding of the sonic character of a piece of equipment. A good assessment is one that stands the test of multiple listenings and listeners. If your conclusions about a piece of equipment stand up after 200 hours or so of use, they were good assessments. On the other hand, if you're in a room with other people, and you're the only one who can hear something, it is, by definition, "in your head."

Here are seven ideas for getting the most from critical listening:

- Acknowledge and debunk your prejudices. The object of the game is to learn something, not to confirm your preconceived notions.
- Listening is like swimming: You get better with practice, and you shouldn't go in alone. A good listening buddy can keep you from getting in over your head. If you both come up with the same *unprompted* assessment, it most likely contains useful information. You don't need "golden ears," but you have to be willing to listen carefully and critically.
- Don't worry about minutiae. You're never looking for tiny differences between two things. If you need repeated double-blind test trials to differentiate A from B, then it is probably a distinction not worth making.
- Turn only one knob at a time. Everything about your listening environment should be as constant as possible throughout the assessment. Familiarize yourself with the room and monitors you use in any evaluation. Ensure that the room is free of distractions.
- Choose well-recorded, familiar source material. Listening to something for the first time is in effect turning a knob. Sparse arrangements are more useful than spectrally dense "walls of sound" because they don't mask distortions. For the same reason, clean, crisp, transparent recordings of acoustic instruments are more telling than tracks with a lot of synthesizers or effects.
- Watch your monitoring levels. Listening fatigue can set in within 10 minutes or so at elevated levels. You can work effectively for far longer if you keep the levels down.
- Carefully note what you think you're hearing and come back to the same setup and retest your perceptions. Listening tests are difficult to control, and the greatest and most dangerous variable is you. Use whatever analytic equipment is available. The most useful assessment is one you can correlate to a measurement.

situation is not true. On the other hand, if you disturb either the output voltage or timing of, say, the SPDIF (Sony/Philips digital-interface) data stream from your CD player's digital output, those disturbances do not affect the data as long as their magnitudes are within the system's voltage and timing tolerances.

You can make a simple pulse-width modulator with a comparator and a triangle generator (**Figure 1**). Although this circuit's transfer function is nonlinear, it is an analog circuit just the same. The modulator produces a pulse train with continuously variable pulse widths from an analog input. PWM generators that take digitally encoded inputs, such as PCM (pulse-code-modulated) streams, are DACs, but rather than representing their outputs as discrete points on a voltage continuum, they represent their outputs as discrete transitions on a time continuum. Digital goes in; analog goes out.

The positioning and marketing do little to move the discussion of Class D amplifiers' merits, which are numerous. The benefits of Class D amplifiers' efficiency extends to smaller power supplies, little or no heat-sinking, lower product weight, and longer per-charge life for battery-powered designs. Early designs may have lacked sonic performance, but the technology's promise wasn't lost on an industry eager for low-cost, low-power *everything*.

The five amplifiers of 2 to 50W that I used in listening sessions for this article produce more than adequate sound quality for a variety of high-volume applications. Several of the chips take digital inputs; others take analog-audio feeds. Of those that take digital-audio inputs, some offer a range of auxiliary signal-processing functions that go well beyond those of a power amplifier. In some applications, these integrated facilities greatly simplify the system design. In others, they add little value. As is often the case, there's no right part for everybody, but a number of good parts are available to choose from.

As for sonic performance, the demo amplifiers as a group performed well. I might have reservations about using some in the most demanding applications, such as primary studio-monitoring. However, for the largest segment of the consumer market for which these parts were designed, they are more than adequate.

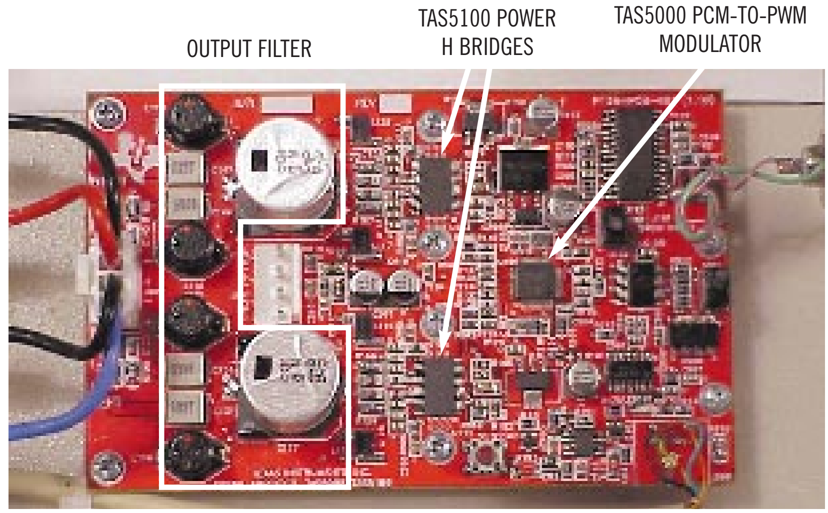


Figure 2 **A little amplifier that can: TI's TAS5000/5100 demo packs signal processing and amplification onto less than 15 in.[2] Of that, about one-third is filter.**

All of the manufacturers submitting parts for listening tests provided demo boards constituting their reference designs. The consensus that developed after speaking to three of the design teams involved suggested that many customers would likely use the reference designs as they were, adding their own value elsewhere.

In addition to providing basic amplification, demo boards taking SPDIF inputs must also provide a means for controlling the signal level because the incoming data stream knows only one level: full-blast. Because making a simple volume control requires a certain amount of digital-processing capability, most also include the rudimentary functions available on most stereo preamplifiers—a source selector and basic treble and bass controls.

For the most part, all the demos produced good sound quality, particularly when you account for the fact that their manufacturers target these devices toward the inexpensive end of the home-entertainment market, not studio monitoring. In keeping with this idea, I listened to each amplifier through a pair of Paradigm Mini Monitors, which serve as the "small speakers" at the Cove Arts postproduction suite where the listening sessions took place. The Paradigms have proved reasonably accurate for near-field monitoring during mixing sessions and the resulting mixes translated well to both higher and lower quality speakers elsewhere. They are the younger siblings to the main monitors I use for mixing at Cove Arts, and after several hundred hours with both, I felt comfortable that the selection of monitor was not itself "turning a knob" (see **sidebar** "Seven tips for listening").

This monitoring issue was apparent when I visited three of the design groups, each of which had listening rooms equipped with speakers I'd never heard. Under those circumstances, it was difficult to attribute what I heard to its source.

Overall, the listening sessions were encouraging. Any of these amplifiers would suffice for casual listening, though each brings its own set of features, so you may find that you have your own preferences. The differences I noted occurred in amplifier controls, noise, and the quality of stereo imaging. None of the Class D amplifiers performed poorly or even marginally.

THE DANISH INVASION

The TAS5000/5100 demo from Texas Instruments measures 4.75×3 in., excluding the detachable control pad. These measurements make it one of the smallest of the demos in the tens-of-watts power class. The TAS5000 PWM processor operates in a four-chip set for stereo applications. The chip set starts with one of several signal processors that interface to digital-audio sources, such as SPDIF, IEEE1394, and USB; strip digital subcodes and decode surround-sound signals; implement the volume and tone or equalization functions; and output one or more PCM streams to the PWM processors. The TAS5000 converts the PCM-encoded audio to PWM-control signals, which drive a pair of TAS5100 H-bridge output stages. The design allows for easy expansion to multichannel architectures, such as for Dolby 5.1 surround sound using a small set of building blocks. The amplifier is rated at 0.08% THD at 1 kHz into 6Ω.

The $3.41 (1000) TAS5000 PWM processor in a TQFP-48 package and the two $3.15 (1000) TAS5100 power H bridges each in HTSSOP-32 thermally enhanced packages, constitute a small minority of the board area, despite their ability to deliver 30W per channel to the speakers (**Figure 2**). The parts use technology that a Danish design team at Toccata, a company TI acquired last year, developed. TacT markets the high end of the Toccata technology in the form of the M1 power amplifier, which weighs roughly 60 lbs, costs thousands of dollars, and targets "golden-eared" home-stereo enthusiasts. The TAS5000/5100 design, which costs and weighs only a small fraction of the M1, is far less ambitious and more practical for the broad consumer market. The LC output filter, more or less typical of the Class Ds, takes up somewhat less than a third of the pc-board area. The TI demo dissipates about 3.7W while idle with valid SPDIF code running and with the volume control at its minimum. When the incoming SPDIF stream stops, the amplifier's dissipation drops to about 1.8W. The unit came with an RCA jack input for standard copper SPDIF inputs.

Like its brethren Class D amplifiers, the TAS5000/5100 compresses rather than clips when their power supplies reach their limit. This behavior might be worrisome in a studio environment in which any departure of the monitored signal from the program feed is cause for alarm. But in almost every other application, volume compression is a desirable feature that guards against the violent reaction speakers have to the ungraceful application of dc due to clipping. Should the power supply fall from its nominal 20V, the TAS5000/5100 continues to operate, albeit at lower volume, until it mutes its outputs at about 10V. Muting and demuting are fairly clean, despite the fact that they occur due to out-of-spec operating conditions.

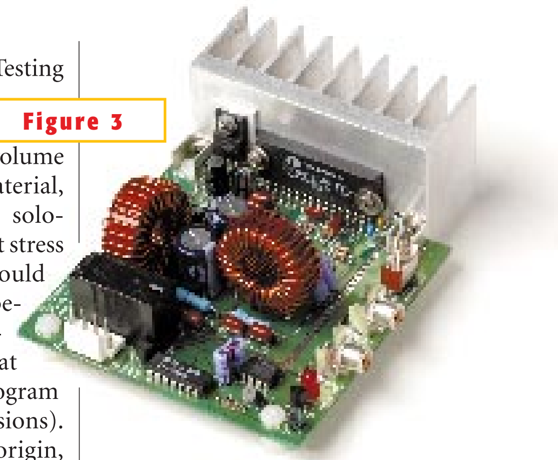
One disturbing attribute of the TI demo was that it distorted audibly when the bass control was set to boost. Testing with a 100-Hz tone revealed readily audible distortion with any amount of base boost at any volume level. With musical-program material, such as Tommy Emmanuel's solo-acoustic-guitar tracks, that did not stress the bottom of the spectrum, I could raise the bass level about halfway before the program became distorted audibly (see this article at www.ednmag.com for a list of program material used in the listening sessions). Though I could not confirm its origin, this behavior's independence of level suggests that it is a correctable error in the tone-control algorithm and thus needn't count against the PWM processor or H-bridge components.

My other criticism of this amplifier is that, on power-up, it assumes a level that is shy of but well on its way to full-blast. This behavior was immediately apparent when I power-cycled the system—amplifier and CD player—together, as you might do in either an all-in-one or component stereo system. Most CD players automatically play if you turn them on with a CD installed. Such was the case with this system, and with a pair of efficient near-field monitors, the results were jarringly loud. Although software people would probably call this behavior a "feature," it is *not* a TI exclusive. Both demos that use increment/decrement switches for their volume controls display the same behavior. Again, you could reprogram the processors in question to power up at a more appropriate level, such as fully attenuated or well on the way to that extreme. OEMs can add nonvolatile memory to maintain volume settings through a power cycle and more properly emulate what the lowly potentiometer has done for years. More than anything, this behavior points out that OEM designers should be prepared to do a certain amount of detail work and not plan on simply copying the reference design lock, stock, and initialization code.

WITHIN THE AUSTIN CITY LIMITS

Cirrus Logic's $3.95 (10,000) CS44210 is a Class D controller that mates with a pair of International Rectifier IRCS8001 bridge-driver ICs. The 44210 includes three serial digital-audio inputs with a multiplexer and host interface in a TSSOP-24. The $3.55 (10,000) IRCS8001 bridge-drivers, available in SOIC-16 packages each drive either four $1.08 (10,000) IRCS8101 MOSFETs in a full-bridge configuration, or two $1.45 (10,000) IRCS8102 MOSFETs in a half-bridge. International Rectifier mounts both MOSFET models in D-Pak packages. Cirrus's CS44210 can drive 50W into 8Ω speakers in either configuration. The amplifier's THD is listed at 0.01% at 1 kHz at 1W; 0.1% at full power.

Figure 3



Just power, filter, and go: Tripath's TA2022 is a single-chip Class D amplifier that takes analog inputs.

The CS44210 demo amplifier fits onto a 6.5×4.5-in. board, which is divided into a 2.5×4.5-in. input interface-and-control section and a 4×4.5-in. power-conditioning-and-amplifier section. The input-and-control section accommodates both copper and fiber-optic SPDIF inputs. I prefer fiber because it galvanically isolates the source and amplifier, eliminating any concerns about ground loops without resorting to pulse transformers. Additionally, I've used the same plastic fiber and Toslink transmitters and detectors in multichannel audio for data densities at least four times greater than those that SPDIF requires, so data-signal integrity through the link isn't a concern, even on fairly long feeds. The input-and-control section also provides increment/decrement switches to control volume, treble, and bass and also provides three other pushbuttons for shutdown, mute, and reset. The amplifier came with a separate power supply that provided ±35V for the output bridges and 5V for logic.

The Cirrus/International Rectifier combination works well. The horn section on Charles Brown's *Just a Lucky So and So* sounded slightly tighter, smoother, and better articulated than on most of the demo amplifiers. The Cirrus amplifier also produced among the best stereo imaging of the amplifiers in this study. The issue here is not of stereo separation or the apparent width of the stereo field, but rather the clarity of a source's location within it. If you listen to, say, a vocal recorded with a single microphone, you should be able to locate the position of the sound source in the stereo field without ambiguity. I've also noticed that the quality of the stereo image coincides with the relative clarity and transparency of sounds in the upper midrange and top end. Although I've not taken the time to tie the two to a measurable parameter, I've observed this characteristic frequently enough to form a hypothesis for such a tie.

Like all the amplifiers that took SPDIF inputs directly, this amplifier's mute function followed the acquisition or loss of a valid input signal. The manufacturer warns that you'll hear a "small click" in the speakers during these transitions, and, true to its word, a click is what you get. This small flaw is not fatal by any stretch, but it is mildly annoying after several hours to have the amplifier punctuate in living stereo every manipulation of the CD player. Further, because this behavior was unique to the CS44210 demo, it is likely that Cirrus could eliminate it in a future spin.

The Cirrus amplifier demutes a bit late when it detects incoming SPDIF code. In most cases, this behavior is innocuous and might have entirely escaped my notice because most CDs have lead-time between each cut's start marker and the actual program start. But it happened that I picked a certain Lyle Lovett track with a particularly tight cue, and the amplifier cut the initial "Hello" from Lovett's *Hello it's Me*.

When powered, like the TI demo, the Cirrus also assumes a level that is a bit aggressive for those listening with efficient monitors. The bass and treble controls were subtler than most I've heard, which you should read as a compliment.

NEW ENGLAND-STYLE CLASS D

Apogee Technology takes Class D in a slightly different direction for its 35W-per-channel amplifier. The \$6.98 (1000) DDX2000/2060 chip set, packaged in a PQFP-44 and a thermally enhanced 36-pin power package, forms a ternary, not binary, amplifier. This distinction allows Apogee to improve efficiency while performing a first-order correction for an important distortion mechanism: the finite rise and fall times of the switching devices (**references 3** and **4**).

The DDX2000 PWM controller maintains a minimum pulse width, so that the trapezoidal pulse train can never decay to triangular. An opposite-polarity, minimum-width compensation pulse follows every signal pulse, which gives a first-order correction to the rise- and fall-time skirts and allows the controller to generate equivalent pulses shorter than the minimum width. For example, if the minimum width is k, then the DDX2000 can generate a short pulse, δ, by generating a k+δ followed by a k-wide compensation pulse.

Between pulses, the speaker is not driven. The third state of the ternary drive connects both speaker terminals to ground and leaves both outputs of the amplifier unloaded. At idle, the Apogee amplifier dissipates 1.6W, less than half that of the TI amplifier. Such comparisons are of dubious value because they ignore substantial differences in digital-signal-processing capability. On the other hand, it's unlikely that TI is burning over a watt and a half just to include equalization.

Like the Cirrus demo, the Apogee Technology amplifier accepts either copper or Toslink fiber-optic connections for its SPDIF input. As in the previous case, I used the Toslink. Unlike the previous amplifiers, the DDX2000/2060 demo has two potentiometers—one for volume and one with a center detent for balance—which an eight-pin Microchip PIC processor reads. The volume potentiometer provides smooth control, reminiscent of an analog amplifier and, best of all, has no trouble remembering its position through power cycles.

The DDX2000 can adjust the digital-audio signal over −82.5 to +12 dB. It includes an anticlipping function that guards against combinations of gain and input signal that would cause the amplifier to run out of codes. The Apogee's output had somewhat-less-audible hiss than the Cirrus but didn't match the Cirrus' stereo-image quality.

CLASS D FOR ANALOG

Unlike the previous three amplifiers, the TA2022, a \$16.00 (1000) stereo integrated modulator and output stage from Tripath takes analog inputs. The company refers to its design as "Class T." Tripath is not the first company to identify its amplifier with a novel class designation. It makes the distinction to reflect the fact that its modulator uses a spread-spectrum switching pattern rather than a fixed switching frequency. Since amplifier operating class, even for Class D, describes how the output devices are biased and driven, not the spectral nature of that drive, this looks like a Class D amplifier to me. Tricked up, perhaps, but Class D just the same.

The TA2022 has a THD of 0.015% to

FOR MORE INFORMATION...

For more information on products such as those discussed in this article, go to www.ednmag.com. When you contact any of the following manufacturers directly, please let them know you read about their products in *EDN*.

Apogee Technology
1-781-440-9528
www.apogeeddx.com
Enter No. 315

Cirrus Logic
1-512-445-7222
www.cirrus.com
Enter No. 316

Maxim Integrated Products
1-408-737-7600
www.maxim-ic.com
Enter No. 317

Texas Instruments
1-800-336-5236
www.ti.com
Enter No. 318

Tripath
1-408-567-3000
www.tripath.com
Enter No. 319

OTHER COMPANIES MENTIONED IN THIS ARTICLE

International Rectifier
1-310-252-7105
www.irf.com

Mackie Designs
1-425-487-4333
www.mackie.com

Microchip
1-480-792-7668
www.microchip.com

Paradigm
1-905-850-2889
www.paradigm.ca

TacT
1-201-440-9300
www.tactaudio.com

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45W in 8Ω rising to 0.1% at 60W. Due to power-supply constraints, the demo board operated at ±25V during the listening sessions for this article, which limited the output power to 25W per side into 8Ω at 0.015% THD. At that voltage, the idle dissipation measured 3.4W. The demo amplifier fits on a 3.5×3.75-in. board, not including the heat sink (**Figure 3**). Tripath packages this amplifier in an SSIP-32 plastic package with a 20°/W junction-to-ambient thermal resistance. To meet its top rating of 80W per channel into 4Ω at 85% efficiency, the manufacturer recommends a heat-sink-to-ambient thermal resistance of 1.63°/W or better, allowing 0.2°/W for the interface case-to-heat-sink thermal resistance.

Because the TA2022 demo amplifier takes an analog input but has no onboard level control, I drove its inputs from the control-room outputs of a Mackie 1604VLZPRO. With the CD player's analog outputs brought in through the two-track tape return, this signal path was the shortest that I could conveniently arrange that provided a level control.

During the listening sessions, the TA2022 was quiet and produced an excellent stereo image. The amplifier rendered the drum solo on Tommy Flanagan's *Verdandi*, an excellent recording of Lewis Nash's brushwork, with greater clarity and transparency than many of the other amplifiers. The experience was probably more fun than you're supposed to have on the job, so I postponed the second and third listening to after-hours.

A LITTLE GOES A LONG WAY

The smallest of the Class D amplifiers offered for this project came from Maxim. The MAX4297 can deliver 2W per channel into 4Ω albeit at about 2% THD. The 1-kHz, 1W distortion spec is typically 0.4%. This spec may sound like "flea power" after the 30 to 50W brutes, but for PDAs (personal digital assistants), laptops, telephones, and a host of other portable applications that lack full-range speakers, a watt can be a lot. Even when driving the Paradigm monitors, the 4297 could reach volume levels that would suffice for, say, a small desktop stereo or powered PC speakers. With a 2.7 to 5.5V power-supply range, the maximum open-circuit idle dissipation is only 40 mW; that figure drops to 75 μW in shutdown mode.

The demo circuit covers less than 4 sq in., of which the input and output connectors and the volume control occupy half. The $2.45 (1000) MAX4297 itself is packaged in an SSOP-24. A mono version, the $1.45 (1000) MAX4295 is available in a QSOP-16. A pair of logic inputs allows you to select the PWM frequency from 125 kHz, 250 kHz, 500 kHz, and 1 MHz. Though the data sheet suggests that the amplifier gives its best overall performance at 125 or 250 kHz, the setup I used worked best at 1 MHz and worst at 125 kHz where it tended to distort at lower volumes than at the higher frequency. Results may have been different if I had auditioned the amplifier with something better approximating its intended load.

Not surprisingly, the listening experience could not compete with that of the other amplifiers, but it was good enough for around the office. In a laptop, PDA, or Walkman-type design, the transducers limit the fidelity, so the MAX4295's size and idle power could be a good fit.□

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References

1. Self, Douglas, *Audio power amplifier design handbook*, Newnes, 1996.

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Author's biography

When not serving as Technical Editor for EDN, *Joshua Israelsohn produces on-location and studio-based digital recordings for Cove Arts, an independent audio-production house. He has more than two decades of production experience and 15 years in design. He is never bald-faced.*

The sources

While preparing this article, I spent tens of hours listening to the subject amplifiers. The quality of information gleaned from your listening experience depends on the qualities of the source materials you select. I chose music from a wide range of styles and orchestrations. I wanted to ensure that I gave the subject amplifiers every opportunity to expose their traits, and I know of no single music that is best at doing that.

For example, the voice tends to expose imaging more reliably than, say, a piano track. This phenomenon may be due to the fact that the voice is monophonic—in this case, as opposed to polyphonic–and less percussive. It may also be due to the fact that a solo voice is virtually always recorded with a single microphone, so no ambiguity should exist in its position within the stereo field as is often the case in setups with multiple microphones. Brass instruments, on the other hand, have rich harmonic series and, if well-recorded in the first place, can expose nonlinearities in the upper midrange and top end of the spectrum.

Another reason for choosing a range of musical styles is that it helps ward off listener fatigue. In my experience, if you listen to the same music or even the same type of music over an extended period of time, you will likely spend less time listening critically than if you change styles regularly. The material I selected ranges from choral music to jazz to rock and a number of artists, such as Lyle Lovett and Patty Larkin, whose musical range, thankfully, defies categorization (**Table A**).

All of the selections have a few things in common: They are all well-recorded works, and they are all recordings of people with excellent musical skill. This point about skill is important, because the quality of the sound starts with the musician and, in a perfect world, would end there—an oft-forgotten fact in the razzle-dazzle world of technology merchandizing. The selected pieces tend to have uncluttered arrangements with little “spectral filler,” such as synthesized string sections. They are works and artists with which I am familiar. Of the titles listed, only one, the Tommy Emmanuel, has been in the collection for less than a year. Only three—Chris Isaak, Johnny A, and Morphene—are performers I’ve not heard in concert.

When planning your own listening, develop clear criteria for choosing your source material. Choosing material you like helps, but don’t let your musical tastes steer you toward selections that don’t suit your goals. For example, I had planned on using a cut or two from Pink Floyd’s *Dark Side of the Moon*. In the end, despite David Gilmore’s iconic guitar playing and Alan Parson’s pristine recording, the work did little to inform the study, and I dropped it from the list. I took only minutes to realize that the same sad fate awaited my entire John Coltrane collection and most of the other jazz recordings made before about 1980.

During my visits to three amplifier-design groups, I noticed that my selection criteria aligned closely with those the design engineers had employed. All three groups depended on cleanly recorded music, predominantly though not exclusively of acoustic instruments and voice, and a range of represented styles.

Table A: Selected music used in listening sessions

Artist	Cut Title	Composer	CD Title	Label	Release #
A, Johnny	Sometime Tuesday Morning	Johnny A	Sometime Tuesday Morning	Aglaophone	AR 1027
	You Don't Love Me	Cobbs			
Brown, Charles	Driftin' Blues	Brown, Moore, Williams	Just a Lucky So & So	Bullseye Blues	BB 9521
	Just a Lucky So & So	Mack, Ellington			
Cambridge Madrigal Singers	Alons, gay, gay	Costeley	Pour le Temps de Noel	Cove Arts	DCA 104
	Ce jour de l'an	Dufay			
	Hodie Christus natus est	Poulenc			
Emmanuel, Tommy	Those Who Wait	Emmanuel	Only	(vanity)	TE 001
	I've Always Thought of You	Emmanuel			
	Mombasa	Emmanuel			
Flanagan, Tommy, Trio	Sea Changes	Flanagan	Sea Changes	Evidence	ECD 22191 2
	Verdandi	Flanagan			
Isaak, Chris	Baby Did a Bad Bad Thing	Isaak	Forever Blue	Reprise	9 45845 2
	Things Go Wrong	Isaak			
	Goin' Nowhere	Isaak			
Larkin, Patty	Pundits and Poets	Larkin	Running Angels	High Street	72902 10318 2
	Booth of Glass	Larkin			
Lovett, Lyle	Here I Am	Lovett	and his Large Band	MCA/Curb	MCAD 42263
	It Ought to be Easier	Lovett	The Road to Ensenada	MCA/Curb	MCAD 11409
	Promices	Lovett			
Morphine	A Head with Wings	Sandman	Cure for Pain	Rykodisc	RCD 100262
	Buena	Sandman			
Test Signal Disc	various test tones		Mix Reference Disc		