

NOW IN THEIR THIRD GENERATION, CLASS D AMPLIFIERS HAVE FOUND COMPELLING COMBINATIONS OF CAPABILITY AND APPLICATION REQUIREMENTS.

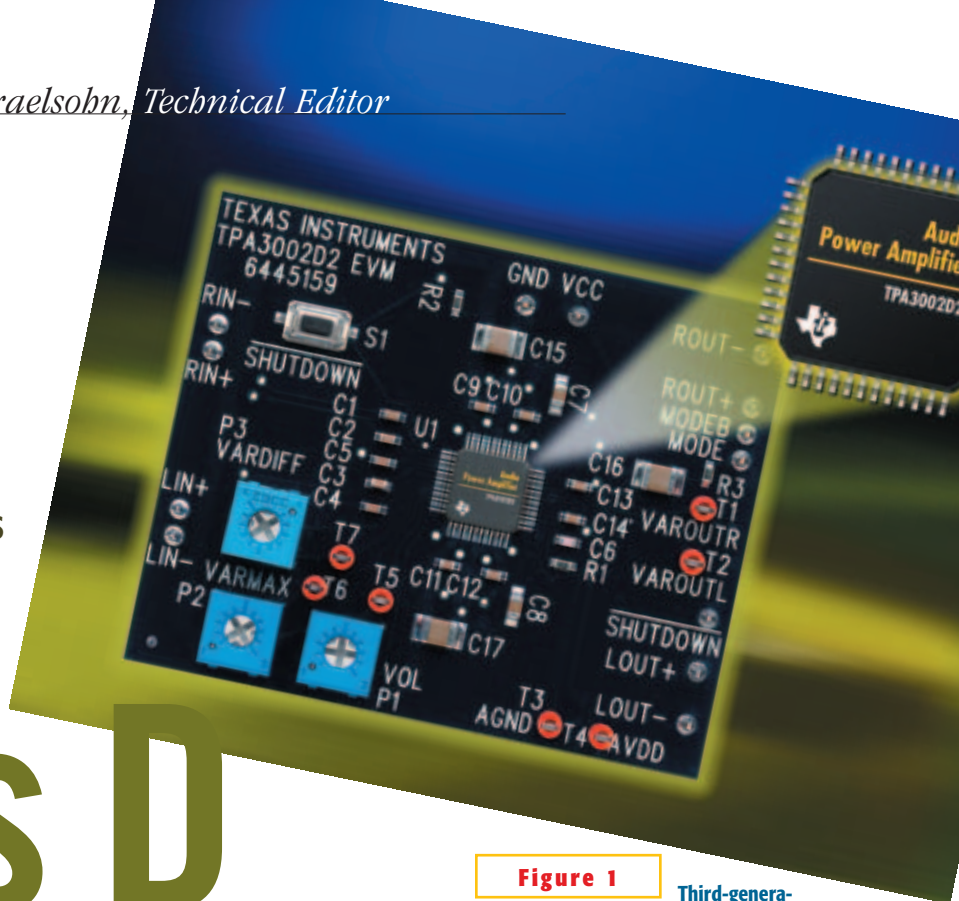


Figure 1 Third-generation Class D amplifiers are far smaller and simpler to design into your product than were second-generation amplifiers.

Class D Gen 3

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IF YOU THINK ABOUT IT, an audio amplifier is as conceptually simple a device as a person could imagine. Traditionally, it

has performed precisely one function—multiplication by a constant—albeit to a high degree of precision. Tolerable harmonic-distortion levels in audio amplifiers, at least for the purpose of spec-sheet presentation, are limited to roughly 1 to 0.0003% for the least and most demanding common applications, respectively. In that context, Class D amplifiers—which have the unique trait of rapidly slamming their output stages from one power rail to the other and back—appeared as unlikely approaches when first suggested in 1958 (Reference 1).

There are lots of ways you might choose to segment Class D amplifiers' commercialization history—none of them definitive. From my perspective, the

marketplace has witnessed three generations of designs. The first, exemplified by the Toccata-designed TacT Millennium, proved the concept and answered with finality arguments that the technology could not yield adequate performance. This first generation of amplifiers moved the focus from questions of possibility to those of practicality and the goal from making something that works to making something with broad market appeal.

The second-generation Class D amplifiers addressed the broad-market-appeal challenge by being comparatively compact and affordable and by performing as well as or better than most consumer-grade Class AB designs on lower power budgets. OEMs saved on power-supply and packaging costs but had to contend with unfamiliar topologies, sensitive layouts, and controller ICs that required many surrounding parts (Reference 2).

Second-generation Class D amplifiers offered a range of capabilities that corresponded with the manufacturers' mar-

ket focus. Typical of this second generation were devices from Tripath, which combined a relatively straightforward PWM for analog source signals with an integrated output stage and off-chip filter. Other manufacturers, notably Texas Instruments, Cirrus Logic, and Apogee Technology emphasized amplifiers with off-chip output stages for PCM sources. These amplifiers required more sophisticated front-end functions—those that you might otherwise associate with a stereo preamplifier, not the power amplifier, such as source selection, volume, balance, and tone controls—because PCM sources themselves often have no user-accessible controls of their own. These additional functions underscored the additional complexity that digital-signal paths demand for want of a *simple* digital replacement for the lowly potentiometer.

More recently, significant synergies have emerged between OEM- and Class D-chip designs. The trend's origins are doubly rooted: OEM designers better understand Class D's advantages and have provided chip makers with insights into their market opportunities and requirements. Meanwhile, chip designers have taken the market feedback to heart, addressing several of the most important criticisms of second-generation designs. The result is a spate of third-generation parts that focus better on specific applications' needs than did their predecessors.

MUST BE CLASS D

Monitors and televisions that feature LCD panels—virtually all such screens,

AT A GLANCE

▶ The signals may be audio, but Class D amplifiers can produce RF artifacts. Consider a Class D chip's EMI performance and be aware of its layout requirements.

▶ An amplifier's efficiency rating may not be a clear indicator of per-charge operating time in portable products. Remember to examine the idle current and be sure to compare devices under the same conditions of supply, signal, and load.

▶ Third-generation Class D amplifiers are designed with a sharper focus on specific applications. Performance, packaging, and features all reflect this trend.

now that the day of the glass blower is past—exemplify one trend. Most of these systems integrate small speakers that require amplifiers that deliver a half dozen to some teens of watts—not in and of itself a serious challenge. However, the LCD's pixel color is a function of operating temperature, and, though the current state of pixel-color correction provides for linear-compensation profiles, it does not accommodate hot spots. The system-packaging goal requires designers to minimize the dimensions beyond those needed by the LCD panel on all three axes. With no design budget for a heat sink's extra space, weight, or cost, these requirements combine to form a simple rule for this application: Thou shalt not dissipate.

The benefit of Class D's superior effi-

ciency extends beyond cooler operation. The power supply, be it integral to the display or a wall wart, is smaller and less expensive for a Class D design than for one that uses a comparable Class AB amplifier.

As a result, recent market introductions include Class D amplifiers that focus on LCD-based entertainment and computer displays. Several of these devices exploit improved modulation schemes that eliminate the output filter and its attendant bulk and cost. Despite the amplifiers' lack of output filtering, some manufacturers claim EMI emissions on par with or less than those of second-generation amplifiers equipped with output filters (see **sidebar** "Route out or root out EMI").

A few examples of amplifiers suitable for flat-panel display applications include Maxim's MAX9713 mono and MAX9714 stereo filterless amplifier chips, both available in TQFN-32 packages. These devices offer three pin-programmable modulation frequencies and a spread-spectrum mode that reduces EMI by spreading the emissions energy over a $\pm 7\%$ band centered on a nominal 335 kHz. The amplifiers pass FCC emissions standards with speaker runs as long as 14 in.

Labeled as nominal 6W amplifiers, that quantity reflects a typical value into 8Ω at 10% THD. Though the 6W claim may strike you at first as a transparent bit of specsmanship, quoting the nameplate power at 10% THD is a common—if lamentable—industry practice. The 9713's typical distortion spec, by con-

ROUTE OUT OR ROOT OUT EMI

As much as chip makers try to portray Class D amplifiers as highly efficient drop-in replacements for their Class AB counterparts, fundamental differences between the two amplifier types eventually belie the image. That said, the concept isn't entirely bankrupt, and third-generation chips extend it further than did second-generation parts.

One distinction that you cannot ignore is EMI emission. Linear audio amplifiers produce no EMI, so many designers don't think about the issue outside the context of the power supply. A Class

D amplifier's spectrum, on the other hand, extends far beyond the audio range. The modulators typically operate at hundreds of kilohertz with edge rise and fall times that you can measure in nanoseconds and spectral artifacts well into the megahertz.

Though the EMI energy may originate with the chip amplifier, the issue invariably becomes a system-design concern. Examine the broadband spectra supplied on prospective parts' data sheets and consider a Class D amplifier's EMI emissions as part of your overall EMI-management plan.

Some amplifiers allow you to select the modulator frequency, which can allow you to move emission harmonics out of an application-sensitive spectral region. Some amplifiers include spread-spectrum modulators that can smear the emission energy over a band and, in so doing, reduce the peak.

Among the chip makers' goals for a reference design is to communicate a layout that minimizes EMI emissions. In the end, however, your application will determine the allowable emission level, and your design will have to

ensure its compliance. Common high-speed-layout techniques apply, including keeping traces to external components short and taking care in ground-system design.

Filterless Class D amplifiers may require limited lengths for speaker feeds, forcing you to locate the amplifier close to its load. The advent of mono-amplifier chips facilitates this arrangement and leads to Class D structures that are smaller, less costly, and more efficient than their Class AB counterparts.

trast, is 0.07% at 4W into the same 8Ω. The power-supply range extends from 10 to 25V; the manufacturer gives performance specifications at a 15V nominal supply. Reading a bit deeper into the spec table, it appears that the output is current-limited rather than compliance-voltage-limited, which suggests that you'll not beat the power numbers by using 4Ω speakers.

The \$1.60 (100,000) MAX9713 and \$2 (100,000) MAX9714 also provide pin-selectable gains from 13 to 22 dB in 3-dB increments. The amplifiers provide differential inputs, but you can drive them

with single-ended sources by ac-coupling the unused input to ground.

Like most recently introduced amplifiers, the 9713/14 can complete a power-up sequence without generating clicks or pops on their outputs. At this point in their evolution, there is little reason to choose an amplifier that doesn't provide "clickless/popless" operation.

The TPA3002D2 from Texas Instruments is a 12V amplifier that provides two 9W channels and a pair of preamp outputs for an optional headphone feed (Figure 1). Unlike amplifiers that implement a volume control through a logic

interface, a dc voltage controls the 3002's volume. Your user interface can be as simple as a potentiometer, or you can drive the control node with a DAC. The amplifier's gain range extends from -40 to +36 dB.

The \$3.49 (1000) TPA3002D2 offers a 96-dB SNR and better than 0.25% THD driving 3W into 8Ω. The evaluation board in Figure 1 has been living at the EDN editorial offices for the last several months, where it has served as the "after-five" house system driving a pair of Tannoy Proto-J monitors—not the load that the amplifier's designers envisioned. To

IN SEARCH OF AN EFFICIENT ZERO

As is the case with many single-valued descriptions of complex behavior, amplifier-efficiency ratings tell part of but by no means the whole story. In a fashion, they don't even tell the most interesting or most relevant part, just the easiest to calculate and most attractive to present.

Amplifier efficiency is defined as the fraction of the power delivered from the power supply that reaches the amplifier's load at the amplifier's full rated output power. But if you examine typical audio signals, you'll find that both music and voice exhibit large crest factors, and, as a result, an amplifier spends more of its working lifetime near zero than near its full-rated output power.

From the perspective of per-charge operating time, therefore, the amplifier's efficiency rating isn't necessarily a good predictor of an end user's experience. This point is particularly true if you compare amplifiers with different modulation schemes—a common condition when comparisons cross vendor boundaries.

The problem lies in making an efficient zero. The simplest Class D implementation uses a simple pulse-width modulator and output bridge (Figure Aa). At zero signal, no power couples to the acoustic environment through the speaker. Nonzero currents in the filter, however, give rise to I²R losses.

Damped ternary modulation, an Apogee Technology scheme

that is also available in DDX-branded amplifiers from STMicroelectronics, reduces dissipation near zero signal by discontinuously operating the output devices (Figure Ab). The small minimum duty cycle imposed by damped ternary allows the system to reduce residual distortion near the signal crossover by canceling switching nonidealities. According to Apogee, this modulation approach reduces carrier energy by 16 dB, compared with the straight binary approach.

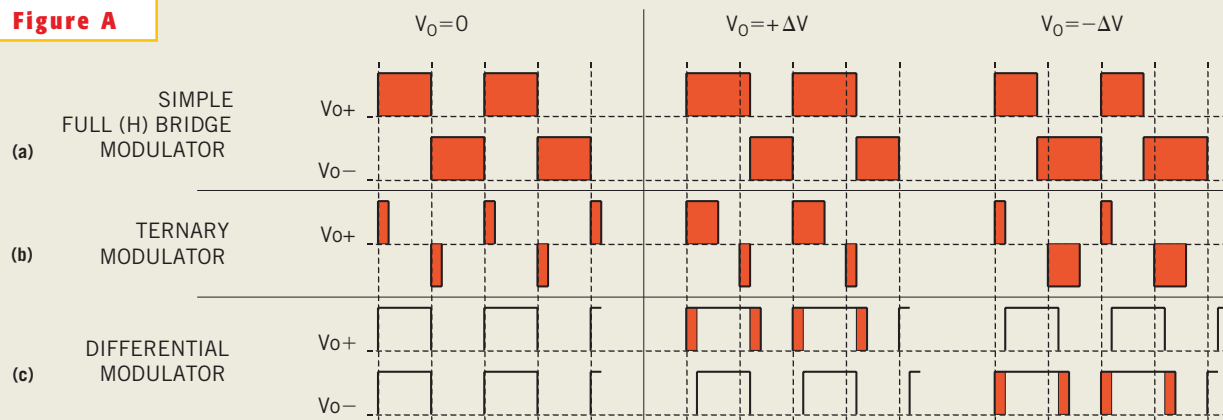
A drawback of damped ternary in portable applications is that it requires a bipolar power supply.

A relatively new approach from Texas Instruments for filter-

less outputs operates the amplifier outputs in common mode at zero signal (Figure Ac). Ideally, this arrangement results in no dissipation at zero signal. As the signal voltage moves away from zero, the pulse widths for the two outputs diverge, resulting in a differential signal across the speaker terminals.

When evaluating competing amplifiers, examine both the full power efficiency and idle current. Ensure that amplifiers you are considering quote these specifications at equivalent supply, signal, and load conditions. For example, you cannot compare an amplifier that quotes idle current at no load with another that quotes the quiescent with its load connected.

Figure A



The basic PWM switches current into and out of the reconstruction filter, engendering non-negligible I²R losses (a). Apogee Technology's damped ternary modulator eliminates much of the waste but does so in the context of a bipolar power supply (b). A differential modulator eliminates virtually all of the clock-rate loss at the signal zero-crossing (c).

make matters worse, the power supply it's been using is a bit too small for the job. Beyond the expected behavior—clipping at lower volume than would have been the case with a properly sized power supply—the noise floor “swells up” in sparsely arranged music following transients. Originally, I had assumed that this behavior was an unfortunate product of the amplifier that might have been masked had I not been using full-range monitors. Later, however, I came to suspect that the excess noise is directly related to the soft power supply's output behavior following large signal transients and that a proper supply would provide both better distortion at greater volumes and less noise.

MOBILE D

Little drives the portable-appliance market harder than per-charge runtime. As a result, new filterless Class D amplifiers are displacing previously entrenched Class AB devices at power levels as low as a few watts. TI's TPA2010D1 exemplifies the new Class Ds designed for mobile phones, smart phones, and PDAs. The 2.5W mono amplifier operates on 2.5 to 5.5V supplies and idles at a maximum of 3.2 and 4.9 mA, respectively, at the two ends of the supply range (see sidebar “In search of an efficient zero”). A logic input controls the shutdown-state in which the quiescent current drops to a maximum of 2 μ A.

The 2010 fits into a nine-ball WCSP that measures 1.45 mm on a side. The 55-cent (1000) amplifier requires only three external components: a pair of resistors to couple your signal source to the amplifier's differential input and a supply-bypass capacitor. The differential input rejects artifacts resulting from rectified RF—a common problem in TDMA and GSM phones. Your choice of input resistor sets the amplifiers' gain to 300 k Ω divided by the input resistance. You can accouple single-ended sources and sum combinations of differential and single-ended signals.

As do other manufacturers, Texas Instruments sets the nameplate power based on typical performance at 10% THD, in this case with a load impedance of 4 Ω . TI also specifies typical output power at 8 Ω and at both impedances operating to 1% THD. If you back off the output power, the 2010 achieves typical THD figures of 0.2% or better, pushing

1W, 0.5W and 200 mW into 8 Ω with 5, 3.6, and 2.5V supplies, respectively.

In contrast to the MAX971x, the TPA2010D1's supply voltage rather than its output current constrains its ability to deliver power to its load. This behavior is expected for either Class D or AB amplifiers within their safe-operating area from which two application-specific design decisions follow: speaker impedance and operating voltage. Ignoring its out-

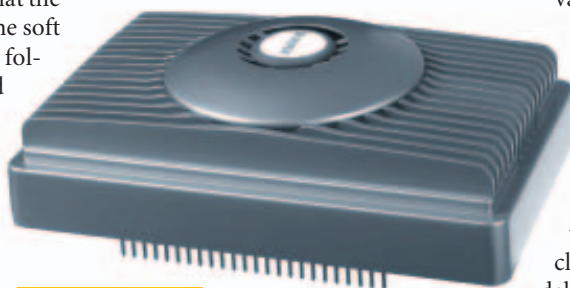


Figure 2 The D2Audio amplifier is a drop-in modular amplifier. With the problems of the gain-block solved, you are free to concentrate on differentiating your product with the features and functions most important to your customers.

put impedance for a moment, the maximum power the amplifier can deliver at a given supply voltage is inversely proportional to the speaker impedance. Your ability to push this trade-off is limited, however. The 2010 is specified for operation into either 4 or 8 Ω . Reducing the load impedance below 4 Ω increases the dissipation in the amplifier's output devices. The output FETs' on-resistance, which typically varies with supply voltage from 400 to 700 m Ω , increasingly cuts into the amplifier's efficiency and eventually threatens the amplifier's internal dissipation limit.

Similarly, if your application needs the 2010's small package but not the full rated output power, you can improve operating efficiency at lower power levels by switching to a lower supply voltage.

Filterless output stages have driven down the power level at which Class D amplifiers can compete successfully against Class AB alternatives; witness the LM4667 1.3W chip amplifier from National Semiconductor. National, which has a long line of successful Class AB chip amplifiers, a couple of years ago extended its Boomer line to include Class D chips. According to the chip maker, the move into Class D neither marks an end

to the AB line nor even cannibalizes it to any significant fraction but rather allows the company to sharpen the application focus for each new device. Other chip makers echo this posture in a growing competitive market, and it reflects in the range of features, power levels, and packaging options each manufacturer brings to the market.

In the case of the 4667, National's focus on mobile phones and PDAs motivates the nine-bump micro-SMD package that measures 1.5 mm on a side and 0.6 mm high, including the bumps. Two of those bumps connect logic signals that select the amplifier's gain—6 or 12 dB—and activate the amplifier's shutdown mode. Wakeup time is typically 5 msec, and transitions are clickless. The 48-cent amplifier uses a delta-sigma modulator, which according to National, yields better noise and distortion performance than traditional PWMs. The THD+N is 0.35% when driving 100 mW rms from 3V supplies.

LOTS O' WATTS

On the other end of the power range are amplifiers designed for home-stereo, home-theater, and other greater-than-personal-stereo applications. Though some of the structures may appear reminiscent of second-generation designs, these new chips offer enhanced features and improved performance.

Class D amplifiers much above a few tens of watts often comprise a controller chip and a separate power stage. The digital content of the controllers for these applications is substantial, and many manufacturers have concluded that an intelligent segmentation for the system is one that collects the logic functions on one chip fabricated in vanilla CMOS and the power devices on another chip built in a less dense, high-voltage process.

The WM8608 from Wolfson Microelectronics exemplifies this trend, packing a lot of features into a 7 \times 7-mm TQFP-48. The 8608 accepts four stereo PCM inputs, which may be coded as standard stereo or 5.1, 6.1, or 7.1 surround. The \$4.17 (10,000) Class D controller provides seven PWM outputs—six full-bandwidth and one reduced-bandwidth for a subwoofer channel. The chip can map 5.1-, 6.1-, or 7.1-surround sources onto 5.1- or 6.1-surround speaker arrays.

You can configure the outputs as either CMOS or LVDS. This flexibility helps you manage EMI with less stringent constraints on your pc-board layout. The WM8608 is compatible with power stages from Texas Instruments and STMicroelectronics. Alternatively, you can also use a driver and FETs, available from Zetex, Vishay, or Fairchild. The controller in combination with an integrated power stage typically yields 96-dB SNR and 0.1% THD at 30W.

The controller provides a four-band equalizer and selectable high-frequency compensation for various speaker types. Independent volume controls for each channel cover the -103.5- to +24-dB range in 0.5-dB steps. The 8608 also provides dynamic peak compression on each channel to prevent digital clipping at combinations of gain and equalization greater than 0 dB.

You can control the chip's internal features through a simple serial interface. The controller can accommodate 16-, 20-, 24-, and 32-bit words and all standard word rates from 32 to 192 kbps.

Zetex's entry into the Class D market is the ZXCW8100S28 stereo controller. This chip forms a complete two-channel amplifier with the addition of drivers and power FETs—also available from Zetex. This is one of three Class D amplifiers I auditioned while researching this article by means of a ZXCW502CEVAL evaluation board. The 8100 on the 502 evaluation board is an impressively clean and quiet amplifier with an SNR of 118 dB and THD of 0.021% at 1W into 4Ω. THD+N curves stay below 0.1% through 10W into 8Ω, and rise to only 1.2% at 20W.

The 8100 accepts 16-, 24-, and 32-bit data at rates from 32 to 192 kbps. The chip's 32-bit signal processor provides volume, base, and treble control; clipping

control; and switch-device compensation. The chip also implements a proprietary filter algorithm, ZTA, that the manufacturer claims improves transient resolution and stereo imaging.

Texas Instruments and National Semiconductor offer controller/power-stage combinations, albeit with different capabilities and prices intended for different applications. TI's TAS5508 accepts eight PCM channels at standard word rates of 32k to 192 kbps, inclusive. TI builds the TAS5508 into a TQFP-64 and around a DSP with an internal 32-bit datapath, 48-bit audio processing, and 76-bit accumulators.

The gain control operates over -100 to +36 dB in 0.25-dB steps. Two-pole treble and bass controls and six-band parametric equalizers provide flexibility for user-interface controls and speaker/environment compensation. Compression, loudness compensation, bass management, and sample-rate conversion round out the \$6.30 (1000) controller's feature list.

You can scale TAS5508-based products by selecting among TI's line of power stages, which includes the \$3.50 (1000), 100W RMS TAS5121. As noted, manufacturers liberally quote power ratings for Class D amplifiers; the numbers may be simultaneously attainable and undesirable. Keep this fact in mind when interpreting manufacturers' data sheets. In the case of the TAS5508, the 100W rms is the 10% THD point driving a 4Ω load. Under the same load, the power stage contributes 0.2% THD at 80W rms and 0.05% at 1W rms. Also as noted, the crest factors of most music and speech suggest that full-power operation is only an occasional phenomenon and, if expected, should call the speaker performance into question as well.

National Semiconductor's \$3.25 (1000) LM4651 controller and \$2.75 (1000) LM4652 power stage form a single-channel, analog-input amplifier chip set for powered subwoofers, automotive booster amplifiers, and self-powered full-range speakers. Despite the now rarely seen through-hole packages, MDIP-28 and TO-220-15 for the 4651 and 52, respectively, the chip set forms an inexpensive 170W amplifier that is suitably compact for its target applications. As usual, the nameplate power rating is given at the 10% THD point—here, driving 4Ω—falling to 125W to capture the 1% THD point. The A-weighted SNR is 92 dB with respect to a 125W output into 4Ω.

JUST THE FAQs, MA'AM

If you're a regular reader of *EDN* or if you attend industry conferences, you may have noticed that the topic of "intelligent segmentation" arises with regularity. Often, the question arises when IC designers reconcile the disparate needs of functional blocks with the process or packaging technologies available for their implementation—all very high-concept stuff.

But the fact remains that segmentation decisions that IC designers make ripple across to the design tasks that OEM designers must take on. The problem is that an IC designer's notion of intelligent segmentation—well thought through and defensible though it may be—may not coincide with the OEM designer's idea of intelligent segmentation from a system perspective.

To their credit, IC manufacturers expend substantial energy and resources making reference designs, building and stocking evaluation boards, training and staffing application-support engineers, and disseminating the answers to frequently asked questions on their Web sites. The level of customer support available from IC makers selling Class D chips is impressive. Experience shows that the need for this level of support exists whether the amplifier in question is a single-chip device, a controller/power-stage chip set, or a controller/driver/power-FET design. Indeed, the top three questions that application engineers field when supporting Class D amplifiers are:

- How should I route the board traces?
- Where should I place the external passive components? and
- What are the best values for the external components?

FOR MORE INFORMATION...

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

Apogee Technology
www.apogeeddx.com

Cirrus Logic
www.cirrus.com

D2Audio
www.d2audio.com

Fairchild Semiconductor
www.fairchildsemi.com

Maxim Integrated Products
www.maxim-ic.com

National Semiconductor
www.national.com

STMicroelectronics
www.st.com

Texas Instruments
www.ti.com

Tripath
www.tripath.com

Vishay
www.vishay.com

Wolfson Microelectronics
www.wolfsonmicro.com

Zetex
www.zetex.com

Careful reading of the manufacturer's data sheet and application notes should answer these questions for most applications, but, because amplifiers must fit into a context, your available board space may look different from the reference design. You may also be concerned about interaction with surrounding circuitry, either by parasitic coupling or through the power supply—concerns that are specific to your layout and that the reference design does not reflect.

THE BOX FROM OUTSIDE THE BOX

In an industry that no longer blanches at the thought of 10-Gigabit Ethernet or 40-Gigabit fiber-optic communications, it is remarkable how much time and energy are consumed by a simple gain block that can with reasonable fidelity replicate signals in the narrow band from 20 Hz to 20 kHz. Manufacturers of high-end and professional box-level audio components recognize that the gain block is a means not an end. At some level of performance, the selling point is no longer the performance but other attributes that combine to define the product's function and form the end user's experience. As the sonic quality of a dedicated audio component rises, its features, flexibility, and user-interface design become more identifiable points of comparison. Paradoxically, perhaps, if the amplifier is good enough, it reverts to the role of system enabler, and OEM designers can concentrate on the features, flexibility, and user interface that distinguish high-end products in their marketplace.

D2Audio offers a series of drop-in amplifiers that free OEM designers to concentrate on critical-path tasks and reduce overall time to market (Figure 2). Based on a common architecture, D2Audio's various models offer performance, attributes, and channel counts geared toward specific applications, such as the XR125 for A/V receivers and home theater, XM100 for multiroom distributed audio, XC100 for distributed-commercial-sound amplification, and XS250 for powered monitors. Amplifier layout, component selection, gate-drive integrity, and filter design are all issues that reside inside the module. You connect audio inputs, speaker outputs, a power supply, and a simple control interface and are free to concentrate your design efforts on your customers' needs, not the

YOU MAY BE CONCERNED ABOUT INTERACTION WITH SURROUNDING CIRCUITRY, EITHER BY PARASITIC COUPLING OR THROUGH THE POWER SUPPLY—CONCERNS THAT ARE SPECIFIC TO YOUR LAYOUT.

amplifier's. D2Audio implements output stages on plug-in daughtercards, which provide instant output-power scalability. Because the daughtercard resides within the D2Audio controller's feedback loop, the controller and interface software need no alteration to accommodate different output stages.

The input structure is as versatile as you'll find on any Class D amplifier: You can connect digital audio through balanced AES/EBU, unbalanced SPDIF, or raw I²S interfaces with 16- to 24-bit samples at 32 to 192 kbps. Optional interfaces include an analog port to support legacy signal sources.

Whichever source you choose, the input signals pass through a sample-rate converter and an adaptive PCM-to-PWM converter, a level-shifter, a gate-drive circuit, the output-power MOSFET devices, and a filter. The amplifier's signal-processing core is a proprietary ASIC that integrates the low-level signal chain and adds a DSP, proprietary drive-signal compensation, and optimization blocks that provide a feedback signal to the DSP. The feedback path improves the amplifier's SNR by 23 dB and reduces THD+N at 1 kHz by 15 to 19 dB. The DSP has sufficient excess processing power to run OEM algorithms for product differentiation.

The \$150 (10,000) XR125, for example, provides seven channels at 125W into 8 Ω . Distortion is less than 0.05% at the 1W level. The amplifier's SNR is better than 105 dB, and the frequency response is flat to within 0.5 dB from 20 Hz to 20 kHz.

The XR125 provides programmable tone controls, volume control, and a five-band parametric equalizer. Zone outputs feature dynamic-range compression to prevent clipping, adjustable time delay to compensate for speaker location, and a three-band parametric equalizer to compensate for speaker deficiencies and

anomalous room acoustics.

You control the amplifier through a standard two-wire serial interface and software-control API. D2Audio also provides the D2Audio Canvas GUI, which can speed product development and customization.

I auditioned a prototype of the D2Audio with its chief designer in an unfair comparison with a Perreux Class A amp whose cost is the neighborhood of 20 times that of the D2Audio amp. No, the D2Audio prototype did not win the contest. Though easily detectable, the sonic difference seemed small compared with the difference in the price tags.

The only disappointment during the audition came when we listened through D2Audio's optional ADC, which lacked both the transparency and the stereo image the amplifier produced when operating through its digital inputs. Because the chief designer had been in the room with me, it came as no surprise to learn days later that the ADC we had heard was no longer on the print and that the design team had selected a new device that better matched the amplifier's sonic quality. For the remainder of the audition, we piped program material directly from a CD player through a coaxial SPDIF connection with satisfying results. □

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REFERENCES

1. Stanley, Gerald, "Audio power amplifiers: a brief history," presented to the Chicago Section, Audio Engineering Society, April 10, 2002.
2. Israelsohn, Joshua, "Listening to Class D," *EDN*, Aug 30, 2001, pg 65.

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TALK TO US

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