Digital Audio’s Final Frontier
Mini but mighty
Class D amps
are forging
audio's future

BY BRUNO PUTZEYS
DIGITAL TECHNOLOGY continues its march from media like CDs and DVDs toward your audio speakers. Today, amplifiers based on digital principles are already having a profound effect on equipment efficiency and size. They are also beginning to set the standard for sound quality.

An old idea, the Class D amplifier has taken on new life as equipment manufacturers and consumers redefine the musical experience to be as likely to occur in a car, on a personal stereo, or on an airplane as in a living room. For most consumers today, portability and style outweigh other factors in the choice of new audio gear.

Class D amplifiers are ideally suited to capitalize on the trend. They are already starting to replace conventional high-fidelity amplifiers, particularly in mobile and portable applications, where their high efficiency and small size put them in a class by themselves. For example, they are fast becoming the dominant technology for entertainment systems in cars, where passengers are now apt to watch a DVD—and expect from the vehicle's compact, ill-illuminated electronics the same musical surround-sound experience they get at home.

The new amplifiers can provide it. They are typically around 90 percent efficient at rated power, versus 65-70 percent for conventional audio amps. Such high efficiency means, for one thing, that the amplifiers can get by with much smaller heat sinks to carry away the energy they waste. Also, portable devices like MP3 players can go much longer on a battery charge or can be powered by lighter batteries.

Class D amplifiers have been used for decades in industrial and medical applications where high efficiency is key. They have been applied with great success in devices as small as hearing aids and as large as controllers for heavy motors and elevators. They blossomed as a significant force in high-fidelity audio a few years ago when Class D power amplifier chips were released by companies like Texas Instruments, Cirrus Logic in the United States, Philips and STMicroelectronics (partnering with Agere/DUX) in Europe, and Sansui (partnering with Fujitsu & Olympus) in Japan.

More recently, Class D amps have expanded beyond the hi-fi niche, showing up in MP3 players, portable CD players, laptop computers, cellphones, even personal digital assistants (PDAs). At the same time, they have been making forays into the world of home audio in the form of products based on these new chips. Notable entries include amplifiers from Bel Canto Design Ltd, (Minneapolis, Minn.) and PS Audio (Boulder, Colo.). Sharp Corp., (Osaka, Japan) is also pushing aggressively into the home audio market with a somewhat different line of amplifiers ranging in price from US $500 to $15,000.

The company's use of delta-sigma technology, which has the effect of limiting the amplifiers' efficiency.

By the way, the "D" in Class D does not stand for digital, but was simply the next available letter for classifying amplifiers. What distinguishes Class D amplifiers from all others is that their power transistors are always operated either fully on or fully off. This is the only and complete definition of Class D; its significance will become clear later.

Nonetheless aside, Class D amps do point to an eventual sea change for audio. In their most obvious form, these systems will be able to accept a stream of bits from a CD or MP3 player and convert it into an analog signal that can drive a set of speakers. Contrast these with conventional amplifiers, which are entirely analog and can amplify digital signals only after those signals have been converted into analog form. Usually in the CD player or MP3 unit (see figure opposite).

Class acts

Today, Class D amplifiers generally work with low-level analog signals, which they pump up to current and voltage levels high enough to drive audio speakers. Remarkably, the heart of such an amplifier does this by switching between only two signal levels. That feat is achieved through various pulse modulation schemes, in which a continuously varying analog waveform is changed into one that switches between just two values—a binary signal, in other words. Some parameter (or parameters) of this binary signal—for example, the relative duration of its on- and off-states—conveys the information contained in the smooth curve of the analog input wave (see illustration just below the caption on p. 39). Before delving more deeply into the workings of this scheme, called pulse-width modulation, let's briefly review conventional amplifier circuitry.

Engineers have divided amplifier circuit types into different topologies and classes, depending on how much current is allowed to flow through the transistors or tubes when they aren't delivering power to the speakers. Most conventional audio amplifiers are a variation of a topology known as push-pull. Its most elementary form, it uses two transistors (or, if you prefer, vacuum tubes) one for pushing current, or sourcing it, toward the speakers; the other for pulling it back, or sinking it. To get a (very rough) idea of how it works, think of two lumberjacks sawing a tree, one on either end of a long, old-fashioned saw.

In Class B push-pull, output current is delivered by only one transistor at a time, while the other is completely inactive. In the case of the lumberjacks, only one could be pushing on the saw at any given instant—and they would have to time their exertions very precisely to work efficiently. Similarly, in a Class A amp (see figures, p. 39), note that all of the output current (green waveform) comes either from the current-sourcing transistor (red waveform) or from the current-sinking one (blue waveform), but never from both at the same time. This mode of operation is neither efficient, it can theo-
critically deliver 74 percent of the power it receives from the power supply to the speakers, dissipating the rest as heat. But in this kind of amp, 100 percent of the current that goes through the transistors is delivered to the speakers. So why isn't the amplifier 100 percent efficient? Recall that power is the product of current and voltage. If our amplifier has a maximum output voltage of, say, 40 V and at some point it is called upon to put out, say, 8 V across the 8 ohm speakers, the output current would be 1 A and the output power would be 8 W. But since the supply voltage would have to be 40 V to properly handle higher instantaneous signal amplitudes, that instant 32 V would be dropped across the transistor while the very same 1 A of current flowed through it. The result: 30 W of power consumption—8 W to power the speaker and 32 W to heat the room.

The main problem with Class B is crossover distortion, which occurs when the output switches from one transistor to the other. If this instantaneous switching of the two devices is not coordinated with absolute perfection, the discontinuity creates distortion that can be heard in the sound coming out of the speakers.

So for audio engineers, Class AB amplifiers are almost certainly best in your house. In Class AB (see diagrams on p. 39), both transistors carry current even when no net current is delivered to the load, doing so in what is known as the crossover point. Remember how hankering we had for those vintage Class AB amplifiers because they seemed so much more like the speaker? How they sent that same raw signal straight to the speaker, with no crossover spikes? Well...now we have the crossover back. Cross-over takes place when the output current is near zero, some current flows from one transistor to the other and never gets to the load at all.

The powers that "D"

Class D amplifiers also use two transistors, but instead of amplifying analog signals, which can assume any value, they switch between just two voltage values, +40 V and -40 V in this case of the amplifier referenced above. One transistor can connect the output to the +40 V rail, the other to the -40 V rail. Theoretically, neither transistor wastes any power.
For distortion

Second, Class D designs are prone to distortion, chiefly from imperfect power supply regulation and timing error. Since the output voltage of a Class D amplifier is directly proportional to the power supply voltage, any error in that voltage modulates the output voltage. Power supply variations caused by variations in the amount of current drawn by the amplifier show up in the output as distortion. Instabilities in the supply itself, such as power line ripple, show up at the output as noise, or hum. Building a power supply so that voltages remain rock steady in spite of fluctuations in output current is not a trivial task.

The other source of distortion is timing error, due to variation in how long MOSFETs take to switch from on to off, which in turn depends on how much current the amp is being called on to deliver. This error causes the output duty-cycle to deviate from the input duty-cycle, such that the output signal shape differs from the input signal's shape. Timing error causes distortion directly proportional to the duty-cycle error - the ratio of the timing uncertainty to a single switching period.

The greater the timing uncertainty and the higher the switching frequency (the higher the frequency of the triangle wave in the PWM circuitry), the worse the distortion.

An IC power stage optimized to minimize its timing uncertainties, like the one from Texas Instruments, can sport errors as low as 10 µs. With a typical switching frequency around 350 kHz, this corresponds to a distortion figure of around 0.1 percent. Such figures are certainly acceptable for inexpensive multichannel audio. But high-performance audio amplifiers for items such as MP3 players and home stereo need to reduce distortion to as little as 0.01 percent. Fortunately it's possible to correct for this rather inexpensively.

Frequency response is another prime performance issue for Class D amplifiers. The all-important low-pass output filter, which recovers the original audio signal from the PWM waveform, is passive, and its frequency response is flat only when it drives a purely resistive load of a specified value. Since the impedance of real loudspeaker loads varies between 4 and 16 ohms at various frequencies, a selector is obviously needed.

These performance issues persuaded many engineers to adopt a stance of technical conservatism that delayed the commercial introduction of Class D amplifiers for many years. But practical demonstrations and listening tests have revealed that these amplifiers are now capable of delivering performance that rivals that of vacuum tubes.

Because modern signal sources are digital, it seems natural to derive the PWM signal directly from the digital source, without converting it to analog first. Indeed, the early 1990s saw the development of several schemes for converting the digital outputs of equipment like CD and DVD players into PWM waveforms suitable for driving switching power devices. Their development reflected a belief that digital-to-analog (D/A) conversion incurs quality loss and that D/A converters are necessity expensive.

Future digital signal processors will include digital signal processors to correct the power stage's inevitable analog errors.

The output of this analog comparator, then, is a waveform that has the information of the original analog signal and yet switches between just two values—in other words, it's precisely what we need. In a Class D amplifier, the PWM waveform acts as a binary control signal that switches the transistor on and off depending on the amplitude of the analog input. Changing the power supply voltages changes the amount of amplification.

Of course, what the transistors produce is a higher power version of this same switching waveform, which would overheat the speakers dreadfully if allowed to reach them, containing as it does amplified versions of both the original audio input signal and myriad higher-frequency components arising from the PWM process. So after the amplification, that PWM waveform has to be passed through a filter that lets lower-frequency signals through while weakening the higher-frequency ones. This lets the input filter does by smoothing the switching waveform, in effect suppressing the rapid changes in the output waveform and leaving only its average value. At the same time, happily enough, it blocks out noises caused by the switching process itself.

Class D audio amplifiers are reasonably simple, conceptually. The trick is building one that is affordable and that performs well. First of all, it requires inexpensive, low-cost, fast-switching transistors that are easy to drive. Such devices became available only with the maturing of metal-oxide semiconductor field-effect transistors (MOSFETs) in the early 1990s.
When a D Lands You At the Head of the Class

When a push-pull amplifier (right) is operated in Class B, all of the output current (green waveform) comes either from the current-sourcing transistor (red waveform) or from the current-sinking device (blue waveform) but never from both at the same time. Class AB exhibits less distortion by allowing the transistors to work together when the output signal is near zero, in what is called the crossover region.

One way to get a two-stage, on-chip, amplifier to handle analog audio signals is through the use of pulse-width modulation (PWM), which converts the audio input (green) into a binary PWM signal (red) by comparing it with a triangle waveform (blue) as shown in the circuit diagram (below). The PWM signal turns the sourcing power transistor on and off, while its inverse does the same for the sinking power transistor. The gain of the amplifier is varied by varying the value of the 40 V supply voltages.

The red waveform below carries all the information in the green analog signal by spending, on average, more time at its upper value when the analog signal is larger and less time there when it is smaller.

Obviously, both premises are relative. As noted earlier, an output stage driven directly by a PWM signal is unlikely to yield harmonic distortion figures much better than 0.1 percent—and it needs an expensive regulated power supply to do even that well. Digital PWM generators are presently available, but only as fairly expensive stand-alone chips. Currently available D-A converters easily deliver harmonic distortion as low as 0.01 percent for as little as US $50 per stereo chip.

Feedback is key

The three major problems confronting high-resolution Class D operation (power supply modulation, timing errors, and load dependent frequency response) can be cheaply and effectively solved using an analog feedback system. Several varieties of these are available, all of them compensating for output-stage distortion and some addressing the frequency-response problem as well.
The Once and Future Audio Amplifiers

BY DAN C. SWEENEY

The recent trend in consumer audio has been towards higher-end components. This has become a more interesting and enjoyable experience. The manufacturers and retailers are doing a great job of promoting these products to the public. However, there is a limit to what can be done with current technology. As the audio market becomes more competitive, the pressure to develop new and innovative products increases.

At one extreme, the consumer electronics market is dominated by high-end, high-quality products. These products are often expensive and require careful selection of components to achieve the desired sound quality. At the other extreme, there is a trend towards lower-end, more affordable products. These products are often marketed as alternatives to high-end products, offering similar sound quality for a lower price.

In some cases, however, the pressure to develop new and innovative products can lead to the development of products that are not well-suited for the intended market. For example, some high-end products are designed to be used in conjunction with other high-end components, such as high-quality speakers and headphones. These products are often difficult to use and require careful selection of components to achieve the desired sound quality.

In contrast, lower-end products are often marketed as alternatives to high-end products, offering similar sound quality for a lower price. These products are often designed to be used in conjunction with a wide range of components, allowing users to create a system that meets their individual needs.

In conclusion, the audio market is becoming more competitive, and manufacturers and retailers are doing a great job of promoting these products to the public. However, there is a limit to what can be done with current technology, and the pressure to develop new and innovative products increases. As the audio market becomes more competitive, the need for careful selection of components to achieve the desired sound quality becomes more important.
20 kHz. Transients are somewhat blurred with this amp, and resolution of inner detail is not as good as that of the other amps in this survey.

Measured distortion is quite high, nearly 2 percent at 40 W and 1 kHz. Nevertheless, some listeners will be so seduced by the sound that they won't give these numbers a second thought.

Present and accounted for

The Gamut C100 is like most amps made today in its use of solid-state circuitry throughout. It operates in Class AB, which trades off some efficiency to achieve good, linear amplification. The circuit design is quite sophisticated, with extensive current and voltage regulation throughout. The Gamut also distinguishes itself by the use of two exotic ultra-high-current MOSFETs per output channel; these devices handle peak currents of 300 A. In comparison with the SA-90, a far less typical of the garden-variety MOSFETs found in ordinary amplifiers.

The Gamut II has been roundly praised, and justifiably so. It is remarkably free from the hard, metallic sound characteristic of the traditional solid-state set, and comes close in transparency to my reference Wocal amplifiers, which cost roughly twice as much. It also provides a sense of power and authority, though not as much as its 400-W-per-channel rating. Design and build quality are about average for an amplifier in this price range. Distortion is a low 0.01 percent at 100 W and 1 kHz into 4 ohms.

Future perfect?

When a surprise the digital Sharp amp wasn't! The company is known for high-end audio, but the SH-90X may change that.

The amp makes use of a variation on a technique called direct-current digital (DCD) modulation, which is also used in the Sony-Philips Super Audio CD system. DCD is a form of pulse-density modulation, in which a sequence of pulses of equal duration and equal amplitude encode an analog waveform. The peaks and troughs of the analog sound signal are encoded as differences in pulse density, or periodicity. So, in a general sense, DCD is like a form of frequency modulation. Strictly speaking, the SH-90X's circuit is a form of pulse-code modulation (PCM).

Incidentally, the Sharp technology should be confused with the pulse-width modulation schemes used in many amplifiers today, as discussed in the main article. The scheme records an extremely high clock frequency, and the SH-90X's internal clock runs at 2.8 MHz. Since the technology dictates that the output devices always switch at full power, the demands on the output transistors are considerable. Sharp uses power FETs optimized for high-power, high-frequency operation to do the job.

The advantage of DCD, as I alluded in theory, is that the linearity of the output devices essentially becomes a non-issue. The accuracy of reproductions is determined entirely by the precision of the clock and the modulator circuitry. So how does the Sharp perform? Spectacularly well in my ears, much less so in my test instruments. The amp reproduces with startling fidelity, and the most complex details of orchestral scores are uncannily correct and in tune, while at the same time there's not a trace of glare.

On the other hand, noise approaches 2 percent of the signal level at 20 kHz, dropping by more than two orders of magnitude at 1 kHz. Also, at a mere 90 W per channel, power is quite limited for an amp at this price; I generally like about five times that much power. Styling is handsome, but the fittings aren't quite up to the level that you expect in a $4,000 amplifier.

The verdict

In extended listening sessions, I preferred the Sharp to the rest, though the other test bench results are below the line. It shouldn't sound so good, but it does (and listeners of limited home for the file agreed with me).

Note that the SH-90X is the least expensive in a whole new series of Sharp DCD amplifiers, which range in price from $1,000 to $15,000. I can't wait to hear its higher-powered brethren.

Growing up

In spite of its current deficiencies, digital Class D amplification is still well on its way to ascendance. Multimedia devices and manufactured home-theater-in-a-box systems have no strong requirements for high-resolution audio performance, although many do have system integration and miniaturization imperative.

Future research efforts in digital amplification will increasingly focus on integrating digital signal processing (DSP) to correct the power stage's inevitable analog errors. Several companies are already well down that path, building controllers that sense the power rail voltage and modify the PWM signal accordingly.

The distant future will almost certainly involve circuits that perform digital modulation on the fly by measuring analog error data from the power stage and modifying the switch control signals accordingly. Preliminary work from several researchers suggests that such schemes may ultimately deliver performance that is simply out of the question any other way.

To Probe Further


For specifics on Philips' completely integrated Class D amps—among the most powerful around—see http://www.sonicmag.com.

Visit http://www.st.com to learn about Texas Instruments' strong portfolio of analog and digital Class D products.

Home of the world's first fully-integrated Class D amp, STMicroelectronics has a site at http://www.st.com.

Tripath's Class D chips and drivers incorporate some nifty tricks borrowed from analog-to-digital and digital-to-analog converter design. Check them out at http://www.tripath.com.