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# coverstory

# FROM EDN EUROPE: Digital amplifiers challenge analogue audiophiles

Audio amplification has long been the one area that's successfully held out against digital technology. Now, semiconductor manufacturers are introducing switch-mode amplifiers with specifications at least as good as their linear counterparts. Can digital designs finally claim audio dominance, too?

# David Marsh, Contributing Technical Editor

Despite product and technology allegiances that can border on religious zealotry, any true audiophile quickly acknowledges that no equipment is perfect. Equipment manufacturers respond to esoteric demands with some apparently retrograde steps, such as Quad's offering vacuum-tube designs alongside the company's iconic current-dumping transistor technology. In the meantime, the mass market for audio power amplifiers lies with far less exotic devices to service needs such as mobile phones and in-car entertainment systems. These applications also require decent sound quality but stress criteria such as low weight and minimum power consumption, promoting switch-mode-amplifier topologies to prominence. But don't imagine that switch-

AT A GLANCE

Switch-mode amplifiers improve real operating efficiency to 50 to 95%.

Sonic performance routinely equals midrange linear amplifiers.

Implementation issues include power-supply design and EMC approval.

Advanced modulation schemes yield audiophile performance.

Signal paths move toward digital exclusivity.

Digital amplifier displays audiophile quality

mode-amplifier topologies to prominence. But don't imagine that switch-mode suits only low-end sonic duties; equipment manufacturers such as TacT Audio offer all-digital power amplifiers with claimed performance figures at least as good as today's best linear designs.

Most audio power amplifiers use a class-AB topology that's a compromise between linearity and power consumption. Class-A designs permanently bias their output devices into conduction to avoid nonlinearities when switching the output signal through zero, minimising crossover distortion at the expense of efficiencies as low as 5%. Biasing the output devices just into linear conduction under quiescent conditions creates a class-AB design that restrains crossover distortion to levels that feedback can reduce—routinely achieving total-harmonic-distortion-and-noise (THD+N) figures better than 0.1%. Under signal conditions, the output devices still dissipate the difference between power-supply voltage and output-signal level as heat, typically constraining efficiency to less than 35%. Quad's classic current-dumping design couples a class-A low-power stage with a class-C output-transistor pair in a feedforward bridge arrangement, dispensing with bias adjustments and setup and drift considerations (Reference 1). The result is audiophile THD+N performance of less than 0.01%. However, there's still no real efficiency gain, and maximum inefficiency lies at about 35% of maximum level—right where most listening occurs.

#### **PWM slashes power consumption**

Classic "digital" audio amplifiers are class-D designs that have more in common with switch-mode

power supplies than with traditional hi-fi equipment. Similar PWM techniques achieve efficiencies that range from 50% at low volume to 80 to 95% as output levels rise. This radical improvement in conversion efficiency comes from switching the output power devices fully on or off, constraining I<sup>2</sup> R losses due to partial conduction. The basic class-D scheme converts analogue-input signals to rectangular-wave outputs by comparing the audio signal with a sampling signal; input-signal amplitude thus modulates output-signal duty cycle (Figure 1a). Unlike a switch-mode power supply that typically uses a sawtooth ramp, class-D designs prefer triangular sampling signals that ease comparator design. However, it's vital to eradicate sampling noise and ramp nonlinearities that create distortion. Comparator-design challenges include achieving reference-voltage stability and providing sufficient common-mode rejection for stable threshold triggering across the input signal's dynamic range.

The sampling frequency is typically an order of magnitude higher than the human auditory limit of about 20 kHz, demanding power MOSFETs for the output drivers. In the simplest arrangement, the loudspeaker's inductance—and your ears—filter the out-of-band components so that you perceive linearly changing audio-frequency information; typically, a passive inductor-capacitor (LC) filter between the amplifier and the speaker attenuates high-frequency energy. The filter also smoothes the high-power current pulses that arise from switching-power-supply voltages to the load, easing power-supply design and limiting emissions that cause EMC problems. Crucially, class D's efficiency lengthens battery life and can allow you to dispense with heat sinks without unacceptable sound-quality sacrifices. THD+N figures of approximately 0.15% are common to about 15 kHz, equalling many hi-fi vacuum-tube designs. Typical IC amplifiers demonstrate noise-floor figures greater than 90 dB and crossover-distortion artifacts are almost completely absent. And frequency-response performance is impressive and easy to achieve. Class-D amplifiers are typically flat to less than 1 dB across the 20-Hz to 20-kHz audio-frequency range.

#### PWM efficiency doesn't come free

Switch-mode amplifiers have inherent drawbacks that result from the sampling process. Even when no input signal exists, class D's typical antiphase output-driver arrangement generates current flow at the switching frequency (Figure 1b). At today's PWM frequencies, this switching energy causes negligible speaker-cone movement as deflection is proportional to  $1/f^2$  for frequencies above the audio band. But if you're using miniature components, such as earphones, check that they can handle the additional switching power that dissipates as ripple currents in the load. You can calculate switching power per channel by removing any output filtering and measuring the difference between the quiescent currents that the amplifier draws with and without a load. Switching power is this difference current multiplied by the power-supply voltage. The results can be dramatic: With no external filtering, a typical 1W design can easily dissipate greater than 100 mW of switching energy in the speaker.

As with ADCs, the sample rate reconstructs a more faithful version of the input signal at low signal frequencies and starts to lose information toward the maximum-input-frequency range. The sampling process also generates low-level images of the signal frequency at submultiples of the sampling frequency. Typical class-D-distortion characteristics are best at low frequency (approximately 1 kHz), then rise to a peak at around 7 to 8 kHz before falling as the output filter starts to reject high-frequency artifacts (Figure 2a). In some designs, intermodulation distortion—the product of mixing the harmonics of two unrelated frequencies—can be high even when THD+N is low, leading to harshness that fatigues listeners. Because intermodulation distortion is a function of nonlinear-transfer characteristics, good open-loop performance is essential for best fidelity.

For optimum efficiency and sonic quality, the output devices should switch in zero time. Of course, it's impossible to switch the output devices instantaneously or completely cleanly; there are finite rise/fall times and potential overshoot as the pulse reaches its extremities, together with conduction losses that dissipate as heat. Residual crossover-distortion performance depends on the dead time between switching the output-power devices that's essential to prevent device destruction by mutual conduction. As a rule of thumb, allow about 15 to 20 nsec of dead time at 200 kHz and 5 to 10 nsec at 500-kHz PWM rates. Any transient mutual conduction ("shoot-through") contributes distortions, and the transient switching currents can couple to the output to further degrade fidelity. Also note that the onset of duty-cycle limitation and power-supply induced clipping causes distortion figures to rise exponentially toward maximum output-power levels (Figure 2b). Notice that several datasheets quote maximum output power at 10% THD+N, but a 1%-distortion ceiling is far more realistic.

You have to consider power-supply and pc-board design to handle high-frequency currents, together with techniques for constraining interference to meet EMC requirements. Separate ground and power-supply planes, together with low ESR capacitors that handle high-frequency current peaks, are mandatory. Although many EMC specifications ignore emissions below 30 MHz, the energy that a switch-mode amplifier radiates easily interferes with AM-radio receivers. Noise that couples back into the supply lines can radiate at frequencies that power-supply EMC specifications tackle (150 kHz). Output filtering is critical above a few watts of output power, and you must keep the trace length from the power device to the inductor as short as possible. Another technique to reduce emissions is soft-switching the power devices at the cost of a slight efficiency loss during partial conduction. All of these considerations compel vendors to provide reference designs and evaluation kits that help reduce development times.

### Mobiles drive low-end class D

With unchallenging audio performance but stringent efficiency and packaging requirements, batterypowered equipment is a prime target for class-D designs. Vendors such as Cirrus Logic, Maxim, National Semiconductor, and Texas Instruments (TI) offer devices for mono and stereo applications at 1 to 2W output-power levels (Table 1). TI's low-power amplifier-IC family includes the single-channel TPA2000D4 that primarily targets mobile handsets. Like most class-D designs, the device has differential inputs, and its outputs drive the loudspeaker in a bridge-tied-load configuration. Differential inputs provide optimum noise rejection for the analogue-input signal, and the bridge-tied-load configuration drives both speaker terminals from an H-bridge MOSFET stage. This driver arrangement permits dc coupling and delivers maximum load current from the device's 2.7 to 5V dc single-rail supply. Other representative features include output short-circuit protection to the load and either supply rail, a shutdown input that reduces quiescent current drain, and integral declick circuitry to suppress clicks and pops between active and power-down states. The device has four link-selectable gain settings and an open-loop configuration that makes feedback circuitry unnecessary.

The TPA2000D4's differentiating features include a modulation scheme that suits filterless operation. Carsten Oppitz, a strategic marketing engineer at TI, explains that conventional class-D modulation applies antiphase signals to the speaker. For no input signal, the filtered average voltage across the load is 0V, but considerable triangular-wave switching current still exists due to the speaker's resistive element. "The TPA2000D4's outputs are in-phase, with the duty-cycle difference between the outputs providing the drive signal. This arrangement also applies 0V for no input signal but removes switching currents due to the loudspeaker's resistive term, reducing power consumption and EMC emissions." If you have long pc-board traces or connecting wires, you may need ferrite-bead inductors to suppress EMC emissions less than 1 MHz, but traditional LC filtering is unnecessary. It's also unnecessary to rate the speaker for extra switching currents.

Cirrus has announced its CS44L10 headphone amplifier and expects to sample by the time you read this article, but no firm data was available at press time. The CS44L10 is the first member of a device family that shares a common technology base across wide output-power levels. Available now, Maxim's MAX4295/97 ICs include programmable PWM frequency selection from 125 kHz to 1 MHz that lets you balance audio performance against output-filter-component size. One input and one feedback resistor per channel establish circuit gain, as for an inverting op amp. If you require amplifier/headphone outputs, consider National's LM4663 IC that shrinks stereo dual-source selection, a  $2W/4\Omega$  amplifier, and an 80-mW headphone driver into a 24-pin SOIC package.

#### Families tackle wide power levels

Cirrus isn't alone in developing a technology family that tackles wide-ranging output-power levels. Behringer, Microsemi's Linfinity division, Philips, Tripath, and Zetex all offer products that control a few watts to as much as 2 kW. Controller ICs allow significant latitude in output power that's mainly a function of power-supply level and output-device selection. Behringer may be an unfamiliar name, but this company is a market leader in professional audio equipment and acquired its class-D technology from Intersil. Behringer currently offers the HCA8001A controller IC. However, Dirk Linnenweber, the company's digital-amplification-division director, promises an ambitious development road map. Plans for the next 18 months include an HCA8001 derivative, 75 and 200W gate drivers, and a magnetic drive module to extend power levels to 1 kW. Behringer currently offers two HCA8001A reference designs that you can licence and that require about \$20 of extra components for  $220W/4\Omega$  or about \$30 for the 1-kW version.

Philips' plans for switch-mode amplification exploit the company's silicon-on-insulator (SOI) fabrication technology. Today, the company is close to releasing its TDA8929 controller IC and the TDA8926/27 power stages for the 50/100W market. But Jean van Eeghem, marketing manager for home and personal audio ICs, anticipates a product line that spans power levels reaching several hundred watts. "Our SOI process allows us to integrate small-feature logic circuitry with high-voltage power devices to offer single-chip solutions at power levels that most competitors can't approach." Van Eeghem considers that the IC designers' main challenge is to economically integrate the controller-driver combination, which isn't easy when you remember that the TDA8927 continuously switches 6 to 7A peak currents at about 300 kHz. IC packaging and pin layouts are critically important to reduce emissions. The company's development experience compelled Philips to produce a reference design. "It can take 12 to 18 months for users to develop a class-D design from the ground up, which isn't acceptable in today's market. Manufacturers must provide support that slashes the development time frame."

Building on the company's expertise with switch-mode-power-supply controller ICs, Microsemi's Linfinity division is a semiconductor class-D pioneer. The company's original LX1720 is still available for \$2.95 (10,000), but Mike Tanaka, Linfinity's product-marketing manager, claims that this first-generation product is too noisy for today's market. Experience from the LX1720 contributes toward the 50W LX1711 and its low-voltage but quieter derivative, the LX1710. Major changes include swapping the LX1720's sawtooth modulation signal for a triangular shape, thereby improving distortion performance by as much as 50%. Linfinity also uses a proprietary output-device sensing technique that monitors operating conditions and dynamically adjusts switching times for optimum performance. Like several other vendors, Linfinity is understandably reluctant to discuss the details. But watch out for a new version of the LX1720 (tentatively, the LX1721) that will appear later this year, as well as an ultralow-power IC for hearing aids that extends battery life by as many as six times.

The reference design for Zetex's ZXCD1000 controller IC highlights class-D-driver considerations (Figure 3). The controller's PWM outputs are in-phase to achieve minimum switching-frequency ripple across the load, requiring antiphase input signals to provide the correct phase relationship at the speaker terminals. This approach also contributes a degree of distortion cancellation. An op-amp stage performs single-ended-to-antiphase conversion, ensuring the correct phasing for the ZXCD1000's audio inputs and providing supply-rail-independent dc biasing. The controller softswitches the output-power devices through a capacitively coupled level-shift stage that includes speedup diodes to drive the MOSFETs' gates. Gate-damping resistors prevent parasitic oscillations. The complementary ZXM64-series devices have gate capacitances of less than 1 nF that reduce drive requirements and power dissipation. The custom-built ZXFN1000 inductors provide linear response from quiescent to maximum output current that's essential to avoid distortion in the filter stage. The amplifier's noise floor is less than -105 dB, and frequency response is less than 0.4 dB across the audio band. Andrew Stewart, strategic marketing manager at Zetex, notes that many linear designs have open-loop-distortion figures of 0.5 to 1% that then require 20 to 40 dB of feedback to achieve acceptable results. "The ZXCD1000's novel design achieves a THD+N of less than 0.2% open loop or less than 0.1% with 10-dB feedback—significantly better than most other class-D solutions. And crossover artifacts are eliminated, so you don't need any dead-time correction."

#### Modulation schemes improve quality

With major mass-market design wins to its credit, Tripath's class-T technology combines analogue and switch-mode techniques to provide near-audiophile performance at consumer prices. Class-T efficiencies range from 80 to 90% with representative THD+N performance of less than 0.08% over the audio bandwidth and high-frequency intermodulation distortion of less than 0.04%. Tripath's architecture couples an analogue-signal buffer to an adaptive signal-processing stage that includes predictive processing and digital conversion. A proprietary modulation scheme adaptively drives the output devices with a variable-frequency waveform that resembles the spread-spectrum techniques that communications engineers employ. Today's class-T switching frequencies reach 1.5 MHz, and the average is about 600 to 700 kHz, reducing output-filter-component size. Scott Bobo, Tripath's applications manager, notes that the high switching frequency also minimises intermodulation

distortion. "Since we have a much higher switching frequency than class D, we reproduce the 19- and 20-kHz test tones with lower harmonic products. A fundamental with low harmonic content naturally reduces the harmonics after mixing. Also, our integration scheme provides very high loop gain and thus much better rejection of lower frequency harmonics" (Figure 4).

Backward-compatibility considerations compel most designers to accommodate analogue-input voltages before the switch-mode power-conversion stage, often adding an unnecessary A/D- or D/A-conversion step. Although most vendors expect consumer sound sources to be fully digital very soon, the slow take-up of digital audio broadcasting challenges this view. But vendors such as Cirrus Logic and TI respond that there's no shortage of high-resolution audio-frequency ADCs to perform glue-logic functions for legacy technologies. At the true audiophile end, it's clear that all-digital pathways extend the technology's reach from studio to loudspeaker (see <u>sidebar</u> "Digital amplifier displays audiophile quality"). Many elements from the development programme behind the amplifier in the <u>sidebar</u> now appear in TI's TAS5000/5100 ICs. The TAS5000 controller similarly employs PCM-to-PWM conversion that accepts serial digital-audio inputs. The company is reticent about describing its intellectual property but publishes much useful information on its Web site, including how to make switch-mode-amplifier measurements (Reference 2).

Cirrus Logic's upcoming device family also features serial digital inputs and a nonlinear delta-sigma modulator that performs PCM-to-PWM conversion. A synchronous sample-rate converter upsamples the S/PDIF (Sony/Philips-digital-interface) audio-data stream ahead of the modulator, which includes noise shaping and feedback circuitry. Output options include bridged and ground-referenced connections. Mike Taylor, the company's director of applications engineering, notes, "Our delta-sigma modulator resembles single-bit DAC techniques and overcomes the fundamental THD+N limits that sample-ramp linearity imposes in PWM designs." Cirrus tackles class D's intrinsic distortion-versus-dead-time trade-off with yet another proprietary technique for dynamically adapting switching times. Details are similarly sketchy, but you can assume that the technique monitors output-stage voltages and currents to automatically calibrate the power devices. As a result, feedback performance is independent of load impedance. The technique compensates for power-supply fluctuations and permits frequency locking between the controller IC and a switch-mode power supply to avoid beating. It's also compatible with active power-factor-correction schemes, which is a big consideration at high power levels. The resulting engineering prototypes demonstrate SNRs of better than 120 dB, to equal many first-class linear designs.

# Digital amplifier displays audiophile quality

Looking at today's sound recording and reproduction chain, any engineer is bound to question why he should convert analogue to digital then back to analogue, when each step introduces additional artifacts and circuit complexity. Ideally, the chain should remain digital from the recording studio's encoder to the consumer's loudspeaker terminals. At the audiophile end of the market, debate continues regarding the relative merits of the negative-feedback technique that Harold Black patented in 1937. Oft-cited problems include phase and transient response as well as unconditional stability into reactive loads. Feedback loops can't respond instantaneously, so no feedback means that there's less opportunity for slew-rate-induced distortion. Loads can present difficult impedances and also voltage reflections from low-frequency speakers that upset feedback amplifiers. Such considerations led engineers at TacT Audio to design the \$9700 Millennium amplifier, which is digital from input to output and which many regard as one of today's best power amplifiers (Figure A and Table A).

The heart of the Millennium is a proprietary PCM-to-PWM converter that dispenses with feedback loops in the signal path to guarantee freedom from any feedback-induced artifacts. This Equibit converter originates from Toccata Technology (now part of Texas Instruments) and is implemented with an Actel FPGA, partly for the FPGA's switching speed but primarily to secure the company's intellectual property. TI's TAS5000 controller IC offers similar conversion technology. In the Millennium, the master-clock frequency synchronously adapts to the audio datastream, and the timing sequence changes to reduce phase shifts at high audio frequencies. Peter Lyngdorf, TacT Audio's chairman, explains: "For 44.1-kHz CD formats, we upsample by eight times to 352.8 kHz, where we modulate to pulse width in 256 steps—corresponding to 8 bits. The resulting 90.316-MHz master clock is 2048 times faster than the CD format. For 96- and 192-kHz DVD audio, we upsample by four and two times, respectively, but still use 256 different pulse widths, giving us a

master clock of 98.304 MHz. Overall, the Equibit system's advantage is that it allows very high resolution with relatively little noise by minimising switching-polarity changes."

A passive-component filter with its –3-dB point at 65 kHz rejects sample-frequency artifacts and provides linear phase response throughout the audio band. The filter's second-order (–12-dB/octave) response minimises time-domain problems, such as ringing, which characterise higher order designs. Uniquely, the amplifier controls the upper 30 dB of its output level by adjusting the output devices' power-supply voltage, which can reach ±58V to deliver 165W into 8 $\Omega$ . At output levels below 30 dB, digital attenuation in the signal path provides low-level volume adjustment and preserves the Millennium's noise floor to allow greater than 134-dB dynamic range. With no feedback in the signal path, the variable-voltage volume-control approach requires an exceptionally quiet power supply to avoid noise coupling to the output terminals. The power supply is a three-stage design that cascades linear prefiltering and linear regulation ahead of the variable-voltage switch-mode stage that supplies the output rails; ripple rejection is greater than 138 dB.

Lyngdorf claims that the Millennium's principal design challenges were the Equibit converter technology, the power-supply system, and the output filter. The output devices are undisclosed components that operate with just 0.1% dead time. Also, a patented output-protection circuit can shut down in 1 µsec. Says Lyngdorf, "Before one of Toccata's engineers thought of this idea, any short circuit was guaranteed to destroy the outputs!" He notes that EMC concerns were challenging in the design phase. "Due to meticulous design, it is now easy to control EMC in terms of chassis design; we don't need any special shielding." Major contributors are the output devices' H-bridge design and millimetric-component-placement accuracy. Lyngdorf observes that the key to the Millennium's sonic performance lies with a total absence of ringing. "Our switching technique and our steady-state power supply quite simply mean that there isn't anything to under or overshoot." You can expect a lower cost but similar performance version of the Millennium later this year, when TacT releases its \$2995 M2150 amplifier.

For more information		
When you contact any of the following manufacturers directly, please let them know you read about their products in EDN Europe.		
Apex Microtechnology	Behringer	Maxim
www.apexmicrotech.com	www.behringer.com	www.maxim-ic.com
Microsemi (Linfinity)	National Semiconductor	Philips Semiconductors
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#### REFERENCE

- 1. Walker, PJ, "Current dumping audio amplifier," *Wireless World*, December 1975 (see <u>www.jps.net/shilohz/</u>).
- 2. Neesgaard, Claus, "Digital Audio Measurements," Texas Instruments, Copenhagen, January 2001 (see SLA114.PDF at <u>www.ti.com</u>).

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