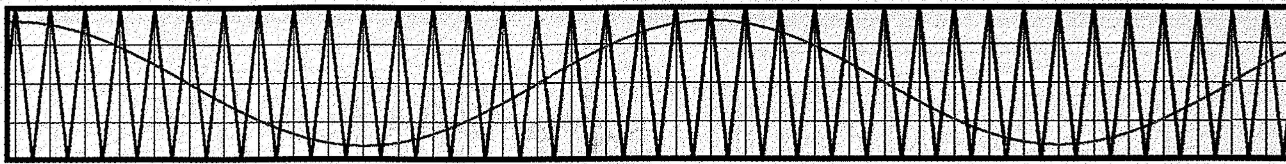

Audio Power Amplification

By Staff Technical Writer

To many people, power amplifiers are seen as a necessary but rather uninspiring part of the audio signal chain—black (or perhaps silver) boxes that make small signals bigger. Nothing should be simpler, one might think. Until recently amplifier technology had remained relatively unchanged for many years, with any advances involving what was really a number of variations on a theme. For a long time there had been no fundamental paradigm shift in the way in which amplifiers make small signals bigger. Now, however, a quiet revolution is taking place.

Owing mainly to the need for highly efficient amplifiers that can be incorporated into compact devices, switching amplifiers, sometimes called digital, Class D, or pulse width modulation (PWM) designs, are gradually taking up a larger share of the market. For example, in "The Industrial Dynamics of Open Innovation: Evidence from the Transformation of Sound Amplification from Linear Solid-State to Class D Technology" (AES 27th Conference paper), Christensen and Olsen quote market research that shows the analog amplifier market has a value between $2.1 to $3 billion in 2003. In the same year the digital amplifier market accounted for only 2 to 3 percent of that total, or around $100 million. However, by 2006 the digital amplifier share is expected to rise to $515 million or 15 percent of the total market. Another analysis shows the potential market for Class D amplifiers rising to $800 million by 2008. Clearly this technology is here to stay now that initial difficulties in designing practical systems have been addressed.

Research now centers on optimizing performance, power efficiency, and compactness. New knowledge and skills are required of designers, and the procedures for optimizing performance are somewhat different from those required for traditional analog amplifiers. In this article we describe some recent developments in this field, looking mainly at papers from the AES 27th International Conference in 2005 on efficient audio power amplification.

WHY THE NEED FOR NEW AMPLIFIER TECHNOLOGY?

Traditional amplifier designs have worked successfully for many years, and it may be hard for some people to fathom the need for so-called digital amplifiers. Loudspeakers, after all, are fundamentally analog devices, aren't they? Indeed the field seems to be quite strongly polarized between those who favor traditional designs and those pioneering the new approach. Initially it seemed that PWM-based amplifiers simply promised more complexity and higher distortion than their traditional counterparts, and the need for something digital in an essentially analog part of the signal chain could have been questioned. That, however, would have been missing the point, and applying the term digital to such devices may have been confusing. As Andersen points out in the AES 27th paper "Efficient Audio Power Amplification—Challenges," the primary motivation for such developments during the 1980s and 1990s was the increased availability of compact consumer audio and video equipment, together with an apparent demand for ever higher output powers from related audio amplifiers. He also points to the increasing trend toward the employment of active instead of passive loudspeakers. The current prevalence of compact but highly powered subwoofers in consumer home cinema systems and 5.1 studio installations is a case in point.

The problem is that if high-powered amplifiers are to be incorporated into small spaces within compact products, traditional designs no longer fulfill the requirements. It is impossible to make traditional devices small enough, partly because they tend to need large heat sinks to dissipate the heat that results from their relatively inefficient methods for converting electrical power into signals for driving a loudspeaker. Self points out in his *Audio Power Amplifier Design Handbook* (Newnes 2002) that Class B amplifiers represent by far the most popular mode of operation and "probably more than 99% of amplifiers currently made are of this type."

The Class B amplifier operates in such a way that each of a pair of output transistors is active for half of the audio cycle time, so considerable power is dissipated. Some variations on the Class B theme have been attempted in a quest to improve efficiency, such as using more transistors in the output stage and a division of the supply voltage across different pairs so that only specific pairs are carrying current at any time. The Class G amplifier, developed by Hitachi in 1976, extended the principle to switching the power sources under the control of the input signal, but this apparently just moved the power loss problems from the output stage to ➡

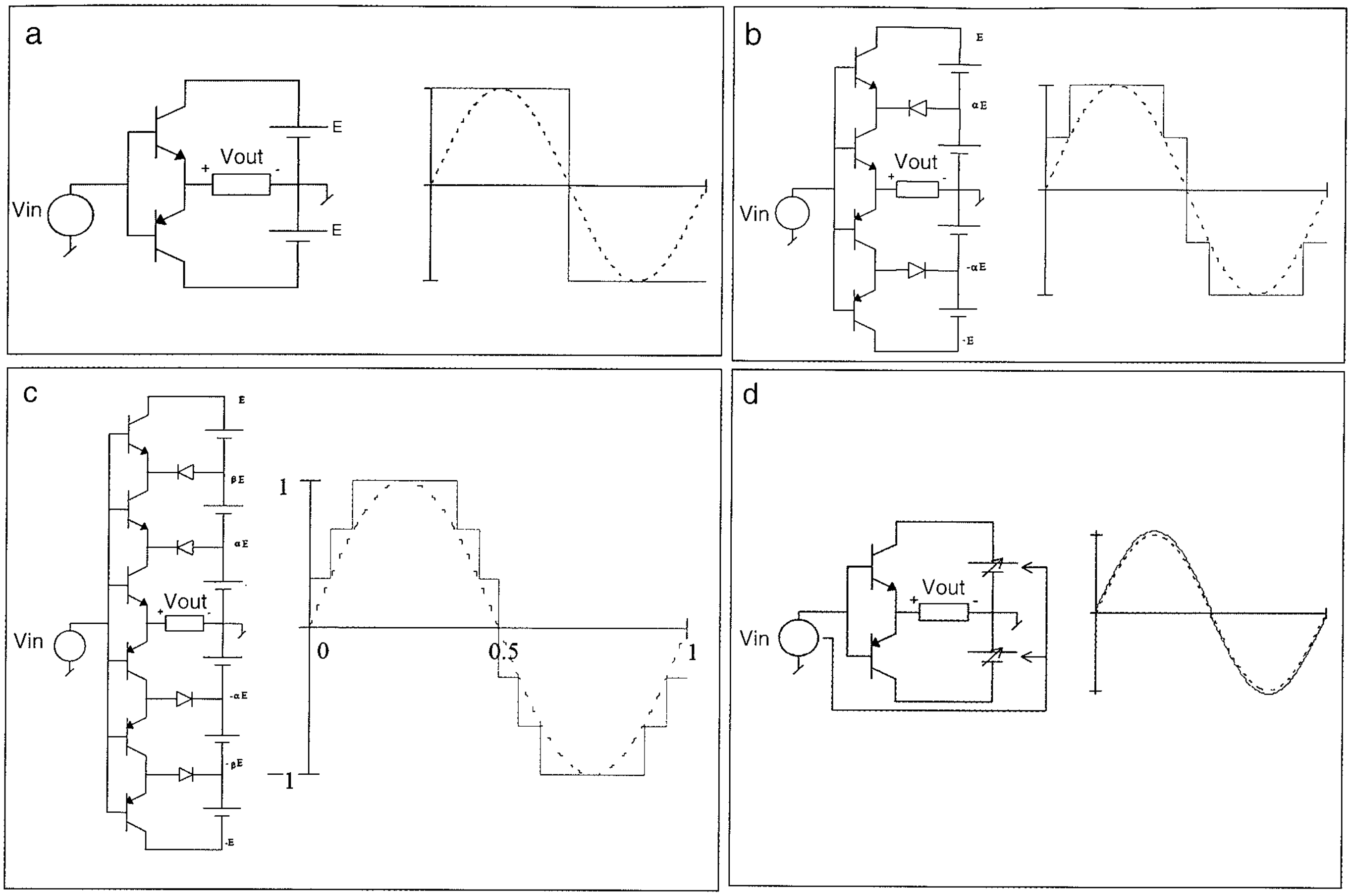

Fig. 1. Some amplifier topologies aiming at increasing power efficiency: (a) Class B; (b) Class B2; (c) Class B3; (d) Class G (Figs. 1–3 courtesy Andersen).

the power supply. Some comparisons of these approaches, as described by Andersen, are shown in Fig. 1.

The Class D design, pioneered by Sony in the 1970s, used the audio signal to modulate the pulse width of a high-frequency pulse train, and required a filter at the output stage to remove noise and high-frequency switching components, as shown in Fig. 2. Early designs suffered from inadequately fast power transistors. However, the efficiency of Class D designs compared with other approaches appears to be indisputable. Andersen provides a graph showing efficiency versus relative output level for five amplifier classes in Fig. 3.

IMPROVING THE PERFORMANCE OF SWITCHING AMPLIFIERS

As Lambruschi et al. point out in their AES 27th paper, "A 97% Efficiency Power Stage Design for High Performance Class D Audio Amplification," typical Class D power amplifiers are built using a so-called totem-pole configuration of N/N MOSFETs (see Fig. 4). A key problem to be solved in design is that of how to eliminate simultaneous conduction by the two transistors at the switching point, resulting either from nonsimultaneous switching times or the effect of parasitic capacitance (Miller effect). This can be tackled by inserting "dead time" at the switching point, on the order of tens of nanoseconds, but this does not solve the whole problem owing to side effects. The result can be considerable power loss and distortion. By adopting a source-follower configuration, as shown in Fig. 5, the gates of the upper and lower MOSFETS are connected together, which, it is claimed, eliminates the simultaneous conduction issue. The gates of the two MOSFETs are also reverse biased during their off state, which decreases their turn-off time. A design based on this approach, with a power output of 150 W is said to be 97 percent efficient; its THD versus output power measurements are shown in Fig. 6.

Antunes et al., in their AES 27th Conference paper "Digital Audio Power Amplifier Based on Multilevel Power Converters," explain that converting the PCM audio signal from a Compact Disc directly to PWM for amplification would result in pulse widths as short as 0.346 ns, which is far too short for existing semiconductors. As a result, most amplifiers tend to resort to lower resolution, but compensate for the side effects by employing aggressive noise shaping and interpolation. For example, it would be common to employ a PWM modulator running at 352.8 kHz with eight bits of resolution (see Fig. 7). In all practical cases considerable distortion can be introduced in the process of conversion from PCM to PWM because the process is nonlinear. Typically, the distortion added as a result of the inadequacy of the output stage switching of such a power amplifier is dealt with by using low-pass filters and analog feedback loops, but this rather contradicts the idea of an all-digital power amplifier.

The solution suggested by the authors is to replace the typical output stage that consists of a full bridge inverter (capable of producing a bipo-

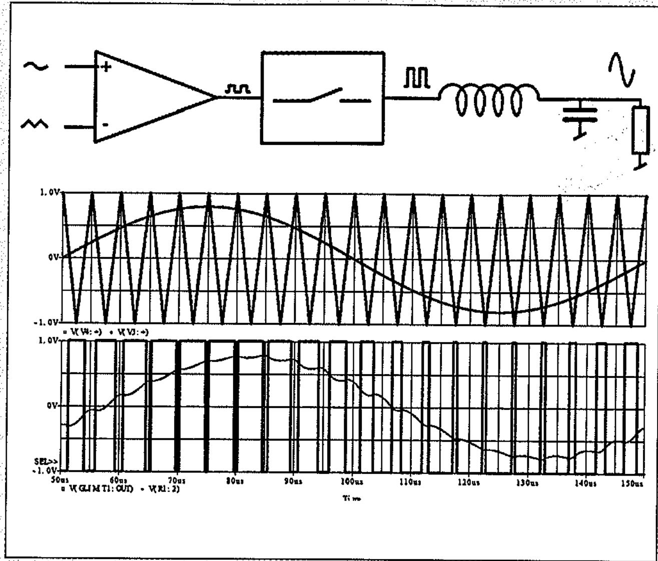
Fig. 2. Class D principle

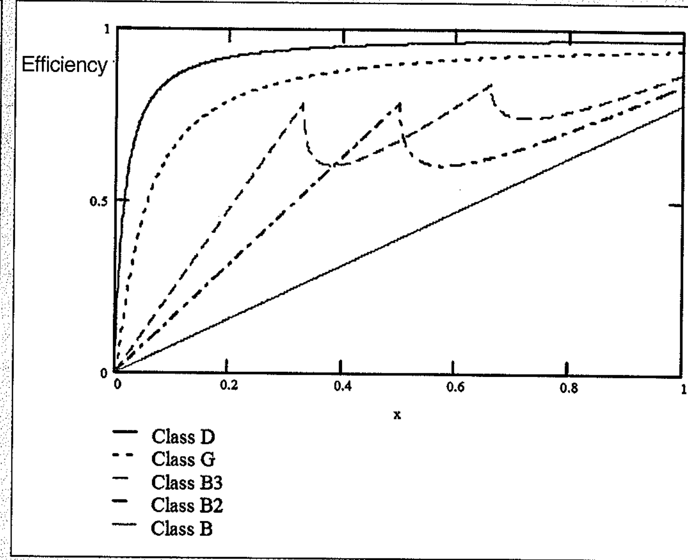

Fig. 3. Comparison of efficiency versus relative output level for different classes of amplifier

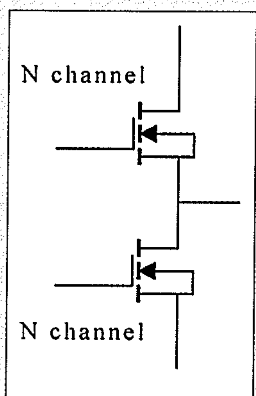

Fig. 4. N/N MOSFET totem-pole configuration (Figs. 4–6 courtesy Lambruschi et al.)

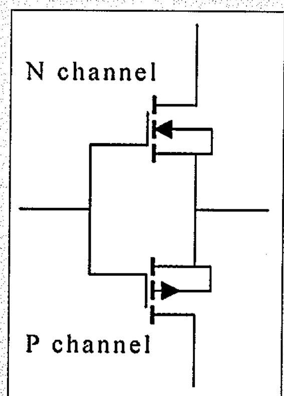

Fig. 5. Source-follower MOSFET configuration

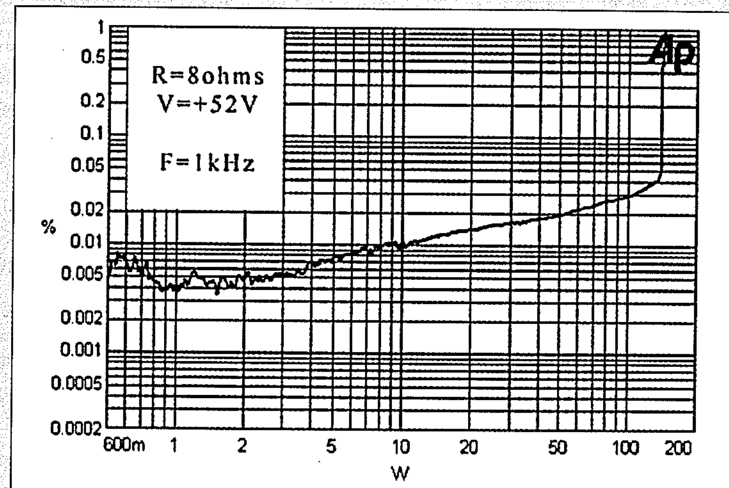

Fig. 6. THD versus output power of source-follower-based power amplifier

lar output with an intermediate zero state) with a multilevel power inverter, such as the basic version shown in Fig. 8. The output waveform is obtained by summing the outputs of each cell, and it is possible to minimize the number of inverters employed by using unequal voltage sources. Some examples given include voltages in geometric ratios of two or three. Using a ratio of three, a system with nine levels is possible, (waveform shown in Fig. 9). The authors use this approach to power amplification in conjunction with a novel approach to the conversion between PCM and PWM, known as narrow pulse elimination technique (NPE). When distortion is introduced due to nonideal output switches, it affects narrow pulses more than wide ones. Therefore the authors devised a technique that puts restrictions on the maximum and minimum duty cycle of PWM pulses. Naturally this introduces errors in the conversion from PCM that have to be corrected, so they calculate the PCM value of the error signal arising from the approximated PWM conversion and use it as the feedback signal in a 2nd-order sigma–delta noise-shaping modulator.

Using NPE the performance of the multilevel digital power amplifier was improved considerably. Without NPE the noise floor lay at minus 90 dB, and the THD+N figure was 0.2 percent.

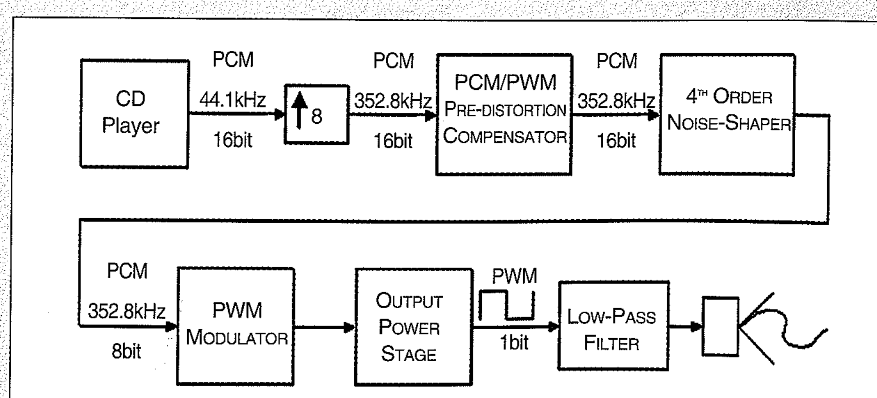

Fig. 7. Typical simplified block diagram of a digital audio amplifier (Figs. 7–9 courtesy Antunes et al.)

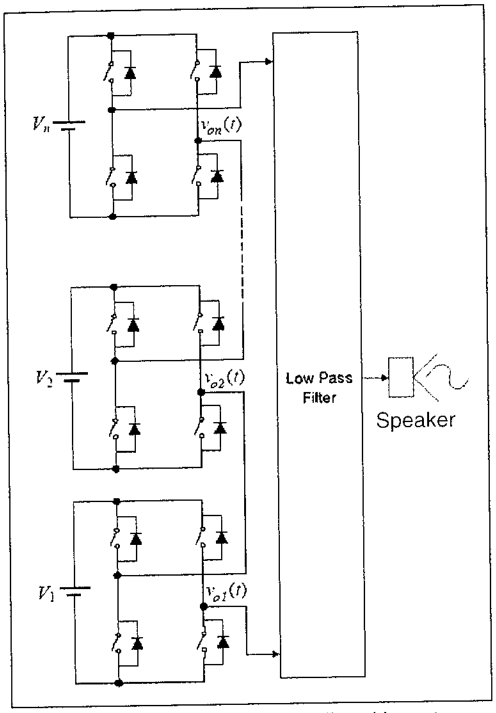

Fig. 8. Basic structure of a multilevel inverter with n series connected full bridge cells

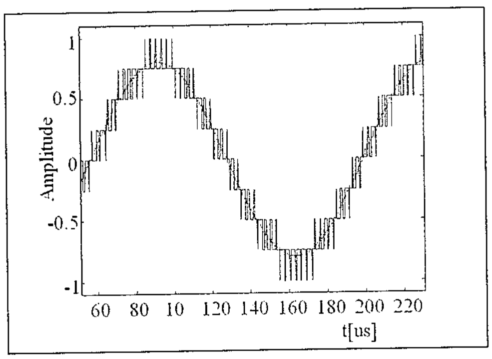

Fig. 9. Typical output voltage waveform of a multilevel inverter with nine normalized levels

Using NPE the noise floor was reduced to minus 100 dB and the THD+N to 0.089 percent. Experiments on a model built using these techniques showed results that compared favorably with several commercial high-end amplifiers.

In systems such as these the two parts consisting of switched-mode power supply and Class D amplifier are often treated separately. Direct-energy-conversion power amplifiers are a means of modulating a power supply with an audio signal, and as such represent a total integration of switched-mode power supply and Class D audio amplifier, as suggested by Ljusev and Andersen in their AES 27th Conference paper, "Self-Oscillating Modulators for Direct Energy Conversion Audio Power Amplifiers." In the SICAM (single conversion stage amplifier) project, described by Ljusev and Andersen, a direct-energy-conversion audio power amplifier is developed from the HF-link power converter topology, which has been proposed for use in uninterruptible power supplies. It differs from a typical Class D power amplifier as shown in Fig. 10, where it will be seen that the LPF and capacitor marked B are removed, thus improving integration of the device and removing bulky components. The authors describe the use of self-oscillating modulators in SICAMs, as opposed to PWM modulators with a carrier frequency derived from an external clock. Such self-oscillating modulators can operate in either a normal variable-frequency mode or a locked constant-frequency mode. The performance of the system is analyzed in both modes.

SWITCHING AMPLIFIERS FOR HIGH-RESOLUTION AUDIO

During a workshop at the AES 119th Convention last year in New York, Vicki Melchior chaired a panel of experts who discussed the application of switching amplifiers in high-resolution audio. She explained that a PWM amplifier can be used with analog or digital (LPCM, linear pulse-code modulation, or DSD, direct stream digital) inputs. DSD precludes the need for any sort of preamplifier or separate D/A converters. Modern amplifier technology of this sort is now comparable in performance to traditional designs using Class AB.

Panelist Karsten Nielsen of Bang & Olufsen ICEpower pointed out that such power amplification involves an essentially analog process, although by virtue of its switching operation it might be thought of as digital. The filter in a typical switching amplifier is very sensitive to the load impedance, its Q being dependent on this factor, so this part of a system tends to be problematic and a cause of electromagnetic interference (EMI), which is also generated by other parts such as the modulator. Filterless technologies are based on synthesizing multilevel pulse outputs and feeding them directly to the transducer. He proposed integrating such processes with the transducer architecture, which would represent a distinct paradigm shift in system design. Components could be mounted within the magnetic structure of the loudspeaker, for example, providing residual cooling features.

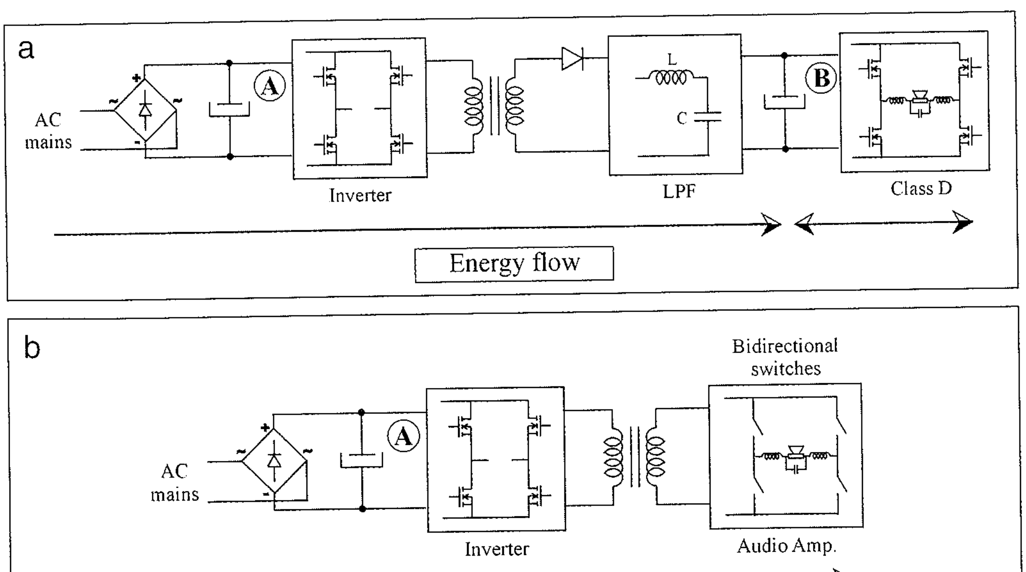

Fig. 10. (a) Conventional Class D audio power amplifier; (b) Direct conversion audio power amplifier (courtesy Ljusev and Andersen).

The papers reviewed in this article, and all AES papers, can be purchased online at http://www.aes.org/publications/preprints/search.cfm and http://www.aes.org/journal/search.cfm. AES members also have free access to a large number of past technical review articles such as this one and other tutorials from AES conventions and conferences; go to http://www.aes.org/tutorials/.

Nielsen described a number of different classes or types of switching amplifier, concluding that much recent research has been centered on a fifth class of so-called feedback oscillating modulators, similar to the SICAM approach mentioned above. In other classes of amplifier relying on digital modulators it is difficult to get the dynamic range to be better than 110 dB. In amplifiers based on feedback oscillating modulators, the modulator is set into a self-oscillating mode by means of a 180° phase shift in the feedback loop, and noise shaping is used to process the noise spectrum. The switching frequency is variable, such that when currents are high the switching frequency is low, and the modulation frequency looks like a sine wave. Distortion components below 0.01 percent at all frequencies are possible using such a design, and extremely high dynamic ranges result. In the final reckoning, however, he noted that the primary challenge is nearly always to realize a practical design that can be implemented and approved for EMI regulations. The more sophisticated the system, the more problematic the EMI issues generally become. This is a major engineering challenge and not to be taken lightly.

Skip Taylor of D2Audio offered a different view on the problems discussed by Nielsen. In the real world some of the biggest constraints that designers run into in systems generating over 50 W of power are a result of imperfect components, such as unmatched output FETs. As mentioned above, the timing of the "dead time" is a crucial factor, as are the various filter components. In open-loop amplifier designs there is inherently no power supply rejection, but power supply interference can be cancelled to some degree (up to 40 dB as a rule) by using full-bridge as opposed to half-bridge output stages. Such solutions are inherently stable. In closed-loop designs the problem is somewhat different, but nonlinearities are still present and contribute to instabilities in the feedback loop. Most of the closed-loop designs currently in existence use analog feedback approaches, and a common problem is one of how to manage the delay in any loop that employs digital feedback. The impedance characteristic of the output load, and any protection circuitry employed in the loudspeaker, is one major factor to be taken into account in this respect; a means of dealing with such matters in a graceful fashion is a key challenge. Users would not, for example, be content with an amplifier that closed down every time it encountered a load that it did not like.

Taylor described a feed-forward approach that attempts to model and characterize the various distortions inherent in a digital amplifier and feed them forward so that they can be corrected using DSP. Jitter, for example, is a major problem and is one of the main aspects to be addressed. 10 ns of jitter can give rise to a 77 dB noise floor, but the use of a digital sampling-rate convertor can lower the noise floor to over 100 dB and make it independent of the jitter. A real-life switching waveform, for example, is not a nice tidy square wave. The varying impedance characteristics of the output load can be modeled and fed into the PWM modulator as correction factors, as can nonideal aspects of the output filter. Power supply characteristics can also be modeled and incorporated in a similar way. Once the system model is built it can be run as an open-loop design. Even if a closed-loop model is eventually chosen, it seems important to solve the open-loop problems first so that any nonlinearities are dealt with up front. By means of this type of adaptive DSP modeling, THD+N figures can be reduced to around 0.013 percent.

Lars Risbo, manager of a strategic research division of Texas Instruments in Denmark (formerly Toccata Technology), commented that he had so far tended to favor the open-loop design approach. This was partly because digital gates are cheaper and cheaper these days and audio quality can be made high when a system is simple. However, he wished to concentrate on how to bring Class D systems to the mass market. Only recently had they been made available at a low cost in modular form. Cost-effective integration into some sort of IC form is important, including a lot of functionality to suit the consumer market. On an IC it would be possible to have fast clock rates without giving rise to EMI problems. Furthermore, DSP would be easy to implement and cheap. But passive devices such as inductors and capacitors will be much more difficult, large, and expensive. Timing is preeminent in a digital amplifier, but the crystal oscillators that would be used are expensive to implement on chips; even an extra $10 to $20 on the price can make or break a design.

John Oh of Pulsus Technologies in Korea tried to alleviate potential concerns that digital power amplification is too difficult for the average engineer to implement. His company has always tried to provide easier and more inexpensive solutions to the customer, he said, and there are lots of options to make this possible with digital amplifiers. Most of the leading home theater manufacturers tend to go for a fully digital integrated solution in their systems, for example, partly because the manufacturing process is easier to control. He showed an example of a 6-channel 100cm^2 power amplifier module with 100 watts per channel, which is a remarkable achievement. Equalization and other DSP functions such as bass management can be handled on board such an integrated system, as well as high-resolution audio inputs up to 192-kHz sampling rate.

CONCLUSION

Given the rate at which Class D and related approaches to amplification are improving in the commercial field, it seems likely that their market share will continue to increase, not least because of device compactness and power efficiency. Highly powered amplifiers no longer need to generate excessive heat and weigh as much as a pile of bricks.