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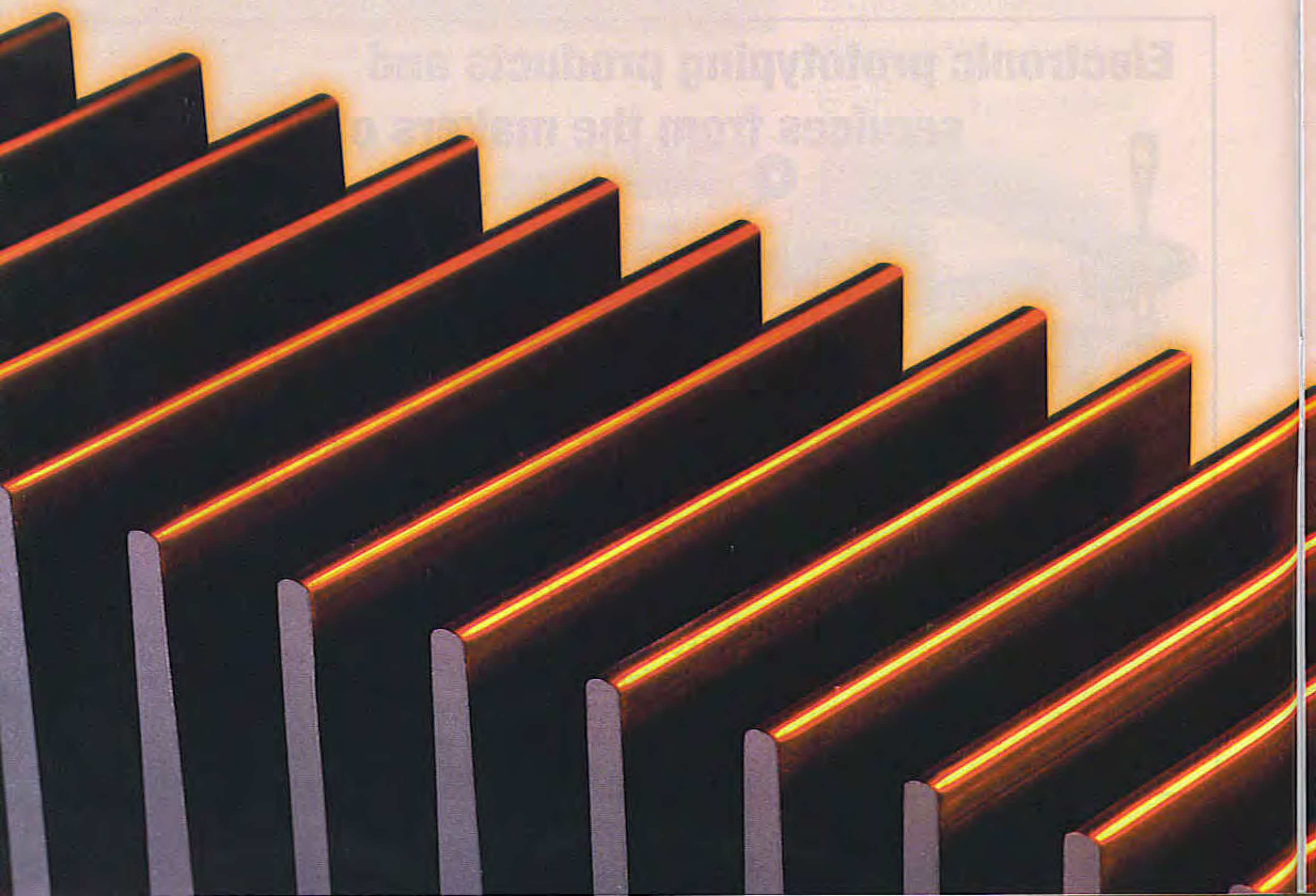
THAT'S CLASS...

audio amplifiers from A to T

The final amplifier is the power source of every audio installation. Its job is to convert a small alternating voltage into a powerful signal suitable for driving loudspeakers, with as little distortion as possible. During the years since the invention of electronic audio systems, designers have come up with various approaches to this problem. It all started with Class A...



A little more noise, a lot more power

For many people, the amount of power an amplifier can produce is an important factor in judging its characteristics (So your amplifier delivers 2 × 40 watts? Mine does 2 × 70!). But in practice, power plays only a minor role.

You can already make a lot of noise with just a few watts. If you use a set of loudspeakers that can produce a sound pressure level of 86 dB at 1 W (which is a value commonly stated by manufacturers in speaker specifications), you can manage 90 dB with just 2.5 watts. With 25 watts, you have enough for 100 dB. That's already rather loud (and harmful to your ears as well!).

Our ears perceive each 6-dB increase in the sound pressure level as a doubling of the volume level, but this requires increasing the power by a factor of four. This means that if you really want to have a bigger final amplifier with more power than what you presently have, you will need an amplifier with at least four times as much power to actually notice any difference.

Delivering a large amount of power is not a simple task for an amplifier. Voltage amplification and current amplification are both necessary in order to provide sufficient power to the speakers connected to the amplifier. This is because loudspeakers have an efficiency of only a few percent, which means that several watts are certainly necessary to generate an adequate sound pressure level in a living room. In the case of concerts or outdoor events, quite a bit more is required, and the necessary power can easily amount to several kilowatts. To produce power amplification in a final amplifier, various concepts have been developed for using transistors or FETs to generate high-quality output signals and/or improve the efficiency of the output stage. (Here we leave valve amplifiers out of the picture.) When devising an output stage, the designer must take into account the specific properties of the semiconductor devices being used. If we could work with 'ideal' transistors or FETs, it would be much easier to build good amplifiers. Unfortunately, all semiconductor devices suffer from non-linearity in their amplification characteristics, and this causes major problems, especially for processing analogue signals. These problems can be minimised by using properly dimensioned feedback. There are also other nasty side effects that also occur, depending on the selected configuration, such as the notorious problem of crossover distortion.

Especially with large amplifiers, heat generation is another factor that must be taken into account. It can lead to far-reaching thermal effects, such as drift of the quiescent current setting and thermal modulation distortion.
Final amplifiers are normally classified according to the configuration of the output stage. This largely determines their efficiency and quality, since the output stage is where the actual power amplification takes place.

The various amplifier configurations are designated using the letters of the alphabet, although the letters do not say anything about how they work. It all just started with the first letter of the alphabet.

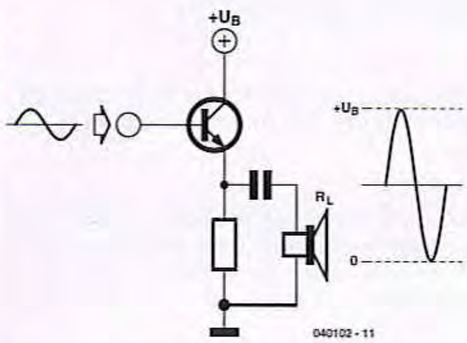


Figure 1. A Class A amplifier has very low efficiency, but it is totally free of crossover distortion.

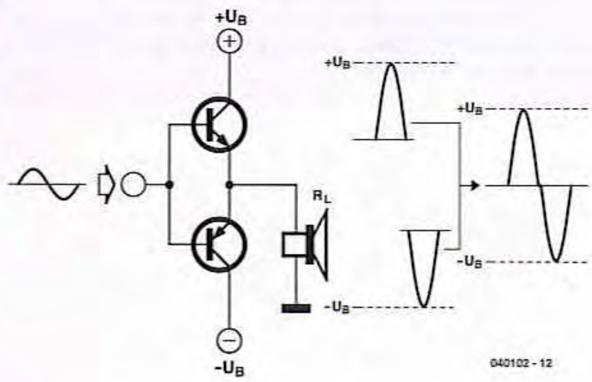


Figure 2. In a Class B configuration, each transistor conducts for half of the sine-wave cycle. Here problems arise around the zero-crossing point.

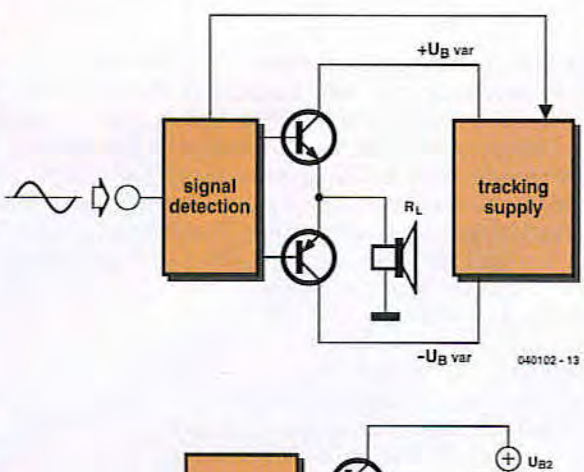


Figure 3. Class G uses a tracking power supply whose voltage is continuously adjusted to match the signal amplitude.

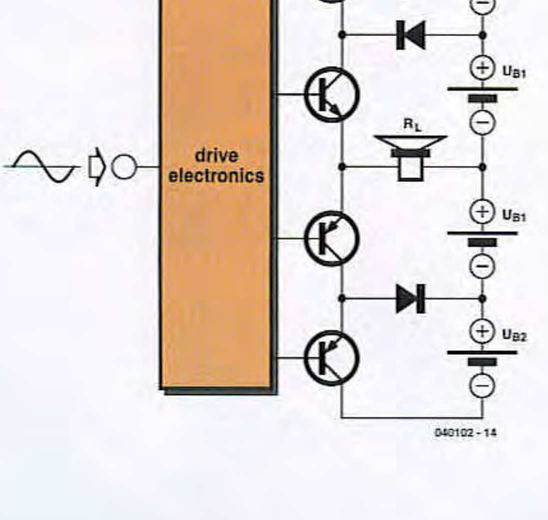


Figure 4. Class H limits itself to switching between several different supply voltages (in this case two).

Class A

Here we begin with the simplest configuration, the Class A final amplifier, which is also one of the best configurations for high-quality audio reproduction. In its basic form, this configuration can be implemented using a standard emitter follower (**Figure 1**). The quiescent current through the transistor is equal to the peak AC output current, which means that the transistor is biased in the middle of its working range and simply conducts more or less current when driven by an alternating voltage. The efficiency is very low: 25 % at maximum output amplitude, and even less at low signal levels. The efficiency can be improved by using a symmetrical design with two transistors, but even then the highest efficiency that can be achieved is 50 %.

Class B

The Class B configuration employs two transistors, each of which conducts for exactly half of the signal cycle (**Figure 2**). In the quiescent state, no current at all flows through the transistors. The efficiency of a Class B output stage is around 78 %, but the primarily disadvantage of this configuration is the 'transfer distortion' that occurs each time the load must be transferred from one transistor to the other. Due to the sharp bend at the bottom end of the transfer characteristic, the two halves of the signal waveform do not properly align with each other. This leads to the notorious problem of crossover distortion, which is a quite audible degradation of the signal waveform.

To solve this problem, Class A and Class B were combined to produce Class AB. This is a Class B configuration in which a small quiescent current flows, causing the output stage to actually work in Class A at low power levels. This approach is presently used in various forms in most final amplifiers. The efficiency remains approximately the same as for Class B.

G and H

Hey, just a minute! Haven't we skipped a few classes? We have indeed, but we did so on purpose. Classes C, E and F also exist, but they are actually only suitable for high-frequency applications, which means they more or less fall outside the scope of what we're talking about here. And the design of Class D amplifiers is so different from Class A and Class B that it we decided to deal with it separately. So, let's first look at classes G and H, which have an important feature in common. This is that in both of these classes, the supply voltage is adjusted according to the magnitude of the output signal. In Class G (**Figure 3**), the supply voltage is continuously adjusted to match the desired amplitude of the output signal. Such a 'tracking' supply voltage can be implemented relatively easily using modern switching power supplies, although it is of course important to have a good regulator circuit to allow the supply voltage to respond sufficiently quickly to changes in the amplitude of the signal generated by the output stage.

In Class H (**Figure 4**), what happens is essentially the same as in Class G, except that here the supply voltage is switched between several distinct levels (usually two) instead of being

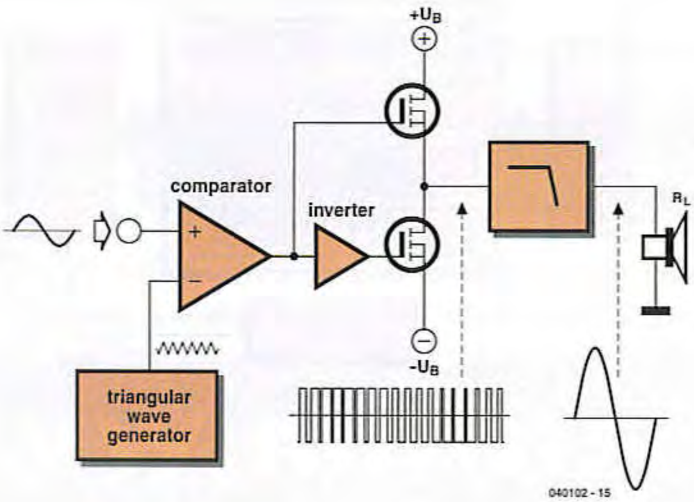


Figure 5. A Class D amplifier consists of a pulse-width modulator with a power output stage and a low-pass filter.

continuously varied. This allows the dissipation in the output stage to be considerably reduced, especially when large amounts of power are involved.

Class D

With the Class D amplifier configuration, the letter 'D' doesn't have anything to do with 'digital' (that's just a coincidence). It refers to a switching amplifier that uses pulse-width modulation (**Figure 5**). The input signal is compared with a triangular waveform, and the signal from the comparator switches the output stage to the positive or negative supply voltage. This is done using a very high switching frequency, which is usually ten times or more higher than the audio bandwidth (which means 200 kHz or above).
With this form of modulation, the pulse width depends on the level of the input signal. If a low-pass filter is placed after the output stage, the pulse-width signal is integrated and what is left is an analogue signal with the same form as the input signal, but of course amplified.
As the output stage only has to switch, its efficiency is very high. However, there are also a number of drawbacks to this approach. It is rather difficult to keep the signal waveform free of distortion, a hefty output filter is required, and drastic measures must be taken to limit radiated interference. For low-distortion amplification, it is always necessary to use negative feedback (analogue or digital).

Classes S and T

Although the working principle of the Class D amplifier is already several decades old, it never managed to become truly established in hi-fi applications. This was primarily due to excessive distortion and a lack of good semiconductor devices (fast power FETs). In the meantime, several manufacturers have devised variations on this theme, and in many cases they have given them their own designations. For instance, Crown came up with the Class I amplifier, Sony developed its S-Master technology, and Tripath has devised its Class T amplifier. Unfortunately, the nice alphabetic sequence has been abandoned in favour of manufacturer-specific abbreviations.
In its S-Master technology, Sony combined several techniques to make the Class D configuration suitable for domestic hi-fi applications. Here the process of converting the incoming signal into a corresponding pulse-width signal is called 'complementary pulse length modulation' (C-PLM). Extensive atten-

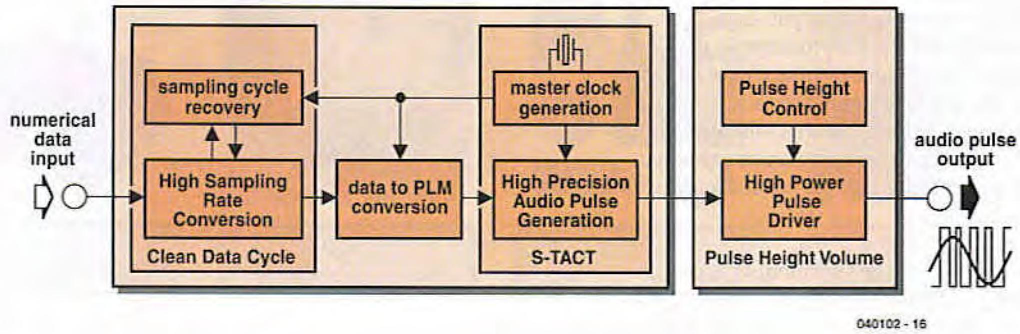


Figure 6. Block diagram of a Sony S-Master amplifier.

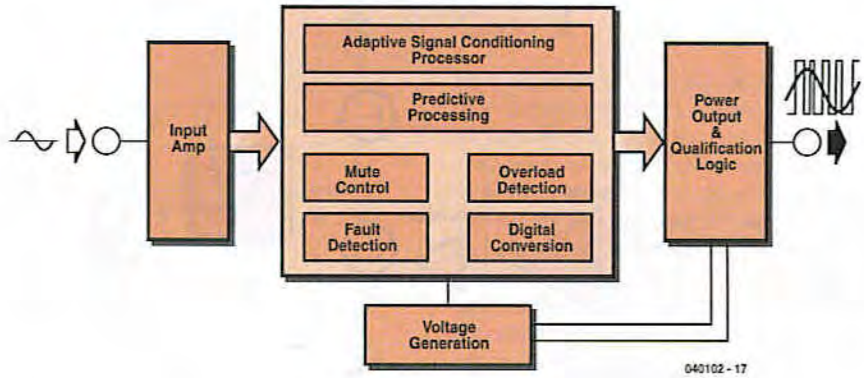


Figure 7. The Tripath Class T amplifier is an 'intelligent' elaboration of the Class D principle. Here a processor continuously monitors the input signal and adjusts the switching signals.

tion was given to suppressing jitter. This was accomplished by using an extremely accurate clock signal and a circuit called 'clean data cycle' that corrects the positioning of the output pulses if necessary (see **Figure 6**).

The method used to implement volume control is certainly an unusual feature of the Sony approach. In a normal Class D design, the full pulse waveform is always present at the output, with an amplitude of 50 to 100 volts peak-to-peak. Particularly with small output signals, it is very difficult to fully eliminate all residual components of the pulse waveform from the filtered signal. In the Sony design, the volume is regulated by adjusting the supply voltage for the output stage. This prevents any information from being lost at low signal levels. This technique has an effective range of 50 dB.

Another company, Tripath, has developed a technique that according to them combines the signal quality of Class A and AB amplifiers with high efficiency (around 80–90 %). This is done using a combination of analogue and digital circuitry, together with digital algorithms that modulate the input signal using a high-frequency switching waveform. The algorithms developed by Tripath are derived from adaptive and predictive algorithms already used in telecommunication systems.
With the Tripath amplifier, the majority of the analogue and digital circuitry is housed in a single IC, which may also include the output transistors (depending on the power). The block diagram of the amplifier is shown in **Figure 7**. The input signal is first buffered by an input stage. From there it passes to the Digital Power Processing block, which contains the signal processor, a digital conversion function, mute switching, overload protection and error detection. The output stage is driven via the qualification logic, and the loudspeaker is connected to a filter following the output stage.
Thanks to its special algorithms, the processor in the Class T amplifier can shape the drive signal for the output stage to match the specific characteristics of the transistors used in that stage. This shaping takes into account non-ideal switching behaviour, mismatching of the complementary output transistors, dead-time distortion and the residual energy of the oscillator in the audio band.

The switching frequency of a Class T amplifier is continuously adapted to the magnitude of the input signal. At low input levels, the switching frequency is quite high (around 1.2 MHz). This has a beneficial effect on signal quality. The switching frequency gradually drops as the input level increases, in order to increase efficiency. The switching frequency ultimately reaches its lowest value (around 200 kHz) when the output is driven to maximum amplitude. Besides this, a form of noise shaping is applied to the peaks of the output signal in order to improve the signal waveform. As a result of all these measures, the Class T amplifier can deliver a sound impression that reminds listeners of audiophile analogue amplifiers.

The future

With the steady advance of digital audio, some form of digital output stage will ultimately be found in many consumer amplifiers. The reasons for this are higher efficiency, smaller size and lower manufacturing cost. It's difficult to estimate whether this development will also come to prevail in the high-end realm. There are presently only a few high-quality digital amplifiers on the market. But if you'd like to try it for yourself, you can start by building the Clarity amplifier described in this issue.

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