Class Distinctions

Understanding power amplifier topologies

By Tommy O'Brien

The class of an audio power amplifier describes the way its output devices are used. Back in the mid-1900s, amplifier classes were defined as Class-A, Class-B, Class-D and several others (Class-C, -G, -S, etc).

Class-A, Class-AB, and Class-D are the primary topologies found in modern amplifiers, with the other classes largely variations of the three fundamental topologies. Let's briefly look at the basics of each design.

Figure 1 shows an amplifier output stage. A resistor (RL) is the load. However, real loads are loudspeakers, and present a reactance – not a pure resistance – to the amplifier's output.

A reactive load can sink or source current, independent of instantaneous voltage. An amplifier's ability to control its output into a reactive load contributes significantly to its sonic character. Other factors, such as linearity and transient response, also play roles in sound quality.

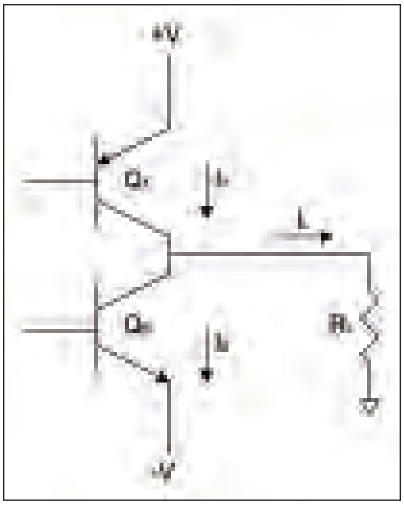

Figure 1: Basic amplifier output stage.

Also note that the active devices in **Figure 1** are shown as bipolar transistors. This is not always the case, although the majority of amplifiers are still outfitted with such devices due to their maturity and low cost. MOSFETs and vacuum tubes are examples of alternate output power devices.

All classes of amplifiers can be divided into two broad categories: linear and switching amplifiers. Class-A and Class-AB designs, along with their variants, are linear, while Class-D designs are switching.

In fact, the formal definition of a Class-D amplifier requires only that the output devices be used as switches. Many fictitious class designations have been conjured up by switching amplifier manufacturers, such as Class-T, Class-J, Class-X, etc, but they are still simply variations of Class-D amplifiers. Other non-class names have been given to amplifier topologies, such as BCA, which is a Class-D topology, and BASH, which is a Class-AB topology.

Figure 1 also depicts a half-bridge topology, with bipolar power supplies. Other variations exist, such as bridging, which drives both sides of the load using two output stages for double the effective output voltage swing.

Multiple output stages connected in parallel (to share the output current) are also used in some cases to allow inexpensive transistors in high-power designs. Whatever the "trick," these special adaptations still fall within the three main classes of audio amplifiers, and we will refer to them simply as "variations."

If linear amplifiers were 100 percent efficient, there would be no reason for this discussion. It's generally assumed that there is a compromise between efficiency and performance (qualitative or quantitative), and that one must trade performance for efficiency based on the application, but in light of recent technology advances, this is a generalization. The missing factor here is cost, which can be loosely correlated to efficiency and performance.

MAXIMUM POWER OUTPUT

Efficiency can be measured several ways, but the official definition of efficiency is power out divided by power in. The problem arises when choosing a power level for measuring efficiency. The efficiency of an amplifier changes over frequency, amplitude, temperature, and even signal history.

In addition, "real source material," such as music and speech, are rarely used to measure efficiency due to their transient nature, although one would think this is most important.

Efficiency is typically measured at maximum power output, where it's usually at a maximum, and is related to efficiency during playback of real signal material. For example, if Amplifier A has higher efficiency at maximum power than Amplifier B, then Amplifier A will likely be more efficient than Amplifier B when playing real source material.

However, there are cases where

higher efficiency at maximum power doesn't translate to higher efficiency while playing actual source material. In fact, the choice of source material can make significant differences in realized efficiency. Efficiency at low signal levels is largely controlled by quiescent ("quiet") power loss, which also plays an important role in "real world" efficiency.

With all this said, it's regrettable that we must generalize with maximum power efficiencies, but it helps contain the scope of the discussion to a manageable level.

Let's suppose we are comparing two amplifiers – Amplifier A is 50 percent efficient and Amplifier B is 100 percent efficient. This means that Amplifier A wastes as much power (as heat) as it delivers to the load, and Amplifier B wastes no power at all, generating no heat and delivering everything it consumes directly to the load.

Let's also suppose these are 100-watt amplifiers, and that we want the power supply to be able to run 100 watts through the amplifier to the load indefinitely. There are tricks often played with the power supply's actual capability in the audio realm, but we will assume here that none of those tricks are being used.

Amplifier A requires a 100-watt heat sink, and Amplifier B requires no heat sink. (A 100-watt heat sink is quite large, and has significant weight.) In addition, Amplifier A requires a 200-watt power supply as opposed to Amplifier B's 100-watt supply. That's twice the power to drive the same load at the same power, plus a large heat sink, all so we can throw heat off into the air!

As we approach 100 percent efficiency, heat sinks and power supplies shrink, enabling more compact and lighter products.

IN SESSION

Class-B topology is the simplest to explain. Refer again to **Figure 1** – QT and QB are used to direct current from the voltage rails (+V and -V) into, or out of, the load. A drive circuit controls the transistors.

Assume the desired output is a voltage, and the current through the transistor, multiplied by the voltage across it, is power, dissipated as heat. This power is wasted energy because it doesn't contribute to driving the load. The heat generated is an undesirable side effect of this circuit topology and must be dissipated into the air (by a heat sink). In most cases, the desired output voltage is only a fraction of the voltage rails.

The remaining voltage (from the rail to the output) is across the transistor that is currently conducting. The top transistor conducts during positive voltage output, and the bottom transistor conducts during negative voltage output. The current, in the case of a resistive load, is a multiple of the output voltage, resulting in large power losses. It all boils down to another shortcoming of this class.

Class-B amplifiers are inherently problematic when driving the load close to zero output. When the output is close to zero, there is uncertainty concerning which transistor drives the load because there is no way to perfectly transfer control from one output device to the other. This is operation on the fringes, and creates "zero crossing distortion".

It's why purely Class-B is not used for audio power amplifiers. In addition, the efficiency is relatively low (theoretically 78 percent) at maximum power out.

With *Class-A*, there is significant current flowing through both transistors because the bias current is at least as great as the maximum output current. In "pure Class-A" both transistors are driving current all the time, regardless of the output.

Class-A controls the load better than Class-B, and generally, Class-A is considered to offer the highest performance and best sound. There are various "levels" of Class-A operation, depending on the amount of current through the transistors at idle (zero output).

A pure Class-A amplifier is actually not the least efficient (50 percent at maximum power) that this topology has to offer. In "heavy-bias Class-A," the transistors are driving each other as well as the load, and the amount of "extra current" dedicated to this lowers efficiency below 50 percent. Why would anyone design an amplifier with "heavy bias"? Simply, better control over the load, especially when the load is reactive.

Class-A topologies that are biased with less current than is required to be classified as pure Class-A are called *Class-AB*. They are a cross between Class-B operation and Class-A operation because the transistors both conduct through the zero crossing point. Class-AB can be "biased heavily" to produce a near-Class-A effect, while the efficiency of Class-AB into reactive loads depends on biasing. Heavier bias currents allow better control, but as mentioned, present an efficiency penalty.

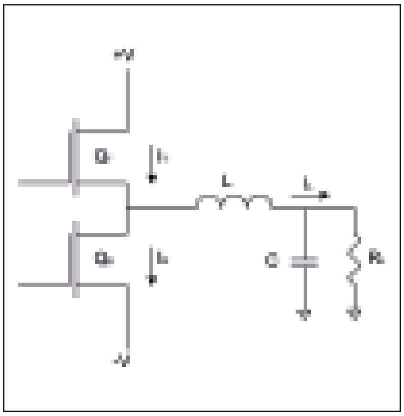

Figure 2: A Class-D fixed frequency PWM half-bridge output stage.

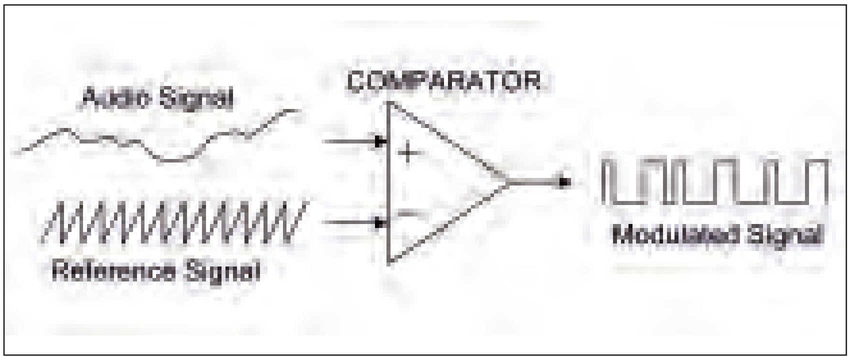

Figure 3: Slowly moving analog signal compared to a faster moving "carrier" signal.

The majority of amplifiers produced today are Class-AB, with efficiency at maximum power varies between 50 percent and 78 percent, depending on the strength of the bias current. This class is very mature in audio, and therefore inherently low cost.

SWITCH OR DIGITAL?

Class-D designs are sometimes called "digital amplifiers." Note that the "D" in Class-D does not stand for "digital," and this is a common misconception. (D was just the next letter in the alphabet when class designations were handed out!)

Class-D amplifiers operate the output devices as switches. The state of a switch can be thought of as a binary value (1 or 0), as in digital circuits like computers. Some Class-D amplifiers take in digital inputs. In that case, they're often referred to as "true digital" amplifiers, adding to the confusion.

Other classes of amplifiers, by the way, do not share this possibility. The bottom line here is that Class-D amplifiers are switching amplifiers, and they have much in common with switching power supplies.

Switches theoretically dissipate no heat. When a switch is closed (the "on" state), there is no voltage across it, and when it is open (the "off" state), there is no current through it. Because power is the product of voltage and current, there is no dissipation in the switch, thus a theoretical 100 percent efficiency for Class-D.

However, transistors are far from ideal switches, especially at the typical 200 kHz to 2 MHz switching frequency. The transistors used in almost every Class-D amplifier today are MOSFETs due to their fast transition speed and low "on" resistance. Despite these limitations, Class-D amplifiers are typically 90 percent efficient in practice.

It's important to note that a certain amount of power is lost during every switching transition. This is due to the imperfections of MOSFETs used as switches, such as finite switching times, dissipation of fly back current from the output filter through on state resistance, reverse recovery charge, and other practical circuit limitations. In essence, the higher the switching frequency, the lower the efficiency.

Class-D amplifiers have historically found homes in subwoofer applications due to conveniently low required bandwidth (allowing lower switching frequencies) and lower expectations regarding noise and distortion performance as opposed to raw power output. Using Class-D in this application shrinks the power supply considerably and doesn't stress the amplifier's performance potential. Using Class-D in a full range application is more demanding.

There are many variations of Class-D, so we will discuss the simplest form of Class-D, the fixed frequency PWM half-bridge, as an example. The output devices are switched with varying duty cycle, and the duty cycle of the transitions determines the average output voltage. An LC filter (called the output filter) is used to remove the carrier, and you are left with the desired output voltage, plus switching ripple. **Figure 2** depicts this type of amplifier output stage.

Also in **Figure 2**, the transistors (MOSFETs) are turned on and off in a manner that prevents them from both being on at the same time (a condition known as “shoot thru”). The LC filter smoothes these transitions to the load.

OTHER ADVANTAGES

Class-D amplifiers handle reactive loads differently than linear amplifiers. Operating the transistors as switches minimizes the voltage across them when reactive currents are present, keeping dissipation low.

Switching amplifiers have a few other advantages over linear amplifiers such as the elimination of zero crossing distortion and excellent transient response. We haven’t discussed the transient response of the other amplifier classes, but will contrast Class-D with respect to the other classes regarding this important sonic quality. In a Class-D amplifier, the power supply is not loaded heavily until the output is close to its limits; therefore, it has more power “on reserve” to tackle the sudden demand for current to the load.

Sonically, Class-D performs very well when compared to Class-AB or Class-A amplifiers of the same price range, while providing much higher efficiency. What, then, is the downside to Class-D? Why aren’t all amplifiers Class-D? Theoretically, it is the perfect amplifier, dissipating no heat, and performing its amplification function as desired.

However, the audio performance of a Class-D amplifier has much to do with its practical implementation. The physical limitations of MOSFETs and the circuits that drive them make it difficult, but not impossible, to match the performance of Class-AB or Class-A.

The specifications where these limitations come into play are noise and distortion. It’s a difficult engineering task to produce a low-distortion low-noise high-power Class-D audio amplifier at reasonable cost. In fact, even in cost-no-object designs, achieving high performance is certainly challenging.

Timing accuracy is important to Class-D amps because the relative timing of the transitions directly determines the average output voltage. This means that any error in timing produces an associated error at the output.

If the errors appear as “jitter,” it creates noise at the output. If the timing error is dependent on the input signal, it appears as non-linearity at the output. Achieving perfect timing is not possible, but the closer you get to perfect timing, the closer you get to perfect output.

In theory, a Class-D output stage drives the output filter with a perfect square wave. In practice, the average output current affects not only the timing of the pre-filtered output, but also the “squareness.”

For instance, rise and fall times vary with average current, and so do some of the undesirable effects of component limitations, such as overshoot or undershoot. These problems are usually corrected with feedback such that the average output voltage is accurate despite the instantaneous inaccuracies associated with generating a high power square wave.

One important factor in Class-D performance is the "modulation technique" that is used to control the output stage. Modulation is the act of converting the analog (or digital) input to a pulse train. Modulation technology is a growing field, and complexity varies greatly from technique to technique.

Class-D amplifiers drive their output devices with a modulated signal, essentially a conversion of the analog audio (variable voltage) to digital (again, "1" or "0"). As mentioned, there are a variety of ways to perform modulation, but for sake of simplicity, the "comparator method" will be explained.

Referring to **Figure 3** (page 52), the slowly moving analog signal is compared to a faster moving "carrier" signal, typically between 200 kHz and 1 MHz. If the audio signal is a greater voltage than the carrier signal, the comparator outputs a "1." If the audio signal is a lower voltage than the carrier signal, the comparator outputs a "0."

The output of the comparator can then be averaged, or filtered, to "demodulate" this digital signal back to analog. This filtering is performed by the amplifier's output filter (after the output stage). The performance of the modulation process, as measured in terms of noise and distortion, depends greatly on actual circuit implementation.

Other concerns with Class-D have caused difficulty in certain applications. One such concern is Electromagnetic Interference (EMI) caused by the switching action of the amplifier. This makes it difficult to place a Class-D amplifier near RF devices, such as wireless microphone receivers unless extra care is taken to contain the radiated field of the output stage and output filter. EMI is also an issue when passing regulatory guidelines.

OPEN LOOP

Feedback is used in most modern amplifiers to correct for power supply variations and output device non-linearity. An amplifier that incorporates feedback is called "closed loop". An amplifier that doesn't use feedback is called "open loop."

The use of feedback is a broad topic with many intricacies. However, it's notable to mention that almost every amplifier class relies heavily on feedback to make them perform well. Some amplifiers operate open loop with good performance, but this is a difficult result to achieve.

In the case of Class-D, feedback is especially difficult to implement when the output filter (and its associated phase response) is part of the "loop," and in many Class-D amplifiers, the feedback is taken from before the output filter.

New topologies are being invented all the time. Combinations of Class-D with linear supplies or switching supplies, Class-AB amplifiers driven from Class-D amplifiers (used as supplies), multiple moving rails, Class-AB driving the power supply rails to control the load from the power supply center tap, and many other novel techniques exist. Covering them all, even briefly is a big chore.

For now, it comes down to this: Class-A topologies are associated with the "best" sound but are the most wasteful in terms of heat. Class-AB is the most common in conventional amplifiers, and produces "good" sound quality with "better than-Class-A" efficiency.

And Class-D – the newest topology – is almost a sure bet to become the most common in the future as the technology progresses. ■

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