Amplifier Classes
From A to H

What is an amplifier and how can you choose the right class type for your application? This article defines amplifiers and describes different classes to help you find the best combination of performance, efficiency, and cost for your design.

By Robert Lacoste (France)

Welcome back to the Darker Side. In my last article, I explained how to wire a simple bipolar transistor to build an amplifier (“Bipolar Transistor Biasing” Circuit Cellar 279). Even if I didn’t state it explicitly, these amplifiers were part of the so-called Class-A amplifiers.

If you are an audiophile, you know there are plenty of other amplifier classes. They are identified by a long list of letters including the common A, AB, B, and D, but also more exotic letters including G, H, and even T. Their advantages and pitfalls are commonly discussed amongst audio geeks. Try to Google “best audio amplifier class” and you will receive more than 39 million answers.

Of course, amplifiers are used for more than just audio applications. Innovative amplifier designs are also required for RF systems, where efficiency is a key concern. Engineers have worked to develop some additional amplifier classes, namely C, E, and F.

This may seem overwhelming, so I thought it was a good subject for this column. Have a seat, as I try to explain the key differences between all these amplifier families.

AMPLIFIER BASICS

Let’s start with some basics. For simplicity, I will assume that the amplifier’s inputs and outputs are AC coupled through properly sized capacitors, so DC offsets are not a concern. I will also assume that the power supply could be either unipolar (V) or bipolar (±V) and that the signal’s polarity is not a problem (if so, it could easily be inverted with another small transistor used as a preamplifier).

What is an amplifier? As shown in Figure 1, amplifiers are devices that must accept a given input signal $V_{\text{IN}}$—for example, ranging from $-V_{\text{P(IN)}}$ to $V_{\text{P(IN)}}$—and generate an amplified version $V_{\text{OUT}}$ of the input with some DC offset. So the amplifier’s output should be something like:

$$V_{\text{OUT}} = V_{\text{OFFSET}} + \text{gain} \times V_{\text{IN}}$$

You may argue that this is more than simplified. You would be correct, as an amplifier can’t be restricted to a voltage amplifier. There are concerns regarding input and output impedances, which would lead to the associated current and power gain. Anyway, this article is only about amplifier classes and talking about voltage gain will be enough for this discussion.

Theory is easy, but difficulties arise when you actually want to design a real-world amplifier. What are your particular choices for its final amplifying stage?

CLASS A

The first and simplest solution would be to use a single transistor in linear mode. This
is what I presented in my previous article. **Figure 2** shows a refreshed version.

Basically the transistor must be biased to have a collector voltage close to $V_{CC}/2$ when no signal is applied on the input. This enables the output signal to swing either above or below this quiescent voltage depending on the input voltage polarity.

If you take another look at the schematic, you will understand that a current must continuously flow through the transistor to achieve this biasing set point. This current must never go down to zero, as you want it to stay in the transistor’s linear zone.

You could replace the bipolar transistor with some fancy MOSFET or use a pair of transistors to adopt a variant, but the concept would stay the same. It is a Class-A amplifier as long as the transistor is always conducting.

This solution’s advantages are numerous: simplicity, no need for a bipolar power supply, and excellent linearity as long as the output voltage doesn’t come too close to the power rails. This solution is considered as the perfect reference for audio applications. But there is a serious downside.

Because a continuous current flows through its collector, even without an input signal’s presence, this implies poor efficiency. In fact, a basic Class-A amplifier’s efficiency is barely more than 30%. It could be increased to 40% if the collector load is replaced with a coupling transformer, but that’s still low. And this poor efficiency means heat.

As an example, assume you want to build a stereo 2 × 100-W Class-A amplifier. If you do the math, you’ll find that around 300 W would be continuously dissipated in the amplifier’s transistors. This would be enough to keep your room at a warm temperature.

**CLASS B**

How can you improve an amplifier’s efficiency? You want to avoid a continuous current flowing in the output transistors as much as possible.

Class-B amplifiers use a pair of complementary transistors in a push-pull configuration (see **Figure 3**). The transistors are biased in such a way that one of the transistors conducts when the input signal is positive and the other conducts when it is negative. Both transistors never conduct at the same time, so there are very few losses. The current always goes to the load. Plenty of variants exist (e.g., with coupling transformers), but a Class-B classification always means each transistor conducts 50% of the time.

A Class-B amplifier has more improved efficiency compared to a Class-A amplifier. This is great, but there is a downside, right?

The answer is unfortunately yes. The downside is called crossover distortion. The problem is that it is impossible to ensure that one transistor will start to conduct exactly when its counterpart stops, whatever its external conditions (e.g., transistors’ temperature, dispersion in transistor’s behaviors, input signal frequency, etc.). Therefore, a Class-B amplifier suffers from a small distortion at each signal’s zero crossing. It could be small when the amplifier is properly designed, but it will never be as good as a Class-A amplifier.

**CLASS AB**

As its name indicates, Class-AB amplifiers are midway between Class A and Class B.
Have a look at the Class-B schematic shown in Figure 3. If you slightly change the transistor’s biasing, it will enable a small current to continuously flow through the transistors when no input is present. This current is not as high as what’s needed for a Class-A amplifier. However, this current would ensure that there will be a small overall current, around zero crossing.

Only one transistor conducts when the input signal has a high enough voltage (positive or negative), but both will conduct around 0 V. Therefore, a Class-AB amplifier’s efficiency is better than a Class-A amplifier but worse than a Class-B amplifier. Moreover, a Class-AB amplifier’s linearity is better than a Class-B amplifier but not as good as a Class-A amplifier.

These characteristics make Class-AB amplifiers a good choice for most low-cost designs. Nearly all low-to-medium-range audio amplifiers used to be Class AB, at least until the proliferation of Class-D amplifiers, as I will explain.

CLASS C

Rather than continuing to talk about audio amplifier with Class D, I found it more fun to follow the alphabet. Now I’ll discuss Class-C amplifiers (see Figure 4).

There isn’t any Class-C audio amplifier. This class is devoted to RF amplifiers. Why? This is because a Class-C amplifier is highly nonlinear. How can it be of any use?

An RF signal is composed of a high-frequency carrier with some modulation. The resulting signal is often quite narrow in terms of frequency range. Moreover, a large class of RF modulations doesn’t modify the carrier signal’s amplitude.

For example, with a frequency or a phase modulation, the carrier peak-to-peak voltage is always stable. In such a case, it is possible to use a nonlinear amplifier and a simple band-pass filter to recover the signal!

Class-C amplifiers go one step further than Class-B amplifiers. With Class A, the transistor always conducts; with Class B, it conducts half of the time; and with Class C, the transistor conducts only a small percentage of the time (i.e., when the input voltage is close to its peak).

The voltage on the transistor’s collector is then sharply dropped one time per period. As illustrated, this signal then has high amplitude and the same frequency as the input signal. However, it is far from a clean sine signal. A proper band-pass or low-pass filter can recover the desired amplified signal.

As you may imagine, designing a properly working Class-C amplifier is a bit more complex than designing with Class A, B, or AB, but the advantage is efficiency.

A Class-C amplifier can have good efficiency as there are no lossy resistors anywhere. It goes up to 60% or even 70%, which is good for high-frequency designs. Moreover, only one transistor is required, which is a key cost reduction when using expensive RF transistors. So there is a high probability that your garage door remote control is equipped with a Class-C RF amplifier.
COLUMNS

CLASS D

Class D is currently the best solution for any low-cost, high-power, low-frequency amplifier—particularly for audio applications. Figure 5 shows its simple concept.

First, a PWM encoder is used to convert the input signal from analog to a one-bit digital format. This could be easily accomplished with a sawtooth generator and a voltage comparator as shown on Figure 5. This section’s output is a digital signal with a duty cycle proportional to the input’s voltage. If the input signal comes from a digital source (e.g., a CD player, a digital radio, a computer audio board, etc.) then there is no need to use an analog signal anywhere. In that case, the PWM signal can be directly generated in the digital domain, avoiding any quality loss.

This PWM signal is then amplified. The beauty of the Class-D architecture is that amplifying a digital signal can be done with high efficiency since the transistors are used as full on/off switches. Power dissipation only occurs during the transitions from one logic state to the other and can be minimized if the transistors are fast enough.

After this amplifying stage, there is a high power signal, but it’s still in a PWM format. How can you reconstruct the high-power analog output you were looking for? You can simply use a low-pass filter! Moreover, if you are designing an audio amplifier, then the loudspeaker itself could be part of the filter (it is an inductor), which reduces the bill of material.

As you may have guessed, Class-D amplifiers aren’t free from difficulties. First, as for any sampling architecture, the PWM frequency must be significantly higher than the input signal’s highest frequency to avoid aliasing. Nyquist sampling theorem says twice higher, but this would require a perfect brick-wall output filter. To keep the filter cost under control, the PWM frequency must be a lot higher, usually 10 times higher than the input signal frequency.

For a 10-Hz-to-20-kHz audio signal, this translates into PWM frequencies in the hundreds of kilohertz, (e.g., 200 kHz). Such a frequency corresponds to a 5-µs period (i.e., 1/200,000).

Now assume you want a 16-bit amplitude accuracy. This implies the position time of each PWM edge must be precise with a 1/65,536 relative accuracy. Therefore, the switching stage’s design must not introduce any timing errors.

This PWM signal is then amplified. The beauty of the Class-D architecture is that amplifying a digital signal can be done with high efficiency since the transistors are used as full on/off switches. Power dissipation only occurs during the transitions from one logic state to the other and can be minimized if the transistors are fast enough.

After this amplifying stage, there is a high power signal, but it’s still in a PWM format. How can you reconstruct the high-power analog output you were looking for? You can simply use a low-pass filter! Moreover, if you are designing an audio amplifier, then the loudspeaker itself could be part of the filter (it is an inductor), which reduces the bill of material.

As you may have guessed, Class-D amplifiers aren’t free from difficulties. First, as for any sampling architecture, the PWM frequency must be significantly higher than the input signal’s highest frequency to avoid aliasing. Nyquist sampling theorem says twice higher, but this would require a perfect brick-wall output filter. To keep the filter cost under control, the PWM frequency must be a lot higher, usually 10 times higher than the input signal frequency.

For a 10-Hz-to-20-kHz audio signal, this translates into PWM frequencies in the hundreds of kilohertz, (e.g., 200 kHz). Such a frequency corresponds to a 5-µs period (i.e., 1/200,000).

Now assume you want a 16-bit amplitude accuracy. This implies the position time of each PWM edge must be precise with a 1/65,536 relative accuracy. Therefore, the switching stage’s design must not introduce any timing errors.

This PWM signal is then amplified. The beauty of the Class-D architecture is that amplifying a digital signal can be done with high efficiency since the transistors are used as full on/off switches. Power dissipation only occurs during the transitions from one logic state to the other and can be minimized if the transistors are fast enough.

After this amplifying stage, there is a high power signal, but it’s still in a PWM format. How can you reconstruct the high-power analog output you were looking for? You can simply use a low-pass filter! Moreover, if you are designing an audio amplifier, then the loudspeaker itself could be part of the filter (it is an inductor), which reduces the bill of material.

As you may have guessed, Class-D amplifiers aren’t free from difficulties. First, as for any sampling architecture, the PWM frequency must be significantly higher than the input signal’s highest frequency to avoid aliasing. Nyquist sampling theorem says twice higher, but this would require a perfect brick-wall output filter. To keep the filter cost under control, the PWM frequency must be a lot higher, usually 10 times higher than the input signal frequency.

For a 10-Hz-to-20-kHz audio signal, this translates into PWM frequencies in the hundreds of kilohertz, (e.g., 200 kHz). Such a frequency corresponds to a 5-µs period (i.e., 1/200,000).

Now assume you want a 16-bit amplitude accuracy. This implies the position time of each PWM edge must be precise with a 1/65,536 relative accuracy. Therefore, the switching stage’s design must not introduce any timing errors.

This PWM signal is then amplified. The beauty of the Class-D architecture is that amplifying a digital signal can be done with high efficiency since the transistors are used as full on/off switches. Power dissipation only occurs during the transitions from one logic state to the other and can be minimized if the transistors are fast enough.

After this amplifying stage, there is a high power signal, but it’s still in a PWM format. How can you reconstruct the high-power analog output you were looking for? You can simply use a low-pass filter! Moreover, if you are designing an audio amplifier, then the loudspeaker itself could be part of the filter (it is an inductor), which reduces the bill of material.

As you may have guessed, Class-D amplifiers aren’t free from difficulties. First, as for any sampling architecture, the PWM frequency must be significantly higher than the input signal’s highest frequency to avoid aliasing. Nyquist sampling theorem says twice higher, but this would require a perfect brick-wall output filter. To keep the filter cost under control, the PWM frequency must be a lot higher, usually 10 times higher than the input signal frequency.

For a 10-Hz-to-20-kHz audio signal, this translates into PWM frequencies in the hundreds of kilohertz, (e.g., 200 kHz). Such a frequency corresponds to a 5-µs period (i.e., 1/200,000).

Now assume you want a 16-bit amplitude accuracy. This implies the position time of each PWM edge must be precise with a 1/65,536 relative accuracy. Therefore, the switching stage’s design must not introduce any timing errors.

This PWM signal is then amplified. The beauty of the Class-D architecture is that amplifying a digital signal can be done with high efficiency since the transistors are used as full on/off switches. Power dissipation only occurs during the transitions from one logic state to the other and can be minimized if the transistors are fast enough.

After this amplifying stage, there is a high power signal, but it’s still in a PWM format. How can you reconstruct the high-power analog output you were looking for? You can simply use a low-pass filter! Moreover, if you are designing an audio amplifier, then the loudspeaker itself could be part of the filter (it is an inductor), which reduces the bill of material.

As you may have guessed, Class-D amplifiers aren’t free from difficulties. First, as for any sampling architecture, the PWM frequency must be significantly higher than the input signal’s highest frequency to avoid aliasing. Nyquist sampling theorem says twice higher, but this would require a perfect brick-wall output filter. To keep the filter cost under control, the PWM frequency must be a lot higher, usually 10 times higher than the input signal frequency.

For a 10-Hz-to-20-kHz audio signal, this translates into PWM frequencies in the hundreds of kilohertz, (e.g., 200 kHz). Such a frequency corresponds to a 5-µs period (i.e., 1/200,000).

Now assume you want a 16-bit amplitude accuracy. This implies the position time of each PWM edge must be precise with a 1/65,536 relative accuracy. Therefore, the switching stage’s design must not introduce any timing errors.

This PWM signal is then amplified. The beauty of the Class-D architecture is that amplifying a digital signal can be done with high efficiency since the transistors are used as full on/off switches. Power dissipation only occurs during the transitions from one logic state to the other and can be minimized if the transistors are fast enough.

After this amplifying stage, there is a high power signal, but it’s still in a PWM format. How can you reconstruct the high-power analog output you were looking for? You can simply use a low-pass filter! Moreover, if you are designing an audio amplifier, then the loudspeaker itself could be part of the filter (it is an inductor), which reduces the bill of material.

As you may have guessed, Class-D amplifiers aren’t free from difficulties. First, as for any sampling architecture, the PWM frequency must be significantly higher than the input signal’s highest frequency to avoid aliasing. Nyquist sampling theorem says twice higher, but this would require a perfect brick-wall output filter. To keep the filter cost under control, the PWM frequency must be a lot higher, usually 10 times higher than the input signal frequency.

For a 10-Hz-to-20-kHz audio signal, this translates into PWM frequencies in the hundreds of kilohertz, (e.g., 200 kHz). Such a frequency corresponds to a 5-µs period (i.e., 1/200,000).

Now assume you want a 16-bit amplitude accuracy. This implies the position time of each PWM edge must be precise with a 1/65,536 relative accuracy. Therefore, the switching stage’s design must not introduce any timing errors.
random variations higher than 76 ps (i.e., 5 µs/65,536) This is quite low, isn’t it? This explains why building a good Class-D amplifier is a little more complex than drafting its block diagram.

The second concern with Class-D amplifiers is related to electromagnetic compatibility (EMC). Switching power signals at high speeds is the best way to generate strong electromagnetic fields (EMFs). This is especially true when the output is connected to long unshielded wires such as the ones between an audio amplifier and the loudspeakers.

Class-D amplifier designers must be cautious regarding the output low-pass filter’s performance. And they must check the compliance to EMC standards as soon as possible in the design cycle.

Class-D amplifiers are now used everywhere, so solutions do exist. However, their implementation is not as straightforward as it looks. As always, the best solution is to use a pre-integrated solution (e.g., a Class-D IC from one of the many suppliers such as Texas Instruments, Maxim Integrated, and Linear Technology). Then you must look at its reference design and either carefully reproduce it or understand exactly what you are doing before making a modification.

**CLASS E AND CLASS F**

Remember that Class C is devoted to RF amplifiers, using a transistor conducting only during a part of the signal period and a filter. Class E is an improvement to this scheme, enabling even greater efficiencies up to 80% to 90%. How?

Remember that with a Class-C amplifier, the losses only occur in the output transistor. This is because the other parts are capacitors and inductors, which theoretically do not dissipate any power.

Because power is voltage multiplied by current, the power dissipated in the transistor would be null if either the voltage or the current was null. This is what Class-E amplifiers try to do: ensure that the output transistor never has a simultaneously high voltage across its terminals and a high current going through it.

This may seem impossible, but remember that an RF amplifier is only used for a narrow frequency band around the RF carrier frequency. This enables you to use two tricks to design a Class-E amplifier.

First, as for Class-D amplifiers, the transistor must be used as much as possible as an on/off switching device and not in its linear region (as with Class-C amplifiers). This requires a proper biasing and a very fast transistor.

Second, a specific tuning of the output impedance-matching filter could ensure that the voltage across the transistor is minimal when the transistor switches from one state to the other. Basically the output filter must have the proper complex impedance to introduce the required phase shift between current and voltage.

Going into further detail would require a full article. If you are interested I recommend you read N. O. Sokal’s “Class-E RF Power Amplifiers” (QEX magazine, 2001).

I have a final word about Class F. This is an improvement to Class E, optimizing the impedance-matching circuits one step further to reduce the effects of harmonics and increase the efficiency a little more. Similar to Class E, Class F is only suitable for narrow-frequency bands, so it is mainly used for RF applications.
**CLASS G AND CLASS H**

Class G and Class H are quests for improved efficiency over the classic Class-AB amplifier. Both work on the power supply section. The idea is simple. For high-output power, a high-voltage power supply is needed. For low-power, this high voltage implies higher losses in the output stage.

What about reducing the supply voltage when the required output power is low enough? This scheme is clever, especially for audio applications. Most of the time, music requires only a couple of watts even if far more power is needed during the *fortissimo*. I agree this may not be the case for some teenagers’ music, but this is the concept.

Class G achieves this improvement by using more than one stable power rail, usually two. Figure 6 shows you the concept.

As long as the input signal is low enough, the amplifier is a classic Class-AB amplifier and uses a low-voltage power supply. When the signal amplitudes are higher, another couple of transistors go into action, supplied by a higher voltage power.

The detailed design is more complex, in particular to reduce the distortion during the transitions between each mode. The two power supplies must be generated through high-efficiency DC/DC switching converters, as you don’t want to reduce the transistors’ losses by increasing losses in the power supply section.

Class H goes one step further in the same spirit. Rather than using two or more power supply rails, it uses a continuously variable switching DC power supply. The power supply’s voltage is dynamically adjusted to whatever is required for optimal performances depending on the required signal amplitude. This enables a better efficiency; however, the power supply’s performances are deeply correlated to the full amplifier’s performances.

**WRAPPING UP**

The list of amplifier classes seems endless. While preparing this article, I even found some references to a so-called “Class-T” audio amplifier. It appears to be a custom architecture developed by Tripath Company—which is now owned by Cirrus Logic—and is somewhat close to Class D. Very little information is available, but at least you known what the “T” stands for.

Amplifier design is an infinite quest for the best balance between performances, efficiency, and cost. An engineer’s goal is to find the best mix.

As an example, Figure 7 shows a block diagram of Maxim’s MAX98090 audio amplifier chip. In the same piece of silicon, there is a stereo audio amplifier that could be configured as a 3-W Class-D speaker amplifier, as a Class-AB amplifier for line output, and an independent ultra-low-power stereo Class-H headphone amplifier. This chip also integrates some preamplifiers as well as an ADC, a DAC, and a DSP, but that’s another story. It’s not too bad for less than $3.

I’m sure you will find your way through a manufacturer’s website for even more impressive silicon, but I would like to wrap up this article by clarifying an important point. Designing an amplifier isn’t just about selecting which class is the best for your application. This may be enough for simple projects, but as a good engineer, you need to check if a more clever approach may be appropriate.

Do you want some examples? In the RF field, an efficient solution was invented in 1936 at Bell Labs by William H. Doherty. His idea was to associate two amplifiers, one in charge of the RF carrier amplification and one providing some extra power when the input signal is strong. Each section could then be independently optimized for best

---

**RESOURCES**


H. Theiler, H. Lenhard, and H. Gether, “Class-G/H Amplifiers: How Well Do They Deliver on Their Promise of High Audio Quality and Low Power Consumption?” ams AG.


—–, “Class-T amplifier.”


**SOURCES**

D-Premier amplifier and ADH technology
Devialet | http://en.devialet.com

MAX98090 Ultra-low-power stereo audio codec
Maxim Integrated, Inc. | www.maximintegrated.com
ABOUT THE AUTHOR

Robert Lacoste lives in France, near Paris. He has 24 years of experience in embedded systems, analog designs, and wireless telecommunications. A prize winner in more than 15 international design contests, in 2003 he started his consulting company, ALCIOM, to share his passion for innovative mixed-signal designs. His book (Robert Lacoste’s The Darker Side) was published by Elsevier/Newnes in 2009. You can reach him at rlacoste@alciom.com if you don’t forget to put “darker side” in the subject line to bypass spam filters.

performances (e.g., with a Class-AB amplifier for the carrier and a Class-C amplifier for the peaks). This so-called “Doherty amplifier” is still currently in use.

Here are a couple more audio field examples. Yamaha’s “Yamaha Power Amplifier White Paper” explains how the company merged a Class-AB and a Class-D amplifier for best cost/performance ratio. Basically, a Class-D amplifier provides a coarse output at high efficiency and is improved by a Class-AB final stage providing fine-tuning at low loss. To me, it seems close to a Class-H amplifier.

In the same spirit, but a more higher-end version, look at Devialet’s award-winning D-Premier amplifiers. Its patented ADH technology (which stands for analog digital hybrid) combines a Class-A amplifier used as a reference and eight Class-D digital amplifiers per channel, achieving a total harmonic distortion below 0.001% and an amazing 130 dB of signal-to-noise ratio, which is more than impressive with 240 W of output power!

In such a short article, my goal was not to help you to design a amplifier. I just wanted to give you some information about how to select an adequate topology for your application.

Amplifier classes have been a hot topic since the early days of electronics, so there are plenty of publications and books on this subject. The information available in the Resources could be your starting point.

All rights reserved. No part of this publication may be reproduced or transmitted in any form or by any means, electronic or mechanical, including photocopying, recording, or any information storage and retrieval system, without permission in writing from the publisher.

www.circuitcellar.com/subscriptions

Now offering student SUBSCRIPTIONS!

When textbooks just aren’t enough, supplement your study supplies with a subscription to Circuit Cellar. From programming to soldering, robotics to Internet and connectivity, Circuit Cellar delivers the critical analysis you require to thrive and excel in your electronics engineering courses.