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A Comprehensive Study of Class D Amplifier Technology

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Abstract

This paper aims to provide a comprehensive technical overview of Class D audio amplifier operation and design. The fundamentals are presented with a focus on conceptual clarity rather than mathematical rigor. The main components of this technology, including modulation scheme, topology, and output stage, are discussed in detail. Important characteristics such as efficiency, distortion, and EMI are also thoroughly covered. Additionally, feedback control and robustness are discussed, and an effort is made throughout the paper to demonstrate the interconnections between the various aspects of Class D design.

Table of Contents

1. Introduction

Although the theory behind Class D amplification has been known for decades [1, 2], the technology to make it practical has only emerged in recent years. Conventional Class AB linear amplifiers have traditionally dominated the market in nearly all applications, but Class D switching amps have begun to succeed them, especially in portable and high-power applications where efficiency is a top priority.

Class D amplifiers offer efficiency improvements by utilizing Pulse-Width Modulation (PWM) to operate the output transistors in switch-mode. With zero current during off-mode and zero voltage during on-mode, power dissipation in the output devices is minimized, thus providing a theoretical efficiency of 100%. Although practical losses prevent this ideal value from being realized, substantial efficiency improvements can be achieved over conventional linear operation. As a result, the reduced heat dissipation affords smaller heatsink and package sizes.

Due to increasing concern over energy consumption and a growing market for portable electronics, Class D technology is expected to continually proliferate into the future. Currently viable markets include high-power professional installations, automotive audio, powered speakers and subwoofers, multichannel home theatre receivers, flat-panel televisions, and portable media players. Eventually, it is envisioned that the majority of amplifier applications will utilize Class D switching technology.

2. Overview & Fundamentals of Operation

Before exploring the details of Class D technology, the basics will be discussed so that the more in-depth topics can be more easily appreciated. Alternative amplifier classes and efficiency comparisons, as well as basic Class D operating principles, will be discussed here.

Class D amplifiers offer advantages over conventional linear amplifiers due to the improved efficiency of switch-mode operation. Higher efficiency can result in less power consumption, smaller size, and lighter weight for any given level of performance. Total system cost can often be reduced and fidelity is becoming increasingly competitive with linear designs. A brief comparison with alternative classes of operation will serve to contrast operating principles and help substantiate the efficiency advantage of Class D.

Although the market share for Class D amps is increasing, the majority of audio amplifiers today are of the Class AB variety. This linear topology combines the fidelity advantage of Class A with the efficiency advantage of Class B.

Class A amplifiers are typically biased at half the supply voltage so that the output transistors conduct for the entire duration of the audio signal. In fact, a very large current flows through the device even with no input signal. This large bias current causes excessive dissipation in the output devices and leads to very poor efficiency, with a 25% theoretical maximum and a more realistic maximum efficiency of 20% or less [3].

Class B amplifiers have the output devices configured for complementary push-pull operation, where one device conducts during positive portions of the signal and the other conducts during negative excursions. In this way, the bias current is eliminated and Class B amps claim a theoretical maximum efficiency of 78.5%, although in practice this maximum is closer to 50% [3].

Class B amps, however, are plagued with crossover distortion, which is an unpleasant artifact occurring near the zero-crossing points where the signal is below both device thresholds, causing neither transistor to conduct. Introducing a small bias current can ensure both transistors conduct during this region, effectively eliminating crossover distortion. This is the configuration known as Class AB, and it has traditionally proven to be the most acceptable compromise between fidelity and efficiency. However, current Class D amplifiers are challenging this supremacy and revolutionizing amplifier design.

Class D operates very differently from the linear classes of operation described above. Instead of having continuously varying levels of current through and voltage across the output devices (causing power dissipation), Class D amps utilize the output MOSFETs as switches driven by a Pulse Width Modulation

(PWM) square-wave. In switch-mode, current and voltage ideally never occur simultaneous in the devices, and this fundamental detail is what permits Class D to claim 100% theoretical efficiency. Current designs are capable of achieving maximum efficiencies greater than 90%, and device technology improvements will continue to drive that value closer to the theoretical limit [4]. More information is presented in the section on Efficiency.

Before continuing to describe the Class D amplifier, a few other classes of operation should also be mentioned. Class C, E, and F are designed primarily for high-frequency (RF) applications and will not be further discussed since the focus here is on audio applications. There are also two other classes, G and H, which share the goal of increased efficiency, but their fundamental operation is essentially that of a Class B amplifier with supply rails modulated to minimize voltage while accommodating the input signal [4, 5]. Although Class G and H offer efficiency improvements, Class D has become a much more popular technique.

A simplified block diagram of a basic Class D amplifier is shown in Fig. 2.1 below. Class D configurations are often referred to as "digital amplifiers," but although this description may seem appropriate given their switch-mode operation, PWM is fundamentally an analog modulation procedure. As shown in Fig. 2.1, PWM can be generated by inputting the audio signal and a triangle/sawtooth waveform into a comparator. The comparator produces a "high" when one of the inputs is higher than the other and then swaps to a "low" during the other condition. The details of PWM are described in detail in the following section on Modulation Schemes.

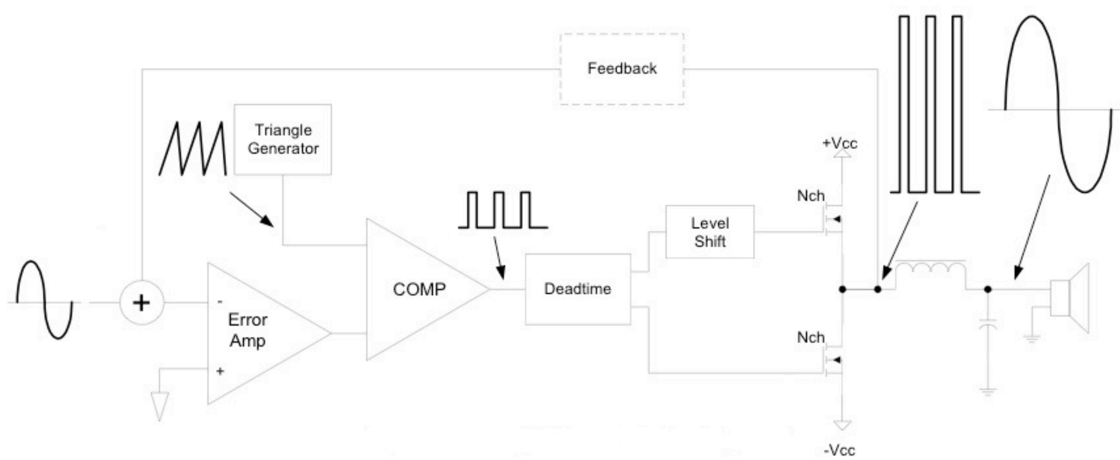

Fig. 2.1 - Simplified Block Diagram of a Basic Class D Amplifier [6].

The PWM signal from the comparator is not at a high enough voltage, so this signal is then used as the drive to complementary output MOSFETs that switch on and off the larger +Vcc and –Vcc supplies in accordance with the PWM signal. The high frequency modulation components of the PWM square wave, as shown in Fig. 2.2, are then removed with a demodulating LC lowpass filter to produce the final amplified audio output to the speaker. The output stage is discussed in the corresponding section below.

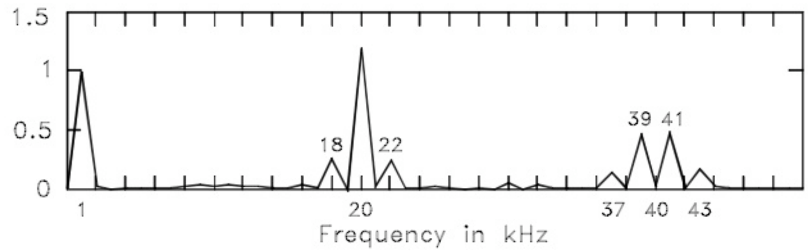

Fig. 2.2 - PWM Spectrum Before Low-Pass Filtering [7].

Errors and nonlinearities can occur throughout the circuit, but there is also an inherent distortion that occurs due to added dead-time, as addressed in the section on Nonlinearity & Distortion. To compensate for these problems, feedback is often utilized. Additionally, EMI radiation can be problematic and should be considered. These issues are also discussed in the sections below.

Throughout this paper, consideration of this operational overview should aid in the understanding of more detailed sections.

3. Modulation Schemes

The modulation scheme, truly at the heart of the Class D amplifier, refers to the technique by which the audio signal is encoded as a modulated square wave. There are various schemes, but the two most popular are Pulse Width Modulation (PWM) and Sigma-Delta (ΣΔ) Modulation. PWM is the most common technique [1], and will be the primary focus of this section.

a. Pulse Width Modulation (PWM)

Pulse Width Modulation is a general technique, and there are various aspects that can be altered for specific purposes. However, before discussing the variants of PWM, it will be of benefit to start the discussion with a conceptual generalization.

PWM encodes the amplitude values of the original audio signal in the pulse widths of a square wave output, as depicted in Fig. 3.1 below. When the input signal (v_S) is positive, the square wave output (v_O') has a duty cycle greater than 50%. In this condition, the square wave spends more time at the positive rail (V_{OP}), so when the output is low-pass filtered, the resultant average value at this point will be a positive value. Similarly, when the input is negative, v_O' has a duty cycle less than 50%, and a zero input results in 50% duty cycle.

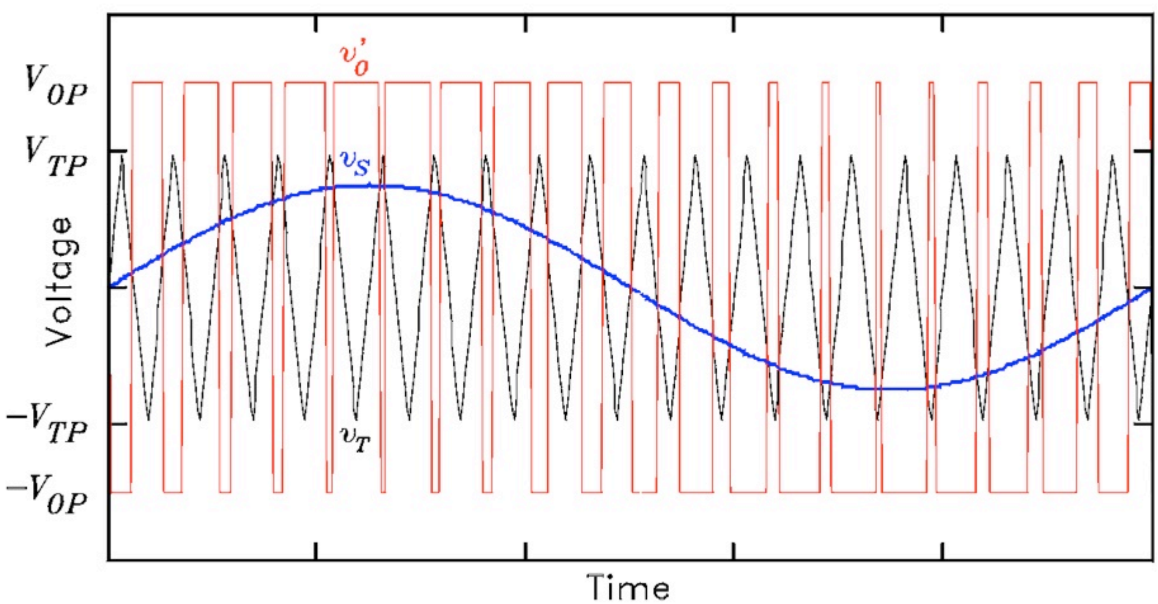

Fig. 3.1 - PWM Waveforms [7].

As mentioned in the Fundamentals of Operation section, PWM can be accomplished by sending the audio signal and a triangle/sawtooth carrier waveform into a comparator. When the audio input (v_S) exceeds the instantaneous value of the carrier (v_T), the comparator outputs its positive rail voltage. Likewise, the comparator outputs its negative rail voltage in the opposite condition, as illustrated in Fig. 3.1.

It should be noted that the carrier in Fig. 3.1 is depicted at a relatively low frequency for demonstrative purposes and in practice a much higher frequency is used. Since the carrier in essence is sampling the input, the Nyquist criterion must be satisfied; that is, the sampling frequency must be at least twice as large as the highest frequency signal component in order to avoid aliasing distortion. An additional constraint on the carrier frequency is that the slew rate of the carrier must be greater than the slew rate of the audio signal [4].

Another implicit requirement in this scheme is that the signal amplitude never exceeds that of the carrier. Full modulation occurs when the ratio of the input signal amplitude to the carrier amplitude, known as modulation depth Δ [8], equals 100% [1]. The amplifier will be unable to reproduce levels beyond full modulation, resulting in sharp clipping and harsh distortion. In practice, modulators can skip pulses for Δ less than 100%, and an appropriate margin should be maintained to avoid instabilities and distortion.

The above discussion has been generalized, but there are various permutations of PWM that will now be discussed. The classifications for PWM are based on sampling method, switching technique, and edge modulation [4]. Each of these factors is discussed below, followed by concluding comparisons.

i. Sampling: Natural (Analog) or Uniform (Digital)

The sampling method essentially involves this characteristic of the input signal, although the distinction has important implications and should be considered with the modulation scheme. The sampling refers to the amplitude representation in the modulation process (i.e. the comparator sampling the input signal through comparison with the carrier). Quite simply, analog signals result in 'natural sampling' and digital signals lead to 'uniform sampling' [4]. This

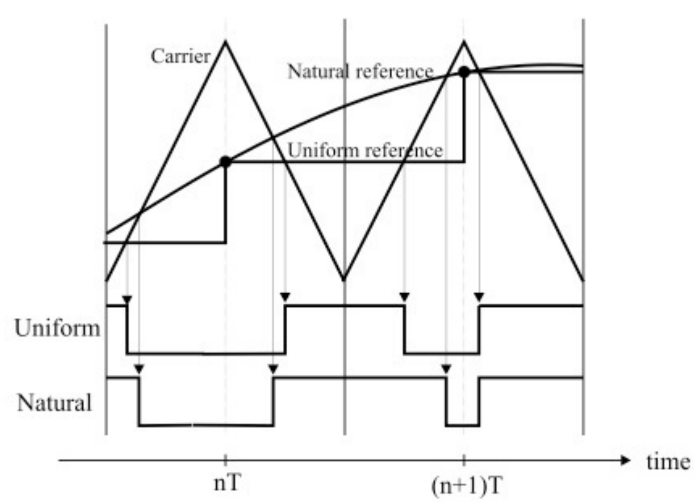

Fig. 3.2 - Sampling Methods [4].

is depicted in Fig. 3.2 and discussed more below.

Natural sampling is straightforward and follows the procedure discussed in section 3a above. This modulation theoretically enables an exact replica of the input signal to be reconstructed. With uniform sampling, however, exact replication is not possible and significant distortion can result.

Uniform sampling occurs for input signals that have already been sampled by other means (i.e. digital input signals). However, Pulse Code Modulated (PCM) representations typical of digital audio must first be converted to an appropriate form for PWM application [4], namely a "stair-step approximation."

The main advantage of this configuration is a potential simplification in system design, since a comparator and a separate digital-to-analog converter (DAC) are not necessary. Also, since many modern audio sources are digital, some may believe that retaining the signal in the digital domain provides the highest accuracy. While an "all-digital" amplifier certainly has high marketing appeal, the reality is that uniform sampling produces relatively poor performance in switching amplifiers. Harmonic distortion is typically much higher in a uniform sampling scheme than an equivalent natural sampling system [4].

Although the errors inherent in uniform PWM are severe, there are methods to compensate for these nonlinearities. In particular, digital signal processing (DSP) can be performed on the digital signal prior to modulation. Also, oversampling and noise-shaping can improve results. Perhaps the most successful technique, however, is to use a hybrid approach – to make uniform sampling appear more like natural sampling. This can be achieved through interpolation of the sampled uniform signal, and acceptable performance can be achieved with such hybrid schemes [4].

ii. Switching: Class AD or Class BD

The switching method is classified into two basic groups: class AD and class BD, which produce two-level and three-level modulation, respectively. The basic method described previously was Class AD, as this two-level modulation is conceptually simpler, with the output square-wave switching between positive and negative power supplies. The two-device half-bridge configuration, described previously and discussed in more detail in section 4b, requires the use of class AD modulation, but this switching technique can also be implemented with a full-bridge or H-bridge design, as shown in Fig. 3.3a below. This extension simply operates both legs of the H-bridge complementary, which in effect always connects a path between supplies but switches the direction of current in the load, forming a bridge-tied load (BTL) configuration.

However, the use of a four-device full-bridge configuration permits the use of Class BD [1], as shown in Fig. 3.3b. Depicted in this figure, and more rudimentary in Fig. 3.4b, is the driving method of achieving this three-level modulation. The two devices on the left, the side marked A, are switched exactly as described before for class AD. However, the devices on the right, marked B, although still complementary to each other, are no longer the simple complement of the A side. Instead, the B devices are driven by a modulated driver derived from an inverted version of the audio signal. The effect of this is that the B side switching node waveform is inverted in time within the sampling period with respect to the A side waveform. This behavior is perhaps more clearly shown in Fig 3.4.

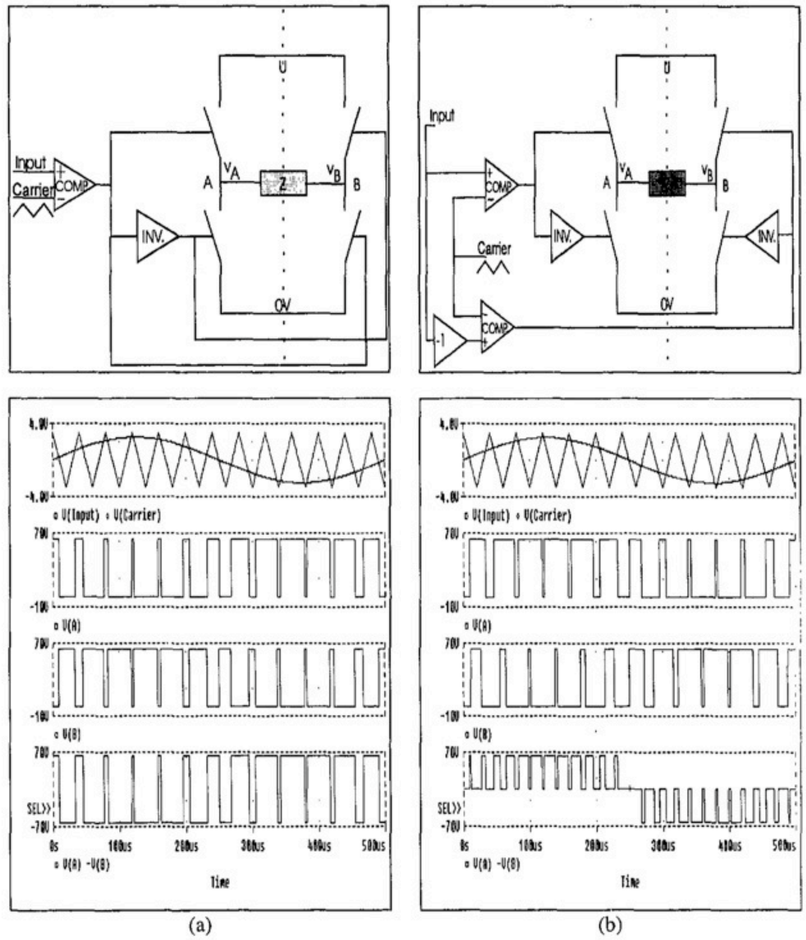

Fig. 3.3 - Class AD (a) and Class BD (b) Switching [9].

In class BD, since the two legs of the H-bridge do not exactly complement each other, the signal across the load, which is developed by the voltage at the A switching node minus the voltage at the B switching node, does not simply switch between the two supplies. Instead, there is a third state – zero differential voltage – established when both switching nodes are at the same potential, which is never possible in class AD.

One immediate effect of this is that class BD has a significant common-mode component, while class AD exhibits no common-mode voltage. Class BD also requires more precise switching control and can introduce added complexity compared to class AD [9].

Despite these disadvantages, performance of class BD is generally considered superior to class AD [4]. The main advantage of class BD involves the high frequency switching components. In particular, for low or idle input signals, the output contains very little high frequency energy [10]. This introduces advantages in efficiency and EMI, especially for signals like music that typically have very high crest factors [2].

Additionally, since the differential output step in BD is half that of AD, the maximum ripple current in the load decreases [9], and the demands on MOSFET settling and slew-rate is decreased as well [10]. Although class BD generally exhibits higher performance, the given application should ultimately dictate which scheme to use.

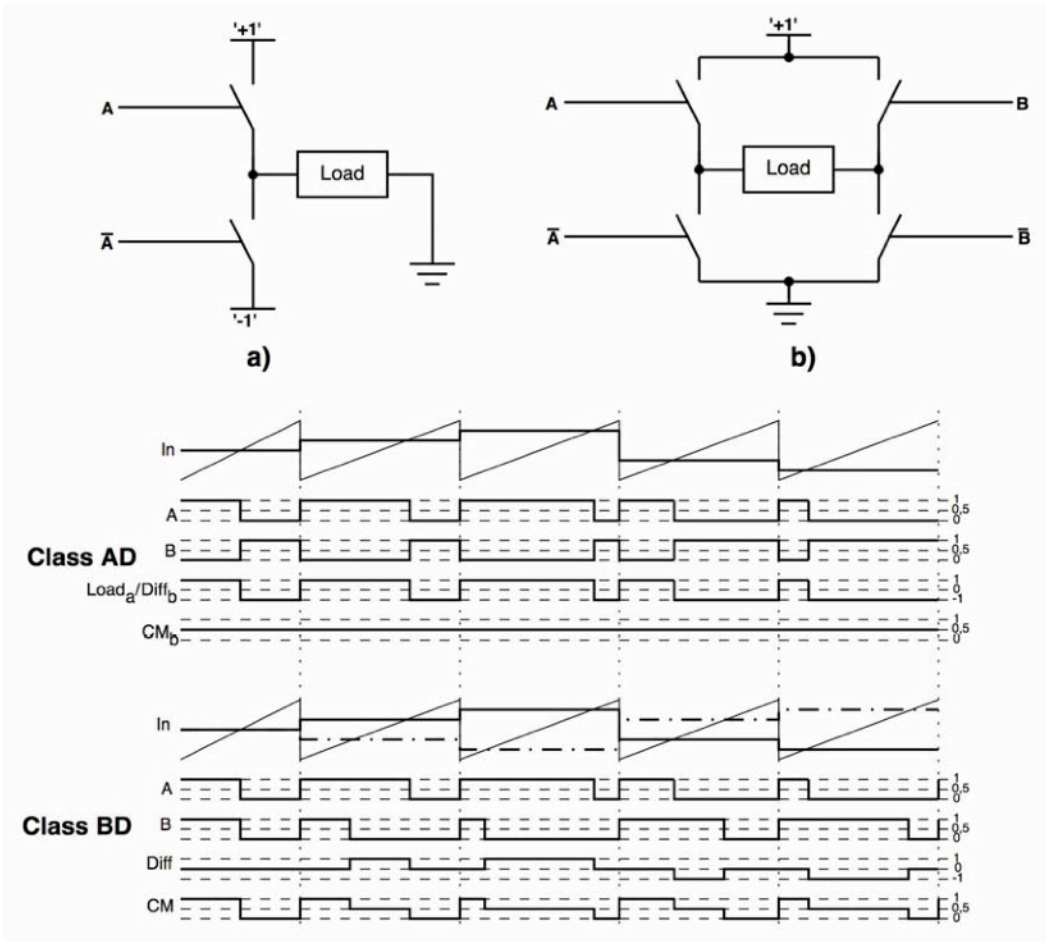

Fig. 3.4 – Half-Bridge Class AD (a) and Full-Bridge Class BD (b) Configurations with Corresponding Waveforms [10].

iii. Edge: Single-Sided or Double-Sided

Although the "edge" descriptor may seem a bit confusing, the single-sided and double-sided modulation to which it refers is very simple. Essentially, the distinction is drawn from whether the modulating carrier waveform is sawtooth or triangular, which produce single-sided and double-sided modulation, respectively [9]. Although the designation is rather simple, the implications are significant.

It should be noted that single-sided modulation can be performed with ascending or descending sawtooth carriers. The resultant trailing edge and leading edge modulation share the same properties [10], so the choice is inconsequential.

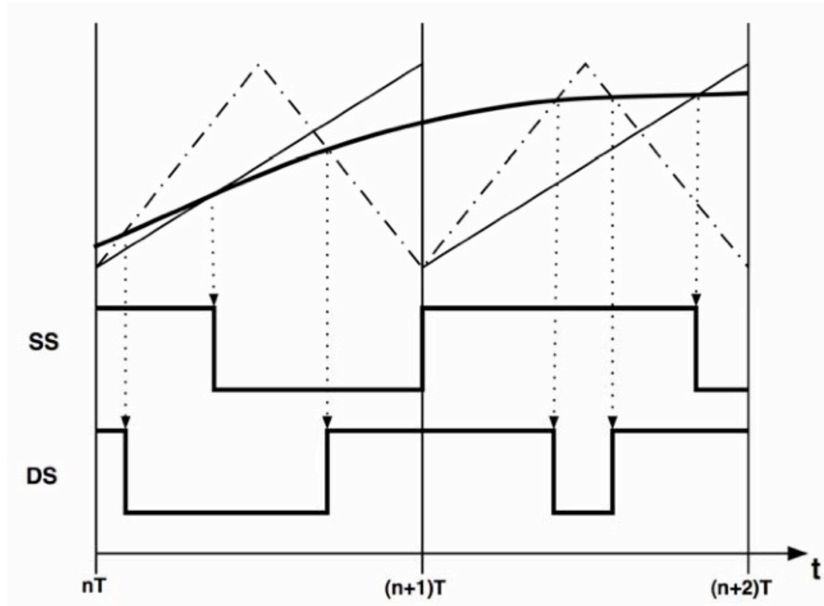

Fig. 3.5 - Single-Sided and Double-Sided Modulation [10].

As Fig. 3.5 demonstrates, double-sided modulation results in two samples per sampling period. This results in twice as much information being stored for any given sampling rate [4]. Essentially, the effective sampling frequency is doubled without increasing the actual switching rate.

This is a strong advantage because it permits a lower switching frequency to be used, which leads to smaller switching losses and thus higher efficiency. The need for a sharp roll-off output low-pass demodulation filter is also reduced, which can result in lower cost. Understandably, the advantages of double-sided modulation make it a very attractive method.

iv. Comparisons

After covering the three PWM classification groups, some important conclusions can be made. It was already discussed how each group generally has a preferable option, namely natural over uniform sampling, class BD over AD switching, and double-sided over single-sided edge modulation.

Also, a naming convention has been introduced to concisely identify these characteristics. This convention follows the format {sampling}{switching}{edge} and uses appropriate abbreviations [4] for natural (N) or uniform (U) sampling, AD or BD for switching class, and double-sided (D) or single-sided (S) edge modulation. So, for example, NBDD indicates natural sampling, class BD, and double-sided modulation.

Now that an encapsulating convention has been issued, the combined characteristics can be ranked accordingly. These conclusions follow those drawn earlier but offer the additional assessment of priority among classification schemes. First off, natural sampling is much preferable to uniform sampling, while hybrid approaches offer compromised performance. Within these main groups, the other characteristics rank in a similar manner, so the remaining scope can be minimized to consider only natural sampling. In terms of modulation quality from best to worst: NBDD, NBDS, NADD, NADS [4]. From this, one can conclude that sampling method is of the highest priority, followed by switching class and finally by edge modulation.

b. Sigma-Delta (ΣΔ) Modulation

Sigma-Delta modulation has gained great acceptance for use in analog-to-digital converters (ADC), but it is also an alternative to PWM for class D power amplifiers. Sigma-delta modulation is actually a form of Pulse Density Modulation (PDM), which is comparatively depicted in Fig. 3.6b.

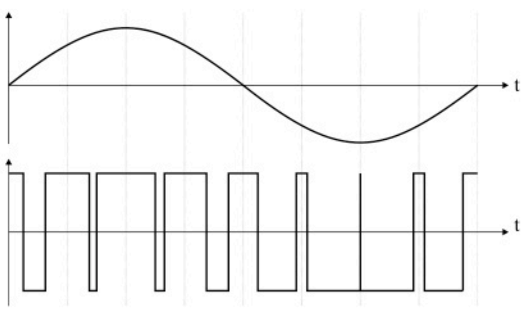

Fig. 3.6a - PWM Waveforms [4].

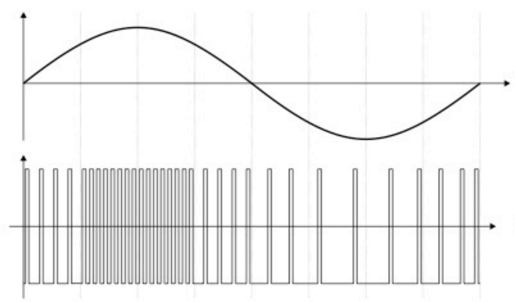

Fig. 3.6b – Sigma-Delta Waveforms [4].

Instead of a continuously varying pulse width corresponding to signal amplitude, PDM pulse widths are all equal and the signal amplitude is encoded in the presence or absence of pulses. A high density of pulses corresponds to a high amplitude, and a low amplitude is encoded with a low density of pulses.

Fig. 3.7 shows a basic block diagram of a first-order sigma-delta modulator. Although 1-bit solutions are conceptually simple, in practice they are only stable to 50% modulation [1], so multi-bit modulators are often required.

In sigma-delta modulation, oversampling is typically used to extend quantization noise bandwidth, thereby reducing in-band noise. The sigma-delta scheme performs noise-shaping, which distributes quantization noise disproportionally towards higher frequencies, as shown in Fig. 3.8, further lowering in-band noise. Upon low-pass demodulation filtering, the high frequency quantization noise is removed, leaving only the desired audio signal.

A significant advantage of sigma-delta modulation is that the high frequency modulation components are distributed over a wide range of frequencies, rather than being concentrated in tones around multiples of the carrier frequency like in PWM. This presents a potential reduction in EMI, since noise levels are typically lower than that for PWM concentrated components, as demonstrated in Fig. 3.9.

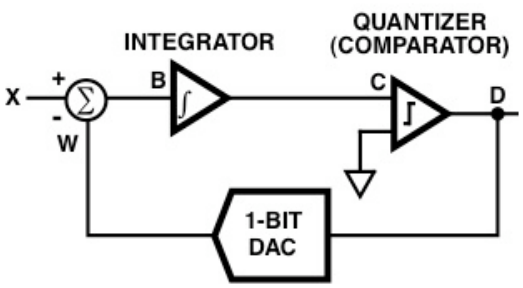

Fig. 3.7- Block Diagram of a First-Order Sigma-Delta Modulator [11].

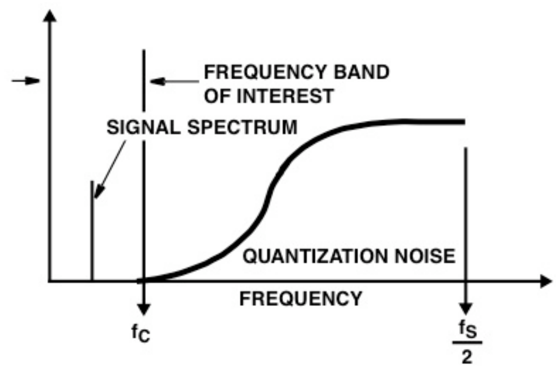

Fig. 3.8 - Noise-Shaped Quantization Noise Spectrum [11].

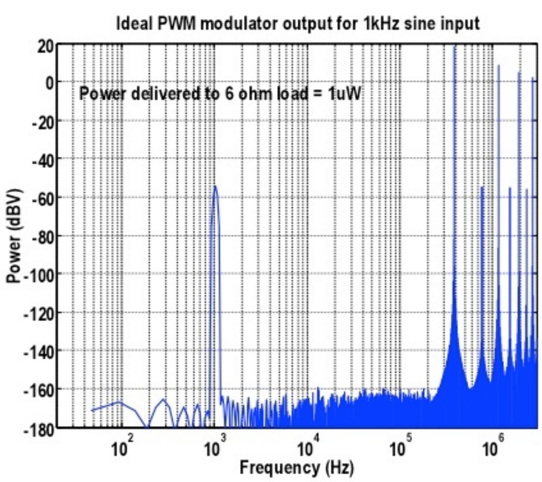

Fig. 3.9a - Unfiltered PWM Spectrum [12].

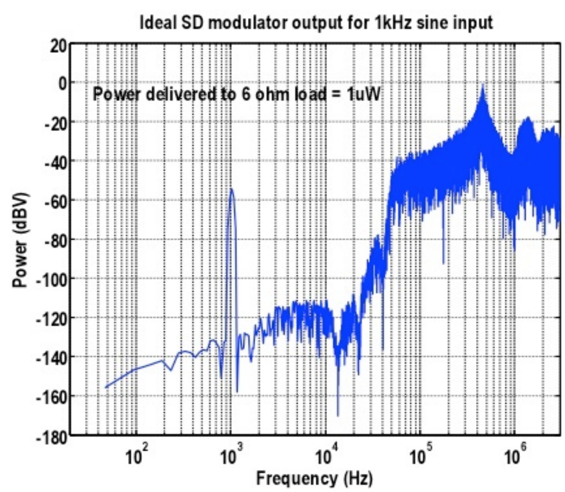

Fig. 3.9b - Unfiltered Sigma-Delta Spectrum [12].

4. Topologies & Output Stage

The previous section dealt with modulation technique, which is certainly the core of class D operation. However, there are many other considerations in a complete design. This section discusses issues of implementation in semiconductor ICs and larger systems, power FET characteristics and configurations, and output low-pass filter details.

a. System Implementation

Although discrete designs exist and can be practical (most notably Hypex's UcD technology [13]), the majority of class D applications utilize integrated circuit (IC) solutions. This approach makes sense for applications like mobile phones and flat-panel televisions where an inexpensive, compact, efficient, and easily implemented package is requisite.

In this case, class D design can generally be divided into two groups: integrated circuit (IC) and final product. Both groups certainly require design from a systems perspective, but the scale on which they operate is entirely different. Engineers working on a final product, such as an mp3 player or a home theatre receiver, must incorporate class D ICs into the complete design. It is critical then that both groups maintain lines of communication, since one's success is inextricably dependent on the other's.

Class D design is a very multidisciplinary practice, relying on disciplines of circuit design, RF principles, material science, and layout, among others. The skill set necessary differs from conventional linear amplifier requirements.

Circuit layout must be carefully considered in a class D design. Generally, traces should be kept as short as possible so as to minimize EMI issues from high frequency radiation, and also to reduce losses for higher efficiency. These considerations contribute to the advantage of integrated packages.

Many low powered devices are able to use a complete class D chip, but amplifiers above a few tens of watts require a controller separate from the power stage [2]. Other solutions also use separate gate drive circuitry in addition to the controller and power stage [2]. The need for this separation typically involves limitations on IC dissipation and size.

Before discussing some important aspects of external power FETs, an interesting example of increased integration should be mentioned. Since a class D amplifier is very similar to a synchronous buck converter (a common DC-DC converter used in switching power supplies), and since switching power supplies are commonly used with class D amps, it is natural to consider combining the power supply and amplifier into one integrated unit. This can be accomplished by replacing the intermediate DC bus with a high frequency AC link. This configuration is referred to as SICAM (SIngle Conversion stage AMplifier) [14], and its main advantage is higher efficiency through direct energy conversion from AC mains to audio output. The basic configuration is compared in Fig. 4.1.

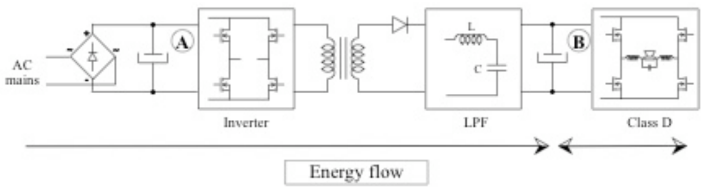

Fig. 4.1a - Conventional Class D [14].

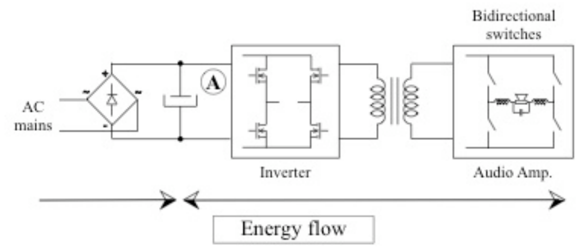

Fig. 4.1b - SICAM [14].

b. Power FETs

The power semiconductor switches are predominantly MOSFETs, due to their superior switching speed over alternatives like BJTs [15]. In fact, modern advances in device performance are largely responsible for stimulating class D development.

The design and operation of the output FETs is a large contributor to efficiency and EMI characteristics. However, instead of discussing these important topics here, the issues are presenting in the corresponding sections (5.a. and 5.d.) below. It should be noted, though, that losses within the power FETs (although minimum compared to dissipation in linear devices) often require heatsinks, and this should be considered when designing.

Section 3.a.ii. on class AD and BD modulation addressed the distinction between half-bridge and full-bridge (also known as H-bridge or bridge-tied-load - BTL) output FET configurations. As mentioned, class BD modulation requires the use of an H-bridge [1], and this is certainly one advantage of the BTL configuration, since class BD is generally preferable [4].

Just as with conventional amplifiers, full-bridge class D designs permit twice the output voltage and hence a 4x theoretical increase in maximum power output (since $P=V^2/R$) compared to a half-bridge operating from the same supply [16]. This is particularly advantageous in automotive and portable applications where high voltage supplies are generally unavailable. Additionally, H-bridge designs do not require DC-blocking capacitors when operating from a single supply, as necessary for half-bridge configurations [16].

Another advantage of full-bridge configurations is their immunity to power supply bus pumping [1], which can occur in half-bridge designs. Since energy flow can be bi-directional (from supply to load and vice-versa) in a half-bridge design [6], and since most power supplies are unable to sink current, bus capacitors can charge up and result in increased voltage [8]. This phenomenon, known as bus pumping, is depicted in Fig. 4.2 and occurs primarily at low frequencies and for low load impedance [6].

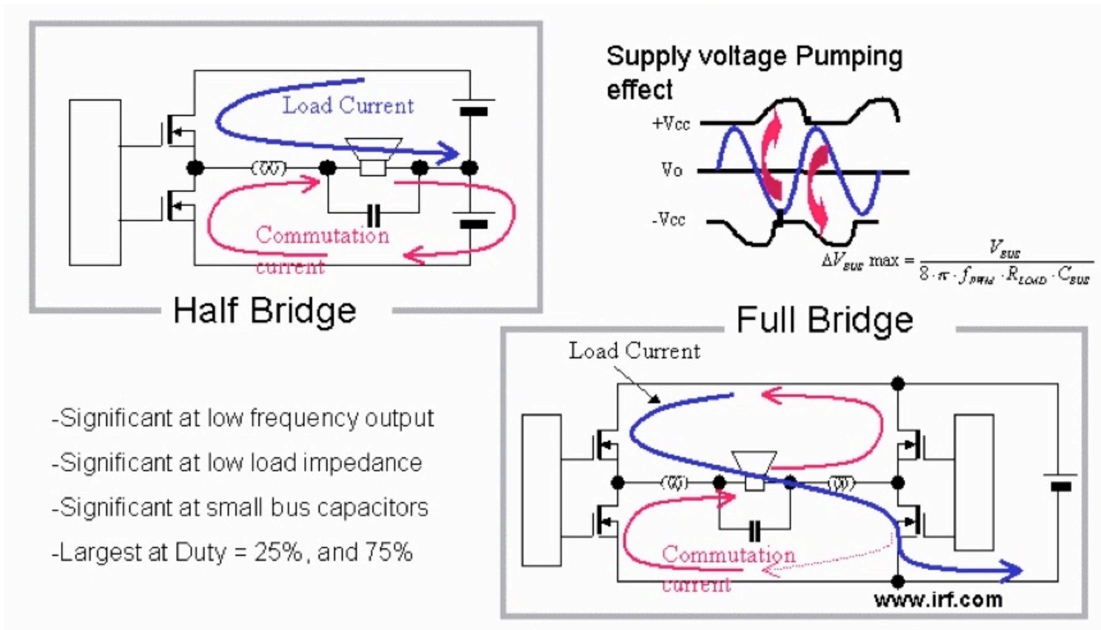

Fig. 4.2 - Bus Pumping [6].

However, full-bridge designs are not always the best choice. The fact that they use twice as many devices can influence cost as well as efficiency. More devices means more losses, but this is generally only significant for high-power amplifiers (greater than 10W), and consequently, half-bridge designs are usually used for high-power applications [16].

Since the bus pumping issue discussed above involves the power supply, this seems an appropriate place to mention another important point about class D amplifiers. For open-loop (i.e. no feedback) designs, the rail voltage directly establishes the level for the PWM output signal, resulting in a power supply rejection ratio (PSRR) of essentially 0dB [6]. Since a large PSRR value is desired, this issue must be carefully accounted for through control techniques.

c. *Output Low-Pass Filter*

With a design as complex as a class D amplifier, it may be surprising that a component as simple as an LC low-pass filter could be so important, but this should certainly not be overlooked. This filter has the critical purpose of demodulating the audio signal from the PWM square wave by removing high frequency switching components.

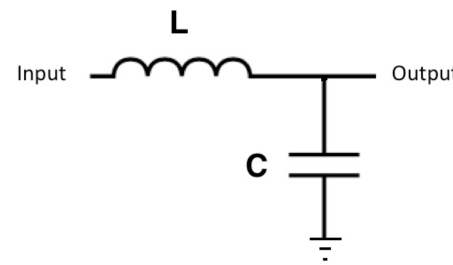

Fig. 4.3 - Second-Order LC Low-Pass Filter

As vital as this filter is, it can have a significant impact on the amplifier's price, size, and distortion performance. In order to minimize cost and board space, many class D amps use a simple second-order LC filter [1], as shown in Fig. 4.3.

Unfortunately, these simple non-ideal components can introduce additional distortion. This can be reduced by including the filter in a feedback loop, but this is not always practical. The most important design aspect to ensure is that the inductor current rating exceeds the highest current expected of the amplifier. This is because inductor cores can magnetically saturate with excessive currents, resulting in a drastic reduction of inductance [1].

Some applications, like portable devices, require the elimination or drastic reduction of the LC filter. "Filterless" designs can be achieved using advanced modulation techniques [16], although a modest filter is still often used with these schemes. In fact, the inductance of the loudspeaker load acts as a filter, as does the human ear's response [16]. However, EMI concerns must always be addressed when designing the filter scheme.

5. Performance Characteristics

The previous sections of this paper have dealt with the operation and design of class D amplifiers. Now that the fundamentals have been covered, some very important performance characteristics can be addressed. Efficiency, distortion, and electro-magnetic interference (EMI) are discussed in this section.

a. Efficiency

Efficiency is arguably the most important specification for class D amps because high efficiency is the fundamental advantage they offer over linear amplifiers. Increased efficiency provides other important advantages like longer battery life and smaller size.

In a linear class AB output stage, efficiency is dependent on the output level relative to the supply voltage. In this case, efficiency increases to a maximum theoretical value of 78.5% when the peak output level reaches the supply voltage [17]. It is important to realize that this theoretical maximum is rarely approached when amplifying music (as opposed to pure sinusoidal signals), due to the large crest factors typical of musical programs [2].

Class D, on the other hand, boasts a theoretical maximum efficiency of 100%. Although practical losses will reduce the achievable value, efficiency generally remains very high for class D amps, even for low output signal levels. Losses are an extremely important aspect of this issue and will be discussed shortly. Typical efficiency versus output values are compared for Class D and Class AB in Fig. 5.1 below.

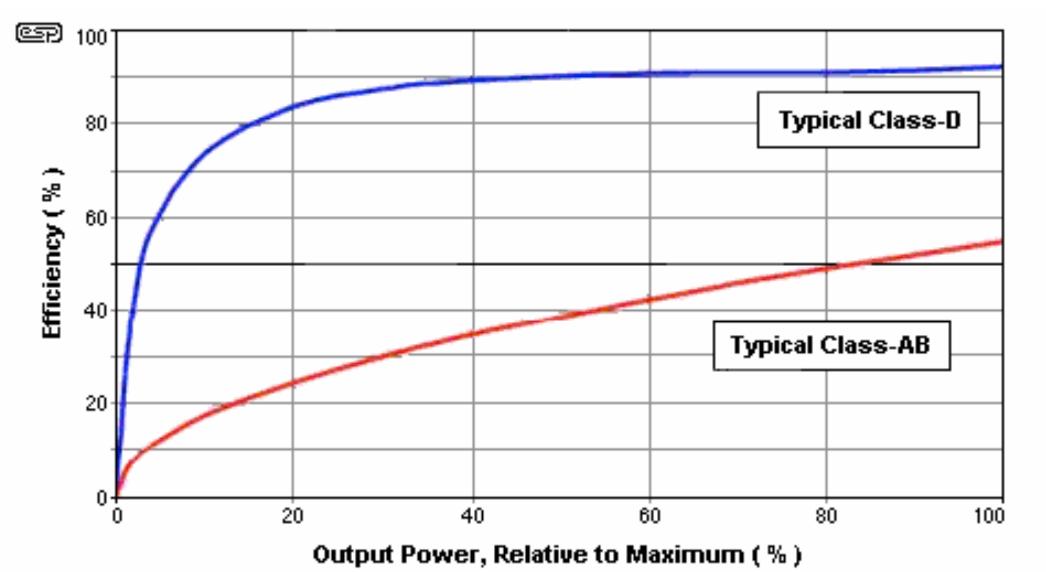

Fig. 5.1 - Efficiency comparison for typical Class D and Class AB [18].

Before discussing inherent losses in class D designs, another issue should be demonstrated. Although purely resistive loads are often used for testing purposes, practical speaker loads are typically reactive. This can have drastic effects on efficiency of class AB amps. Fig. 5.2 shows efficiency versus output level and load phase angle for class AB and class D amplifiers. Notice that class D achieves higher efficiency over a relatively large region compared to class AB.

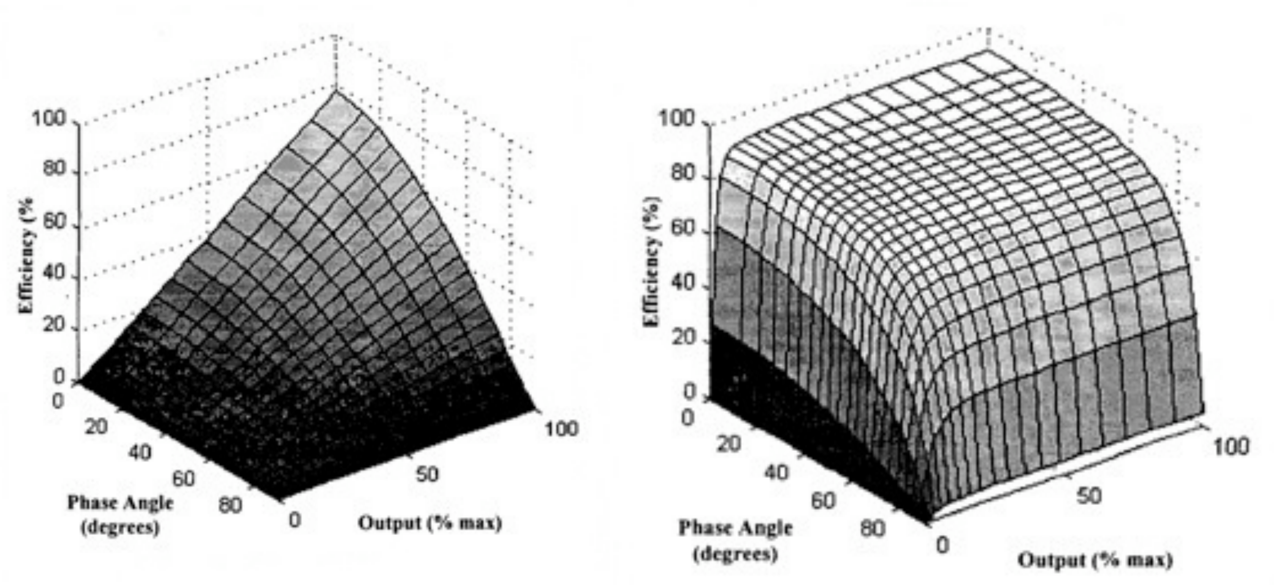

Fig. 5.2 – Typical efficiency of Class AB (left) and Class D (right) for reactive loads[17].

The reason class D has increased efficiency for higher phase angles of reactive loads has to do with the way it handles currents. The complex impedance of a reactive load produces in-phase and quadrature components in the load current. However, only the in-phase component produces power in the load, while the quadrature component will attempt to flow back and forth between the source and load every cycle. Since class AB amps only permit power flow from source to load, the energy associated with the quadrature component cannot return to the supply. Instead, it is dissipated in the output devices, reducing efficiency and putting extra stress on the output devices [17].

Class D amps, on the other hand, are required to allow bidirectional current flow in the output devices due to their requisite inductive low-pass filter. This capability permits class D amps to re-circulate quadrature load current back to the supply, thus providing the efficiency improvement across load phase angle shown in Fig 5.2 [17].

Practical losses reduce the efficiency of a class D amplifier. Future device technology improvement will likely reduce these losses, but any design can still be optimized for best efficiency. The two main sources of loss in a class D output stage are conduction losses and switching losses. There are also losses associated with the gate drive, but these are another form of switching loss. Conduction losses dominate at high output and switching losses dominate at low levels [1]. These sources of loss combine to form the total loss. When designing, the given application should dictate the optimal balance between these sources of loss.

$$P_{TOTAL} = Pcond + Psw + Pgd$$

Conduction losses arise from the finite output resistance, $R_{DS(on)}$, of the output MOSFETs, as shown in the equation to the right [6]. This resistance is ideally zero, but typical values are around 0.1 to 0.2 Ω [1, 19]. When the MOSFET conducts current, this finite output resistance causes power dissipation in the device. This conduction loss is more significant for higher output levels because the power dissipation increases with output current. Reducing $R_{DS(on)}$ increases efficiency.

$$Pcond = \frac{R_{DS(ON)}}{R_L} \cdot Po$$

Switching losses, along with gate drive losses, are more significant at lower output levels where the conduction losses are less severe. The equations below [6] demonstrate the dependence on switching frequency, as well as other parameters. Some of the more unfamiliar parameters may be C_{OSS} MOSFET output capacitance, t_f fall time, and Qg gate charge.

$$Psw = C_{OSS} \cdot V_{BUS}{}^{2} \cdot f_{PWM} + I_D \cdot V_{DS} \cdot t_f \cdot f_{PWM}$$

$$Pgd = 2 \cdot Qg \cdot Vgs \cdot f_{PWM}$$

The most significant relationship here is that switching losses and gate drive losses increase with switching frequency f_{PWM}. A high switching frequency is generally desired to increase bandwidth and reduce the low-pass demodulation filter requirements, but the higher the switching frequency the higher the switching losses. There is a trade-off between these factors when designing.

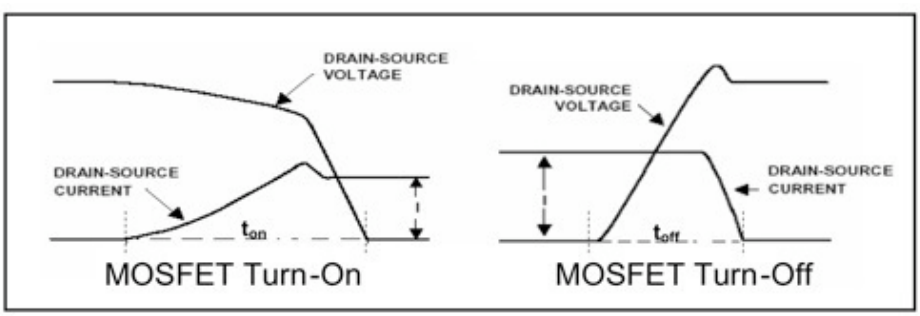

Fig. 5.3 - V*I Switching Losses During Turn-On and Turn-Off [15].

Switching losses can be conceptualized by considering the drain-source voltage and current during turn-on and turn-off, as shown in Fig. 5.3. Since neither of these characteristics can change instantly, there is a finite length of time over which current and voltage are both present, producing power dissipation. Since this loss occurs for every transition, it should be clear why switching losses increase with switching frequency.

Gate charge Qg, the amount of charge required to fully turn on the MOSFET, is another important parameter because it also has an effect on efficiency. A lower Qg value will result in lower switching losses and hence higher efficiency [15].

There is a trade-off between Qg and $R_{DS(on)}$ to consider when designing or choosing MOSFET die size. A larger die will result in lower $R_{DS(on)}$ (lower conduction loss) and higher Qg (higher switching loss), while a smaller die will have the opposite effect. Due to the distribution of these effects, as shown in Fig. 5.4, there is an optimal die size for minimizing losses.

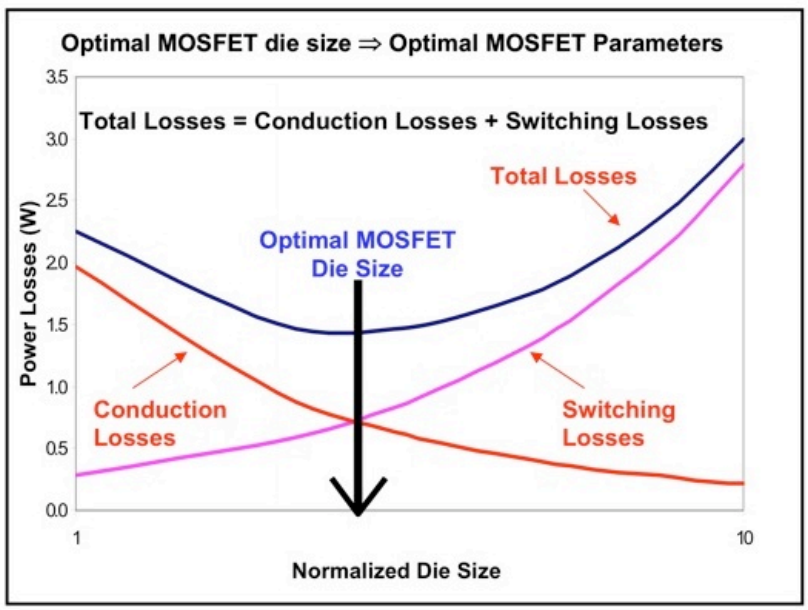

Fig. 5.4 – Trade-Off Between Conduction Losses and Switching Losses when Designing Die Size to Minimize Losses [15].

b. Distortion

The primary figure of merit in assessing the audio quality of an amplifier is its distortion or nonlinearity, often measured as total harmonic distortion (THD). There are a number of sources of distortion in a class D amplifier, and these generally differ from those typical of a linear amplifier. The primary source of distortion is due to the deadtime necessary to prevent shoot-through current. Nonlinearity of LC filter components can also be a significant cause of distortion. Variations in supply voltage, finite rise and fall times, parasitic oscillations, and modulator clipping can also contribute to distortion in class D amplifiers.

The MOSFETs in a class D amplifier switch between the positive supply and negative supply to create the PWM output. During normal operation, only one device conducts at a time. However, during the transition there is the potential for both devices to conduct simultaneously, providing a path between the two supplies and causing excessive current flow, known as shoot-through or cross conduction [1, 8]. As a worst case, this shoot-through can permanently damage the devices, but even modest amounts of shoot-through can reduce efficiency significantly [1].

In order to prevent shoot-through current, a "break-before-make" approach ensures that both transistors are off before the next one turns on. The time during which both FETs are off is called deadtime, and this unfortunately leads to pulse-shape errors in the PWM waveform [1, 8], as shown in Fig. 5.5. The small rectangular notches on the PWM waveform are caused by the forward bias voltage of the backgate diodes [8].

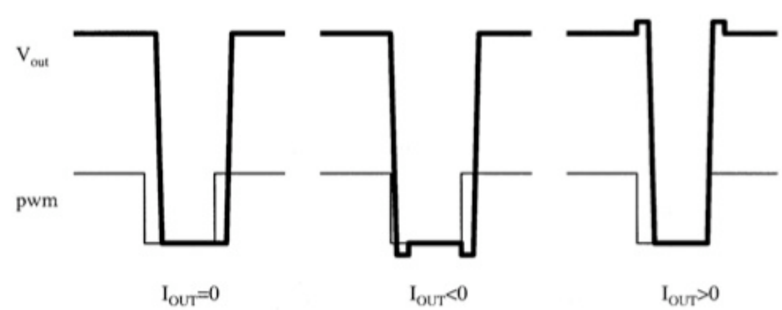

Fig. 5.5 - Deadtime Distortion [8].

Deadtime should generally be made as small as possible to avoid shoot-through [1, 8]. As an example, a 40ns deadtime can produce 2% THD, while that figure can be decreased to 0.2% by lowering deadtime to 15ns [6].

The relatively simple LC lowpass filter may not seem too suspect as a root of distortion. However, this is actually one of the main sources, due to hysteresis and saturation in the inductor's magnetic core [20].

In an open-loop class D amplifier, the output LC filter is essentially connected to the power supplies through the small resistance of the FETs, producing very little power supply rejection (PSR). Therefore, any noise or changes in the power supply voltage couple directly to the output voltage, causing distortion [1].

Finite rise and fall times also produce distortion, so larger dV/dt values are desirable [10]. Also, parasitic components can cause transient edge ringing [6]. Another type of distortion occurs for excessive input levels that push the modulator past the limits of its modulation depth. This can produce severe waveform clipping, similar to that for linear amplifiers, and should be avoided whenever possible. Some modulators can even skip pulses when approaching full modulation, creating additional waveform distortion.

Feedback is often used to correct for these various nonlinearities, but there are important considerations when designing a closed-loop class D amplifier. Most importantly, stability must always be ensured under all operating conditions [16], as with any feedback system. This can be achieved through careful design and compensation.

Continuous-time analog feedback is usually necessary, but this can unacceptably increase cost for some IC applications [1]. One alterative is for a digital open-loop modulator to employ an analog-to-digital converter (ADC) to sense power supply variations and compensate the modulator's behavior accordingly. This can improve PSR but unfortunately not distortion performance [1]. Another approach with a digital modulator is to precompensate for errors and nonlinearities. This can not, however, address all distortion mechanisms [1]. Some form of feedback is usually necessary to achieve high audio quality.

To overcome some of the problems with conventional single-loop feedback, a double feedback loop architecture can be used, as shown in Fig. 5.6. This technique involves an inner current loop and an outer voltage loop [9, 20].

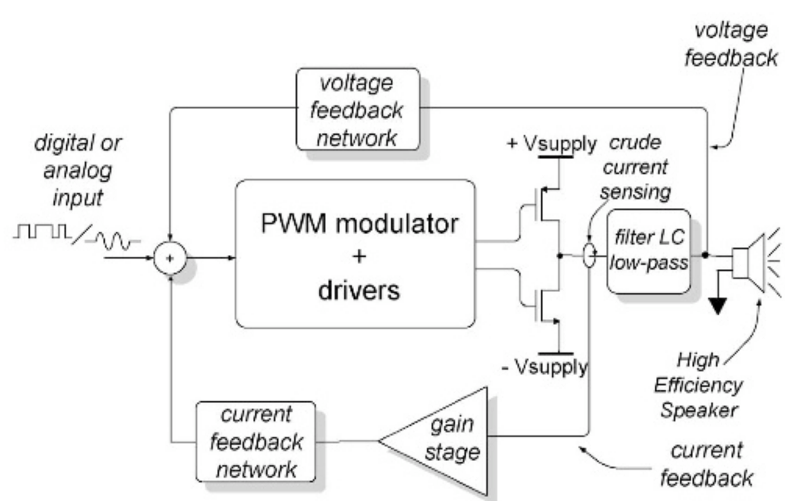

Fig. 5.6 - Double Feedback Loop Architecture [20].

Although the current loop increases the output impedance of the inner loop, the outer loop provides low overall output impedance for the amplifier, as desired [9]. One advantage of the double loop configuration is that the current feedback acts as an intrinsic current limiter, allowing low impedance loads (below 1Ω) to be driven and eliminating the need for overcurrent protection circuitry. The system behaves like a current-mode or transconductance amplifier for low load impedances and like a voltage amplifier for high load impedances [20]. Also, the current feedback corrects power stage errors close the their source, which is desirable [9].

In a class D amplifier with traditional feedback, the LC lowpass filter is typically outside the feedback loop. The consequence of this is that the feedback cannot compensate for the nonlinearity of the filter components [9, 20]. Additionally, the amplitude response is dependent on load impedance in this case, which is unsatisfactory for many applications where the load is often unknown [9]. The double loop architecture is able to include the LC filter within the feedback path, correcting for its nonlinearity and providing greater insensitivity to load impedance [9, 20]. Additionally, since the double feedback scheme compensates for LC filter nonlinearities, smaller cores can be used in the filter inductor, providing potential advantages in size and cost [20].

In addition to feedback control for improved linearity, other protection mechanisms should be incorporated into any amplifier design. These include protection for high temperature, over/undervoltage, overcurrent, short-circuit (low impedance) output, click and pop suppression or prevention, and general robustness.

c. Electro-Magnetic Interference (EMI)

Electro-magnetic interference (EMI) can be a serious problem in class D designs and is arguably the most significant drawback of a switching amplifier. However, certain design techniques can be used to reduce EMI.

EMI is caused by high frequency energy radiated from signal wires and traces. An example of EMI on a time domain signal is shown in Fig. 5.7 as prominent spikes.

The high frequency switching components, although suppressed by the output low-pass filter, form the baseline spectrum for EMI emissions.

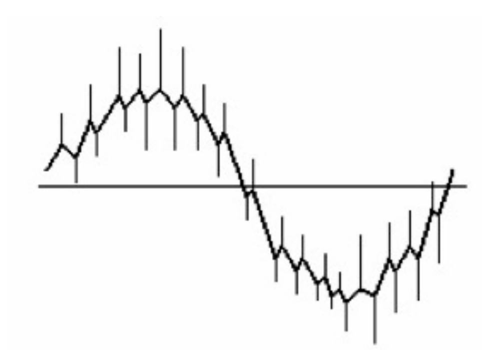

Fig. 5.7 - EMI Spikes Visible on a Signal [6].

The primary source of EMI in class D amplifiers is the reverse recovery current associated with the backgate or body diodes of the MOSFETs [6, 8]. Fig 5.8 depicts the process. In step 1, the low-side FET is conducting, but then in step 2 since the inductor current must continue flowing during the dead time when both FETs are off, the body diode consequentially conducts. Then in step 3, when the high side FET turns on, the body diode is still conducting in order to wipe away minority carrier charge (Qrr or reverse recovery charge) stored during forward conduction, forming a momentary short between supply rails. This causes large, high frequency currents which can excite resonances and lead to EMI [1, 6, 8].

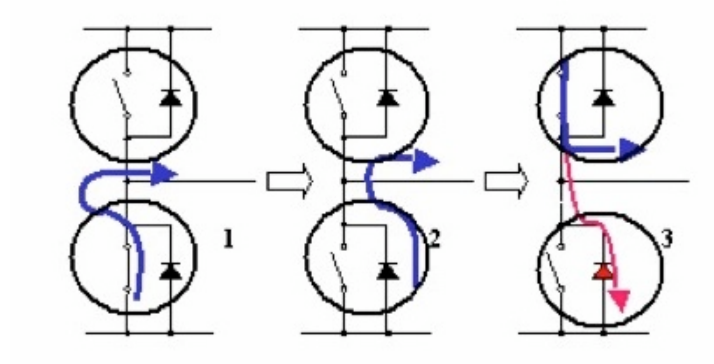

Fig. 5.8 - Reverse Recovery Current [6].

Reducing dead time can reduce EMI from reverse recovery current [1, 15] as well as reduce distortion, but dead time must be sufficiently large to prevent shoot-through, as discussed in the section above on distortion. Another approach to reducing EMI from reverse recovery effects is to put Schottky diodes in parallel with the MOSFETs' intrinsic body diodes. This prevents the body diode from ever conducting by diverting current into the Schottky, which is intrinsically immune to reverse recovery effects [1]. This has an additional benefit of improving efficiency [19].

Another way to reduce this source of EMI is to dampen the resonances with external snubber networks across the FETs or power supplies [8]. Additionally, board-level layout design techniques can reduce EMI by minimizing the area of loops that carry high-frequency currents. One way to do this is to keep traces for current drive and return paths close together. Also, the LC filter should be laid out compactly and located close to the amplifier [1]. Long speaker wires cause more EMI radiation, so these should be kept as short as possible for the given application.

Another technique for reducing EMI is to use an alternative modulation scheme called spread-spectrum modulation. This technique dithers or randomizes the switching frequency by up to +/-10% of the nominal value. Although the period varies randomly, the duty cycle remains unaffected. Effectively, the switching energy is distributed or spread throughout a wider spectrum, rather than being concentrated at a set switching frequency and its harmonics. This reduces the peaks of the high frequency switching energy, minimizing EMI emissions.

6. Conclusion

Class D switching amplifier technology has seen tremendous growth in the past decade, and the future looks just as promising. As demand increases for smaller, more efficient electronic devices, class D amplifiers will likely increase in popularity. This, in turn, will stimulate further development of the technology.

This paper examined the operation, design, and performance of class D amplifiers. Modulation schemes, including Pulse Width Modulation (PWM) and Sigma-Delta ($\Sigma\Delta$) were discussed. Different PWM methods were compared, including natural & uniform sampling, class AD & BD switching, and single-sided & double-sided edge modulation. Design topologies and output stage, including power FETs and low-pass filter, were discussed. Finally, the paper concluded with a section addressing performance characteristics, namely efficiency, distortion, and electro-magnetic interference (EMI).

The aim of this paper was to introduce the reader to class D amplifier design through a general, comprehensive overview. It is the hope that the information provided be accessible to novices yet useful for more advanced readers as well. Thorough investigations of the topics discussed are available in the literature, and readers are encouraged to consult the following references for more information.

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