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<u>PLEASE NOTE</u>: This is NOT a complete paper. This paper is unfinished and more like an *outline* of what I learned after conducting a comprehensive research of Class D amplifier solutions, their implementation in consumer applications, and the practical considerations of using them.

The reader is welcome to skim this document for quick information about class D amplifiers, but I strongly recommend exploring the extensive list of references accompanying this document. *Not just the references at the bottom of this outline, but there is also a separate 6-page document full of URL links to relevant class D material I found useful during my research.

Class D Amps

Class AB

For decades, the class AB amplifier configuration has been the favored solution in audio power output stages. Class AB is implemented as a push-pull class B with two output devices, but the output devices are slightly over-biased so that they both conduct when the push-pull output voltage is in the crossover region near signal ground. In other words, the two output devices operate with conduction angles slightly greater than 180 degrees so that the device conduction cycles overlap briefly to ease the transition from high-side device to low-side, and vice versa.

The class AB output stage has been so popular because it can achieve linearity comparable to class A, but with efficiencies levels closer to class B, which has a theoretical efficiency of 78.5%. Theoretical class B efficiency assumes lossless components and the case of 78.5% efficiency occurs only when the output is a sine wave driven to peak rail-to-rail amplitude without clipping. In reality, components are not lossless and audio amplifiers rarely operate at continuous peak output power. Consequently, it has been estimated that a typical class AB audio amplifier used at nominal output levels has an average efficiency of 40 – 50%. [2,4] Efficiency levels under 50% have traditionally be seen as a necessary tradeoff in achieving highly linear power amplification. But over the past decade the growing market emphasis on mobile applications along with advancements in switching technology have led to a new generation of highly efficient class D amplifier solutions.

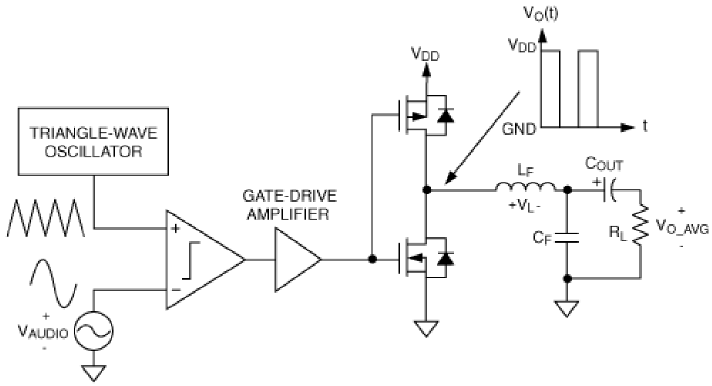


Figure 1 - Simplified class D with half-bridge output stage [1]

Class D

A simplified class D amplifier is illustrated in Figure 1. A class D audio amplifier is inherently a “switching” amplifier because it reproduces an audio input signal as a pulse-width modulated (PWM) output. Switching amplifiers operate on the same fundamental principles as a synchronous step-down “buck” converter [2]: The output is switched between the high and low supply rails and the duty cycle of each switching period is made proportional to the instantaneous input voltage. Figure 2 illustrates the pulse-width modulation.

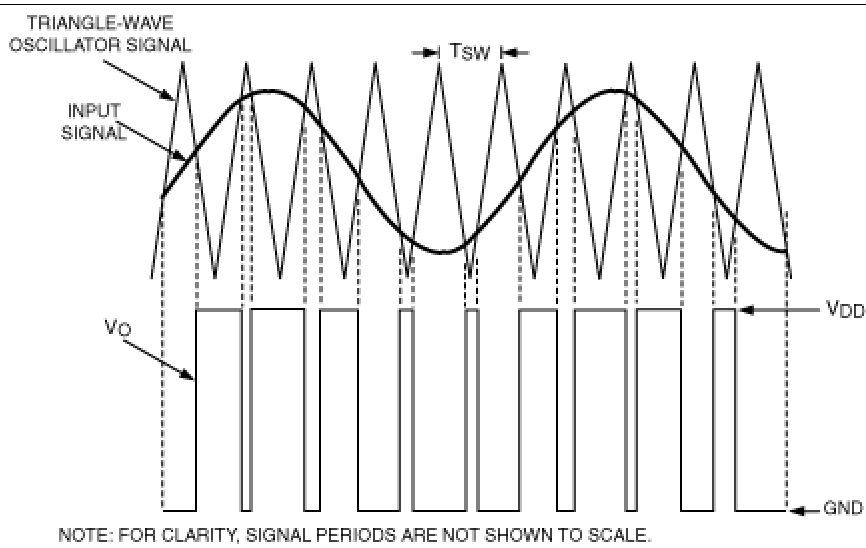


Figure 2 - The output-signal pulse widths vary proportionally with the input-signal magnitude. [1]

Typical switching frequencies (fs) in class D amps range from 250kHz to 1.5MHz. [1] If 20kHz represents the highest audio input frequency, then the minimum number of PWM samples per audio period could be between 12.5 and 75, with lower audio frequencies having even better resolution. As the output stage generates the PWM pulse train, the LC output filter is connected to the speaker load to form a low-pass RLC filter with cutoff frequency set just above the audible band. The output filter is 2nd-order and significantly attenuates the HF switching frequency (fs) and its harmonics, leaving the reconstructed audio waveform in the pass band to drive the loudspeaker load.

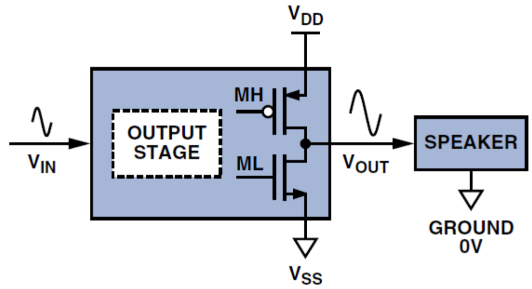


Figure 3 - Linear Amplifier [5]

Like the buck converter, the class D amplifier realizes significantly higher efficiencies (>90 %) across a wider operating area compared to linear alternatives. A fundamental part of how linear amplifiers and regulators function is that the output devices operate in the linear region, drawing load current from the supply rails through the output devices, during which the voltage magnitude is dropped from |VDD| or |VSS| down to |Vout|. Significant power is dissipated through this |VDD|-|Vout| drop in each of the output device when they are conducting load current. This is why a linear class AB amplifier has a theoretical max efficiency of 78.5%, while the theoretical max efficiency of a class D amp is 100%.

Early Class D Amplifiers

The concept of a switching amplifier was first proposed in 1958 [6]. It took a couple decades until fast enough transistors were developed that could switch at a high enough frequency to obtain sufficient samples of the audio input waveform. For this reason, class D designs were first successfully commercialized in automotive subwoofer applications [4]. Subwoofer amplifiers typically have a bandwidth of only a few hundred hertz, so the switching frequency (fs) of a switching subwoofer amplifier could be a low as a couple tens of kilohertz. Working with such a low switching frequency could greatly simplify system design by reducing the amplifiers sensitivity to slow MOSFET rise and fall times as well as reduce switching losses.

The performance of early class D amps was less than impressive. The amplifiers were open loop and as a result they had nonexistent (0dB) power supply rejection (PSR). With zero feedback, any other noise signals injected into the system could not be compensated for. Class D solutions were offered for general purpose and not designed with specific applications in mind. Device manufacturers encouraged application engineers to implement their amps with a control loop (to increase linearity and provide PSR), but manufacturers could offer little advice as to how to begin designing such a complicated loop. It was often too much of a chore to try to design a feedback loop and it would have added too many additional parts and taken up too much additional board space. [4]

With zero feedback, the THD+N of many of these early designs was typically greater than 0.5% [4], which is considered to be relatively poor linearity. The PWM outputs also required large LC filters to extract the audio, further increasing the size and cost of the application. The size of these switching amplifiers prevented their use in small, battery-powered applications (where their efficiency really pays off), and their poor THD+N made them undesirable compared to class AB amps. Because of all these complications, class D amplifiers were unpopular in the beginning, gaining acceptance mostly only in very high power applications. Applications such as subwoofer systems and PA systems were perfect to benefit from being able to squeeze a much higher power amplifier into a much smaller volume, reducing the size, cost, and weight of the entire application. But improvements in switching technologies and new design techniques would later seed the emergence of well-designed application-specific integrated class D solutions.

Modern Class D

- more efficient
- better linearity ➔ low THD+N
- many single-chip devices have integrated feedback [4]
    - National Semiconductor, LM4673, boasts about integrated global negative feedback taken from the H-bridge output: [4]
        - *"Now the effect of any mismatch, jitter, finite rise/fall time, or supply noise present at any point in the amplifier signal path is reduced resulting in class D amplifiers with class AB quality audio."* [4]
        - THD+N < 0.02%
        - PSRR = 78dB @ 217 Hz
        - see additional data plots ➔ [4]p4-5

- Filterless output
- small size

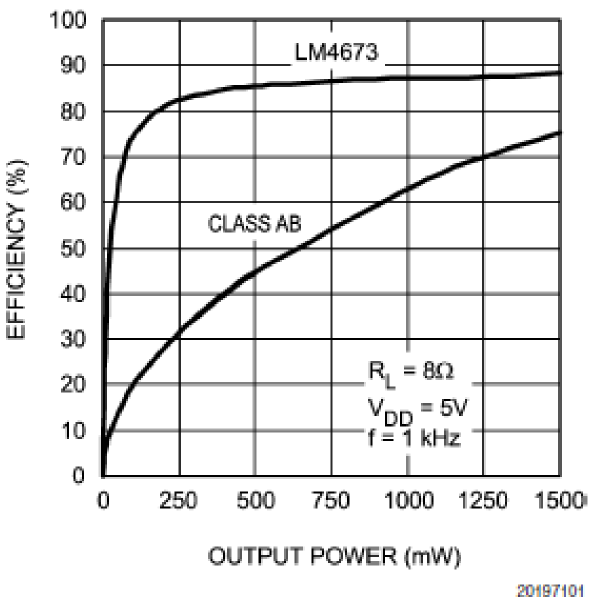
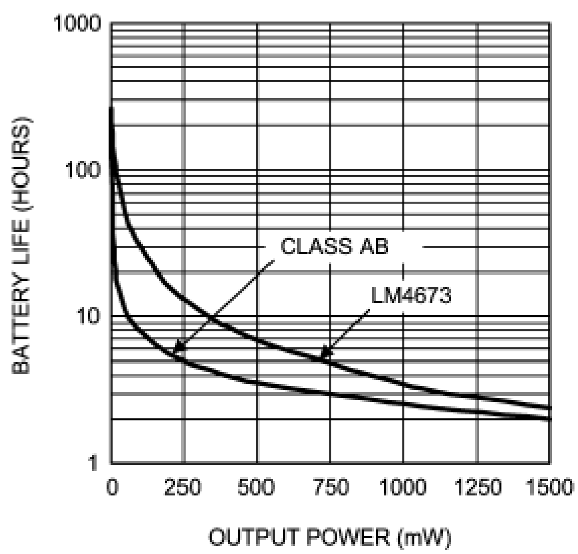


FIGURE 1. LM4673 vs. Class AB Efficiency



Class D vs. Class AB on Efficiency & Battery Life [4]

More on Class D Operation

There are generally two methods of output stage (OPS) switching. The simpler of the two is the complementary push-pull output stage shown in Figure 1, called a half-bridge. The second and often preferred type of output stage is called a full-bridge (or H-bridge) and is composed of two half-bridges driving the load differentially in the bridge-tied-load (BTL) configuration. Both of these techniques are discussed below.

Half-Bridge:

A half-bridge output stage consists of a single pair of complementary MOSFET in a push-pull configuration that can switch the output back and forth between the two supply rails. The output is single-ended and slightly more efficient than the full-bridge because it has only one switch RDS(on) in the output path, whereas the full-bridge has two. For this reason a half-bridge output stage is sometimes preferred in power outputs in excess of 10W. [1] At these medium-high output powers the load current becomes more significant and MOSFET conduction losses via RDS(on) can become relatively substantial. In these cases, a full-bridge output would experience double conduction losses compared to the half-bridge.

In addition to having one half the conduction losses of a full-bridge OPS, the half-bridge also has the advantages of simplicity, small size, low parts count, and along with low parts count comes the cost savings of only needing one half-bridge and one level-shifter / gate driver for each output channel. In high power applications, the cost of power MOSFETs can be substantial (on the order of several dollars per device), so too would be the cost savings from having the simpler half-bridge output stage.

Cons: - cannot implement 3-level PWM
- kick-back energy can damage power supply (“bus-pumping”)

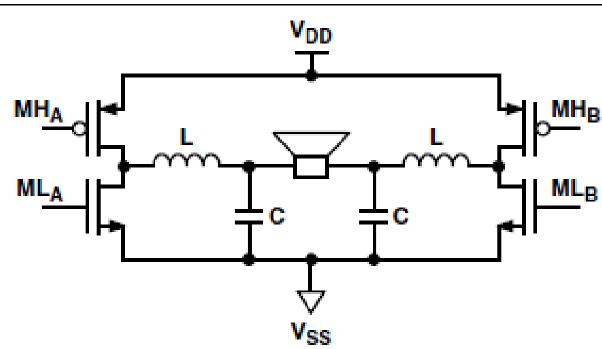


Figure 4 - Full-Bridge BTL Output Stage with LC Output Filter [5]

Full-Bridge: (or, **H-Bridge**) [1], [2],

As was already mentioned, full-bridge BTL is the preferred configuration for class D output stages. This is true because of a few small but important benefits offered by the full-bridge OPS. First, the differential nature of the BTL output connections permit not only traditional 2-level PWM, but also a 3-level PWM scheme that employs a third quantization level at signal ground. This 3-level PWM technique has greatly advanced the performance of modern class D amps enabling such features as filterless output and more advanced feedback with noise-shaping, all of which will be explained later.

And additional feature of the full-bridge BTL OPS is one that finds particular use in low-voltage single-supply applications. In such an application, the BTL OPS can switch each of the load terminals between VDD and GND. In one orientation the voltage across the load is +VDD. In the reverse orientation the voltage across the load is -VDD. Therefore, a BTL single-supply OPS is capable of driving the load with a peak to peak amplitude of 2*VDD. This is twice the Vout swing of the half-bridge OPS, and by Ohm's Law (P = V^2 / R), doubling the output voltage swing effectively translates to a theoretical quadrupling of the peak output power. This makes the full-bridge BTL configuration an ideal solution for mobile applications operating from a low-voltage single-supply.

An additional feature of the full-bridge BTL OPS is that the load is driven directly by the two H-bridge outputs and therefore does not require an output coupling capacitor to block a DC offset, because there is no offset. A half-bridge OPS running on a single supply would have to reference the output PWM signal to a mid-supply reference voltage and an output coupling capacitor would be mandatory.

Some disadvantages of the full-bridge OPS is first the fact that it is essentially composed of two complete half-bridge output stages, with double the size and cost of a single half-bridge. As already noted, the full-bridge is also slightly less efficient than the half-bridge because it places two switches in the load current path effectively doubling the conduction loss [1]. However, the conduction losses in a full-bridge OPS only become significant at relatively high output currents. Thus, for low-power portable applications, the extra conduction loss in the full-bridge is negligible and much more important is the full-bridge's ability to drive a load to double the supply voltage, a very useful feature for any low-voltage application.

> **Pros:** - H-bridge can be implemented with improved modulation schemes ➔ like *3-level PWM* which has 3 output states: V_S+, 0V, and V_S-. This 3-level modulation scheme has the advantage of an output stage that idles at a constant 0V DC output (instead of 50% duty cycle pulses, as in traditional 2-level PWM). This type of modulation has significantly lower output filtering requirements.

Modulation Schemes

2-Level, Pulse-Width Modulation

The most common type of modulation used in class D amplifiers is pulse-width modulation (PWM). [5] Referring back to Figure 2, this type of PWM can be classified as 2-level PWM because the output pulse train has two states, VDD and GND. This technique is easy to implement with a triangle wave, audio input, and comparator, but a somewhat undesirable effect of 2-level PWM is that even when the input is zero, the PWM output will continue switching

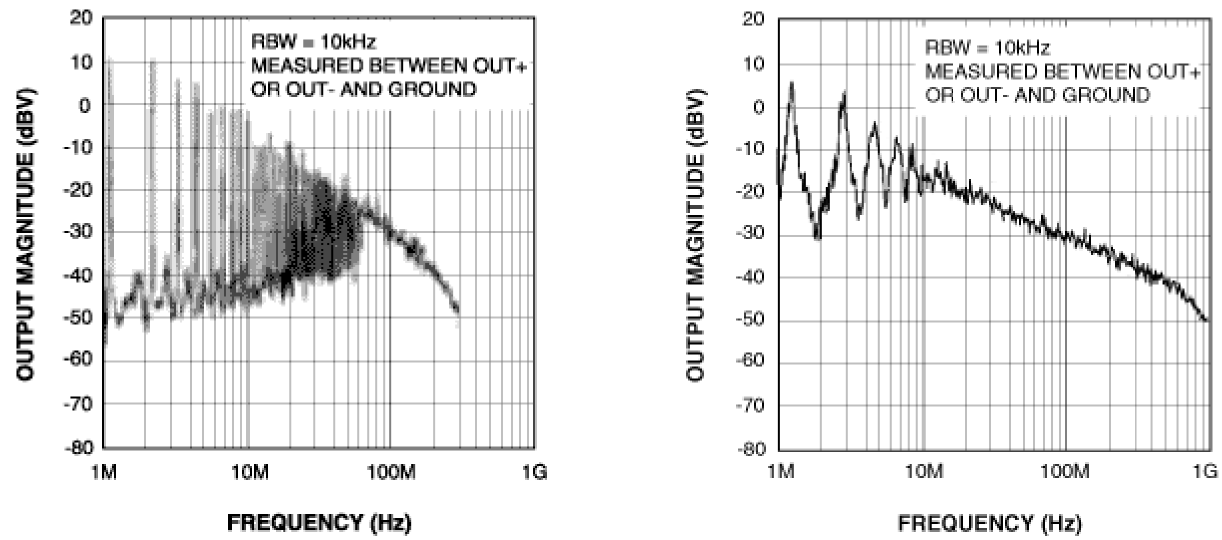
3-Level, Pulse-Width Modulation

A more advanced modulation scheme is 3-level PWM, which requires a full-bridge OPS to implement. 3-level PWM works by

- **Traditional** scheme (**Natural Sampling**): 2-level PWM with a single half-bridge OPS [4]
    - duty cycle idles at 50% for zero input
    - these +/- Vs output pulses need a filter to extract the audio before sending it to the load
        - Filterless is not an option
            - dissipate HF energy in speaker load ➔ bad for EMC
    - high and low-side switches are ~180 degrees out of phase (complementary)

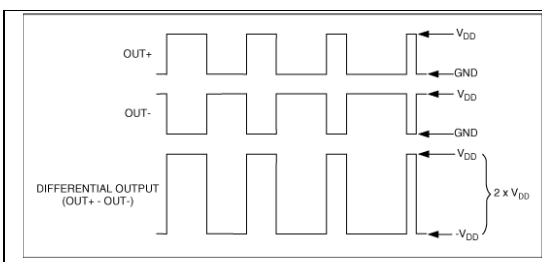
- **Three-Level Modulation**: requires an amp w/ H-bridge BTL output stage ➔ in which both speaker terminals are driven by a half-bridge.
    - When idle, H-bridge differential outputs {Vo1, Vo2} are in phase ➔ voltage across load idles at 0Vdc.
    - greatly reduces filtering requirements
    - to further reduce switching noise emissions, manufacturers have implemented a variety of <u>noise-shaping</u> & <u>noise-suppression</u> techniques in the control loop. ➔ important part of achieving filterless design
        - Maxim uses its patented "<u>Spread-Spectrum" modulation</u> to spread out the fs spectral energy on the output. [1]
            - randomly varies fs by +/-10% from nominal value
            - reduces the magnitude of spectral energy spikes
            - total spectral energy is the same, but it's been spread out into higher frequencies with reduced energy peak (at the nominal fs and its harmonics)
                ➔ reduced spectral energy peaks = reduced risk of EMI emission
        - Maxim's spread-spectrum modulation significantly reduces EMI
            - but you should still minimize cable lengths to ensure you pass FCC EMC inspection

- if output cables are long, add ferrite bead to the output wiring near the amp to reduce cable emissions.



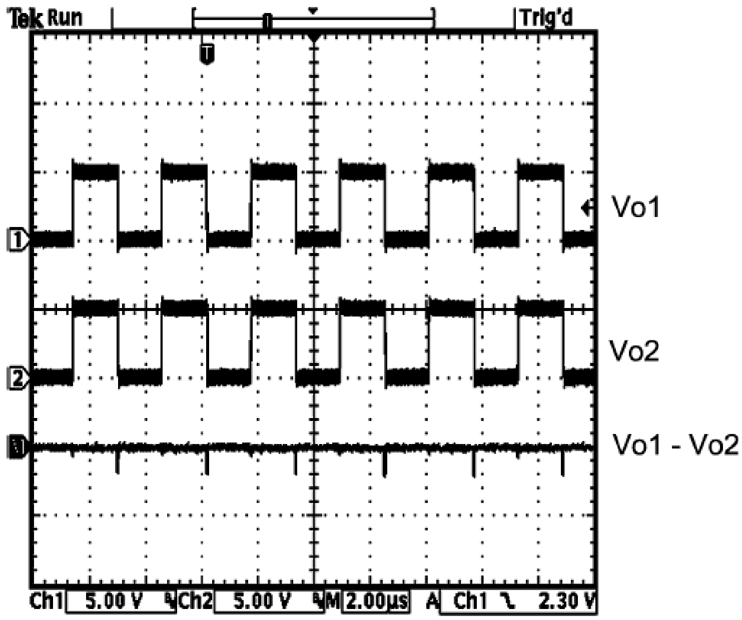
Output spectra of class D amps:
- the amp in the Left plot has a fixed fs,
- the amp in the Right plot has Maxim's spread-spectrum noise-shaping
    -

- 3-level PWM is more efficient than 2-level PWM: [4]
    - in 3-level PWM: Vo1 and Vo2 idle in phase
        ➔ voltage across the load idles at zero Vdc
    - in 2-level PWM: Vo1 and Vo2 are always 180 deg out of phase
        ➔ differential output voltage is constantly switching between 0V and 2*Vs
        ➔ differential output idles at 50% duty cycle
            - each output pulse drives the RLC filter back and forth
            - as energy goes back and forth in the filter, power is wasted in the filter's parasitic [4]



Traditional, 2-level BTL PWM waveform [1]

- In an H-bridge driven w/ 2-level PWM, the Vo1 and Vo2 PWM outputs remain in complementary states and always switch together.
- Differential voltage across load continually switches between 0V and 2*Vs



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Differential Outputs in 3-level PWM @ Zero Input Voltage [4]

- Vo1 and Vo2 are in phase
- Differential [load] voltage idles at 0Vdc

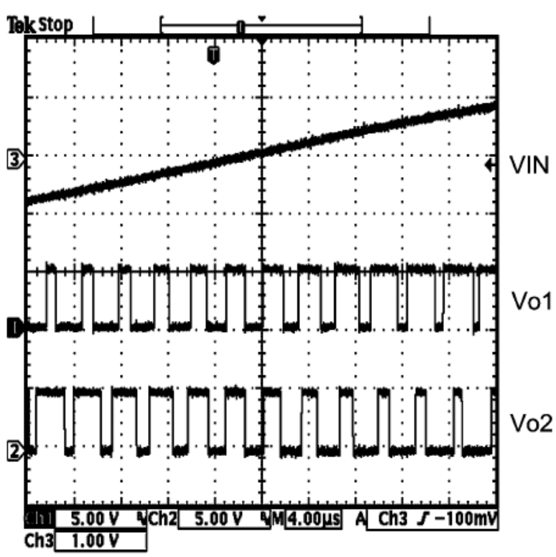

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FIGURE 7. Switching Waveform with Input Signal

3-level PWM driver signals (Vo1, Vo2) and audio input (VIN) [4]

- if VIN goes positive ➔ D_{Vo1} goes > 50%, D_{Vo2} < 50%

Feedback: (from PWM output, to the audio input)

- improves linearity
- can achieve PSRR > 80dB ➔ not bad… getting better [3]
- allows noise-shaping in the feedback loop ➔ further improves linearity & PSR [1]
    - modern class D amps use multi-order noise-shaping topologies [1]
- control loop must be carefully compensated to ensure stability [1]
    - open-loop class D amps are perfectly stable b/c feedback is zero
- if amp has digital input, the input must be converted to analog before adding feedback [3]

Battery / Efficiency

are capable of very high efficiencies (around 90%) due to the switching nature of the output stage.

As illustrated in Figure 1, the basic class D output stage is a complementary pair of transistor switches (usually MOSFETs) arranged in a push-pull configuration. The high-side and low-side complementary MOSFETs switch the output level between the positive rail and ground, with a duty cycle proportional to the audio

Class D: (switching amp) – output devices are either fully on or fully off (like switches)

- *see graph* for class D vs. class AB efficiency data
    - at maximum Pout: 88% vs. 75%
    - at nominal Pout: 85% vs. 44%
- gain is proportional to supply voltage [2]
- pure digital amplifier has no PSR (0dB) [2]
- modern designs use feedback to achieve high PSRR. Feedback compensates the system for analog errors such as the noise injected by power supply ripple.
- similar to Synchronous Buck Converter [2]
    - Buck Converter: the reference voltage V_{REF} is usually a slow-moving (often fixed) DC reference potential, producing a relatively fixed output duty cycle.
    - Class D: the V_{REF} input in an audio class D amplifier is the incoming audio signal. The audio signal has a constantly changing voltage which produces a constantly varied duty cycle.

Basics: (Intro) Class D first proposed in 1958 [5]

- simple PWM modulation with triangle (sawtooth) wave [1]
- fs = 250kHz to 1.5MHz [1]

Output Duty Cycle (D):

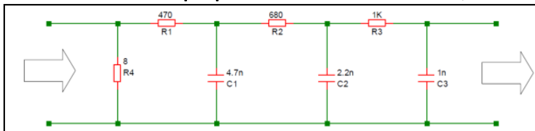
- D is proportional to input voltage level [1]
- D idles at 50% [1]

Output Filter:

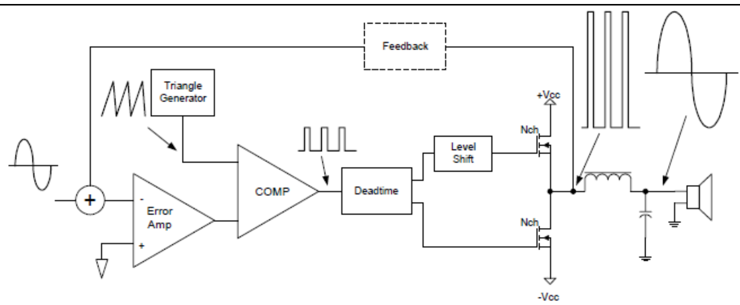
- extracts the audio waveform from the output [1]
- prevents high frequency energy from being dissipated in the load [1]

Measuring the Audio Output: [2]

- to measure the audio waveform→ must remove the HF switching component first
    - switching noise in the output spectra will affect audio measurements
    - modern audio measuring equipment already has input filter, usually AES-17 brick wall filter
    - if equipment has no filter, use a discrete LP filter



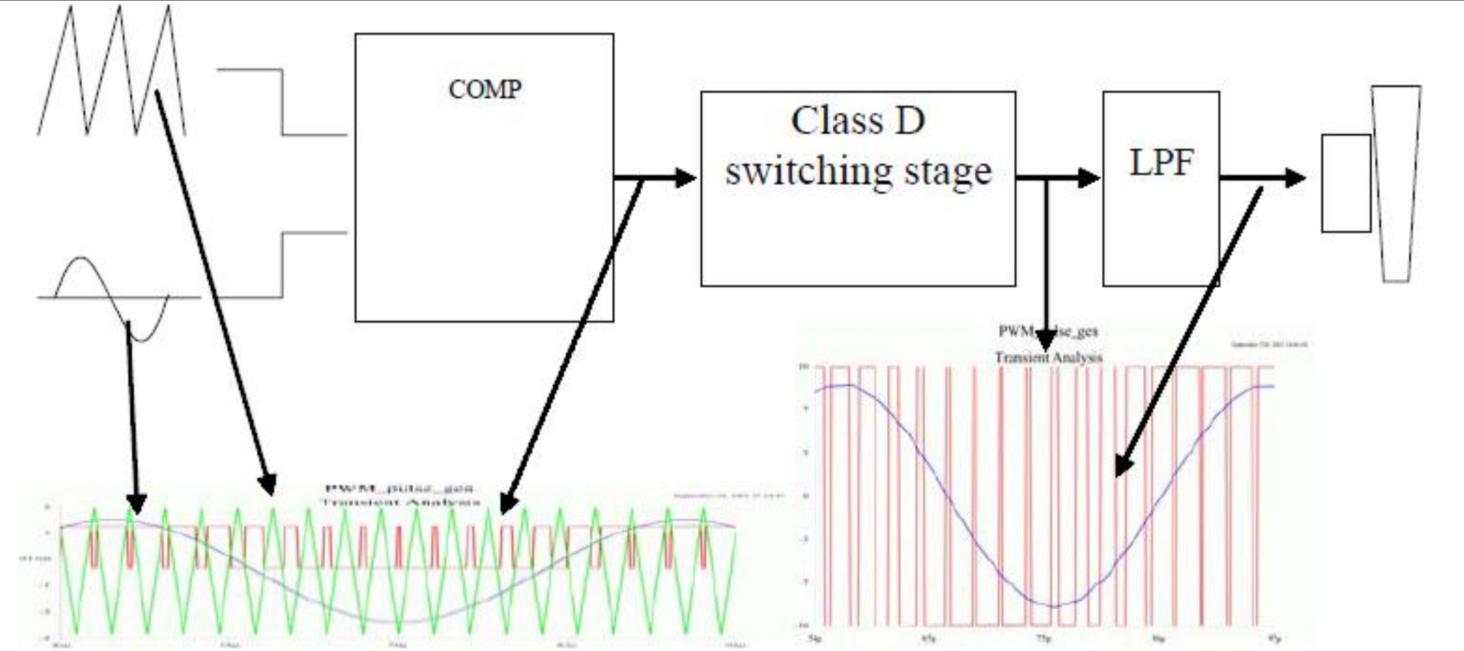
LP Filter for measuring audio output [2]



Simplified Block Diagram of Class D Amplifier [2]

Class D Applications: MP3 player, mobile phone handsets, laptop, LCD TVs, home theater, high-power audio, etc.. [1,3]

- main advantages in:
    - Portable Devices: higher efficiency ➔ longer battery life ➔ very small size
    - Plug-In Devices: higher efficiency ➔ less power consumption/dissipation
        ➔ power supplies become smaller & less expensive
        ➔ less heat dissipation = smaller heatsinks ➔ smaller size
            ➔ more output power per unit volume of amplifier



Simplified Block Diagram with Illustrated Waveforms [2]

Switching Frequency: (f_S)

- clock (f_S) stability is crucial [3]
    - timing errors cause nonlinearity (distortion)
    - clock is usually on-chip, low-noise PLL
- higher f_S :
    - greater switching losses in OPS
    - reduced filtering requirements:
        - filter can be smaller & less expensive
    - better audio SNR due to higher PWM resolution (like oversampling with a sigma-delta filter)

Error in Class D:

- comparator offsets
- device mismatch
- oscillator jitter ➔ timing error
- finite rise time
- supply noise

Output Stage (OPS):

Linear OPS:

- transistors operate in linear region ➔ large bias current ➔ huge power loss

Switching OPS:

- high efficiency
- sources of power loss: [1]

- transistor on-resistance R_{ON} (conduction loss)
- switching loss
- quiescent (idle) current overhead

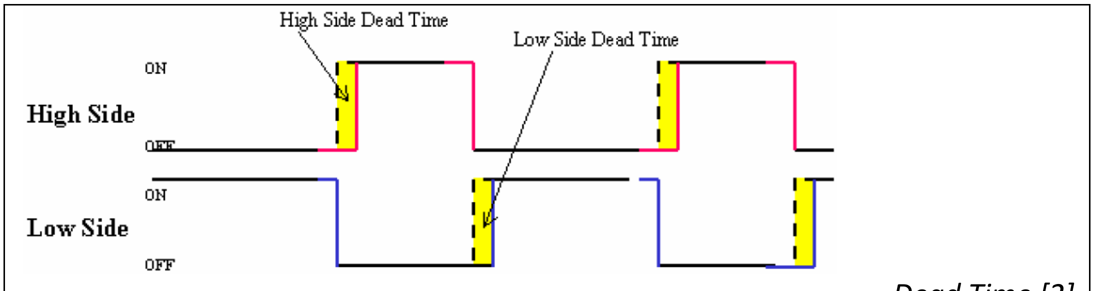
- total power loss in the output devices: [2]
  - PT = Psw + Pcond + Pgd
  - $Psw = C_{OSS} \cdot V_{BUS}^2 \cdot f_{PWM} + I_D \cdot V_{DS} \cdot t_f \cdot f_{PWM}$

$$Pcond = \frac{R_{DS(ON)}}{R_L} \cdot Po \qquad Pgd = 2 \cdot Qg \cdot Vgs \cdot f_{PWM}$$

As can be seen in a Class D amplifier the output losses are dependant on the parameters of the device used, so optimization is needed to have the most effective device, based on Qg, $R_{DS(on)}$, C_{OSS}, and t_f. Fig 6 below shows the power losses vs K for the Class D amplifier.

Class D - sources of distortion: [2, unless otherwise stated]

- ideal class D: no distortion, no noise, no loss (100% efficiency)
- actual non-idealities:
  1) nonlinearity in the PWM signal → due to limited resolution or timing jitters
  2) timing errors added by gate-drive → e.g. dead time, t_{ON} vs. t_{OFF}, t_R vs. t_F
  3) non-ideal switching characteristics → e.g. nonzero Rds(on), finite switching speed, rise & fall times, gate capacitance
  4) parasitic components → causes ringing on transient edges
  5) power supply fluctuations → nonzero output resistance
  6) nonlinearity in the output filter → inductor DCR, capacitor ESR
- main source of distortion: Timing Errors in gate drive (associate w/ dead time)
  - accurate switching time is essential!
  - timing errors cause bad distortion
    - 40ns dead time can easily generate THD>1%



Dead Time [2]

- Dead Time: brief time interval when both high-side & low-side switches are OFF (not conducting)

- must have at least some dead time to ensure high/low switches are never ON at the same time → shorts the supply rails
- seamless transition between high- & low-side conduction cycles → essential for good linearity.

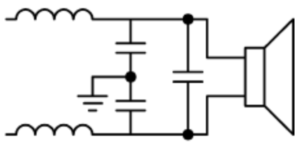
THD vs. Dead Time:

15ns → 0.18% THD

40ns → 2.10% THD

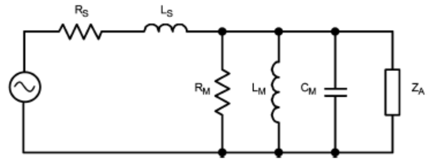
Output Filter: - to extract the audio waveform & sufficiently attenuate f_S component to pass FCC EMC certification [3]

- external LC filter → large size, high cost (especially in high-power amps)
  - filter components must be rated for peak output power
- usually RLC 2^{nd}-order low-pass w/ cutoff frequency just above the audio band



2^{nd}-order LRC Output Filter [4]

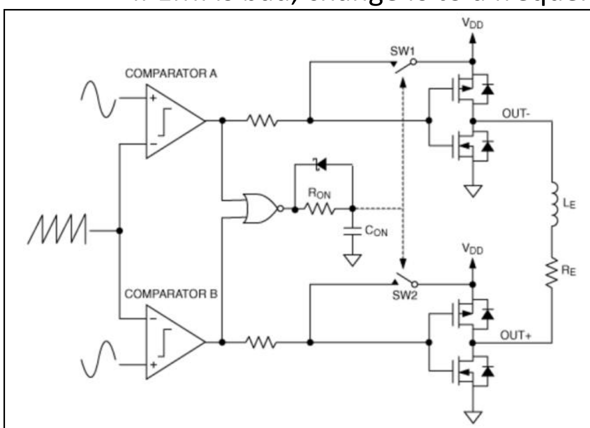
- LC component parasitics
  - reduce efficiency (especially in high power apps) [4]
    - inductor DCR (dc resistance)
  - component parasitics & nonlinearities increase THD+N [4]
- switching freq:
  - Higher fs → easier to filter but less efficiency (more switching loss)
  - Lower fs → more efficient but harder to filter (filter must be larger, more expensive)
- Speaker Has Finite Frequency Response:
  - voice coil inductance and DC resistance → 1^{st}-order LR low-pass filter
    - helps to further attenuate HF on the output signal
  - audio transducers are acoustically designed for audio frequency response
    - driving a speaker with unfiltered PWM output probably won't cause any fs vibration [4]
    - even if a speaker could reproduce fs → nobody would hear it! (inaudible)



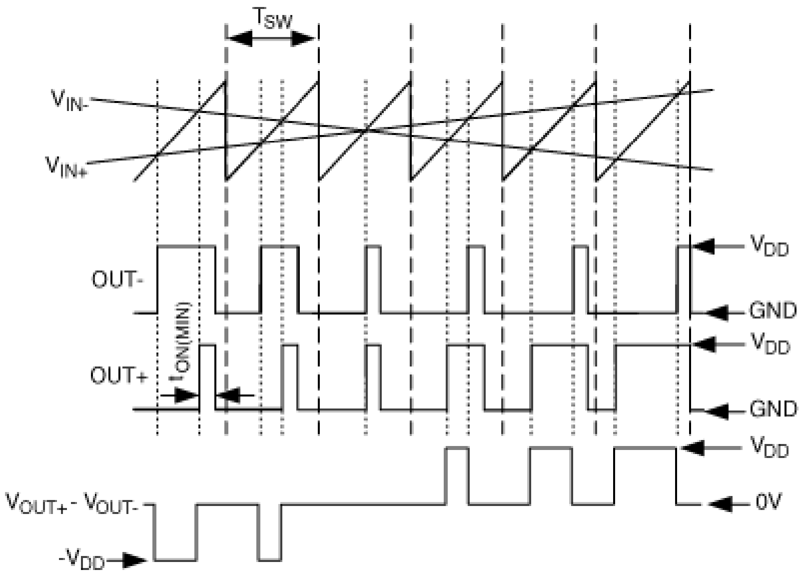
20197104 *Simplified Model of Speaker Load* [4]

Filterless Class D: 3-level PWM controller idles at Vout=0 and needs minimal filtering (speaker is enough)

- if amp is very close to speaker ➔ parasitic R and L of speaker can be suitable RL filter [3]
- if amp is far from speaker ➔ ferrite bead can add inductance to improve EMC [3]
    - reduces RF emissions on long speaker cable
    - can help pass FCC EMC tests
- EMI is still a big challenge in filterless Class D amps
    - w/out a filter on the output, HF energy (fs and harmonics) will radiate down the speaker cable
    - manufacturers are designing filterless class D with noise-shaping in the feedback loop to further reduce .
    - Maxim uses patented "Spread-Spectrum" modulation to spread out the fs spectral energy on the output.
        - randomly varies fs by +/-10% to reduce the magnitude of spectral energy spikes
        - distributes the noise more evenly ➔reduced spectral energy peaks = lower EMI emission.
    - some switching amps have programmable fs
        - if EMI is bad, change fs to a frequency that does not cause EMI in your application



Simplified functional diagram of MAX9700 filterless Class D modulator topology, manufactured by Maxim [1]



The input and output waveforms are shown for the MAX9700's filterless modulator topography [1]
- illustrates 3-level PWM differential output waveform (idle at zero)

Low Power Class D: (P_{OUT} < several Watts) ➔ good for battery life, small size, low cost

- class D advantage:
    -
- usually a single chip, no external heatsink
- for highly-integrated portable applications ➔ class D amps are available as part of an entire audio CODEC or Subsystem
-

High Power Class D: (P_{OUT} > several Watts) [3] ➔ good for less power dissipation, smaller heatsink size, high-power amps can be more compact (i.e. more watts output power per unit volume of amplifier)

- separate controller and OPS
    - OPS = MOSFET switches + level-shifter/drivers
    - OPS switches can be discrete or integrated
        - usually integrated b/c very important to have FETs w/ matched parameters
        - OPS transistor switches must have matching high- and low-side switching delays
            - otherwise, mismatched delays can cause the ON-times of two complementary devices to overlap and momentarily short the supply.
                - distortion
                - supply noise: short-circuit pulses produce dips on the supply rail [3]
                - devices get hotter ➔ wasted power; reduced efficiency
    - an integrated audio channel has 4 FETs and level-shifter/driver
        - connect directly to the IC controller outputs

Power Supply:

Open-Loop Amps: good supply regulation is ESSENTIAL!
- a completely digital amp has PSRR = 0dB [3]
- audio-band supply ripple will modulate the output signal directly! [3]
    - PSRR is 0dB b/c output is continually switching between the supply rails
    - supply must have good load regulation in the audio band
        - especially if 2+ audio channels share the same supply
            - driving the speaker (load current) causes audio-band supply ripple
            ➔ crosstalk between audio channels that share supply rails

Feedback Amps:
- feedback provides PSR ➔ compensates for supply fluctuations

Power Supply HF Transient Performance: must be good at high frequencies for OPS to accurately reproduce PWM waveform.
- supply current pulses with fast edges ➔ caused by switching on/off load current
    - supply must react quickly to sudden changes in supply current
    - slow output rise time will cause distortion
- linear amps have supply currents mainly in the audio band
    - power supply does not need to be much faster than 30kHz [3]
    - power supply is easier to design
- improve HF transient response by adding high-value, low-ESR caps
    - expensive - capacitors may be too large for some portable apps
- reduce peak supply current spikes in multi-channel systems:
    - clock each channel with a phase-delayed clock so the amps don't switch in unison [3]
        - reduces peak instantaneous load current
        - reduces crosstalk
        - employed in Wolfson WM8608 5.1- to 7.1-channel digital power amp [3]

Switching Power Supplies:
- common w/ class D amps b/c of efficiency & small size[3]
    - National Semi offers a class D chip w/ integrated dc-dc converter supply
- added source of EMI; does NOT make it easier to pass FCC law
- Amp & Power Supply should be clocked together (same frequency)
    - if not, intermodulation can produce tones that might be audible [3]

Thermal Considerations: [7, unless otherwise noted]

- test thermal performance using real audio (voice or music) instead of sinewaves
  - crest factor: CF = (peak amplitude) / (RMS amplitude) *[reference: my EE Dictionary]*
  - real audio (music/voice) has higher crest factor than sinewave:
    - sinewave CF is approx 3dB
    - voice CF is approx 12dB
    - music CF is approx 20dB
    - ➔ testing with real audio is crucial for accurate thermal testing
  - sinewave represents worst-case scenario for thermal

PCB design:

TQFP - microcontroller

  - TQFP (*Thin Quad Flat Package*): exposed pad on bottom side is primary path of heat transfer.
    - PCB & copper traces are primary heatsink
    - solder the exposed pad of TQFP to a large copper polygon on the PCB (under the IC)
      - add lots of copper to the polygon, especially where it connects to other traces and pins
      - add vias from the polygon to the other side of the board where there is a similar polygon
    - add as much copper as possible without upsetting signal routing
    - make copper paths as wide as possible (for heat transfer)
    - make all TQFP pins traces as wide as possible
      - pins are not primary thermal path, but still dissipate some heat.
    - use external heatsink for higher output power
      - mount to copper polygon, not the device' plastic top
- MOSFET conduction losses, Pcond, are a function of Rds(on) and drain current, Id
  - Pcond can be reduced by selecting a speaker with higher impedance
    - higher Z ➔ lower current ➔ lower Pcond
    - as long as your supply rails can deliver your desired Pout to the higher-Z load
    - remember that speaker Z varies widely w/ frequency.

Class D APPLICATIONS: MP3 player, mobile phone handsets, laptop, LCD TVs, home theater, high-power audio, etc.. [1,3]

- main advantages in:
    - Portable Devices: higher efficiency ➔ longer battery life ➔ very small size
    - Plug-In Devices: higher efficiency ➔ less power consumption/dissipation

➔ power supplies become smaller & less expensive

➔ less heat dissipation = smaller heatsinks ➔ smaller size

➔ more output power per unit volume of amplifier

NOTE: “clickless” and “popless” signal switching has become standard in virtually all integrated audio solutions. [6]

Examples:

LCD TV monitors: [6] with integrated stereo speakers ➔ perfect for class D!

- typical application requires 5 – 15 watts peak Pout per channel
- want to keep the TV enclosure as flat as possible, preferably not beyond the dimensions required for the LCD display module.
    - no space for large heatsink
- want to minimize heat
    - LCD pixel color is temperature-sensitive [6]
    - LCD drivers can easily compensate for temperature changes that are uniform throughout the display;
        - but not good at compensating for hot spots [6]

➔ Now explain the class D solution

Mobile Phones, Smart Phones, PDAs: [6] Application Specs:

- max Pout = 2 or 3 watts
- low quiescent supply current
- low supply voltage range
- low-power shutdown
- differential audio inputs → to reject RF artifacts, such as those from TDMA & GSM cell phones
- minimal external components

➔ Now explain the class D solution

CONCLUSION:

- Advantages of Class D:
  - high efficiency
  - small size
  - low heat dissipation
- well-designed class D can achieve THD & SNR comparable to that of most consumer linear amps [1]
- I have found that the past decade has been a time of great evolution in class D
  - some of the class D solutions being praised in 2006, but 3 years later the company has been restructured and have a completely different product line now.

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