



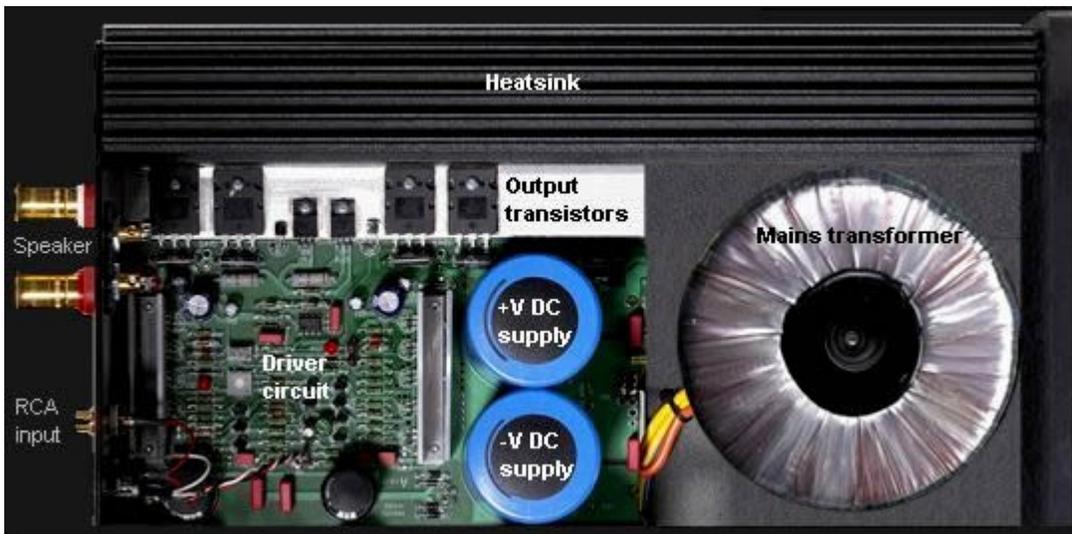
## **Solid State Amplifiers**

Solid State amplifiers have superior technical specifications compared to valve amplifiers. But when solid state amplifiers were first introduced it was noticed that they sounded flat and lifeless in comparison to well made valve amplifiers. Also a solid state amplifier had to be twice as powerful as a valve amplifier to sound as loud - Why? Unfortunately little to no research was done.

Amplifiers and speakers have traditionally been marketed independently of each other. There has been little interest in how amplifiers and speakers interact. Valve technology was tediously assembled with manual labour whereas Solid state technology is mass produced at a fraction of the cost. Economical rationalism and modern marketing easily pushed aside the older valve technology. Page 8 has a detailed explanation of voltage-drive and current-drive, which enables us understand why Valve and Solid state amplifiers sound so different.

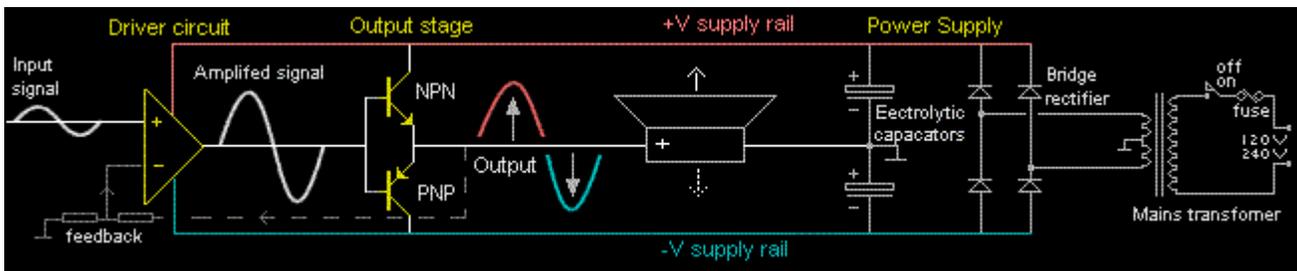
### **Basic technical principles**

Many professional audio providers and audiophiles have mis-conceptions about how a solid state (transistor) amplifier functions. Mis-conceptions occur because basic principles are not understood. Please be attentive when reading these pages as each step adds to the next and becomes complex very quickly. If a step becomes difficult to understand, take time to review the previous steps.



A solid state amplifier consists of 3 sections.

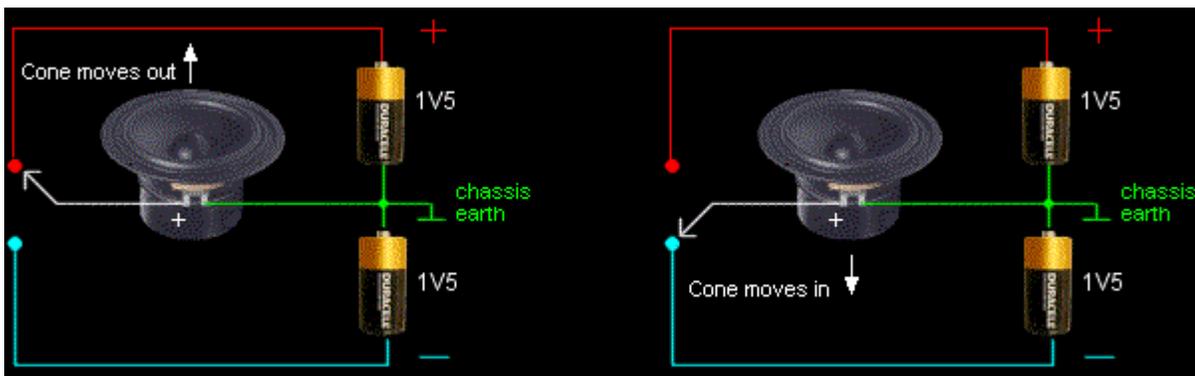
- 1 The input driver circuit amplifies the small input signal to a larger size, approx x20 to x50.
- 2 The large output transistors add current to the amplified signal to be sent to the speaker.
- 3 The power supply converts 110V / 220V AC mains to 2 DC supply Voltages that power the amplifier.



The principles of each section will be described, beginning with the output stage. The output stage consists of 2 or more large output transistors bolted to a heat sink (NPN and PNP).

A solid state amplifier has 2 (DC) power supplies (+V and -V). The 2 power supplies are connected in series. The middle is connected to the chassis. One terminal of the speaker is connected to the middle chassis and other speaker terminal is switched between the 2 supplies. We shall begin with batteries as the power supply.

As the speaker is connected from chassis across the +V supply the cone moves out.  
 As the speaker is connected from chassis across the -V supply the cone moves in.

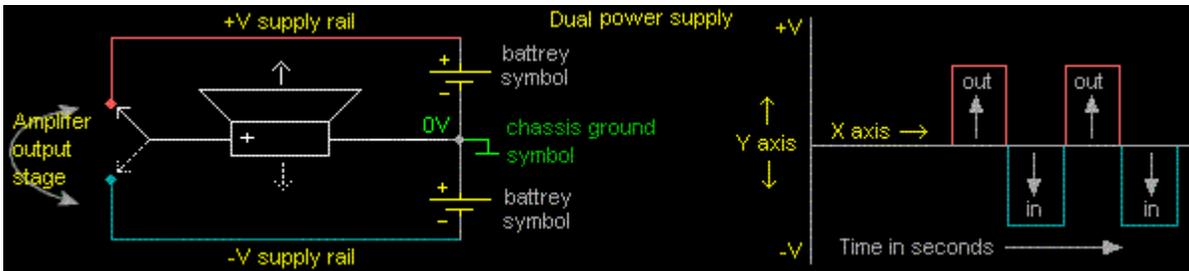


The speaker is connected across one power supply at a time. Therefore the maximum Voltage across the speaker can be no greater than 1V5 at any one point in time.

**Note** – In electronic circuits we try not to use decimal points (1.5V) because grammatical full stops and cockroach droppings .... are the same size .... 1.5V is written as 1V5

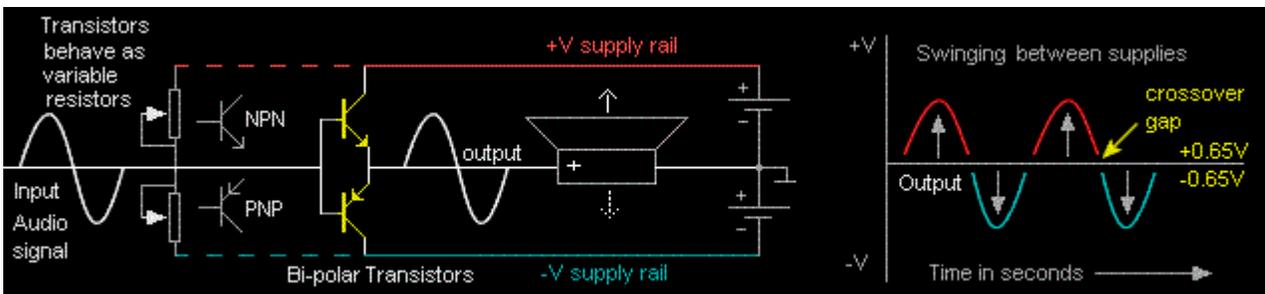
**Speaker polarity** One of the terminals on the speaker has a polarity + identification mark. The + symbol identifies which terminal has to be connected to a +Voltage to cause the cone to move out. In stereo systems or paralleling speakers it is essential that all speakers move together (in and out) in the same direction. When only 1 speaker is used as mono it makes no difference which way around the speaker terminals are connected.

We must redraw the same circuit and remember the correct symbols and names for each part of the circuit. Circuits drawings do not use colour. The colour in the pics is for temporary assistance only.



- Dual power supply
- Battery long line is + short line is -
- Supply rails +V and -V from the dual power supply.
- Chassis and ground symbol. Is 0 V
- Graph Y axis is the dual power supply Voltage, X axis is time.
- Graph shows the + – Voltage switched to the speaker over time.

The speaker is not directly switched on and off between the 2 power supplies, but gradually connected to each power supply in turn through solid state devices that behave as variable resistors, in a way that follows the audio signal. These solid state devices behave as Transfer resistors therefore are named Transistors.



The output transistors swing the speaker between the +V supply then to the -V supply, in-sympathy with the music. The output transistors provide current from the + – V supply rails to drive the speaker.

Output transistors do not increase the size of the music signal. The output transistors behave as variable resistors conducting current from each power supply in turn and get hot, very hot. They are bolted to a large piece of aluminum with fins (heat-sink). Heat is the enemy of transistors. The maximum Power output of an amplifier is dependent on the Current and Voltage from the power supplies.

NPN and PNP transistors are polarity complements of each other. PNP is positive-negative-positive and vice-versa. The input signal has to reach 0.65V (650mV) before each transistor will start to conduct. This means there is a total gap of 1V3. This gap is named ‘crossover distortion’ and was a

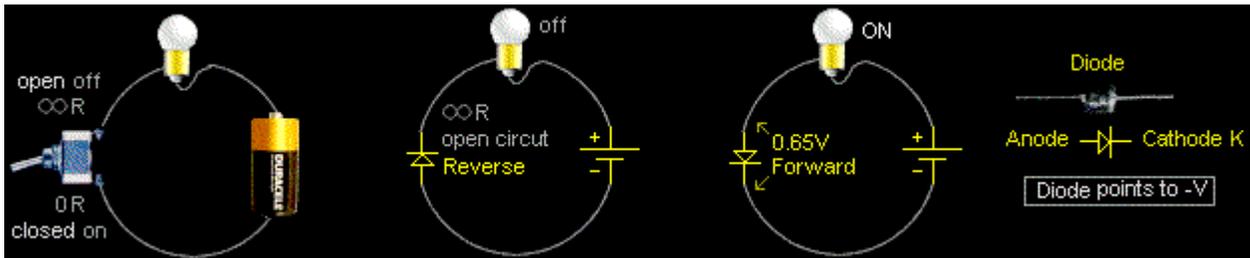
major problem of solid state amplifiers when they were first introduced. It took many years before this problem was correctly solved.

## Transistors and Diodes

Electromagnetic energy is a mystery. Descriptions and drawings of electricity flowing in a direction are not used in this text. Electricity functions at the speed of light, where time and direction do not exist. A red arrow is used in a circuit to represent an electric current is functioning, regardless of polarity.

The cathode K of a diode and the emitter of a transistor have an arrow head symbol. The arrow head points (forward) to the -V of a battery or Voltage supply in the circuit for the diode or transistor to function.

A circuit is a circle, so it does not matter where in the circle each item is placed. It is easier to read a circuit if everyone draws it the same way and places the power supply vertically on the right, with +V at top and -V at bottom. Switch, diode, transistor or control circuit at left.



The left side of the above pic shows a battery, light bulb and switch. When the switch is open, the resistance across the switch is infinite  $\infty R$ . When switch is closed the resistance across the switch is zero  $0R$ .

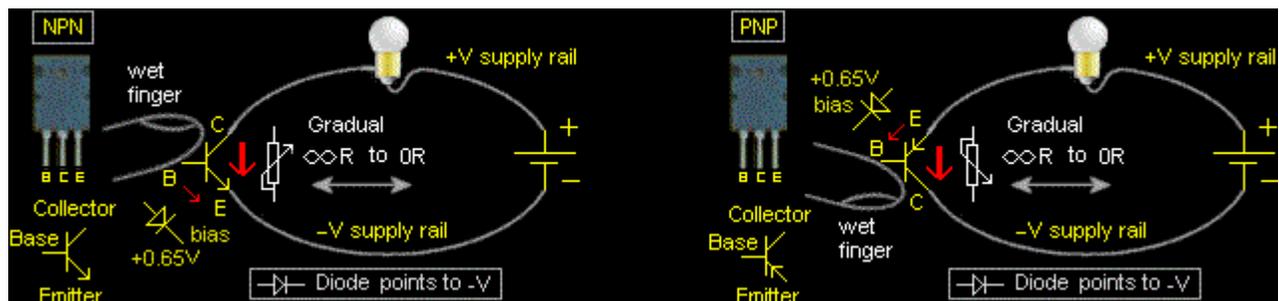
$\infty$  Resistance can be described as an open circuit. (Infinite resistance)

$0$  Resistance can be described as a short circuit.

A diode acts as a open circuit if the arrow points toward the +V in the circle (Reverse).

A diode acts as a short circuit if the arrow points toward the -V in the circle (Forward).

In the forward direction a diode requires 0.65V (650mV) across it, to activate. Less than 0.65V (650mV) the diode will be an open circuit. An activated diode it will remain locked at 0.65V across it, regardless of the amount of current the diode conducts. Therefore a diode will get slightly hot as the current increases.



The diode is replaced with an NPN transistor on the left and a PNP transistor on the right. NPN and PNP transistors are polarity complements of each other, which enables them to be used from a +V or -V supply. A transistor can be made to behave as a resistor that changes its value between an open circuit

( $\infty$  Resistance) to almost a short circuit (0 R), therefore controlling the brightness of the light bulb.

By placing a finger on the Base causes a very small amount of current between the Base and Emitter, which will then enable a larger amount of current to flow between the Collector and Emitter, enabling the brightness of the light bulb to be adjusted. Notice that the red arrows are symbolising that the transistors are actively functioning. Whereas the transistor emitter arrow head points to the -V in the circuit to represent the correct polarity for it to function. A small amount of current between the Base and Emitter, enables a larger amount of current between the Collector and Emitter.

HFE (Hybrid Forward Emitter) is a technical term to describe the current gain of a transistor. The HFE (gain) of a transistor increases with temperature but decreases with current, which makes it difficult to control. Small signal transistors can have a HFE (gain) of 100 to 200, whereas large power transistors will have a low HFE of approx 20 to 100.

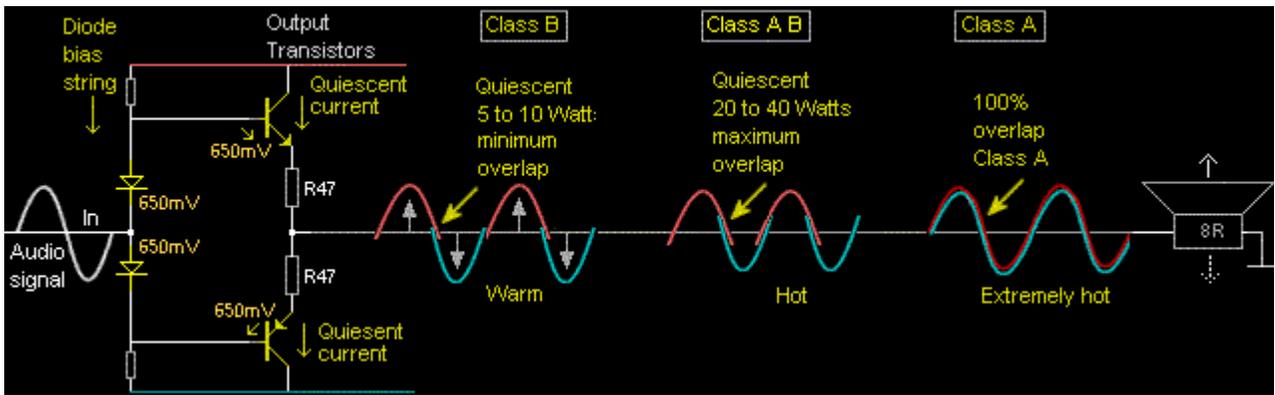
Bias 0.65V (650mV) The transistor junction between Base and Emitter acts like a diode. Nothing will happen until 0.65V (650mV) is reached between the Base and Emitter, only then is the transistor activated. When the Bias of 0.65V is reached the transistor is activated and the Base\_Emitter will always be locked together at 0.65V (650mV). Therefore the Emitter of the transistor in an amplifier will always follow the input signal on the Base (less the 0.65V Bias) with added current from the Collector. Review the previous steps.

Bias voltages of diodes, transistors and amplifier designs are generalised in this text. Detailed engineering reference for electronic components and amplifier design is on [www.sound.westhost.com](http://www.sound.westhost.com)  
[wikipedia.org / Transistor](http://wikipedia.org/Transistor)  
[wikipedia.org / Bipolar Transistor](http://wikipedia.org/Bipolar Transistor)

### **Output stage class B class AB**

Crossover Each transistor controls each half of the sine wave. This is described as Class B. The crossover gap is 650mV and 650mV across each Base\_Emitter. Eliminating the crossover gap between each half of the audio signal is the most critical part of solid state amplifier design. The distortion created by the crossover gap, generates an annoying 1/3 harmonic sound imposed in the music, similar to the sound of tearing paper. A reduced gap of 1/000V (1mV) is still audible. The gap must be closed, plus the current through each transistors must slightly overlap the other to insure there can be no crossover gap.

A bias string using 2 diodes is placed before the output transistors so 650mV + 650mV is already causing each of the output transistors to be activated with quiescent current flowing between the collector and emitter. This quiescent bias current closes the 650mV gap of each transistor. Some amplifiers use 2 diodes in a string, but the majority have a more complex circuit that allows the quiescent current (overlap) to be adjusted.

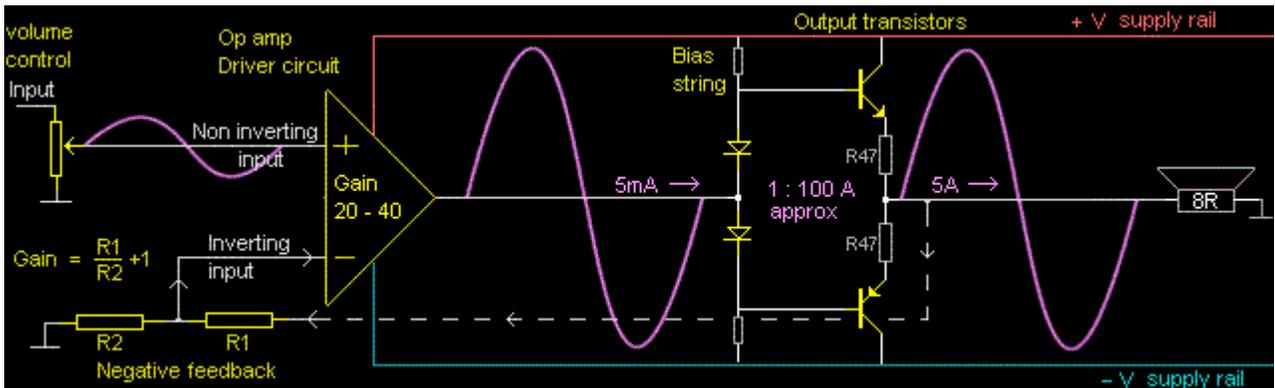


Providing the gap is closed with a small quiescent current through the output transistors to provide a small overlap, the crossover distortion is eliminated. Quiescent current generates waste heat in the output transistors, approx 3 to 5 Watts per transistor. This waste heat is also described as the quiescent temperature. A Hi-powered amplifier with many output transistors will create a lot of quiescent heat and need to be fan cooled. There is a R47 (0.47R ) resistor in series with each Emitter. Emitter resistors help stabilise the quiescent current. They are physically large, and have a very low resistance 1/2R approx.

**Class A** Some audiophile amplifier designs increase the overlap to 100% (Class A) by increasing the quiescent current to the maximum amount. This causes the output transistors to get very very hot, often well beyond the safety temperature parameters. Because Class A amplifiers have an excessive quiescent temperature they are mostly designed for low power applications approx 20 Watts. Class A amplifiers are marketed as having a magical sound. No one has proven in a university controlled test (with double blind A B comparison) that magical sound exists. Audiophile fanatics who believe in magical sound are similar to religious fundamentalists who can become aggressive, if challenged to provide proof.

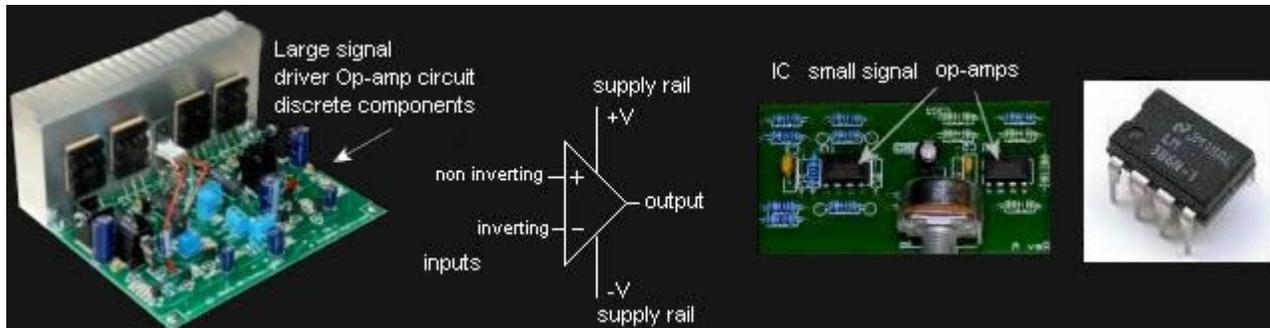
## Driver Circuits and Op-amps

A small input line level signal 100mV to 1V is amplified by the op-amp driver circuit. A correctly designed driver circuit is capable of amplifying the signal (perfectly) to the exact height of the + – V rail supply. Op-amp driver circuits are not capable of driving a speaker directly because it can not provide sufficient current (Amperes) 10mA approx. Output transistors do not increase the size of the signal but simply add current (Amperes) 100:1 approx to drive the speaker. The bias string eliminates the crossover gap as previously described.



The driver circuit is described as a high voltage op-amp (Operational amplifier). Op-amps are drawn as a triangle with 2 inputs. The inputs are described as Non-inverting with a + symbol and Inverting with a – symbol. The input signal goes to the Non-inverting + input. The output of the amplifier is feed back to the – Inverting input, described as Negative feedback.

Gain The Voltage gain of an op-amp can be adjusted by resistors R1 and R2. Op-amp gain =  $R1/R2 + 1$ . Power amplifier gains will typically be between 20:1 and 40:1. There is no international standard for the set gain of amplifiers. In domestic system amplifiers the loudness level is simply adjusted with the Volume control. In large sound re-enforcement installations many amplifiers are placed in racks. Not to have an international agreed ‘gain standard’ for professional amplifiers is irresponsible and a nightmare to manage.



Large driver op-amps for power amplifiers are mostly made from discrete (separate) components and designed specifically for each application. IC (integrated circuit) small signal op-amps are used in almost all audio pre-amplifiers and signal processing equipment. Small signal IC op-amps are designed to be powered from + – 15V supply rails, or less. IC op-amps are available from many suppliers. They vary in cost from approx 50 cents to \$50 each depending on application. Hi-precision ultra low noise op-amps are understandably more expensive and are designed for instrumentation measurement. The average low cost IC op-amp is used for general domestic audio equipment. The basic operational principle and internal circuit of all op-amps is similar and their performance attains to what is described as ‘the perfect amplifier’.

The perfect operational amplifier would aspire to have infinite attributes. Infinity does not exist (universe excepted) but the ability to approach infinity does exist. Asymptotic is a technical term that means approaching infinity.

1. Infinite power
2. Infinite gain
3. Infinite bandwidth
4. Infinite high input impedance
5. Infinite low output impedance
6. Infinite low distortion
7. Infinite low noise

(1) Infinite power is unrealistic but hi-powered amplifiers are available with more power than is required, therefore power is not a limitation. In a past era when only valve amps existed power was a limitation.

(2) Infinite gain is not necessary for any known audio application. The maximum gain normally required may be 100. Cheap op-amps are capable of a gain of 1 million. Special op-amps can be much higher.

(3) Infinite bandwidth is also un-necessary as the audio spectrum is from 20Hz to 20KHz. The

cheapest op-amp is capable of 1MHz. Special op-amps can go higher than 100MHz.

(4) Infinite high input impedance. In normal audio applications an input impedance of 1M Ohm is sometimes required. Special op-amps can easily achieve 100M Ohm and higher.

(5) Infinite low output impedance is achievable but rarely required. It is possible to make an op-amp circuit achieve negative output impedance.

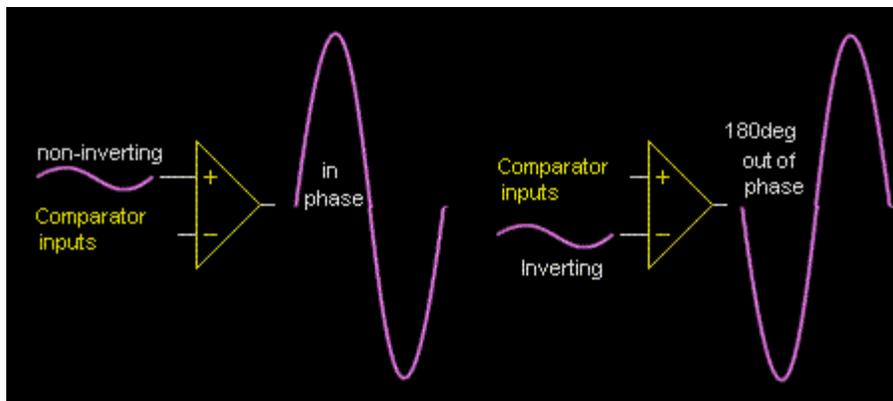
(6) Infinite low distortion. Cheap op-amps achieve distortion figures magnitudes below what is audibly detectable, and special op-amps achieve low distortion figures at the threshold of what is measurable.

(7) Infinite low noise. The lowest noise is thermionic electron noise of components limited by the laws of physics. There are precision low noise op-amps available that approach this threshold.

There is no such thing as an audiophile product constructed with 'magical' components (pure gold wire etc) regardless of cost, brand name and marketing hype that can be proven to be auditory different (with a double blind A B comparison) than the same circuit correctly constructed with equivalent general purpose components from the nearest electronic component supplier. Audiophiles who believe in magical brand names and components not only trivialise electronic technology but show appalling dis-respect for the engineers and scientists who dedicate their lives to the research and development of component and IC engineering.

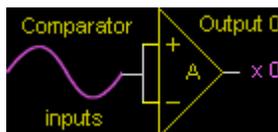
### The op-amp miracle

Modern electronics and ICs are possible the closest thing we can describe as miraculous. The following description of op-amps is an overview and appreciation of how with a few simple components a beautifully complex circuit can function.



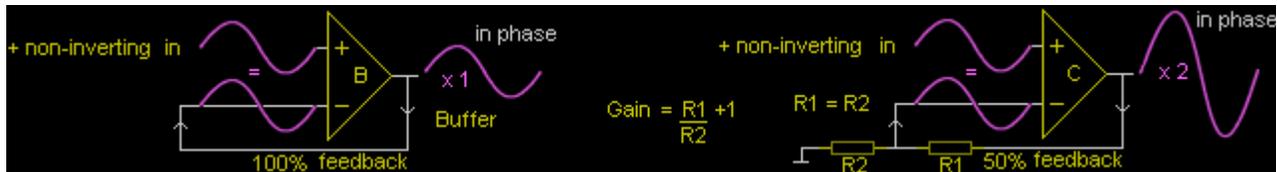
An op-amp has 2 inputs described as a comparator. One non-inverting input (+ symbol) and one inverting input (- symbol). An op-amp has almost infinite gain by comparison to what is required.

When a signal is put into the + non-inverting input the output will be in phase with the input and amplified toward infinity. When a signal is put into the - inverting input the output will be 180deg out of phase with the input and amplified toward infinity.

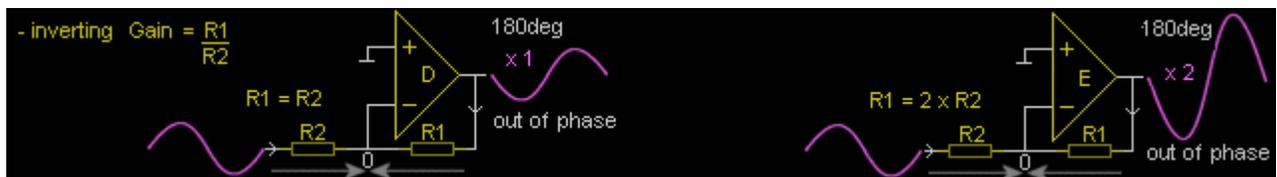


The gain of the op-amp responds to the difference between the 2 inputs which is described as a comparator. In the right pic Op-amp A has both inputs joined together. The joined non-inverting and inverting inputs cancel each other, therefore no output.

In op-amp B the output is directly fed back (100% feedback) to the -inverting input which results in the same signal on both comparator inputs. This is like trying to describe which comes first (chicken or egg) at the speed of light. At the moment a signal appears at the input, the output will attempt to have infinite gain. But the moment the signal at the inverting input is at the same size as the incoming signal on the + non-inverting input, the op-amp ceases to have any further gain.



Op-amp C has two resistors in series to ground. The junction of the resistors is a voltage divider, which is connected to the -inverting input. Because both resistors are the same value, the voltage at the junction will be 1/2. Therefore the amplified output signal will have to be x2 before the signal at both inputs will be the same, causing the op-amp to have no further gain.



Op-amps D and E have the input signal fed to the -inverting input. The output will be 180deg out of phase with the input. The output is also fed back (feedback) to the -inverting input via R1. The moment the fed back amplified output signal is the same size as the incoming input signal via R2, they will both cancel each other out, causing both inputs to be the same, hereby causing the gain of the op-amp to cease amplifying the signal any further. The gain is adjusted by simply changing the values of R1 and R2.

These basic descriptions of how an op-amp is managed to give the signal gain required are analogous and all that is required to make use of them in circuit applications. But this is not the descriptions given in academic text. In academic text extensive mathematical formula are used to give a more detailed account. In reality the 2 inputs do not reach absolute 0 difference between them when the required gain is obtained. There has to be a difference between the inputs for the op-amp to function. The difference between the inputs is asymptotic to zero, dependant on the open loop gain.

A simple principle taught to electronic students about op-amps. “In a circuit an op-amp will attempt to make both inputs have the same Voltage. If not, the output will take the sign of whichever input is most +ve”.

### Inside the op-amp

An op-amp can be constructed with 3 small 10 cent signal transistors and three 2k2 resistors. However most IC op-amps have 30 to 100 internal transistors. The below pic is a simple circuit that will actually work and is an ideal beginning for electronic students to construct and experiment with to obtain an understanding of the basic principles. The below circuit will work on any Voltage between 5 to 30 Volts.

The first 2 NPN transistors with their emitters joined together (long tail pair) is described as a differential input. The 3rd PNP transistor is described as a Class A driver, which means that the emitter is connected to the +V rail supply and the collector is the output. The output Voltage is the result of a difference between the two inputs. A very small difference voltage between the two inputs of micro Volts will cause the output to flip to the +V rail or to -V rail. An op-amp used to switch

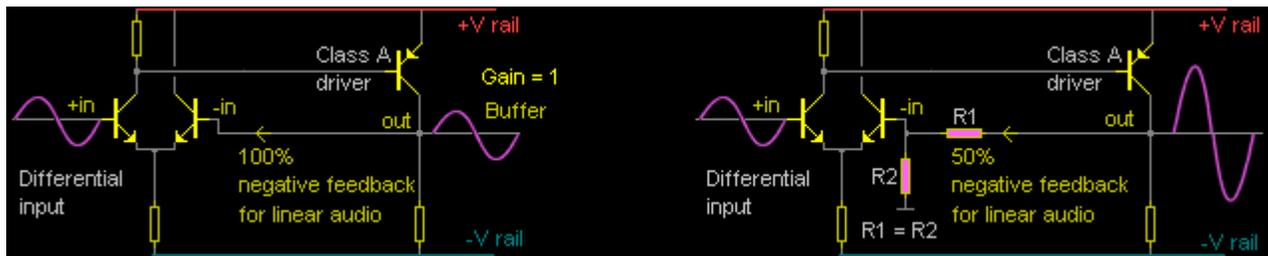
between the rails is described as a comparator.

The same circuit can be constructed with complement NPN for PNP transistors and +V -V rails reversed.



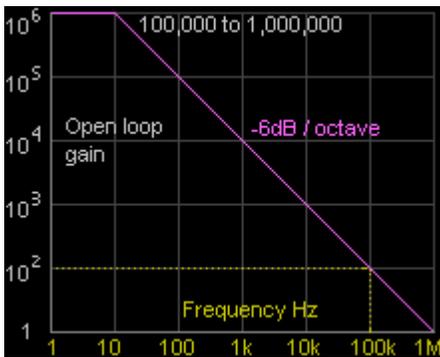
**Comparator** A small +V on the +non-inverting input (compared to the -inverting input) will cause the first transistor to conduct. The Voltage at the collector will drop toward the -V rail. This will turn on the NPN Class A driver causing the collector (output) to flip to the +V rail. Vice-versa for a small -V on the +non-inverting input.

A small +V at the -inverting input (compared to the + non-inverting input) will cause the emitter second transistor of the differential pair to follow its base within 650mV. Because both emitters are joined this will cause the emitter of the first transistor to decrease below the 650mV threshold which will turn off the first +non-inverting transistor (open circuit) causing the voltage at the collector to go directly to the +V rail. This will turn off the Class A driver (open circuit) as a switch causing the collector (output) to drop to the -V rail. Vice-versa for a small -V on the -inverting input.



**Linear audio** The output can be connected directly to the -inverting input, or through R1 R2 to provide gain. A small +V on the +non-inverting input (compared to the -inverting input) will cause the first transistor to conduct. The Voltage at the collector will drop toward the -V rail. This will turn on the NPN Class A driver causing the collector (output) to attempt to flip to the +V rail. But the output is feed back to the base of the -inverting input transistor. The emitters of the differential pair are joined together. Therefore what happens on the emitter of the -inverting input also happens on the emitter of the +non-inverting input. Both base and emitters are locked together within 650mV.

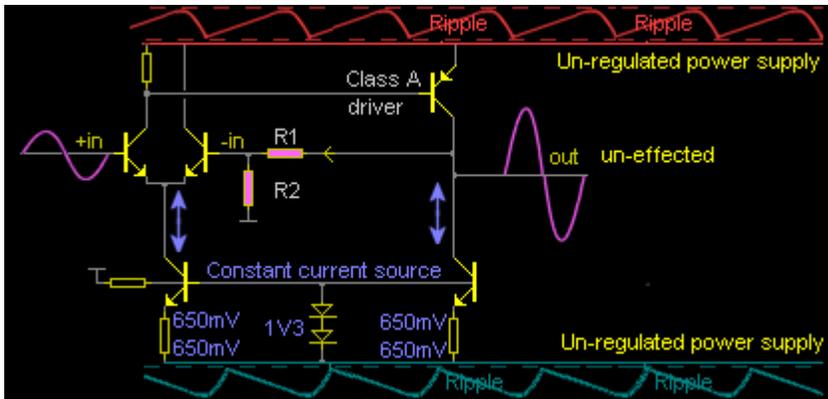
The emitter of the +non-inverting input is influenced from the output attempting to cause the 650mV difference between base and emitter to be reduced or negated therefore counteracting the effect of the incoming signal. This counteracting effect of the feedback causes the output to follow the input directly. Only by reducing the amount of feedback through R1 R2 can the output have gain.



Open loop gain is the maximum gain of the op-amp without negative feedback and is specified at DC. From approx 10Hz the open loop gain will decrease at 6dB / octave as a result of slew. At 10kHz most op-amps in audio circuits will have a gain of approx 1,000 and at 1MHz will have unity gain. The right pic shows an average \$1 op-amp is capable of a flat response to 100kHz with a gain of 100. The maximum signal gain required for audio application with negative feedback is often less than 100.

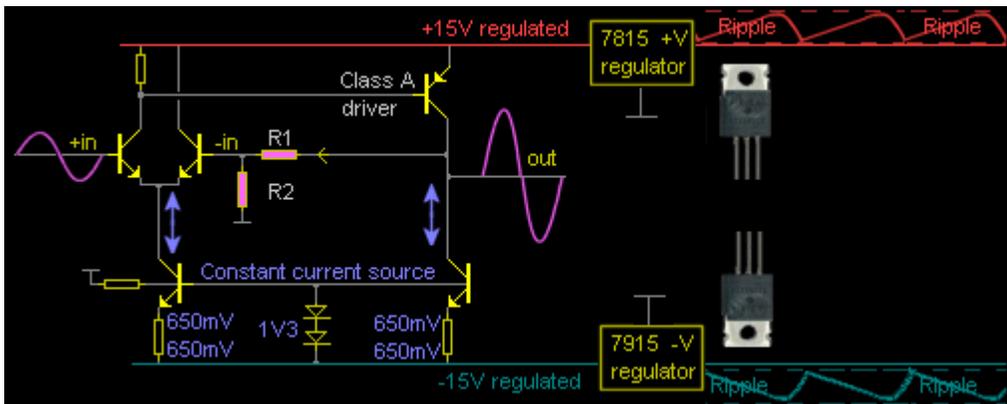
Common mode rejection Absolutely no variations on the supply rails can effect the audio signal. The op-amp output can only respond to the signal between the inputs and not to any variation on the supply rails. Nothing else whatsoever can effect what happens at the output.

Current source Op-amp performance is greatly enhanced by a constant current source from the -V rail to the differential inputs and Class A driver. The constant current source insures that the current through the differential inputs and the Class A driver remains absolutely constant regardless of any changes in the supply rails. This enables the op-amp to perform perfectly from any voltage between 5V to 30V. The constant current source absolutely insures that supply rail noise and ripple can not in any way effect the audio signal.



A simple constant current source consists of transistor with its base locked by small fixed reference Voltage. The 1V3 (650mV + 650mV) of two diodes acts as a fixed reference. There is 650mV between the Base and emitter of a transistor and therefor another 650mV must be across the resistor between the emitter and the -V rail. The fixed 650mV across the resistor insures a fixed reference current through the resistor which is also the fixed current through the transistor. Some constant current source circuits can be more complex.

[wikipedia.org / Op-amp](http://wikipedia.org / Op-amp)



7815 Voltage regulators All quality audio pre-amplifiers and signal processors have power supplies with additional Voltage regulators that provide a smooth ripple free + – 15V rail supply to all op-amps. Special low Voltage op-amps in Ipods etc are not discussed in this text. The specified + – 15V rails allows the correct line level with sufficient dynamic range without the signal being clipped into the rails.

Some audiophiles are being conned by dishonest audio technicians claiming that auditory fidelity in pre-amplifiers and CD players can be improved if the 7815/7915 Voltage regulators are replaced by a special audiophile brand at high cost \$x\$. The only audio requirement for Ultra low noise voltage regulators and op-amps is in phantom power supplies for low noise microphones pre-amplifier circuits, and phono pre-amps for transformer-less moving coil magnetic cartridges.

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## Amp Parameters

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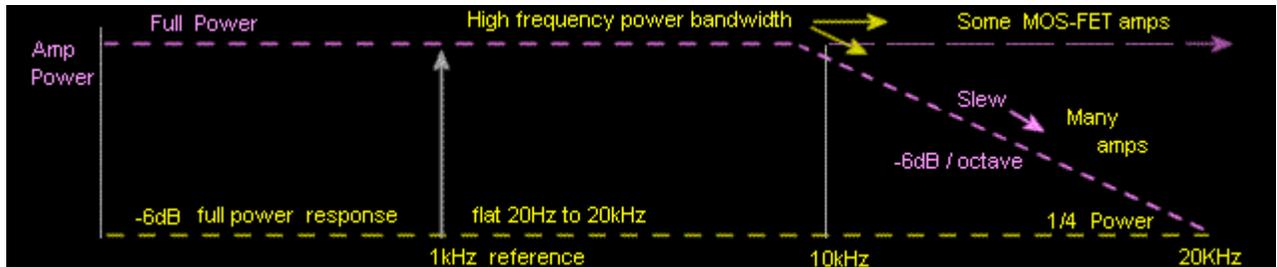
Almost all modern amplifiers today including the cheapest domestic ghetto blasters have distortion figures that are magnitudes below what can be audibly discernible. Amplifier distortion figures are now at the threshold of what can be scientifically measured. An amplifier with a THD of 0.001% is of no discernible difference to one of 0.0001%. Amplifier manufactures today have virtually no influence on these figures. These extremely low distortion figures are now governed by the engineering physics of the components that are common to all amplifier manufactures. Quoting comparative distortion figures with claims of an auditory difference has now become a face of marketing deception.

If there is an actual audible difference between 2 amplifiers (separate from power) then either one or both amplifiers has a design mistake, or one or both amplifiers is faulty. Any modern amplifier designed and made correctly that do not have a design fault or technical fault, cannot be audibly different from any other modern amplifier that is designed and made correctly that does not have a design fault or technical fault, regardless of brand name, marketing hype or cost. Almost all technical specifications of modern amplifiers are asymptotic toward infinity, therefore auditory indiscernible and meaningless by comparison (except for power).

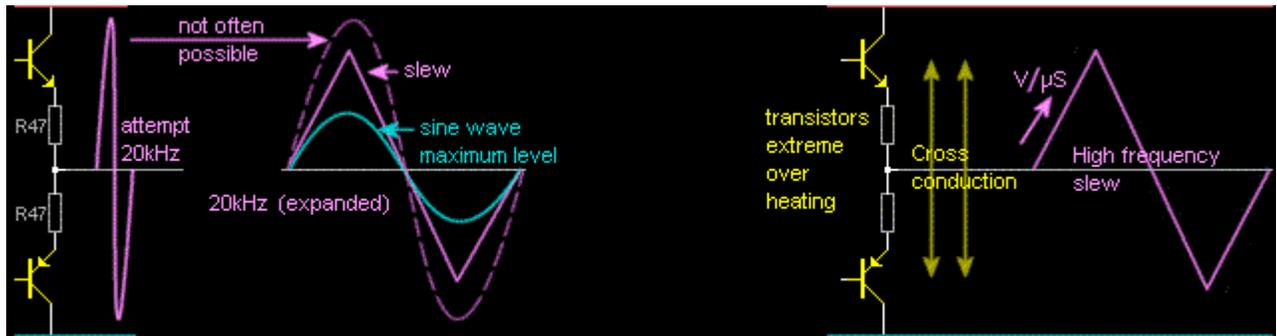
Slew

Frequency response and distortion figures of amplifiers are normally measured at 1 Watt. The maximum power of an amplifier is normally measured at 1kHz. At 20kHz most amplifiers can not

deliver more than 1/4 of the power quoted at 1kHz as in the graph below. The limitation of power restriction of amplifiers above 10kHz is described as Slew. It is not possible for any electronic circuit or individual component, transistor, FET or mechanical switch to be able to turn on and off instantly at the speed of light. Slew is the highest speed an electronic circuit, transistor or FET can change from on or off.



Volts / micro second is Slew rate. With an amplifier of  $\pm 60V$  supply rail, the leading edge of a 20kHz sine wave would have to change from 0V to 60V in  $1/80,000$  of a second ( $12.5\mu S$ ). Therefore it has to have a slew rate of at least  $0.2V / \mu S$  to reach 60 Volts in time. But if the amplifier output can only change at  $0.1V / \mu S$  the maximum voltage it could reach is 30V which is half way and therefore 1/4 power.



Many amplifiers that use power MOS-FETs have very high slew rates and are easily capable of delivering 20kHz at full power. But many amplifiers that use output transistors are restricted in being able to deliver full power at 20kHz. Slew can be intentionally applied in a driver circuit of an amplifier (dominant pole capacitor) to insure high frequency stability and freedom from internally generated parasitic oscillations.

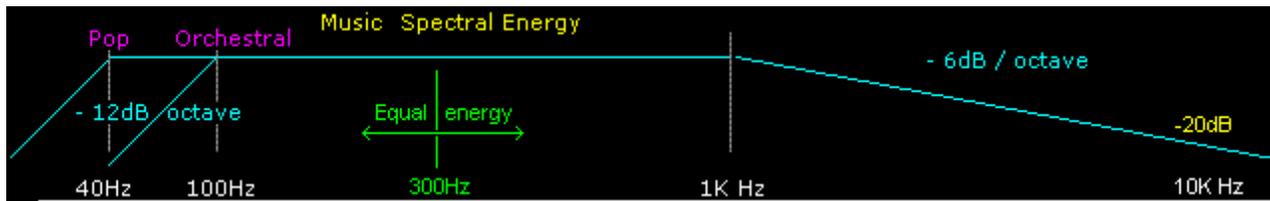
But there is another unique problem of slew that is caused by the internal limitation of one or both output transistors not being able to turn off as quickly as they can turn on. This problem causes a large quiescent current cross conduction through the output transistors that generates extreme heat. An amplifier with slew limited by output transistor cross conduction can be destroyed by high frequency oscillation caused by a reactive capacitive speaker load or Rf (radio frequency) getting to the input from an external source.

### Music power bandwidth

Almost all amplifiers including cheap domestic amplifiers have a frequency capability from 0Hz to 100kHz at low power. The gain of an amplifier should be rolled off below 20Hz to stop un-wanted sub-sonic frequencies getting to the speaker. In a previous era (before solid state technology) when only valve amplifiers existed, the BBC in London did extensive research about how high in frequency response an amplifier needs to be to give faithful music reproduction. They found no evidence that an amplifier which was designed to continue amplifying frequencies above 20kHz was musically any different to an amplifier deliberately designed to restrict frequencies above 20kHz. Amplifiers which

were capable of amplifying supersonic frequencies above 20Khz were not only un-stable but were also susceptible to Rf (radio frequency) interference. Hence the statement “The larger the window the more sh\*t gets in”.

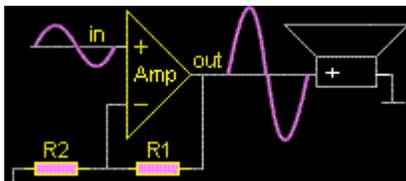
In this era of magnetic recording, the limitation of high frequency response was governed by the tape speed and the high frequency bias oscillator. The high frequency bias oscillator acted as a carrier for the audio signal to be magnetised on to the recording tape. The highest audio frequency that could be transferred on to a recording tape was 1/4 of the bias oscillator frequency. Bias oscillators in the majority of professional recording machines in many of the worlds top recording studios were set at 60kHz which restricted the highest audio frequency that could be recorded at 15kHz. Special mastering recording machines had bias frequencies of 80kHz to 100kHz which enabled 20kHz to be recorded on to a tape. But as a general rule, the higher the bias frequency the less audio energy could be recorded and therefore the higher the signal to noise ratio. Only very expensive tape recording machines were not restricted in this way.



The spectral energy of music is flat to approx 1kHz. The highest notes on any musical instrument is approx 2kHz. Above 2kHz are the harmonics which decrease in energy at approx -6dB per octave. At 10kHz the harmonic energy of music is approx -20dB which is 1/100 of the power compared to the middle sector of the voice range. Therefore very little energy exists above 2kHz at these extreme high frequencies.

By combining all the previous information knowing that no audio information is traditionally recorded above 20kHz and the spectral energy of music decreases at -6dB / octave above 1kHz and the slew limitation of many amplifiers only enable a flat power bandwidth to 1/4 full power demonstrates that slew restriction above 10kHz is almost never a factor in limiting an amplifiers ability to reproduce music with normal use. However in a high power 4 way active system the amplifier driving a tweeter from 6kHz up should not have slew restriction.

### Amplifier in-stability

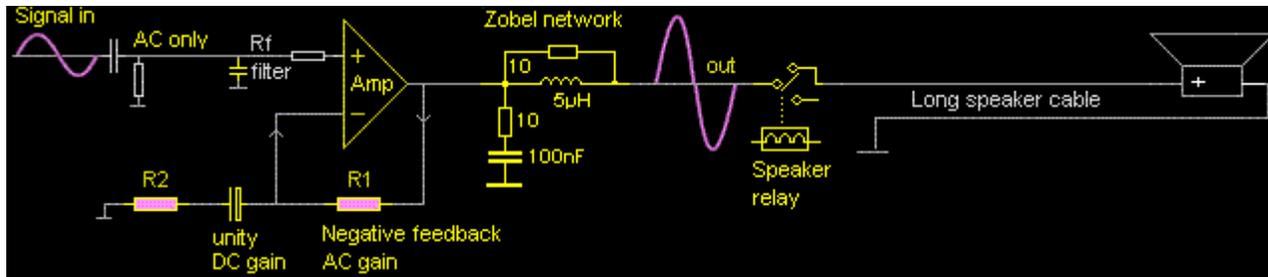


Power amplifiers are simply large op-amps with the capacity to drive a speaker, simplified in the right pic. The lower pic shows the external management detail.

(1) A capacitor in series with the input signal only allows AC to pass and blocks any DC from the getting to the amplifier’s input from an external source. A small capacitor to ground shorts out any Rf (radio frequencies) picked up by the input lead.

(2)  $Gain = R1/R2 + 1$  There is no agreed set gain for amplifiers. Gain will typically be between 20:1 to 40:1. A capacitor must be series with R1 R2 to stop the amplifier from being able to have DC gain,

to help avoid DC offset at the output.

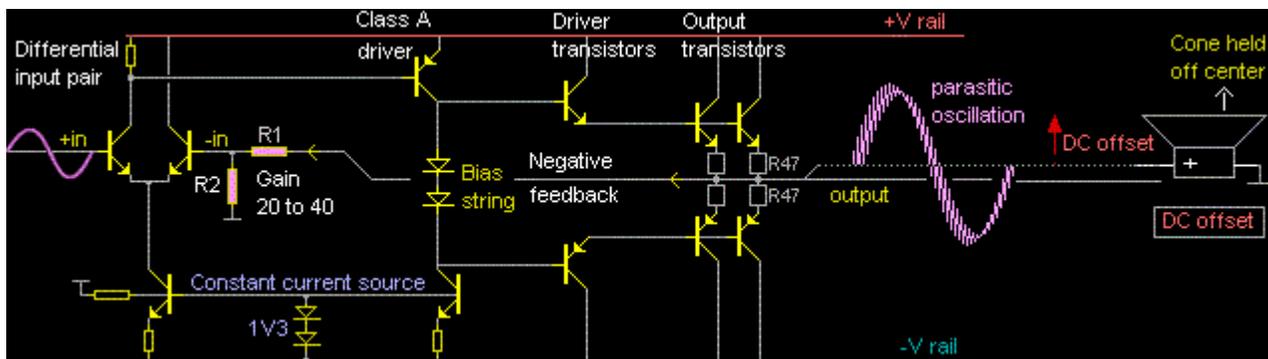


(3) A Zobel network of various complexity is placed at the output of every solid state amplifier to minimise the possibility of parasitic oscillation, and damp the capacitor reactance of the speaker cable.

(4) A speaker relay with a delayed turn on is used in some amplifiers to avoid turn on and turn off thump into the speaker. Some relay circuits also sense DC offset. Most amplifier failures are the result of a short circuit output transistor. This causes the speaker to be connected across one of the V rails hereby destroying the speaker. The relay circuit should also detect a short circuit output transistor and disconnect the speaker.

### Parasitic oscillation

The combined small phase shifts within the circuit and internal parameters of the transistors is compounded and fed back through the negative feedback, un-stabilising the amplifier causing 50kHz to 1MHz Parasitic oscillations. Parasitic oscillations are almost impossible to control but are minimised with stabilisation capacitors across the Class A driver and driver transistors, reducing the slew of the amplifier and further reducing the open loop gain above 20kHz, but often with only limited success.



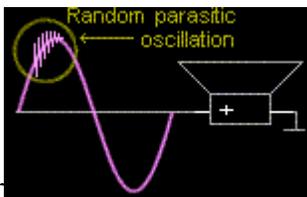
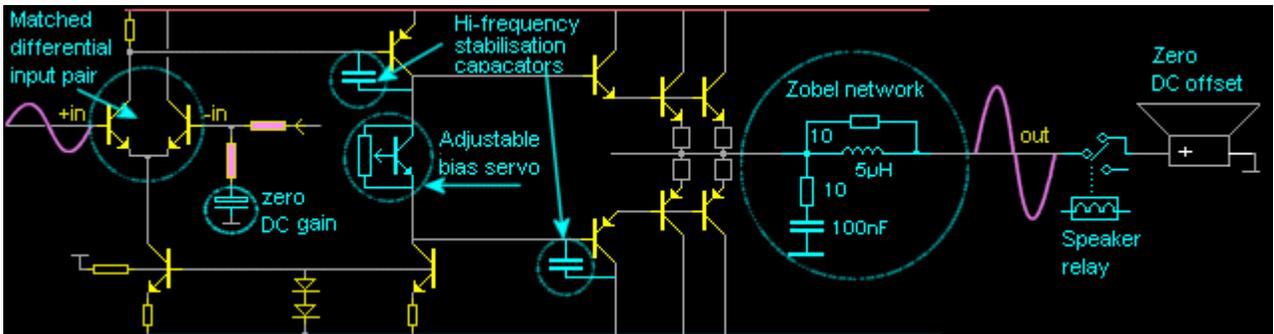
The higher the open loop gain the more responsive, but the more potentially un-stable an amplifier becomes. A stable amplifier may have a low open loop gain of 20,000 with a flat response to 20kHz.

But its response above 50kHz may be poor in comparison to an audiophile amplifier with an open loop gain of 100,000 that has a flat response to 100kHz. There are many un-substantiated claims in audiophile marketing that an amplifier with a flat response to 100kHz will sound better than another amplifier with a flat response to only 50kHz. Some audiophile amplifiers are so potentially unstable that the excessive capacitance of a magical speaker cable cause parasitic oscillations within the amp overheating and destroying output transistors.

Zobel network is at the output of all solid state amplifiers. Solid state amplifiers are only stable with a perfect Resistive load that has no capacitive reactance. Anything that causes a slight high frequency rotation of phase at the amplifier output is fed back through the negative feedback path to the -inverting input. The comparator can only function correctly if the phase is exactly correct across all frequencies. If not, the negative feedback becomes positive feedback causing the amplifier to

oscillate.

Zobel network design is based on transmission line theory. All wires including printed circuit tracks have specific lengths that randomly co-inside with fractional numbers of  $R_f$  (radio frequencies) wavelengths. This can cause slight rotational phase shifts typically between 50kHz to 1Mhz. The Zobel network represents a phase correcting load across the amplifier's output. It attempts to keep the combined small capacitive effects of the amplifiers printed circuit tracks, internal wiring to the speaker terminals, and the speaker cable appearing as a phase coherent resistive load. Only a very few professional amplifier designs have achieved excellent performance stability under most load conditions.

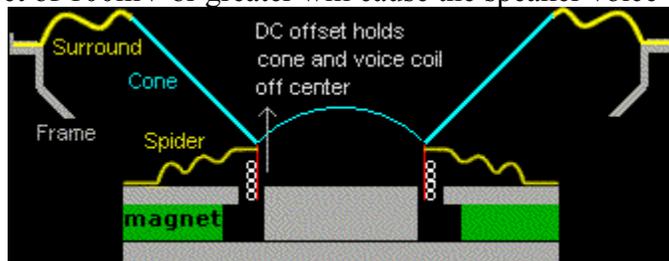


Similar to marriage and helicopters there is no such thing as a naturally stable solid state amplifier. All solid state amplifiers are parasitically unstable once a speaker cable is connected, its only a matter of degree. Cables of differing lengths and styles can cause bursts of parasitic oscillation to randomly appear on different parts of an audio waveform. Changing cable may stop or shift the parasitic to different positions. Parasitic sub-harmonics can be heard within the music.

There is un-intentional incorrect information on Zobel networks on many web sites. A Zobel network in an amplifier does not provide correction for impedance variation of a speaker. Some passive crossovers include a correcting network for impedance variation of a speaker inside the speaker box also described as a Zobel, but it is not the same Zobel as in the amp.

### DC offset

It is essential for the output of an amplifier to be exactly at 0V when there is no signal. The differential input pair have to be exactly equal for DC offset to be zero. DC offset can also be caused by un-equal quiescent current through the output transistors. As transistors gets hot, the HFE (gain) increases, and each transistor will behave slightly differently. This may cause a DC offset 1mV to 1V at the amplifier output. A + or - DC offset of 100mV or greater will cause the speaker voice coil to be



biased slightly off center, either in or out.

With the cone biased off center, the cone will move back and forth non-linearly. The low frequency

With

efficiency of the speaker will be reduced and distorted at higher power.

This problem is often mistaken for a faulty speaker. First, turn the amplifier off. Then turn the amplifier on (without music) and notice if the cone moves slightly, in or out. A small DC offset is often overlooked by most service technicians because it does not effect the power performance or other specifications of an amplifier. Also DC off set may appear only when the amp is hot. DC offset was a major problem in many early solid state amplifiers. The negative feedback R1 R2 must have a 100uF capacitor in series to ground to stop the amp from amplifying any DC that gets to the input from an external source. Also the output of the amp may have a speaker relay with a delay turn on to avoid turn on and off bangs and detect DC offset.

End of page 3

## Power and Heat

RMS (Root Mean Square). A sine wave is similar to an audio wave that constantly changes from 0 to maximum. This is similar to driving a vehicle and constantly stopping at red lights. Therefore the peak speed of the vehicle has to be greater than another vehicle that is not stoping at lights, traveling the same distance in the same time. RMS is the AC (alternating current) equivalent to DC (direct current) as resultant Power to generate the same amount of heat. The exact formula is based on the square root of 2.  $\sqrt{2} = 1.414$  or  $1/\sqrt{2} = 0.707$ .

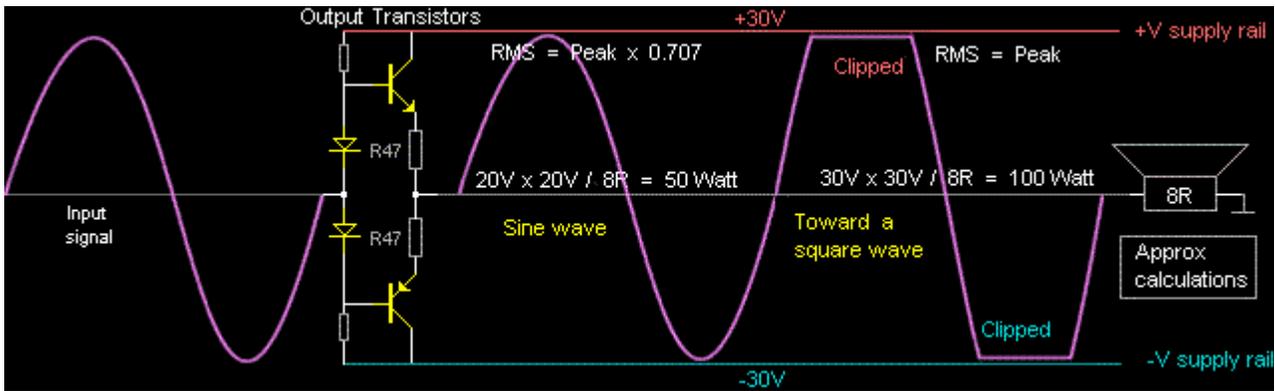
Resistance R is the same as  $\Omega$  Ohms (constant over frequency). Impedance is Resistance that changes with frequency. A speaker voice coil changes Resistance with frequency, 8 Ohm or 4 Ohm measured at 400 Hz.

The large output transistors are bolted to a heat sink. The output transistors provide current from the V supply rails to drive the speaker. Output transistors do not increase the size of the music signal. The maximum power available to drive the speaker is dependant on the Voltage and Current from the supply rails. The average hi-power professional amplifier will have + – 75V rails. Rarely do supply rails exceed + – 100V.



The modern marketing trend with many domestic amplifiers is to state Power output figurers that have no technical meaning. A small 3 Watt computer speaker with an internal amplifier can be marketed as 1,000 Watts. Only by testing, or looking at a circuit diagram, is it possible to know the power of an amplifier.

The supply rails in an average 50 Watt per channel domestic amplifier will be approx 30V. There are 2 supply rails +30V and -30V. This is correctly stated as + – 30V. The difference between the supply rails is 60V. The speaker is connected by one transistor to one supply rail at a time. The maximum Voltage to the speaker can be no greater than 30V. That is +30V or -30V at any one point in time.

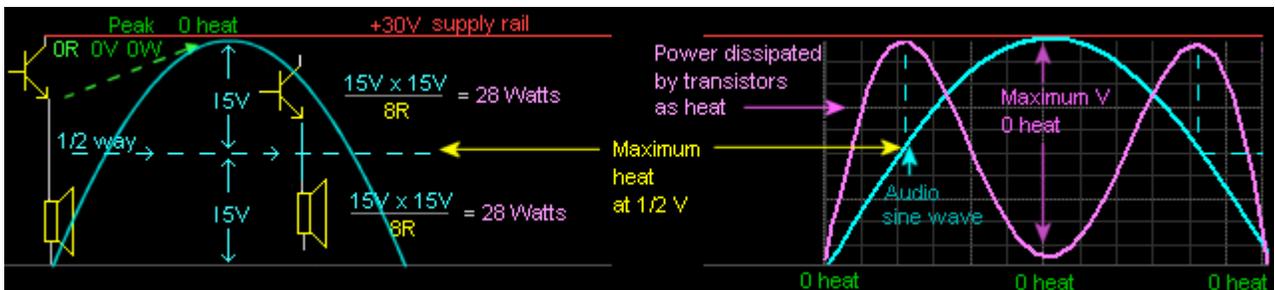


The peak of the sine wave is always slightly less than the rail Voltage, but we will begin with peak of the sine wave as being the same as the rail Voltage. Because a sine wave is constantly changing, (between zero and maximum) the RMS Voltage or power over 1 second has to be calculated. This is calculated from measuring the peak of the sine wave. Peak 30V x 0.707 = 21.21V RMS (20V RMS approx with losses).

Power Watts can be directly calculated from the output RMS Voltage and the Speaker R as  $V^2/R$ . Speaker 8R Resistance, 8Ω Ohms.  $20V \times 20V / 8R = 50$  Watts.

**Clipping** If the audio signal is driven into the rail supply, then the speaker is held at 30V for a longer period of time. The more the sine wave is driven into clipping the more it will change shape toward a square wave. The RMS formula (peak V x 0.707) for a sine wave, no longer applies, because the Voltage into the speaker will remain at 30V, flipping directly between +30V or -30V, dependant of frequency.  $30V \times 30V / 8R = 112$  Watts (100 Watts approx). We can now see that power to the speaker can be doubled by simply driving the amplifier into extreme clipping or overload. Many guitar players drive their amps in clipped distortion 100% of the time. Guitar amplifiers have multi stage high gain pre-amplifiers to enable the output to be easily driven into clipping. The distorted guitar sound now remains at a constant level, and is described as 'sustain'.

**Heat dissipation** Many amplifiers have insufficient heat sink or fan cooling. This may happen because of incorrect calculations, but is mostly the result of heat sink being expensive to purchase. Almost all output transistors are destroyed by over heating. Heat is the enemy of transistors. No solid state amplifier needs to be warmed up. The colder they are the better they will perform and the more reliable they will be. Many Audiophiles believe amplifiers sound magical when warm. Science has no meaning to fanatical believers.

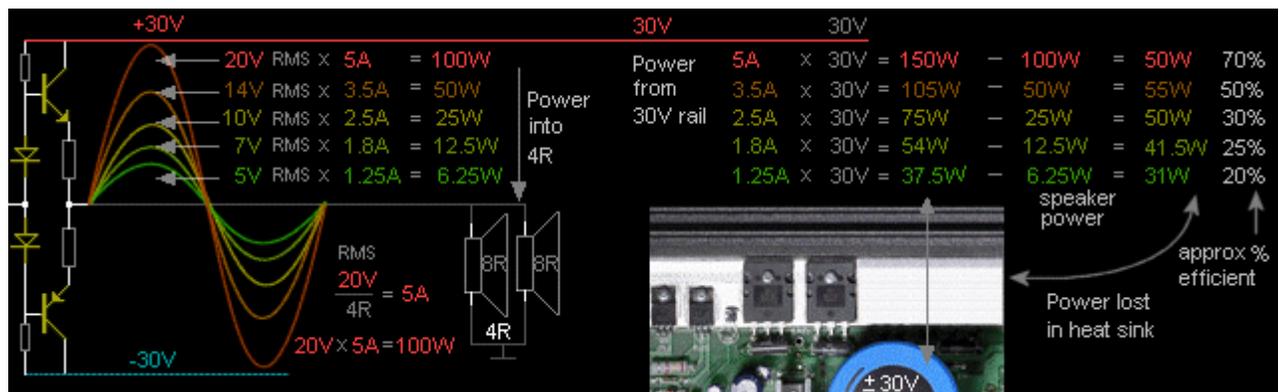


The above pic is a simplified example, showing transistor heat generated in one half of the sine wave. The resistance between collector and emitter changes from an open circuit to a short circuit. At the half way point, (at one moment in time) the resistance between collector and emitter equals the resistance of the speaker. Therefore 28W is dissipated in the speaker, and 28W is dissipated in the transistor (at one moment in time). When the sine wave has reached the 30V rail, the transistor is now a short circuit. Similar to an off on switch in the closed position, therefore no heat is dissipated across

the transistor.

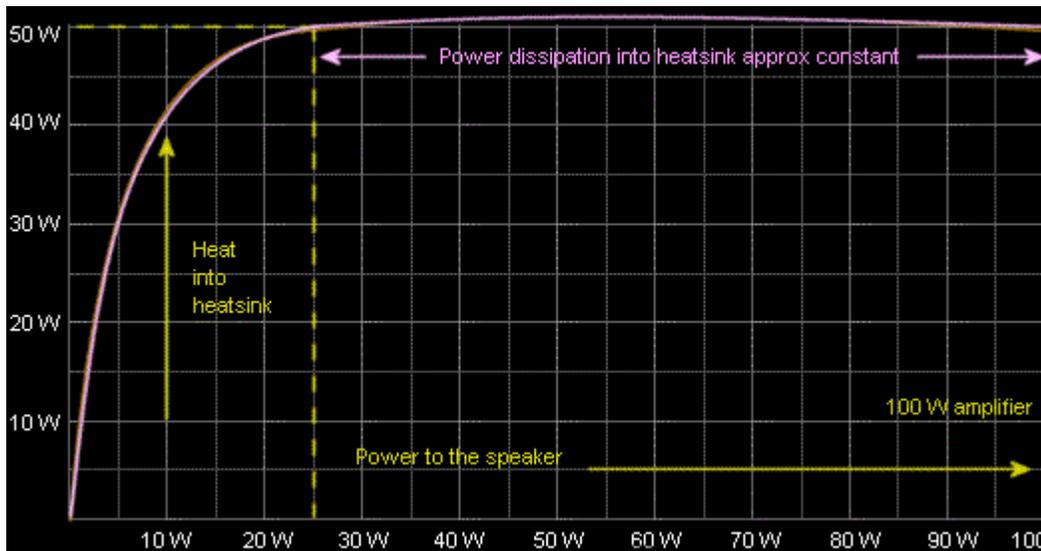
At the half point, maximum heat is dissipated by the transistor. At full power, less heat is dissipated by the transistor. When the amplifier is driven hard into clipping, very little heat is dissipated by the transistor. However this calculation is simplified to instantaneous points in time, to give easy examples. Real RMS calculations are taken over 1 second and require the Ohms law formula.

**Amplifier efficiency** With Ohms law we can calculate the RMS Power into the speaker. Then we can calculate how much power is taken from the power supply. The difference between the power to the speaker, and the power from the rail supply, is the power lost in the heat sink. The below description shows 2 x 8R speakers in parallel = 4R. 50 Watts into each 8R speaker, is the same as 100W into a single 4R speaker, and 4R is used in this description to give 100W. Using 100W as full power is easier to describe % efficiency.



20V RMS is the maximum level before the peak of the sine wave clips the 30V supply.  $20/4R = 5$  Amperes of current into the speaker.  $20V \times 5A = 100$  Watts. The 5A is supplied from the 30V supply (through the transistor) to the 4R speaker.  $5A \times 30V = 150W$ . 50W heat is dissipated by the transistor into the heat sink. Therefore at full power the amplifier is approx 65% to 70% efficient.

**70% efficient** Class B solid state amplifiers are described as being 70% efficient at full power. In academic text, the peak of a 20V RMS sine wave is 28V (not 30V).  $5A \times 28V = 140W$  which results in a 70% efficiency calculation. It is not possible for output transistors to reduce to absolute 0R, at the peak of the sine wave enabling the speaker to reach the 30V rail. There is a small residual on resistance in the output transistor at the peak of the sine wave. The peak of the sine wave can only get to within approx 2V to 6V of the 30V supply rail. This limitation results in another 4% approx loss. At lower power the efficiency decreases. At 1/2 power the amplifier is approx 50% efficient, at 1/4 power the amplifier is approx 30% efficient.



The graph on the right shows heat dissipated into the heat-sink from the output transistors remains (approx) constant between 1/4 power to full power. Maximum heat is created at approx 1/2 power.

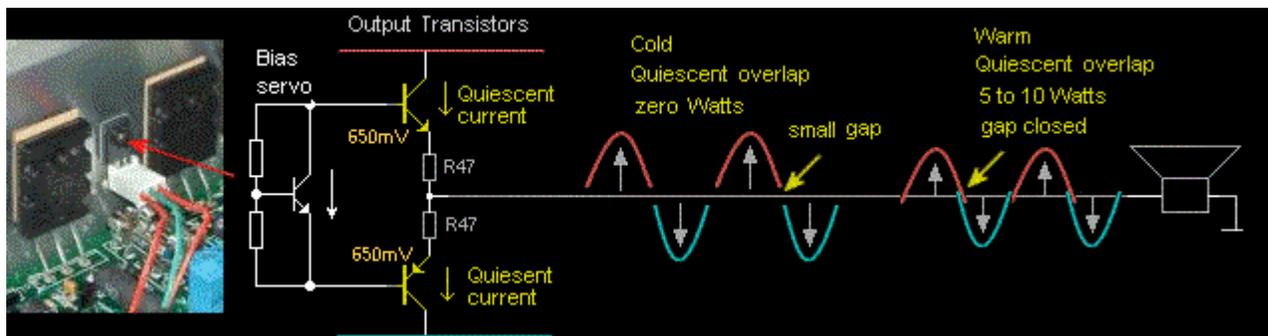
The % efficiency between power to the speaker and power lost as heat in the transistor, changes between 50% to 70%. However the most important thing to understand is, heat delivered into the heat-sink by the output transistors remains approx constant between 1/4 power to full power.

[www.sound.westhost.com/amp-efficiency](http://www.sound.westhost.com/amp-efficiency)

Whatever the full power rating of an amplifier is, approx 1/3 of that power figure is converted into heat by the output transistors. This 1/3 of the full power figure as heat into the heat sink remains approx constant between 1/4 power to full power. Only when the power to the speaker is reduced below 1/4 of full power, does the heat delivered to the heat-sink start to reduce.

### Why do some amps sound better when warm ?

If there is a verifiable audible difference between hot and cold in an amplifier then there is a fault condition which may or may not effect the reliability of the amplifier. This could be because of a circuit design fault, a component fault, or a connection fault. Also the internal parameters of all solid state devices (transistors and FETs) change with temperature. A common example is the quiescent bias current through the output transistors which often increases when the operating temperature increases.



A small Bias servo transistor is sometimes bolted to the heatsink between the output transistors as shown in the above left pic. In many amplifiers the heat sink is too small and the temperature may rise to 20deg to 40deg above ambient before there is sufficient thermal difference to the surrounding air to achieve stable dissipation. The bias servo transistor maybe calibrated to provide the correct quiescent

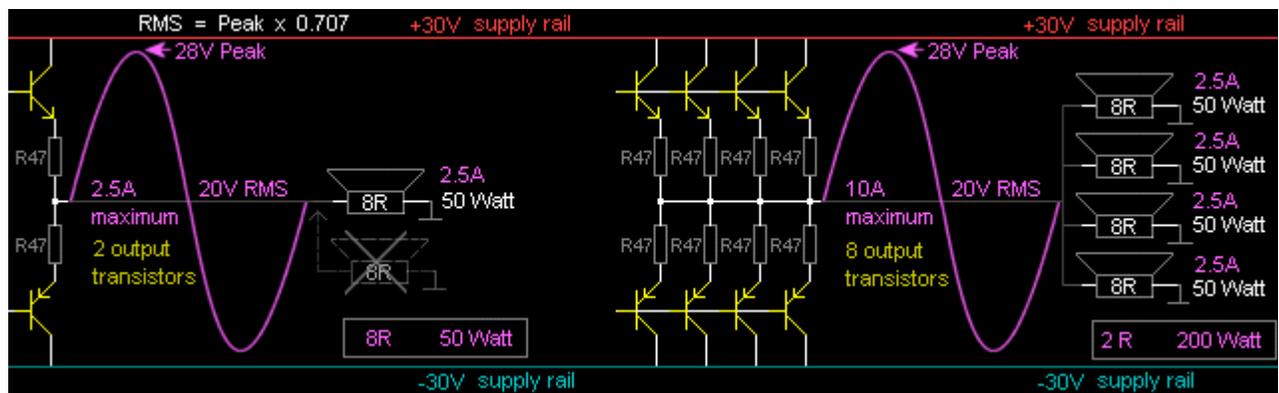
current through the output transistors (to insure there is zero crossover distortion) only when the heatsink temperature has risen to an unfortunate higher but stable level. These problems can also be the result of manufacturing cost cutting, where the extra profits can be used to market the benefit of an amplifier that must be warmed up before it will perform correctly.

## Speaker load Power output

In a capitalistic society, writing large numbers in front of Watts in marketing brochures, results in larger profits. Amplifiers can be marketed as 1,000 Watts, or 1,000,000 Watts. The real output power (Watts) of an amplifier is the result of the maximum Voltage and Amperes available from the supply rails and the speaker Impedance R. The reliability of an amplifier is mostly dependant the quality and number of output transistors, power supply regulation and size of heat sink. Increasing the number of output transistors distributes the current (Amperes) between them and provides better thermal dissipation into the heat sink.

An amplifier marketed as 200 Watt may or may not be able to deliver 200 Watts into a 8R speaker. It may be only able to deliver 50 Watt into a single 8R speaker. However it may be able to deliver 200 Watt into 2R which is the same as 4 x 8R speakers in parallel = 2R.

With a + - 30V rail supply the maximum output is 20V RMS. The on resistance of output transistors limits how close the peak of the sine wave can get to the 30V rail supply. The below pic shows a 2V difference. In many amplifiers the difference V between the peak of the audio wave and the V supply rail can be much greater. The below example assumes a + - 30V rail supply that does not change between 0 power to full power.

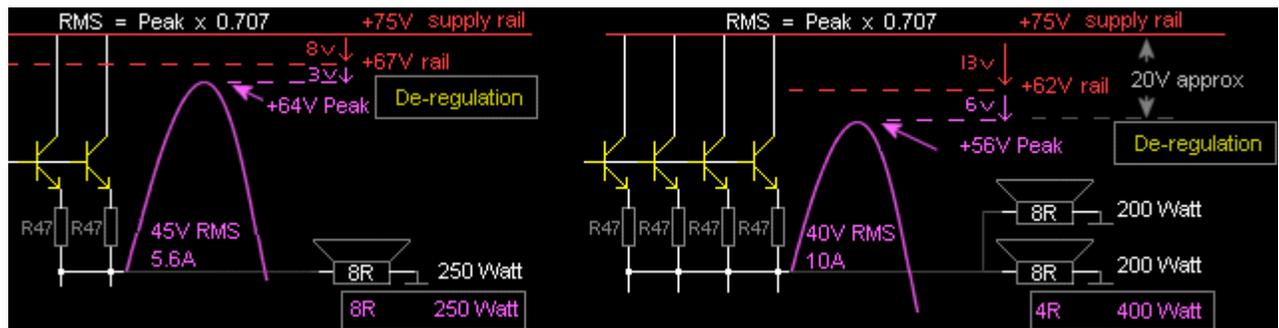


There is limit to how much current A (Amperes) 1 pair of output transistors can conduct and dissipate heat into a heat sink. Paralleling 2 x 8R speakers = 4R and may overheat 1 pair of output transistors and cause them to fail. If an amplifier is to drive a 4R or 2R load by paralleling many 8R speakers, then it is essential to parallel more output transistors to provide more current (Amperes) therefore dissipating more heat into the heat-sink. Also the heat-sink will have to be larger.

1 x 8R speaker	= 8R	20V RMS	x 2.5A	= 50 Watt
2 x 8R speakers	= 4R	20V RMS	x 5A	= 100 Watt
3 x 8R speakers	= 2.6R	20V RMS	x 7.5A	= 150 Watt
4 x 8R speakers	= 2R	20V RMS	x 10A	= 200 Watt

In an imaginary perfect amplifier the + - V supply rail would remain constant ( under load ) at full power. The on resistance of the output transistors would be 0R and the peak of the sine wave which would reach the + - V supply rail. In a real amplifier the supply rails would be approx + - 35V at 0

Power ( 0 load ), and collapse to  $\pm 30V$  at full power ( under load ). This is described as ‘Power supply de-regulation’ Therefore it is essential to measure the  $\pm V$  supply rails at full power ( under load ) to calculate maximum power.



45V RMS into 8R = 250 Watt. The peak of a 45V RMS sine wave is 64V. A typical amplifier with  $\pm 75V$  rail supply will collapse to approx  $\pm 67V$  ( with an 8R load ). With 2 8R speakers in parallel (4R) the  $\pm 75V$  rail supply will collapse down to  $\pm 62V$ . Because of the higher current (Amperes) the transistor on resistance will now have a greater Voltage across it (approx 6V) causing the peak of the sine wave to be reduced to 56V. Total power into 4R is 400 Watt. Therefore power to each 8R speaker is now 200 Watt.

## Output Stage

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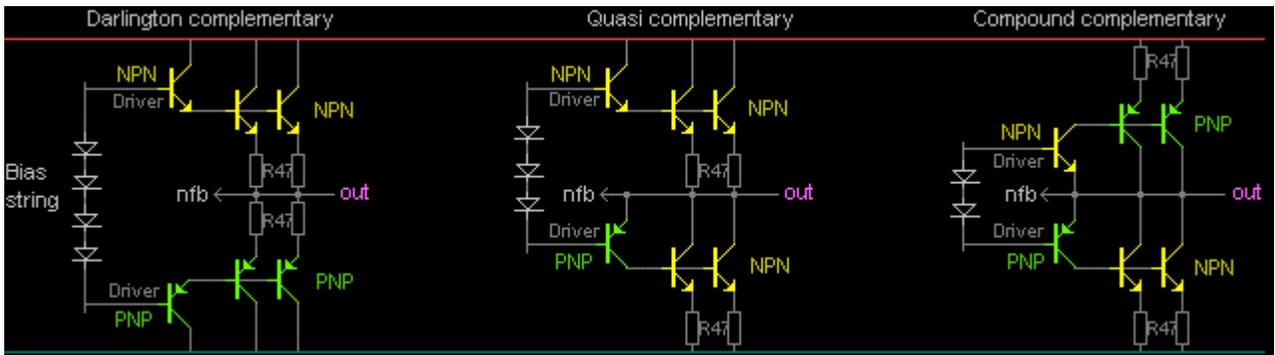
### Output stage MOS-FET Bridge Class G-H

In many power amplifiers the op-amp circuit is constructed with discrete components specifically designed for higher rail Voltages. Output transistors are added to provide extra current to drive a speaker. Large output transistors only have a small HFE current gain, therefore driver transistors are placed in front of the output transistors to increase to total current gain to approx 200. Amplifiers that use power MOS-FETs do not require driver transistors. The bias string can now be placed in the Class A driver circuit.

Output transistors can be arranged in three different ways. This description is a basic overview. Electronic design detail including PCBs for constructing power amplifiers is available on [www.sound.westhost.com](http://www.sound.westhost.com)

The first transistors were germanium which worked well for low power transistor radios in the 1960s and 70s. But germanium transistors were unstable and not reliable. Reliable Silicon transistors were invented later. High power amplifiers could only be built with silicon transistors.

The below pic shows parallel output transistors. Some large power amplifiers use many parallel output transistors. Large wire wound resistors  $1/2\Omega$  (R47) are placed in series with the emitters. These emitter resistors force the output transistors to equally share current and therefore will be equal in heat dissipation. Because a small amount of power is lost across the emitter resistors some amp designs use  $1/4\Omega$  (R22).



Darlington complementary is the basic order in how the output stage of an amplifier is taught. All output transistors are arranged as emitter followers. The collectors are connected directly to the rails. The emitter follows the signal on base within 650mV. The output transistors do not increase the size of the audio signal. Output transistors can only add current.

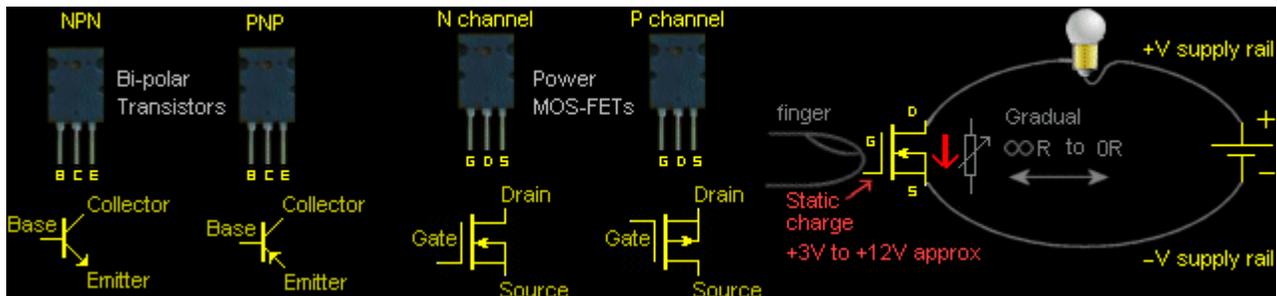
Quasi complementary is used in the majority of amplifiers. The first large silicon transistors (2N3055) enabled power amplifiers to be capable of 50 Watts but were only available as NPN and not as PNP. When PNP power transistors (2N2955) became available they were twice the price. The output transistors on the -V rail appear not to be wired as emitter followers. However the PNP driver transistor manages the output transistors collectively as a single compound large Emitter follower with a high HFE current gain.

Compound complementary NPN and PNP complementary output transistors designed for audio amplifiers are now available from many manufactures. Both NPN and PNP driver transistors manage the NPN and PNP output transistors collectively as compound single large Emitter followers with a high HFE current gain. The compound complementary arrangement has two advantages over the Darlington and Quasi complementary arrangements. Compound complementary has superior quiescent bias stability and the peak of the audio signal can get closer to the + - V rails, therefore slightly more power.

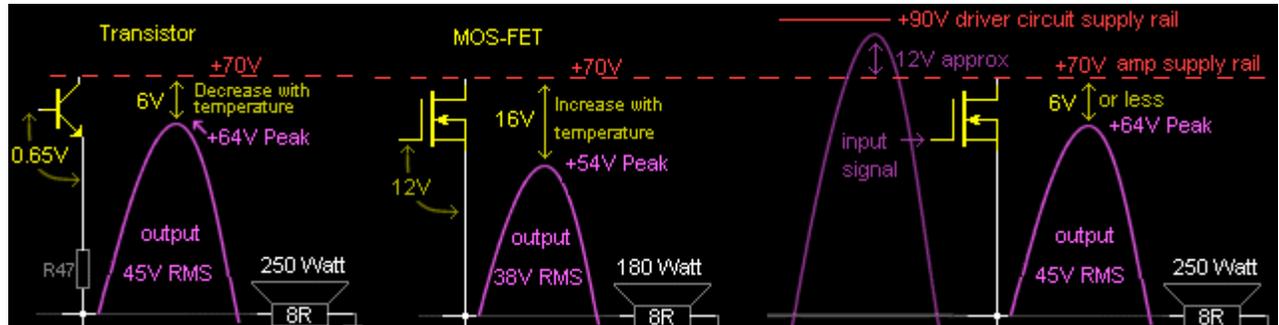
The negative feedback must be taken from the output of the amplifier. Power amplifiers have a signal gain of approx 20 to 40 (adjusted by R1 R2). But the additional driver and output transistors are now contained within the negative feedback loop, and this causes all power amplifiers to become unstable.

## Power MOS-FET

MOS-FET Metal Oxide Semiconductor Field-Effect Transistors are a variation of a Bi-polar transistor and are used in some amplifiers. A transistor functions by having a small amount of current between the Base\_Emitter to enable a larger current between Collector\_Emitter. FETs only require a static electrical charge as a Voltage (3V to 12V) on the Gate to enable a current to flow between Drain\_Source.



MOS-FETs can be easily controlled to turn on and off at high speed (Mega Hz) and are mostly used for switch-mode power supplies in computers etc, and are named Vertical MOS-FETs. Lateral Power MOS-FETs were developed for audio amplifiers during the 1990s. The primary disadvantage of FETs is that they deliver less power than a Bi-polar transistor amp using the same supply voltage. They are expensive, difficult to manufacture and only a few companies supply them. Less amplifier manufacturers use power MOS-FETs.



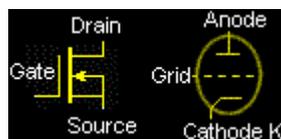
The above pic shows the difference between transistors and FETs using the same + – 70V supply.

- (1) At the peak of the sine the on resistance of a transistor decreases with temperature.  
At the peak of the sine the on resistance of a MOS-FET increases with temperature.
- (2) At the peak of the sine the transistor Emitter follows the Base within 0.65V  
At the peak of the sine the FET Source follows the Gate by approx 12V

The Source (output) will be 12V less than at the Gate. In technical terms a specified MOS-FET has a rated  $V_{ds}$  (saturated voltage, Drain to Source) of 12V at full current, which is subtracted from the DC value of the supply voltage. In the above example the amplifier using Power MOS-FETs will deliver 60 Watts less power than the same amplifier using transistors.

To solve the 12V loss problem requires the Gate to be driven 12V above the 70V rail supply at the peak of the sine wave. To achieve this the driver circuit would need to be powered from a separate higher rail supply (90V example), to enable the input signal to reach at least 12V above the 70V amp rail supply. The 6V difference between the peak of the sine wave and the rail supply shown in the above pic could be reduced to a lower voltage enabling greater power. Most amplifiers that use power MOS-FETs do not have this extra circuitry.

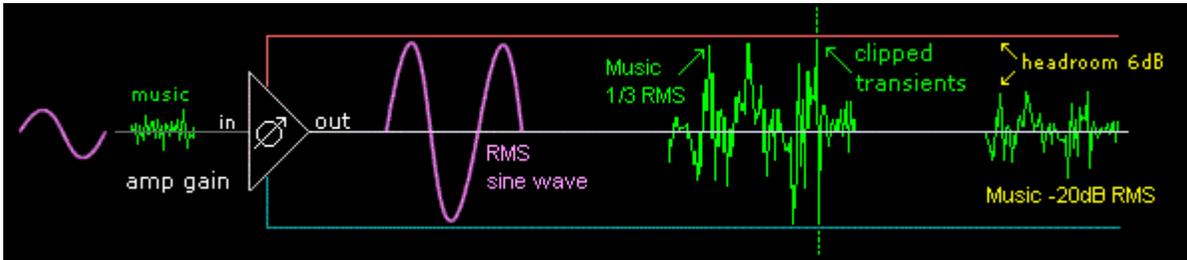
[wikipedia.org / MOS-FET](http://wikipedia.org / MOS-FET)



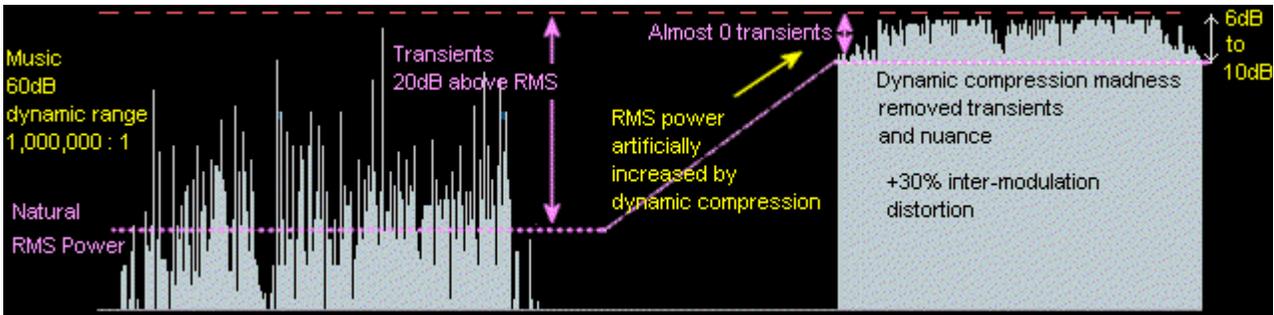
Paranormal beliefs exist about FETs sounding like valves, but what are the facts? The only similarity with FETs and valves is that the input Grid of a valve and the Gate of a FET require no current (Amperes) to function. Outside of this single point Valves and FETs have no similarity whatsoever. Without being previously informed, it is impossible to hear or scientifically test if the output devices in an amplifier are Transistors or MOS-FETs. The \$1,000,000 prize offered by the [James Randi Educational Foundation](http://James Randi Educational Foundation) for anyone providing proof of the paranormal, should also include any Audiophile who can prove under a double blind (A B comparison) to hear a difference between Transistors and FETs.

## RMS power and music compression

From the previous description about the amount of heat generated by the transistors into the heat sink, the question arises – How is it possible for the majority of amplifiers not to be destroyed by overheating?



Music is capable of a 60dB (1,000,000:1) dynamic range. The transients in music are very small in energy but are approx 20dB above the RMS music level. The average RMS power of fully dynamic music can not go above -20dB of the amplifiers full power capacity without the transients clipping the rail supplies. 20dB is 100:1 so therefore a 100 Watt amplifier should not be driven above 1 Watt of RMS music level (over approx 1 minute of time) to avoid transients being driven into rail clipping. A 100 Watt amplifier can only be used at an average of 1 Watt with fully dynamic music. For this reason amplifiers less than 60 Watts should not be considered as audiophile status, but unfortunately many are.



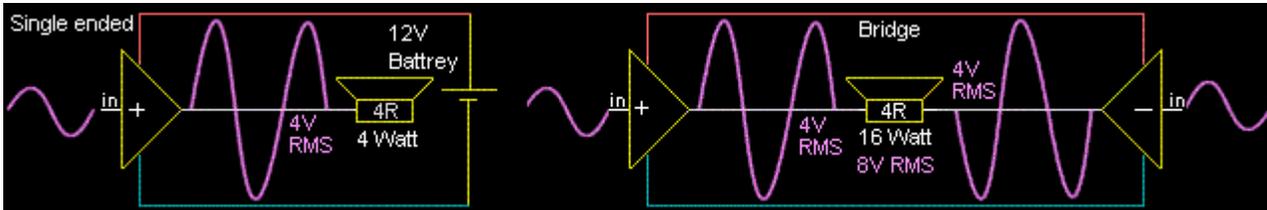
The modern digital recording trend is to dynamically compress music in an attempt to remove all the dynamic range which includes transients. Dynamic compression allows music to be played at higher power without transient clipping. However excessive dynamic compression imposes extreme inter-modulation distortion. Voices and instruments are squashed and mangled together, rendering articulation of voices and instruments so removed from sounding natural that it is often difficult to recognise. The largest problem of this irresponsible recording behaviour in pop recordings, TV programs and films is that it makes it difficult to understand the words being sung or dialogue spoken. The inter-modulation distortion including the removal of articulation caused by dynamically compressing music is so great (approx 30% distortion), that audiophiles and professional sound installers pretending to be concerned about inaudible time alignment differences of speaker driver components on a baffle board is delusion to say the least.

Also dynamic compressed music is already so distorted by the dynamic compression in the recording process that it can be driven into supply rail clipping without being audibly noticed in comparison to the distortion created by dynamic compression. The maximum level an amplifier can be driven with dynamically compressed music before the added distortion caused by clipping into the rail supplies becomes objectionable, is 1/3 of the equivalent energy of a sine wave at full power. Worse still, in most live concerts the music is further compressed so the average RMS power can be taken close to 1/2 full power of the amplifiers capacity. In this condition many professional high power amplifiers

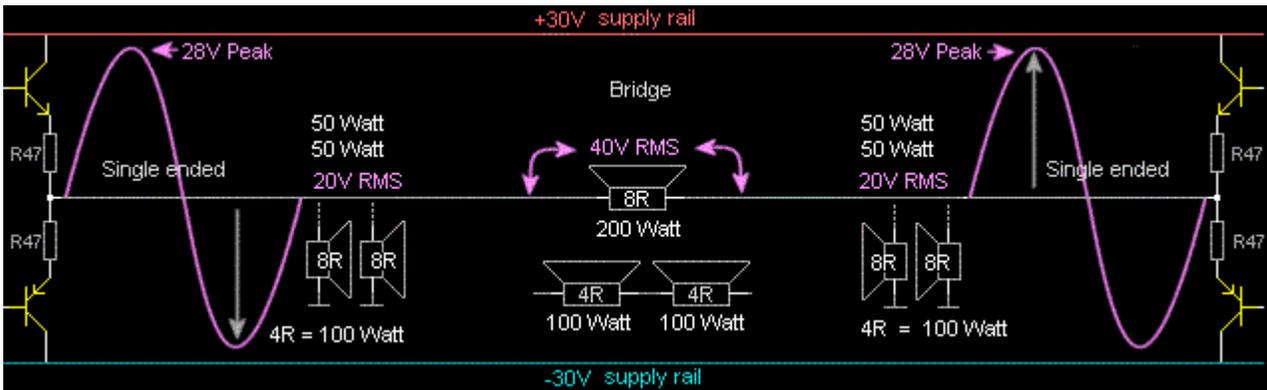
will shut down from overheating.

### Bridge amp advantage

Single ended is 1 amplifier driving a speaker. Single ended is the most commonly used application. However a speaker can be bridged between 2 amplifiers. Bridging a speaker between 2 amplifiers is one of the least understood concepts about amplifier management. Bridging 1 speaker between 2 amplifiers is commonly used in sound systems for vehicles where the supply Voltage is limited by the 12V battery.



From a 12V DC supply 4V RMS is the maximum that can be achieved from a single ended amplifier.  $4V \times 4V / 8R = 2 \text{ Watt}$ . The reason 8Ω speakers are not used in vehicles.  $4V \times 4V / 4R = 4 \text{ Watt}$ . The reason 4Ω speakers are used in vehicles. Bridging a speaker between 2 amplifiers and driving one amp in opposite phase  $4V + 4V = 8V \text{ RMS}$ .  $8V \times 8V / 4R = 16 \text{ Watt}$ . Therefore many vehicle sound systems use bridge amplifiers to power speakers.

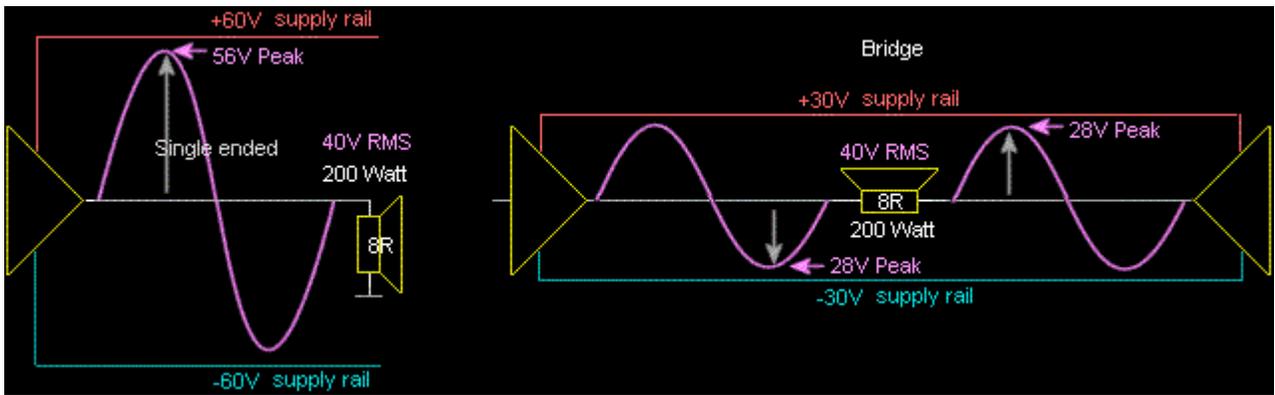


A popular belief is that 4 times the power is achieved from bridging 2 amplifiers in comparison to single ended application. This is partially true, but there is no such thing as something for nothing. The above pic shows two 100 Watt amplifiers with + – 30V rail supplies. 20V RMS is the maximum from a + – 30V rail supply.

20V RMS into a 8Ω speaker is 50 Watt.  
20V RMS into a 4Ω speaker is 100 Watt.

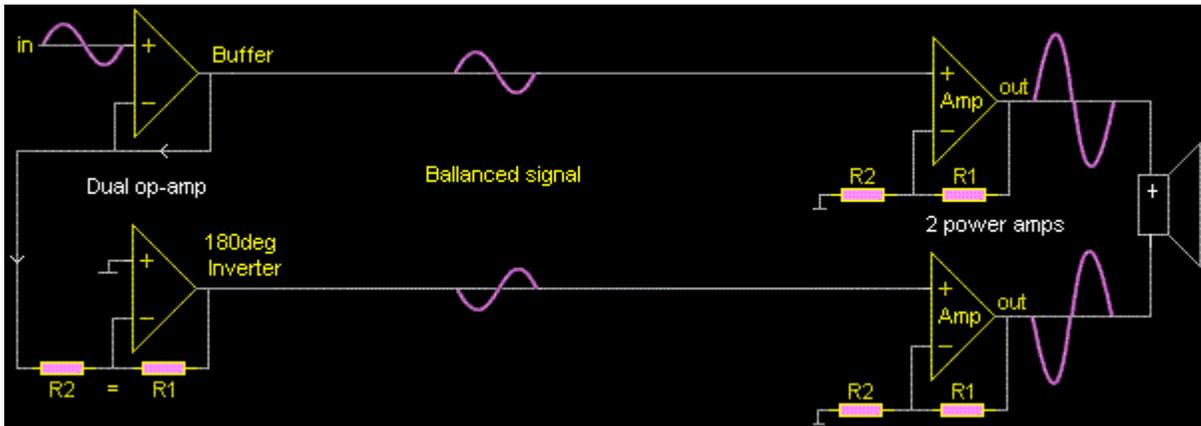
Bridging two amplifiers  $20V + 20V = 40V \text{ RMS}$ .  
 $40V \times 40V / 8R = 200 \text{ Watt}$ .

By paying close attention we can see that 4 times the power is achieved from bridging 2 amplifiers delivering 40V RMS into a 8Ω speaker (200 Watt) if we are comparing it to a single ended amplifier delivering 20V RMS into the same 8Ω speaker (50 Watts). However when comparing a bridge amplifier delivering 40V RMS into a 8Ω speaker (200 Watt) to a single ended amplifier delivering 20V RMS into a 4Ω speaker (100 Watt) then bridge only appears twice as powerful.

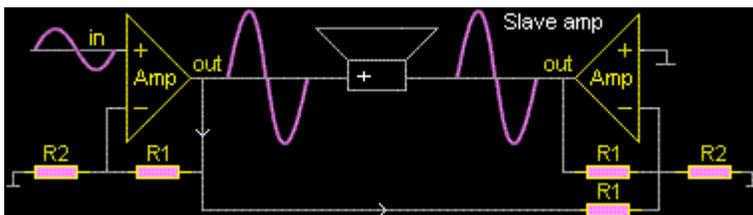


The advantage of bridge is that it delivers the same power as a single ended amplifier with only half the rail Voltage. (40V RMS into 8R = 200 Watt) 40V RMS from a single ended amplifier requires + – 60V rails, whereas 40V RMS from bridged amps only requires + – 30V rails. With bridged amps the speaker is powered from both + – V rail supplies at the same time, instead of alternate between supply rails as with a single ended amp. Therefore Bridge amps make more efficient use of the rail supplies. Also the maximum Voltage across the transistors is half by comparison to single ended amp. Bridge is the most effective method to drive a speaker. The only disadvantage is higher cost.

Bridge management Bridging a speaker between two amplifiers is the most effective means to power a speaker. The power is supplied from both + and – V rails at the same time enabling twice the voltage across the speaker in comparison to using a single amplifier. The only disadvantage is cost.



A dual op-amp is often used to create a balanced signal. The first op-amp acts as a buffer with unity gain. The output of the buffer is sent to an inverting buffer to flip the signal 180deg. A perfectly balanced signal is then sent to the power amps.



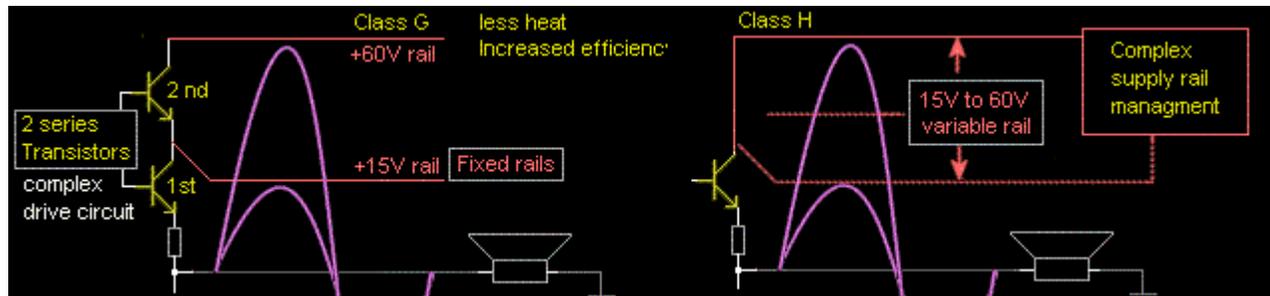
There is also an alternate method that does not require a dual op-amp to create a balanced signal to be sent to the amplifiers. The output of the first amplifier is sent to the -inverting input of the second amplifier through a resistor that is the same value as R1. The second power amp now acts as an inverting slave. This is the simplest means to bridge 2 amplifiers as it only requires the addition of a single R1 resistor. The only disadvantage is that any distortion in the first amp is sent to the second

amp, causing the distortion to be doubled.

[www.sound.westhost.com/amp\\_design](http://www.sound.westhost.com/amp_design) Advanced essential reading  
[www.sound.westhost.com/amp\\_sound](http://www.sound.westhost.com/amp_sound) more advanced essential reading

## Class G Class H

There have been many attempts by amplifier designers to reduce the 30% to 50% wasted heat across the output transistors. By keeping the rail supply close to the peak of the sine wave, the heat dissipation across the transistors is kept to minimum. These designs require greater circuit complexity.



Class G has 4 fixed rails. 2 x +V supply rails and 2 x -V supply rails. There are 2 transistors in series for each + -V supply rail. The above pic only shows the +V supply only. The same arrangement is applied to the -V rail supply. At low level, power is taken from the lower Voltage rail by the 1st transistor. As the audio signal increases the second transistor connected to the higher Voltage rail starts to conduct. All the current flows through the 1st transistor to the speaker.

Class H gives a similar result to Class G and is slightly more efficient. The +V rail and a -V rail change voltage and increase when required. Class H requires the circuit to predict when a high transient input signal is about to appear. The rail Voltage must increase ahead of the audio signal for it not to clip. Because this is not always possible transient clipping distortion does happen.

Both Class G and Class H are sometimes used by hi-powered amplifiers that are expected to be used at low power for most of the time, hereby minimising the amount of heat wasted by being able to function alternatively from a higher to a lower rail Voltage. Class G is also used for domestic amplifiers with a small heat sink.

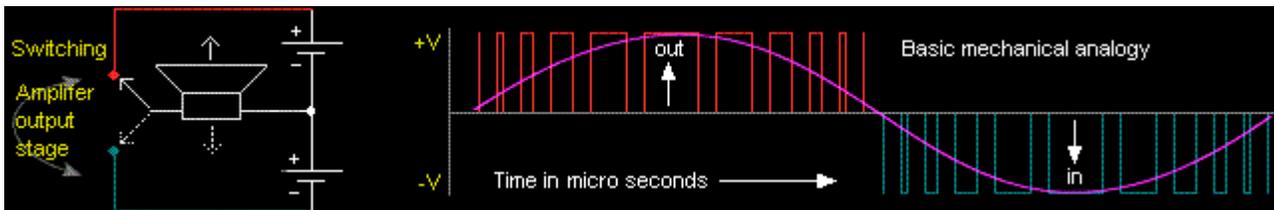
[www.sound.westhost.com/amp-basics](http://www.sound.westhost.com/amp-basics)  
[www.sound.westhost.com/amp-design](http://www.sound.westhost.com/amp-design)

## Class D

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### Class D Pulse Width Modulation

Class D amplifiers are described as PWD Pulse Width Modulation amplifiers. Class D is similar to a Switch Mode Power supply. Switch mode power supplies and Class D amplifiers are low mass, generate very little heat and approx 90% efficient. The majority of small amplifiers less than 20 Watts in computers and domestic application including TVs etc are now Class D amplifiers. Above 20 Watt the complexity and problems of Class D increase exponentially. Similar to switch mode power supplies, Class D amplifiers are virtually un-serviceable and life after 7 years considered a blessing.

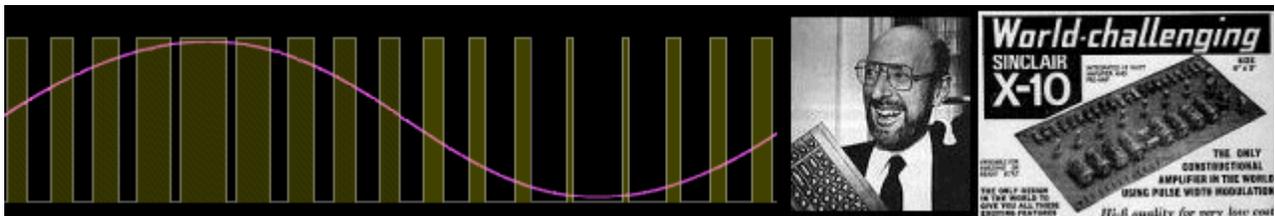


Switching on and off a light bulb, electric motor, electric heater etc at high speed to control power is not rocket science to understand. With a little imagination a speaker could be switched on and off across 2 batteries and by varying the on time compared to the off time (providing the switching speed is above hearing range) the mass of the cone will average the energy to represent an audio signal.

Pre 1970 Valve technology reigned supreme. Outside of experimental computers the first transistors were used in small cheap radios which is how the Japanese electronics industry and Sony began. Transistors do and not have a linear function similar to valves, but are ideally suited to switching on and off. Class B analogue transistor amplifiers above 10 Watts were originally unreliable. It was assumed that transistors would not be suited for making high powered class AB amplifiers.

In 1964 Clive Sinclair an electronic inventive genius along with fellow engineer Gordon Edge developed the first PWM amplifier. The X-10 was marketed as 10 Watts but only produced 2 to 3 Watts. The Z-12 arrived shortly later which reliably produced 12 Watts, and the race was on. Almost every young electronic audio enthusiast started experimenting with PWM amplifiers. It was believed that PWM amplifiers capable of 100s of Watts would be available at low cost before a man would set foot on the moon.

[www.nvg.org/PlanetSinclair](http://www.nvg.org/PlanetSinclair)  
[wikipedia.org/CliveSinclair](http://wikipedia.org/CliveSinclair)

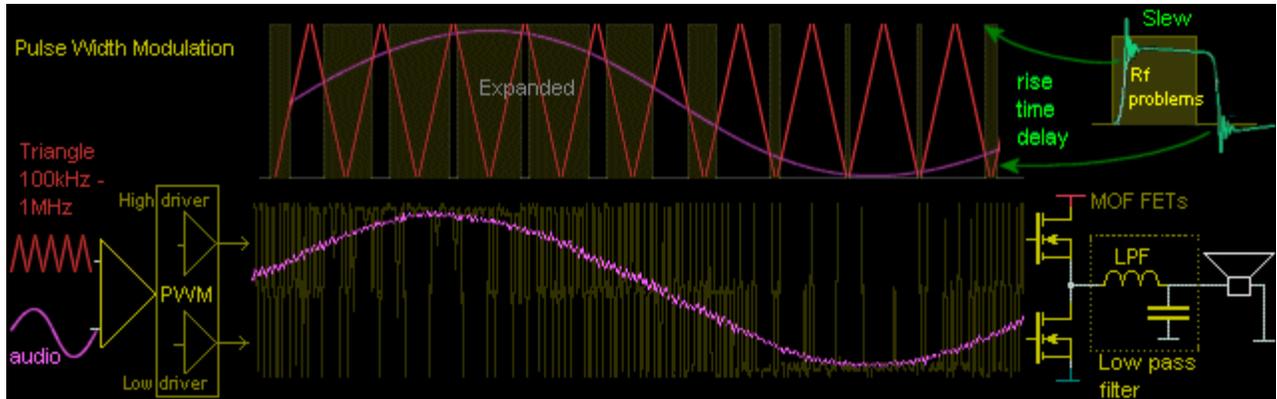


Similar to Dr Frankenstein everyone was originally unaware of the monstrous lurking problems that were about to be unleashed. Any radio or TV in close vicinity to a PWM amplifier was blocked out by the Rf noise interference generated by the high frequency switching of the transistors. Test equipment capable of analysing switching frequency rise times that approached the limits of physics was not readily available. Government communication authorities quickly outlawed the use of any device that interfered with Rf transmissions. Another 30 years would pass before the science for controlling PWM Rf noise radiation was better understood. Very few high power PWM amplifiers today are CE [Communication emission](#) certified.

Understanding how a PWM amplifier works requires a lateral shift in thinking, but once the basic principle is seen it appears almost obvious. PWM does not require separate switching for each half of the wave form as in the first pic. A fixed high frequency square wave 100kHz to 1MHz is width modulated by the audio sine wave. At the output is a filter stopping everything above 20kHz from getting to the speaker.

The above pic is expanded to show only 16 pulses across an audio sine wave. When the square pulse widths are equal the averaged energy output is zero. As the pulse widths become wider than equal the averaged energy shifts toward the +Ve half of the audio sine wave. As the pulse widths become less than equal the averaged energy shifts toward -Ve half of the audio sine wave. But how is this

achieved?



(1) To make the transition from a audio sine wave to a fixed 100kHz to 1MHz modulated square wave that simulates the energy of the sine wave, requires an intermediate step. First, a fixed 100kHz to 1MHz triangle wave is generated. The triangular wave must be approx x 100 the highest audio frequency. As the sine wave moves across the high frequency triangle wave a comparator recognises which part of the sine wave is above or below the points crossed on the triangle wave.

(2) As the triangle wave is crossed by the sine wave the MOS-FETs are turned on or off generating a high energy square wave of varying width. The MOS-FETs are not capable of turning on and off instantly. In the real world the speed of light is a theoretical concept. The Slew of the MOS-FETs governs how quickly the change from off to on and off again can happen.

(3) This transition from off to on represents the leading or trailing edge of an extreme high frequency which is magnitudes higher than the 100KHz to 1MHz fixed switching frequency. As the corners of the leading and trailing edges of the square wave make the transition from vertical to horizontal Rf ringing noise is generated. This Rf ringing also creates secondary problems inside the MOS-FETs that result in excessive heating and potential breakdowns.

The lower part of the above pic shows a more realistic view of the hi frequency square wave in comparison the comparatively low frequency audio sine wave. It clearly shows the noise of the switching frequency imposed on the audio sine wave that has to be filtered out. The high frequency modulating frequency has to be filtered out leaving the resultant energy of the amplified audio. The LPF Low pass filter only allows everything below 20kHz to get to the speaker.



Class D amplifiers are similar to the switch mode power supplies in computers and therefore non user serviceable. Failures are simply dealt with by warranty replacement or purchasing a new amplifier. The technology of professional high power Class D amplifiers is so complex and fragile that reliable operation of after 7 years should not be expected.

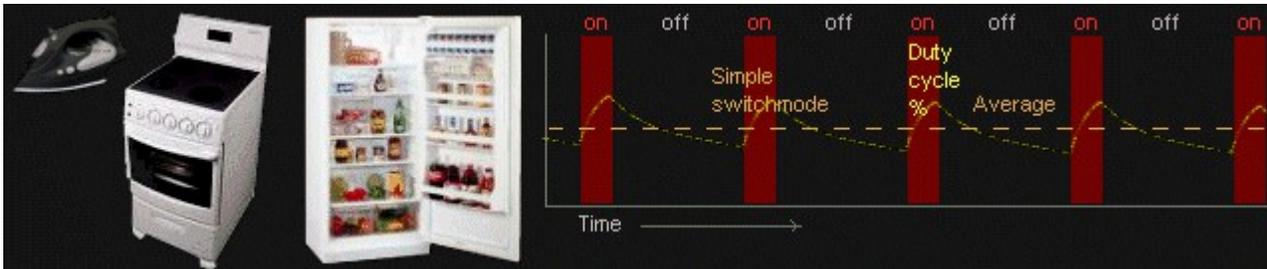
A deeper technical understanding of PWM amplifiers and their limitations is on this PDF published by component manufacturer IR (International Rectifier)

[Application Note AN-1071](#) by Jun Honda and Jonathan Adams

[www.sound.westhost.com/PWM](http://www.sound.westhost.com/PWM) An excellent technical review of PWM problems and solutions.  
[wikipedia.org/ Switching amplifier](http://wikipedia.org/Switching amplifier)  
[www.digiamps.com](http://www.digiamps.com) PWM Amplifier Modules  
[www.amplifier.cd](http://www.amplifier.cd)

### Switch mode power supply

Many household white goods, refrigerators, steam irons and electric ovens are simply controlled by being switched on and off. As the refrigerators warms up it is switched on until it cools down, then it is switched off again and so on. Electric ovens and steam irons function in reverse way. The time it is switched on compared to the time it is switched on is the Duty cycle. Duty cycle is measured as % of on time compared to off time. A refrigerator with a permanently open door would have a Duty cycle of 100%. With the door closed the Duty cycle may be only 10% depending on the quality of insulation. A refrigerator may be on for 5 minutes and off for 50 minutes and so on to maintain cold temperature. The longer the time span (clock speed) of the duty cycle the greater will be the temperature difference between on and off. The fastest time a refrigerator may be able to turn on and off would be approx 1 minute. Therefore it would have a clock speed of 1 minute. If the refrigerator had a higher clock speed of 1 second (1Hz) so it could be turned on for 1 second and off for 10 seconds maintaining the same Duty cycle of 10% the temperature would be controlled more evenly.



With a conventional ‘Capacitor Input’ power supply the mains frequency of 50Hz or 60Hz can be described as the clock speed. A 50Hz or 60Hz clock speed is very slow and the reason for the large ripple in the supply rails. If the mains frequency was 1kHz then the ripple on the supply rails would be virtually non existent. Switch mode power supplies in computers and most home entertainment systems DVDs and plug packs etc have clock speeds of 25kHz to 100kHz. A computer switch mode power supply does not use a large heavy mains transformer. The advantage of a switch mode power supply is reduced mass, but its circuitry is extremely complex and virtually un-serviceable.

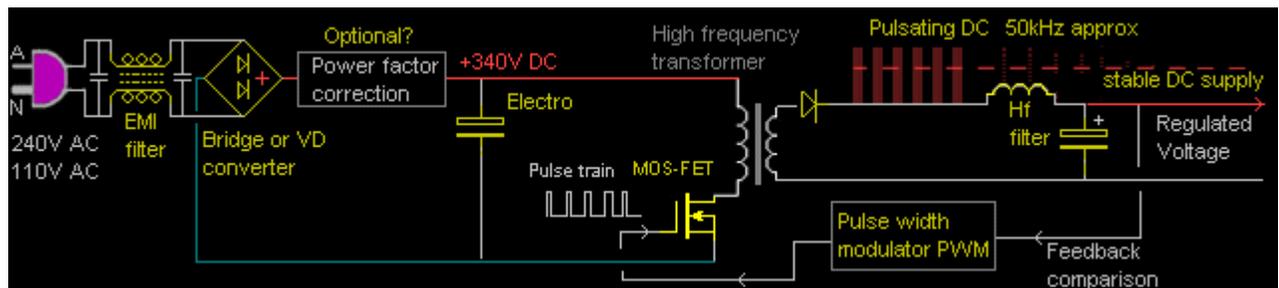
The mains Voltage is fed into through an EMI Electro magnetic Interference filter into a Bridge or VD Voltage Doubler rectifier and converted directly to DC. Most switch mode supplies are internally designed to function between 280V DC to 340V DC.

240V AC mains x 1.414 = 340V DC with Bridge rectification.

110V AC mains x 1.414 x 2 = 310V DC with Voltage Doubler rectification.

The EMI filter stops Rf (radio frequency) switching noise generated by the switch mode power supply

from getting back out into the 110V 240V mains system effecting other equipment connected to the mains. There are strict international regulations governing the maximum Rf radio frequency noise that electronic products must comply within. This regulation is named C-tic.



- (1) The mains rectified 300V DC approx is switched on and off through a transformer. The MOS-FET act as a switch. The switching speed can be between 25kHz and 100kHz. The PWM (pulse width modulator) controls the duty cycle of the switching frequency which is similar to a square wave that has variable width. The transformer is specifically designed for switching frequencies of approx 50kHz. High frequency transformers are very small and therefore have little mass.
- (2) The secondary of the transformer reduces the switched AC pulses to a lower voltage, plus isolating the mains to ensure electrical safety. The diode then converts the secondary pulses to the required polarity. The secondary Electrolytic capacitor smoothes out the 50kHz DC pulses to an almost perfectly smooth DC supply.
- (3) The secondary Voltage is fed back into a comparator within the pulse width modulator (PWM). The PWM then automatically adjusts the width (Duty cycle) of the turn on and off pulses to the power MOS-FET which acts as the high frequency switch, controlling the current Amperes through the primary of the high frequency transformer, which in turn corrects the voltage at the secondary to insure that the output DC Voltage is exact.

The high switching frequency gives the switch mode power supply a unique advantage of providing a near perfect ripple free smooth and regulated DC supply Voltage that does not decrease in Voltage as extra current is required.

However switch mode power supplies are extremely complex. The high switching frequency technology is in the Rf radio frequency band. This causes major headaches to technically manage containing Rf radiation from the printed circuit tracks and connecting wires within the circuit getting to the outside world and causing Rf noise contamination in radio and TV. Also the circuit operation is extremely fragile, and if any fault occurs (no matter how small) it contaminates almost every sector of the circuit.

Switch mode power supplies are non user serviceable, failures are simply dealt with by trashing the power supply and purchasing a new one. Attempting to service a switch mode power supply is virtually out of the question, except for fanatics who have infinite time. The technology is so complex and fragile that reliable operation after 7 years is not always expected but considered a blessing. As the world converts to using switch mode power supplies and PWM Class D audio amplifiers, they will eventually end up as the dominant mass of rubbish land fills.

[www.hardwaresecrets.com/Switching Power Supplies](http://www.hardwaresecrets.com/Switching Power Supplies)  
[www.sound.westhost.com/power-supplies](http://www.sound.westhost.com/power-supplies)

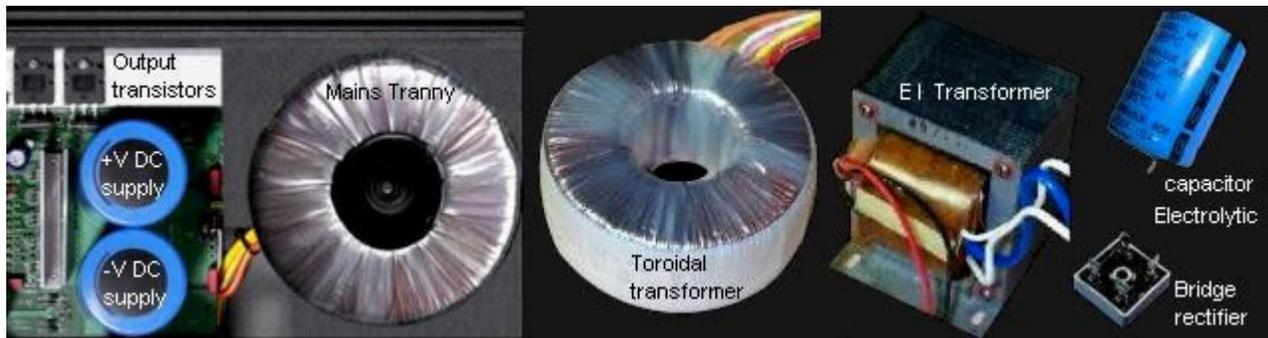
## Power Supply

When looking inside an amplifier the power supply is easy to recognise. The majority of power supplies in amplifiers are described as a 'Capacitor Input' supply. The description below is an overview.

- 1 Large mains tranny Toroidal or EI.
- 1 Bridge rectifier
- 2 Electrolytic capacitors

The mains tranny has a primary winding and two isolated secondary windings. The mains transformer steps down the 110V or 240V AC to 2 equal lower AC voltages. The power of a mains tranny is directly proportional to its physical size. However a 200 Watt transformer is described as 200 VA and not 200 Watts. (Volts x Amperes = Watts). The reason the power of a transformer is described as VA and not Watts is described in detail on [www.sound.westhost.com / Transformers](http://www.sound.westhost.com/Transformers)

The iron core can be a donut Toroidal shape or a square shape described as EI. The iron core consists of thin layers of silicon soft steel, which has an excellent ability to conduct magnetic energy without retaining being magnetised. Therefore it can be alternately magnetised N-S and S-N at 50 or 60 times a second. 220V – 240V AC 50Hz UK Aus Europe or 110V AC 60Hz USA Canada etc.



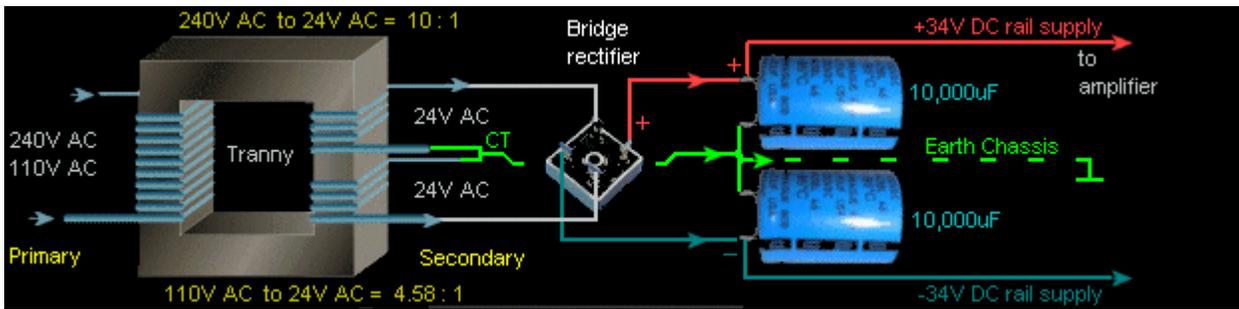
A common mis-conception is that Toroidal transformers are more efficient than square EI transformers, but this is not so. For the same mass a square EI tranny will be slightly more powerful and have superior regulation, but EI transformers externally radiate the AC magnet field which can induce eddy current loops into the chassis and nearby components causing hum problems. EI trannys are cheaper. Toroidal transformers are more expensive to make. The primary advantage of Toroidal is that they radiate virtually zero external magnetic field, therefore correctly suited for audio amplifiers.

The traditional way to draw a power supply is from left to right. The example tranny in the below pic clearly shows that the primary and secondary windings are completely isolated from each other. Primary Secondary winding isolation is also essential for electrical safety. The AC Voltage on the secondary winding is simply adjusted by the ratio of turns between primary and secondary. As an example, if the primary has 1,000 turns and the secondary has 100 turns (Ratio 10:1) 240V AC on the primary will result in 24V AC on the secondary. The mass of copper wire on both primary and secondary must be approx equal.

Primary 240 Watts at 240V AC = 1 Ampere.

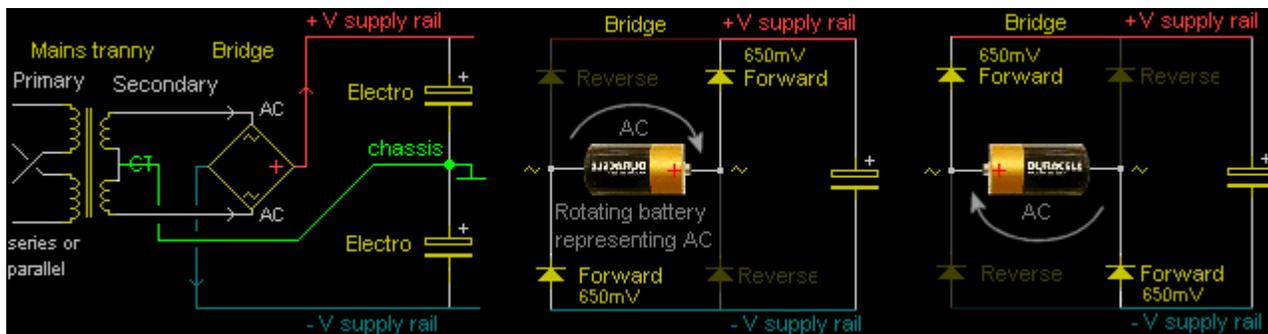
Secondary 240 Watts at 24 V AC = 10 Ampere.

Therefore the secondary wire gauge must be a larger diameter to conduct 10 Amperes. The description given is a guide however in a real transformer there are losses of efficiency which have to be calculated for. Efficiency loss of magnetic coupling between the primary and secondary windings plus the Resistance of the wire length in all windings that has to be adjusted for in the turns ratio.



The Bridge rectifier (Bridge) converts the 24V AC to DC. The electrolytic capacitors are similar to storage batteries that can be almost instantly charged and discharged repeatedly for an indefinite period.

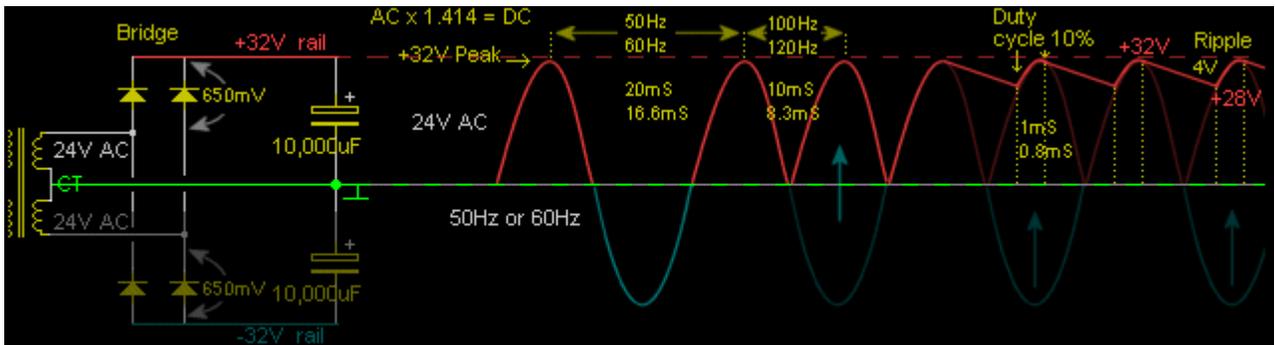
The power supply must be mechanically wired in the exact order shown in the above pic. The two secondary windings are in series. The point where they are connected is named CT (Center Tap). The CT is wired directly to the junction of the two Electors. Only from the junction of the Electors is the CT connected to chassis. If the CT was connected to chassis at the transformer an induced hum loop could be injected into the amplifier. Also the supply rails must be wired directly to the terminals of the Electors and not from the Bridge. Mechanical wiring order is uniquely essential to power supplies. Many early musical instrument amplifiers had hum loop problems caused by external magnetic field of EI trannys and power supply mechanical wiring.



A bridge rectifier has 4 diodes. A Bridge can be as a single unit with 4 internal diodes or 4 separate diodes. The diodes are arranged in order so no matter which way around the Voltage polarity appears at the junction of each pair of diodes, from AC or a rotating battery, 2 diodes will be in the forward direction pointing toward the -V end of the example AC rotating battery, and 2 diodes will be in the reverse direction (open circuit). The polarity at the supply rails will always be the same and therefore correct. Remember there is a 0.65V (650mV) loss across each diode in the forward direction.

A single primary winding is rarely used because of different AC mains Voltages throughout the world. Therefore most mains trannys have two primary windings which can be connected in series or parallel.

For countries with 110V 60Hz AC USA Canada etc the two primary windings are connected in parallel. For the countries with 220V or 240V 50Hz AC Europe UK Aus etc the two primary windings are connected in series.

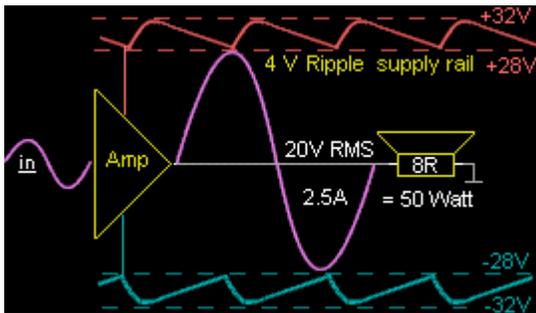


The above pic shows how 24V AC is converted to the +V supply (vice-versa for the -V supply). The +V peaks of the sine wave appear 50 or 60 times / second (50Hz or 60Hz) which is each 20mS or 16.6mS.

mS = milli-Second.

The bridge rectifier reverses the polarity of the -V half of the sine wave so the +V peaks appear 100 or 120 times / second (100Hz or 120Hz) which is each 10mS or 8.3mS.

This is now described as pulsating DC.



The electrolytic capacitor charges up (similar to a battery) to the peak of each +1/2 of the sine wave. As the pulsating DC drops back to zero the Electrolytic capacitor provide power to the amplifier. The capacitor starts to discharge from +32V down to +28V which is a drop of 4V (12.5%). This is described as Ripple Voltage and should not decrease less than 10%.

However 10mS or 8.3mS later the next peak of the +V 1/2 of the sine wave charges up the capacitor again and again and again. By looking closely at the drawing each charging cycle only takes place for a very small period of time approx 1mS to 0.8mS. This small charging time is named the Duty cycle.

Also it is only during this small time that power is provided from the mains transformer. The duty cycle charging time is only 1/10 of each cycle, therefore the charging current Amperes has to be x 10 greater than the average being supplied to the amplifier from the capacitor.

When no power is being delivered to the speaker the ripple decreases to almost zero and only increases with the current Amperes demanded upon the power supply. Increasing the size of the capacitor reduces the ripple Voltage. As a rule of thumb 2,000uF is the minimum value for each 1 Ampere of current. A power supply that has to provide 5 Amperes will need a 10,000uF capacitor. Obviously the larger the capacitor the better.

Because the power supply is only charging the capacitors at the peak of each cycle for only 1/10 of the time means the overall efficiency of a 'Capacitor Input' power supply is approx 70%. This also means that a mains tranny rated as 200VA (200 Watt) will only be able to provide approx 0.707 of its rated value. If the amplifier is approx 70% efficient and the power supply is approx 70% efficient then the power taken from the 110V or 240V mains supply will be at least x 2 greater than the power delivered

to the speaker.

However because an amplifier can not be driven with music greater than 1/3 of its capacity with a sine wave the size of a power supply does not need to be greater than its full power rating into a speaker. A very large power supply has better + – V rail supply regulation. The only disadvantage of a large power supply is mass. The mass conventional power supply will be slightly less than 1kg / 100 VA (Watts).

### Power factor

Power factor is the result of Volts x Amperes = Watts. This may appear obvious but when AC is converted to DC to make a + – V supply, power factor requires attention and further understanding. We shall begin with a simple battery and light bulb.

1V5 x 0A = 0 Watt. Open circuit across the battery. maximum V but 0 Amperes.

1V5 x 1A = 1.5 Watt. With a light bulb across the battery

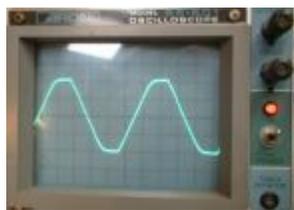
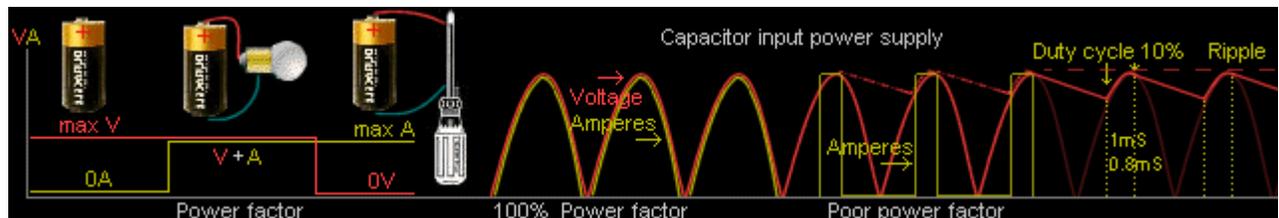
0V x ∞A = 0 Watt. With a screw driver across the battery 0V but maximum Amperes.

Volts without Amperes, and Amperes without Volts, results in 0 Watts. The electronic description is, Volts and Amperes are out of phase with each other. The degree to which V and A are out of phase, results in a decrease of power Watts. Power factor phase is an in depth study in electronics. Hopefully the simplified explanation on this page is sufficient to obtain a basic understanding.

1V5 x 0A = 0 Watt. V A is -90deg out of phase.

1V5 x 1A = 1.5 Watt. V A is in phase.

0V x ∞A = 0 Watt. V A is +90deg out of phase.

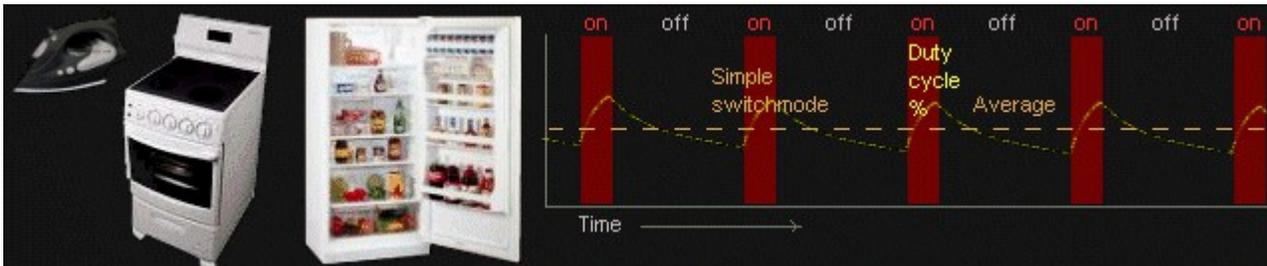


In a 'Capacitor Input' power supply the current Amperes from the secondary of the mains tranny through the bridge rectifier is only charging the electrolytic capacitor for 10% of the time. Therefore Amperes and Voltage are only in phase for 10% of the time during the whole sine wave cycle. This 10% Power factor is reflected back through the primary winding to the 110V 240V AC mains supply.

The oscilloscope in the right pic shows flattened tops on the mains supply system. This is the result of vast numbers of 'Capacitor input' power supplies of all the electronic equipment on the AC mains supply. Taking current from only 10% of the mains supply sine wave instead of taking current continuously across the sine wave has created large mains power regulation problems throughout the world.

## Switch mode power supply

Many household white goods, refrigerators, steam irons and electric ovens are simply controlled by being switched on and off. As the refrigerator warms up it is switched on until it cools down, then it is switched off again and so on. Electric ovens and steam irons function in reverse way. The time it is switched on compared to the time it is switched off is the Duty cycle. Duty cycle is measured as % of on time compared to off time. A refrigerator with a permanently open door would have a Duty cycle of 100%. With the door closed the Duty cycle may be only 10% depending on the quality of insulation. A refrigerator may be on for 5 minutes and off for 50 minutes and so on to maintain cold temperature. The longer the time span (clock speed) of the duty cycle the greater will be the temperature difference between on and off. The fastest time a refrigerator may be able to turn on and off would be approx 1 minute. Therefore it would have a clock speed of 1 minute. If the refrigerator had a higher clock speed of 1 second (1Hz) so it could be turned on for 1 second and off for 10 seconds maintaining the same Duty cycle of 10% the temperature would be controlled more evenly.



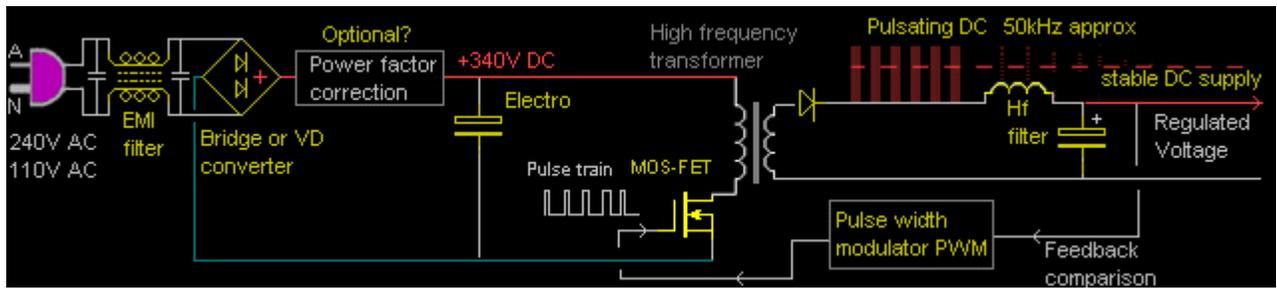
With a conventional 'Capacitor Input' power supply the mains frequency of 50Hz or 60Hz can be described as the clock speed. A 50Hz or 60Hz clock speed is very slow and the reason for the large ripple in the supply rails. If the mains frequency was 1kHz then the ripple on the supply rails would be virtually non-existent. Switch mode power supplies in computers and most home entertainment systems DVDs and plug packs etc have clock speeds of 25kHz to 100kHz. A computer switch mode power supply does not use a large heavy mains transformer. The advantage of a switch mode power supply is reduced mass, but its circuitry is extremely complex and virtually un-serviceable.

The mains Voltage is fed into through an EMI Electro magnetic Interference filter into a Bridge or VD Voltage Doubler rectifier and converted directly to DC. Most switch mode supplies are internally designed to function between 280V DC to 340V DC.

$240\text{V AC mains} \times 1.414 = 340\text{V DC}$  with Bridge rectification.

$110\text{V AC mains} \times 1.414 \times 2 = 310\text{V DC}$  with Voltage Doubler rectification.

The EMI filter stops Rf (radio frequency) switching noise generated by the switch mode power supply from getting back out into the 110V 240V mains system effecting other equipment connected to the mains. There are strict international regulations governing the maximum Rf radio frequency noise that electronic products must comply within. This regulation is named C-tic.



- (1) The mains rectified 300V DC approx is switched on and off through a transformer. The MOS-FET act as a switch. The switching speed can be between 25kHz and 100kHz. The PWM (pulse width modulator) controls the duty cycle of the switching frequency which is similar to a square wave that has variable width. The transformer is specifically designed for switching frequencies of approx 50kHz. High frequency transformers are very small and therefore have little mass.
- (2) The secondary of the transformer reduces the switched AC pulses to a lower voltage, plus isolating the mains to ensure electrical safety. The diode then converts the secondary pulses to the required polarity. The secondary Electrolytic capacitor smoothes out the 50kHz DC pulses to an almost perfectly smooth DC supply.
- (3) The secondary Voltage is fed back into a comparator within the pulse width modulator (PWM). The PWM then automatically adjusts the width (Duty cycle) of the turn on and off pulses to the power MOS-FET which acts as the high frequency switch, controlling the current Amperes through the primary of the high frequency transformer, which in turn corrects the voltage at the secondary to insure that the output DC Voltage is exact.

The high switching frequency gives the switch mode power supply a unique advantage of providing a near perfect ripple free smooth and regulated DC supply Voltage that does not decrease in Voltage as extra current is required.

However switch mode power supplies are extremely complex. The high switching frequency technology is in the Rf radio frequency band. This causes major headaches to technically manage containing Rf radiation from the printed circuit tracks and connecting wires within the circuit getting to the outside world and causing Rf noise contamination in radio and TV. Also the circuit operation is extremely fragile, and if any fault occurs (no matter how small) it contaminates almost every sector of the circuit.

Switch mode power supplies are non user serviceable, failures are simply dealt with by trashing the power supply and purchasing a new one. Attempting to service a switch mode power supply is virtually out of the question, except for fanatics who have infinite time. The technology is so complex and fragile that reliable operation after 7 years is not always expected but considered a blessing. As the world converts to using switch mode power supplies and PWM Class D audio amplifiers, they will eventually end up as the dominant mass of rubbish land fills.

[www.hardwaresecrets.com/ Switching Power Supplies](http://www.hardwaresecrets.com/Switching Power Supplies)  
[www.sound.westhost.com / power-supplies](http://www.sound.westhost.com/power-supplies)

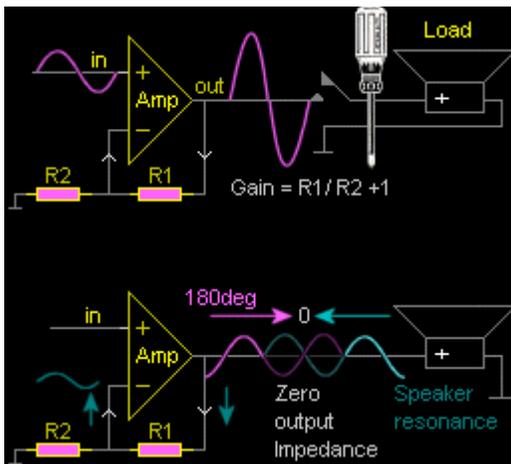
## Solid State V Valves

This page consists of an explanation of Voltage-drive and Current-drive, which enables us understand why Solid-state and Valve amps sound different. When solid-state amps were introduced there was

little interest taken in how amplifiers and speakers interact. Manufacturers of amplifiers and speakers have little in common. It serves the interest of marketing to separate speaker systems and amplifiers into different categories and brand names enabling retailers to have greater control in how customers are influenced and profits made. This imposed marketing separation also suits those who attain notoriety as [Thiel Small](#) loudspeaker design experts. Academic mis-application of Thiel Small parameters has resulted in a bewildering convolution of speakers designs many of which perform poorly. This is also the reason for the traditional marketing hostility against integrated active sound systems, but this is starting to change.

The complimentary page (Valve / Solid-state) in the Valve amps chapter, approaches the difference between these amps from the perspective of the physics of a valve amp, whereas this page describes the difference from a technical perspective of a solid-state amp.

**Comparator** The first stage of the amplifier is called a comparator, and has 2 inputs. The +non-inverting input is connected to the incoming signal. The -inverting input is connected to the output of the amplifier through R1 (negative feedback). The comparator compares the 2 inputs, and forces the amplifier output to match the incoming signal, except for increased size. The way the comparator is set up defines the operation of the amplifier as voltage-drive or current-drive.

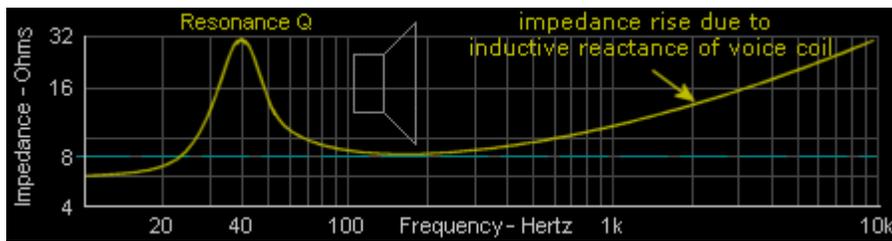


**Voltage drive** is the established conventional method of connecting R1 directly to the output of the amplifier. The gain of the amplifier is controlled independently by the ratio of R1 and R2. Anything (including speakers) connected to the output of an amplifier has no effect on amplifier gain. It makes no difference to the gain of the amplifier if there is a load, or no load, or what the load is. The output voltage will be the result of the ratio of R1 and R2. If a screwdriver is placed across the output, the amplifier will attempt to provide the correct output voltage as a function of R1 and R2. If the short circuit protection in the amplifier does not activate quickly the amplifier will self destruct.

**Zero output Impedance** A voltage drive amp gives the appearance to the speaker (load) that it has a short circuit is across it. Any electrical signal AC or DC that is externally applied to the output of an the amplifier is fed back through R1 to the -inverting input, which is amplified in the reverse direction (180deg) and immediately cancels out (shorts out) the signal form the external source. The voice coil of speaker also generates electricity as it moves in and out. The voice coil generates maximum electricity at the speakers Resonance frequency, described as Back EMF (electro motive force). Any electrical signal from the speaker is fed back through R1 to the -inverting input, which is amplified in the reverse direction (180deg) and immediately shorts out (damps) the resonance of the speaker. Zero output Impedance is also described as 100% Damping factor.

**Speaker Impedance  $\Omega$**  Impedance is resistance R that varies with frequency and this includes all cone speakers. Speaker Impedance is traditionally symbolised with the Greek letter Omega  $\Omega$  to represent

its Resistance R. Speaker Impedance can be written as  $8\Omega$  or  $8R$ . The DC resistance of the voice coil in most  $8\Omega$  speakers is approx  $6R$ . Most cone speakers are  $8R$  between  $400\text{Hz}$  to  $600\text{Hz}$  only.

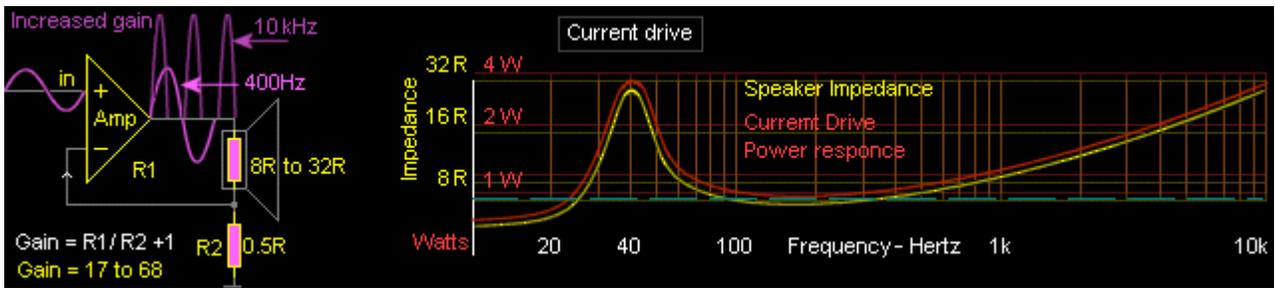


At fundamental resonance (Fs) the speaker Impedance will rise steeply to  $30R - 40R$  as in the above pic. Above  $600\text{Hz}$  the Impedance of a voice coil in most cone speakers will begin to rise due to its inductive reactance and may reach  $20R$  to  $60R$  at  $10\text{kHz}$ . The Impedance of a speaker varies by magnitudes  $500\%$  approx across the frequency range. Some cone speakers have an internal magnetic shorting ring which attempts to minimise Impedance variation, but at the cost of reducing efficiency.

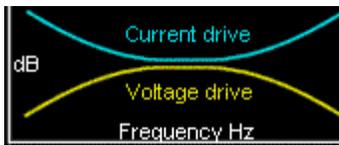


Power into the speaker is dependant on its Impedance at any particular frequency. As the speaker Impedance increases, power to the speaker decreases. In the above pic the speaker Impedance at Resonance is  $32R$  therefore power decreases to  $25\%$ . Only between  $200\text{Hz}$  to  $600\text{Hz}$  is the speaker  $8R$ . At  $10\text{kHz}$  the speaker Impedance is  $32R$  and the power decreases to  $25\%$ . Across the frequency spectrum the average impedance of a speaker is above  $8R$ . Due to the changing Impedance of the voice coil, a cone speaker receives maximum power at mid frequencies but minimum power at bass and high frequencies.

Current drive and Constant current. is technical term to describe the inverse of Voltage drive. The ratio of  $R1$  and  $R2$  controls the gain of the amplifier  $\text{Gain} = R1 / R2 + 1$  as previously described.  $R1$  and  $R2$  are in series to ground and also act as a load to the amplifier. In current drive  $R1$  is the  $8\Omega$  speaker  $R2 = 0.5R$  the gain of the amplifier is  $17$  ( $8\Omega / 0.5R + 1 = 17$ ). However the speaker Impedance varies with frequency, and as the Impedance of the speaker increases so does the gain of the amplifier. At Resonance the speaker is  $32R$  gain and power is increased  $\times 4$ . Between  $200\text{Hz}$  and  $600\text{Hz}$  the speaker Impedance will be  $8R$  gain will be normal. At  $10\text{kHz}$  the Impedance of the speaker is  $32R$  gain and power will be increased  $\times 4$ . Also any electrical signal AC or DC that is externally applied to the output of an the amplifier from the speaker (load) as back EMF is not fed back through  $R1$  to the amplifiers -inverting input. Therefore the output Impedance of the amplifier is Infinite, Damping factor is zero.



Current drive comparison If the speaker Impedance remained a constant 8R over the frequency spectrum there would be little to zero auditory difference between Voltage-drive and Current-drive. All cone speakers vary Impedance over the frequency spectrum (approx 500%) and the auditory difference between Voltage-drive and Current-drive is very noticeable to extreme. With most cone speakers, Voltage-drive appears to sound flat and dead, the bass and high frequencies appear reduced, as if a blanket has been placed over the speaker.



1. Voltage-drive power is inversely proportional to speaker  $\Omega$  therefore power decreases as the speaker Impedance rises.
- Current-drive power is directly proportional to speaker Impedance therefore power increases as the speaker Impedance rises.

In Current-drive the bass and hi-frequencies appear to spring to life with clarity and detail. If given the opportunity for comparison, every human with ears regardless of age, sex, creed or color will hear the difference and will immediately choose Current-drive as the preferred listening option, if the option was available. Every musician (guitar and bass player) instantly hears the brighter response of Current-drive and will immediately choose it, if given the opportunity to compare.

The previous description is by A B comparison only. It is possible for speaker to be designed to give a flat response from a conventional solid-state amp in Voltage-drive. The bass and high frequencies becoming exaggerated when driven by an amplifier in Current-drive.

Valve amplifiers naturally function in quasi Current-drive and have a higher audiophile musicality status. It is virtually impossible to make a Valve amplifier function in 100% Voltage-drive in the same way as a solid state amplifier. Many manufacturers of musical instrument amplifiers (Fender, Marshall etc) stayed with Valve technology for this reason.

Why is this fundamental difference between Voltage and Current drive rarely understood or spoken about by the majority of self proclaimed audio experts, text books, web sites or forums. Rarely is there any connection to this description as being the primary cause for the difference between how Valve and solid state amps sound. Almost all explanations about differences between solid state and Valve amps is based on romanticised subjective twaddle. Why was Voltage drive originally chosen as the convention for solid state amplifiers ?

Questions of Voltage and Current drive

- 1 In Voltage-drive the amplifiers specifications of flat frequency response, gain and internal distortion

figurers are not influenced by the speaker. Except for total power all amplifiers in Voltage drive behave identically. Resonant peaks within the speaker system are damped and not amplified. Tweeters and compression drivers preform better when driven with an amp in Voltage drive and not current drive. Voltage drive enables speaker system designers to have independence with predictability and simplicity. This enables ported boxes to be easily designed and allows high order complex passive crossovers, which are highly reactive, requiring maximum damping from the amplifier to work correctly. Because all amplifiers are made in Voltage drive it does not need to be mentioned or explained. Amplifiers and speakers are marketed independently of each other, therefore it is simply pragmatic.

2 In current-drive the amplifiers frequency response, gain and distortion figurers are dependant on the speaker system. The amplifier and speaker are now interactive and complex to understand. Resonant peaks within the speaker system are not damped and therefore amplified. Current drive is not suited for most applications of ported boxes and complex passive crossovers. Current drive as with most valve amps is best suited to sealed critically damped speaker cabinet designs, with constant impedance passive crossovers, and active systems where each amplifier is matched to its speaker. Early passive crossovers were mostly second order, and critically aligned to avoid resonant effects. Some earlier speakers had copper caped pole pieces, which helped damp impedance variations. Valve amplifiers, which naturally function in quasi current drive, are sensitive to resonances in crossovers and impedance variations. Therefore a flat impedance was synonymous with a flat frequency response.

3 The technical differences between Voltage-drive and Current-drive is neither right or wrong but the result of application management. However the failure for not allowing solid state amps to be optionally available with switchable Current-drive management can be challenged in reference to technical educational attitudes attaining to moral righteousness, similar to religious beliefs.

#### **Technical reference PDF download**

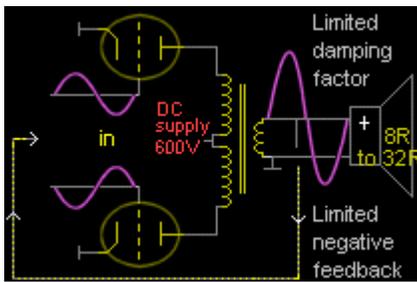
[Distortion Reduction in Moving-Coil Loudspeaker Systems Using Current-Drive Technology. PDF](#)  
by P. G. L. Mills and M. O. J. Hawksford.

#### **Amplifier history overview**

Untill the 1970s the majority of audiophile Valve amplifiers were approx 15 Watts per channel. Only a small % of audiophiles had 30 to 60 Watt amplifiers.  $16\Omega$  was the standard Impedance for speakers and was later changed to  $8\Omega$  to suit solid state amplifiers. Speakers were made as efficient as engineering would allow. The average speaker was 6dB to 10dB more efficient than the majority of speakers today. The gap tolerance was less than a bees dick to obtain the highest BL flux density possible without the voice coil touching the pole piece. Paper fibres for the cones were selected to achieve maximum stiffness with minimum mass. The external compliance and spiders were as soft as could possibly be achieved while still holding the cone centred. A speaker needing to handle power above 50 Watts was rarely required. 100 Watt Marshall guitar amps had 1 quad box consisting of 4 x 25 Watt Celestion 12in speakers.

High Q speakers Speaker boxes were made as large as possible to achieve maximum efficiency at bass frequencies. High efficiency speakers in very large boxes also means high Q (Fundamental resonance) which was tuned to achieve the maximum efficiency at the lowest musical bass frequencies.

Q is quality of resonance (energy stored / energy lost). Many speaker systems had an excessively high Q of approx 2 to 4. The average audiophile bass speaker system of today has a Q of approx 0.5 to 0.7. Excessively high Q causes the bass notes to sound annoyingly resonant. Excessive bass resonance is controlled by reducing the amplifiers output Impedance, described as Damping factor.



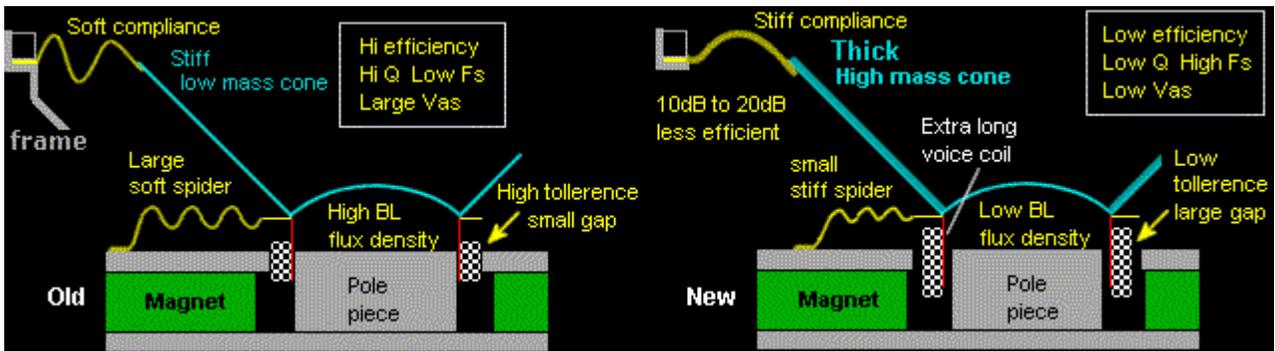
**Damping factor** Valve amps naturally function in quasi Current-drive having a high output Impedance with minimal damping. Negative feedback from the speaker terminals is fed back to the Cathode of the first input valve causes the Valve amplifier to shift slightly toward Voltage-drive reducing its output Impedance and introducing Damping factor to control the excessive Q Resonance of the speaker. Valve amps are naturally noisy (hum) and negative feedback quietness the amplifier and reduces measured distortion.

**Negative feedback** Academic engineers became obsessed with finding ways to increase negative feedback in Valve amplifiers to achieve a greater damping factor and lowest distortion figures. But Valve amplifiers will only tolerate a minimum amount of negative feedback before the amp is caused to oscillate, hereby restricting the amount of Damping factor that can be achieved. When solid state amplifiers arrived with the ability to have 100% Damping factor, interest in Valve technology was discarded without further thought.

Pre 1970s many if not most large manufacturing companies maintained a strong class demarcation of management and engineers against trades and production workers. This demarcation was also reflected in music. Management and academic class was identified with Classical and Popular conservative music similar to Frank Sinatra or Julie Andrews “The sound of music” etc mostly mid range with minimal bass energy. The working class identified with music of the devil, Sex drugs Rock’n Roll, drums bass guitar, Jazz, Blues with intense bass energy.

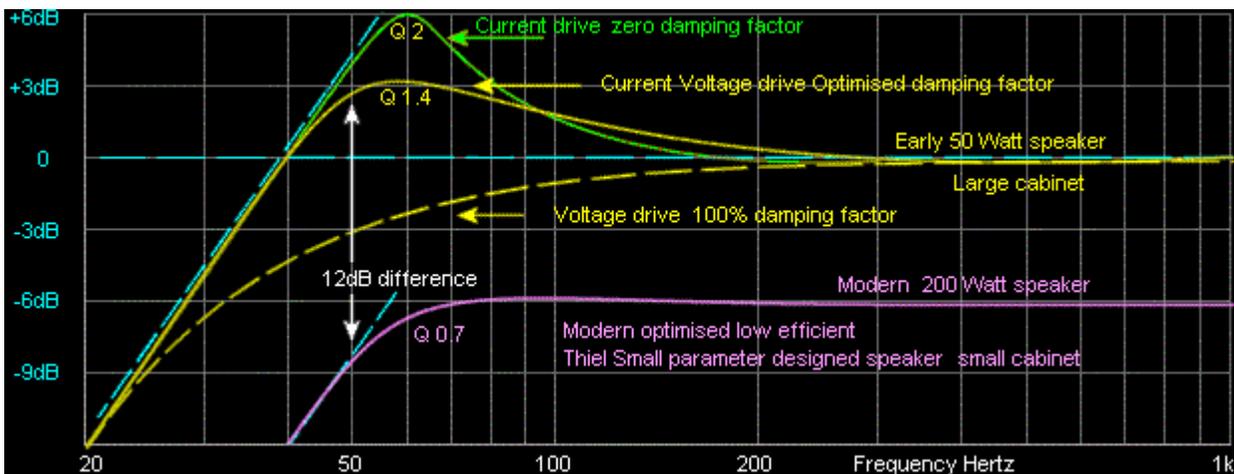
The rock recording industry noticed that the new solid state amps caused most speakers to sound flat and lifeless. Without any standards in place, recordings had and have extensive EQ applied to boost the bass and hi-frequencies to compensate. Many guitar and bass amplifier designers also noticed that the new solid state amplifiers (with 100% Damping factor) had a suppressive effect on the musicality of a speakers performance and stayed with Valve technology. But academic engineers of HiFi amps were blindly obsessed with the almost perfect zero distortion figures and 100% damping factor solid state amps could achieve. The concept of introducing variable current drive to manage speaker Damping was negated.

**Thiel Small parameters** [www.wikipedia.org / Thiel Small](http://www.wikipedia.org/Thiel%20Small) Thiel Small parameters for speaker design were developed in the early 1970s, but little interest was taken until computers were available in the 1980s. Each parameter of a speaker is compiled into a singular complex algorithm that enables a speakers performance to be accurately defined. The parameters assume the speaker is being driven with zero output Impedance (100% Damping factor) and will also optimise the performance of a speaker in a ported cabinet. Thiel Small parameters provided a new way to manage speaker design and re-obtain the lost bass energy caused by solid state amps in Voltage-drive, but mostly at the cost of reduced efficiency.



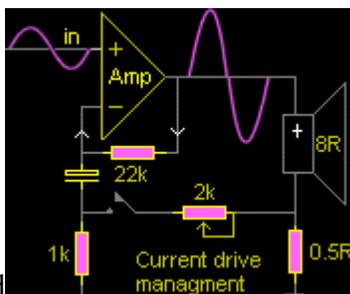
High power solid state amps were becoming available at low cost and the latest chemical engineering of epoxy compounds enabled speakers to be designed that were capable of 100s of Watts. With the applied use of Thiel Small design parameters it was seen that by increasing the mass of the cone, and decreasing the flux density BL by increasing the gap tolerance, coupled with a stiff suspension to enable speaker manufacturers to assemble speakers with wider tolerances enabling lower cost mechanised production the end result is a speaker that is less efficient compared to previous times, but with cheap high power amps the result can be accurately calculated to give the low frequency performance required.

The below pic shows a generalised graph of an early design 50 Watt high efficient speaker (yellow) with a low mass cone and soft compliance in a typical large sealed cabinet. With zero damping the resonance Q2 the speaker is extreme +6dB. With a set amount of damping provided by the Valve amplifiers negative feedback the resonant Q1.4 is reduced to +3dB at 60Hz, and is flat again at 40Hz and only down -3dB at 34Hz. The music will sound deep and warm with extra responsive bass energy.



When the same high efficient speaker (yellow dash) is driven by a solid state amp (100% Damping factor) the speakers resonance is completely damped and the bass response begins to roll off from above 100Hz. Bass energy at 50Hz reduced -6dB. The music will sound flat and shallow.

With a modern less efficient 200 Watt Thiel Small designed speaker with a heavy cone placed in a small sealed cabinet driven by a solid state amp (100% Damping factor) the response is almost flat down to 60Hz and reduced -6dB at 40Hz. The modern speaker has an academically flatter response



and the music sounds as we hear it to be.

By comparison many earlier speakers in larger cabinets were 6dB to 12dB more efficient than most modern speaker systems today. Many modern speakers now have to be driven with x 4 power to sound as loud. High powered solid state amplifiers are now cheap and wasting power is symbolic of modern times. History has been forgotten and there is little interest in reflecting on what was not understood during the transition from valve to solid state technology.

All that was required was a simple addition to the gain management of solid state amps (in the right pic) to enable Voltage to Current drive adjustment so the Damping factor of speakers could be effectively managed.

The historical purpose of organised religions was to manage society to un-questioning obey authority and live within a moral code. Over thousands of years religious conditioning has become so entrenched within human consciousness that it no longer requires us to worship Deity's or repeat ancient text as a guide for obedience. Marketing and consumerism is the religion of modern times.

Modern man is so conditioned by his past, that he will blindly accept anything marketed to him, including identifying with brand names, model numbers and celebrities. He also infallibly believes in the propagandised ideal of Democratic consumerism as a disguise for self-interest. Above all, western man remains a contradiction to himself as he will always choose to be right, rather than choose to be happy.

Voltage drive may not always give a pleasing sound but it provides superior technical measured performance of zero output Impedance (100% damping factor) and the primary reason it was unquestionably chosen, reflecting one of many examples from our historical religious conditioning now expressed as a self-righteous belief in an imposed technical morality.

[www.sound.westhost.com](http://www.sound.westhost.com) is a parallel site by Rod Elliott that has a detailed description of amplifier design, and is essential reading for those who require an advanced mathematical and electronic design explanation of the amplifier principles described on this page.