

Last updated 19 November 2005

## Disclaimer

The information presented below is just that, *information*. **Use it at your own risk.** *You* (the reader) are responsible for anything foolish that you might do as a result of reading this article. *You* assume complete and total responsibility for your actions! The information presented below is believed to be technically correct (however mistakes occasionally are made, this being the "real world"). In simple terms, if you do something foolish (as a result of reading this article), *don't blame me!* This statement is necessary because of (in the US anyway) too many lawsuit happy people (and certain lawyers who flock to represent such people!)

# Audio Power Amplifier Fundamentals

---

## Note:

If you are reading this article you may also be interested in my article [Audio Power Amplifier Power Rating Mysteries Explained](#) (it has a lot more detail regarding power ratings of amplifiers).

## Introduction

The term *amplifier* is very generic. In general, the purpose of an amplifier is to take an input signal and make it stronger (or in more technically correct terms, increase its *amplitude*). Amplifiers find application in all kinds of electronic devices designed to perform any number of functions. There are many different types of amplifiers, each with a specific purpose in mind. For example, a radio transmitter uses an RF Amplifier (RF stands for Radio Frequency); such an amplifier is designed to amplify a signal so that it may drive an antenna. This article will focus on *audio power amplifiers*. Audio power amplifiers are those amplifiers which are designed to drive loudspeakers. Specifically, this discussion will focus on audio power amplifiers intended for DJ and sound reinforcement use. Much of the material presented also applies to amplifiers intended for home stereo system use.

## Basics

The purpose of a power amplifier, in very simple terms, is to take a signal from a source device (in a DJ system the signal typically comes from a preamplifier or signal processor) and make it suitable for driving a loudspeaker. Ideally, the **ONLY** thing different between the input signal and the output signal is

the *strength* of the signal. In mathematical terms, if the input signal is denoted as  $S$ , the output of a *perfect* amplifier is  $X*S$ , where  $X$  is a *constant* (a fixed number). The "\*" symbol means "multiplied by".

This being the real world, no amplifier does exactly the ideal, but many do a very good job if they are operated within their advertised power ratings. The output of *all* amplifiers contain additional signal components that are not present in the input signal; these additional (and unwanted) characteristics may be lumped together and are generally known as *distortion*. There are many types of distortion; however the two most common types are known as *harmonic distortion* and *intermodulation distortion*. In addition to the "garbage" traditionally known as distortion, all amplifiers generate a certain amount of *noise* (this can be heard as a background "hiss" when no music is playing). More on these later.

All power amplifiers have a *power rating*, the units of power are called *watts*. The power rating of an amplifier may be stated for various *load impedances*; the units for load impedance are *ohms*. The most common load impedances are 8 ohms, 4 ohms, and 2 ohms (if you have an old vacuum tube amplifier the load impedances are more likely to be 32 ohms, 16 ohms, 8 ohms, and maybe 4 ohms). The power output of a modern amplifier is usually *higher* when *lower* impedance loads (speakers) are used (but as we shall see later this is not necessarily better).

In the early days, power amplifiers used devices called *vacuum tubes* (referred to simply as "tubes" from here on). Tubes are seldom used in amplifiers intended for DJ use (however tube amplifiers have a loyal following with musicians and hi-fi enthusiasts). Modern amplifiers almost always use transistors (instead of tubes); in the late 60's and early 70's, the term "solid state" was used (and often engraved on the front panel as a "buzz word"). The signal path in a tube amplifier undergoes similar processing as the signal in a transistor amp, however the devices and voltages are quite different. Tubes are generally "high voltage low current" devices, where transistors are the opposite ("low voltage high current"). Tube amplifiers are generally not very efficient and tend to generate a lot of heat. One of the biggest differences between a tube amplifier and a transistor amplifier is that an *audio output transformer* is almost always required in a tube amplifier (this is because the output impedance of a tube circuit is far too high to properly interface directly to a loudspeaker). High quality audio output transformers are difficult to design, and tend to be large, heavy, and expensive. Transistor amplifiers have numerous practical advantages as compared with tube amplifiers: they tend to be more efficient, smaller, more rugged (physically), no audio output transformer is required, and transistors do not require periodic replacement (unless you continually abuse them). Contrary to what many people believe, a well designed tube amplifier can have excellent sound (many high end hi-fi enthusiasts swear by them). Some people claim that tube amplifiers have their own particular "sound". This "sound" is a result of the way tubes behave when approaching their output limits (clipping). The onset of output overload in a tube amplifier tends to be much more gradual than that of a transistor amplifier. A few big advantages that tube amplifiers have were necessarily given up when amplifiers went to transistors. First, tubes can withstand electrical abuse that would leave even the most robust transistor completely blown. Also, tube amplifiers use an output transformer to interface to the speaker; such a device provides an excellent buffer (protection to the speaker) in the case of internal malfunction. Modern amplifiers (with no output transformer) occasionally fail in a way that connects the full DC supply voltage to the speaker. If the amplifier does not have adequate protection circuitry built in, the result is often a melted woofer voice coil.

Power amplifiers get the necessary energy for amplification of input signals from the AC wall outlet to which they are plugged into. If you had a *perfect* amplifier, all of the energy it took from the AC outlet would be converted to useful output (to the speakers). However, in the real world no amplifier is 100% efficient, so some of the energy from the wall outlet is wasted. The vast majority of energy wasted by an amplifier shows up in the form of *heat*. Heat is one of the biggest enemies to electronic equipment, so it is important to ensure adequate air flow around equipment (especially so for those units using convection cooling). Most amplifiers in the 200 watt per channel range (and up) have forced air cooling (via fans) in order to prevent excessive heat buildup.

Many amplifiers have a number of features to help monitor the status of the amplifier and also to protect speakers (and the amplifier itself) in the event of an overload condition. Some features include power meters, clipping indicators, thermal overload shutdown, over current protection, etc. Features vary from manufacturer to manufacturer. In addition, there are many variations in how protection circuits are implemented and how much "safety margin" they allow. For example, I tested the clipping indicator on one particular amplifier. The clipping indicator did not come on until there was a substantial amount of clipping actually occurring (as viewed on an oscilloscope). In this case, I did not notice a significant degradation of the sound quality despite the clipping. The manufacturer in this case chose to "allow a little more volume" before actually lighting up the warning light.

Power amplifiers intended for DJ use have power output ratings starting from around 75 watts per channel to over 1000 watts per channel. *However, keep in mind that MORE POWER DOES NOT NECESSARILY MEAN A SUPERIOR AMP OR BETTER SOUND!* A well designed amplifier in the 200 watt per channel class may be better investment than a marginally designed 500 watt per channel unit.

## What are the functional blocks of an amplifier?

All power amplifiers have a *power supply*, an *input stage*, and an *output stage*. Many amplifiers have various protection features which fall into a category I refer to as *housekeeping*.

**Power Supply:** The primary purpose of a power supply in a power amplifier is to take the 120 VAC power from the outlet and convert it to a DC voltage (VAC is an abbreviation for *Volts Alternating Current*, and DC is an abbreviation for *Direct Current*). Conversion from AC to DC is necessary because the semiconductor devices (transistors, FETs, MOSFETs, etc.) used inside the equipment require this type of voltage. (By the way, FET stands for Field Effect Transistor, and MOSFET stands for Metal Oxide Semiconductor Field Effect Transistor). Many different types of power supplies are used in power amplifiers, but in the end they all basically aim to generate DC voltage for the transistor circuits of the unit. The very best of amplifiers have two totally independent power supplies (they do share a common AC power cord though). Such amplifiers are really just two monaural amplifiers mounted in a single case.

**Input stage:** The general purpose of the input stage of a power amplifier (sometimes called the "front end") is to receive and prepare the input signals for "amplification" by the output stage. Most

professional quality amplifiers have various input connectors; typically they will have XLR inputs, "quarter inch" inputs, and sometimes a simple terminal strip input (although these tend to be found on amplifiers intended primarily for public address systems). XLR and most quarter inch inputs are *balanced* inputs (as compared to *single ended* inputs. Balanced inputs are much preferred over single ended inputs when interconnection cables are long and/or subject to noisy electrical environments because they provide very good *noise rejection*. The input stage also contains things like input level controls. Some amplifiers have facilities for "plug in" modules (such as filters); these too are grouped into the input stage.

**Output stage:** The output stage of an amplifier is the portion which actually converts the weak input signal into a much more powerful "replica" which is capable of driving high power to a speaker. This portion of the amplifier typically uses a number of "power transistors" (or MOSFETs) and is also responsible for generating the most heat in the unit (unless the amplifier happens to have a very bad power supply design). The output stage of an amplifier interfaces to the speakers.

## What are Amplifier Classes?

The **Class** of an amplifier refers to the design of the circuitry within the amp. There are many classes used for audio amps. The following is brief description of some of the more common amplifier classes you may have heard of.

- **Class A:** Class A amplifiers have very low distortion (lowest distortion occurs when the volume is low) however they are very inefficient and are rarely used for high power designs. The distortion is low because the transistors in the amp are biased such that they are half "on" when the amp is idling. As a result, a lot of power is dissipated even when the amp has no music playing! Class A amps are often used for "signal" level circuits (where power is small) because they maintain low distortion. Distortion for class A amps increases as the signal approaches clipping, as the signal is reaching the limits of voltage swing for the circuit. Also, some class A amps have speakers connected via capacitive coupling.
- **Class B:** Class B amplifiers are used in low cost, low quality designs. Class B amplifiers are a lot more efficient than class A amps, however they suffer from bad distortion when the signal level is low (the distortion is called "crossover distortion"). Class B is used most often where economy of design is needed. Before the advent of IC amplifiers, class B amplifiers were common in clock radio circuits, pocket transistor radios, or other applications where quality of sound is not that critical.
- **Class AB:** Class AB is probably the most common amplifier class for home stereo and similar amplifiers. Class AB amps combine the good points of class A and B amps. They have the good efficiency of class B amps and distortion that is a lot closer to a class A amp. With such amplifiers, distortion is worst when the signal is low, and lowest when the signal is just reaching the point of clipping. Class AB amps (like class B) use pairs of transistors, both of them being biased slightly ON so that the crossover distortion (associated with Class B amps) is largely eliminated.
- **Class C:** Class C amps are never used for audio circuits. They are commonly used in RF circuits.

Class C amplifiers operate the output transistor in a state that results in tremendous distortion (it would be totally unsuitable for audio reproduction). However, the RF circuits where Class C amps are used employ filtering so that the final signal is completely acceptable. Class C amps are quite efficient.

- **Class D:** The concept of a Class D amp has been around for a long time, however only fairly recently have they become commonly used in consumer applications. Due to improvements in the speed, power capacity and efficiency of modern semiconductor devices, applications using Class D amps have become affordable for the common person. Class D amplifiers convert the input signal to a Pulse Width Modulation (PWM) type signal (at a very high frequency) and then this signal is filtered and sent to speakers (this is a VERY high level idea of how they work)! Class D amps are (today) most often found in car audio subwoofer amplifiers. Class D amplifiers have very good efficiency (due to the fact that the semiconductor devices are ON or OFF in the power stage (resulting in low power dissipation in the device as compared to linear amplifier classes). Due to the high frequencies that are present in the audio signal, Class D amps used for car stereo applications are often limited to subwoofer frequencies, however designs are improving all the time. It will not be too long before a full band class D amp becomes commonplace. Class D amps find use in many other applications besides audio.
- **Other classes:** There are a number of other classes of amplifiers, such as G, H, S, etc. Most of these are variations of the class AB design, however they result in higher efficiency for designs that require very high output levels (500W and up for example). At this time I will not go into the details of all of these other classes at this time. Suffice to note that Class D (among A, B, AB, D, S, G, H classes) is the class that represents a major delta in the way it operates as compared to the other audio amplifier classes. Sometimes the marketplace promotes Class D amplifiers as being "digital". The marketplace tends to toss around the word "digital" a lot, there is no really standardized definition (that I am aware of) that deems an amp "digital". To find out what a vendor means when they use the word digital with regard to an amp requires research into the design of the amp.

## Why do Amplifiers have different power ratings for different "ohms"?

Power amplifiers are typically rated for "8 ohm" and "4 ohm" loads, and some also give ratings for "2 ohm" loads. If you have ever looked at a spec sheet, you probably noticed that the power output of an amplifier is higher when the load impedance (number of ohms) is lower. *Important:* a load with a low number of ohms is a more difficult load than one with a higher number of ohms! (that is, a 4 ohm speaker is harder for an amplifier to drive than an 8 ohm speaker). The performance of an amplifier with low impedance loads is closely related to the capabilities of its *power supply*.

If we had a *perfect* amplifier (*and it was plugged into an outlet that had unlimited current capability*), its output power rating would *double* each time the load impedance was *halved*. For example, let's say the amplifier puts out 200 watts per channel at 8 ohms. At 4 ohms, it would put out 400 watts per channel, at 2 ohms it would put out 800 watts per channel, and at 1 ohm it would put out 1600 watts per channel. For the *perfect* amplifier, one could keep going with this until the load impedance approached zero, at which time the amplifier output would approach infinity! On the other side, if the load impedance was 16 ohms,

the amplifier would put out only 100 watts per channel. In this direction, one could keep raising the load impedance, and the power output would grow smaller and smaller.

The power supply of the perfect amplifier generates a DC voltage that *does not change* no matter how much current is demanded from it. This means that the perfect amplifier can drive an unlimited number of speakers. In the real world, amplifiers have real power supplies which *do* have limits as to how much current they deliver. For such typical amplifiers, the 4 ohm power rating is usually about 50% more than the 8 ohm rating (and if a 2 ohm rating is given, this is maybe 50% more than the 4 ohm rating). Amplifiers with exceptional power supply designs will do better than this, but eventually a limit will be reached (if by nothing else the AC outlet can only deliver so much current!). Lesser designs will "run out of juice" when driving the heavier loads. Stay away from amplifiers that have a 4 ohm rating that is less than 25% greater than the 8 ohm rating!

Amplifiers utilizing exceptional power supply designs will invariably be the more expensive units available, and possibly the (physically) heavier designs. This is because good power supply designs usually require heavier and better (low loss) "magnetics". All power supplies utilize some combination of *transformers, rectifiers, capacitors*, and in the case of so called "digital" amplifiers, *switching* components.

**"Analog" Amplifiers:** ALL amplifiers in use by DJ's today process *analog* input (music) signals. An analog signal is a *continuous wave* signal, a *digital* signal is an analog signal which has been converted to a sequence of *numbers*. *Analog* when spoken in terms of power amplifiers typically refers to the design of the power supply, and most analog amps are those with a straight Class AB design. A so called analog amplifier has a power supply which typically uses a large *power transformer*, a *rectifier circuit*, and large *capacitors*. These three basic devices convert the AC voltage from the outlet to a lower voltage (more suitable for the internal needs of the unit), change it from AC to DC, and filter and store energy. These types of power supplies have been around for *many* years; they are simple and reliable. The downside is that the power transformer is usually large and quite heavy (the transformer *core* utilizes a considerable amount of iron), and the capacitors (a minimum of two are normally used) are also large and bulky.

**"Digital" Amplifiers:** When the term *digital* is associated with a power amplifier, it sometimes refers to the design of the power supply and may that the power supply is of the *switching* type (sometimes referred to as a *DC - DC converter*). The term "digital" is sometime associated with amplifiers of the more exotic classes (class G, H, S, and especially D). Class G, H and S amplifiers use special switching circuits that change the power supply voltage to the output stage on the fly such that higher efficiency is maintained. Class D uses a totally different scheme for amplification (and is the most legitimate class to be termed "digital"). NOTE: A digital amp It *in no way* means that it is inherently better at producing sound from "digital" sources such as CD's and DAT's!!! Most all car stereo amps use a switching power supply. What advantages does a switching power supply offer? For car audio (which runs on a 13.8 VDC power source) there is no way to get high power to speakers without boosting the voltage to higher levels. Switching power supplies are used in some conventional (home or pro audio) amps as well. Switching power supplies use much smaller transformers and capacitors (as compared to conventional

amps), and are therefore considerably smaller and lighter than an equivalent analog power supply. The concepts behind switching power supplies have been known for many years. However, until fairly recently the components necessary for switching power supplies were unable to be produced cheaply enough for consumer use. Advances in transistor technology have made the necessary devices available at a cost which permits their widespread use. (Note: *ALL* of the "super systems" heard in many automobiles today are powered by amplifiers using switching power supplies).

On the minus side, switching power supplies are a great deal more complicated than their analog counterparts. They work basically by first creating a "crude" DC voltage. This crude voltage is applied to a circuit which uses a specially designed high frequency transformer. A control circuit monitors the output voltage of this stage and makes adjustments "on the fly" in order to keep the final DC output voltage as close to the design value as possible. So, the advantages of lighter weight and smaller size come at the expense of increased parts count (which ultimately might translate to less reliability if the parts are of lesser quality). Also, switching power supplies are harder to repair if they fail.

Many "digital" amplifiers also use a "multi-rail" power supply system. Such systems are more complicated than conventional amplifier designs, however they offer considerable improvements in amplifier efficiency. The amplifier selects a "rail voltage" based on the output demands of the amplifier. Higher efficiency is achieved by minimizing the voltage drop across the amplifier's output transistors. Since less of the amplifier's power is wasted as heat, the power supply and transistor heat sinks do not have to be as large as those in a "conventional" design. As before, the theory behind "digital" designs has been known for decades, but until recently components necessary to make unaffordable design were unavailable.

## "Analog" vs. "Digital" ... Which is better?

Many of the amplifiers on the market today are of the "digital" type, using switching power supplies and/or special power supplies that maintain high efficiency at high outputs. Some people believe that "digital" amplifiers are not so good at producing powerful bass notes. While it is true that there probably some marginally designed "digital" amplifiers which *do* have less than ideal bass response, weak bass response is not *anecessity* of digital designs. The dominating factor in performance comes back to the *ability of the power supply to provide adequate current*; a solid design means adequate current is available for loud bass notes and/or difficult speaker loads. In addition, a second important factor is the *adequacy of the AC power outlet*. Two well designed amplifiers (one of each type) operated on an AC outlet which doesn't "sag" (see my article on AC Power) should both provide excellent sound quality. Many of the higher power amplifiers available today are of the "digital" (switching power supply) design. But keep in mind that this does not necessarily make them better or worse. Stay with vendors that have proven track records of reliability and you should have few problems with either type of design.

## Power Ratings

Two amplifiers with the same power rating put out the same power, right? Not necessarily.

Manufacturers vary as to how conservatively they rate their amplifiers. As an example, I measured one particular amplifier, rated at 350 watts/channel, and found it actually was able to put out 450 watts/channel! Manufacturers often understate what their units will actually put out. It would be a bad idea to publish the "absolute maximum power" that the unit could put out, since a margin needs to be allowed to insure that all production units will meet published specs. In addition, a manufacturer may publish a very conservative 8 ohm rating in order to make the 4 ohm rating look better (a really terrible amplifier will put out LESS power into a 4 ohm load!).

Amplifiers are generally rated in watts per channel, at several load impedances, with both channels driven, over a frequency range of usually 20 Hz - 20,000 Hz, at some amount of total harmonic distortion. Most amplifiers will put out slightly *more* (but not a tremendous amount more) power when only a *single* channel is driven. This occurs because the power supply only has to provide power for a single channel, and its DC voltage doesn't sag as much. The exception are amplifiers which use dual independent power supplies (since each of their supplies only has to supply power for one channel anyway).

## What about 2 ohm ratings?

Many of the amplifiers on the market today are touting excellent performance with 2 ohm load impedances. Some also state that "continuous operation" with 2 ohm loads is possible. While such statements are probably true, it is not really a good idea to run under such conditions!

First, a word on speakers is in order (for much more detail see my article on speakers). All speakers have a characteristic known as impedance (measured in *ohms*), with most speakers being either 8 ohms or 4 ohms. *Lower* impedances represent more difficult loads for amplifiers to drive. Two 8 ohm speakers connected in *parallel* will result in a 4 ohm load at the amplifier. And, two 4 ohm speakers (wired in parallel) result in a 2 ohm load. In actuality, speaker impedance can vary *by a factor of 10 or more* over the audio frequency range. When a speaker is said to be 8 ohms, it is understood that this is a *nominal* or *approximate* rating (the same goes for 4 ohm speakers). An 8 ohm speaker could have an impedance as low as 2 or 3 ohms and as high as 50 ohms (impedance is frequency dependent)! Further, a speaker load is not the same as a resistive load, speakers are *reactive* loads. A reactive load is a load that has *inductive* or *capacitive* properties. Depending upon the input signal frequency, speaker loads may be resistive or resistive with an inductive or capacitive component. Without going into a ton of technical explanation, what this means is that speakers are often difficult loads for amplifiers to drive. *Driving difficult speaker loads is where better amplifiers are separated from lesser designs.*

Even though an amplifier may be rated for continuous use at 2ohms, there are several reasons why this is not the best thing to do:

- **Paralleled speaker loads may be lower than you think:** No speakers that DJ's are likely to use have 2ohm ratings. However, a pair of 4 ohm speakers paralleled will yield a nominal 2 ohm rating. As stated before, the actual impedance varies and the minimum impedance may dip

considerably below 2 ohms at certain frequencies. Lower impedance loads mean more losses and more heat dissipation in the amplifier (see next item).

- **Heat Considerations:** Operating an amplifier with a low impedance load increases the heat dissipation of the amplifier (try it if you don't believe it!). This is because low impedance loads require more current, which taxes the amplifier's power supply more severely. More current means more losses (which translates to more heat). Excessive heat is unhealthy for electronic devices and should be avoided.
- **Increased Line Losses:** As the speaker impedance is lowered, more of the audio signal is lost (in the form of heat) in the speaker cables! This can become significant if you run long cables. Speaker wires have resistance (the value depends on the thickness and length of the cable); if the speaker impedance becomes very low the resistance of the speaker wire may no longer be insignificant. To prevent this problem, the cross sectional area of the speaker cable conductor must *double* for each *halving* of speaker load impedance! In other words, running 2 ohm loads means using VERY heavy speaker cables.
- **Damping Factor degradation:** Using super low impedance loads on an amplifier will degrade the system's *damping factor* (discussed in detail below). Degradation of damping factor means that the amplifier will have less "control" over the speaker system, possibly resulting in "boomy" bass response.

So, just because an amplifier has a super powerful 2 ohm rating, don't look for ways to wire up multiple speakers in order to "use" this power! Treat the 2 ohm rating as "headroom" and know that your amp has the ability to more easily handle the most difficult "normal" speaker loads that you are likely to ever encounter. If you need more power, get a second amp. Two medium powered amps are better than one monster (what if your one big amp dies? With two smaller amps at least you can still run!).

## Noise

All amplifiers generate a certain amount of electrical noise. Generally, the more powerful the amplifier, the more noise. If you turn on an amplifier (with the input jacks disconnected) and listen to a speaker you can clearly hear a hissing sound. This pretty much represents the noise floor of the amplifier. For a powerful system, the noise might seem pretty obvious; however when actual music is playing the noise will be totally masked.

All electrical circuits generate a certain amount of noise. Better designs minimize the amount of noise, however no matter how good the design there will always be some. The noise is generated by the movement of electrons in the system and cannot be eliminated (unless you chill your equipment to absolute zero!). The noise floor of an amplifier by itself is usually not obviously audible in a typical room (unless you are standing right next to a speaker). However, the remaining components in a system (preamp, equalizer, processor, etc.) each add in some noise. So, the total system noise (when no music is playing) *might* be objectionable. If this is a serious problem, a device called a *noise gate* can be used. Such a device is essentially a "squelch" which is wired in just before the power amps (or electronic crossover in multi-way systems). The device is basically cuts noise from upstream components when no music is playing. Most noise gates have adjustable controls to set the threshold at which noise cut begins

and also to set the amount of desired noise cut. Most DJ systems probably do not need noise gates unless they are very high powered systems with along signal chain (or noisy components).

The noise floor of an amplifier is relatively constant, meaning it does not increase with increasing output signal (unless the amplifier has a poorly regulated power supply). In other words, the amplifier's noise floor is pretty much the same whether or not music is playing loudly or softly. So, when music is playing softly, the noise will be proportionally larger. When music is playing loudly, the noise is essentially "buried" or masked.

As stated, an amplifier with a poorly regulated power supply can create some additional noise. If the filtering of the power supply is marginal, the "smoothness" of the DC power supply voltage will be degraded when the amplifier is playing loudly. This will result in additional noise being added to the system (generally in the form of 60 Hz products). This type of noise isn't really part of the noise floor. Such noise is often inaudible when music is playing loudly. It can be clearly heard however when playing test tones at levels near the output limit of the amplifier (don't try this unless you are thoroughly familiar with testing practices... blown speakers will otherwise be the result!).

## Distortion

*ALL* amplifiers alter input signals, generally in two ways: they make them stronger (*amplify*) them, and they add characteristics *which did not exist* in the original signal. These undesirable characteristics are lumped together and called *distortion*. Noise can be considered a type of distortion and was discussed in the above section.

Everyone is familiar with gross distortion, the sound quality that results when turning up a radio or boom box to "full blast". An excessive amount of amplifier *clipping* (see section below) results in hideous distortion that would be totally unsatisfactory for a DJ sound system (as well as a listener's ears). However, not all distortion is blatant. In addition, there are several types, two of which will be discussed. Knowing what causes distortion will help you to prevent it from occurring. Knowing how to control distortion is important because excessive distortion can be detrimental to speaker systems (and your reputation).

**Harmonic distortion:** One common type of distortion is *harmonic distortion*. *Harmonics* of a signal are signals which are related to the original (or *fundamental*) by an integer (non decimal) number. A *pure tone* signal has no harmonics; it consists of only one single frequency. If 100 Hz pure tone signal was applied to the input of an amplifier, we would (upon measurement with special test equipment) find that the output signal of the amplifier was no longer pure. Careful measurements would likely show that several "new" frequencies have appeared. These new frequencies are almost certainly to be integer multiples of the original tone; they are the harmonics of the original signal. In the case of a 100 Hz input tone, we might expect to find tones at 200, 300, 400, 500 (etc.) Hz. We would also probably notice that the *odd* harmonics are much stronger than the *even* harmonics (we will not go into the reasons why in this article). In a good amplifier, the harmonics will be *much* weaker than the original tone. By much weaker, we

mean on the order of a thousand times for decent amplifiers.

All amplifiers are generally rated for *Total Harmonic Distortion* (or *THD*), usually at full power output over a given frequency band with a particular load. Good values are anything less than 0.5 % THD. When an amplifier is measured for THD, a pure tone is applied to the input and the output is measured with special test equipment. The energy of the pure tone is measured, and the energy of the harmonics is measured. Those two values are compared, and a THD rating is calculated. A THD rating of 1% means that the *total* energy of all the harmonics *combined* is one one-hundredth of the energy in the *fundamental*.

Harmonic distortion (although certainly undesirable) is one of the more tolerable types of distortion as long as it is kept reasonably low. Distortion levels of 10% may be very tolerable with *music* so long as the 10% level is only "occasional" (10% THD on a *pure tone* can *easily* be heard by the human ear... but who listens to pure tones?). The reason that a seemingly high value of THD is acceptable for music is partially because many sounds in nature are rich in harmonics. Also, most decent cassette decks (which most people agree sound pretty good) have THD (off the tape that is) of *several* percent. Worse, even *good* speakers can have THD up to 10%, especially at low frequencies! All in all, the human ear can tolerate a fair amount of THD before it becomes objectionable.

**Do two amplifiers with identical THD ratings sound the same, everything else being equal?** *Not necessarily* (but differences will be subtle). The reason is that the THD specification states nothing about *where* the harmonics are in the frequency band. For example one amplifier could have a dominant harmonic at one frequency and a second amplifier could have a dominant harmonic at a very different frequency. Or, one amplifier could have a few "big" harmonics while a second has many weak ones. These situations could easily result in identical THD ratings. The variations could be easily measured with laboratory equipment. However do not be overly concerned. Minor variations in THD ratings will not cause *major* differences in sound when listening to *music*. With pure tones as input signals it might be fairly easy to discern which of two amplifiers was used (but again, who listens to tones?).

**Intermodulation distortion** Intermodulation distortion is the second "major" type of distortion that is often specified for amplifiers. Intermodulation distortion is much more objectionable to the human ear because it generates non-harmonically related "extra" signals which were not present in the original. It is analogous to someone singing way off key in a choral group

Intermodulation distortion (sometimes abbreviated IM) is more complicated to test for and specify. Basically, two pure tones are simultaneously applied to the input of the amplifier. If the amplifier were perfect, the two tones (and only the two tones) would be present at the amplifier output. In the real world, the amplifier would have some harmonic distortion (as described above), but careful observation of the output signal (using laboratory equipment) would reveal that there are a number of new tones present which cannot be accounted for as a result of harmonic distortion. These "new" tones are called "beat products" or "sum and difference" frequencies, and are a result of the interaction of the two pure tones within the amplifier. No amplifier is perfect, all have some *non linear* characteristics. Whenever two signals are applied to a nonlinear system, new signals (in addition to the original two) are generated. For

a good amplifier, the new signals are *very* small in relation to the two original tones. This is fortunate, since the ear can detect much lower levels of intermodulation distortion as compared to harmonic distortion.

It should be noted that distortion measurements on amplifiers are made with test tones. These tones are usually sine waves (pure tones), which represent the simplest possible test signal to measure and quantify. A music signal is an *extremely* complicated waveform consisting of many constantly changing sine waves. Since music has so many harmonics and frequencies present, quantifying how two different amplifiers will sound by using simple THD and IM specifications is extremely difficult. In other words, just because two amplifiers have the same published specs for THD and IM does not mean that they are equivalent. Fully and completely quantifying the technical performance of an amplifier would be extremely complicated and costly (and would probably have little benefit in the end). Most amplifiers available today (from reputable manufacturers) have THD and IM levels low enough to yield excellent performance (so long as they are not overdriven). This leads nicely into our next topic...

## Clipping: What is this?

*Clipping* is a term which many people have probably heard, but may not fully understand. Very simply, clipping of an amplifier occurs when one tries to get a larger output signal out of an amplifier than it was designed to provide.

As stated before, all power amplifiers have a DC power supply which provides power to (among other things) the output stage of the amplifier. For most amplifiers, the power supply consists of a "plus" supply and a "minus" supply. The two voltages are often referred to as "rail voltages" or simply "rails". As an example, a 200 wpc amplifier (at 8 ohms) might have a power supply voltage (rails) of +/- 60 volts DC. This means that the output voltage which drives the speaker can *never* exceed + 60 or - 60 volts. If the amplifier is playing at near full volume, and someone "cranks up the volume", the amplifier will attempt to put out more power. However, the power required to meet the sudden new demand for more volume cannot be met by the power supply voltage, which has limits of +/-60 volts in this example. The result is a waveform with the top portion (or peak) "clipped" off (hence the term "clipping"). Such clipping represents a distortion which is added to the waveform (and if it is severe enough it will be clearly audible). If a signal is severely clipped, the waveform takes on the shape of a "square wave", and the resulting sound will be absolutely hideous. Clipping can be easily observed using an oscilloscope attached to the amplifier output.

Clipping is not usually a major problem for amplifiers (unless it is extreme), but it can be very detrimental to speaker systems. Whenever clipping occurs, two things happen: (1) the spectral content of the music signal is altered (high frequency components are generated), and (2) signal compression occurs. If excessive clipping occurs, tweeters will be the first to blow followed by midrange drivers. Woofers are best equipped to survive clipping (unless the abuse is blatant).

In general, clipping of an amplifier should be avoided. Use an amplifier that has clipping indicators, and

pay attention to them! Occasional clipping is OK and probably not very audible. However if you find yourself clipping the amp most of the time, you should consider obtaining stronger (or additional) amplifier.

## Damping Factor... What is this?

The *Damping Factor* of an amplifier in general refers to the ratio of the amplifier's output load impedance (the speaker, nominally 8 ohms) to the output impedance of the amplifier. Ideally, the damping factor would be infinity (in other words, the ideal output impedance for an audio amplifier is zero ohms). Damping factor, like many amplifier specifications, is a function of many factors and is thus difficult to quantify with a single number. As such, "low end" manufacturers can have a "field day" with this spec, publishing fantastic numbers (however with no information as to how the measurement was made).

The damping factor if an amplifier depends greatly upon the speaker to which it is connected, the wire connecting the speaker to the amplifier, the signal frequency that the amplifier is sending to the speaker, and the power level at which the amplifier is operating, among other things. Damping factor is most critical at low frequencies, generally 100 Hz and below (i.e. frequencies that a woofer reproduces). At such frequencies, a high damping factor is desirable in order to maintain a "tight" sound. If an amplifier/speaker pair has a low damping factor, the bass response is likely to be "boomy", "uncontrolled", and "loose" sounding.

Specifying damping factor as a simple single number does not really tell the whole story. Damping factor is a ratio of two numbers, one of which (the speaker impedance) varies by a large amount depending upon frequency. This being the case, the damping factor will also vary considerably as a function of frequency. Most of the variation in damping factor is due to the characteristics of the speaker connected to the amplifier. The wire which connects the speaker to the amplifier has finite resistance which must be accounted for; basically it is lumped in with the impedance of the speaker. So, it is wise to use heavy speaker wire in order to minimize degradation of the damping factor.

As mentioned, the output impedance of an amplifier is ideally zero. In the real world, this is never the case. The next best thing would be a very low constant (non changing) impedance. Again, the real world does not allow this either. The output impedance of most amplifiers is relatively constant except for when they approach the last 10% or so of their voltage output. This is due to the nature of the waveform from which most power supplies obtain their energy (especially analog supplies) . What this means is that the output impedance of an amplifier tends to rise considerably as it approached its output limit. As the amplifier's output impedance increases, the damping factor must decrease proportionally. In my opinion, if manufacturers specified the *output impedance* of their amplifiers, there would be a lot less ambiguity among the numbers.

High damping factor numbers go hand-in-hand with amplifiers that can drive very low impedance loads (these are amplifiers with power supplies capable of delivering tremendous current). If you want to

"artificially" degrade the damping factor of your system (to hear the effects), a simple test can be done. Listen to your system at a "healthy" volume (use a CD with lots of low, tight percussion type sounds); be sure to use a heavy gauge short length speaker wire. If you have a sound level meter, note the sound level you listened at. Then, connect your speaker up through a 100 foot (give or take) wire with much smaller gauge (use #20 or higher). Play the same music as before, but *make sure the volume (to your ears, not the volume control!)* is the same (this is where the sound level meter comes in handy). The volume control on the amp will have to be turned up a bit to overcome the power loss in the smaller wire. You should be able to tell that the sound has changed (for the worst, in most people's opinion).

Do not be *terribly* concerned with damping factor when choosing quality equipment. Most of the good amplifiers and speakers available today will yield excellent sound when used together. To avoid degrading the damping factor of your system, simply follow these (easy) steps:

- Don't load up an amp with multiple pairs of low impedance speakers
- Use heavy gauge speaker wire, ESPECIALLY in long runs
- Never wire resistors in series with your speakers (you can't change a 4 ohm speaker to 8ohms by doing this!)
- Use a heavy duty (i.e.12 gauge or heavier) extension cord when plugging your amp into the wall outlet.

## Can I get a shock from the speaker connections on my Amp?

YES! Amplifiers in the 400 plus watt per channel range are not uncommon today. Such an amplifier will put out about 50 to 60 volts RMS to a speaker. While this is only about half the amount that comes out of a wall socket, it's definitely enough to be unpleasant if you are holding on to it! Note: The US Military defines any voltage in excess of 30 volts as hazardous. Such a voltage can be generated by any amplifier in the 100 + watt per channel range.

As a side note, it's not a good idea to plug or unplug speakers when the amplifier is playing at high volume. The "make and break" of connectors can cause momentary short circuits, as well as voltage and current transients (none of which is healthy for the amp). The preferable procedure is to make all speaker connections (and disconnects) with the amp turned OFF.

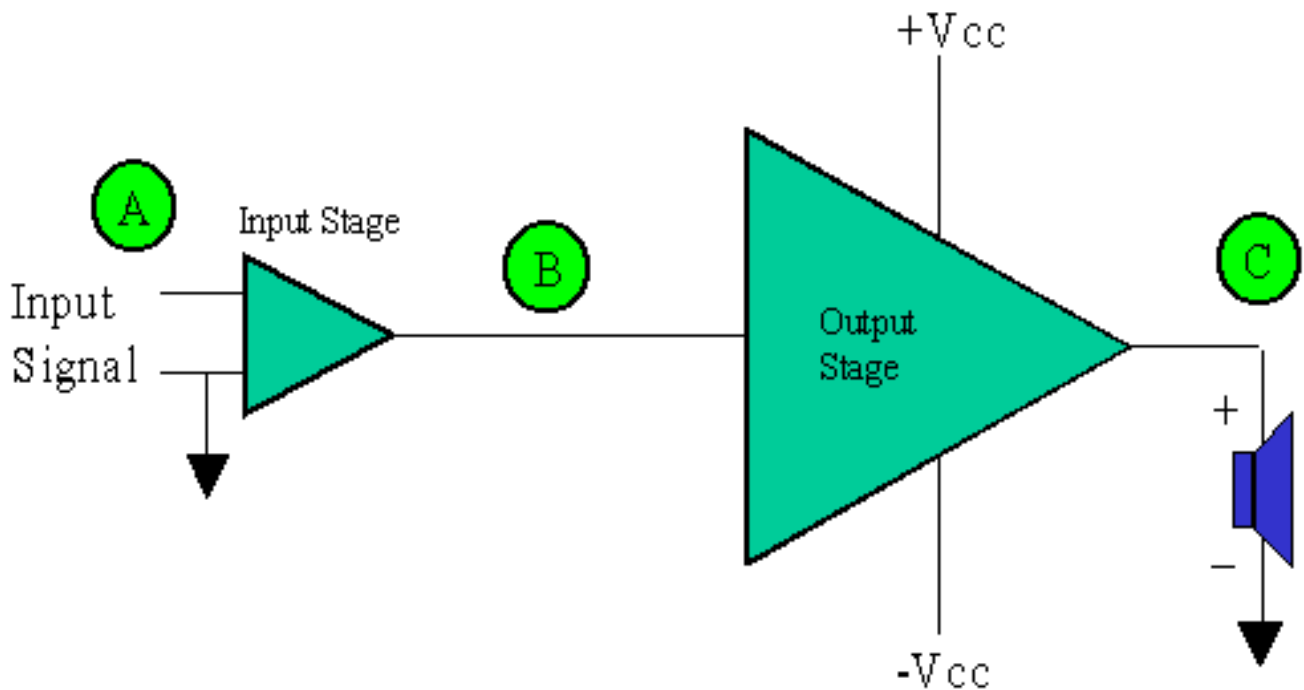
## What is "Bridging"?

*Bridging* an amplifier refers to configuring a two channel (stereo) amplifier to drive a *single* load with more power than the sum of the two original channels combined. For an example, a 100 watt per channel amp may put out 300 watts(one channel) after bridging.

There are important things to know about running an amplifier in the bridged mode:

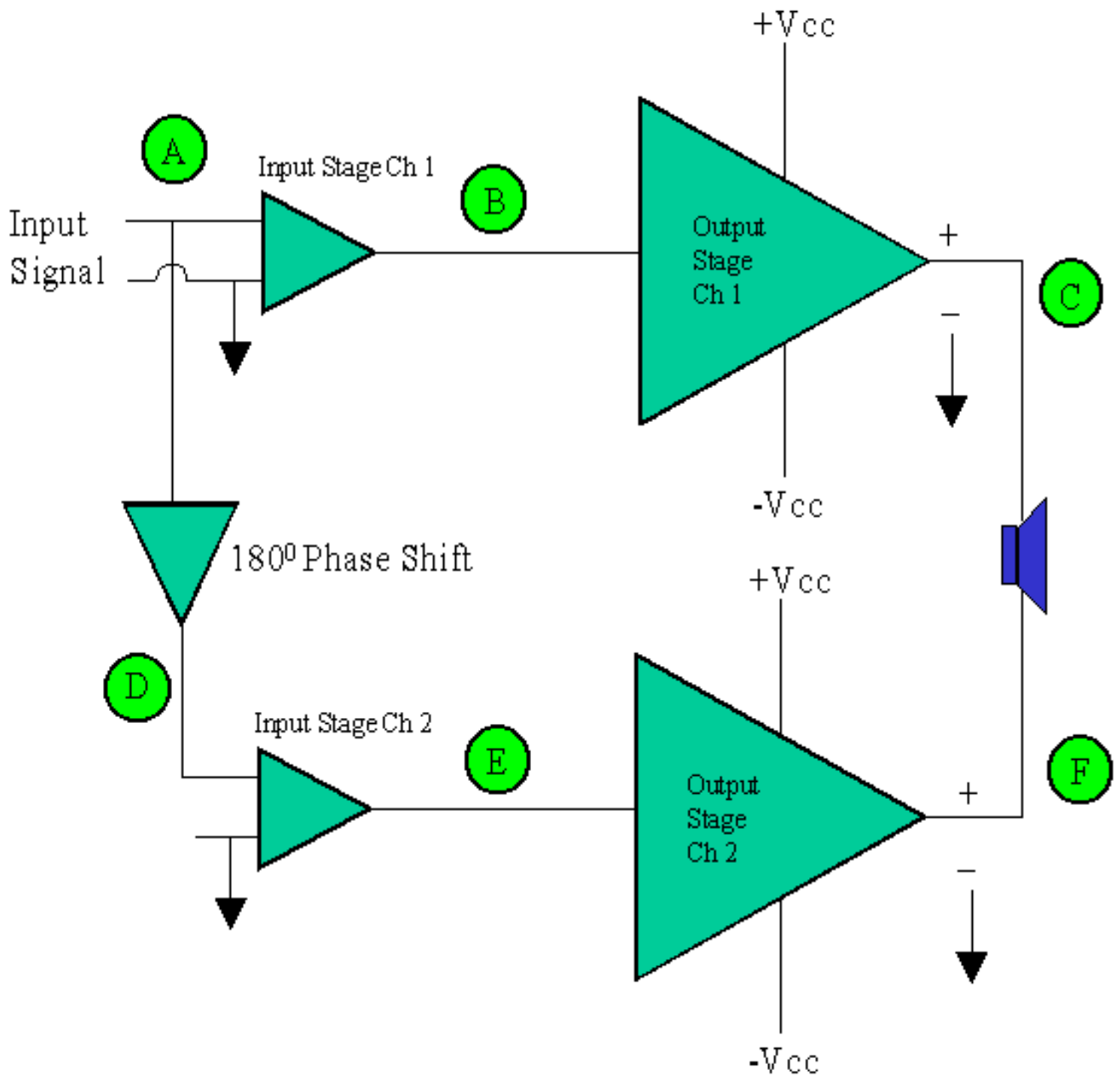
The following diagrams will be used to try to explain what goes on when an amplifier is operated in

bridged mode. The letters in circles are designators, when I have time I will generate waveform plots that show the signals at each of the stages shown. In the meantime, we first we show a diagram for an amplifier operating in "normal" mode:



**Figure 1. Class AB amp running in "normal" mode.**

In Figure 1 above, we show one channel of a two channel amp. Note how the speaker is connected to output of the amplifier.



**Figure 2. Class AB amp running in "bridged" mode.**

Figure 2 above shows a concept diagram of a two channel amplifier operating in bridged mode. Note that there are many ways to do the 180 degree phase shift, the concept diagram above is but one way of accomplishing this task. Note that the speaker is connected between the two "plus" terminals at the output of the amp! Bridged amplifiers work basically as follows: A single input signal is applied to the amplifier. Internal to the amp, the input signal is split into two signals. One is identical to the original, and the second is also identical except that it is passed through a "phase flip" circuit. The original signal

is sent to one channel of the amp, and the inverted signal is applied to the second channel. Amplification of these two signals occurs just like for any other signal. The output results in two channels which are *identical* except one channel is the *inverse* of the other. The speaker is connected between the two *positive* amplifier speaker output terminals. In words, one channel "pulls" one way while the second channel "pulls" in the opposite direction. This allows considerably more power to be delivered to a single load.

Some facts about amplifiers operating in bridged mode:

- An amplifier running in bridged mode has *one* output channel to which a load (speaker) can be connected. It is no longer a two channel (stereo) amp as far as input signals and loads are concerned. If you have two speakers and want to use bridged amplifiers, you will need *two* stereo amplifiers.
- Amplifiers running in bridged mode accept a *single* input signal. Normally, the input will be "Channel 1" or "left". Manufacturers vary however, so check the instruction manual for the proper input wiring procedures.
- If the amp you want to run in bridged mode *does not have built in facilities* for doing so, you should not attempt to use it in this manner (unless you are thoroughly sure of what you are doing).
- If you run bridged amplifiers, you must pay close attention to speaker wiring *phasing*, otherwise, you may have "hollow" or "weak" sound. The speaker cable connection in bridged mode connects to the two *positive* (usually red) speaker connection terminals on the amplifier (the ground (black) connections are not used). The manufacturer will state which red terminal is really the "positive" connection.
- The speaker output signals of a bridged amplifier are *floating*; such connections must *never* be connected to any grounded device (such as an external accessory power meter, for example). If you do make such an illegal connection, one amplifier channel is basically short circuited (worst case result is a blown amplifier!).
- Amplifiers running in bridged mode are usually capable of doing so only with speakers that have impedance of *twice* the minimum impedance the amp is rated for when in normal mode. In other words, if an amp is rated to handle 4 ohm loads in normal mode, in general the minimum load for bridged mode will be 8 ohms. Check your amp's specifications for details.

If we had our *perfect* amplifier, upon bridging it we would have a single channel amplifier with exactly *four times* as much power as any *one* channel of the amplifier in "normal" stereo mode, assuming an 8 ohm speaker load. This is because the effective output *voltage* available to drive the speaker has *doubled* as a result of bridging. A doubling of *voltage* on a given load results in a *fourfold* increase of *power* delivered to that load. If we used a 4 ohm load on the *perfect* bridged amplifier, the output *power* would be a very substantial *eight* times the normal stereo single channel 8 ohm output! These numbers should give some clues as to why real world amplifiers cannot meet such expectations. Once again, we are back to limitations of the *power supply*. In reality, most amplifiers in bridged mode will put out about 3 times the power as any one channel of the amp in normal stereo mode. The fourfold increase cannot be achieved because the power supply is unable to provide the current required for such performance. With 4 ohm loads, the situation is compounded. The amount of current required to drive a

4 ohm load when in bridged mode will tax the amplifier's power supply to its absolute limits. Not to mention, the output stage may not be able to safely handle the extra heat that will be dissipated. Bottom line: stay away from 4 ohm loads if you are running an amplifier in bridged mode!

## Bi-Amplification: what is this?

Bi-amplification is (in most cases) defined as using more than one amplifier to power a speaker system (typically one amp for bass and one for mid/high frequencies). Similarly, some systems are tri-amplified and once in a while you may see a Quad amped system (although the latter is usually reserved for only the most powerful touring type sound systems). Bi-amplification has many significant benefits:

- **Lower Distortion** By using a separate amp for bass and a separate amp for mid/highs, the bass amp can clip without adversely impacting the sound quality. Chances are that the ugly sounding distortion produced by clipping an amp will not sound as bad coming out of a woofer (because woofers don't reproduce the high frequency distortion signals very well). Not that this means you should consider clipping a good thing, it's just that the impact won't be as bad.
- **More total output** Let's say you have a 400 watt single amp and 300 watt bass amp and 100 watt high/mid amp for use in a bi-amped configuration. You will be (in most cases) be able to get a higher volume out of the bi-amped system as compared to the single 400 watt system. Why is this? Isn't the total power the same? It is. However, due to the nature of musical waveforms, you'll be better able to handle large peak signals better. For example a loud bass sound and a loud HF sound at the same time may require a signal level that would clip the 400 watt amp. However, because in the bi-amp system the signal is split into different frequency bands, there's a better chance of handling the signal without clipping. Note that you can't expect to get a large amount of increase in sound level by using bi-amped systems, but in general you will get somewhat more capacity and at the same time reduced levels of distortion.
- **Better Crossover performance** The bad news is that bi-amplification requires that the signals to the amps be split into low and high bands BEFORE they are input to the amp. This requires what is called an "electronic" or "active" crossover network. The bad news is that this means more money, but the advantages and flexibility are attractive. Using an electronic crossover eliminates the need for large, expensive, lossy and non linear passive low frequency crossovers in bass systems. Also, it is easy to design steep slope crossovers using active electronics, allowing for very sharp cutoff of lows and highs (a good thing for HF systems). Also, most active crossovers allow one to adjust the crossover frequency, adjust the level of the signal and vary the phase. For high power systems, bi-amplification is the norm (and tri-amplification is not at all uncommon).

Bi-amplification is common among car audio systems (the powerful ones, not so much with the stock stereo that comes with most cars). Many home theater systems use a separate amp to drive a subwoofer. These systems are a form of bi-amplification, although in most cases the amp is a multi channel amp (not two separate amplifiers) and also they generally don't have nearly the flexibility of a conventional bi-amped system. Nonetheless, many of the benefits of bi-amplification are realized. The same applies to powered subwoofers for the most part. (I'll expand this section a lot more when I have time, just wanted to get the basics in there for now, JR)

## Maximum Power Transfer Theory and Efficiency

Note: This section is intended primarily for engineering students or those with a deeper technical interest. The purpose is to provide a "real world" explanation of the Maximum Power Transfer theory and why it is NOT used in amplifiers designed for stereo systems!

Second year electrical engineering students have most likely covered the theory that basically states "maximum power is transferred to a load when the output impedance of the source is identical ("matched") to that of the load". The connection that some people fail to make is that maximum power transfer *doesn't mean maximum efficiency!* At best, if the maximum power transfer theory is used, efficiency will be only 50% (not such a good figure for an audio amplifier). In other words, if an amplifier is designed for maximum power transfer to a load, fully one half of the energy required by the amplifier's output stage will be dissipated (i.e. wasted) in the source impedance!

For amplifiers used in stereo systems (audio amplifiers), the goal is to have the amplifier output impedance be as *low as possible* (ideally zero, but this is never achieved). If an amplifier were to have an output impedance of 8 ohms (a common value for speakers), maximum power transfer would occur. However two other bad things result. First, the efficiency of the amplifier is at best only 50%, meaning that the amplifier will generate a lot of heat. Second, the amplifier/speaker system will have a terrible *damping factor*. Damping factor basically refers to the ratio of speaker impedance to amplifier output impedance; high numbers are better. A low damping factor will not damage anything but it will tend to louse up the sound considerably. To maintain a "tight" sound, it is important to have the output impedance of the amplifier be as low as possible with respect to the speaker. Otherwise, the amplifier will not have as much control over the speaker. Speakers, being highly complicated electromechanical devices with reactive impedance properties, behave better when they are connected to an amplifier with an extremely low output impedance. Speakers tend to electrically "buck and kick" an amplifier when in operation; the best way to tame this behavior is to put a heavy "load" (i.e. an amp with a very low output impedance) on the speaker! An amplifier/speaker combination with a low damping factor will tend to have a "boomier" sound and poorer transient response, (such a sound is not always bad, some people actually prefer it!).

There is a quick test anyone can do to get a feel for what affect the damping factor has on a speaker system. Disconnect your speaker system from the amplifier, remove the grille, and *gently* tap on the woofer cone. You will hear a low frequency sound, this is the "resonance frequency" of the system. Note the characteristic of the sound as you tap the cone. Now, connect the speaker up to the amplifier, and turn the amplifier ON (but leave the volume at zero). Now tap on the speaker cone as before. You will observe that the sound has changed considerably. The sound will be much "tighter", and the cone will seem harder to move. This is because the amplifier has in effect "loaded" the speaker system. The case where the speaker was disconnected from the amplifier represents the worse possible damping factor (zero).

Anyway, back to the topic of this section. Although there are many applications where maximum power transfer is desired, audio amplifiers are not one of them. Audio amplifiers generally deal with a considerable amount of power, so high efficiency is a more important design consideration.. In addition, to maintain high quality audio, an audio amplifier ideally has an output impedance which is VERY small compared to the impedance of the speaker it will be driving. Note that using 4 ohm speakers on an amplifier will degrade the damping factor as compared to using 8 ohm speakers.

## Questions / Other

If you have questions you can e-mail me. Please note that I receive a considerable volume of questions as a result of my web pages, and I may not always be able to answer promptly (although I try). I provide answers only in areas where I am qualified (I will let you know if your question is not one I can answer). Also please note that I in general cannot answer questions regarding specific brands and models of equipment. With so many kinds of equipment available, it is nearly impossible to know all of them! Questions I can best answer involve fundamental performance characteristics of equipment (why low impedance loads are difficult, what is bridging, etc).

[E-mail to Joe Roberts](#)

[back to the Audio Tech Stuff page...](#)