Analog Ethos™

PACIFIC 63

Stereo Tube Amplifier Kit

Part 1 - Build Instructions
Part 2 - Tube Amplifier Explained

WARNING: RISKS OF ELECTROCUTION OR FIRE

Be safe with this amplifier kit! Follow all instructions carefully in this manual, and for your safety, please take seriously the risks and follow the precautions below. By purchasing this kit and assembling it yourself, you have personal liability to ensure safety in the building and operation of the amplifier. DIY electronics is a great hobby, but it has risks including but not limited to those below.

Important Risks:

- This amplifier uses mains voltage (120/240 VAC) which can kill you if you touch it and current passes through you.
- This amplifier uses a power transformer to create a high voltage power supply up over 400 volts DC which will kill you if it passes through you, will destroy your finger or screwdriver if you touch it and create a short circuit. Do not think this voltage is safe in any way to touch with your hand or a tool.
- This amplifier uses capacitors that are charged when the amplifier is on and may not be discharged even when the amplifier is off and unplugged, even hours or days later, if not bled properly. These capacitors hold sufficient energy at high voltage to seriously injure or kill you.
- Vacuum tubes and power transformers create high heat which can burn you if you touch the surfaces that reach hundreds of degrees Fahrenheit. There is a risk of fire if the amplifier is in an enclosed space without adequate ventilation, or other objects are touching or near the tubes or transformer.
- You will need to utilize a soldering iron to build your amplifier, which operates at a very high temperature and will burn you if you touch it, or can cause fire if not properly used.
- The amplifier is heavy, and it requires a sturdy table or stereo shelf to hold it safely.

Follow these general safety precautions:

- Never plug in or operate your amplifier with the chassis open. Do not attempt to trouble-shoot the amplifier while turned on. Use safe inspection methods only when the amplifier is off, unplugged, and capacitors are discharged. Never assume insulated wire or components inside the amplifier are safe to touch while the circuit is live.
- Do not leave an exposed circuit accessible to other people, especially children or pets.
- Keep a clean work space, with no wires or other objects near your amplifier or soldering iron.
- Follow all safety instructions for your soldering iron. Unplug when not in use. Allow safe time and space to cool.
- Utilize a non-conductive work table and mat below your chair or standing position.
- When turning on your amplifier for first use after building, follow instructions to monitor for smoke, smells, sounds, or other indicators of a problem. Immediately turn off power if you detect any.
- Never leave the amplifier turned on and unattended. Always turn it off when you leave the room or your home.
- Operate your amplifier on top of a table or sturdy stereo shelf with at least 12 inches of space above the top of the vacuum tubes, and 6 inches of space around each side of the amplifier chassis. For proper ventilation, ensure the bottom panel is not obstructed and the amplifier rests on the chassis feet.
- Do not place the amplifier inside of an enclosed cabinet or stereo console that has limited ventilation. The amplifier gets hot and requires air flow to stay at a safe operating temperature.
- If you have a child in your home, do not operate the amplifier in a location where the child can reach it. It takes only a second to get a serious burn from the tubes which will glow and may attract the interest of a child.

CONTENTS

INTRODUCTION

So, you are ready to build a DIY hifi stereo tube amplifier? Maybe you have read about tube amplifiers or are a curious learner and have been wanting to make the jump into this hobby, or perhaps you have already built DIY electronics in the past and want to learn more and build a new great-sounding amplifier with high quality design and components. Maybe you are experiencing something in your life and need an escape from the day-to-day and you want to put your mind and hands to work on something to take you away from it for a while.

Whatever your situation, I'm glad you decided to give this a try! Here are the goals I had when I created this kit. Hopefully this matches what you are looking for.

GOALS FOR THIS KIT:

LEARNING

You will truly learn how a vacuum tube amplifier works.

This is not simply a step-by-step guide, although you could use it that way. My intention is to show you exactly how the entire circuit works, how each component is selected and the role it plays. You will walk away with knowledge that empowers you. This may be the only kit you build, or it may be one stop on a journey of learning and building. Perhaps your next build will extend on what you have learned, and you could customize a circuit or come up with your own designs. If you want to learn, this is the kit for you!

As you can see, the instructions are extensive and filled with illustrations, photos, and explanations. Some kits include poorly written instructions that are difficult to follow, and you might get the amplifier to work, but only after frustration, perhaps a few mistakes, and certainly without learning anything new. I want your experience with these instructions to be the best you've ever had in a kit!

HIGH QUALITY AUDIO

You will get high audio quality, with reasonable circuit complexity and cost.

First, I am using a straightforward circuit, mostly to achieve the objective above for you to learn how it works. Before moving on to more complex circuits that could provide higher output power or other characteristics, you should know how a tube amplifies the signal, understand rectification and filtering of a power supply, basics of negative feedback, and several other elements. The circuit used here is customized for the components included, but is not unique, and is based on designs that have been used for many years and proven to work very well. I have carefully tested it and made adjustments.

One of the benefits of a kit is that you have everything you need in one box, and don't need to go through the difficult job of sourcing components, often from multiple suppliers, paying multiple shipping costs and needing to identify precisely the right part from a multitude of options, sometimes testing and needing to try again with a different selection. This work is done for you so you can save time, money, and avoid mistakes.

I have sourced high quality components for this amplifier from carefully selected manufacturers, for the best possible quality without entering into areas of excessive cost. There are many "audiophile" products sold at outrageous prices, and with dubious claims of performance. I'm sure you've seen them—fancy and enormously thick speaker wire or interconnects, rare new-old-stock tubes, magical capacitors, and other items that add up to many thousands of dollars. This kit amplifier will sound excellent and be reliable, at a cost that is reasonable without compromises. There are cheap components available and used in other tube amplifier kits that you may find through other distributors. I did not try to build the cheapest possible kit, and I assume you are someone who is willing to pay for quality.

This amplifier is perfect for typical listening in a home, with relatively low distortion and minimal hum, though it is not a high power amplifier. If you build this kit, learn how it works, and want to build something for louder output or lower sensitivity speakers, you can expand to other more advanced circuits, or try other tubes and components.

VISUAL DESIGN

How it looks is as important as how it sounds.

Great visual design is important to have an end-product you are proud of. This is a showpiece in your home that your friends and family will see and ask about. I wanted to create a custom-designed chassis to hold the components functionally but also beautifully in context of your listening environment. Inspired by the original Dynaco ST-35 amplifier layout, the kit includes a chassis that perfectly fits the transformers, tubes and physical layout of the circuit with colors and finishing touches that make it unlike any other amplifier!

For those of you who don't have extensive workshop tools and finishing capabilities, this allows you to have a great-looking amplifier and not settle for an off-the-shelf, generic metal box.

But for those of you who do have tools and skills of design and construction, this kit can serve as your entry into building a tube amplifier. You may in the future build a custom design of your own for your next amplifier and put some of your own inspiration into it! Tube amplifiers are both art and science, and I hope you'll lean into your creative side. Along the way, I'll share a few thoughts on design, inspiration and craftsmanship.

WHAT YOU NEED TO GET THE MOST OUT OF THIS KIT:

- Basic soldering skills (and a few hand tools). You will need to solder components onto printed circuit boards, and connect some wires and small components in enclosed spaces. It's not always easy, and I'll share tips along the way. You don't need extensive soldering experience, but should be able to get good and clean connections in order to have a safe and functional amplifier.
- High quality, relatively efficient speakers. The Pacific 63 is a push-pull amplifier that can drive any 4 or 8-ohm impedance speakers, but of course your sound quality will depend on having good quality speakers. The amplifier will provide up to around 15 watts of power per channel for a sufficient line level source. This is more than a single-ended tube amplifier would typically provide, but is still not a huge amount of power and you should not expect to be rocking a neighborhood party with this amplifier! More efficient speakers above 90 dB SPL will help convert power to listening volume, but you can use the amp with any speakers.
- Patience! Take your time and enjoy the process. Read and observe carefully to put the right parts in the right places. If you make a mistake along the way, take a step back and try again. There are few mistakes that can't be corrected if you catch them in your process, but if you don't follow the instructions and checks carefully, it's possible you will damage components along the way or when you turn on the amplifier.

NOTES ON SAFETY

As indicated on page 2 of this manual, there are important risks to be aware of with an electronics hobby. Hopefully you know that already! Most importantly, tube amplifiers operate at very high voltage. In this case, steady state operating around 350 volts DC, which is enough to kill you. These instructions do not have you doing any work or testing on a live circuit.

This manual gives you instructions to build the amplifier in a safe manner using a limited set of common tools and a soldering iron. Along the way, I will provide additional tips and suggestions for builders who may be interested in expanding into a more extensive hobby, or in the theory and testing methods behind the design of this kit. This manual cannot possibly go into all the details of safe testing methods that could involve use of multimeters, oscilloscope, spectrum analyzer, variable transformer, and others. If you are a trained technician or have experience working with high voltage, it is at your own risk (I know you know this!) if you choose not to follow these instructions or elect to perform other tests using additional equipment.

When I build and test amplifiers, I do sometimes access an exposed circuit using test equipment. You should not expect to do this at all with this kit. However, so that you can understand some elements of safety, below are examples of protective steps I take:

- I use an isolation transformer so that any exposed mains voltage is less likely to have a path to ground if I were to accidentally touch it.
- I use a variable transformer ("Variac") to slowly bring up voltage to test a circuit
- Turning on an amplifier on my work space requires multiple separate power switches, so others in my house can't easily turn on a high voltage circuit. I keep these all off when I leave the room.
- I have a rubber mat beneath my work space as a ground insulation.
- I manually discharge capacitors using a resistor after turning off a circuit, or double-check the discharge is done even if a bleeder resistor is in place.
- I never, ever put more than one hand over a circuit to probe with a multimeter. I force myself to hold a small object in one hand to avoid instinctively reaching in with it (some people follow the rule of putting one hand in their pocket). Current from one hand to the other would pass through my chest (heart) and has greatest risk of death. Brushing across a high voltage potential with one hand could still cause injury but is less likely to be lethal.

You can learn more about safety through many other sources. A healthy respect for electricity and the risks outlined on page 2 are very important. I hope you'll have an enjoyable hobby, but please be safe, for your sake and those around you!

PARTS LIST

Note on parts: Most of the small parts are provided in labeled bags. At times I buy different brands of resistors or capacitors, which means they may look different than the pictures, but they will always be the same type and rating. I specify power rating of resistors only where it matters (where not specified, power dissipation is low and I usually include 1/2W or 1W). Resistors are all metal-film or wirewound types for lowest possible noise. Some components may vary depending on special options I may make available. The important thing to remember is to carefully identify the part using the label on the bag, and to not get resistors confused, especially those marked using color bands instead of printed values. Putting the wrong resistor in the wrong place in the circuit will cause serious problems or damage to components. Double-check to ensure the correct part.

Chassis 	Left and Right Channel PCBs 	2-terminal blocks (13) 	RCA Input Jacks (2 pairs) 	Wood Side Panels 
Power Transformer 	Tube Sockets (6) 	3-terminal blocks (2) 	Speaker Binding Posts (2 pairs) 	CL-120 Thermistor 
Output Transformers (2) 	Power Indicator Lamp 	120uF 450V Capacitors (2) 	0.1uF Coupling Capacitors (4) 	Impedance Switch 
AC Power Cord 	Power Switch 	47uF 450V Capacitor 	68pF Film Capacitors (2) 	UF4007 Rectifier Diodes (2) 
IEC Power Inlet 	Chassis Feet (4) 	33uF 450V Capacitors (2) 	220uF 50V Capacitors (2) 	Volume Knob 
Power Supply PCB 	EL84 Tubes (4) 	12DW7 Tubes (2) 	Input Selector Switch 	2A Slow-Blow Fuse (2) 
100k Dual Audio Potentiometer 	Resistors (various) 100 ohm 7W resistors (2) 200 ohm 5W resistors (2) 220k 2W resistor 5.6k 2W resistors (2) 300k 2W resistors (2) 30k 2W resistors (4) 120 ohm resistors (4) 470k resistors (4) 3.9k resistors (2) 1k resistors (4) 1.2k resistors (2) 1M resistors (2) 11.5k resistors (2) 	Hardware (various) 1/2" Aluminum Spacers (10) 1/4" Nylon Spacers (4) #6 3/8" Wood Screws (4) #4 1/4" Filler Screw, washer, nut #8 star washer Ground Bolt with Star Washer Power Transformer Hardware Output Transformer Hardware #4 nylon washers (10) #4 1/4" truss head screws (10) #4 1/4" pan head screws (10) #6 Nylon Washers (4) #6 Nuts (4) Ring terminals (2) 	Hookup Wire (various) 2-Conductor Shielded Wire 	Heat Shrink Tubing Zip Ties High Voltage Sticker 

TOOLS AND WORKSPACE

You will need the following tools to assemble the amplifier:

- Phillips and flat head screwdrivers of several sizes
- Nut drivers/sockets or wrenches for #4 and #6 nuts
- Wire stripper for 18-22 gauge wire (kit includes all wire you will need)
- Soldering iron with medium chisel tip. I suggest a 40 watt pencil-style iron, or a temperature-controlled solder station. Have a sponge or wire tip cleaner, too.
- Electrical solder. I suggest 0.8mm 63/37 tin/lead rosin core solder.
- 90-99% isopropyl alcohol with soft brush or lint-free wipes to clean flux from PCBs
- Side-cutters or other small snips for trimming leads
- I suggest small needle-nose pliers for shaping leads
- Ideally you have a digital multimeter for double-checking resistance of individual components and connectivity (while the amplifier is off)
- You could also consider some type of "helping hands" tool with alligator clips or other method to hold items in place as you solder them

An ideal workspace is a flat table or work bench with good lighting and room for your soldering iron to sit in its holder without risks of touching other objects or being bumped into your lap. There should generally not be a cat residing on this table. Have plenty of space to lay out the instructions and components you are working with. If you don't have a dedicated work bench, the kitchen table is fine. Tell your family that your amplifier is more important; they will understand.

To hook up and operate your amplifier, you will need:

- A pair of 4 or 8 ohm speakers and speaker wire (never operate the amplifier without speakers or load hooked up, or you may cause damage to the amplifier). For best results, I recommend speakers around 90 dB rated sensitivity or higher, and speaker wire 16 awg or heavier.
- A line-level audio source (CD player, computer audio output, phone/tablet with headphone output, etc.) and audio connector wire with RCA jacks. This amplifier does not have a phono stage and will not directly amplify a turntable unless you have a separate pre-amplifier for the phono source.

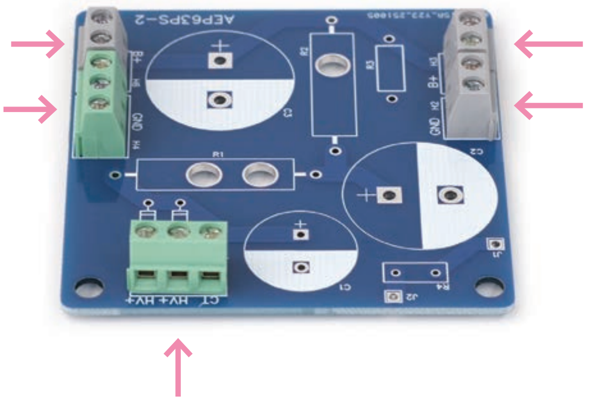
To enjoy your amplifier in operation, you will need:

- A great album to play. I think The Beach Boys Pet Sounds would be a good first listen.
- Comfy chair, preferably with cat in your lap.
- Celebratory drink. I recommend a gin & tonic with fresh lime.

PART 1
BUILD INSTRUCTIONS

STEP 1 - Power Supply PCB

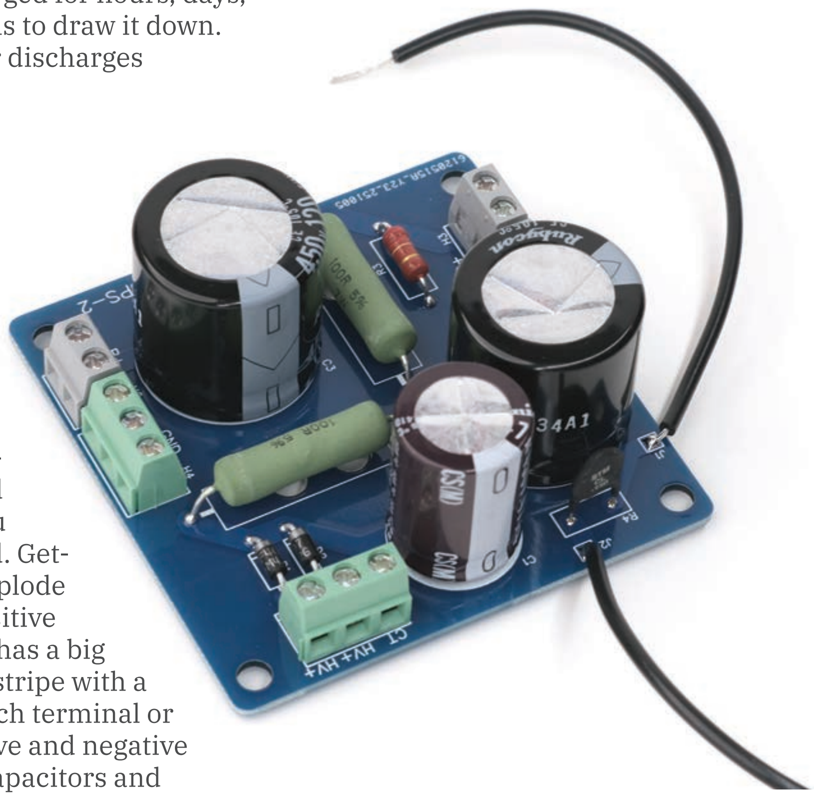
Let's start with the power supply PCB. Most of these components are rectifying and filtering high voltage from the power transformer. This one is pretty easy—only a few components. I'll give you a suggested order, and see the soldering tips on the next page if you are new to soldering components on a printed circuit board.



1. Start with the 3-terminal blocks to H1 and H4, and the 2-terminal blocks to H2, H3 and H6. Take a look at the photo so you orient them correctly. (Ignore the PCB outlines that might imply facing them a certain way.) The open side where the wire goes needs to face outward. You can put these down on a hard surface and put the PCB on top like it's a little table on legs—this even becomes a nifty way of holding up the PCB for some other components. Solder all the terminals in place. Yes! You are on your way.

2. Next, solder in the two rectifier diodes to D1 and D2. High voltage from the power transformer will go through these as part of the process to convert from AC to DC. Note the gray stripe on one side of the rectifier, which MUST be oriented in the direction shown on the circuit board (away from the terminal block). Really important—double check you got it right, then solder & trim the leads.

3. There's a 220k ohm 2W bleeder resistor that goes in position R3. A bleeder resistor provides no function to the amplifier itself but is used to deplete the charged capacitors when the amp is turned off. In some situations, a capacitor could stay charged for hours, days, or longer if there were no other circuit connections to draw it down. This becomes a safety risk, so the bleeder resistor discharges the capacitors slowly after shutoff.



4. This power supply uses two big 7 watt 100 ohm resistors as part of the filter stages to smooth the high voltage supply (these are part of the "RC filters" with the capacitors). Solder these into R1 and R2. These will get hot, so give them a little breathing room; don't mash them down flat touching the PCB. You'll notice I put some holes in the PCB to promote airflow also.

5. Ok, now we're ready for the capacitors. Most of the capacitors in the amp are polarized electrolytic capacitors. This means they have a positive and negative lead. This is SUPER IMPORTANT that you orient it correctly with the negative side to ground. Getting this wrong has a big risk the capacitor will explode when you run the amp. The PCB will mark the positive side with a plus (+) symbol, and the negative side has a big white fill area. The capacitor itself will have a big stripe with a negative (-) symbol on one side that indicates which terminal or lead is negative. Orient the capacitor so the positive and negative match the PCB. Snap in both of the 120uF 450V capacitors and solder to C2 and C3. Then put the 47uF 450V capacitor in C1. Sweet capacitance means a clean power supply, ready to have an amplified signal!

6. This PCB uses a little space to mount a negative temperature coefficient (NTC) thermistor. This is a device that goes on the primary side of the power transformer. When it's cold, it provides some resistance to slow down the current that will immediately be drawn as the power transformer and capacitors charge up. When it gets hot, resistance drops to be negligible. Solder the CL-120 thermistor to R4.

7. There are also two holes for wires that connect to the thermistor. Trim and strip a 5" black wire and solder to J1, and put a 4" black wire on J2. These will go to the AC power inlet and the power switch, putting the thermistor in series with the primary side of the power transformer.

All done! Inspect the whole PCB to make sure it looks good and then clean it with some isopropyl alcohol or other cleaning solution so you don't have any messy flux left. This baby's ready to go.

SOLDERING TIPS

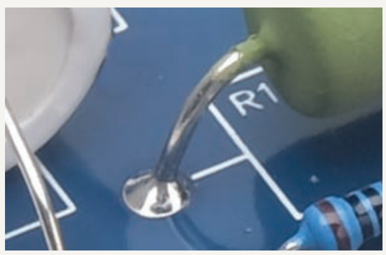
I have to say, I really love soldering components onto printed circuit boards. It's satisfying, a little therapeutic even. If you have never done it before, here are a few quick tips to help you.

Because a PCB allows components to be placed close together, there can be some tight quarters that could be difficult to reach with a soldering iron if you put components on in the wrong order. My PCBs are not super cramped but still I suggest you follow my suggested order, which is generally going from smallest to largest components, with a few exceptions.

You will bend the leads of certain components like resistors so they fit, while other components just drop in place. You insert from one side of the PCB to get the component in place and then solder the leads on the opposite side of the board. For long-lead components, I often will bend them a bit after they are in place so that when the board is upside down, they don't just fall out from gravity. For other components, they will fall out when you turn the board upside down, so you may need to find ways to rest the component and board in place so they aren't moving while you solder. Some resistors can get hot as they dissipate power, so for any of the larger resistors I suggest you put them snug in place but not mashed down flat touching the PCB surface—give them just a little air around them for breathing room.

You solder only one side of the board, not both--the hole draws the liquid solder in so it's filled up. You want the component to stay as still as possible—a component in motion while the solder cools could result in a poor solder connection. Hold the very tip of the soldering iron touching the intersection of the component lead and the tinned hole on the PCB, and then feed the solder right into that joint so it melts into place, and then remove the soldering iron. Clip off the leads and that's it!

It's a bit of an art to time this just right to have the perfect amount of solder and get it to flow through the hole so both sides of the lead are secure. Hold the iron on for about 1-2 seconds to heat up the joint, feed the solder in until it is sufficiently full, then hold the iron for another 1-2 seconds to allow the solder to suck itself into place through to the other side, then remove. Got it? Hold the iron to heat up, feed in the solder, hold the iron a bit longer, remove. Too fast and you won't get enough solder in. Too slow and you'll have too much solder starting to blob up. Smaller components need less; larger components need more. Don't hold the iron in place too long as you don't want to melt things. Some components are sensitive to heat if you go too long.



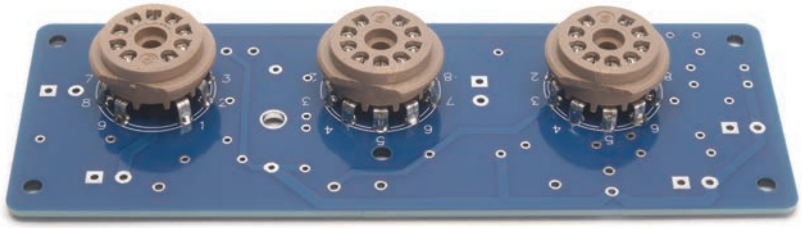
Check your work and you'll start to get a feel for it. You want to see a little "volcano" of solder sloping up from the pcb pad onto the lead. It shouldn't look like a crater (not enough solder) while it also ideally doesn't end up like a boulder blob (too much). If it's not enough, just touch it up quickly on either side of the PCB so it's a good connection.

You got this!

STEP 2 - Left and Right PCBs

Ok, good work on the first one. Now let's do the left and right amplifier circuit boards. A typical stereo amplifier will have a single power supply and then two identical circuits, one for the left channel and one for the right. That's what these circuit boards are. They will hold the tube sockets and associated components to amplify the input signal. You will note that they are not identical! One is marked with a label "AEP63-L" and the other "AEP63-R". They each have a different physical layout, though in general they have the input tube components on one end and the two output tubes on the other. They aren't identical because the wire connections need to be oriented toward the middle of the amp, while the tube sockets have to face the same chassis-top direction, so this requires a unique layout for each. I'll provide instructions starting with the left (L) board, and then you'll repeat the steps for the right board.

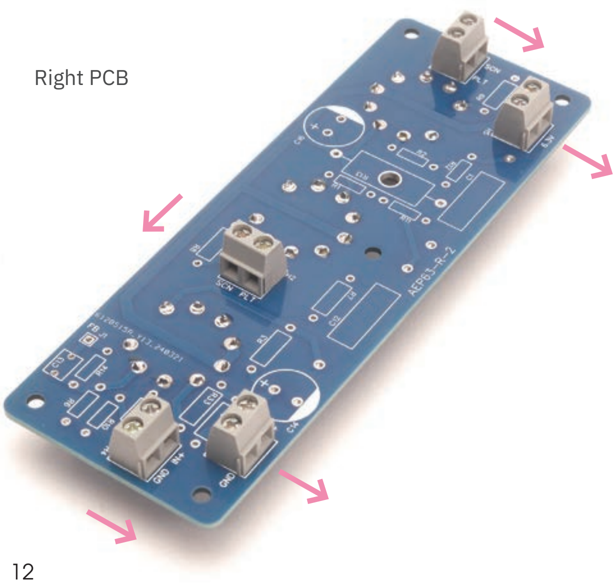
You will also note that the photos and layout diagram show the amp from the bottom, where the left channel is actually mounted on the side of the amp that is visually on the right-hand side of the image. When the amp is turned upright, the boards and tubes for L and R are oriented toward their actual left and right channel directions.

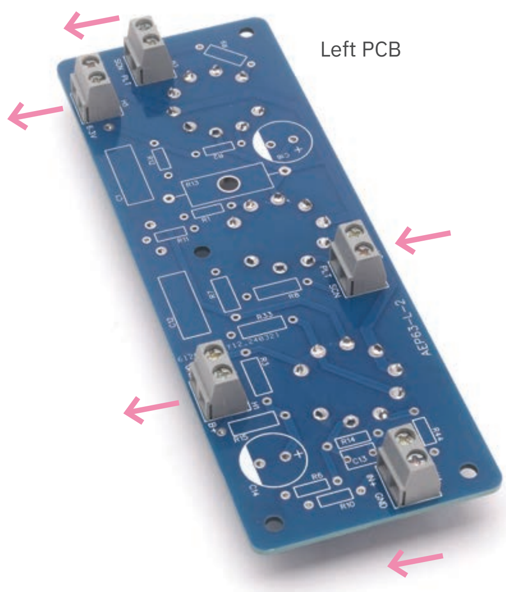


I really want you to understand what all these components are, so I will make a light reference here, and it's all explained in detail in Part 2 of this instruction manual!

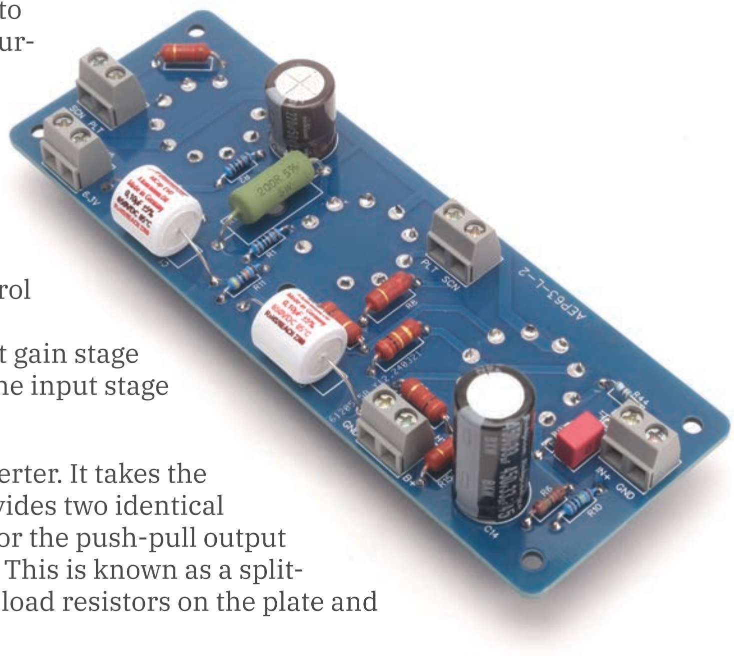
1. Start with the tube sockets. These are the only components soldered on one side of the board, while all the others will be mounted on the other side where most of the circuit markings are. You can picture how this will work—the tube sockets need to face the chassis top (exterior) of the amplifier and everything else will face the interior of the amp. Later we'll put some spacers on to screw it in place. Fit the tube sockets securely in their holes so they are flat. They should stick in place and then you can turn the board upside down so it is resting on the sockets, while you solder all of the pins in place. They will take a good amount of solder to get the filled and secured well. Inspect from both sides to make sure it's all looking good.

2. Next, on the other side of the board, solder the five 2-terminal blocks in their positions. Note the photo and layout diagram that shows the orientation where the wire holes need to be. Double-check to get these all right! If you put a few of these in and turn the board upside down, it can rest on the terminals while you solder.





3. Resistors are next, going generally from smallest to largest, but I'll group these to help explain their purpose briefly. First is the input gain stage, which is one half of the 12DW7 tube. It takes the small input signal and amplifies it large enough to drive the output tubes.



    a. 1M to R10 – grid leak resistor providing a ground reference to the grid
    b. 3.9k to R44 – grid stopper resistor, helps control oscillation
    c. 1.2k to R6 – cathode bias resistor for the input gain stage
    d. 300k to R33 – anode (plate) load resistor for the input stage
    e. 11.5k to R14 – negative feedback resistor

4. The other half of the 12DW7 tube is the phase inverter. It takes the amplified signal from the first gain stage and provides two identical outputs in opposite phase to drive the two tubes for the push-pull output stage. Solder two 30k ohm resistors to R3 and R7. This is known as a split-load or concertina phase inverter, using identical load resistors on the plate and cathode of the tube.

5. Now solder the components for the two EL84 tubes that are the output stage. Push-pull amps have two tubes that work together in opposite phase. The output transformer recombines the amplified signals into a single output that drives the speaker. Each tube will end up getting the same resistors, so there will be two of each below, with one exception.

    a. 470k to R11 and R12 – grid leak resistors for a ground reference to the grids
    b. 1k to R1 and R2 – grid stopper resistors to control oscillation on the input grids
    c. 120 ohm to R8 and R9 – screen grid stopper to control oscillation on the screen grids
    d. There is a single 200 ohm 5W resistor shared by both tubes as the cathode bias resistor. Solder to R13.

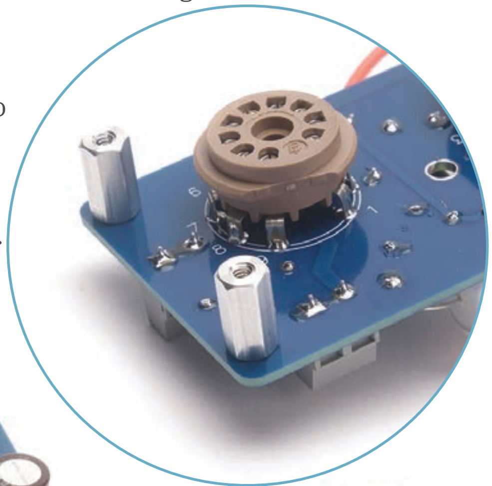
6. There is one resistor left, a 5.6k 2W resistor to R15. This is part of an RC filter that smooths the B+ high voltage and drops it a bit further to be more appropriate for the input gain stage and phase inverter.

7. Next up is the capacitors. I love capacitors. I can't wait to tell you about my favorites.

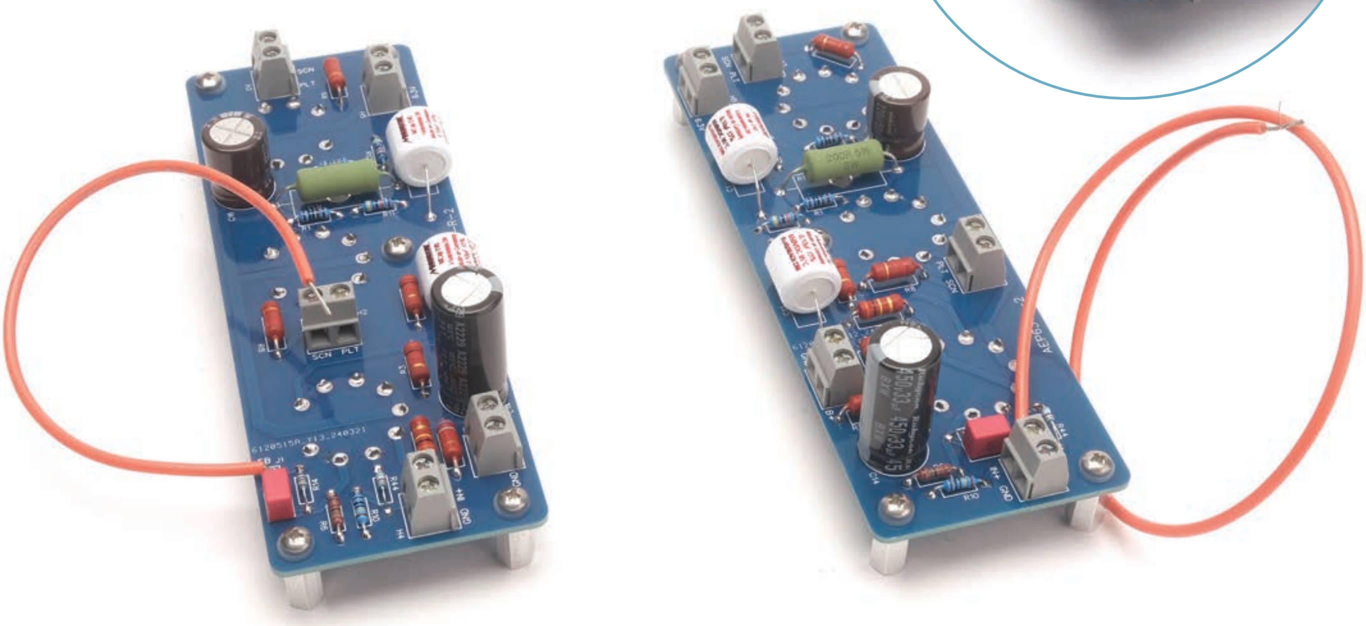
    a. The cutest little one is the 68pF capacitor to C13. This is part of the feedback network, it helps stabilize the amplifier at extremely high frequencies above the audible band, by allowing them to bypass the feedback resistor so the amplifier rolls off at those high frequencies and doesn't risk oscillation. This is a film capacitor with no polarity, so direction doesn't matter.
    b. We'll put a cathode bypass capacitor 220uF 50V on C16. This optimizes our output gain and low frequency extension. It's polarized, so make SURE you get the positive and negative sides aligned properly on the board.
    c. We need to couple the phase inverter to the output tubes using coupling capacitors. I love these. Solder the 0.1uF coupling capacitors to C1 and C12. These are film capacitors and do not have polarity so you can solder them in either orientation. Note that I do offer different variations of this amp, so the coupling caps you have may or may not look like the ones in the picture or diagram. I love these Mundorf ones that look like little marshmallows! One day I think I might just pop one into my mouth.
    d. Finally, the 33uF 450V filter capacitor to C14. It goes with the resistor to keep the high voltage really clean for the input gain stage. This is electrolytic polarized, so unless you like firecrackers and hum, make sure to get the positive and negative sides correct!

8. Instead of wiring the feedback network using a terminal block, you'll solder a feedback wire directly to the board. These will connect from the output transformer back to the feedback resistor going into the cathode of the input gain stage. Cut and strip an orange wire 11.5" long for the Left board, and an orange wire 7" long for the Right board. Solder to J1.

9. Last step for these boards is to put some spacers on. These will be used to screw the board to the chassis at the right distance so the tube sockets are aligned with the holes. Use 5 of the aluminum hex spacers and attach them to the same side as the tube sockets, securing from the other side using #4 nylon washers and ¼" #4 pan head screws. Nice and snug. Give yourself a hug.



All set. Repeat the instructions above for both boards. Inspect and clean the boards. Can you believe how much you've done already?



STEP 3 - Chassis Side Panels

You'll start to prepare the chassis. Mount the wood side panels using two 3/8" #6 truss head screws on each panel. The wood pieces are cut with a sloped angle, so orient them so that they slope backward away from the front of the amplifier. Nice and tight, you don't want them to slip later. Cool, man. Grab your surfboard and meet me at the beach.

STEP 4 - Chassis Back Panel Components

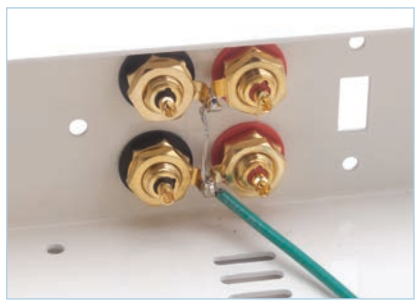
Ok, the back of the chassis has various components you'll attach.

1. Mount the IEC Power Inlet using the #4 oval head screws, lock washers and nuts. Orient it so that the middle earth ground post is in the downward direction (toward the top of the chassis when it's upside down).
2. Attach the #6 7/16" ground bolt with the star lock washer and nut. This goes in the hole next to the IEC connector and you'll notice some of the powder coat of the chassis is scraped off here. This is a safety earth connection for the chassis, in case any high voltage were ever to connect to the metal of the chassis, it will be grounded to the IEC earth connection. This also allows the chassis to act as a shield by being at a ground potential relative to the circuit.

3. Mount the RCA jacks in the four holes, putting the black jacks in the holes marked for Left and the red jacks for the ones marked Right. Admittedly, these are a little difficult to put in because there's not much space. Be patient, you'll get it. The jacks have plastic washers that need to go on each side of the chassis that isolates the jacks from the chassis. There are two nuts for each jack. Orient the jack so that the solder cup faces upward as you look at it and tighten the first nut as secure as you can get it. I use a small socket wrench, or mini adjustable wrench. Then put on the ring terminal that will be for the common ground of the input signal. There's no right or wrong way to do this, but it is difficult to wire up later, so my advice is to bend the leads slightly so that you can align them in pairs in the middle. See my photos below. Tighten the second nut, trying to hold them in place so the terminals and jacks don't rotate while you tighten.
4. Now will be the best time to solder a wire to those ring terminals; things get crowded later. Trim a green wire 6" long, and strip one end normally with about 1/2" exposed wire, but on the other end strip it very long, like 1" or even a little longer. What you will do is feed this through the bottom pair of the common ground terminals and then up through the other, and solder these together. I like to put a very tiny amount of solder on the very tip of the wire first, so the strands don't come apart--but don't do much! The solder will wick into the wire and make it hard to bend. See my photo. Take your time and get this nice and clean. This will eventually connect the input signal to the circuit ground, and we'll later connect some shielding to this wire. Ok, I owe you a beer after these RCA jack steps. Not easy, I know.

a. Mount the jacks so the solder cups face up, the insulating washers are centered, and secure with the first nut.

b. Bend the ring terminals slightly and orient them in pairs to the middle so you can eventually put a wire through the pairs of holes. Secure with the second nut.



c. Run the long-stripped wire through both pairs of terminals, connecting the common grounds all together. Solder both pairs.

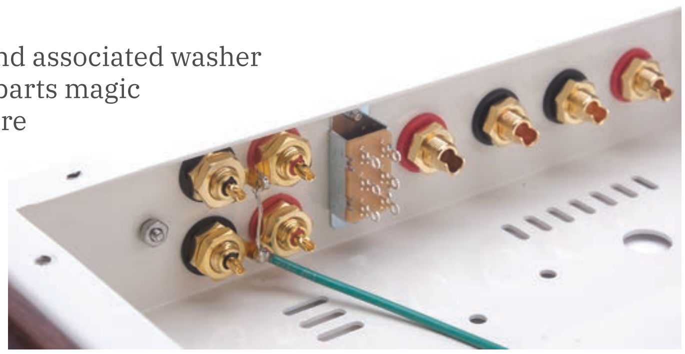
5. Mount the speaker binding posts, putting the black posts in the middle and red on the left and right sides. There are two different versions of these posts I include with different kits. Use the instructions matching the ones you have.

   Option 1 - These posts each have a pair of plastic insulating shoulder washers that go on the inside and outside of the chassis. The raised center area keeps the washer and post from moving around inside the hole so the post doesn't make contact with the chassis. On a thicker chassis, you could have both of the washers facing inward to the hole, but this chassis wall isn't thick enough for both of these raised areas inside the hole. So the washer on the exterior side of the amplifier will have the raised edge go into the hole, while the washer on the interior side of the amplifier will have the raised edge facing the nut and the flat side against the chassis wall. Align so the solder cup areas face upward. Tighten securely.

   Option 2 - These posts have a pair of plastic insulating washers that go on the inside and outside of the chassis. One of them is a shoulder washer with a raised area that keeps the washer and post from moving around inside the hole so the post doesn't make contact with the chassis. Put this one on the binding post on the outside of the chassis, and the other flat plastic washer on the interior. These posts have a screw that allows you to tighten a wire later. For now, the screw needs to be in enough to not obstruct the binding post nut. Align the screw in the upward direction so that later you can reach it with a screwdriver. Tighten the nuts securely.

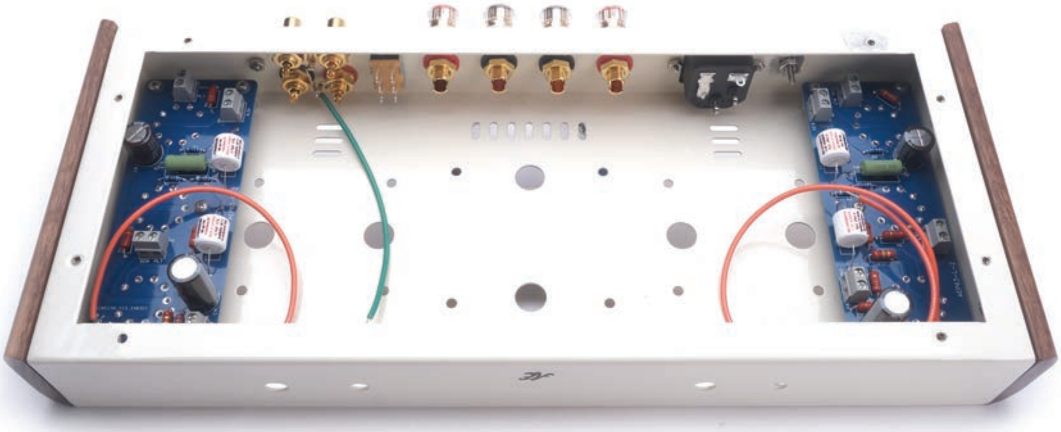
6. Mount the impedance switch. The holes in the switch are threaded, so you just need to put the M3 screws through the back of the chassis and tighten in place. Give the switch a flip or two and pretend something cool happens.

7. Attach the #4 1/4" Magic Screw. This screw and associated washer and nut are made with a special alloy that imparts magic sound in your amplifier. Ok, not really. They are just to fill in an unused hole on the side of the chassis near the RCA jacks. I put this hole here as a secondary option for how the input and main audio circuit could be grounded to the chassis, but as you'll see I chose to just run a wire over to the other side of the power connector.



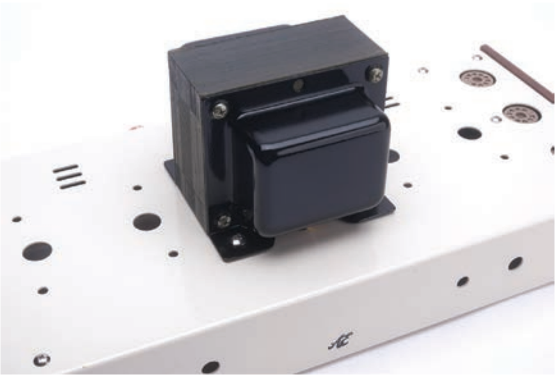
STEP 5 - Mount Left and Right PCBs in the Chassis

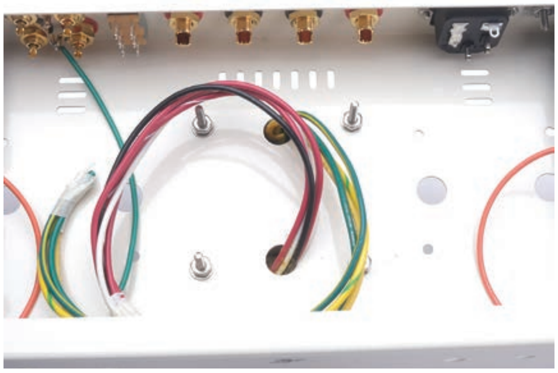
Now it's time to put the left and right channel PCBs into the chassis. You already prepared them with standoffs that should align to the holes. When you put them in the chassis, you'll need to drop them in at an angle and then twist to fit in place. You'll use 1/4" #4 truss head screws to tighten in place. Notice there's a small amount of tolerance in the holes so you'll want to align them so the tube sockets appear centered in their holes. Put all the screws in loosely until you have it positioned right, then tighten them snug, but not overtightened. Get yourself a nice lemon-lime soda and chill. Tell your family you are pretty awesome.



STEP 6 - Mount the Power Transformer

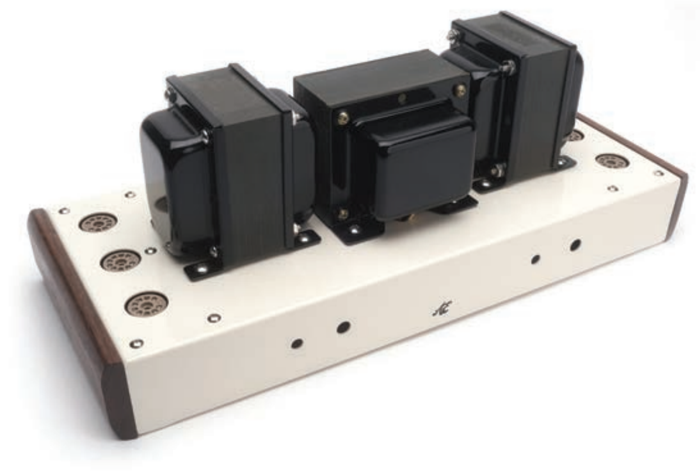
Let's get the big boy on here and then we'll really be getting going! The Power Transformer is mounted with 7/8" #6 truss head screws. Red wires toward the front of the amplifier and green & yellow wires in the back. You will put black fiber washers in between the transformer and the chassis to absorb a little physical vibration and to protect the chassis surface a bit. The interior gets a flat washer, then a lock washer, then a nut. This can be a little tricky to manage. You can either put the transformer upside down, set the fiber washers on it, and lower the chassis onto it, running the wires through the holes, or try it the other way, lowering the transformer onto the chassis with the washers in place. Align it so it is straight and tighten the nuts on the interior using a nut driver or wrench; don't tighten by turning the screws because they could damage the paint finish of the transformer as they move.





STEP 7 - Mount the Output Transformers

Output Transformers are mounted in a similar way to the power transformer. They will use 1/2” #6 truss head screws, also with fiber washers on the exterior, and a flat washer, lock washer and nut on the interior. The orange, black, and brown wires should be in the holes closer to the center of the amp. Align so they are straight and tighten down. It’s starting to look like an amp!

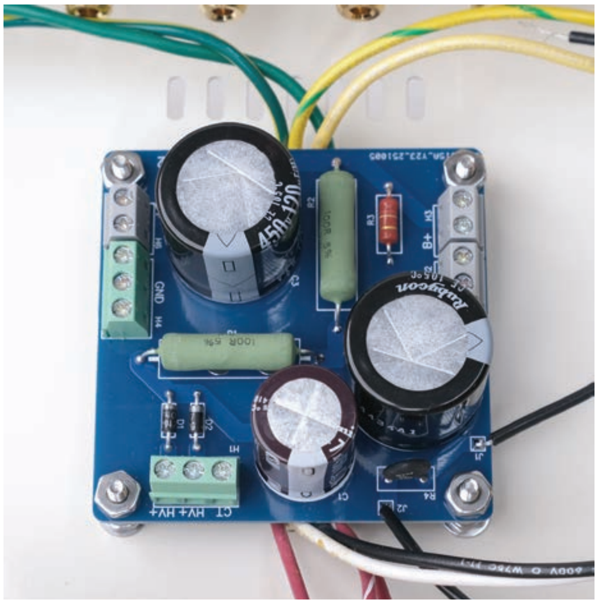


STEP 8 - Mount the Power Supply PCB

The Power Supply PCB is mounted on the same screws used to mount the power transformer. First, put on the 1/4” nylon spacers which gives some separation space. Then put the PCB down so that the terminal blocks are aligned as shown in the photo--one set of terminals will face either side of the amp, and the third set faces the front of the amp.

As you put the PCB in place, neatly align the power transformer wires underneath it. The yellow wires will eventually go to one side and the green wires to the other. These are the 6.3V supplies for the tube filaments. The red wires are the high voltage supply and will go together to the front of the PCB. The black and white wires are the primary side (mains) voltage, and will go toward the right-hand side of the amp as you look at it from this orientation.

Use #6 nylon washers and nuts to gently secure the PCB in place. Make your best “power-up” sound effect.

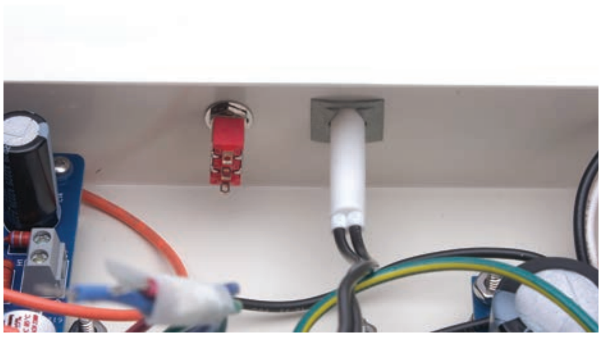


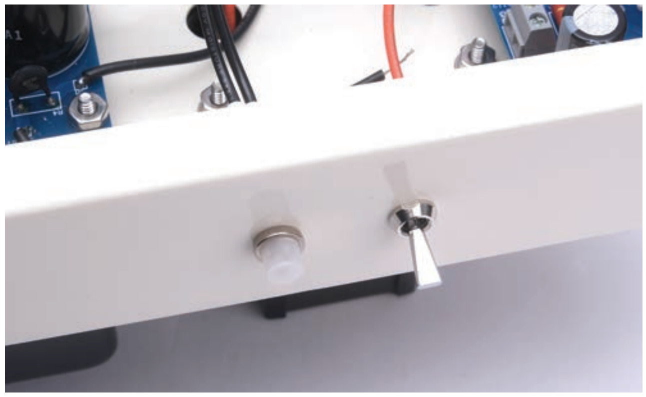
STEP 9 - Power Switch and Indicator Lamp

On the front of the chassis, install the power switch and power indicator lamp.

The power switch has a nut and lock washer that go on the interior of the chassis, and a decorative nut goes on the exterior, with a flatted side for a wrench to tighten it. Adjust the interior nut so the right depth of threads are exposed--the decorative nut has a limit to how deep it can go. There's an additional washer that I usually don't use. It can protect the chassis surface, but if you are careful you can tighten the outer nut securely without damaging the surface, and I think it looks better without the washer.

The indicator lamp has a flexible metal clamp that just pushes up the plastic surface to hold it in place. It's hard to push on or remove once you've pressed it in place, so just do it when you are ready! Get it snug so the lamp doesn't move around.

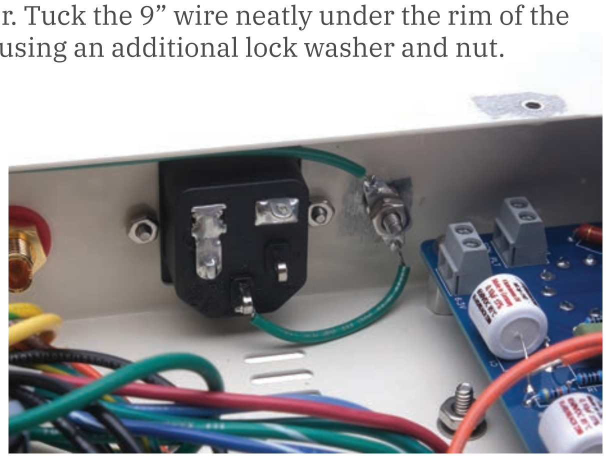




STEP 10 - Safety Earth and Input Ground Wires

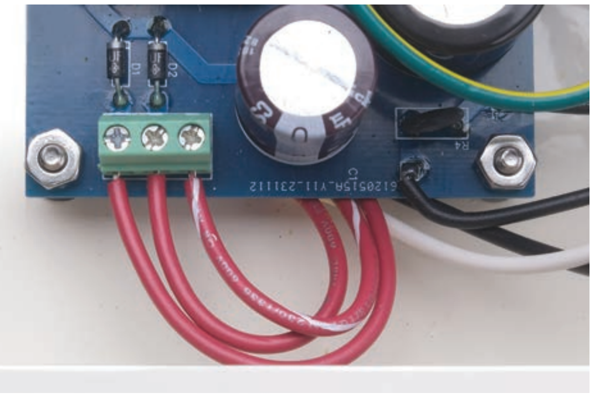
The chassis must be grounded to the IEC earth ground for safety reasons. If there were ever a fault that put high voltage onto the chassis, this will send it to ground and blow a fuse. The chassis is powder coated, which provides a layer of safety, but that is not enough. The amplifier circuit also must have a connection to this ground point, and we'll make that connection at the RCA input jack point.

Prepare two green wires, one 2.5" and the other 9", and solder a ring terminal on the end. The ring terminals will go on the chassis ground bolt next to the IEC connector. Solder the short wire to the IEC earth ground, which is the one in the center bottom. Solder the other end of the 9" wire to the RCA input jacks common wire that connects the four jacks together. Tuck the 9" wire neatly under the rim of the chassis. Secure the ring terminals onto the ground bolt using an additional lock washer and nut.



STEP 11 - Wire the Power Transformer Secondary Power Wires

The power transformer takes the household mains voltage (120V in North America) and transforms it to other voltage levels needed by the amplifier. The tube filaments (heaters) run at 6.3V AC. The main amplifier circuit rectifies AC to DC and runs at about 360V DC after filtering. The input of 120V is considered the "primary" side of the transformer, and the output voltages are called the "secondary" side. This step will wire these secondary voltages.



The high voltage wires use a "center-tap" which means there are two red wires that carry the AC voltage, and a red/white stripe center tap wire that is zero volts in between these two. The transformer manufacturer would refer to this as 330-0-330 volts. Meaning each leg of the secondary is 330V relative to the center tap (or 660V relative to one another). Part 2 of these instructions explains more how this is rectified and filtered to a DC voltage used by the circuit. On the power supply PCB, trim and connect the red wires to the left two terminal block connections as shown in the photo here, and the red/white stripe wire to the right terminal block connection. You'll see how the red wires correspond to the rectifier diodes. The center tap wire is connected to our circuit 0V ground potential. Trim the wires relatively short, but not so short they are too difficult to work with.

The left and right channel will each get their own wires for the tube filaments. The filaments draw a lot of AC current, which will cause a magnetic field to form around the wires, and if you aren't neat about it, this can cause hum if your sensitive audio signal picks it up. Twisting AC wires together keeps the wires close to one another using the opposing phase of the wires to help offset this magnetic field. It also just helps keep your wiring neatly arranged. Twist the yellow and yellow/green-stripe wire together and trim and strip the wires to a length so they reach over to one of the PCB terminal blocks marked for 6.3V. Do the same for the green and green/yellow-stripe wire on the other side. Trim the wires down so they reach comfortably. They shouldn't be so short that they barely reach, but you also don't want excess wire meandering around inside your chassis like a beach bum loafing around your precious amp.

You may need to first loosen the terminal block screws to allow the wire to enter, then tighten down securely with a small screwdriver. You want a very good connection on all of these. Strip your leads just long enough to reach inside the terminal block, but not so long that there is excessive exposed wire. Twist the lead strands so they stays neatly together, with no stray strand anywhere. The red wires in particular are extremely high voltage, so you want to be very careful!

Throughout, remember you can refer to the layout diagram in the back of this manual to double check everything is looking correct!

STEP 12 - Wire the Impedance Switch and Speaker Binding Posts

The output transformers have multiple taps to allow for speakers that are 8 ohms (orange wire) or 4 ohms (brown wire) impedance. The amplifier has a slide switch that will let you choose between these. So the output transformer signals will go to the outer switch terminals, and the center of the switch will go to the red (positive) speaker binding posts using some additional **white 20 awg wire**. The switch is dual-pole dual-throw (DPDT) meaning it allows two different circuits to be used (left channel and right channel) and each one has two different connections that can be made (8 ohm or 4 ohm).

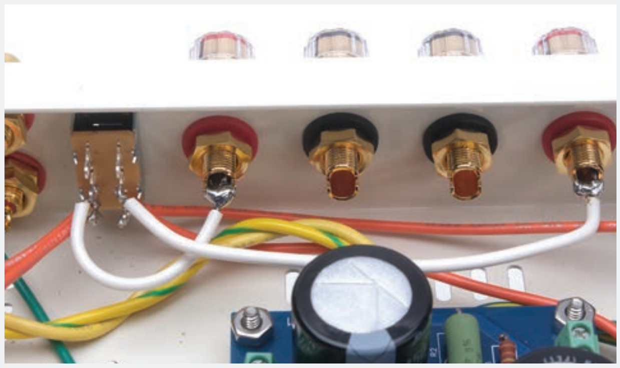
The black wire from the output transformer is the common wire and will go to the black binding posts. This will also need to be connected to circuit ground (0 volts). Cut and strip two 4” green wires that will connect to the power supply PCB ground terminals. Note how on each side of the PCB there is a terminal block marked with GND. The one on the left (H4) has 3 terminal connectors and the one on the right (H2) has two. These blocks are ground connections, you can choose any of the connectors on this block and they are all 0 volts ground.

I like to twist the stripped leads together first to make it a little easier to work with--you would take the black lead from the output transformer, twist it with one end of the green wire, and this is connected to the black speaker binding post. The other end of the green wire is connected to the ground terminal on the PCB.

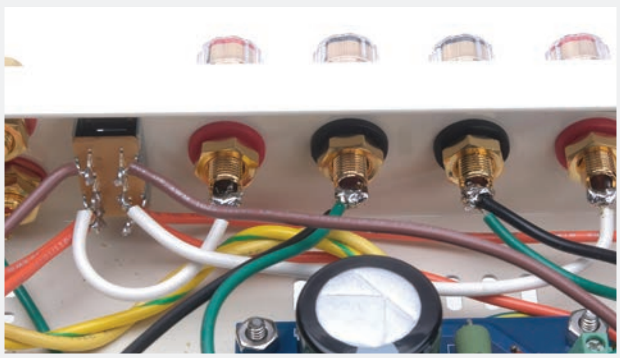
Additionally, the amplifier uses a negative-feedback loop to improve performance and this will always be connected to the 8 ohm tap, regardless of which speaker impedance is being used. This is the orange wire you connected to the left and right amplifier PCBs. This will get soldered to a terminal hole on the slide switch so it’s always connected to the 8 ohm tap.

Depending on which version of the kit you purchased, you will have different binding posts. One type requires the wires to be soldered directly into the cup of the binding post. It acts like a big heat sink, so do something to hold the wire steady in contact

1. Connect the orange feedback wire from the PCB, and orange 8 ohm wire from the output transformer to the bottom lugs of the switch.



2. Use 2.5” and 5” pieces of 20 awg wire to connect the middle lugs to the red speaker binding posts.



3. Connect the brown 4 ohm wires from the output transformer to the top lugs of the switch.

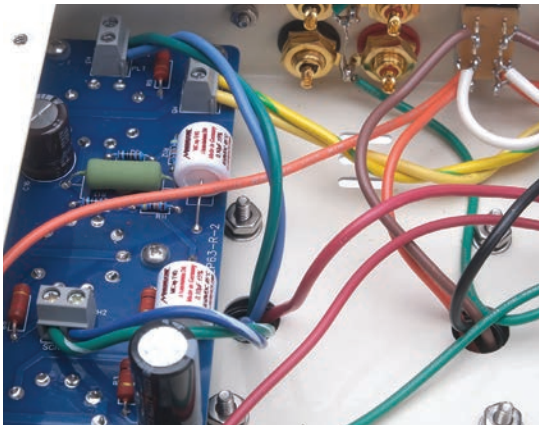
4. Connect the black wires from the output transformer along with green wires from the power supply PCB ground to the black binding posts.

with the post while you feed in quite a bit of solder until you have covered it well enough to make a good connection. Don't hold your iron too long--you don't want to get the post so hot that you start to melt the plastic washers. The other type of post has a screw connection. You can loosen the screw, insert the wire, and tighten down.

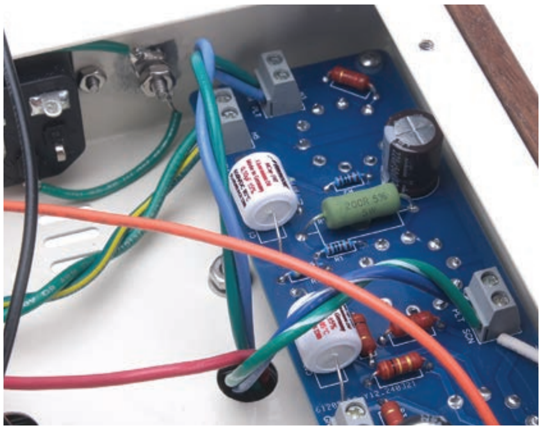
Solder and connect these all up as shown in the pictures here or the wiring diagram at the end of the manual. Trim any excess leads and inspect for good connections. Be aware that the casing of the switch has small metal clasps that wrap around the edge of the switch somewhat close to the terminals. Don't let any wire or solder touch those, otherwise you could accidentally be grounding the signal to the chassis.

STEP 13 - Wire the Primary Side of Output Transformers

You already wired the secondary side of the output transformers which end up going to the speakers. A push-pull output transformer has several wires on the primary side. The B+ high voltage goes into a center tap, and then there are coils that go to the plate (anode) and screen of each of the two output tubes, in opposite phase. The second part of this instruction manual will explain in more detail, but each tube works together but in opposite phase--one "pushing" while the other "pulls." By winding the coils in opposite directions, the output transformer will do the magic to recombine these two AC currents back into a single signal on the output side.



The blue wires will be for the plates, and the green wires for the screens of the tubes. Since we have two tubes, one set of these wires has a white stripe and the other pair will be solid colored. Keep the solid colors together and the stripe wires together--they are a pair to go to each tube. I like to twist them together just to keep them neat. Notice how the PCB is marked with "PLT" for plate, and "SCN" for screen. Connect the blue or blue-white-stripe wires to the plate connections, and the green or green-white-stripe wires to the screen connections. It's important to put the striped ones with the tube toward the front of the amplifier. Since negative feedback is used, if you get these reversed then you would end up with positive feedback, causing the amp to oscillate and squeal like an evil banshee. As always, trim the wires so they are a relatively short length, but not too short.

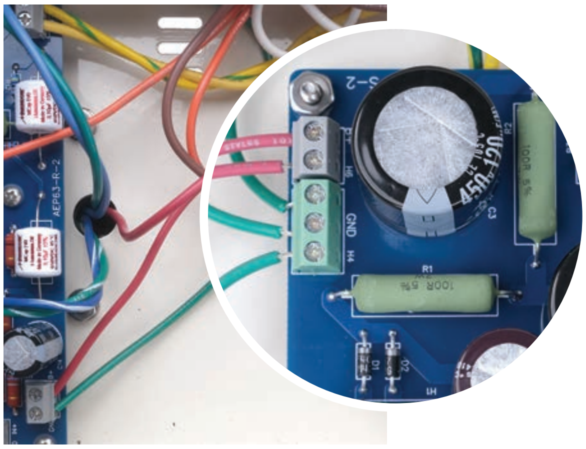


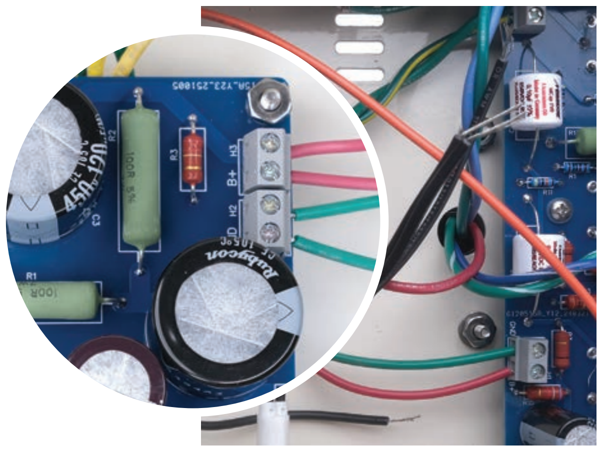
Now I have to confess, a few of these terminal blocks are difficult to reach because of the overhang of the edge of the chassis. A small screwdriver will be best, and you'll just need to angle it a bit to reach in there. Make sure you get these wires tight despite the difficulty in reaching the screw. Inspect closely, tug them a bit to make sure they are secure.

The red wire of the output transformer goes to one of the B+ terminal connections on the power supply PCB on the side facing that channel. There is a second terminal for an additional B+ wire, and a separate terminal block for ground connections. Cut and trim a 5" red wire and a 5" green wire for each channel. These will bring the B+ high voltage and ground connections to the left and right PCB terminal blocks that accept B+ and Ground. Notice that the left and right PCBs orient ground and B+ differently, so be sure to check to get these right.

Also, on the side with the RCA jacks, now is a good time to connect the existing green wire you already soldered to the RCA jack common ground to the power supply PCB ground on the left side in the extra terminal spot. See what's happening here? We are giving a ground connection to each channel's PCB circuit, a ground connection to each of the speaker outputs, and we connecting this central ground point to the RCA input ground (which is sort of the master ground point and this was further sent to earth ground in a previous step). If you are curious, the power supply PCB already connects the left and right side grounds together since they are going to the same power supply circuit.

Take a look at the layout diagram in the back of this manual and the photos here to make sure you have these right.



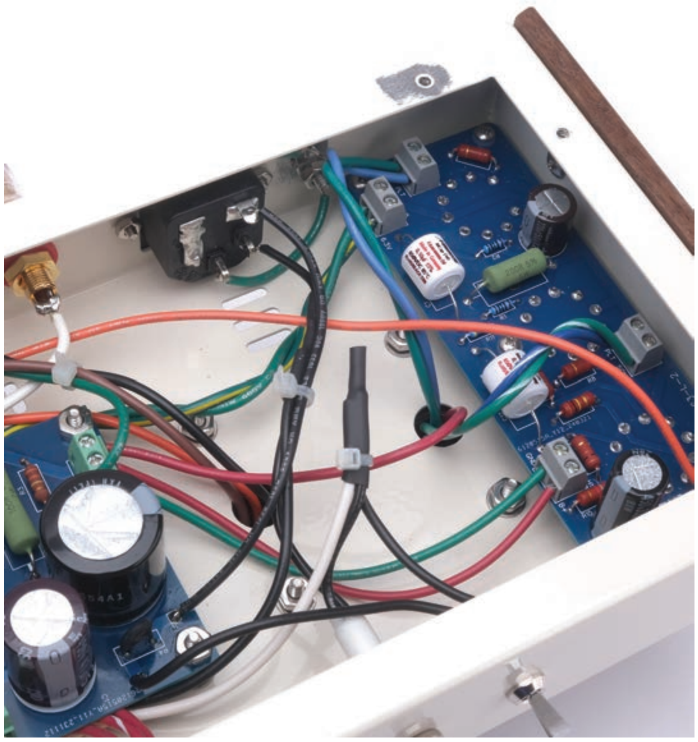


STEP 14 - Power the Primary Power Supply, Switch and Lamp

Your household mains voltage (120V in North America) is going to go from the IEC power inlet through a NTC thermistor and power switch and will then power both the power transformer and the indicator lamp.

A NTC (negative temperature coefficient) thermistor is a device that acts as a resistor when cold, and as it heats up, the resistance drops to near zero. This is a way to slow down the inrush current that will come the moment the amp is switched on as the iron core of the power transformer is magnetized and the capacitors are charged up. You soldered this onto the power supply PCB in a previous step.

Wire as follows:



1. The shorter black wire from the NTC thermistor on the PCB to the middle lug of the power switch.
2. Cut and strip a 3” black wire and solder one end to the top lug of the power switch.
3. Trim and strip one lead of the indicator lamp down shorter. Trim and strip the white wire from the power transformer down shorter. Twist these together with the 3” black wire from the power switch. Solder all three of these wires together, use just enough to run some solder into the wire for a strong connection, no need to make it a huge blob. Cover with heat shrink tubing for safety and use a heat gun or hold the barrel of your soldering iron close (without melting anything!) to shrink the tubing tight. Add a zip tie to give extra strength to hold this bundle together. See what we did here? This is the live 120V AC after the power switch going into both the power transformer and the lamp.

4. Trim and strip the black lead from the power transformer to the right length to reach the IEC power inlet. Twist this together with the other lead from the indicator lamp. Solder these together to the neutral lug of the IEC power inlet. That’s the bottom right lug shown in the picture. (In my photo here, my lamp lead didn’t quite reach, so I merged and extended this with a little extra black wire. No problemo, just be sure to cover any merged wires with heat shrink tubing.)
5. Solder the black wire from the thermistor on the PCB to the live lug on the IEC power inlet. That’s the upper right lug shown in the picture.

You can use another zip tie to neaten up these power wires.

That’s it. You have mains voltage coming into the IEC inlet (going through a fuse first that is integrated into the inlet) then to the thermistor, then to the power switch, then to the power transformer and lamp, and then the circuit is closed from the power transformer and lamp back to the neutral AC inlet. Juice, baby.

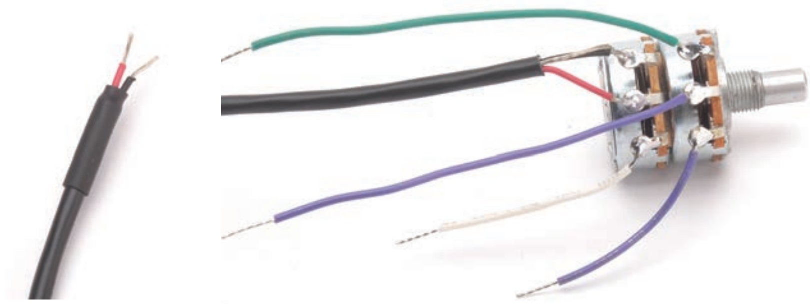
STEP 15 - Prepare the Volume Potentiometer and Input Switch

We have not yet mounted the potentiometer (volume control) or the dual channel switch that toggles between the two sets of inputs. It’s too hard to wire the potentiometer when it is already inside the amp chassis, so you will prepare it ahead of time and then attach it. This is a “dual gang” potentiometer where one knob controls both left and right channels.

Cut and strip a 2” white 24 awg wire, 2” purple wire, 3.5” purple wire, and 3.5” green wire. Solder to the potentiometer exactly as shown in the photo here. I like to use a helping-hands tool to hold the pot and wire steady. The bottom lug here is the input signal and gets the shorter white and purple wires. The middle lug is the attenuated output signal and gets the longer purple wire. The top lug is the ground and gets the green wire.

For the short distance from the potentiometer to the nearest PCB channel, it's ok to use unshielded wires. But for long runs of wire that carry an audio signal, you usually want to shield the wire from interference. So for one channel you will use a piece of 10" shielded 2-conductor wire. Cut the outer layer carefully without cutting into the inner wires--it doesn't take much to remove it. You'll see a bare shield wire. On one end, expose a small amount of the red and black wires, and cut off the shield wire completely. (We will shield the wire only on one end so we don't create a ground loop.) Apply a small bit of heat shrink tubing on the end so the shield doesn't accidentally touch anything. This end is eventually going to the PCB on the other side of the chassis.

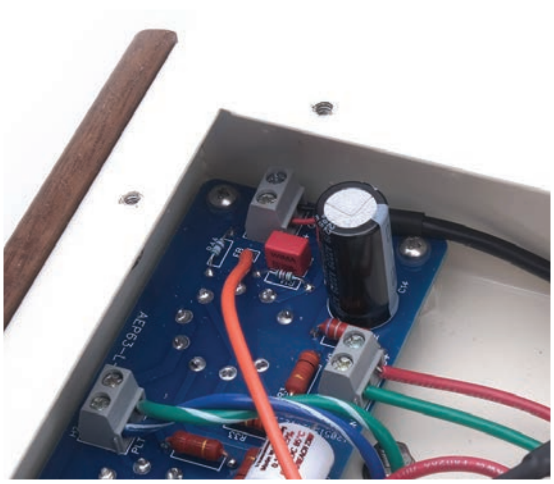
The other end of the shielded wire goes to the potentiometer. Cut some of the outer casing off, and on this side, trim and twist together the shield wire with the black lead. Solder those together to the ground lug of the potentiometer as shown here. Solder the red wire to the middle lug. Red will carry the signal, black will carry ground.



After you have prepared the potentiometer and wires, mount them in the chassis as shown. See how the potentiometer wires are rotated to face the switch, and the switch is oriented sideways (you could optionally orient it vertically if you prefer your switch to go in the up & down direction). The potentiometer has a flat washer and nut to tighten on the outside of the chassis and a star lock washer that goes on the inside. The toggle switch mounts the same way as the power switch, with a decorative nut on the outside, and a nut and lock washer on the inside that can adjust the depth of thread exposed.

Route the green and longer purple wire under the potentiometer over to the PCB and securely tighten into the terminal block, with the purple wire to the terminal marked "IN+" and green wire to the terminal marked "GND". Route the shielded wire along the edge of the chassis over to the other PCB and do the same thing, with the red wire to "IN+" and black wire to "GND" on that terminal block.

This will be how your audio signal gets to each of the amplifier channel circuits after going through the potentiometer.



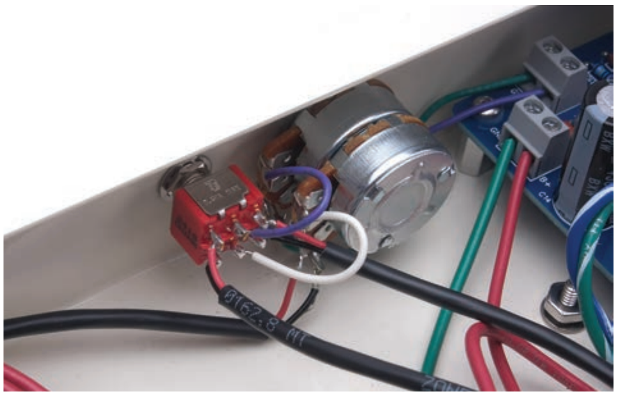
STEP 16 - Wire the Input Selector Switch

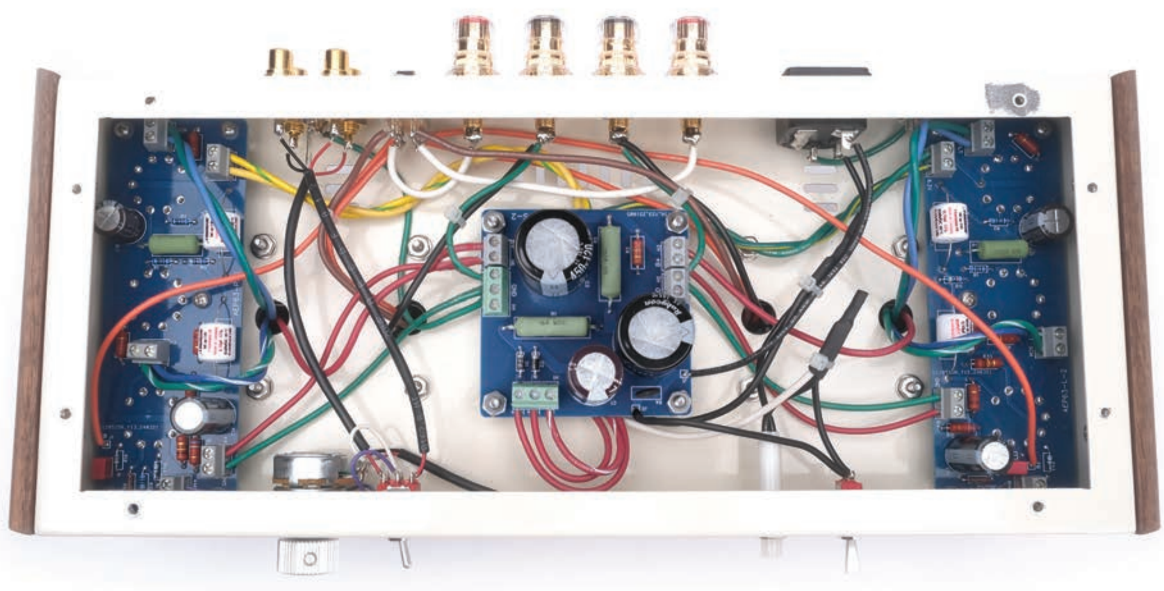
Now that you have the switch mounted and potentiometer wires ready, you can finish the input wiring. Prepare two 5.5” lengths of 2-conductor shielded wire. On one end, remove about 1” of outer insulation, strip the red and black wires, and keep the shield wire. This will go to the RCA jacks. The other end will go to the selector switch. Remove only about 1/2” of the outer insulation and either cut off the shield wire or you could just pull back the wire from the other end, so it’s a little longer. As a reminder, we ground only one end of a wire like this to act as a shield for interference.

At the RCA jacks, prepare the wires to align to their respective black and red jacks. The wires fit into the small center cups of the jack and you put a little solder in place. You may need a technique to hold the wire steady. The shielded wire needs to reach to the center common conductor. I just wrap it around an open spot and drop a little solder in place, then trim any excess. It’s a little tricky to work in this small space. Start with the bottom wire and RCA jacks, then do the top pair.

At the switch, solder the red wires on the top lug and black wires on the bottom, when the switch is oriented as shown in the photo. It’s important to get these to the correct end, so that when you look at the front of the amp when it is right side up, the switch in the left position is input 1 and in the right position it is input 2. As you look at the photo here, input 1 wires should go to the right hand side, and input 2 goes to the left. It isn’t entirely intuitive to the switch, but when the switch toggle moves toward one end, the opposite end is switched on.

The output of the switch comes from the middle lugs and will go to the potentiometer, so solder the purple wire on the middle top lug, and the white wire on the middle bottom lug.



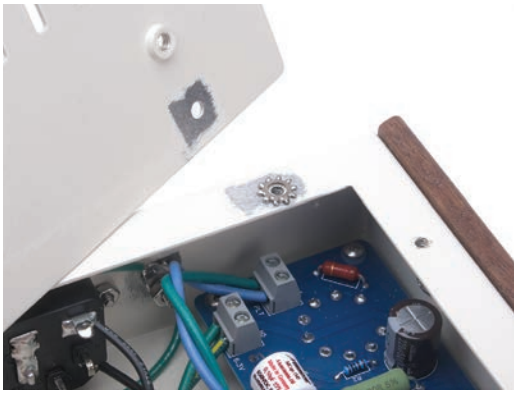


STEP 17 - Review and Close

You are almost done. Time for a full review. Use the layout diagram in the back of this manual and trace through every part in the amp. Make sure you have completed all steps, soldering is clean and neat, wires are organized, nothing is touching things it shouldn't touch, etc. I like to step through a logical flow of electrical current. In a point-to-point amp this is easier than one with PCBs, but we'll do a form of it here.

- ☐ Compare the diagram with your amp. Everything looks identical?
- ☐ Power comes in the IEC connector, goes from the live terminal after the fuse to the thermistor, to the power switch, then to both the lamp and the power transformer? Power returns to the neutral lug of the IEC from the power transformer and lamp?
- ☐ The IEC earth ground securely connected to the chassis bolt? There's a ground connection from the RCA jacks?
- ☐ Your power supply board has all the components on it? Your high voltage red wires are in the correct terminals? Your ground wires are going to the correct places? The terminal screws are tight so the wires are secure?
- ☐ The 6.3V power supply wires go to the correct terminals?
- ☐ Your output transformers are wired correctly on the secondary side to the impedance switch and then speaker binding posts? They are wired correctly on the primary side to the screen and plate terminals? Double check all of these carefully.
- ☐ Your input signals are coming in the RCA jacks, going to the toggle switch, then the potentiometer, then each of the PCBs for left and right channel?
- ☐ Your left and right channel PCBs have all their components in place? The terminal screws are all tight so the wires are secure?
- ☐ No little bits of wire or solder are left anywhere? Shake it out if you need to!
- ☐ You have had a good time? You complimented yourself on what a great job you did?

Cool! Time to close up the bottom of the chassis. There is a star lock washer that should go between the chassis and the bottom panel on the spot that has exposed metal. See how the bottom panel has a spot and the chassis does also? The lock washer adds a bit of insurance that these touch each other. We want the entire chassis grounded. Align these carefully, then secure the bottom with the 8 chassis panel screws. Also screw in the 4 rubber chassis feet. Apply the high voltage warning sticker. Just want people to be safe!



STEP 18 - Volume Knob, Tubes, Fuse

My previous photos show the **volume knob** already attached, but to be specific for the instructions, take the step to do this! Different versions of the kit include slightly different volume knobs, but they all have some type of screw to tighten onto the potentiometer shaft. You'll want to get it aligned so that when the volume is at the minimum (fully counter-clockwise) or maximum (fully clockwise) then the indicator line is where you expect it to be. Tighten it down.

Give it a test spin and pretend like when you had your stereo at mid-volume playing AC/DC, then Mom yelled up to your room, "Turn that down!" and then you cranked it all the way up! Heh.

You're also ready to install the tubes. The smaller **12DW7 tubes** go in the sockets closest to the front of the amplifier. The **EL84 tubes** go in the two sockets toward the rear of the amplifier. Since this is a push-pull amplifier, you really should be using matched tubes for the EL84 so that the tubes are optimally working together in each channel, left and right. I include a matched quad with the kit, meaning all four tubes should be matched, so you don't have to keep track of left vs. right. Any of those four EL84 tubes can go in any of the four rear sockets.

Volume knob

Finally, insert a **2A slow-blow fuse** into the small tray in the IEC inlet. Most tube amps would use a slow blowing fuse because at the moment of start-up there can be a surge of current as the transformers and capacitors charge up and you don't want the fuse to blow unless the amp is already in its steady operating state and then has current over the expected rating. You can use a small flat screwdriver to get the fuse tray to open and slide out. The fuse snaps into a holder in the part of the tray that pushes in farthest to the inlet. Don't get confused and put it in the other section that is really just a box meant to hold a spare fuse.

Fuse tray

You are done! Take a minute to admire your work. Go tell your family or friends how awesome you are because you built a tube amp.

Final finished amp!

Testing the Amp

Plug in a line level audio input signal to the RCA jacks. This could be a CD player, output from a mobile device, a source from a preamplifier, etc. You will need your own RCA interconnect cables.

Connect 4 or 8 ohm speakers to the speaker binding posts. Do not run the amplifier without speakers connected (or a dummy load resistor of a similar resistance and rated for 15 watts or more). Be sure the impedance switch on the back of the amp matches your speaker nominal impedance.

Ensure the power switch is in the off position and plug in the AC power cord.

Now... I suggest you queue up a first-try song. I think for this amp, *Wouldn't It Be Nice* by the Beach Boys might be just perfect. But choose your own, whatever you like. Diana Krall. John Denver. Ozzy. Beatles. You know, whatever.

With the volume at a relatively low setting and an input signal playing, turn on the amplifier. You should see the tube filaments start to glow, and sound start to come up after 5-10 seconds. Watch for these things and turn it off immediately if you notice any:

- Strange and unexpected sounds
- Sparks, smoke or strong burning smells. (Some heat smells are normal as the tubes and components get to operating temperatures.)

Now celebrate! This is the time when you smile. You did it, and should be proud of yourself, right? A bunch of resistors, capacitors, transformers, and tubes, and you get this beautiful music coming out. How cool is that? Tell your friends you built your own amplifier. Turn it up. Get a gin & tonic to celebrate and relax for a while.

You. Are. Awesome!

Test source & speakers

I typically use dummy loads or crappy speakers that I got at a thrift store for $5 for testing until I know things work. And I use an old unused cellphone that plays some music as my source. Then after I know things are working properly, I move to my good speakers and source equipment for listening. It's up to you what you decide to connect, but I would not recommend wiring up expensive speakers and equipment for your first test.

A few quick notes on operating your amplifier:

- The tubes will get very hot in a few seconds, and the power transformer will get hot over a longer time period (maybe an hour or so). This is normal. Obviously you don't want to touch them or allow other things to be close to them. See the safety precautions on page 2.
- Remember it is not a high-power amplifier, so you may need to turn up the volume knob to get to sufficient listening levels, depending on your source signal. But do not overdrive the amplifier with a pre-amp. If you hear distortion, turn it down!
- Always turn off your amplifier when not in use. It is always drawing power and is not only a waste of electricity, but it could wear out your tubes prematurely and has risks if you leave a hot amplifier running for hours unattended.
- When others are in your home and look at your amplifier questioningly, just slip another album out of its cover, take a sip of your Tom Collins and say, "Yeah, it's a tube amplifier that I built myself." They may not understand, but that's cool. You can read Part 2 and tell them how a tube works.

Troubleshooting:

It didn't work? Aw man. Here are a few things to check:

- Your AC cord and connection all ok and definitely providing power?
- Speakers are connected properly?
- Tubes are installed fully?
- Input signal is actually being generated at expected line level voltage? (Input cannot be a turntable unless you are using a pre-amplifier with a phono stage)
- You have the input selector switch set for the input jacks you are using?
- Did the fuse blow? Take it out and check continuity. If the fuse blew, unplug the amplifier and review your circuit. There may be a short somewhere that caused heavy current; see if you can find it and correct it. I gave you a second fuse in case you need it, but don't just put it in and try again. You need to find and fix whatever was wrong.
- Unplug, open the amplifier, and trace back over all the elements of the schematic and assembly diagram. All looks right?
- The power transformer and output transformers are wired to the correct places?
- None of your polarized capacitors are wired in reverse? Your diodes are not wired in reverse?
- Your solder joints are secure, with no shorts, loose leads or wire strands, etc?
- Power switch seems to have continuity as expected?
- Input selector switch and volume potentiometer are wired properly, and signal coming out of it?
- RCA jacks, impedance switch and speaker terminals wired ok? Positive binding posts are insulated with plastic washers and not shorting to chassis?
- Ground bolt ring terminals connected and tightened properly?
- When turned on, do you see filaments glowing in the tubes? If not, tube filament wiring is right?

I am not providing any live-amplifier troubleshooting tips in these instructions. It is critical that you understand safety and testing techniques before you attempt to diagnose problems with a live circuit, and I can't describe that here or be certain builders of the kit have this type of expertise and safety awareness. You should be experienced and comfortable with trouble-shooting a high voltage electrical device to do it. Consult someone who has this experience and training if you have gone through the bullets above and still need help getting it to work. The schematic in the appendix does include typical operating voltages at key points. Never probe around in an operating amplifier if you are unfamiliar with the safety risks or what to look for!

Also, it is possible, but unlikely that a tube or other component is bad. This shouldn't be your first assumption of what's wrong. I have used reputable suppliers and high-quality components. It is most likely something in your physical circuit build. I will of course replace any component that is not working properly. If you are still having trouble, send an e-mail or check the website, and I'll see what I can do to help troubleshoot the issue. I want you to get this working, but at the same time, please know that I'm not there in person, safety is a top priority, and I may not be able to help in all cases.

This concludes the build section. The next section explains how this tube amplifier works, so you can learn in detail what is happening inside the tubes and throughout the circuit. I hope you'll read on, because this is the most empowering part of this kit—understanding how it works so you can do even more in the future!

PART 2
HOW A TUBE AMPLIFIER WORKS

To meet the objectives of this kit, not only inclusive of the assembly instructions necessary for a working amplifier, but to fulfill the learning objective of understanding how the parts and circuit work—this section will touch on some core concepts and then attempt a clear and simple explanation of the circuit. It's so fun to learn new things, you will love it!

I will not attempt a comprehensive explanation of all the physics, electronics theory and math that would make a more robust reference. I'm not best suited to do this, and there are excellent resources online or in books (some noted for reference at the end) that can teach the theory more extensively, and much broader than this one type of single-ended tube circuit you are building. This is intended for a non-technical audience, but I will assume that you have basic knowledge of electric circuits and components such as resistors and capacitors. If you already have a strong knowledge of the theory and function of a triode or pentode vacuum tube and amplifier circuit, you may have bought this kit without a learning objective, and this section may not be necessary for you. But if you want a better understanding of how tubes and this circuit works, read on!

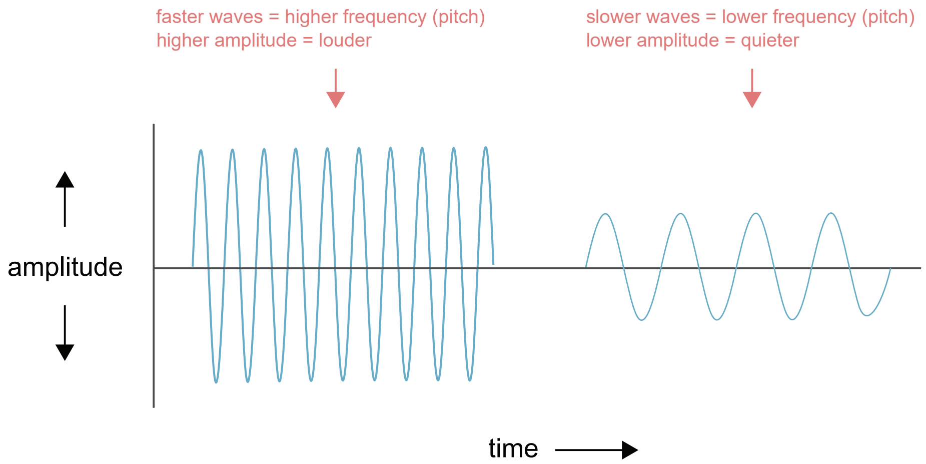
CONCEPTS

What is sound?

Sound is a type of energy created by vibrations. When things collide or vibrate, there is a physical process that moves the air around those things, and sound travels through the air as pressure waves that compress and expand based on the vibration source. Eventually, these sound waves reach and vibrate within our ears, creating signals that our brain interprets as music, speech, or other noises.

This barely describes the phenomenon, but a key takeaway is that sound waves can be caused by vibrations that go fast (high pitch) or slow (low pitch), and that move a small amount (quiet) or a lot (loud). We can describe basic sound waves using these characteristics—**frequency** and **amplitude**—and plot them on a chart.

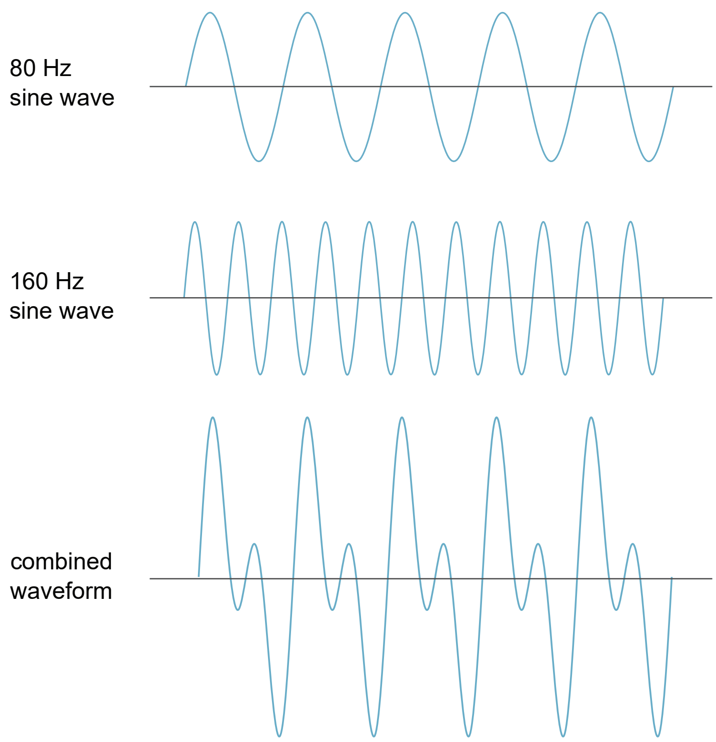
Electronic audio recording equipment converts actual sound waves moving through the air into electric signals using a microphone that reacts to the sound waves creating a voltage at a frequency and level corresponding to the sound. A recording device can capture this stream of varying voltage information, representing the sound waves, saving it for later. A playback device can take the stored information, convert it to a voltage signal again, which is then amplified to a larger, more powerful signal. We then need a device to turn it back into sound waves in the air. This is the loudspeaker, often made using a cone that is vibrated magnetically and this physically moves the air to generate sound waves that a listener can hear. There is much more to this entire process and the physics of loudspeakers, but the important concept is to understand that an audio signal in our amplifier will be a **voltage changing over time**.



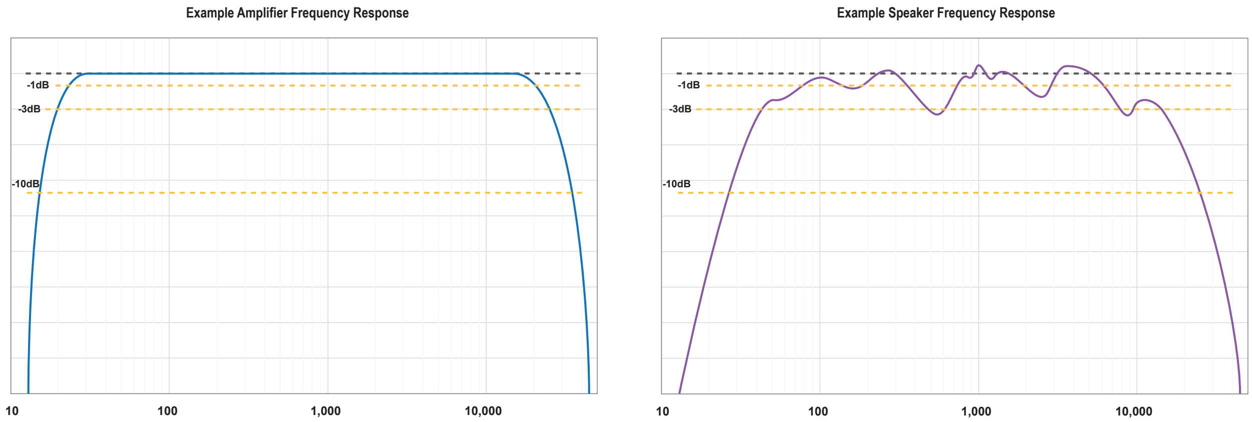
We have two ears that can each hear sound waves separately and interpret spacial location from them, and stereo recordings have two separate signals that are recorded or engineered, and eventually played back, one for each speaker to recreate a form of dimensional soundstage. In most cases through this manual, we will discuss a single process of audio amplification, but know that this process is duplicated, one for each **channel** (left and right).

Pure cycles of rising and falling amplitude over time are **sine waves** and can be described with math formulas that I won't introduce here. But you could think about sine waves as a type of building block that, in complex combinations of frequency, duration and amplitude, can make up music or other audible sound. Combinations of multiple waves will add to a compound wave, as shown in a simple example below. This will be important when we start to discuss distortion and harmonics.

Humans can hear sound only within certain frequencies. To cover human hearing adequately, most audio equipment is designed or evaluated in the range between 20Hz and 20,000Hz. Hertz (Hz) is a measure of the number of wave cycles per second. A vibration at 20 cycles per second is very slow, and we would probably be feeling this very low "thunder" more than we really hear it. On the other hand, 20,000 is such a high pitch that most of us cannot even hear it—especially as we age or lose hearing from exposure to loud noise or bad marching bands. Although it is rare that a music recording spans 20Hz-20kHz or that a loudspeaker could accurately reproduce that wide of a frequency spectrum, we still will design, measure and expect our amplifier to be able to reproduce as accurately as possible within this range of audio frequencies.



The charts of sine waves shown here (and visualized on an oscilloscope) are in the time domain, with frequency illustrated as the number of peaks and valleys over time on the horizontal axis. Another commonly used graph for audio measurements is in the frequency domain—a horizontal axis representing frequency from low to high, and the vertical showing a measurement of amplitude that a device may reproduce at each frequency.



The chart on the left here shows an example ideal flat frequency response that we might have with an amplifier: frequencies across the audible range of 20 – 20,000 Hz can be reproduced at nearly equal amplitude. The chart on the right shows an illustrative speaker frequency response—wow, it's very choppy, huh? There are some peaks and valleys where some frequencies are reproduced louder than others. This is typical of a speaker, where a very flat response is difficult to achieve and involves a wide range of physical implications in the action of the speaker drivers (woofer, midrange, tweeter), crossover points where drivers are transitioning responsibility, diffraction of sound from the shape of the speaker, and many others. In fact, the listening room will have a high impact on how we perceive the sound due to reflections. There is an extensive field of study of loudspeakers, room treatment, and other areas—more fun for the DIY hobbyist to explore! Just note for now that the frequency response of your speakers is incredibly important in achieving high-quality sound. It could require large and often expensive speakers to have a flat frequency response and one that extends down to the lowest ranges around 50 Hz and below.

A common measurement unit for frequency response is decibels (dB), a non-linear unit to measure the intensity of sound relative to a reference. If the reference level of 0 dB represents the threshold of human hearing, then a normal conversation might be 60dB in intensity, while a loud music concert could be 100-120dB. It is on a logarithmic scale because our hearing responds differently to low intensity sounds than it does to high intensity ones: 0dB is near silence, 10dB would be 10 times as powerful, 20dB would be 100 times as powerful, 30dB is 1000 times as powerful, and so on. A rule of thumb is that a 10dB increase is perceived as twice as loud.

In both charts, near the bottom and the top ends you see the frequency response start to fall off. On a response chart, the roll-off point that reaches -3dB point relative to a normal level is referred to as the cutoff point, where calculated power at that point is technically half of the normal reference point. Measuring a response that fluctuates within 1-3 dB might be considered flat with minimal perception of change, but more than that could start to become noticeable in listening. For example, a speaker that could reproduce a flat response across a range, but rolls off with a cutoff at 100 Hz means it is down 3dB at 100Hz relative to the maximum level. As the rolloff continues it might be down 10dB by 80Hz (or half the perceived loudness of the higher frequencies) and you might feel it sounds thin or lacking in bass because these frequencies are not loud enough relative to others.

Analog vs. Digital

I will take a moment to point out that this entire process of generating, transmitting and hearing sound is analog—continuously variable movement and interpretation of sound waves. And the process of recording and playback is also analog using a continuous signal voltage. Conversion of the signal to digital information and then later back to analog is commonly used now in recording and playback, and can be done with sophisticated hardware and software to preserve the original analog information as closely as possible, but the natural physics of sound is of course not digital.

I'm not a purist, and I use digital music streaming among other devices, both digital and analog. But there is a fascinating beauty and pleasure in learning and understanding the physics and electronics of analog music playback using tube amplifiers and passive loudspeakers (and perhaps other analog devices you use, such as magnetic tape or vinyl).

Ohm's Law: voltage, current and resistance

It's worth quickly highlighting the basics of a circuit: a closed loop involving **voltage (V)**, the difference in electric charge from one point to another, **current (I)**, the rate of change of electrical charge measured in Amperes (Amps), and **resistance (R)**, which opposes the current flow and is measured in Ohms. These three are the foundation to understand how a circuit is operating. Some type of power source creates a voltage that is higher in one place in the circuit relative to another. This voltage, sometimes referred to using a metaphor of "pressure", will force electrical charge to flow through the circuit as much as the resistance will allow, and this flow is measured as current. As current flows through the resistance, or the "load" of the circuit, the potential voltage drops until it is used up by the time the circuit is closed back to the power source.

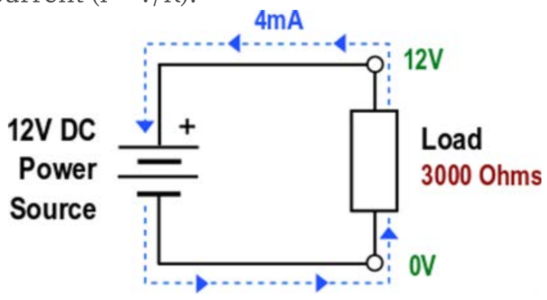
We can use one of the greatest magical formulas ever, **Ohm's Law**, which describes the relationship between these three (and we can solve for the third if we ever know the other two):

$$V = IR$$

$$I = V/R$$

$$R = V/I$$

In a **direct current (DC)** circuit, there is a source of positive charge and electricity flows in one direction. Perhaps the voltage source creates 12V DC, and the resistance in the circuit could be a simple resistor or could be some component that performs work when current is flowing, causing resistance or load on the circuit. In the example below, there is 12V of positive potential on one side of the battery, relative to the other side that closes the circuit. In between is a load that represents 3000 Ohms of resistance. Using Ohm's law, we can calculate that this load will draw 0.004 Amps, or 4mA of current (I = V/R).



If there were no load, or no resistance, (essentially a short-circuit) then Ohm's law would tell us there is 12V divided by 0 Ohms, or infinite current that would flow. This is obviously impossible in the real world, but we know that a short circuit will draw as much current as the power supply can provide, which could damage components, blow a fuse, etc.

Our houses use **alternating current (AC)** as the power source because of limitations that make it difficult to transmit DC over long distances. Alternating current is a voltage that changes rapidly back and forth from positive to negative at some frequency. In North American homes, we have 120V AC at 60 cycles per second (Hz).

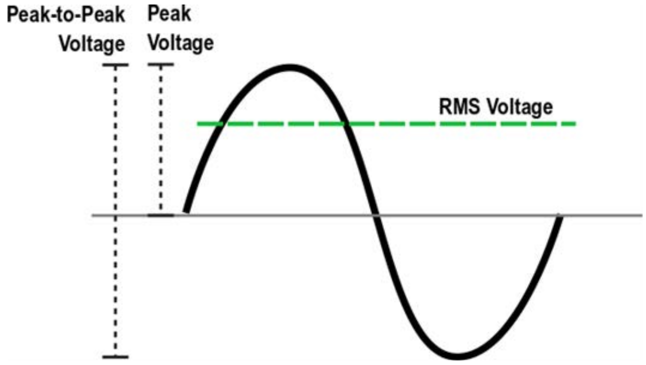
Direction of Current

You will notice I drew arrows in the direction from negative to positive to represent the current. Don't worry about direction. There is a history and convention of thinking about current flowing from positive to negative, and we often refer to a "voltage drop" that happens across the load that makes us think about the starting point being higher and moving to a lower potential. Electrons actually move from negative to positive. Don't let it tangle you up. It's two sides of the same coin. The key point is that there is a difference in charge between the two points in the circuit, and the amount of resistance between those points, as well as the amount of difference in charge (voltage) determines how much current (rate of change) we have.

A circuit can have a load that functions on AC, such as a light bulb. But in many cases, an electronic device will need to convert the alternating current into direct current for the circuit to operate. We will discuss this process later when we go over power supplies and rectification.

Sound waves discussed in the first core concept can also be represented as AC voltages changing over time positive and negative at various frequencies or complex AC waveforms.

AC introduces a question if we wish to perform some measurements or calculations. What is the actual voltage of an AC power source if it is continually changing? We need a few more ways to describe this voltage. **Peak voltage** is the highest positive voltage in the cycle, and **peak-to-peak voltage** is the difference from the highest to lowest point in the cycle (usually twice the peak voltage). **RMS voltage** (root mean square) is a way to measure this AC voltage and express it as an equivalent DC voltage that would produce the same power dissipation. (Some digital multimeters can measure RMS voltage while others use techniques that estimate it assuming an AC sine wave.) RMS voltage can be calculated by multiplying the peak voltage by 0.7071.



So in your home, if you measure the voltage from a wall outlet, it may be 120V RMS, but the actual peak-to-peak voltage is about 340V.

Power

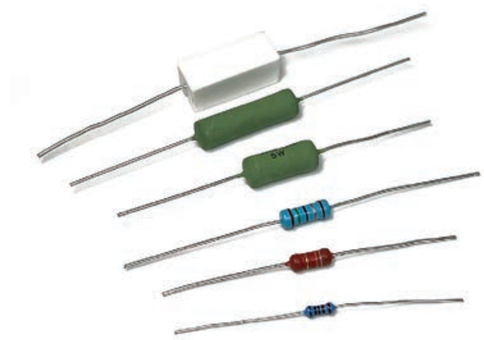
Finally, while we are on the subject of Ohm's law and the voltage and current through a resistive load: power (P) represents the rate that energy is produced or used in a circuit. It is measured in Watts and is the product of current times voltage (P = IV). Or, knowing Ohm's law, you could calculate power knowing other combinations, such as resistance and voltage. For example, a circuit with 500 Ohms resistance and 24 volts of potential would dissipate 1.15 Watts of power. Hopefully this power is doing some useful work! Sometimes power may be transferred to heat energy.

COMMON ELECTRICAL COMPONENTS

Resistors

You are likely familiar with a basic resistor: a device that intentionally holds back current and is rated in Ohms. I won't go over these in much detail other than to mention briefly a few types of resistors and to cover wattage ratings.

Resistors are made in different ways and you'll see them categorized—metal film resistors, carbon film resistors, carbon composition resistors, wirewound resistors, etc. There are simply different ways to construct a resistor for varying objectives and you end up with different attributes, sizes, costs, etc. In my amplifiers I use metal film resistors and wirewound resistors, which tend to have the lowest noise. Noise is an unwanted side effect of a resistor that impacts the signal passing through it. It can be thermal noise, or current noise caused by the structure of the resistive material when current runs through it. While we want to minimize noise, this is not in my opinion the largest problem we need to deal with compared to many other aspects of circuit design and selection of high-quality components such as tubes and transformers. Resistors are relatively inexpensive components and have a pretty easy job to do if selected and rated properly.

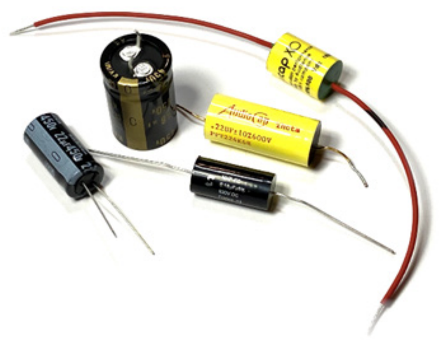


Note that wirewound resistors are generally available in relatively lower resistances due to their method of construction. I use them sometimes for power filtering or cathode bias resistors, which are often less than 1 kOhm. Wirewound resistors can be inductive, but in this application, it will not have a noticeable impact.

Resistors are manufactured with a power rating: 1/2 watt, 1 watt, 2 watts, 5 watts, etc. As voltage is forcing current through a resistor, electrical power is converted to heat energy. The resistor is designed to handle a maximum amount of power before it is destroyed by too much heat—if you've ever made a mistake in a circuit, you will know you can easily burn up a resistor! Consider a circuit that has 100V causing 8mA of current to flow through a resistor that is 12.5kOhms (Ohms law validates these relationships). The power dissipated will be 0.8 watts (P = IV). So you would need at least a 1W resistor to handle this power. But a good rule of thumb is to use a resistor rated for double the power your circuit needs. I would select a 2W resistor in this case. There is no problem using an overrated resistor, other than cost and size. I often use 1 or 2W resistors even when power required is much lower because I buy them in bulk and they are physically larger than tiny ¼ or ½ watt resistors and my fingers can work with them better!

Capacitors

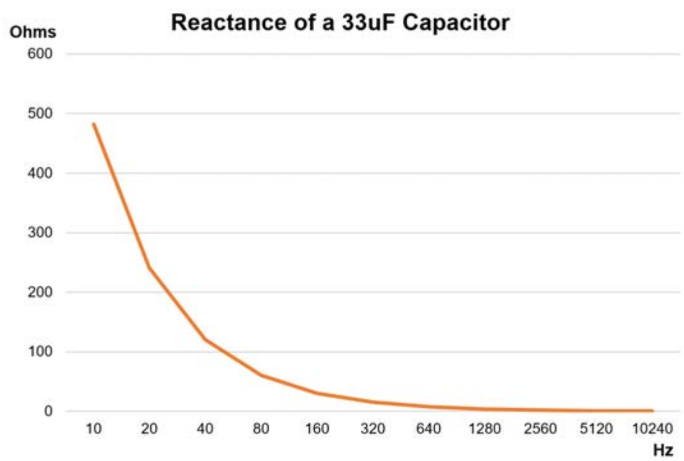
A capacitor stores energy in an electric field. It can be created in various ways, but commonly is done using two conductors (or plates) separated in some way, such as by a film or ceramic material or dielectric. There is no electrical connection between the two conductors, but because they are physically close, a voltage potential between them causes a positive charge to build up on one plate and a negative charge on the other. The physical characteristics of the capacitor determine how much energy can be stored in this way, and we measure capacitance using Farads, or more commonly microfarads (μF or uF), which are one millionth of a Farad.



Since there is no electrical connection, capacitors do not allow direct current (DC) to pass through them. But a change in voltage will cause the plates to charge or discharge, allowing alternating current (AC) to pass across the capacitor as it changes rapidly from positive to negative. This is an important principle: the current in a capacitor is directly related to the rate of change in voltage—a very slow-changing or steady voltage will have low or no current, while a very fast change in voltage will have higher current. We can quantify the action of a capacitor to oppose lower frequency voltage changes using Ohms, the same unit of measure we use for resistance. But in this case we would refer to it as **capacitive reactance** (X_C). I promise I won't use many formulas, but you might find it helpful to see how we can calculate the reactance of a capacitor, since it will vary based on frequency and capacitance:

$$X_C = \frac{1}{2\pi f C}$$

For example, a capacitor that is 33uF in capacitance that has an AC voltage at 120Hz across its conductors would have about 40 Ohms of reactance at that frequency. What would you guess is the reactance at a lower frequency, like 60Hz? Yes, lower frequencies would have lower current passing, so it's a higher reactance of 80 Ohms. How about a very high audio frequency, like 8kHz? Plug that into the formula and you'll see the capacitor would present only about 0.6 Ohms of reactance. Cool, huh? And very useful! Visualized here is a plot of a 33uF capacitor's reactance at various frequencies.



One more example, let's see how this reactance changes based on higher and lower capacitance. Imagine this was a bigger capacitor of 150uF instead of 33. Now, because the capacitor can hold a greater charge, those same frequencies we just looked at have different reactance: 120Hz would have reactance of only 8.8 Ohms and 8kHz would have reactance of 0.13 Ohms. So a higher capacitance will lower the reactance at a given frequency.

Capacitors are rated for a certain voltage, and many are not manufactured with high precision in capacitance, often +/- 20%. Higher voltage rating and capacitance will require a physically larger (and usually more expensive) capacitor. Always choose a capacitor with a higher voltage rating than you expect it to see in the circuit. Exceeding the voltage rating will lead to the failure of the capacitor.

Capacitors, like resistors, come in a variety of types. For high voltage and capacitance values, we typically need to use electrolytic capacitors. These have a few drawbacks in terms of lifespan and some characteristics, but are still our best choice. Most electrolytics are polarized, so you must wire the negative side to the lower voltage potential or it can be destroyed or even explode. Electrolytic capacitors have a liquid inside of them and are sensitive to high temperatures and "ripple current" that can create internal heating of the capacitor, shortening the life if not rated sufficiently for the application.

In some cases when we have smaller capacitance needs, we can use some type of film capacitors. These are better for certain audio purposes than electrolytic capacitors, but would be too large physically for other uses when capacitance needs to be higher. You'll see in this circuit that we use a film capacitor for coupling between the stages of amplification—a sensitive part of the circuit where we don't need much capacitance. Audiophiles love fancy coupling capacitors and you can find some for outrageous prices, probably made by elves using ingredients that cost many gold coins and precious gems. I believe in good caps, but only so far, like most things.

Inductors

Inductors (some called "chokes") also store energy, but instead of an electric field as in a capacitor, it is stored in a magnetic field. These are usually made of some type of insulated wire that is wound into a coil, sometimes around an iron core. When current flows through the coil, a magnetic field is created. We can measure "inductance" with a unit called Henries (commonly using the symbol L) and it is based on number of turns of the wire, length and cross-sectional area and core material.

Under DC conditions through an inductor, current flows and a magnetic field is created. At this point, the inductor acts as though it were a short-circuit, with the only resistance being the natural resistance that the coiled length of wire would have.

However, when there is AC, the current is trying to change rapidly from one moment to the next. But when energy is built up in the magnetic field from increasing current, the inductor will tend to maintain that same level of energy, which is related to the amount of current. Think of it a bit like momentum and inertia: something put in motion will tend to continue in that same motion, or something not in motion will tend to stay not in motion. So when the current rapidly changes, the inductor resists the change. The net effect of this is that the inductor forms a voltage between its two connectors in opposite polarity to the change in current.

I know, this requires some heavy thinking, physics, force fields, ESP, and Jedi training to really get the theory of it, and I have not explained it with much depth. But the main takeaway is that inductors resist changes in current, and the voltage across the inductor is proportional to the rate of change of current. This is exactly the opposite of a capacitor, where current is related to the rate of change of voltage. Inductors have "reactance" similar to capacitors, that varies based on frequency. Higher frequencies will have greater inductive reactance than lower frequencies. The inductive reactance (XL), measured in ohms, based on frequency (f) and inductance (L) is:

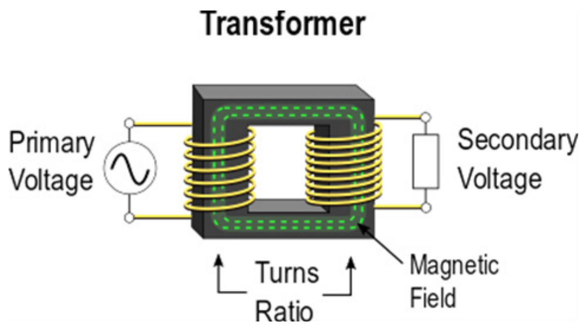
$$X_L = 2\pi f L$$

You may see inductors (chokes) and capacitors in amplifier circuits performing important roles of blocking or allowing DC or AC, or filtering a power source to remove ripples in the voltage or current. I'm not touching on many other aspects of these components, including how phase of AC is altered by these components, but hopefully this gives some of the basics to understand our circuit.

Transformers

A transformer also uses principles of magnetic induction. Two coils of insulated wire can be wound on a core. When a voltage is put across one of the coils (the “primary” side of the transformer), it will magnetize the core and induce a voltage onto the other “secondary” coil. The two coils can have a different number of turns, and the ratio of these turns will result in a higher or lower voltage on the secondary side. This is very useful when we want to “transform” a voltage from one level to another. We use this in several places in an amplifier: to convert household mains voltage from 120V AC (or 220-240V) to some higher voltage needed in the amplifier circuit or to transform it to a low voltage needed to power the tube filaments, and also after amplification to convert from a high voltage signal down to a low voltage usable for speaker outputs.

A transformer works with AC only. There is no electrical connection from primary to secondary, and DC would pass across the primary side as a short circuit. Remember how the inductor resists changes in current and will create a voltage to try and maintain its state of magnetic field? Under AC voltage conditions on the primary, a voltage will be created by the transformer on the secondary side as an inductive response, with current in opposite direction to the primary.



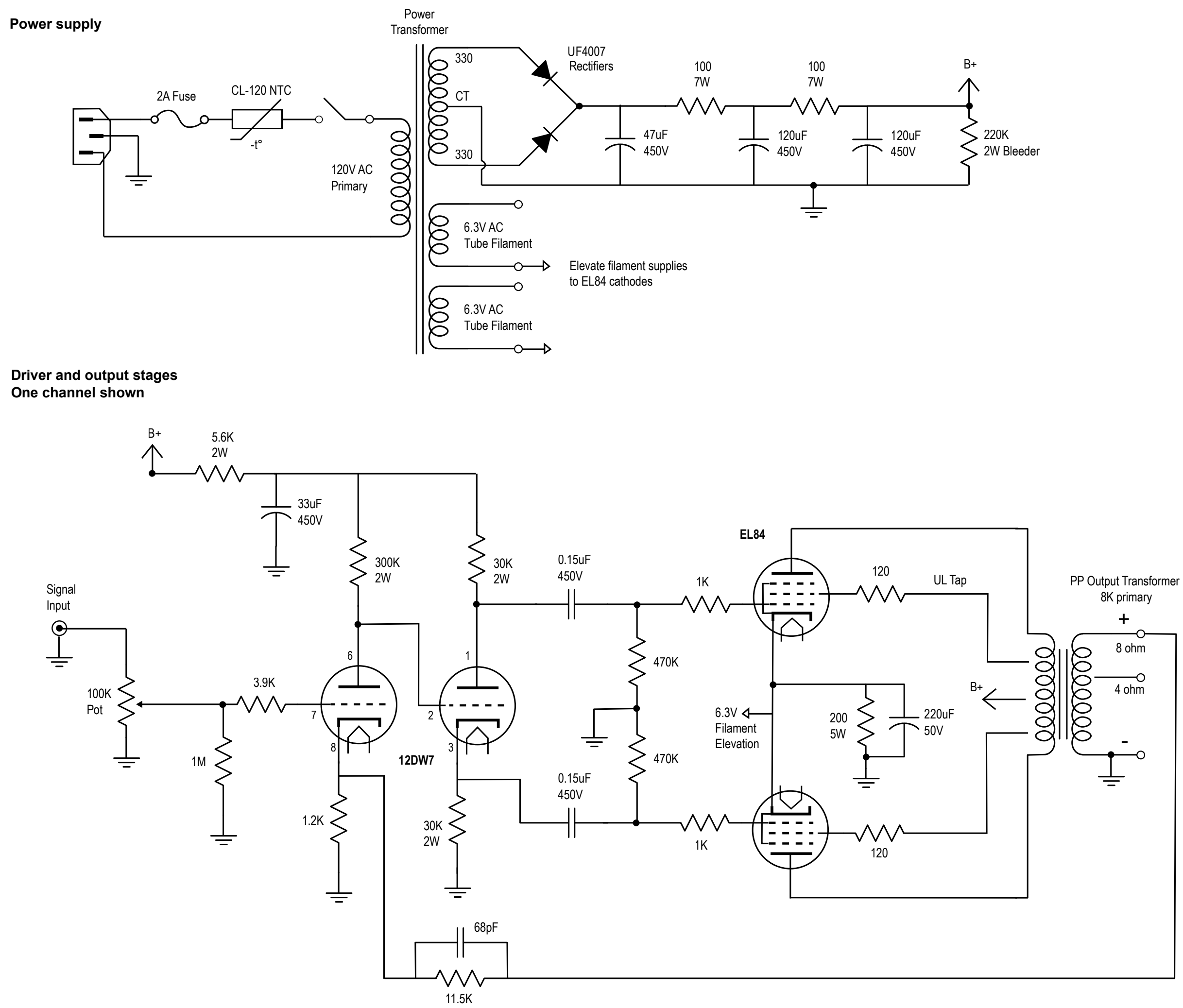
Keep in mind that this is a passive electrical activity (unlike an amplification process). We are putting the circuit load on the secondary side of the transformer where there is going to be a voltage, load resistance and resulting current draw. On the primary side, that load will appear differently due to the turns ratio so the current will also be different. If for example, a transformer has a 1:4 primary to secondary turns ratio, then voltage will transform from 120V on the primary to 480V on the secondary. A load on the secondary that results in a current draw of 100mA would have that current appear on the primary side using an inverse ratio as a draw of 400mA of current.

All that said, this is a simplified view and a transformer is not perfect, there are some losses in efficiency in several ways we won’t discuss. Some power is dissipated as heat, and a power transformer can therefore become hot under a full load, and is typically designed and rated to allow for this.

BIG PICTURE OF THE CIRCUIT

We will go into detail on each part of the circuit, but for now, let's get up on the balcony and take a look at the big picture. Refer to the schematic, and you'll see one section is the power supply and the other is the amplification circuit. The amplification stages are duplicated in the actual amplifier, one for left and one for right channel, but only one is shown for simplicity. There is, however, only one power supply for the circuit.

Pacific 63 Push-Pull Amplifier 12DW7 / EL84



A later section will go over the power supply in detail, but for now, just know that the purpose of this portion of the circuit is to take AC voltage from your wall outlet ("mains" voltage) and convert it to high voltage DC, referred to as "B+". This is a historical term from when batteries were used as power supplies and this was the positive voltage. You may also see this referred to as High Tension or HT, also an older terminology. In our case, in comes the household 120V AC (or 240V), and out goes the B+ of about 350V DC. You'll see later why we need this high of voltage.

The arrow on the right-hand side illustrates that the B+ is an output of the power supply circuit. Similar arrows are shown in two places in the amplification part of the circuit, indicating that the B+ voltage is used to supply power to two parts of the circuit. See those spots? Just a convenience in showing the power supply and amplification circuits separately instead of a single schematic.

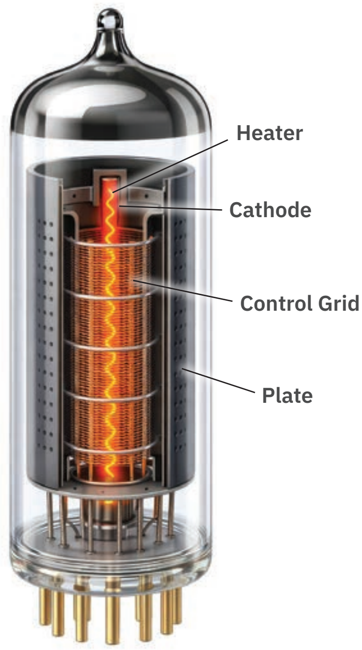
In the driver and output stage, you can see the schematic flows from left to right, with the input signal entering on the left, the 12DW7 tube is a dual-triode vacuum tube that performs a first level of amplification and splits the signal through a phase inverter step, then there are two EL84 tubes that perform a larger ("power" or "output") amplification. We will also go over the details of these tube types and how they work. Finally, there is a transformer to convert back from high voltage to something usable for the output of the amplifier that will go to your 4 or 8 ohm speakers.

As I go through more explanations, I'm going to use schematic excerpts and example tubes that are not exactly what are used here in this schematic and in your actual amplifier. That's for easy illustration purposes, but you hopefully will get the concepts and understand once we bring things back together.

We will go over all of this in detail. But first, let's look closer at the star of the show, the vacuum tube!

HOW DOES A VACUUM TUBE WORK?

At the heart of this circuit is a vacuum tube doing its little magic. I will cover the basics of how vacuum tube amplification works so you can understand what's happening. I won't try to convey all of the physical science involved, but you are a lifelong learner I know, and can read more about it from others who can go into more depth and robust explanation.



Vacuum tubes used in amplifiers are also called "thermionic valves" referring to the way that temperature causes the release and flow of electrons. The basic type we will start with is a triode tube. Inside the glass enclosure of a triode tube are three main components: the cathode, the anode (also called the "plate"), and the control grid. There is often also a filament to act as a heater.

The cathode is typically coated with a certain type of metal, and it is heated up to a high temperature. In some types of tubes, the cathode is directly heated by running current through it, but in most modern triodes (including the tubes in this kit) the cathode is heated using a filament physically close to it, but not connected electrically. The filament is what you see glowing inside of the tube, sort of like a filament inside a light bulb.

As the cathode reaches a high temperature, it begins to emit electrons. They build up in a cloud and, without any other action, eventually there are so many that the space around the cathode reaches a point where no more will be emitted. Why not? Because electrons have a negative charge and they hate being close to other negative charges, and all those other electrons bumping elbows are making the place pretty dang negative. What do electrons love? Positive stuff. They are attracted like crazy to it. So what would happen if we introduced something positive into the mix here? Yeah, those guys would go for it.

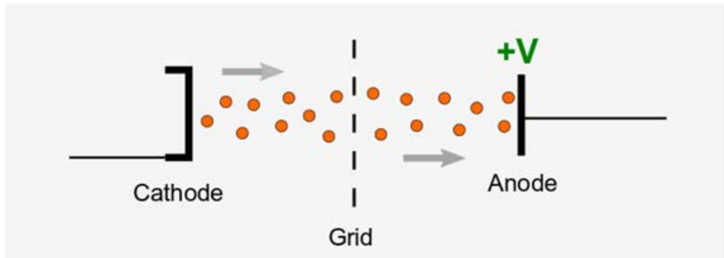
The anode is referred to as a plate because it's a metal plate surrounding the cathode, and we can put on the anode a juicy and delicious positive voltage with respect to the cathode. You will soon start to see why we need high voltage in a tube amplifier. If you just put a few volts on the anode, the electrons say, yeah man, cool, but I don't even get out of bed for that kind of voltage. To be sufficiently attractive, it has to be high.

The glass tube enclosure is sealed and there's a vacuum inside, remember? So those electrons are free to fly around without colliding into air molecules. So when there's a nearby high voltage potential on the anode, they are attracted and fly to it at ridiculous velocity, like around a million meters per second or something. Wow, right?

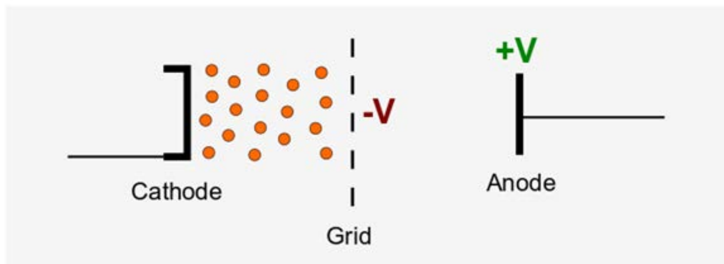
Now let's take a step back and think what's happening here. We heat up the cathode and it emits bazillions of electrons that flow at a million meters per second to the positive potential of the anode, continuously. Sounds a bit like…current? Yes. When the cathode is heated, electrons and current flows. (Don't get tangled up in directivity…the electrons go from cathode to anode, but we sometimes refer to current flow from positive to negative. It's just the way we measure current as a rate of change of electrical charge.) You'll note that it flows only one direction (a "diode" at this point). We heat the cathode and electrons can go to the anode. There is no way for electrons to go the other direction.

Alright, so an operating tube is allowing current flow (the "valve is open"). Now, there's one more component to make this a triode: the control grid. This is a wire mesh in between the cathode and anode that is spaced wide enough to allow the electrons to pass through. But what might happen if you applied a negative charge to it? You son of a turtle, say those electrons! We are not going through that negative fence you set up, we don't care what's on the other side. So now we have a way to control those electrons and if we raise or lower the voltage on the grid we can influence how strongly the electrons are repelled or allowed to pass. A very negative charge? No electrons pass and no current is flowing—we refer to this as "cutoff" of the tube. A less negative charge? Electrons and current flows (maximum flow sometimes referred to as "saturation" reflecting the temperature and physical constraint where the anode is pulling in all of the electrons that the cathode can produce).

1. When the grid is not negative with respect to the cathode, electrons flow to the positive voltage of the anode.



2. When the grid is negative with respect to the cathode, electrons are repelled and stay in the space around the cathode.

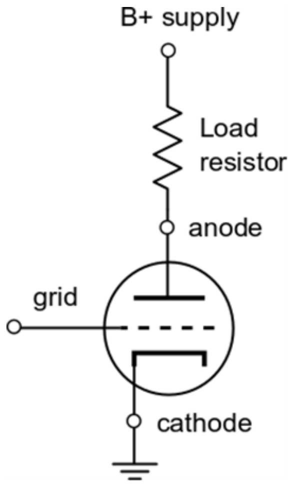


By altering the voltage of the grid, we can "open and close the valve," allowing current to flow more or less. Awesome! This is why a tube amplifier is sometimes referred to as a "valve amplifier." If we were to put an audio signal on that control grid, then the changing voltage over time of the audio signal will allow current to flow in alignment to the audio signal.

We are getting close! But how does this amplify the input signal? The answer is that the voltage change on the grid has a large influence over the current flow. How much current flow? To really understand, we need to get into a circuit a bit more and discuss load lines. You will love it. Hang with me.

First, we need one more quick addition to put the tube in context of a circuit. We already said that we would put a high voltage potential on the anode to attract the electrons and allow current to flow. We also will want a load on the anode, in this case a simple resistor, so that there is sufficient resistance in our circuit so we don't have an unreasonable amount of current flowing and also so that we can make use of the voltage change across that resistor. Ohm's law will tell us there is a known relationship between resistance, voltage, and current, right? And you love Ohm's law, right? Me, too.

Here's a very simplified circuit using a tube symbol. We won't worry about how we generated the B+ voltage or how we are setting the grid voltage, but assume the B+ is some high voltage potential, the cathode is at 0V, and the grid could be a few volts negative. We have enough for a working illustration of an operating circuit. (We usually do not show the heater filament in a tube schematic, so just know the heater is operating.)



For the sake of a quick calculation to get started, let's pretend that we have a 30k Ohm load resistor and 400V supply, and we pick a grid voltage that is allowing some amount of current to flow, let's say it is -2V. How much current is there if we measure the voltage on the anode and it is 250V? Well, 150V must have dropped across the load resistor to get from 400 to the 250, and using Ohm's Law:

I = V/R

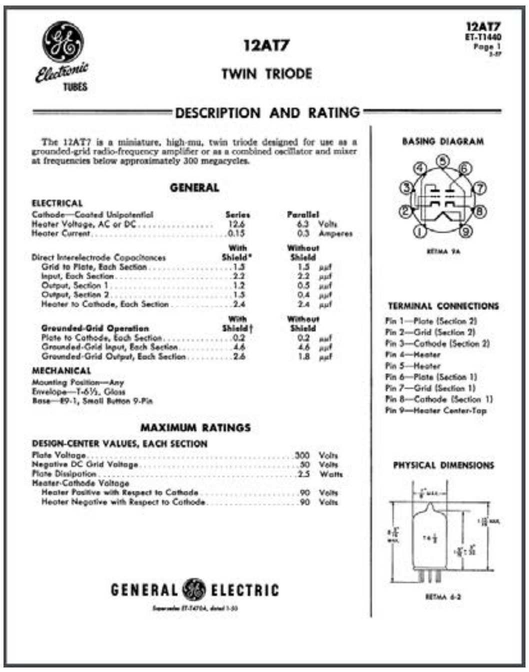
I = 150 / 30,000

I = 0.005 Amps or 5mA

So in this made-up example, -2V volts on the grid is allowing 5mA of current to flow. All we are doing here is using an easy technique to calculate current using a known resistance and measured voltages, using Ohm's law. But I want you to have the load resistor and Ohm's law in mind as we get a step further into the operation of the tube.

TUBE CHARACTERISTICS, LOAD LINES, AND OPERATING POINT

The physical characteristics of each type of tube—materials, how far apart is the cathode, grid, anode, etc.—determine how the tube will operate and the effect of different voltages. Tube manufacturers provide datasheets that include a variety of information about these characteristics and limiting values. Search online and you'll easily find these as PDFs—often looking like bad photocopies of documents from the 1950s—and of course there's a long history of tubes from that era or earlier, so it's no surprise.



Electronic TUBES

12AT7
ET-T1440
Page 1

12AT7

TWIN TRIODE

DESCRIPTION AND RATING

The 12AT7 is a miniature, high-mu, twin triode designed for use as a grounded-grid radio-frequency amplifier or as a combined oscillator and mixer at frequencies below approximately 300 megacycles.

GENERAL

ELECTRICAL

	Series	Parallel	
Cathode—Coated Unipotential			
Heater Voltage, AC or DC	12.6	6.3	Volts
Heater Current	0.15	0.3	Amperes
Direct Interelectrode Capacitances	With Shield*	Without Shield	
Grid to Plate, Each Section	1.5	1.5	μμf
Input, Each Section	2.2	2.2	μμf
Output, Section 1	1.2	0.5	μμf
Output, Section 2	1.5	0.4	μμf
Heater to Cathode, Each Section	2.4	2.4	μμf
Grounded-Grid Operation	With Shield†	Without Shield	
Plate to Cathode, Each Section	0.2	0.2	μμf
Grounded-Grid Input, Each Section	4.6	4.6	μμf
Grounded-Grid Output, Each Section	2.6	1.8	μμf

MECHANICAL

Mounting Position—Any
Envelope—T-6½, Glass
Base—E9-1, Small Button 9-Pin

MAXIMUM RATINGS

DESIGN-CENTER VALUES, EACH SECTION

Plate Voltage	300	Volts
Negative DC Grid Voltage	50	Volts
Plate Dissipation	2.5	Watts
Heater-Cathode Voltage		
Heater Positive with Respect to Cathode	90	Volts
Heater Negative with Respect to Cathode	90	Volts

BASING DIAGRAM

RETMA 9A

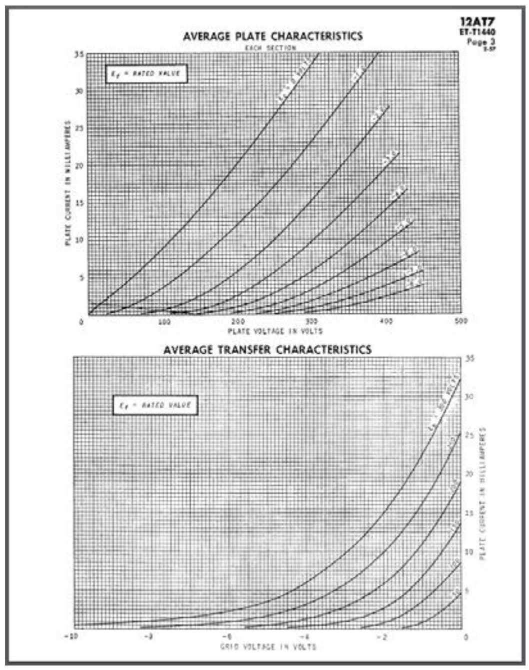
TERMINAL CONNECTIONS

Pin 1—Plate (Section 2)
Pin 2—Grid (Section 2)
Pin 3—Cathode (Section 2)
Pin 4—Heater
Pin 5—Heater
Pin 6—Plate (Section 1)
Pin 7—Grid (Section 1)
Pin 8—Cathode (Section 1)
Pin 9—Heater Center-Tap

PHYSICAL DIMENSIONS

RETMA 6-2

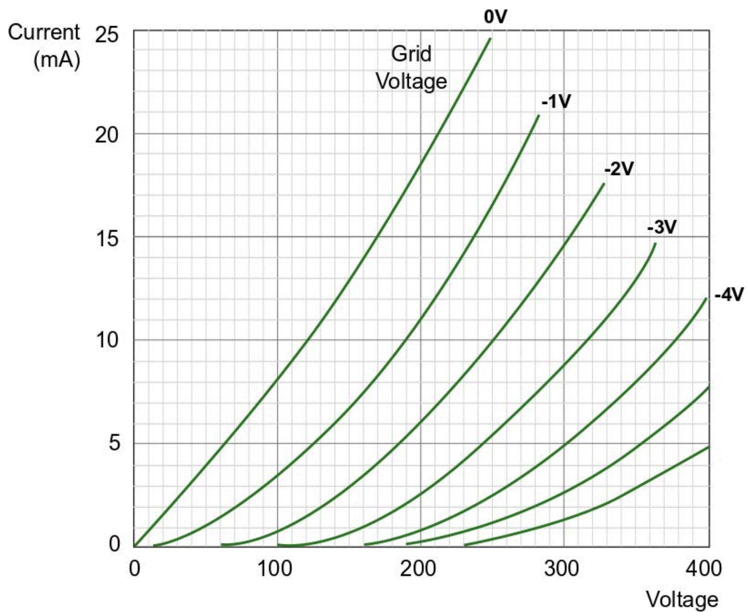
GENERAL ELECTRIC



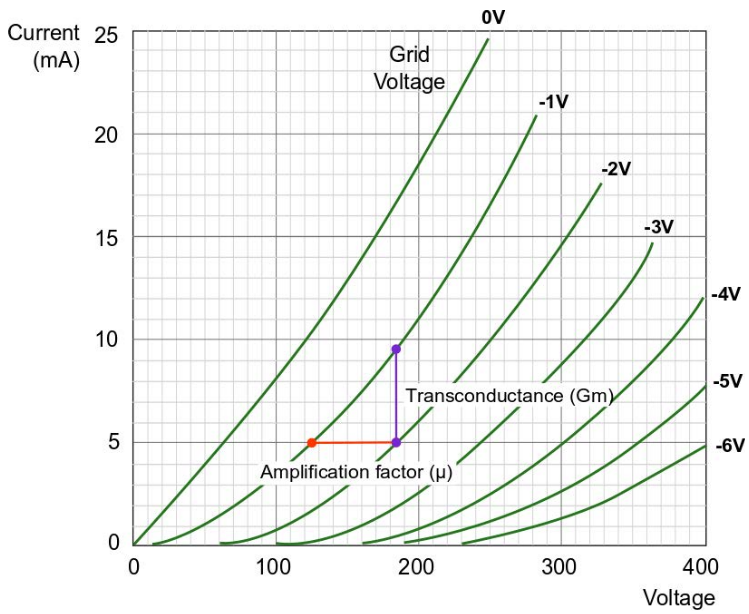
If you haven't studied these, they might seem confusing, but you are a brilliant learner and this is how you will be empowered to begin understanding or even designing your own circuits. Let's look at an example datasheet. I'm going to use a 12AT7, just for illustration reasons. Our amp uses the 12DW7, and we'll talk about this tube soon.

Go search for one of these online (e.g. "12AT7 datasheet") and pull one up for reference. We won't go into all of the details, but you'll find some useful information and I always keep these handy especially for the pin diagrams to tell you which pin number is the anode, grid, etc., or to know how much current the heaters will draw, or what maximum voltages can be used.

One of the charts you will find is the plate characteristics. On the x-axis is voltage and on the y-axis is current. There are a series of curves that represent different voltages that you could possibly have on the grid, and this will tell you what current would correspond to a particular anode voltage. Using the chart below, if you had the grid at -2V and you put 200V on the anode, then the tube (valve) will allow about 6mA of current. You will notice these are curves, not straight lines, because the tube does not operate as a perfectly linear device. We will talk about this later, regarding distortion.



Before we look at an actual circuit and load line, let's consider a few attributes of the valve that we can visualize on this chart. First, remembering that grid voltage controls the amount of current, let's see what this ratio is. If we pick a place on the chart and hold anode voltage constant and measure the distance between two grid curves, we can see that a 1V change in grid voltage results in around 4-5mA of current change. This is the purple line on the chart below, and is referred to as transconductance (Gm), often measured in a funny unit called mhos (reverse spelling of ohm, conductance being the opposite of resistance!) and tube datasheets usually use micromhos (one millionth of a mho). Look on the 12AT7 datasheet and you see transconductance of 4,000 – 5,500 (depending on operating conditions), which would match our chart estimate after converting units.



Now, consider another property called amplification factor (a ratio abbreviated with the Mu symbol: μ). Holding current constant, if we measure between two grid curves, we see that a 1V change in grid voltage will result in a change in anode voltage of 60V (red line on the chart). Aha! Here is the leverage that we have been looking to understand. We could swing 60 volts of anode voltage for every one volt on the grid. Powerful, yah? So the μ of the 12AT7 is 60 and you'll see this on the datasheet. Some tubes have lower or higher amplification factors that could range from around 20 to 100. After we plot a load line and calculate the gain of the amplifier, you'll see that we won't expect to get this full factor of amplification in our application.[1]

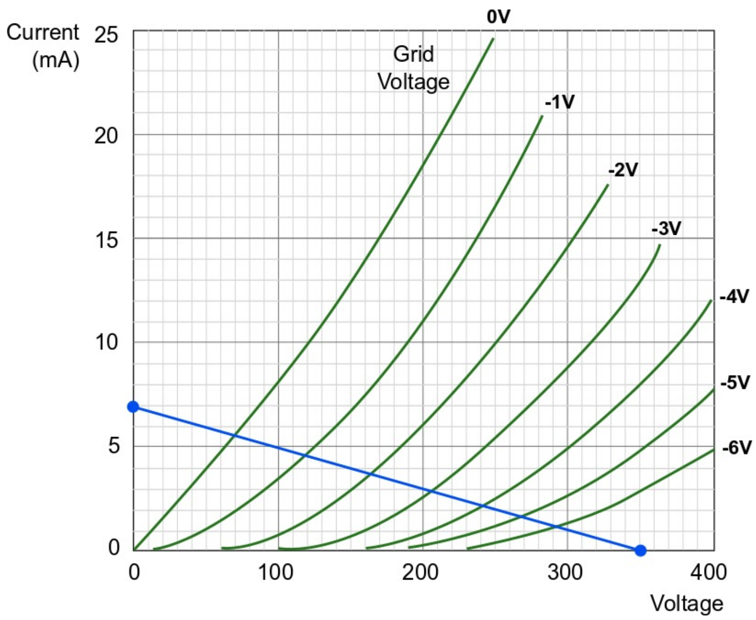
Now let's take our basic circuit and consider what happens when we have a particular anode load. Let's say we had a load resistor that is 50k Ohms and we supplied 350V B+. If there was zero current flowing (tube totally in cutoff), what voltage is on the anode? Ohms law tells us that with no current, there would be no voltage drop across the load resistor, so the anode will have the full 350V. Let's plot a point at zero mA and 350V representing this extreme situation. And imagine the opposite end of the spectrum—what if the valve was wide open so that maximum current flows and the entire 350V were to drop across that resistor, and there was zero volts on the anode. Ohms law in this case to solve for current:

I = V/R

I = 350 / 50,000

I = 0.007A or 7mA

So let's plot another point at 7mA current and 0 volts on the anode. Ohms law is a linear relationship, so this is all we need to draw a load line between these two points.



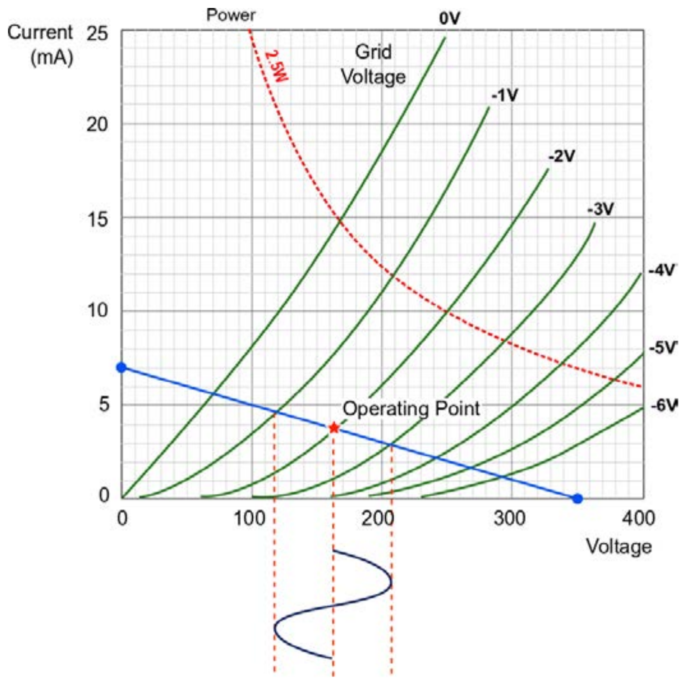
So what we see here is the linear relationship between voltage and current for a 50kOhm load resistance and 350V B+ supply. The tube must operate on this line somewhere. Where? That depends on what voltage we put on the grid. Look again at the grid curves and find the points that intersect with our load line. If we set the grid to -2V, then in this circuit we will have about 4mA of current and 160V on the anode. If we change the grid to -1V (less negative), then we will have 4.5mA of current and 120V. Notice that this is a change of 1V on the grid and a change of 40V on the anode—not the full 60 amplification factor because our load line isn't horizontal, it has a slope. If you were to alter the supply voltage or the load resistance, you can control the position and slope of this line, which impacts gain (and distortion).

Now it's time to visualize your audio signal on this load line. The input signal is an AC voltage that goes peak to peak from, let's just say, +1V to -1V (depending on your source level). If we simply put this signal onto the grid, it would cause the grid to fluctuate both positive and negative. We don't have any

1 Techniques such as a regulated current source can be used to maximize amplification factor but will not be covered.

positive grid curves shown here. Why not? Because when the grid is positive it is attracting electrons just like the anode. This isn't what we intend and will leak current out of the tube through the grid, which would then further impact the voltage potential between grid and cathode, etc. It's a problem we will avoid by keeping our grid operating in a better space, more central on our load line.

Let's pick a spot to represent the operating point, or the quiescent state when there is no audio signal impacting the grid, but there is enough room above and below this point for the audio signal to raise/lower the grid voltage. If we pick an operating point where the grid is -2V, this puts us roughly in the middle of the load line and there will be a steady-state current of around 4mA, and 160V on the anode.



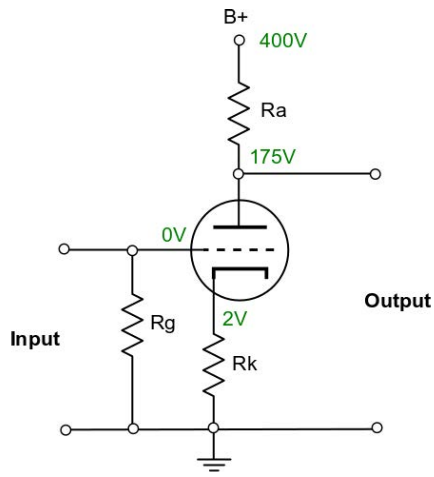
Now envision an AC audio signal on the grid that causes it to have a range between 1V and 3V and the anode voltage will fluctuate from around 120V to 210V. You can see a sine wave visualized on the horizontal axis. You should also recognize that we have current flowing continuously, it just varies in how much, based on the input signal and grid voltage. This load line is demonstrating a Class A type of amplifier, conducting current across the entire input signal, not going into cutoff at any point. You can see that Class A consumes power continuously regardless of input signal amplitude and is therefore less efficient than other options, but it has excellent characteristics for high fidelity amplification and simplicity in our design. A single-ended tube amplifier uses one tube to produce an output. We will talk later about a push-pull design, using two tubes to amplify portions of the signal.

One more consideration regarding load line and operating point. The tube will have maximum rated limits of operation. In the example of the 12AT7, the datasheet tells us that the anode cannot be higher than 300V in steady operation[2]. So the operating point must be left of that point. And the tube is rated for maximum power dissipation of 2.5 watts. The red dotted line on the chart shows where current and voltage would exceed 2.5W of power dissipation on the anode, so the load line and operating point needs to be below that line. (On this small-signal tube, we are not even close to maximum power, but on the output stage you will see we get much closer to it.)

2 This is technically the maximum anode-to-cathode voltage (or "plate-to-cathode")

Now let's figure out how to get our grid to be at our desired voltage for this operating point. One option is to use a negative DC power supply and adjust it so that it is 2V lower than the 0V potential of the cathode. Some amplifiers use this technique and it's referred to as fixed bias. It requires calibration to ensure the right relative voltage between grid and cathode to achieve a target operating current.

In our case we will choose another method. The grid needs to be negative with respect to the cathode in order to hold back the electrons, but we can achieve this in other ways rather than a negative DC voltage on the grid. We could keep the grid at 0V potential and elevate the cathode to a higher voltage. By inserting a resistor between cathode and 0V ground potential, we can put the cathode at a positive 2V potential so that the grid is now 2V relative to the cathode. This is a method called cathode bias. It's sometimes referred to as automatic bias or self-bias because if the steady-state current were to change for some reason (such as the tube aging and changing in properties), then by Ohm's law, the voltage drop across that resistor would change, so our cathode level would "automatically" adjust itself.



Ok, consider the new circuit shown. We inserted a cathode resistor Rk. What value should we use? If we want to elevate the cathode to +2V so that we have quiescent current flowing of 4.5mA, then Ohm's law will tell us that we need a value of:

R = V/I

R = 2 / 0.0045

R = 444 Ohms

Now we have technically increased our total load on the B+ voltage to the sum of the anode resistor and cathode resistor, so our original load line isn't quite right anymore, but this value is very small relative to the 50k Ohm so we won't worry about it, or you could re-calculate the load line with this larger value of 50,444 Ohms. (And of course 444 is an unusual number and we could round this cathode resistor value to one that is typically available.)

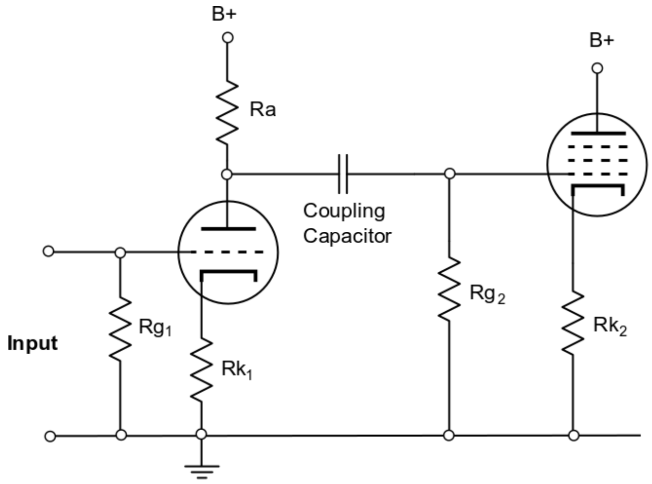
You will also notice a resistor Rg between grid and ground. This is called a grid leak resistor. We need something to reference the grid to the 0V DC potential, allowing the input AC signal to then be applied onto it to control the tube. We want this to be a high resistance value so that we don't attenuate the input signal (keeping the input impedance high relative to the source impedance), but there are some other considerations. There is a very small amount of current that flows in the grid ("leaking") and it could alter the bias of the grid. For a small signal tube like the 12AT7, we might use 1M ohms.

Congratulate yourself for understanding this far. We aren't done yet, but I hope you feel good about what you've learned so far. You might need to re-read a few times to get it, and consider additional resources I reference in the appendix if you want to learn at a deeper level. Pause here, get a drink, scratch dog behind ears, etc.

COUPLING, AC LOAD LINE, AND CATHODE BYPASS CAPACITOR

You have the basics. Now it's time to broaden our circuit a bit more and deal with a few more issues. The triode tube we have seen as an example so far is used as a small-amplification stage in an amplifier with still a relatively low level of current and output voltage. We need to put it through a power amplification stage to really get it to the level of current needed to drive speakers and move some air on the journey of sound back to our ears!

We will take the amplified voltage from the anode, which you already know in this example operates around 160V DC, fluctuating up and down with the audio signal. We can consider this an AC signal riding on top of the DC voltage. We will want to feed this signal into another tube (an output or "power" tube) for further amplification, but we can't put 160V onto the grid. This high voltage will cause all sorts of problems on the grid which is intended to be negative, likely destroying it. We want only the AC portion.



The coupling capacitor is a solution. As you know, capacitors block DC and allow AC. The capacitance value can be small. We simply want to block the DC and allow all audio frequencies to pass. Typical values might be somewhere around 0.1uF to 0.22uF, for example.

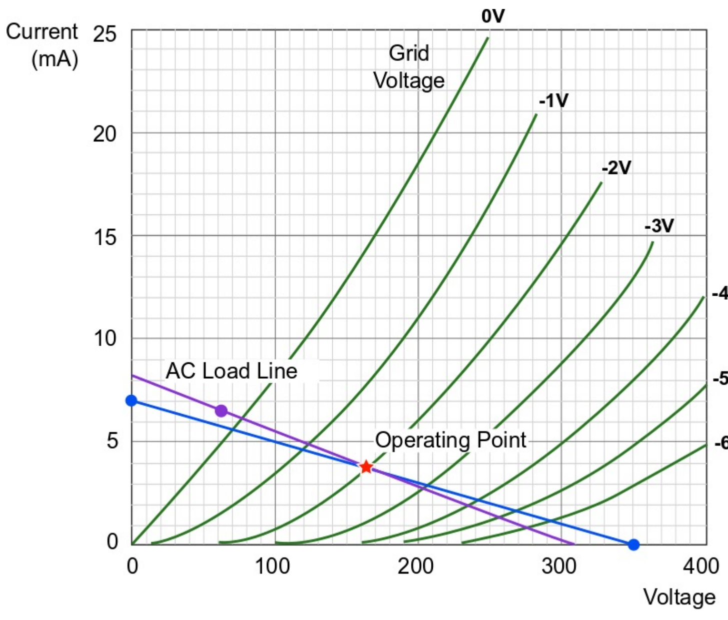
The coupling capacitor is directly in the signal path and so it is a very important component, and we want to use something high quality that will pass the signal without noise or distortion. Some type of film capacitor is best, and there are very high-end audiophile grade coupling capacitors produced, some at outrageous prices. Note that the voltage rating of this capacitor needs to be high enough to handle the entire B+ because the tube takes time to warm up before any current flows, so the full voltage will appear on the anode, at least for a few seconds at startup.

We will get into the output stage soon, but we need to go back now and consider a few things with the driver stage. Our load line was fine for defining an operating point when there is DC voltage without audio (AC) on the grid, but since we are allowing AC voltage to pass on to the next stage through the coupling capacitor, we need another load line to understand the load under operating conditions: an AC load line.

The AC load will be the combination of two things: the anode load resistance we have already been dealing with, but also the impedance of the next stage, in this case a path to ground through the next stage's grid leak resistor. Do you see how Rg2 is in parallel with our driver tube (with regard to AC voltage, not DC)? Let's pretend that Rg2 is 470k Ohms. The formula to solve for parallel resistance is:

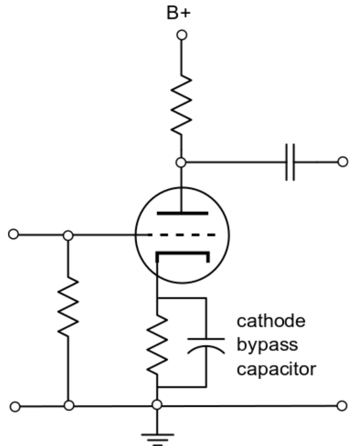
$$R = \frac{R_a \times R_{g2}}{R_a + R_{g2}}$$

Solving this with 50,000 and 470,000 as our Ra and Rg2 values tells us that the AC resistance would be about 45.2k Ohms. Now, let's find our new AC load line. The operating point will stay the same, and we can choose to plot some other point to get our line. Let's pretend under AC voltage the anode drops by 100 volts. Ohms law will tell us that if our AC resistance is 45.2k Ohms, then the current difference for this voltage drop is 2.2mA. So we can plot a point that is 100V lower than our operating point, and 2.2mA higher in current. Now this is what our true working load line will be.



Our circuit is coming together! We will move on to the power stage soon, but first, one more consideration to address. Remember how we wanted our cathode to be +2V relative to the grid. But think for a moment what is happening as the grid rises and falls with an audio signal—use a sine wave as an example and think about the voltage rising and falling over time. When the input signal is +1V, then the grid becomes less negative with respect to the cathode, and more current is allowed to flow. When more current flows, what will happen across our cathode resistor? Dang it. Voltage on the cathode will rise, right? Ohm's law again. On our AC load line, if the grid rises to -1V, then current is somewhere around 5mA. Remember how we solved to find our cathode resistor value and came to 444 ohms so our cathode would be at 2V. Well, 5mA across 444 ohms in Ohm's law is about 2.2V, instead of our intended 2V. The opposite is true when the signal goes the other direction: move down the AC load line, less current flows, which reduces the voltage across the cathode resistor. So the cathode voltage is fluctuating along with the input signal to some degree. Not exactly what we intended, and the net effect is that it counterbalances the amplification of the tube, reducing gain.

In some amplification stages, this can be used to an intended advantage for reducing distortion (referred to as cathode degeneration), and in others we might want to prevent this loss of gain. You'll see in our circuit we will allow it in our first stage, but will use a solution in the output stage: a cathode bypass capacitor. It's best to illustrate now, so you understand it. By putting a capacitor alongside the cathode resistor, we can hold the cathode at a desired DC voltage, but allow AC to pass through the capacitor. As the grid changes in voltage at audio frequencies, and current rises or falls accordingly, the AC currents will no longer impact the cathode voltage, it will remain at the intended DC voltage while the AC current can bypass it.



This is usually an electrolytic capacitor because capacitance needs to be relatively high, say 100uF or more, to allow current from low audio frequencies to pass. The voltage rating needed is relatively low given the expected voltage on the cathode. I often buy 25V or 100V caps for this purpose.

OUTPUT STAGE AND PENTODE TUBES

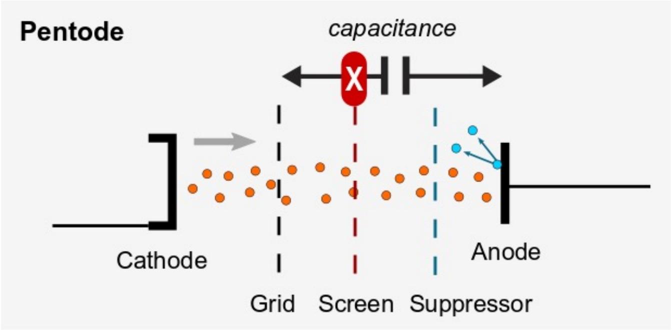
I'm going to take you through a single-ended output stage first so that we can get some concepts across, and then I'll go into a push-pull output stage that is actually used in this circuit, since we need to build on those concepts.

The tube used in our driver stage is a triode, which has great qualities, but also some limitations. You can read further material I reference in the appendix to understand at a greater level, but I will summarize.

A triode has a small amount of undesirable internal capacitance that exists between the electrodes, notably between the anode and grid referred to as Miller capacitance. Historically, this had caused problems in particular when attempting to use triodes for radio frequencies. To deal with this, new types of tubes were invented. First the tetrode introduced a screen in between the anode and control grid to shield the grid from the anode, reducing this capacitance. The screen could be held at a constant voltage while most electrons would still pass through to the anode.

However (there's always a however!) to make it work properly, the screen must be held at a relatively high DC voltage, and this introduces a new complication in our expected current flow in the tube. At certain voltages, electrons will hit the anode with high velocity, dislodging extra electrons which bounce back and are absorbed by the high voltage screen instead of the anode. This is a form of emission from the anode, impacting its voltage at certain amplitudes of the signal, causing a bend in the grid curves, and distortion in the amplification process.

To deal with this side-effect, the pentode was invented[3], which adds yet another grid called the suppressor in between anode and screen, held at a low voltage (often tied to cathode) and this keeps those bounced electrons from being attracted to the screen, and they are then recollected by the anode. The electrons emitted by the cathode are moving at such a high velocity that the suppressor has minimal effect on them, only the deflected electrons.

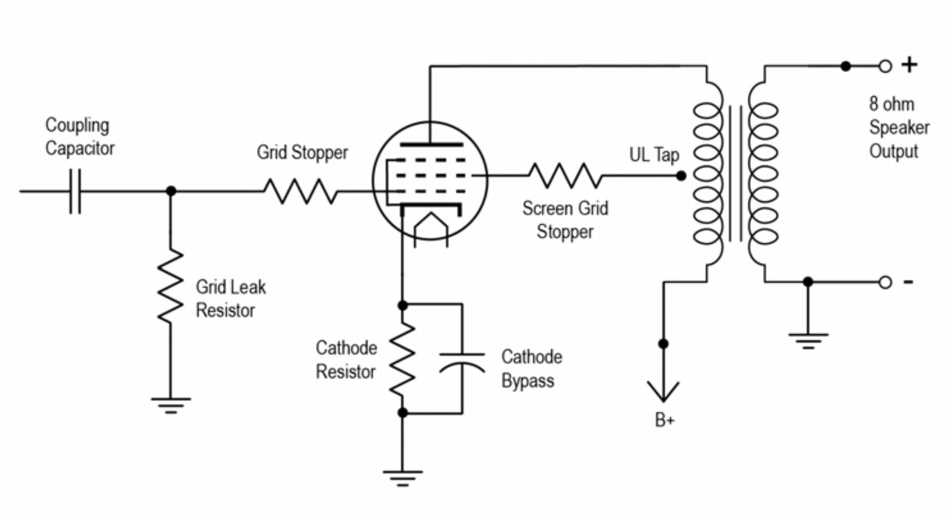


The general principle of the power stage pentode tube is similar to the driver triode tube: put high voltage on the anode and screen, put the grid at a voltage that is negative relative to the cathode and modulate it with the voltage signal from the driver stage. But the output tube will be working with higher current and voltage swings.

The pentode screen grid has another effect on the tube, which is that gain is increased, and this is an attractive feature for a power amplification stage. Because the screen is held at a constant voltage, it continuously will attempt to pull a steady flow of electrons from the cathode toward the anode (as much as the control grid will permit), so the tube can conduct current more efficiently. This is different from a triode where the voltage of the anode rising and falling with the signal is the only form of electron draw—for example, as the triode grid voltage moves more positive, current rises causing a drop in anode voltage across the load, but the lower anode voltage is less attractive to electrons than otherwise could be possible. This is a sort of headwind to realizing maximum triode gain that is solved with the pentode.

In our circuit, we are using one type of pentode tube, the EL84. Notice on the simple schematic shown here the bottom dotted line is the grid, the next one is the screen, and the top one is the suppressor. Note that a pentode tube could be made to operate in triode mode by simply connecting the screen to the anode so they are always at the same voltage. Some amplifier designs do this and we will discuss soon some choices of how we will actually use the screen in ultralinear mode versus pentode or triode mode.

On the schematic, you can see again some familiar components: we have another grid-leak resistor, we have a cathode resistor with bypass capacitor, there are a few "stopper" resistors I will explain later, and we have an anode load—oh, wait…where's the load resistor? The B+ in the output stage is going into the output transformer primary, and out the other side of the primary, to the anode. Aha! We don't have a



3 The beam-tetrode was also invented around the same time, solving similar issues in different ways.

resistive load in this case, we have the speaker as the load, and the fluctuating voltage across the transformer is going to drive—or do the work—of moving the speaker coil. But speakers are not designed for high voltage (and it wouldn't be safe to have this exposed outside of the amplifier anyway), so the output transformer will convert our amplified plate voltage to a different voltage and current level suitable to drive the speakers.

Ok, you are saying, but wasn't the whole point to amplify the signal? Are we reversing our process? No. Remember that the transformer will transfer the same power from the primary side to the secondary side, but the number of windings in the transformer will determine how current and voltage are changed. In our case, we are lowering voltage down to levels below 10V, but for power to be the same, current must increase. So we will have a large AC current driving our speakers. This is a transformation to the final step of our audio signal and we hope our trusty output transformer does this as a faithful and accurate reproduction. The quality and size of the output transformer is very important in maintaining a flat frequency response across a wide bandwidth at our desired output power. The kit includes transformers made in the USA that I believe are excellent quality relative to the cost and size of this particular amplifier kit and also made to be a replacement for the original Dynaco ST-35 transformers known as the Z-565. There are smaller and larger sizes of transformer ("iron" as you might say to your DIY tube amp buddies), and larger ones will be heavier and more costly, but also will ensure a wide frequency bandwidth with lower distortion at higher rated power output. The takeaway is that not all are created equal, so this is one place in your amp where you want to choose carefully a trusted and high quality manufacturer. I'll go into a bit more detail on output transformers when we discuss push-pull circuits. Remember that the illustration here is still a single-ended circuit.

So if the speaker is the load on the power tube, how big of a load is it and can we calculate the load line again? This amplifier is designed to work with either a 4 or 8 ohm speaker impedance, but the windings ratio of the transformer will make this impedance look much larger to the tube which is what we need for the tube to operate properly. We select this ratio when choosing the output transformer. For an example, let's consider a transformer that will be around 5,000 ohms on the primary side of the transformer for an 8 ohm secondary impedance. You could choose a transformer with different values, depending on the load you want on the tube and the impedance of your speakers. An 8 ohm nominal impedance is the most commonly used speaker impedance. And 5k ohms gives us a load that would work effectively for characteristics of the EL84 tube in single-ended circuits.

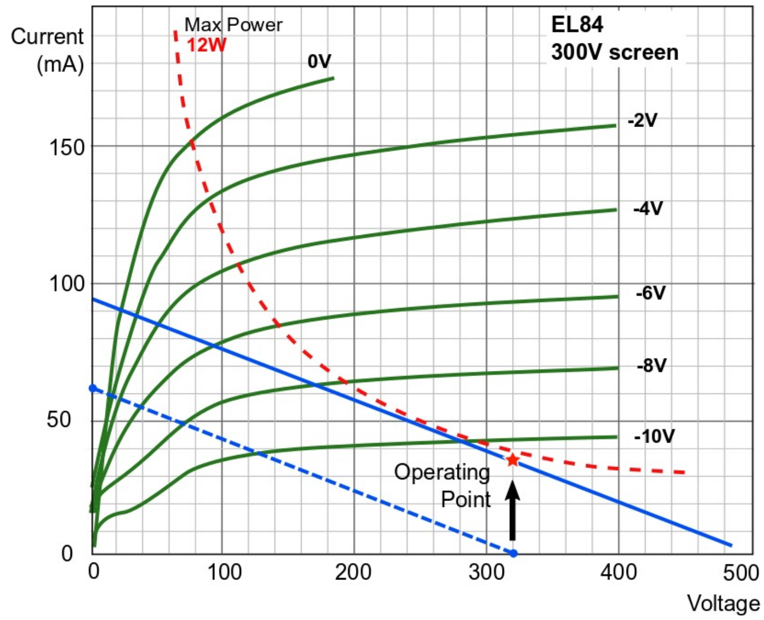
Speaker impedance

It is important to remember that the speaker is not a uniform resistance of 8 ohms. All speakers have impedance that varies with frequency. They are rated typically at 4 ohms, 8 ohms, 16 ohms, but this is just a nominal value generally close to the overall impedance. A typical 8 ohm speaker could actually have a wide range of impedance from 3 or 4 ohms at some frequencies or 10 or 20 ohms or higher at other points. The job of the speaker is to reproduce frequencies as uniformly as possible across the audible frequency spectrum, and there are physical properties to the cone and suspension and a wide range of other things that will cause the impedance to be higher or lower across frequencies.

Let's say in our example that B+ voltage is 310V (you'll see why I chose this in a minute) and we have a 5,000 ohm load, we can start to work on a load line and operating point. Like the triode tube in our driver stage, we can look at the tube characteristics provided by the manufacturer. The process will be a bit different because this is a pentode tube and because of our output transformer and speaker load. One challenge is that the load lines will vary depending on the screen voltage. Sometimes the datasheet will have multiple charts for different screen voltages. Below is one for 300V. Sometimes you might find one for 250V or something else. It's ok if this isn't perfect, we are just getting somewhat close to the theoretical operating point and it becomes important to test and modify the circuit in actual operation to optimize it anyway.

In the chart shown, the blue dotted line is the initial load line for 310V and 5,000 ohms (62mA of current if the entire voltage were dropped). But remember that we do not have a resistive load. (Actually, the transformer has a small amount of DC resistance in the primary winding, but this is so small we will ignore it.) So at the quiescent point with no AC signal, nearly the entire B+ voltage will be on the anode, so our operating point voltage must be around 310V. But if we stay on our initial load line at this point, it would not be useful because it's at cutoff and could then only amplify the positive side of the signal. We need to bias the tube and move this load line up.

What we will do is choose an amount of current that we want at the 310V operating point. To maximize our power from this tube, let's pick a level just below the maximum rating, shown with the red dotted line. The 12W maximum would be at approximately 40mA of current. So we shift our load line up, keeping the same slope until we reach an operating point that is 310V and around 40mA. Finding the nearest grid line, this is pretty close to 10V. See that? But it's a bit below, so let's pick 11V. Now, if we need 11V on the grid relative to cathode, then we can again use the cathode bias technique and hold the grid at 0V while we elevate the cathode to +11V. Using Ohm's law, if we are estimating 40mA of current dropping across 11V, this gives us a resistance of 275 ohms, or let's pick 270 which is a common resistor value. This will dissipate about 440mA of power, so we might choose a resistor rated at least double or more for a margin of safety. I would probably choose 2 watts. (Some power tubes will dissipate much more power, so you should always calculate this!) We will bypass this capacitor with an electrolytic capacitor. An example might be 220uF rated at 50V.



Are you still with me? I know it's a lot, but you are getting it, right? Summarizing for a moment… we put 310V on the anode and because there is little resistive load under DC conditions, this is our operating point voltage. We elevate the cathode to 11V and our tube will be dissipating 40mA of current in its quiescent state, shifting our load line up. Now under AC conditions the signal will vary the voltage of the grid by say 8 volts or so depending on input level and amplification in the driver stage, and this will swing the anode voltage very widely, up to a few hundred volts. And importantly, it will put this alternating current across the transformer, transferred back down to a low voltage and higher current to drive the speakers. Notice here that the anode voltage can actually rise higher than the B+ supply because the transformer stores energy in its magnetic field and it will resist the change in current, causing it to create a temporary higher voltage.

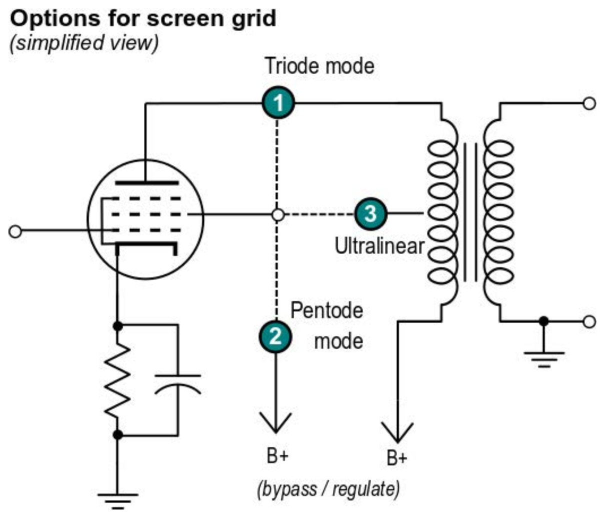
A few more quick notes. We haven't illustrated the AC load line in the chart, and will not try to do this because the speaker impedance varies across frequencies. But you could visualize an AC load line pivoting around this operating point, sometimes a higher load and sometimes lower, altering the slope of the line, but always around the operating point. Also, you may be wondering why I chose 310V but the maximum rating for the tube is 300V. This is actually a maximum voltage relative to the cathode, which we have elevated to 11V, so should be safe. In fact, the EL84 tube is a pretty rugged dude, and the original Dynaco ST-35 ran it pretty hard, so you'll see from voltages in the appendix that we run it pretty high, too.

At this time, we know enough about the operating point of the circuit to have our initial design and can modify if needed. We also need to get back to that screen. You thought I forgot, didn't you? Take a break, get a snack. Almost there!

SCREEN VOLTAGE AND ULTRALINEAR MODE

There are several options for how a pentode tube can be operated. This kit is designed for ultralinear mode, and so I will not go over a circuit for a standard pentode design in much detail. But I will cover a few basics so you understand the options and how ultralinear mode fits into those options.

First, and simply, you can take any pentode tube and operate it like a triode ("triode mode" or "triode strapped") by simply connecting the screen to the anode, and the voltage of each will always be identical. Both the anode and screen will swing together in voltage across the output load as a signal is applied and current passes through both of them. Output power will be lower, and the tube characteristics will generally have more lower order harmonic distortion—some consider this a desirable "sound" for a tube amplifier, though obviously perception of sound is highly subjective. The grid-curve chart shown earlier for the EL84 is not in triode mode and would look different if the tube were triode-strapped.



Second, to operate in pentode mode, you would put a DC voltage on the screen at around the B+ level or somewhat lower, and then you would also bypass the AC by using a capacitor to ground (not shown) so that your screen stays at a constant voltage. Only the anode will swing in voltage as the signal is applied. Pentode mode will have more power output possible, and generally produces more higher order harmonic distortion—some would consider a less pleasing sound, but again a subjective generalization. (I know I have just mentioned a few types of distortion and have not discussed this topic much yet; we will come back to it soon in more detail.)

Ok, these are two fine options. But some clever folks in the 1950s made a discovery of a third option that has very interesting results. You could think about triode mode and pentode mode as two ends of an extreme with respect to the screen voltage. On the triode end, the screen voltage is entirely determined by the anode (plate) voltage which means it changes with the audio signal, and on the pentode end the anode voltage is completely independent and the screen voltage is held at a constant level.

Each of these modes has their own form of non-linear distortion, but they tend to curve in opposite directions. If we were to take a position part-way in between these two extremes and put that on the screen, there's a sort of magic that happens at the right spot where we can get a more linear operation (less distortion) and still get most of the benefits of pentode mode for power. By putting a special tap on the output transformer, we can vary the screen voltage in proportion to the anode but not as much as in triode mode. It was found that a tap around 40-45% of the primary is the ideal spot. This was called "Ultralinear." It could be considered a form of negative feedback.

There is deeper theory of ultralinear operation you can study elsewhere. It's used in our case to have optimized performance: good power output and low distortion. Critics of ultralinear might believe triode mode sounds better or pentode performs better, but in my opinion this is a good option given design objectives of low distortion, reasonable cost and sufficient power output from the selected components.

You can re-wire this kit to operate in triode mode by connecting the screen to the plate (tying off the UL wires unused). You would still have a resistor on the screen pin for stability reasons to prevent oscillation, similar to a grid stopper resistor we will also put on the control grids. The screen is a sensitive part of the tube, and the datasheet will tell you the maximum voltage and power dissipation of the screen.

EL84 tubes are rated for 300V on the screen and about 2W of power—notice that there is some current passing through the screen as some electrons will be hitting it, even while most are passing through and on to the anode.

PUSH PULL DESIGNS

Ok, so far we have only reviewed a single-ended tube output stage. It was important to have this as a starting point. Single-ended designs are beautiful, I love them dearly, but they can't put out a lot of power. I mean, it's one tube, and well, if one tube is good, what might we do with, mmm, let's say, TWO tubes?

Most amps with more than about 10W per channel are going to be in a circuit called Push-Pull. It's two tubes--they are best buddies, working together, but think of them as facing opposite directions. When one pushes, the other pulls. It's not a tug of war, they like each other. No one is trying to pull anyone into a mud puddle. More like cutting lumber with one of those big two-person saws.

Each tube has the audio signal on their grid, but one of them has a phase-inverted version of the signal. When the original audio signal rises in voltage, one tube will conduct more, while the other tube conducts less—push and pull. More on this soon.

Two tubes of course can control more current than a single tube, giving us more power potential. But it's even better than that. The output power can be even higher. I won't go into some of the deeper theory, but I will note a few specific benefits that go beyond just having two tubes.

Let's start with some limitations of the single-ended design and I'll show you how a push-pull amplifier is different. First, regarding the output transformer.

I will not get into the design and physics of a transformer in detail, but remember we need an AC current through a coil of wire on the primary side of the transformer and this creates magnetic flux in the iron core of the transformer that induces a current into the coil of wire on the secondary side of the transformer connected to the speaker. For a single-ended amplifier, there must be a steady state DC current in the primary coil and passing through the tube, and this current is modulated by the AC music signal on the grid. But for a transformer, if a DC current is passed through the coil, then it will saturate the magnetic core. We need it to not be saturated; we need it to be able to charge and discharge this magnetic flux. The solution for a single-ended output transformer is to insert a very small air gap so that the magnetic core is not a continuous path but has a break in it. It's really small, like piece of paper kind of small. But it's enough that the core doesn't saturate and the transformer can do its thing to get the AC from primary side to the appropriately transformed secondary voltage and current.

Multiple Tube Options

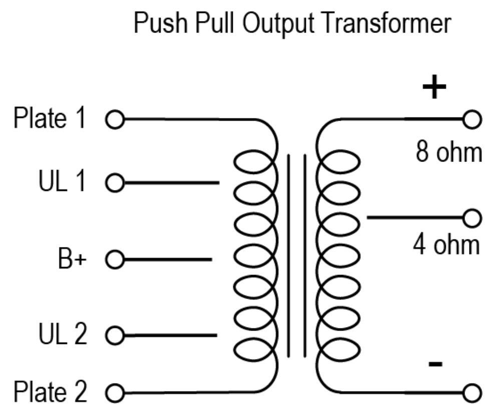
Push-pull is not the only configuration for multiple tubes. You could also have parallel single-ended tube designs which have certain benefits but also some downsides or considerations.

Two powerful tubes like KT-88 or 6550 tubes in a push-pull circuit could produce fairly high output power, let's say 50 or 60 watts or more. But to go even higher, some push pull designs would have a quad of tubes—two push-pull pairs in parallel per channel to reach even higher power or provide other benefits. High powered amplifier designs might typically be set up in fixed bias configurations.

The trade-off to inserting this gap is that the transformer core has lower inductance. We need inductance so that changes in current will create changing magnetic flux that induces a voltage in the secondary (output) side of the transformer. In particular at low frequencies, we need a LOT of inductance. If we have an air gap so the core doesn't saturate with flux, but this negatively impacts the inductance, we'll end up with poor low frequency response. We want good bass, am I right? Ok, then the next solution is to make the core bigger. Bigger iron equals more inductance. So, a single-ended output transformer must be physically large for a given amount of output power, assuming we want good low frequency response. Larger means more expensive, heavier, and there are further down-sides when the core is made bigger that I won't get into here. The point is that the DC current and air gap are a limitation in a single-ended transformer, especially for efficiency and low-frequency performance.

Now for a push-pull output transformer, we can do a little special something. On the primary side of the transformer, each tube gets its own coil of wire (half of the primary) that is wound in a certain direction so that they are in opposite magnetic polarity. They each have DC current going through them, so the tubes can do their thing to modulate it and create an AC voltage, but since these are in opposite polarity, their DC magnetization cancels each other out—no core saturation! Yeah, baby. Just sweet AC signal to worry about. These two halves of the primary get AC signals in opposite phase, but because of the way their windings are connected, the resulting magnetic flux adds in the core rather than canceling—the tubes are working together! The secondary side coil gets an induced current as expected and this goes to the speaker as the combined output signal.

Let's look at a schematic diagram of a push-pull output transformer and you'll see this visualized. See how the B+ is shown on the primary side going into the middle of the schematic diagram (both coils share this starting B+ voltage) and then one tube plate (anode) is at one end, and the other plate is at the other end. This is illustrative of the two coils in opposing polarity. You see here also that each side has its own ultralinear tap, too.



Ok, cool. So that's one benefit: the output transformer solves the problem of DC current saturation, making it more efficient to generate power to the speakers. Next, let's look at the class of the amplifier.

A single-ended tube amplifier must be Class A, meaning it is always amplifying all parts of the audio waveform. The operating point is set somewhere in the middle of the load line and as the grid voltage changes, the current through the tube goes up and down, but there is always some current flowing. And in fact, you really need to keep the tube operating within a reasonable region to avoid distortion at the extreme ends. (You'll understand this more in the next section about harmonic distortion.)

A push-pull amplifier could be operated in Class A, but having two tubes opens the possibility to improve efficiency by allowing each tube to conduct for only part of the waveform at higher signal levels. The first tube says, "Hey man, when the signal is up on the high points, I got this, ok?" And the other tube is like, "Bro, that's cool. I'm gonna chill while you take care of it. I got you in the other direction. When the signal goes down to the lowest peaks, you take a break."

The most common method of doing this is called Class AB, where for small amplitude signals, the amp is fully in Class A with both tubes amplifying, but at higher amplitudes, one of the tubes goes into cutoff.

Take a look at this graph that puts one of our load lines upside down so we are looking at two tubes working together in opposite phase. A sine wave signal is in between them, so you can visualize on either chart the action of the tube as that signal changes in value up or down the load line. The shaded areas show the zone that each tube operates in. Can you see how at the peak of the signal in the first half of the waveform, the first tube amplifies but the second tube reaches cutoff so is passing no current. Then at the peak of the signal on the second half of the waveform, it's reversed where the second tube would amplify while the first one cuts off. If this signal were smaller (lower amplitude), you could imagine how both tubes might be amplifying all of it because it would be within the class A zone where both tubes are still passing current for the entire signal.

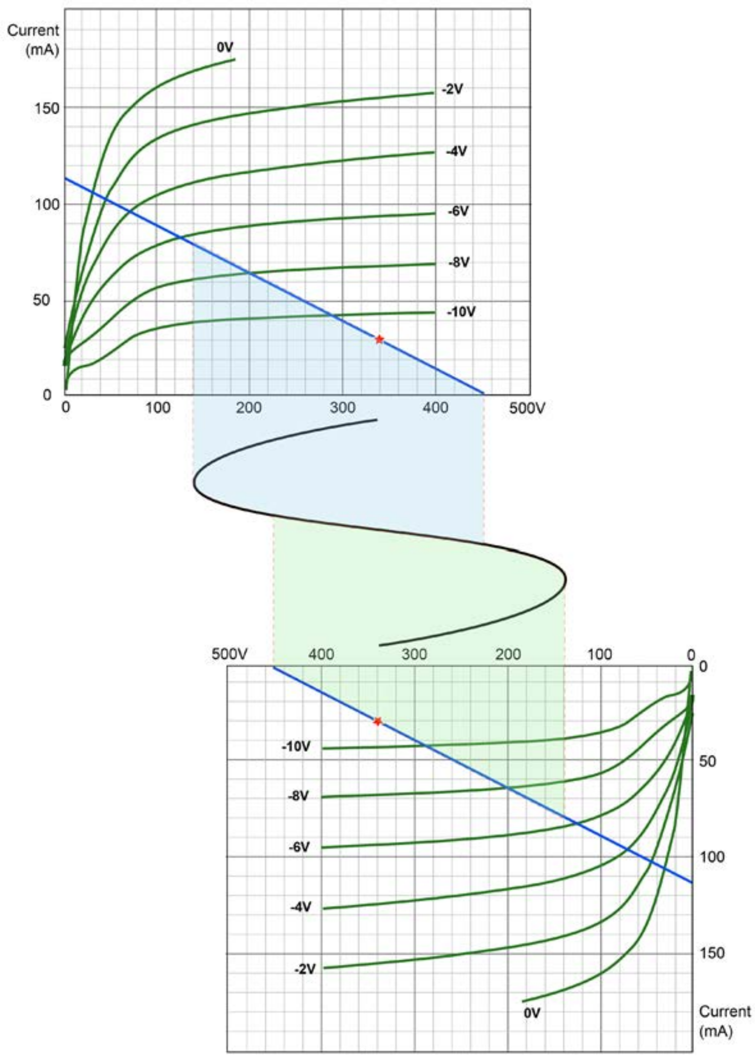
So operating in Class AB allows us to amplify this signal more efficiently using two tubes instead of one. If we want to swing a really big signal (more potential power), we don't need it to ALL fit into the single load line for each tube. We have more usable area of the combined load line for the larger signal.

Because the two tubes work in opposite phase, the voltage swing across the full primary winding is much larger than either tube could produce alone.

Pure Class B would set the cutoff point precisely where the signal switches between positive and negative but has some undesirable qualities in how the tubes transition to be on or off, which is why AB is more often used.

There is much more to this topic I won't go into here and the most technical among you might already be pointing out how this fails to fully address all of the dynamics of a push-pull load line, but you get the point, right? It's enough to get the concepts, and you can always read and research more as you keep learning!

There you go—two really cool features of push-pull amps that result in greater output power when two tubes work together. One is an improvement to the performance of the output transformer, and the other is ability to swing a bigger voltage across the combined load line.



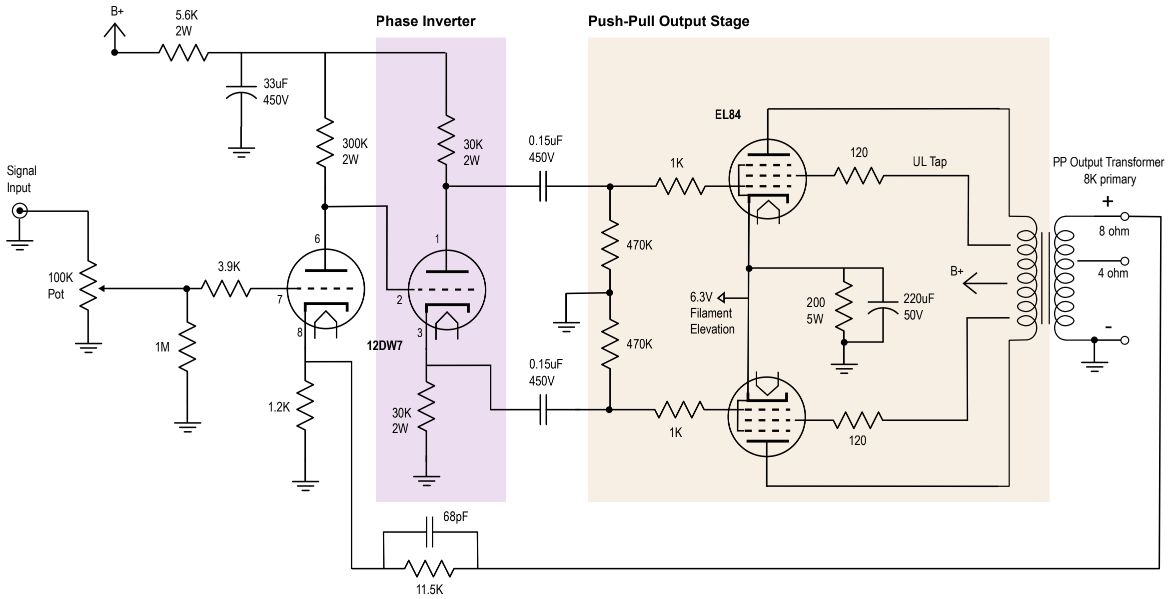
PHASE INVERTER

Now, we have one more piece of this puzzle to fit in. Remember these two output tubes are working together in opposite phase—when one conducts more current, the other conducts less. The way we do this is to put a phase-inverted signal on one of the tubes, so that if the original audio signal is say +1V going into first tube, then the inverted signal is -1V going into the other tube. But how do we set up the signal for this arrangement?

The answer is a phase inverter. Now there are a lot of ways to do phase inversion, and I'm only going to talk about one method used in this amplifier. It goes by various names: split-load, concertina, or cathodyne are all names to describe a method of taking a single input signal and producing two identical but phase-inverted output signals. Here's how it works.

You learned earlier that we can put a resistance load on the anode of a triode tube and as the grid signal changes, the current will rise and fall creating a voltage on this load due to Ohm's Law. You also learned that when we have a cathode bias resistor, the changing current will ALSO cause the cathode voltage to rise and fall with the changing current. (And we sometimes bypass the cathode using a capacitor so that this doesn't happen.) The voltage on the anode is inverted relative to the original signal, while the voltage on the cathode is in the same phase as the original. Mmmm, interesting. If we could adjust the cathode load to be identical to the anode load, then we would have two equal but opposite phase signals coming out of this tube—one from the anode and one from the cathode. And that's exactly what the split-load or concertina phase inverter is doing!

Looking at our amplifier schematic, you can see we put a triode tube in front of the pair of output tubes and give the anode and cathode the same resistance value. Then we can take the signal from the anode and cathode, pass it through a coupling capacitor, and Ta-Da! We have the two signals we need to drive our push-pull output tubes.



Now let's get just a little more specific to our actual application in this amplifier. The original Dynaco ST-35 and my Pacific 63 amplifier use a 12DW7 tube, which is a dual section tube that has an interesting design. One section is a high-amplification triode basically identical to the 12AX7 tube (you may recognize this as a high-gain tube) and the other section is a lower-amplification triode that is basically identical to the 12AU7 tube, commonly used in a buffer stage because it has lower output impedance and other characteristics that makes it suitable for driving an output load.

In our amplifier, we are going to use both halves. The first is a normal gain stage intended to bring our small input signal up to a larger level. Then we will send this into the second half where we split the load as a phase-inverter. From there, we can couple it through a pair of coupling capacitors to the output tube grids.

You will notice something on the schematic here—how do we take our signal from the first gain stage and put it into the grid of the phase inverter stage? No coupling capacitor. This is a directly-coupled stage where we are taking the signal directly from the anode of the first stage and putting that signal on the grid of the phase inverter second stage, even at the relatively high voltage anode level.

This is part of what makes the split-load or concertina phase inverter rather elegant. Since we split the load by increasing the size of the cathode resistor, this means we have a much higher voltage on the cathode than a typical cathode-biased gain stage would have. Imagine we had 4mA of current through that phase inverter, and the two load resistors are 30k ohms each, this

Why is the output signal inverted on the anode, but in phase on the cathode?

The grid regulates the amount of current flowing through the tube. A more positive grid voltage increases tube current, while a more negative grid voltage decreases it.

At the anode, increased current produces a larger voltage drop across the plate load resistor, causing the anode voltage to fall. Decreased current causes the anode voltage to rise. The result is an output signal at the anode that is inverted relative to the grid signal.

At the cathode, increased current raises the voltage across the cathode resistor, while decreased current lowers it. Because the cathode voltage moves in the same direction as the grid voltage, the cathode output signal is in phase with the input.

means the cathode is going to be sitting at 120V DC. This is relatively close to the anode voltage of the previous stage, making it a good candidate for direct-coupling.

By direct coupling, we can avoid the need for a coupling capacitor, which has a few benefits. First, it's simply one less component—reducing cost and keeping the signal path as simple and clean as possible. It also avoids the phase shift introduced by a coupling capacitor, which can have implications for the stability of a global negative feedback circuit. (I'll go over feedback in a later section.)

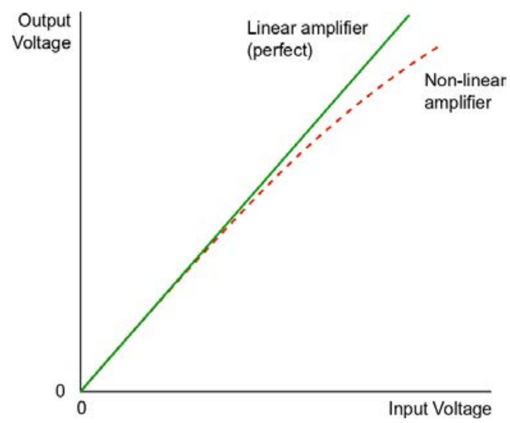
Without going into detail here, the short version is that a capacitor introduces frequency-dependent phase shift in an AC signal. In a complex amplifier with multiple stages, each coupled by a capacitor, these phase shifts can accumulate—especially at low frequencies. When global negative feedback is applied to control distortion, excessive phase shift can reduce stability. In the worst case, if the total phase shift approaches 180 degrees, negative feedback can effectively become positive feedback, leading to instability.

For this reason, minimizing the number of coupling capacitors is a useful design consideration. The ability to directly couple stages in this amplifier is therefore a meaningful benefit.

UNDERSTANDING DISTORTION

It's time to try and cover the topic of distortion. It certainly sounds like a bad thing for a high fidelity amplifier, but what exactly is it? Let's take a closer look, and we may dispel a myth or two in the process. First, I should say that there are multiple types of distortion, and I will mainly be discussing one type here, harmonic distortion, which is commonly analyzed in an amplifier—and even this we will only just begin to explore.

In a perfect amplifier, we would have an input voltage that varies over time (the audio signal) and an amplified output voltage that varies over time exactly proportional to the input voltage but at a larger amplitude. We want this to be a linear relationship: Vout = Vin * gain, no matter what level of Vin. As the input voltage rises, the larger output voltage rises proportionally.



But in the real world things are not perfect. Perhaps over the range of possible voltages, the tube is doing its best (c'mon give the guy a break!) but as the input voltage on the grid moves up and down, the output voltage on the anode is changing in a similar, but not exactly identical way across all voltage levels. Perhaps we would expect 1V input to result in 10V output, but the amplifier actually puts out 9.5V.

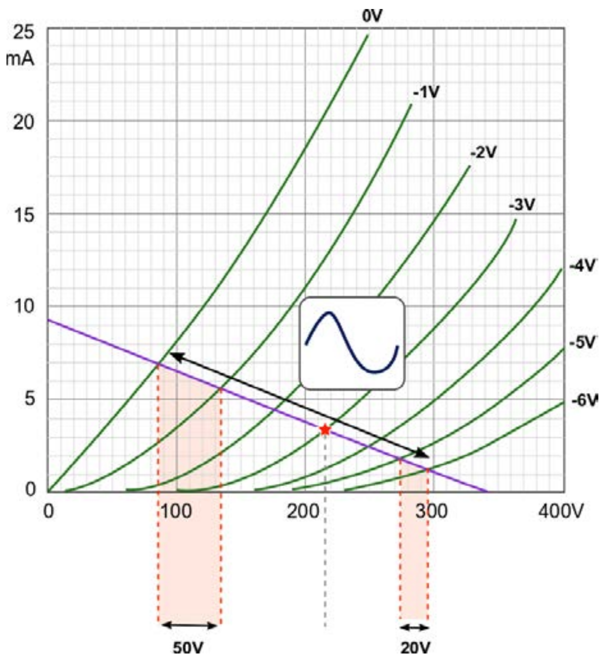
This non-linearity of the range of output voltages relative to the range of input voltages is distortion. This is an attribute of all tube amplifiers, and some might argue part of their "tube sound." In that respect, zero distortion might not actually be the goal, but certainly we want the output to be as close a representation of the original as possible, so we will aim to have relatively low distortion.

Ok, so how can we visualize and understand distortion even more? First, I want to emphasize one important point. The amplifier has no idea about the concept of an audio signal or sine wave, which incorporates time. The tube is not first listening to your Bob Dylan song and then playing it back to you louder, nor is it looking at a sine wave for a millisecond or two and then recreating a new sine wave afterward, hopefully similar. The amplifier sees an input voltage at a point in time and puts out an output voltage at that exact point in time, or close enough to consider instantaneous. This will be important as we get into visualizing distortion and discussing feedback. We use sine waves to communicate concepts of frequency and distortion, but the behavior of the amplifier occurs at a single point in time. That said, there are elements of time in the ability of the amplifier to react quickly to fast changes in voltage and demands on current in order to perform its job properly at the present point in time, but the point I'm making is that we need to consider distortion of a sine wave as an effect that we observe over time, but that is happening at any given instant.

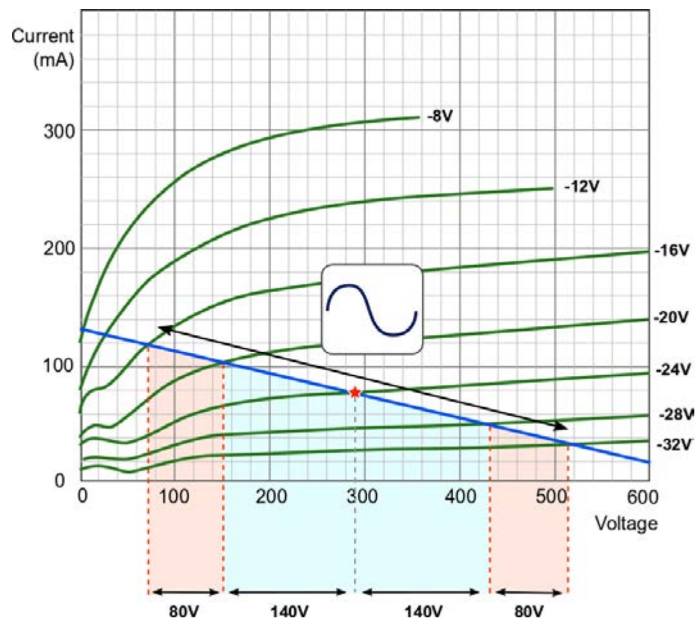
Now let's reintroduce the concept of time and consider a sine wave as a voltage changing over time, and the output voltage isn't perfectly identical in shape to the input sine wave. Remember how we drew a straight load line, expecting that variation of the grid voltage would cause the anode voltage to swing up and down on that line. But if the characteristics of the tube means the grid lines are not evenly spaced apart, then we can't expect the output to be a perfect replica. Two examples are illustrated below, where grid curves are either closer together at one end, or at both ends, causing non-linearity.

Examples of harmonic distortion

If grid curves are wider on one side than another, gain is asymmetrical, creating even-order harmonic distortion



If grid curves are wider in the center than at the extremes, this causes symmetrical, odd-order harmonic distortion



You can see in these examples that an equal change in grid voltage would not cause a proportionally equal change in anode voltage at all areas of the load line. When the gain is asymmetrical around the operating point, the output waveform will be misshapen, or distorted, on one side but not the other. This will cause a type of harmonic distortion that is even-order. If the non-linear effect is symmetrical on both sides of the waveform, then this causes odd-order harmonic distortion.

You might also consider what happens if the operating point were too far to the left or right on the load line, or if the input signal were too large relative to the span of grid curves—if the input voltage pushes into saturation (near or above 0 volts on the grid) or down to cutoff (deeply negative grid curves, with low or no current), then obviously the output signal will be extremely distorted and a sine wave would appear as flattened on top or bottom. This is what we would refer to as clipping. Guitar amplifiers operate closer to these conditions to achieve a desirable overdrive or distortion sound, but obviously in a hifi amplifier, this is undesirable.

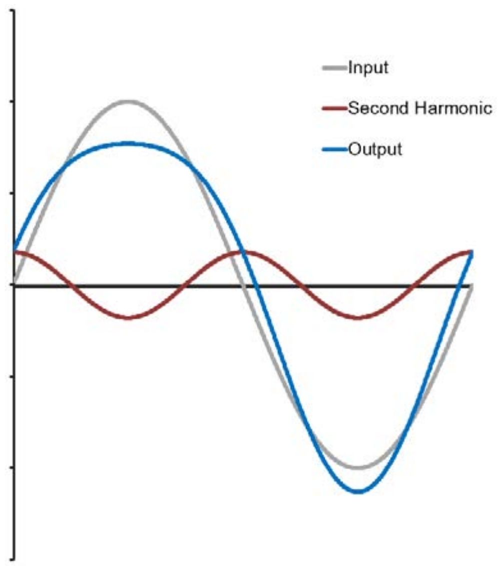
Now, we're ready to look at harmonic distortion another way to visualize these even and odd-order harmonics! I love this part, because the natural world, physics and math are sometimes like magic.

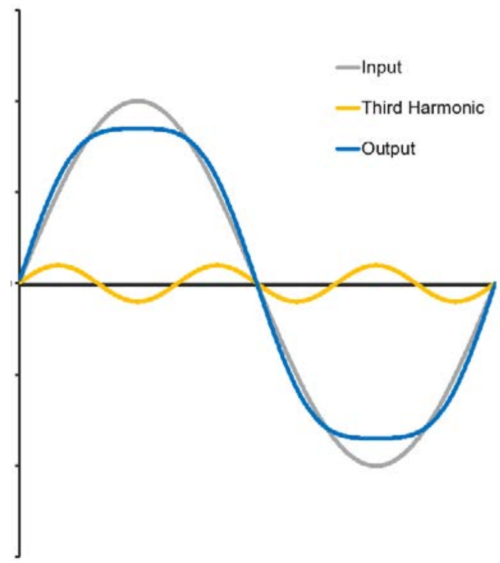
The sine wave that we input is at a frequency, let's say 1000 Hz. We can call this the fundamental frequency (FF). Multiples of this frequency are the harmonics: the second harmonic is 2x the fundamental frequency, or 2000 Hz; the third harmonic is 3x the fundamental frequency, or 3000 Hz; and so on... fourth, fifth, etc.

An output sine wave that has been distorted by a non-linear amplifier will have a waveform that doesn't look exactly like the original sine wave. And this distorted waveform is equal to the fundamental frequency plus some combination of harmonics at lower amplitudes. See the charts below that are illustrative examples if we took a pure sine wave input and add a second or third harmonic at lower amplitude. The output is a combination of those frequencies added together.

Distorted waveforms

In both examples here, the blue distorted waveform is equivalent to the combination of the input sine wave and a second or third-order harmonic (at 2x or 3x the frequency).





Are you getting it? The distorted waveform is **the same** as multiple sine waves at different frequencies and amplitudes added together.[4] Going back to our amplifier, it's important to re-emphasize: the output waveform is distorted due to various causes, and this is equivalent to a fundamental frequency plus harmonics. We could be tempted to have a mental image that the harmonic frequencies are created in some other way and then added to the original signal, with harmonics as the cause of a distorted output. I suggest this is not the best way to think about it. Remember how I pointed out that amplification is happening instantaneously. The waveform is distorted because gain is not uniform across all input levels, and when that happens, we can watch a waveform over time that can be described as the original fundamental frequency plus harmonics. The net effect is, however, exactly the same as if we did truly have multiple sine waves generated at different frequencies, amplitudes and phases and combined together.

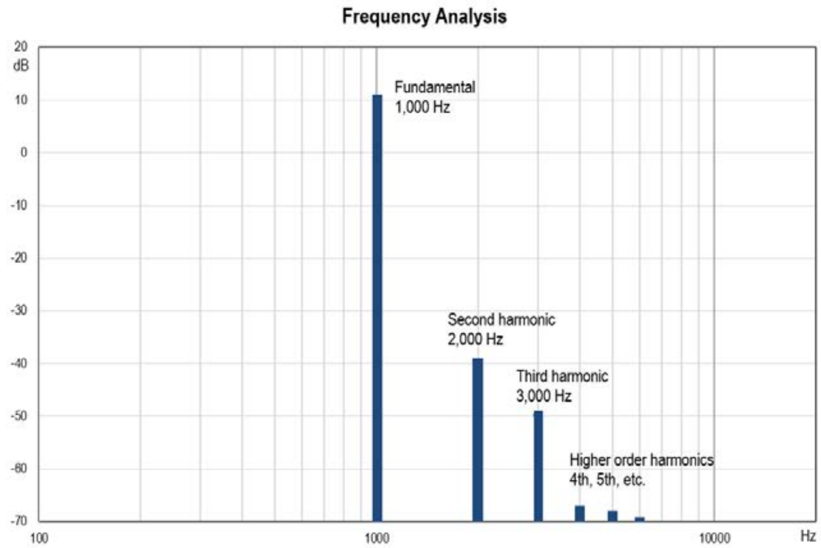
What this means is we have a way to measure the harmonic distortion of the amplifier. While you could look at the waveform on an oscilloscope and subjectively say, "well, it looks pretty close to a sine wave," and this is fine for basic observation and detection of heavy distortion and clipping, you would not be able to easily observe more subtle distortion. We can do better.

Since the distorted waveform is identical to multiple sine waves at different frequencies, we can mea-

4 Harmonics may also be at a different phase than the fundamental frequency, but this is not introduced here. For illustrative reasons, the second order harmonic shown here is offset by 90 degrees in phase.

sure and visualize the amplitudes of those harmonic frequencies. Here's where we need a new graph: instead of showing amplitude versus time of a waveform, we can show amplitude versus frequency. A frequency analyzer can do this for us. Below is an illustrative chart demonstrating measurements of harmonic distortion on the frequency spectrum.

You'll see this uses decibels for the vertical amplitude, with a 1000 Hz fundamental frequency at a certain level, and the second harmonic is much lower about 50 dB below the FF. The third harmonic is lower still, and very tiny amounts of higher order harmonics. This is typical of a single-ended tube amplifier: most distortion is second and third harmonic. I'll soon explain how a push-pull amplifier has a somewhat different profile.



We could then calculate a summary measure to add up all these harmonics and this is what you see reported as Total Harmonic Distortion (THD) usually as a percentage of the fundamental frequency. High quality modern solid state amplifiers will have extremely low values, like 0.01% THD. Tube amplifiers by nature will typically have higher THD, and harmonic distortion will nearly always be proportional to output level—higher volume (higher grid voltage change) will have more distortion.

Now we should address one fact about harmonic distortion related to push-pull amplifiers. The load lines shown earlier illustrated harmonic distortion occurring because of nonlinearity in the tube behavior. But they were a single load line, and a typical single-ended amplifier will have some second- and third-order harmonic distortion. But we learned how in a push-pull amplifier we have two tubes working on phase-inverted signals--when one pushes, the other pulls, and it's like having two load lines, one a mirror image of the other. This means that a push-pull amplifier could have one tube causing nonlinearity in one direction, and the other tube causing nonlinearity in the opposite direction. So what might happen here? Yup, a push-pull amplifier is going to neutralize some harmonic distortion. Specifically, second-order harmonic distortion, which is an even-order effect that comes from asymmetry in the transfer curve. Tube A distorts on the positive swing, tube B distorts on the negative swing, the output transformer flips one of these to recombine, so they cancel each other out.

Now before you get too excited, the implications are different for third-order harmonic distortion, which is an odd-order effect resulting from symmetrical curvature. The distortion changes polarity when the signal changes polarity. So tube A and tube B each might produce third-order harmonic distortion that ends up in-phase and so they are summed in the transformer. Dang it.

All that to say, a push-pull tube stage will have minimal second-order harmonic distortion and more third-order harmonic distortion. Keep in mind, this canceling & summing from a push-pull design is only for the push-pull output stage. The initial gain stage is a triode using single-ended amplification, so the first gain stage might be adding second order harmonic distortion into the signal that is passed along through the phase inverter and then amplified in the output stage. Make sense? Each tube might be adding harmonic distortion along the chain.

We have not yet discussed what harmonic distortion means in terms of how the amplifier sounds. How much distortion would be audible? What does second-order harmonic distortion sound like versus third-order or higher? You can find examples online or create your own if you have computer software or signal generators, to hear a sine wave with a second or third harmonic mixed in. You will notice that second and third order harmonics have distinct tonal qualities. In music, doubling the frequency is the same as a one octave higher note, and some will argue that a second-harmonic is "better" or more acceptable in sound because it is "in tune" whereas some odd-order harmonics are not musically related. There is much debate on this and conflicting tests and research about people's perceptions of which sounds better. There is typically consensus that lower-order harmonics (second and third) are more acceptable than excess amounts of higher order harmonics.

On the subject of how much THD is acceptable, again there is much debate and probably the answer is: it depends, on many factors—the type of distortion, source, etc. Purists will say it should be as close to zero as possible. Some research was done years ago that gave clues many people may not be able to detect it audibly below 0.75% and that it may not be noticed or considered interfering with the sound until 2-3% or even higher with complex sources such as music.[5] This enters into complex or subjective areas of what people believe sounds good or not, learned experience of critical listening, variability in human hearing, etc.

I should emphasize again that there is much depth to this topic and I'm only covering some simple concepts to explain it. There are other types of distortion, such as intermodulation distortion, that can be important, too. For now, I hope you will at least understand some basics of harmonic distortion, examples of how it could be caused and how we can see or measure it.

Measure or Listen?

I believe in measurements to help understand performance of the amplifier or speakers. I've learned too often my ears play tricks on me and my own psychology can lead me to think something sounds good because I want it to (or vice versa). Remember also that one of the most critical parts of how it sounds is your speakers. There's a whole other world of speaker selection (or DIY design!) and room environment and treatments to think about, and that's all part of the fun of this hobby. Also remember, at the end of the day, if you can play a system and it sounds good to you, you win!

NEGATIVE FEEDBACK

If we believe an excess of distortion is not ideal, then the real question remaining to ask is: how can we limit it?

There is a technique we will review in a moment, but first I'll just mention that the component selection and design of the circuit is the most important starting point. This kit uses 12DW7 and EL84 tubes, and is designed with a certain operating point, power supply and filtering, load lines, output transformer, etc. All of these choices result in a certain performance and level of distortion. There are many types of tubes, each with different characteristics for amplification factor, transconductance, grid curves. There are also many other circuit design choices different than ours and with different implications for power and distortion. All that to say, we start with components and a circuit designed as best we can for our intended performance, size or cost objectives, and then we can consider one more tool in our tool belt: negative feedback.

As with many other things, there are varying points of view about the use of negative feedback. Some may view this as an undesirable way to improve the performance of the amplifier, somehow compromising on sound quality or purity of design. Certainly if the circuit design and performance is poor, and negative feedback is used to try and put lipstick on a pig, then I can understand. But to categorically view negative feedback as something to avoid is, in my opinion, missing an absolutely beneficial technique.

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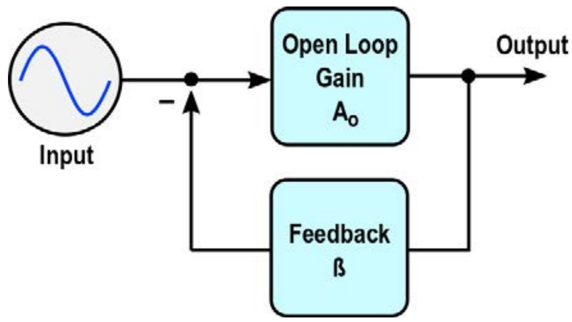
5 Check out the Radiotron Designer's Handbook , published in the 1950s and available online in PDF, for some interesting information about distortion, among other things.

So what is negative feedback and how does it work? I'll try to explain it in the way that makes sense to me, acknowledging there are others who can cover the theory and equations much better if you want to learn more. Feedback as a general concept can be found in all sorts of places where the actual result is compared to a reference level. Consider a few examples:

- A thermostat detects the room is too cold and turns on the furnace until the temperature meets an expected level and then the thermostat turns it off.
- Your car's cruise control measures how fast the car is going and modulates the gas to the engine if the car is going too slow or fast relative to a certain level.
- Your toilet has a float that detects the water level and opens or closes a water valve until the tank is filled to the expected point.

In the amplification process we have looked at so far, we have an input voltage and an output voltage that is larger based on the gain of the amplifier. This can be called "open loop gain" (meaning there is no feedback loop) and we know this gain is not perfectly uniform for all input voltages. It is sort of "out of control," so to speak.



A negative feedback loop can be used to change the circuit and create a different "closed loop gain" that takes a portion of the output voltage—the "feedback fraction"—and subtracts it from the input to create a new control voltage. A general feedback structure is shown here.

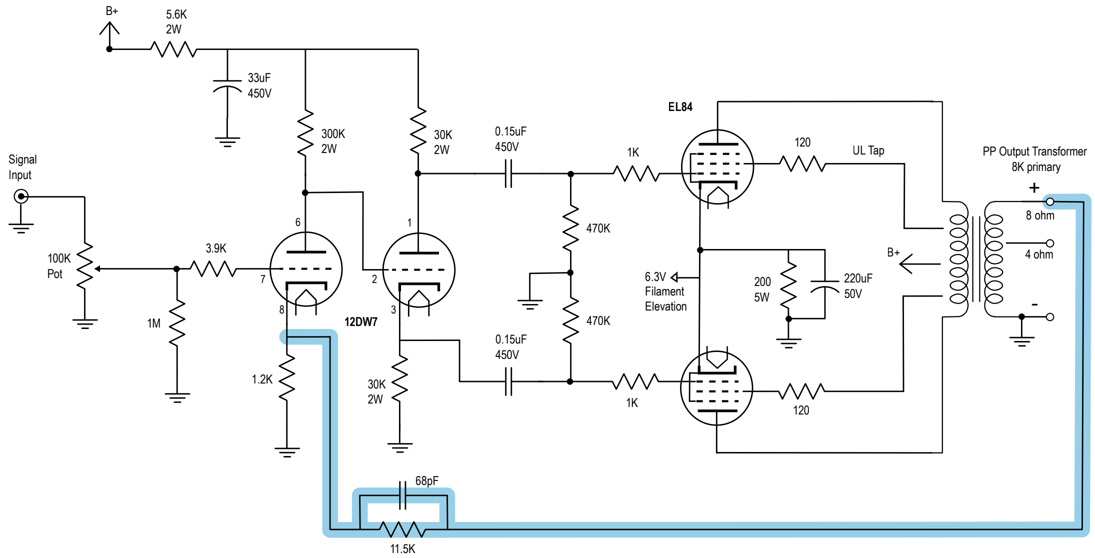
Taking a portion of the output and subtracting it from the input will attenuate the control voltage, lowering overall gain. This is a small sacrifice we make to use negative feedback, and presumes we have a substantial amount of gain to begin with. We have now tied the input to the output at a certain relationship, and the actual gain of the amplified signal is now determined entirely by this feedback fraction, instead of the open loop gain on its own. In order to achieve a state of equilibrium, the control voltage will compensate higher or lower depending on the relationship the feedback loop sees between output and input. This is exactly what we need to deal with non-linearity and is why it will reduce harmonic distortion.

Before we look at our actual circuit again, I want to address one more mental trap that is easy to fall into. It is easy to trace a circuit at our slow and methodical human pace: "Ok, our input signal is coming in here… [finger pointing on schematic] …and then the tube amplifies the signal and it comes out over here… and then the feedback loop sends it down here where it comes back over to the input… and then, um, it goes through the tube again a second time? And over and over?" This is not the right way to think about it. There is not an iterative process happening over time, it is an almost instantaneous influence on the control voltage when we add in the feedback loop.[6] If there were an imbalance based on the feedback, the voltages would compensate faster than any audio frequency that we care about.

Negative feedback can be implemented in various ways: local feedback around one tube, global feedback around an entire circuit, etc. Let's take another look at our schematic and see how it is using negative feedback. Notice that we have a connection from the 8 ohm output of the transformer all the way back, through a 11.5k resistor and to the cathode of the gain stage of the 12DW7 tube. This is one type of feedback, using a portion of that output signal and applying it to the cathode. Remember we chose not to bypass our cathode resistor, so any variance in the cathode is going to impact the voltage difference to grid and affect our gain. So we now have the feedback relationship established with some of the

6 This subject is obviously more complicated than I'm describing here, and time can play a factor in various ways, but the intent is to explain feedback for very basic understanding, not a detailed technical description.

output signal being effectively subtracted from the input stage. The choice of 11.5k is the amount of feedback—we could increase or decrease this resistor, and will have more or less feedback, controlling distortion. The tradeoff is that we give up gain. I selected this value as a good balance to get to 15W of maximum output power with a reasonable input signal voltage, so we get nice control of harmonic distortion but still enough gain to provide the power we want.



How do we know this is negative feedback, subtracting instead of adding? If you have ever hooked up feedback the wrong way, you would know it! Positive feedback creates oscillation as a portion of the output is added to the input, and this can eventually run away and destroy stuff. By adding the positive side of our amplifier output to the cathode of the tube, an increase in this voltage would make the cathode less negative with respect to the grid than it otherwise would be, acting effectively as a subtraction to the voltage that is applied to the grid.[7]

You will notice one more component added to this feedback network: a 68pF capacitor that bypasses the 11.5k resistor. This is common in a feedback configuration to control very high frequencies, related to properties of the output transformer where there can be a peaking or resonance above the audio band. The transformer is well-behaved at normal frequencies, but some parasitic elements (leakage inductance, capacitance, etc.) can exist at high frequencies in the tens of kHz and this would result in phase shift. We need this negative feedback to be, well, **negative**! And if there is a phase shift too large at these high frequencies, then we'll get instability as it turns into positive feedback. So by adding a very small value capacitor here, we compensate for the imperfections and allow higher frequencies to slip right by the resistor and roll off before things get too ugly way up there. Even though it is in ultrasonic frequencies you might think we don't hear, this bad behavior can still have consequences that impacts the performance of the tube in the audible range, adds stress and so on. Best to have the insurance policy!

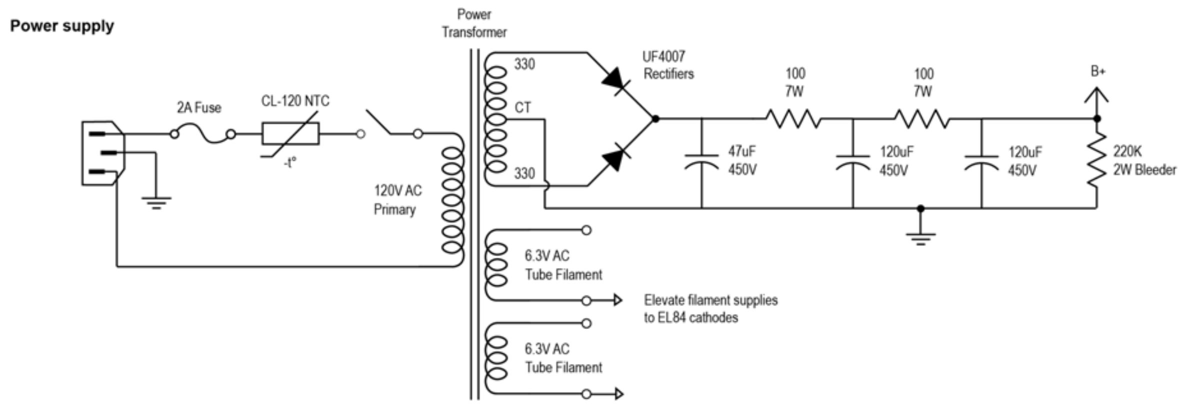
I know I've only barely described this, and poorly! Although we could look at other types of feedback or use some equations, I think I'm going to stop here and not try to go further than this general overview. But I hope you get the general concept: using the output as an influence on the input to achieve a controlled gain level. If you haven't gotten it just yet but want to know more, this could be a topic you choose to study more using additional resources.

7 There are actually multiple inversions of phase of the signal. If you look back at the load line, a positive change in the grid causes a negative change in the anode voltage. This means the output signal taken from the anode of the tube is inverted relative to the input. But we have two stages of amplification, each of which is inverting the phase before finally reaching the output of the amplifier.

POWER SUPPLY, RECTIFICATION AND FILTERING

We have covered all the main parts of the amplification circuit. Now let's back up and understand the power supply. Why are we doing this last? As you saw with the load lines and tube explanations, we have choices that we can make about the supply voltage and operating point that determine the voltage and current demands of the amplifier. Having this information helps understand the choices in the design of the power supply.

Looking at the schematic, let's start on the primary side of the power transformer. Mains voltage comes in from your electrical outlet and we have a few components the hot side passes through in series.



First, a 2 amp fuse is for protection in case something in the amplifier is drawing more current than it should, such as if you had a short somewhere. We would rather the fuse blow than your components to be destroyed or things catch on fire, but even this fuse is not a guarantee that improper wiring or a short wouldn't do damage to components, it just will break the circuit to prevent continuous high current that could be dangerous. This is a slow-blow type fuse, so it will allow a brief heavy draw of current, which is typical from the inrush that can happen when the amplifier is first turned on and capacitors are charged up.

Next, we have a negative temperature coefficient (NTC) thermistor. This is another protection device that has some amount of resistance when cold, and as it heats up over a few seconds, the resistance drops down closer to zero. This is to help reduce some of the inrush current of the amplifier when first turned on as the core of the transformer and capacitors are charged. It's optional and the circuit would work without it, but there is a chance that the initial current is high enough to put a strain on some of the components. (In some circuits with large transformers, the inrush can even cause a circuit breaker in your house to be thrown.)

Next, we have a simple switch to turn on or off the amplifier. When on, the hot side of the mains voltage connects to one lead of the primary side of the power transformer and the other lead of the primary closes the loop back to the neutral side of the mains. (I am not showing on this schematic the power indicator lamp that is also on that same primary side switched in parallel with the power transformer.)

Remember that a transformer uses coil windings at a certain ratio to transfer power from the primary side to the secondary side which will convert the voltage and current to different levels. In our case, we are using a transformer that will take 120V AC (or 240V) household mains voltage on the primary, and we will get 660V AC on the secondary. Yeah, it won't end up being that high in a minute, but for real you still don't want to touch this kind of voltage!

This transformer has additional windings on the secondary side that each will provide 6.3V as the power for the tube filaments (heaters) of the driver or power tubes. It is common to not show the heater wiring in the schematic because it is relatively straightforward and is otherwise isolated from the rest of the amplification circuit. The one thing I will note is that we are going to elevate this 6.3V AC heater voltage by referencing it to a higher DC voltage, around 13V. Remember that AC cycles at 60 Hz, so if we connect one leg of this heater voltage to a DC voltage reference at 13V, it's like that AC is now "riding" on the 13V, fluctuating up and down with the AC changes with respect to 13V. It has to be referenced to something (you should never have a floating heater voltage without any reference to your main circuit) so instead of referencing it to 0V ground, we will choose a higher voltage reference. We will do this for two reasons.

First, in the initial gain stage, the cathode is very close to ground potential, at around 0.8V. Elevating the heater from 0V to about +13V makes the heater sit above the cathode instead of right next to it at nearly the same voltage. That reduces tiny heater-to-cathode leakage/capacitive coupling, so less 60Hz "heater hum" sneaks into the audio path. It's a simple way to lower the noise floor in a high-gain stage.

Second, in the phase inverter stage, remember the cathode sits very high above ground more than 100V, so a heater at 0V puts a large heater-to-cathode voltage stress on the tube which has insulation between the heater and cathode, but there are limits to it. A tube datasheet will usually tell you a maximum voltage allowed between those. Elevating the heater reduces that stress, improving reliability and keeping the phase inverter cleaner and more stable. It's a simple way to protect the tube and lower noise in a sensitive, high-impedance stage.

Out of convenience, we are going to reference the AC heater supplies by connecting one side of it to the cathode of the output stage, which happens to be around 13V DC. That works pretty well for our situation. We could in theory have chosen a higher voltage elevation, added more circuitry to make a voltage divider from the high voltage supply, but it's not really necessary in this case. Elevating to an output stage cathode is a pretty common trick. Tell your buddies about it.

Now, back to the high voltage stuff. You will notice that on this transformer the high voltage secondary has a center tap to use as a 0V potential. This is convenient so we can reference each 330V end of the secondary relative to this point. This becomes our ground reference throughout the amplifier circuit. In some cases, the transformer will not have a center tap and you would use a slightly different type of rectifier to create a 0V reference.

The transformer secondary is still AC, now at a higher voltage, but our amplifier will require a high voltage DC supply. In fact, we need this DC voltage to be as pure and steady as possible. The fundamental activity of the amplifier is to modulate this DC supply voltage based on the input signal. If the supply is not clean, we will not get a high-fidelity output and may even hear an audible hum or buzz. You'll see why soon.

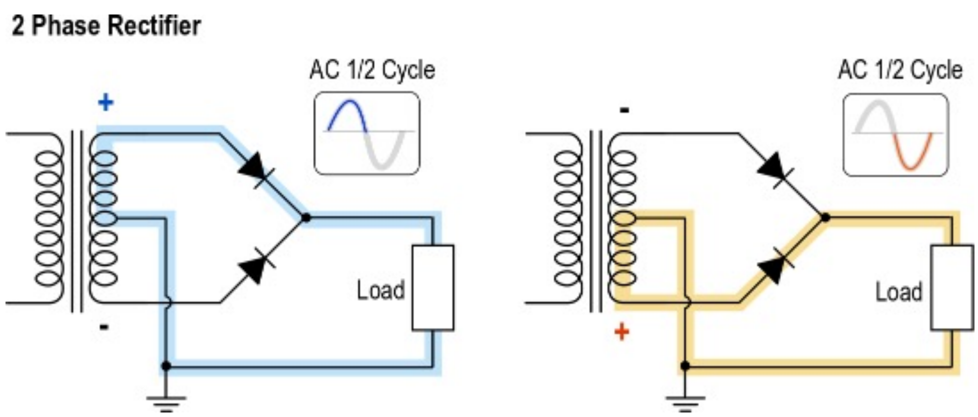
To convert AC to DC, we need a rectifier—something that will allow current to flow only in one direction, so we have only positive current. Amplifiers historically used a tube rectifier. From our earlier topic, you understand how a vacuum tube works—with a cathode emitting electrons and an anode pulling them in, allowing current. Importantly, this current can pass only in one direction. So using a rectifier "diode" tube (no control grid), you can rectify from AC to DC. Rectifier tubes drop a significant amount of voltage, require current to heat the filament, and add cost and physical space required in the amplifier. We have a better and cheaper solution now: silicon rectifier diodes. Once again, there may be debates about whether a tube rectifier is better: does it have a desirable "sag" under current loads impacting the sound and do guitar players prefer it in their amps, is it better to bring up the DC voltage slowly due to the heater warm-up time, what about switching noise of a diode, etc. I won't try to cover the

Tube vs. diode rectification and B+

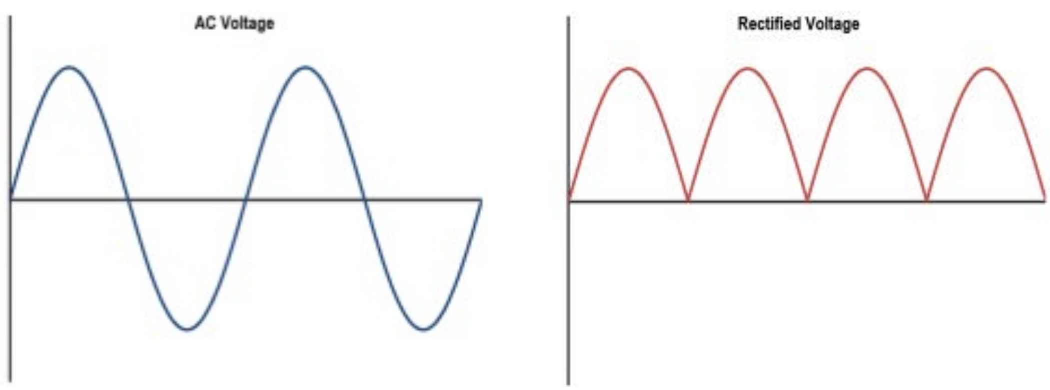
One criticism of silicon diode rectification is that it immediately allows the full B+ voltage to appear on the tube plates before they are warmed up and passing any current. The theory is that this high voltage damages the tubes. A tube rectifier warms up at a similar rate to the driver and output tubes, so the B+ would come up slowly, perhaps extending tube life. I'm not sure there is proof one way or the other, but I have seen experts argue both, and I generally believe that diode rectification and a few seconds of immediate B+ on the plates is not a risk to worry about.

differences, but I will say that I believe silicon diode rectifiers are very good at doing their job and are ideal for this amplifier kit, intended to be simple, high quality, and reasonable in cost. Using silicon diodes to rectify the power supply is very common in tube amplifiers and does not compromise the sound or make this a solid-state or hybrid amplifier in any way.

The AC voltage is cycling at 60Hz (US mains frequency) and each terminal of the high voltage secondary of the transformer is alternating back and forth at this rate, opposite from one another and positive or negative with respect to the 0V center tap. Diodes allow current to flow in one direction only and switch off when current goes the reverse direction. By connecting diodes on each secondary terminal, we create a two-phase rectifier. On the first half of the cycle when voltage swings positive on one terminal (with respect to center tap), that diode switches on and conducts current while the other terminal is now negative and that diode switches off. On the second half of the cycle, the opposite occurs and the other diode will conduct.



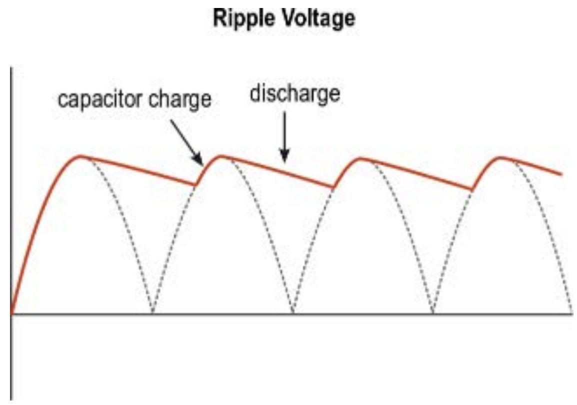
If you visualize the effect of this at the output of the rectifier, the voltage potential with respect to the center tap is always positive—first from current flowing through one diode for the first half-cycle, and then from current flowing through the other diode for the second half-cycle. The rectified voltage now looks like the graph below.



We are making progress, but our intention is to have smooth DC voltage, not large peaks and voids like this. We need to filter this power supply to smooth this out. There are various ways to do this, but the two main components to use are capacitors and chokes (inductors).

By putting a capacitor in parallel with our supply voltage, the rectified voltage will charge it up on the up-cycles and then when the cycle is falling, the capacitor will discharge, supplying current to the amplifier load. You could call this a “reservoir” capacitor because it’s like we have a tank holding a supply of water. While the faucet may be turned on and off continuously to keep it filled, we can tap the barrel from the other side to draw a relatively steady stream out.

A reservoir capacitor will make our DC voltage look like the chart here. Note that we still have some ripple voltage as the capacitor is discharged, but it's certainly better than the peaks we had previously. In our circuit we are using a 47uF capacitor as our initial reservoir capacitor. You could use a lower or higher capacitance. I will not try to describe the calculations of what size ripple voltage you would have for a given power supply, capacitor, and load, but the point right now is that we have more work to do, even after putting in place this capacitor.

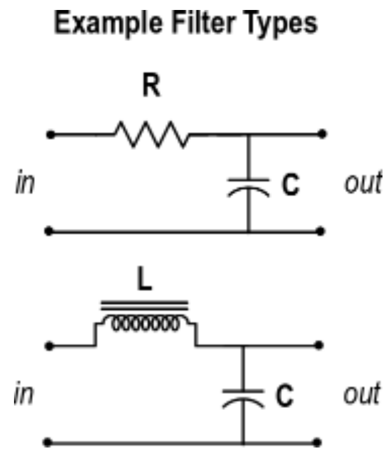


If we used this DC as our B+ supply, this ripple voltage would be modulating our plate voltages by a small amount at a frequency of 120Hz (since this rectified voltage represents two half-cycles of the original 60Hz AC). You would hear this as a sort of buzz or hum in your amplified output.

We can do more to continue to refine this power supply and get the DC supply as clean as possible, with ripple below any audible level. While you might just be tempted to use a bigger reservoir capacitor, there are limits to how much this can reduce ripple, and it has some other downsides I won't go into here.

What we are attempting is to allow DC to pass while we filter out the frequency of this ripple—think of it like AC at 120Hz riding on top of the DC current. The most common techniques are to use low-pass filters using resistors, capacitors, and inductors. The simplest is an RC filter—a resistor and another capacitor to create a type of voltage divider that would attenuate frequencies above a certain point. This type of filter is inexpensive and works quite well. The downside is that it requires some voltage to drop, sort of wasting a bit of the power supply. This also dissipates as heat.

An alternative is to use an inductor, typically called a choke in this application, as an LC filter (inductors denoted with the symbol L). Remember we covered that capacitors block DC but allow AC (a simplistic way of describing it). And inductors are the opposite, reacting against AC current changes while allowing DC to pass. Putting the inductor in series in the filter would have minimal impact on DC, while reacting against the AC change at the ripple frequency. Downsides of chokes are that they can be expensive and physically large when they have enough inductance to do adequate filtering, and low enough DC resistance to not drop much voltage (all wires have some resistance).



In this kit, I was aiming for small physical size and lower relative cost, so I did not use a choke. The EL84 tubes also run on relatively lower B+ voltage compared to some tubes, so dropping voltage is actually not a problem. We use several filtering steps in the power supply: a 47uF initial reservoir capacitor, then two RC filters with 100 ohm resistors and 120uF capacitors.

So looking again at our schematic, we have our rectified power supply going through the CRCRC sequence. The end result is a filtered B+ that should have very little ripple voltage, which means our amplifier should have a clean power supply we can use for an amplified audio signal, and silence when there is no signal.

How much DC voltage will we get as our B+ after this rectification and filtering? First, remember that AC can be measured in RMS volts, sort of like an "equivalent" steady voltage measurement because it's actually changing voltage throughout its cycle. This transformer is designed as 660V center-tapped (usually labeled 330-0-330), meaning 330V RMS on each half-cycle of the secondary that we have rectified to be a positive voltage, which means the peaks you see in the illustration are going a lot higher. A commonly used formula is that peak voltage is the RMS voltage times 1.41, so our 330V RMS is actually a voltage that could possibly reach around 465V at the peaks, and this is what our reservoir capacitor is being charged up with. Now it's not quite that high for a few reasons. Each diode will typically drop a small amount of voltage, and the 330V specification is unloaded, so this transformer under a full operating load will reduce some of its voltage, so it might only be around 380-390V at this capacitor. Then each of those 100 ohm resistors will be dropping some voltage as part of their job in the RC filters. So our final B+ supply turns out to be around 350V.

A few notes on component ratings. First, the diodes used in the kit are UF4007, frequently used rectifiers rated for reverse voltage up to 1000V. The peak to peak voltage from approximately 390V charging capacitor to negative 390V on the reverse cycle of the diode would total 780V, and there can easily be some plus/minus variation in mains voltage or in the transformer, so 1000V gives us some extra room to be safe. Also, the capacitors are rated to 450V. Although our final B+ is just under 350V, at startup with no load and as the capacitors first charge, they can easily reach 450V.

We should also talk about current demand of the circuit. Transformers are rated for a certain amount of current. If we add up the demand of our expected circuit, we would have approximately 65 mA per channel for the power tubes, 4 mA for the phase inverter, and 1ma for the first gain stage. A few mA are given up in a few ways, so maybe this is around 150mA total for the high voltage supply. This particular transformer was made specifically for the Dynaco ST-35 circuit, applying to what we need.

There is also a current demand of the heaters. Each EL84 requires about 760mA and the 12DW7 requires 300mA, for a total of 1.82A per channel. This is rated separately on the transformer.

There is one final component on the power supply we haven't touched on, the 220k bleeder resistor. This is for safety reasons. When you shut off the amplifier, this will take a few seconds to dissipate the energy that has been stored in the capacitors so they are at a safe level. If somehow you had no load on the circuit (such as without tubes installed) but charged up the capacitors, you could turn off the amplifier and unplug it and there could still be a very dangerous high voltage charge in the capacitors hours or even days later. A very small amount of wasted current through this bleeder is worth the safety of not having a surprise shock.

There is far more depth to power supply design than I'm able to cover, and other very different techniques that could be used to regulate voltage or current. In this kit, you are seeing one design and hopefully this helped explain the basic principles of transforming the AC voltage, regulating it to DC, and filtering it.

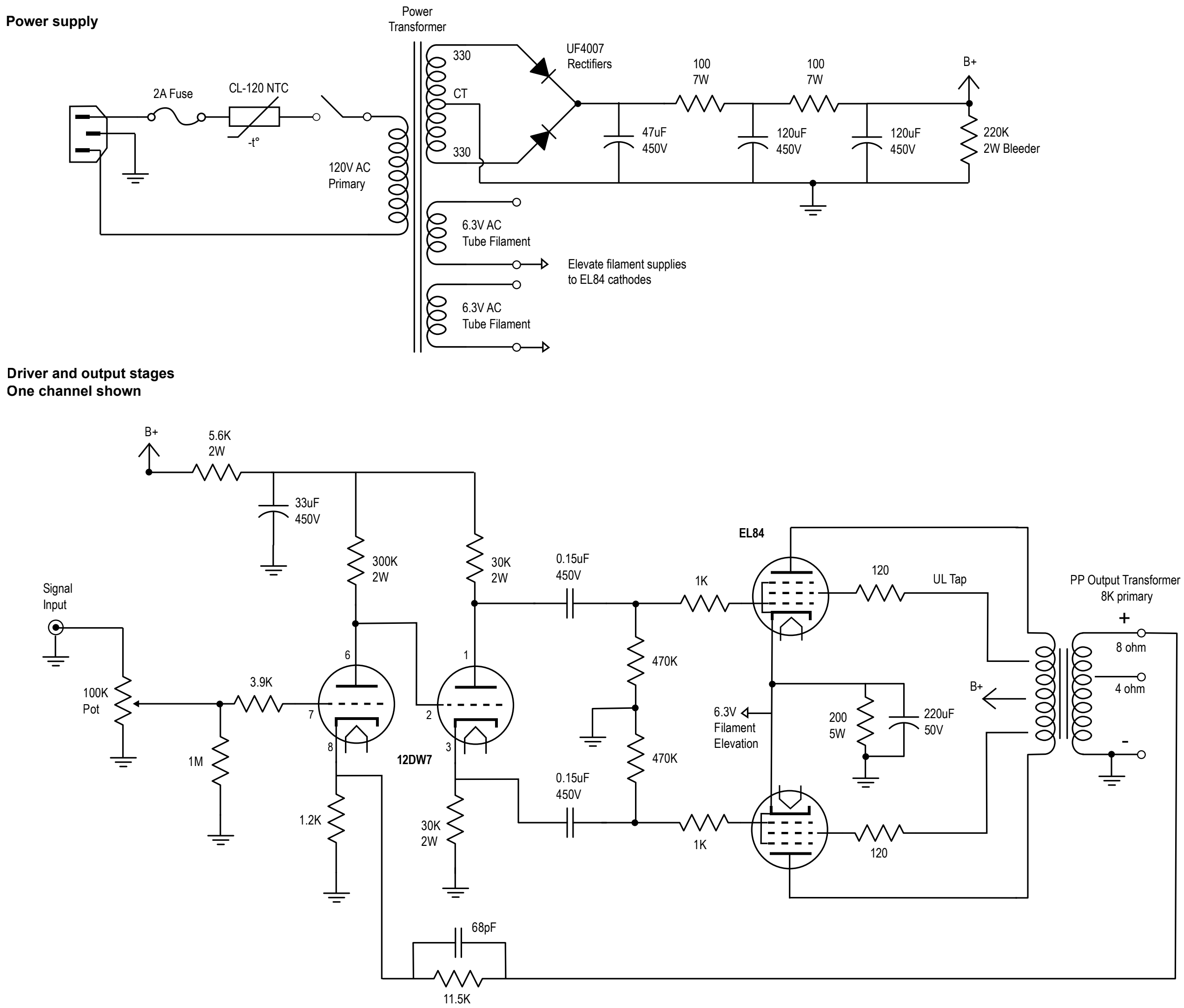
BRINGING IT ALL TOGETHER

Let's look one more time at the schematic, I will point out a few final elements, and you should have a complete understanding of this circuit.

You will see the B+ coming into the first tube and there's an additional RC filter with a 5.6k resistor and 33uF capacitor. This provides some further smoothing of the voltage and decoupling so the input and output stages don't impact one another. The driver stage also does not require as high of voltage as the output stage.

There is a 3.9k resistor leading into the grid of the 12DW7 tube, and a 1k resistor on the grids of the EL84 tubes. These are "grid-stopper" resistors and are used to help avoid stability problems. It combines with the internal capacitance of the tube from grid to cathode to act as an RC filter, small enough that it does not impact audio frequencies, but large enough to stop any very high frequency interference that can create instability that you may not be able to hear but could lead to oscillation and dam-

Pacific 63 Push-Pull Amplifier 12DW7 / EL84



age to the tubes. Don't worry too much about these resistors—they might typically range from 1k to 5k or so. Some designs do not even use them, but I think it's a good idea. The same is true with the screen grid, where you see a 120 ohm resistor.

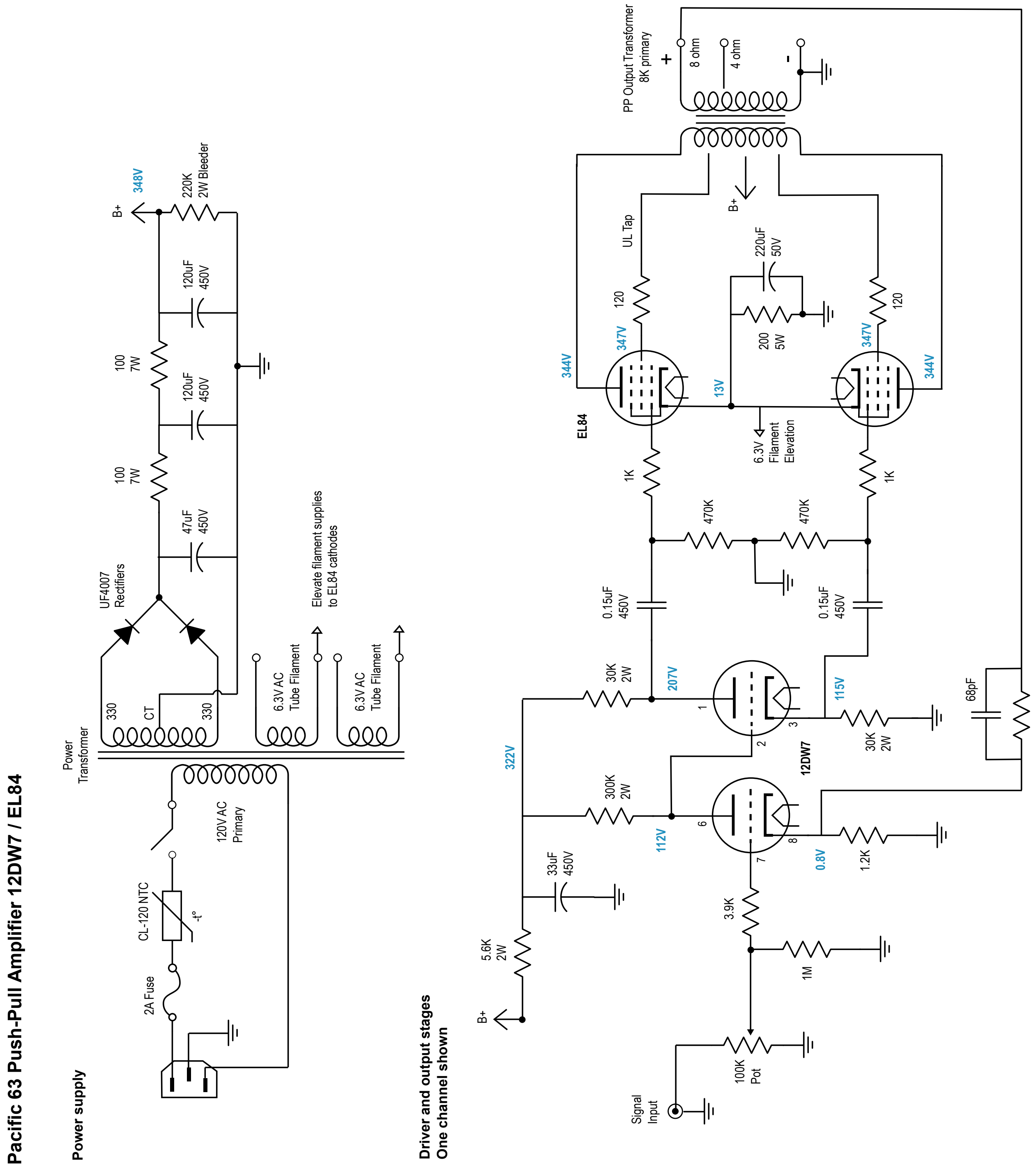
You also see the input signal entering the circuit and going through a 100k potentiometer, which is acting as a voltage divider to attenuate the signal depending on the position of the volume knob. 100k is the amount of resistance this potentiometer uses. When the knob is turned to one end of its range, the signal goes straight to ground. At the other end of the extreme, there is 100k of resistance, allowing nearly the full signal to be applied to the grid. And in between has some variable amount of resistance that can reduce the level of the signal to some degree.

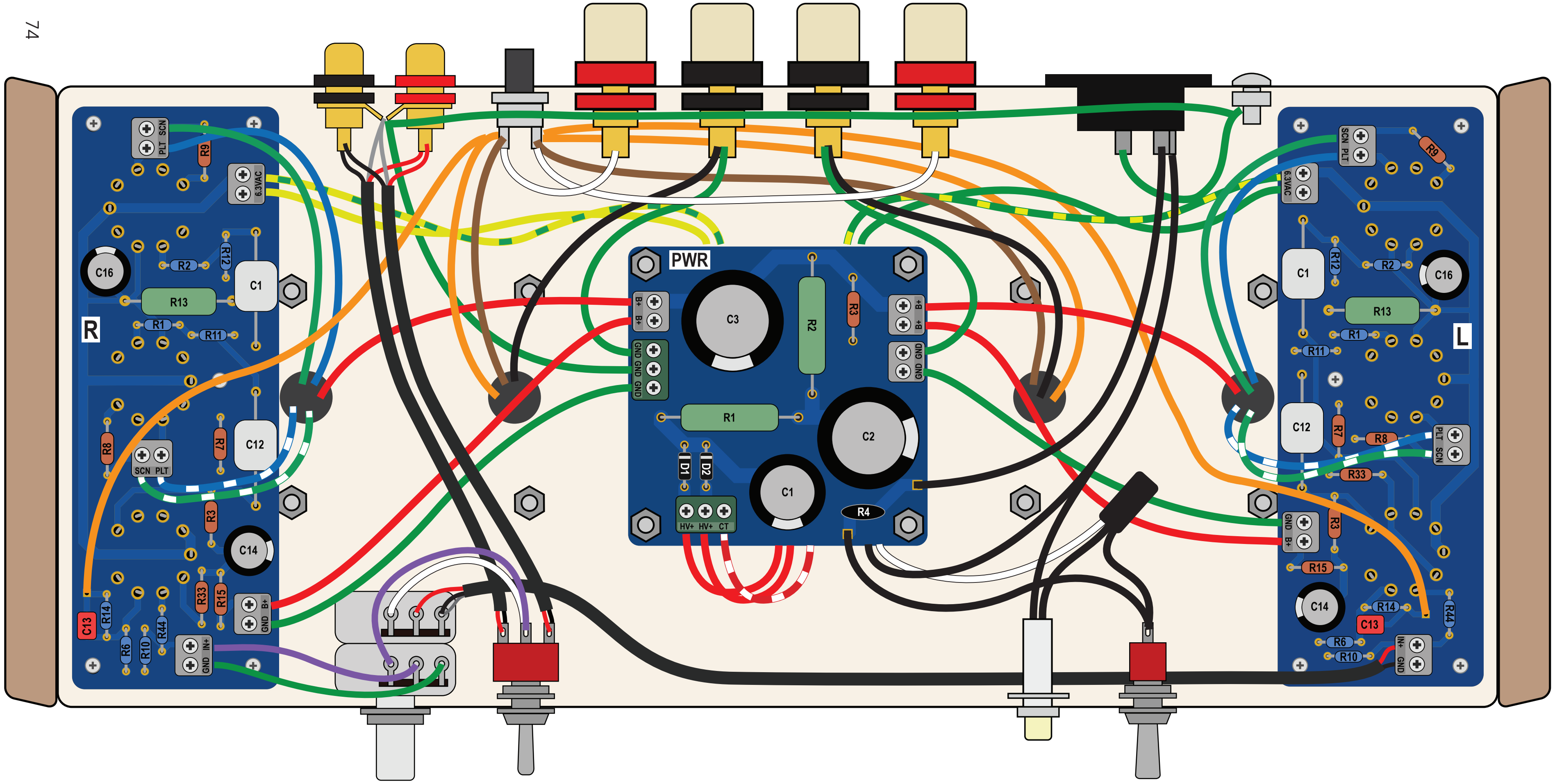
I haven't mentioned the fact that the two output tubes share the same cathode resistor. This is common in some push-pull amp designs. We could give each one their own, but it's convenient to have them share one. It also provides a slight self-stabilizing effect; if one tube starts to draw more current, the total cathode current rises, increasing the voltage for both cathodes. While both tubes would then shift cooler, the hotter tube will shift a bit more for the same change in voltage, giving a mild nudge closer to one another. The tubes still need to be matched pairs so they operate well together; this is not a regulation or forcing of equal currents.

You will notice the output transformer has two different taps—one for 4 ohm speakers and one for 8 ohm speakers. I don't show the switch on the schematic, but you can get the concept that the amp has a slide switch to select which of these is going to the speaker binding posts. Likewise, I don't show the input selector switch on this schematic, where there are two sets of RCA jacks and a toggle switch to select which signal goes into the potentiometer.

That's it! We have walked through the entire circuit and I hope you feel you've had enough overview to understand each part of it. If you need (and want) to, re-read this Part II and think about what's happening in your amplifier. I don't know about you, but I'm the type of person who needs to go over it a few times to really get it. Knowledge is powerful, and it's great fun to learn. I hope you find this useful and perhaps you'll want to keep learning more!

APPENDIX: SCHEMATIC





APPENDIX: LAYOUT DIAGRAM

Left & Right Channels

R1	1k
R2	1k
R3	30k 2W
R6	1.2k
R7	30k 2W
R8	120
R9	120

R10	1M
R11	470k
R12	470k
R13	200 5W
R14	11.5k
R15	5.6k 2W
R33	300k

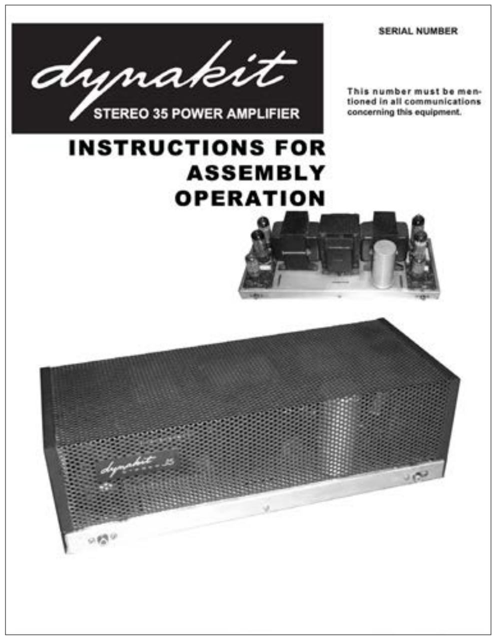
R44	3.9k
C1	0.1uF 600V
C12	0.1uF 600V
C13	68pF 100V
C14	33uF 450V
C16	220uF 50V

Power Supply

R1, R2	100 7W
R3	220k 2W
R4	CL-120
C1	47uF 450V
C2, C3	120uF 450V
D1, D2	UF4007

APPENDIX: 1963, A YEAR TO REMEMBER

My Pacific 63 amplifier was inspired by the Dynaco ST-35 (Stereo 35). Introduced in 1963, the ST-35 was a compact, no-nonsense stereo power amplifier using EL84 output tubes. Dynaco built its reputation on smart engineering and affordability, and the ST-35 had a simple circuit, excellent transformers, and sound quality that punched way above its modest specs. It became a favorite for music lovers who wanted real hi-fi without spending a fortune.



The ST-35 was sold primarily as a kit, which was very much the Dynaco way. Building it yourself was part of the appeal. Factory-assembled versions were also available, though far fewer were sold that way. Either route, the design leaned heavily on elegant simplicity: cathode-biased output stage, a shared cathode resistor for the EL84s, and a straightforward driver circuit that prioritized stability and musicality over complexity.

The ST-35 used a few tricks that let it get up around 17 watts per channel and also ensure sufficient front-end gain so that input sensitivity wasn't too demanding. In particular the use of excellent output transformers (the well-known Z-565 transformers) with ultralinear operation, global negative feedback, and the careful use of positive feedback in one portion of the circuit to augment the stage gain. I chose not to implement a few elements of the original circuit, preferring a bit more simplicity and not feeling like it was necessary in a modern interpretation to worry too much about juicing out the watts or sensitivity levels.

In terms of physical design, the ST-35, like many kit amplifiers of that era, had a rather industrial look to it, with a low-profile metal chassis and perforated steel cage, inputs on the front, and older types of speaker screw connections and power cord on the rear. I've brought my design to a more modern style, allowing multiple inputs, improved connectors, a volume control and other features I think improve the experience and aesthetics.

Decades later, the ST-35 is still cloned, rebuilt, and used as inspiration. It's a reminder that sometimes the "small" amps are the ones that leave the biggest impression.

At the same time, 1963 was when the Beach Boys became a national phenomenon, with Surfin' USA and other releases. Tight harmonies, bright melodies, endless summer! Their records were clean, polished and energetic on the hi-fi systems people were proudly setting up in their living rooms. Stereo and fidelity mattered! So I needed a good musical and visual inspiration, leading to the styling I chose for this amp with a California surf theme, wood side panels and other details.

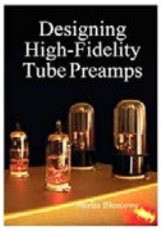
I find having some inspiration always makes a creation better and more fun!

I hope you enjoy it!

Perry Board
Analog Ethos

APPENDIX: ADDITIONAL RESOURCES

Here are a few favorite books and resources that I've found useful or inspiring in case you want ideas of further ways to learn more about making your own tube amplifiers.

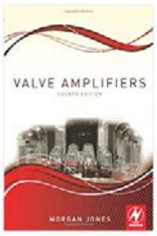


Designing High-Fidelity Tube Preamps

Merlin Blencowe

Also, information on his website: www.valvewizard.co.uk

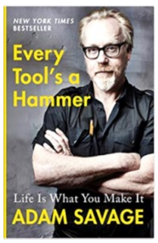
Excellent information and well-written for understanding.



Valve Amplifiers

Morgan Jones

A great book, quite deep on some theory.



Every Tool's a Hammer

Adam Savage

Also, his website www.tested.com and YouTube channel

Great inspiration on the general topic of being a maker!

RCA Radiotron Designer's Handbook

A classic text that has extensive information. It is available online in PDF form if you search for it, or you can find used printed copies.

The Valve Museum website

www.r-type.org

A great collection of information and articles

Online communities:

www.audiokarma.org

www.diyaudio.com

Facebook group: *Tube Amp Builders (DIY)*

OPERATING AND MAINTENANCE NOTES

- **Amplifier location:** see safety precautions above. Locate the amplifier on a flat, secure surface that can handle its weight safely. Ensure plenty of room above the tubes and do not enclose the amplifier without ventilation.
- **Tubes:** Insert the tubes fully into the tube sockets. The pins will fit only in one orientation and it takes some amount of pressure to seat them down into the sockets securely. The smaller 12DW7 tubes go in the front socket and the taller EL84 tubes go in the rear two sockets on each side of the amp.

- **Speakers:** Always ensure speakers are connected to the amplifier when in operation. Running a tube amplifier with no speaker load attached can damage components. A switch on the back of the amplifier lets you choose the nominal impedance of your speakers--either 4 ohms or 8 ohms. The most common impedance is 8 ohms.
- **Input:** The amplifier uses any standard line-level input source. If you wish to use a turntable, you will need a separate pre-amplifier with a phono stage. There are two sets of inputs on the Pacific 63 amp. The front selector switche next to the volume control will switch between them.
- When using a preamplifier with gain, ensure you do not overdrive the amplifier, leading to distortion that could damage your speakers.
- **Warm-up time.** The tubes will warm up after about 10-20 seconds before sound will play. It can take several minutes for tubes to be fully warmed up and performing optimally. Your amplifier includes new tubes; some burn-in time may still be in progress for several hours. All tubes change over time and eventually need to be replaced or can burn out after extensive use.
- The power transformer and top surface of the amplifier can get hot after an hour or two. This is normal.

Specifications

Driver tube: 12DW7

Power tubes: EL84 - you should used matched pairs for the left and right sides (or a matched quad)

Speaker impedance: 4 or 8 ohms nominal (amplifier has a switch to choose the output)

Maximum output power: 15W per channel

Input Sensitivity for maximum output power: 1.5V

Frequency response: 20Hz – 20kHz (<1dB)

Fuse: 2A slow-blow (extra fuse included, but the fuse should generally not need replacement)

Take care and enjoy, and thanks for supporting Analog Ethos, an independent business proudly making hand-crafted amplifiers in the USA.

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