

Designing Vacuum Tube Amplifiers and Related Topics

Written for and about tool-capable guitarists who can use a pocket calculator



*With gratitude to my Uncle, **Charles S. Frazier**, and the friend of my parents, **Robert L. Tabor** (MSEE, deceased), for stimulating two lifelong interests.*

Charles R. Couch, Eureka, California, December 2009

**This work is NOT a commercial enterprise - Except for the cost to
reproduce a single copy for a single individual, the use of this work for
profit is explicitly denied. This writing is NOT in the public domain, no
permission is given - written or otherwise - for any publication for profit.**

<u>Foreword</u>	<u>6</u>
<u>1.0 Vacuum Tubes, Amplification, History and Observations</u>	<u>7</u>
1.1 Early History and Contributors	7
1.2 De Forest's Contributions - Birth of Modern Electronics	8
1.3 An Incremental Improvement	9
1.4 Power Amplification	9
1.5 The Peak Years	10
1.6 The End of an Era?	11
1.7 Current State of the Art.....	11
1.8 Are Good Tubes Still Made?	12
1.9 Reminiscences.....	12
1.10 Influence of the Psychedelic Sixties	13
1.11 Amplifier Modifications	14
1.12 Manufacturers Responded	15
1.13 Getting Along With Today's Vacuum Tubes.....	16
1.14 Further Thoughts, Modifications and Repairs	18
<u>2.0 Biasing the Vacuum Tube</u>	<u>19</u>
2.1 What Is Bias?	19
2.2 How to Bias A Triode	20
2.3 Amplifier Bias Adjustment.....	24
2.4 Tube Amplifier Designs Rarely Change.....	24
2.5 Matched Sets of Tubes.....	25
<u>3.0 How Vacuum Tubes "Amplify"</u>	<u>26</u>
3.1 Definition of Voltage Gain	28
3.2 Other Forms of Gain	29
3.3 Vacuum Tube Configurations.....	29
3.4 Common Cathode Configuration for Voltage Gain.....	30
<u>4.0 Amplifier Functions</u>	<u>31</u>
4.1 Important Performance Parameters.....	33
4.1.1 Noise	33
4.1.2 Noise figure.....	34
4.1.3 Gain	35
4.1.4 Output Power	36
4.1.5 Distortion	37
4.2 Individual Circuit Functions	38
4.2.1 Preamplifier.....	38
4.2.2 Equalization	39
4.2.3 Post Amplification	39
4.2.4 Phase Shifter/Power Divider.....	40
4.2.5 Power amplifier.....	41
4.2.6 Output Transformer	42

4.2.7 Load	43
5.0 Amplifier Specifications	44
5.1 Guitar Pickup Impedance and Signal Level.....	45
5.2 Amplifier Input Impedance.....	45
5.3 Output Impedance, Speaker Impedance, Output Power, Frequency Response	47
5.4 Noise Figure Considerations.....	51
5.4.1 Pragmatic Definitions of Noise Figure	51
6.0 Performance Estimates Using a Spreadsheet.....	53
7.0 Precautionary Information	56
8.0 Designing the Output Power Stage	56
8.1 The "Push-Pull" Output Stage or "Balanced" Amplifier	57
8.1.1 Why Are Balanced Amplifiers More Efficient?	58
8.2 Tubes, Transformers, Power Supply Voltage	59
8.3 Selecting Output Tubes.....	60
8.4 Estimating Plate Voltage and the Tube Data Sheet	63
8.5 Estimating Quiescent Bias Current.....	66
8.6 Quiescent Power Dissipation	66
8.7 Determining Grid Bias Voltage	67
8.8 Screen Grid Current	68
8.9 Estimating Output Impedance.....	69
8.10 Estimating Transformer Requirements	69
8.11 Re-checking Output Power	70
8.12 Output Stage Characteristics, Preliminary	71
8.13 More On Transformers	72
8.14 Input Drive Level, Estimate for Voltage Gain.....	74
8.15 Biassing Power Tubes, Screen Bias	75
8.16 Biassing the Power Tubes, Control Grid and Cathode.....	77
8.17 Biased Grid Configuration	77
8.18 Biased Cathode Configuration.....	79
8.19 Purpose and Selection of Cathode Capacitors	80
8.20 Effect of Cathode Bias Resistors On Plate Voltage.....	82
8.21 Selection of Grid Resistors	84
8.22 Blocking Capacitors.....	86
8.23 Graphical Estimates For Gain and Power	87
8.24 Output Stage Characteristics, Final.....	90
8.25 Reliability and Performance Implications of Vacuum Tube Biassing	91
8.26 Simple Forms of Distortion and Possible Improvements	98
8.27 Negative Feedback.....	102
9.0 Designing Triode Phase Splitters.....	107
9.1 Phase Splitter Without Gain.....	108
9.2 Dual-Triode Phase Splitter With Gain	112
9.3 Practical Example of Phase-Splitter + Power Amplifier	118
10.0 Designing EQ and Post Amplifier.....	120
10.1 EQ And Volume Control	121
10.2 Other Commonly Used Tone Control Circuits	123
10.3 Post Ampifier	124

11.0 Designing the Preamplifier.....	129
11.1 Preamplifier Noise Figure Estimate	132
12.0 Designing the Power Supply.....	134
12.1 Power Supply Functions	135
12.2 Rectifier Configurations.....	136
12.3 Rectifier Circuits.....	137
12.4 Need For the "Bleeder" Resistor.....	139
12.5 Designing a Negative Supply For Grid Bias.....	140
12.6 Hum Reduction and Decoupling.....	145
12.7 Hum Issues Related to Parts Layout	150
12.7.1 The Amplifier Schematic	150
12.7.2 Stray Coupling	151
12.7.3 An Example of a Successful Layout.....	153
12.7.4 Proximity and Orientation of the Transformers.....	154
12.8 Testing the Power Supply Circuit	157
13.0 Completed Guitar Amplifier Circuit.....	158
13.1 Test Strategies.....	160
13.2 Problem Solving.....	164
14.0 Fidelity Considerations	166
14.1 Compression, Single-Tone Distortion	166
14.2 Harmonics in Push-Pull Output Stage	167
14.3 Transient Responses.....	170
14.4 Multi-Tone Distortion	173
14.5 Noise and Hum Resulting From Excess Gain.....	177
14.5.1 Quiescent Noise (No Input Signal)	177
15.0 Design and Parts for Specific Performance Goals	179
15.1 Bright, Clean, Minimally Distorted Sound	179
15.2 Jazz, Standards, Clean Rhythm.....	180
15.3 Distortion	181
15.4 Bass	181
16.0 Versatility of the Beam Power Tube Screen Grid.....	182
16.1 Adjusting Imax By Varying Screen Bias.....	183
16.2 Provision for Variable Screen Grid Bias	184
16.3 Accommodating Existing Transformers	185
16.4 Obtaining Specific Imax Values	188
17.0 Using Beam Power Tubes As Triodes	191
18.0 Hybrid Configurations (Solid State + Vacuum Tubes).....	193
18.1 Solid State Front End Vacuum Tube Power Amplifier	196
18.2 Vacuum Tube Front End, Solid State Power Amplifier	197
18.3 Tremolo Circuits	199
18.4 Reverberation Circuits	203
18.5 Building a Vacuum Tube Preamplifier	204
19.0 Modifying Amplifiers	207
19.1 Making A Practical Amplifier From the Fender "Champ"	208
19.2 Adding "Line Output" or "Preamplifier Output"	211
19.3 Reducing the Power of Large Amplifiers to a Practical Level	214

19.4	Eliminating the Need for Matched Output Tubes	215
19.5	Replacing Tube Rectifier With Solid-State Rectifier	219
19.6	Tube Substitution	223
20.0	<i>Test Equipment, Build or Buy ?</i>	225
20.1	Mandatory Test and Measurement tools	225
20.2	High Voltage Power Supplies	227
20.2.1	Regulated Adjustable Plate Supply	228
20.2.2	Unregulated Adjustable Plate Supply	231
20.3	Signal Generator	232
20.4	High Power Loads (Speaker Simulators)	234
20.5	Pulse Testing	238
20.6	Virtual Signal Generator	242
20.7	Virtual Spectrum Analyzer	244
20.8	Virtual Oscilloscope	246
20.9	Universal "Breadboard"	246
21.0	<i>Amplifier Cabinets and Chassis</i>	247
21.1	The Enclosure or Cabinet	247
21.2	Making Finger Joints	248
21.3	The Amplifier Chassis	252
22.0	<i>Using Spreadsheets, from Concept to Design</i>	254
22.1	The Starting Point	255
22.2	Power Supply Comments	256
22.2.1	Using the Power Supply Spreadsheet	257
22.3	The Output Stage	260
22.4	The Overall Amplifier Block Diagram	265
22.5	Decoupling Power Supply "Hum"	268
22.6	What's Left?	270
22.6.1	Revisiting the Phase-Splitter with Voltage Gain	271
22.6.2	Increasing Preamplifier and Post-amplifier Gain	272
22.6.3	Preamplifier/Post-amplifier Gain Distribution	274
22.6.4	Deciding on Gain Distribution	275
22.6.5	Putting It All Together	276
23.0	<i>Appendix: Abbreviations and Relationships</i>	277
24.0	<i>Appendix: Setting Up A Guitar</i>	279
24.1	Measuring String Clearance/Relief	279
24.2	Guitar Adjustments	283
24.2.1	Nut Adjustment	284
24.2.2	Baseline Configuration	285
24.2.3	Bridge Adjustment	286
24.2.4	Truss Rod Adjustment	287
24.2.5	Intonation Adjustment	289
25.0	<i>Appendix: Comparing Loudspeakers Using SPL Data</i>	291
26.0	<i>Appendix: Inexpensive H-V Power Supply</i>	298
27.0	<i>Appendix: Audio Impedance, A Small-Signal Overview</i>	303
27.1	Attenuation From Impedance Mismatch	304
27.2	Frequency Limitations Resulting From Impedance Mismatch	306

28.0 Appendix: Guitar Trivia	309
28.1 Binding on Guitar Bodies	309
28.2 Neck Binding	315
28.3 Headstock Binding.....	317

Foreword

This is not a *considered* work - it's an impromptu idea, an impulse. My writing skills are minimal and I am not an expert at vacuum tube amplifier design. Formal instruction wasn't offered on this subject when I commenced my engineering education.

I've read about vacuum tubes, both as a guitarist using vacuum tube amplifiers for many years and as a teen constructing and using amateur radio gear (*WV6IJE*). My professional background is solid-state circuit and system design. I'm a retired engineer and have designed many different types of hardware (although my specialty was high frequency/microwave circuits and systems).

I also have many years of interest and experience in woodworking and metalworking. I've accumulated appropriate machinery (plus tooling) to fabricate the mechanical structures required by my various interests, either wood or metal. Similarly, I've acquired adequate electronic test equipment necessary to characterize and test my designs or evaluate modifications and repairs to my musical equipment.

The original motivation for this project was to document how engineers of an earlier time executed the design process - insofar as I could deduce it. I did apply *some* modern techniques, in the form of constructing a few spreadsheets, useful for repetitive parts of the design process and minimizing computational errors. I also used computer circuit analysis simulations to evaluate the performance of circuits discussed here - mainly as a check against the approximations and assumptions that represent a simplified design approach to these niche products.

An audience that I hope to address is the jazz guitarist who is familiar with simple vacuum tube circuits, can read a circuit schematic, has an understanding of how vacuum tubes function, owns some *basic* equipment for measurement of voltages and currents and a *healthy respect for the high voltages always present in these devices*.

Those that, for example, find the technical terms unfamiliar, the units of measure unknown and the concepts forbidding should not read this for any reason other than "entertainment" or as an incentive for seeking more knowledge related to this subject. Don't assume that you can safely, successfully design a vacuum tube amplifier based on my stated opinions without first having some basic knowledge of the topic.

The desired audience is an esoteric group, to be sure, but in my opinion a growing one characterized by curiosity and a desire for experimentation. I hope to encourage curiosity - always a good thing since it results in knowledge.

The thoughts in this book are offered as free speech, without compensation. I wanted to stimulate interest in past technology - perhaps leading to interests in current and developing technology. From early years I thought that being a design engineer - either mechanical or electrical - might be an excellent and rewarding career. I did both and enjoyed almost every minute of my careers in both fields.

I apologize for the primitive graphics and general amateurish presentation of this work - I didn't put much time into making the book attractive. Additionally, many of the chapters were first published as stand-alone topics in internet jazz forums and these may not sequence as well as if the book were written in one entirety.

1.0 Vacuum Tubes, Amplification, History and Observations

1.1 Early History and Contributors

It is reported that vacuum tubes were produced as early as the 1850's, when vacuum pumps capable of evacuating most of the air from a sealable vessel became available. The purpose of these early tubes was probably pure science: studying the behavior of heat flow in a near vacuum, for example. It is doubtful that any experimenters working with early tubes conceived of what the devices might eventually be capable.

We first hear of commercial applications when Edison, after many trials, introduced the first successful light bulb in 1879. (A light bulb is a vacuum tube with only one element: the filament.) In 1883, for reasons that are unclear, an engineer working for Edison, William Hammer decided to introduce another element within the glass envelope, which he called a "plate".

(Interestingly, Edison's successful incandescent light bulb was preceded some twenty years earlier, by early versions of what we now call fluorescent lighting. Both French and German inventors produced primitive versions of gas-filled tubes that glowed when electrical current was passed through them.)

Hammer experimented with his configuration, which we would call a "diode" today. He found that, when a high potential positive D.C. voltage was applied to the plate and the negative return voltage applied to the filament, current would flow between filament and plate.

This phenomenon was called the "Edison Effect" and was not understood at the

time. Edison could see no commercial value in the diode tube but Hammer's experimental results were recorded and, as was Edison's practice, the device was patented in the U.S.

In 1873, Professor Frederick Guthrie, experimenting with "red-hot" iron balls in England had noted a similar effect but he, like Edison, saw nothing of importance in the process. A few years later, also in England, Sir John Fleming duplicated Edison's (actually Hammer's) experiments. Fleming, however, DID see commercial value in the diode tube and patented it in the U.K. in 1905.

In 1901, a paper published by Owen Richardson explained the Edison Effect. This paper describes mathematically what is now known as thermionic emission and is called "Richardson's Law". In non-mathematical terms, plate current flow is the result of the hot filament "boiling off" electrons, which are negatively charged particles. The negatively charged electrons are attracted to the positively charged "plate" electrode and thus current flow is produced.

1.2 De Forest's Contributions - Birth of Modern Electronics

In 1906, Lee De Forest (Ph.D., physics, Yale) was working with Edison's experiment and introduced another element to the diode tube. Perhaps realizing the implications of Richardson's Law, De Forest placed a serpentine bent wire between filament and plate, calling this added element a "grid". De Forest repeated Edison's (Hammer's) arrangement of applying a high, positive D.C. voltage to the plate and the negative return voltage to the filament to establish current flow.

De Forest's modification produced a significant new property of the vacuum tube configuration that came to be called a "triode" because it had three (tri) connections (electrodes). Instead of a fixed current flowing between filament and plate, as Edison had observed, De Forest could change the amount of current flow by varying the voltage of the grid.

Modern electronics was born when De Forest noted that his triode was capable of amplification by means of a characteristic that we now call "transconductance" which means that the triode's conductance (and plate current flow) could be changed by varying the input (grid) voltage. De Forest immediately saw the commercial value of his invention and he patented it in 1907, calling the device the "Audion".

As a consequence of De Forest's invention, long-distance communication was born. The "Audion" became the heart of radio and telephone communications systems of the time and germinated the huge telecommunications industries of today.

It's interesting to note that De Forest didn't actually understand how his "Audion" functioned, despite his extensive education in physics. This was evidenced by the original patent application. It was left to others to determine functional descriptions and operational parameters of the vacuum tube (and the many other applications exclusive of simple amplification).

Edwin Armstrong is acknowledged by most researchers to have been the major experimenter, designer, theoretician and circuit developer of early vacuum tube designs. Legal battles between De Forest, Armstrong, David Sarnoff (President of Radio Corporation of America, "RCA") and others continued even to the time that the vacuum tube was headed for oblivion. Armstrong was long deceased by the time the courts decided in his favor.

1.3 An Incremental Improvement

As the vacuum tube was refined, it became apparent that using the filament as an operating element wasn't a good idea. Filaments require LOTS of current, in order to heat them to operating temperature. The batteries of the day, completely satisfactory for the low currents required by grid and plate, weren't all that suitable for powering filaments unless they were large (and heavy).

The solution was to heat the filaments from "line" voltage, or ordinary 60 Hz A.C. (alternating current). Since the filament was also a circuit element, the use of line voltage introduced "A.C. hum", the line voltage alternating current modulated the direct (non-alternating) current flowing from cathode to plate. In order to minimize the hum, the separate cathode was devised, around 1929.

The separate cathode consisted of a thin sleeve of metal slipped over, and heated by, the filament. Connecting the heated cathode to the negative D.C. return of the positive plate voltage supply re-established current flow. The annoying A.C. hum produced by the filament was greatly diminished because the new cathode had no electrical connection to the filament.

1.4 Power Amplification

Early triodes had many disadvantages, especially efficiency. Although they consumed large amounts of power, these devices could not amplify signals to significantly large power levels. Behavior was mostly linear until the plate current started approaching the limits of either saturation or cut-off. Once these limits were reached, no more linear power could be produced.

Other difficulties in producing linear amplifiers were found to be attributable to the geometry of early vacuum tubes. There was a great deal of internal (and unintentional) capacitance, which caused the tubes to oscillate when amplifier gain was high.

And there was heat – the vacuum tube was similar to a steam engine in that the filament (furnace) had to heat the cathode (boiler) to produce electron flow (steam) and perform work. In order to obtain higher power levels, higher currents were required which implied higher filament temperatures and shorter tube life.

There were other problems such as transformer technology, limited primarily because materials engineering (in the form of appropriate core materials) hadn't yet developed. There had been no previous need for these materials.

To improve performance, in the decade following 1930, experimenters added two new elements within the vacuum tube, naming them "screen grid" and "suppressor grid". The addition of these new elements and the concept of "beam forming" introduced the first vacuum tubes capable of producing more than a watt or so of output power.

Virtually all output tubes used in audio power amplifiers have conformed to this configuration, called "pentodes" from "penta" and "electrode" (five + connections). The ubiquitous 6V6 – the most popular American audio output tube - is in this family.

1.5 The Peak Years

During the years of maximum vacuum tube production, from the late 1920s until the early 1960s (peaking during World War II), the devices were well characterized and manufactured under carefully controlled conditions resulting in superior quality and performance. This equated to predictable, reproducible circuit operation. (Some of the best production vacuum tubes were made in the U.S., the U.K., France and Germany.)

Because of the consistent electrical behavior of vacuum tubes produced under high quality conditions, circuit designs never included provisions for adjusting the bias conditions. It was generally accepted that the inclusion of variable elements to adjust tube bias was evidence of a poor design.

The performance of the vacuum tube expanded in many directions, power levels increased as did operating frequency, and efficiency was incrementally improved. Given the fact that the vacuum tube is at best a primitive, inefficient device, the tasks that it was called to perform were accomplished quite creditably.

In the early 1950s, Bell Laboratories produced the first transistor, solid state electronics was born and the vacuum tube was destined for obsolescence.

Or was it?

1.6 The End of an Era?

Western countries quickly transitioned their electronics industries, embracing solid state technology for efficiency, economics and performance margins over tubes. Countries with less advanced manufacturing base, like the "Cold War" Soviet/soviet-influenced countries, could not easily make this transition, partly for economic reasons and partly for political ones. (Since their factories did not sell at a profit, there was little incentive to invest the huge sums necessary to produce solid state semiconductors.)

Although vacuum tube manufacturing technology was all but lost in the West, it survived in isolated enclaves, usually for the political reasons mentioned above. Designers in the former U.S.S.R. and allied countries used these "antique" devices everywhere – their modern military aircraft and naval vessels reportedly still used vacuum tubes as recently as 1980. But that is no longer true, even former Iron Curtain countries now have little use for vacuum tubes except to export them to the West for musical instrument sound reproduction.

With the rapidly diminishing need for vacuum tubes in military, telecommunications and entertainment equipment, the factories in Russia, China and (formerly) Czechoslovakia reduced their production capability. (Some six or seven different tube types are adequate to support the manufacturers of musical instrument amplifiers.)

As the higher performance tube production lines were shut down and the supporting engineers looked elsewhere for more lucrative, rewarding employment, the quality of the products of these old factories started to decline. It was the requirements of the Soviet military, after all, that had established the higher quality standards that formerly existed.

1.7 Current State of the Art

Today's vacuum tube is a poor substitute for the products made by RCA, Philco, Telefunken, Mullard, Philips, Sylvania, G.E., Tung-Sol and many others in the late nineteen-forties. The eastern European and Chinese products are decidedly inferior in all respects when compared to American and European products. Quality control is lacking, materials are not necessarily optimal and there is little engineering support in the factories. (Factories are no longer subsidized by parent countries because military supply considerations no longer exist.)

The result is that performance (and reliability) suffers. Vital parameters vary, from tube to tube of the same type, to the extent that probably 50% of these tubes would have been rejected from any Western country's vacuum tube production line of the 1950s.

Although there is still appreciable demand for vacuum tubes among musicians,

there obviously exists a price ceiling that defines the quality level of today's vacuum tubes. In other words, if the Eastern manufacturers of modern vacuum tubes made them to the same standards that an American factory employed in the 1950s, the cost of the tubes would drive the price of new guitar amplifiers (modestly-priced products) instantly into the \$1500 - \$2500 range!

1.8 Are Good Tubes Still Made?

That question gets various answers, depending upon who is asked. Personally, I'd say that the quality of current tubes is *adequate* but performance, compared to devices manufactured in the U.S. and in Europe during the decade of 1950, is not comparable. Fortunately, the technical needs of guitar amplifiers are not particularly demanding and the supply and quality apparently is equal to demand.

So how do the brand-name amplifier manufacturers meet their need for quality tubes? Well, sometimes they DON'T – that is, they don't use quality tubes. I'll illustrate that point later. Many manufacturers simply select from the quality distribution curve of the tube manufacturer. As an example, a guitar amplifier manufacturer might send a team to the Shangri-La factory to negotiate a contract for fixed quantities of 12AX7, 6BQ5, 6V6 and 6L6 tubes.

After examining test data from a statistically significant number of tubes from each of the desired types, manufacturing engineers could determine how many tubes of acceptable performance could be expected from each lot. For example, the measured data for 6L6 tubes might indicate that only 300 tubes out of a lot of 1000 would meet the amplifier manufacturer's minimum requirements.

The negotiation would likely produce an agreement whereby the tube manufacturer would screen the 6L6 tubes from production and "cherry-pick" those that met the amplifier manufacturer's standards. For which, of course, an additional cost would be added to the base price of the 6L6 tube.

An interesting question is suggested by the above example: what happens to the other 600 6L6 tubes that weren't good enough for the guitar amplifier manufacturer? *Hint: they weren't thrown away.*

1.9 Reminiscences

Some of us (beyond a certain age) can remember when vacuum tubes could actually be purchased in a grocery store! At the front of major stores was a large self-service tube tester and underneath the tester was a cabinet full of replacement tubes.

Someone was usually standing in front of the tester with a paper bag of television or musical amplifier (hi-fi/stereo) tubes to test. If the meter on the tester read "gassy", "short" or "weak" (the tube tester meters were labeled in consumer-

relative words rather than technical terms), one searched through the cabinet underneath the tester to find the right replacement tube, based on a handy cross-reference chart.

Anyone owning consumer electronic equipment of that era and having to routinely replace one or more vacuum tubes would have been astonished if told that he had to "re-bias" the tube circuit from time to time and ALWAYS whenever a tube was replaced. However, that process is universally accepted when dealing with musical instrument amplifiers at this time.

99.9% of the population wouldn't know what the term "bias" meant. How could the average person be expected to do this and how many might be killed or injured by the high voltages within the chassis? Where would one take that 75 pound television set to have the audio amplifier output tubes "re-biased" each time one was changed?

I exaggerate the bias situation above, still it is obvious that nobody did this. Even high power musical amplifiers of the time didn't require adjustment (even if a person could be found that knew how to perform it). Properly designed equipment did not require changing bias conditions when tubes were replaced. What does this tell us? As previously mentioned, that tubes were more consistently manufactured and circuits more conservatively designed so that normal variations in tube operating parameters didn't significantly affect operation.

So how did we get the idea that guitar amplifiers have to be re-biased when no other forms of vacuum tube electronic equipment required this? Bear with me ...

1.10 Influence of the Psychedelic Sixties

Popular music, beginning in the nineteen-sixties, began to showcase the electric guitar as never before. The economic reasons (low cost of guitars compared to other musical instruments) are apparent but the evolution of "rock" music revealed deficiencies in amplifiers of the day. Audio vacuum tubes and audio transformers, considering pricing strategy, precluded manufacturers from making a linear, high-power vacuum tube amplifier. Some tried to replicate high-power amplifiers as used in audiophile-quality equipment (e.g. "Sunn"). For the most part the need for improving the design of the amplifiers was eliminated by a simple expediency:

Musicians, lacking an amplifier with sufficient distortion-free power, first accepted then EMBRACED the distortion products of the over-driven tube amplifiers.

A technically-oriented band from San Francisco started questioning why equipment couldn't be made better and proved that it could - if one threw enough money at the problems. Several companies (I believe that Alembic was one)

were spun off as a result of continual experimentation with amplifiers, sound reproduction, feedback suppression and personal instrument improvements. An unusual idea: that existing audio equipment could be modified for special needs, started to attract experimenters and tinkerers.

That concept caught on with a vengeance and soon "hot rod" amplifiers began to appear. The vacuum tube amplifier is the simplest design that one could possibly devise for its purpose, there are far fewer parts than a solid state amplifier of comparable performance. Fewer parts and a simpler circuit limit the amount of "improvement" one can make to a standard amplifier.

The limits of the tubes themselves had long ago been reached. The tube designs had already been pushed as far as possible by the most aggressive of all tube customers: military "consumers". So there were only a couple of things for the music store "technician" to tinker with when Ralph Rocquenroll brought in his Fender "Deluxe" and wanted it to have as much volume as a Fender "Bassman".

1.11 Amplifier Modifications

One of the modifications frequently made was to eliminate (or reduce the effect of) feedback circuits in the amplifier. The most common circuit is a resistor (or resistor-capacitor combination) connected between the speaker output and the cathode of the post amplifier stage, the stage immediately following volume control and tone adjustment.

Although no undistorted power increase was possible from this change, the amplifier was less linear, had more gain (not more power) and a different frequency response.) Many people paid a local music store technician for this "improvement".

Diverting for a moment, many people state that a tube amplifier sounds "louder" than a solid-state equivalent amplifier. That's actually a psychoacoustic perspective rather than a scientific one. Varying a periodic waveform from sinusoidal to "square" (as in total distortion), results in more power dissipated by a load - about 40% more (or 1.5 dB). Moderate distortion from a tube amplifier is tolerable by most people, even appreciated.

The same level of distortion produced in a solid-state amplifier is not tolerated well, hence the "loudness" misunderstanding between the two types of amplifiers. (This is usually "explained" by the level of harmonic content and the difference in the way harmonics are generated by different amplifier configurations. We'll get around to that later when we commence the technical discussion and address distortion.)

Let's be clear: undistorted volume levels of vacuum tube amplifiers and solid-state amplifiers are perceived identically by the human ear.

Another similar modification was to eliminate (or reduce the value of) the cathode bias resistors found in output stages. Not necessarily a good choice from a reliability standpoint. The lifetime of the tubes (and maybe the power supply and output transformers) suffered since the plate voltage/current, screen voltage/current increased as well - both plate dissipation and screen grid dissipation were universally exceeded from the manufacturers' specifications.

If the modification was made to a pair of push-pull tubes (virtually all output stages) then it was almost certain that the two tubes were no longer biased to the same conditions. This unbalance might result in distortion and might actually produce LESS output power as a result of the imbalance in the circuit. (Perhaps the enhanced distortion suggested that the amplifier was louder.)

Eliminating the cathode resistors required the inclusion of a negative voltage, applied to the control grid, to establish the correct amount of plate current flow. Including a potentiometer (or two) to adjust this voltage became common practice. This made it possible to "balance" the push-pull output tubes and still eliminate cathode resistors. But the technique is not necessarily the best for bias stability ... the two tubes can become unbalanced over time. (The cathode resistor configuration is always preferred to other methods.)

Most of the modifications people tried on their tube amplifiers did result in one significant change: more distortion. And more distortion was universally perceived as a good thing, at least in popular music of the time.

1.12 Manufacturers Responded

Amplifier manufacturers responded to some of the market pressure and actually did increase the power levels (and headroom?) of some products. For the most part the increase in power, however good it looked on a sales brochure, was not significant. As an example, the iconic Fender Twin in one reincarnation was increased from about 85 watts to around 100 watts, as I recall. That would seem to be a pretty substantial increase, right? Actually, not.

Most authorities of the human ear agree that the *minimum* difference in music power level that the ear can detect is about 1 decibel. The increase in power from 85 watts to 100 watts is 0.71 decibels, so the average listening human ear could not discern whether it was being tormented by an OLD Twin, with all controls dimed, or a NEW Twin, similarly adjusted.

Manufacturers also increased the gain of the preamplifier stages far beyond what was required and introduced the "master" volume control so that distortion could be selectively controlled. Unsaid, however, was that increasing the preamplifier gain also introduced more noise. The vacuum tube amplifiers of the 1970s, in general, produced more "hiss" than their predecessors, all other parameters being equal.

As guitar and amplifier demand and sales increased, amplifier manufacturers were caught up in escalating power level competition. It was demonstrable that a purchaser would always pick the amplifier with slightly higher power (even if design compromises had been made to achieve the higher power and even if the power increase was insignificant).

Some of the more conservative design practices of the past disappeared: feedback and bias networks were altered or eliminated, power supplies were more highly stressed and so were the output tubes. All this was done in an effort to obtain a few more watts of power from older, reliable designs. Bias adjustments started to appear in circuits that employed grounded cathodes, instead of the safe, reliable cathode resistor circuits of the past. The factories started imitating the music store tinkerers. But cost drove everything because cost drove profit.

The profit motive naturally produced some cutting of corners (the CBS association with Fender and the Norlin/Gibson association). All of this, added to diminishing standards of vacuum tube performance, eroded what had formerly been conservative designs typified by amplifiers requiring no expertise to maintain other than the physical coordination required to remove an old tube and replace it with a new one.

1.13 Getting Along With Today's Vacuum Tubes

Given that today's vacuum tubes will probably NEVER be as consistent as those of the past, is it still possible to produce amplifiers that sound as good as the old ones and do not require adjustments ?

Oh yes, but there are compromises, cost being the most apparent. The designs could certainly be made more conservative – sacrificing a little power would make the circuits better behaved, more reliable, tube performance would be less critical and the tubes would last longer. Some manufacturers seem to be doing this, their best selling tube amplifiers are not high-powered, heavy, expensive models but more modestly powered "Blues Junior" types, to use an example with which Fender owners will be familiar.

The lifetime of a vacuum tube is dependant on plate voltage/current, screen grid voltage/current and filament voltage/current. So is the health and well-being of the power supply transformer (providing all power required by the amplifier) and the output transformer (providing the D.C. current of the output tubes and passing the audio output power to the speaker). The circuit components, especially the transformers, are costly and usually selected for a specified current or power rating.

Indiscriminately adjusting bias voltages can lead to failure of tubes and transformers. In other words, allowing persons of questionable qualifications to

make "improvements" and "adjustments" to your tube amplifier can cause financial pain. Even if you never intend to design, build or modify an amplifier, reading this book may allow you to evaluate the music store technician's explanation of a service charge that you may think unreasonable.

For guitarists desiring vacuum tube performance and something approaching solid-state reliability, these features in a tube amplifier may help minimize maintenance:

Power level of 30 watts or less (if the fifty watt level is exceeded, for example, most amplifiers require four output tubes, instead of two. The circuitry and balance requirements are more complex and matched replacement tubes are far more expensive).

6L6 output tubes seem to be universally desirable (all tube manufacturers make them, there is no availability problem if one doesn't mind the expense). Other attractive options are available for those who find that designing and building a personal amplifier is a practical exercise. We'll get into this later when we design an output power amplifier stage.

Output tubes with cathode resistor bias are preferred, rather than a negative voltage control grid bias through an adjustment device. (Cathode bias resistors will accommodate more variation in tube parameters without adjustment). Cathode resistor bias configurations are uncommon in amplifiers of power levels greater than 25 watts.

Negative feedback from the speaker output to the preamp (enhances linearity and flat frequency response, improves distortion caused by tube imbalance).

No master volume control (master volume control implies excessive gain in the preamp stages which can cause additional noise and hum).

The inclusion of solid-state devices in certain areas of vacuum tube amplifiers can be beneficial and we'll discuss this in a future chapter.

Those of us who prefer jazz as our medium of expression generally prefer lightweight amplifiers with moderately large speakers, perhaps because we tend to be older than musicians of other genres. There are many reasons for this, some of which will be explored later.

I mention this because there is no longer a need for high-power, HEAVY amplifiers in a day when most venues include quality sound reinforcement systems. Those who prefer not to manhandle amplifiers heavier than 35 pounds can supplement volume by including a direct output in an amplifier of their

personal design. (Positioning a microphone near the loudspeaker, as most of us have done for years, is also effective.)

1.14 Further Thoughts, Modifications and Repairs

It's best to avoid modification or adjustment to a vacuum tube amplifier unless one has a clear understanding of the various functions that comprise the total amplifier. Voltages present are **LETHAL** and there should be no need for the average user to access areas of the circuit where these voltages are present.

If you are a jazz guitarist and own a vacuum tube amplifier that has been modified, consider having it restored - or doing it yourself, if you're competent - to original configuration. You may be pleasantly surprised at the quality of the sound the original circuit produced.

Amplifiers of around 25 watts generally provide the best package of performance, portability, cost and reliability. Sound reinforcement is simple if the venue requires more volume. Amplifiers in this class frequently accept replacement output tubes without any change in performance or circuit adjustment, although matched tube sets *may* be recommended.

Avoid non-specific recommendations from music store technicians that don't address an audible, measurable shortcoming, the correction of which can be shown to audibly and *measurably* correct that deficiency. Typical might be the suggestion that your amplifier should be "re-capped"- a process that (thankfully) seems to be diminishing since it verges on superstition, in my opinion.

It is pointless to fix something that is not broken. (Claims for the benefits of replacing all of the capacitors in the signal path are wildly inaccurate - remember that if it sounds too good to be true, it probably is NOT true.) Capacitors, at audio frequencies, have no "tone" - get a second opinion, preferably from a knowledgeable person whose "rice bowl" isn't involved in the music industry and who isn't a guitarist.

If you think that your output tubes need replacing, consider having this done locally, rather than ordering tubes and changing them yourself. The argument for having the tubes replaced in the local store is that you can play the amplifier before and after replacement and determine if there is an actual difference in performance.

Perhaps you can make a condition of tube sale that an improvement – as determined by YOU – has to result after tubes have been changed (you need to make this comparison within a minute or so - your ears have a short memory). Other than the twenty-five seconds of time expended by the store technician to replace two tubes, it's a reasonable enough request. I suggest making an

appointment for the tube change, before/after test, rather than leaving the amplifier at the music store.

(Another reason for buying/installing tubes locally is that mail order and internet distributors normally have a no-return-no-refund policy. They need to make this stipulation because of the possibility of fraudulent returns.)

Good vacuum tubes aren't necessarily pricey – nor are the pricey vacuum tubes necessarily the best ones, frequently the converse is true! When buying new vacuum tubes, recall that you're purchasing a product that is inferior in every way to what was installed in vacuum tube amplifiers of the nineteen-fifties and sixties and calibrate your expectations accordingly. After you finish reading this material, you may have altered your opinions about a few things that are "common knowledge", at least I hope so.

2.0 Biasing the Vacuum Tube

2.1 *What Is Bias?*

The term "bias" is not clearly understood, it's not a very precise term in a technical sense. "Bias" could infer something like "prejudice, inclination, pre-disposition, slanted" or something similar. Here's a review of how the word was introduced, in the context of vacuum tubes.

Early experimenters observed that De Forest's "Audion" amplifier behaved in a linear manner only within a certain range of operating conditions. The two extremes of operation were "cut-off" and "saturation". Cut-off describes a point at which current stops (or nearly stops) flowing between cathode and plate and occurs when the grid voltage is most negative (with respect to the cathode). Saturation describes a point of maximum current flow between cathode and plate and occurs when the grid voltage is at the cathode voltage potential or even beyond it.

Engineers quickly found that the most linear operation of the vacuum tube occurred when the plate current was about midway between saturation and cutoff (maximum current flow and minimum current flow). The voltage applied to the grid, which established this linear condition, was labeled "bias". The term suggests that this fixed grid voltage "biased" or "influenced" the vacuum tube toward more linear operation.

(Solid state semiconductors also require a "bias" for optimum operating conditions. Generally speaking, they require more complex circuits than a vacuum tube to achieve linear operation.)

2.2 How to Bias A Triode

Unlike the vast majority of semiconductors, vacuum tube operational parameters are normally determined graphically - we'll explore this in some detail as we design various amplifier stages. During the time of most common usage, it was normal for vacuum tube manufacturers to publish considerable amounts of information regarding audio amplifier design, especially small-signal applications such as preamplifiers. Below is a table of different configurations of small-signal audio amplifiers copied from a "Sylvania" 12AX7A data sheet. Similar tables were provided by all major manufacturers for the convenience of design engineers.

R _p	R _s	R _{g1}	R _{total} (R _p R _s)	Eb = 90			Eb = 180			Eb = 300		
R _k	A _v	E _o p-p	R _k	A _v	E _o p-p	R _k	A _v	E _o p-p	R _k	A _v	E _o p-p	
100,000	100,000	100,000	50,000	1,700	31	14.1	1,000	40	42.4	760	43	84.8
100,000	100,000	100,000	50,000	2,000	38	19.5	1,100	46	56.6	900	50	113.1
240,000	240,000	100,000	120,000	3,500	43	18.4	2,000	54	50.9	2,300	59	67.9
240,000	510,000	100,000	163,200	3,900	49	24.3	4,300	62	53.7	4,800	64	67.9
510,000	510,000	100,000	255,000	7,100	50	20.9	0	53	42.4	3,100	66	110.3
510,000	1,000,000	100,000	337,748	7,800	53	25.7	0	60	53.7	3,600	69	130.1
240,000	240,000	10,000,000	120,000	0	37	25.2	0	61	48.1	0	62	90.5
240,000	510,000	10,000,000	163,200	0	44	15.3	0	69	99.0	0	67	115.9
510,000	510,000	10,000,000	255,000	0	44	14.1	0	66	59.4	0	71	115.9
510,000	1,000,000	10,000,000	337,748	0	49	18.1						

The table provides the necessary information to configure the 12AX7 tube for operation with a predetermined amount of gain, output signal voltage and plate operating voltage. Triodes, being fairly simple devices, were easily characterized and these tables were commonly available.

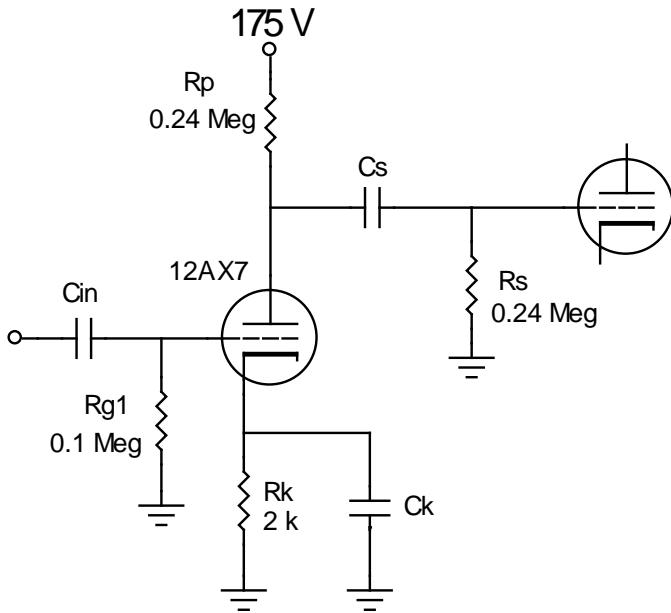
Although we are getting ahead of ourselves, we can illustrate the use of the table by making an assumption that we've performed an analysis, perhaps by using the spread sheet in chapter 6.0, and that the requirements for a preamplifier have been determined to be as follows:

Voltage gain (A_v) = 50
 Plate voltage (E_b) = 175 volts
 Signal voltage output = 10 volts, peak-peak

The closest plate voltage to 175 is 180 volts in the table and the closest voltage gain to 50 is 54. For the amplifier circuit with those characteristics, the output signal voltage is 50.9 Vrms, so there is ample margin for the required output signal voltage of 10 V p-p. We can then obtain from the table the following information:

R_p = 0.24 Meg, R_{g1} = 0.1 Meg, R_k = 2000 ohms

Gain and bias conditions are established by resistors R_p, R_{g1}, R_k and R_s; the circuit for the preamplifier would be as follows:

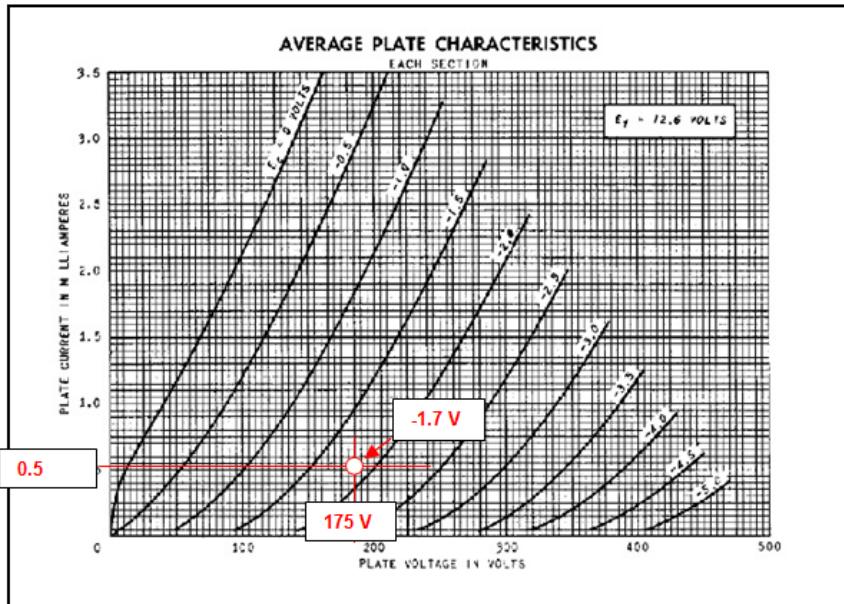


Note that different manufacturers, various texts of the period and even this book are inconsistent in the nomenclature of properties, components and values of components. When tabulated data is used, take care to understand the units of measure that are being employed in the tables (and to what component the reference designators actually refer?). Example, many tabulations mix resistor values between Megohms, Kilohms and ohms; be sure that you understand which component is described by which value.

Resistor R_s represents the grid return resistor for the next stage in the amplifier chain. It is included because the parallel combination of R_p and R_s form the total plate load for the preamplifier, i.e. the plate load is actually 0.12 Meg, which is the parallel resistance of R_p and R_s . This information isn't germane to establishing bias conditions but it is related to the voltage gain of the tube, which will be discussed in detail in a later chapter.

If it is desired to design the preamplifier stage for a specific current, rather than a specific gain (there are some situations where this might be desirable), then the plate characteristics can be used. The following set of plate curves are extracted from the same data sheet as the above table.

Assume that our desired quiescent operating conditions are $E_b = 175$ volts and $I_b = 0.5$ milliamps. Drawing a line on the "Y" axis (the plate current axis) from 0.5 milliampere and then drawing an intersecting line from the "X" axis (the plate voltage) at 175 volts, we define the operating point.



Examining the point on the curve representing 0.5 milliamps and 175 volts we can then estimate (from the curves of control grid voltages, Ec1) that the approximate grid voltage required to produce a current flow of 0.5 milliamps at a plate voltage of 175 is about -1.7 volts.

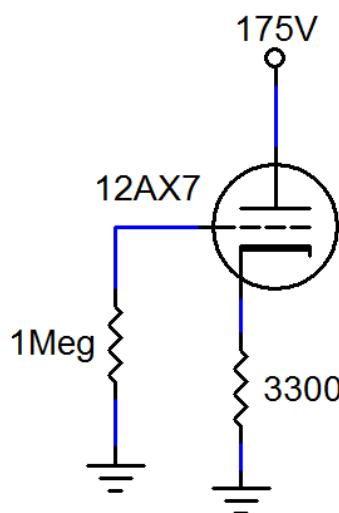
The "-" indication notes that the grid voltage is negative with respect to the cathode of the tube. Another way of expressing this is to say that the cathode must be more positive than the grid by 1.7 volts. The easiest way of obtaining the 1.7 volt offset between grid and cathode is by "grounding" the grid and then allowing current flow through a resistor connected between ground and cathode that causes a 1.7 volt "drop".

Referring to the schematic below, note that the tube grid isn't actually grounded, it is connected to ground through a high value resistor. The amount of current flowing in the grid of the typical vacuum tube is very, very small. The grid therefore represents *extremely* high impedance for audio purposes. Because the grid impedance is so high, a high value of resistance can be used to "ground" the grid. Typical values range from about 47,000 ohms (47k) to over 1 Megohm (1Meg).

To determine the value of cathode resistance required to cause 1.7 volt difference in the cathode to grid potential, we simply use the required quiescent current of 0.5 milliamps and the 1.7 volt potential difference, inserting the values into "Ohm's Law", which relates voltage, resistance in ohms and current in amperes as follows:

$$I = E / R \quad \text{which can be re-arranged to yield} \quad R = E / I$$

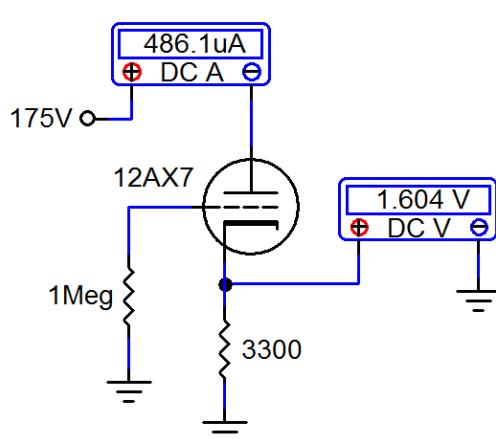
Inserting the known values we obtain $R = 1.7 / .0005 = 3400 \text{ ohms}$ (note that 0.5 milliamps = .0005 amperes) the closest standard value is **3.3 k**



In this simple circuit, the cathode resistor is the component that determines plate current (I_b), the grid resistor as noted previously is not particularly critical (47k to 1.0 Meg). The plate resistor will be selected to obtain the required gain and plate voltage for the tube. This part is not shown in these schematics and will be discussed later.

The following is a computer simulation result for the above circuit, showing the analyzed voltage drop across the cathode resistor and the analyzed current flow through the plate (so that the relationships are more easily visualized).

The relationships between current, voltage and resistance are shown.



$$I = E / R \quad .0005 = 1.7 / 3300$$

$$E = I \times R \quad 0.72 = .001 \times 714$$

and

$$R = E / I \quad 714 = 0.72 / .001$$

These expressions, simple as they are, will be used throughout our discussion and are known as "**Ohm's Law**".

Note that many of the approximations that we use and the graphical means of extracting information from plate curves result in slight errors from the design values. Add the fact that we use "standard" resistor values rather than calculated values (and that those resistance values have practical tolerances) and *it should be apparent that there will be slight errors between calculations and measurements.*

In the above example, variations in vacuum tube transconductance parameters, changing the resistor value from the calculated value result in a change in the design value of plate current from 0.5 milliamperes to 0.486 milliamperes. This is relatively typical of vacuum tube circuits - measured bias voltages and currents can vary by 5% to 15%.

These brief examples discussed *triode* tube bias adjustment only. Biasing of power pentodes (beam power tubes) are slightly more complex and will be covered in the chapter related to the design of the amplifier output power stage. A review of the triode bias calculations will be made in each chapter that requires the use of a triode.

A word of caution about published tables of values for common vacuum tubes. These devices were originally manufactured to tight tolerances and measured to determine conformation to their specifications. That is not necessarily the situation with modern imported parts. Because a vacuum tube is "labeled" with the same part number provides no assurance that it will perform identically with original EIA (Electronic Industry Association) standards or that the tabulated component values available on the internet (and sometimes used in this book) will produce the claimed results.

2.3 Amplifier Bias Adjustment

Generally, adjustment is not required on a regular basis if a few guidelines are followed and output power considerations permit. Preamplifier tubes *never* require bias adjustment. Bias provisions are not included in small-signal circuits; there is no necessity to include them. The "small-signal" description also applies to tubes in reverb circuits, tremolo circuits and phase-shift tube(s). All that remains is the output power stage(s).

Whether bias adjustments will be required for the output tubes depends primarily on amplifier design. If the design cannot accommodate series feedback, in the form of cathode bias resistors, then the necessity for adjustment depends on the tubes that are to be installed. Typically, for higher power amplifiers, one seeks to obtain maximum efficiency without undue cost impact to the power supply circuit and this frequently suggests eliminating cathode bias resistors. If this is the case, then some form of adjustment will be required, as a minimum when the tubes are replaced or start to age.

(Alternatively, additional feedback circuits may be included in the output stage to eliminate the need for tube matching. In a later chapter, one possible solution will be discussed.)

2.4 Tube Amplifier Designs Rarely Change

Since the 1930s, most vacuum tube audio amplifiers producing more than a few watts use an output circuit that we refer to as push-pull, Class "B" or Class "AB". These circuits commonly use two tubes (or two sets of two tubes in parallel, for amplifiers producing over fifty watts). The operation of the tubes will be described later in the chapter relating to the design of the output stage.

This configuration is a typical engineering compromise and therefore has

advantages compromised with disadvantages. Advantages are higher efficiency and less stress on the amplifier power supply (compared to single-ended Class "A" configurations), disadvantages are that the tubes need to be "balanced". Balance means that the change in current of each tube in the output stage should be equal under drive conditions. Distortion is always greater than linear (Class "A") circuits.

If the output tubes are not balanced, greater distortion and a reduction in output power is the result. Adjusting the bias of the individual tubes – in the manner most often described by amplifier manufacturers – usually won't balance the output stage except under a quiescent condition (quiescent means no input drive). For demanding applications, the output tubes should be balanced when they are driven at normal volume levels.

So maybe we should re-bias the output tubes when they are being *driven* in the actual amplifier? That's a better solution than the usual method however there are several tube parameters that affect balance. If the "transconductance" or even the internal capacitances of the output tubes differ significantly, balance can never be achieved, no matter how the bias of each stage is adjusted, except over a narrow range of drive level and frequency range.

Further, balancing the output stages dynamically (i.e. under normal drive conditions) requires special test equipment, training and knowledge – these might not be available at the corner music store. Against the reasons for dynamic balancing, we can pose this important question: is it worth it, will I hear the difference?

Probably you will **not** hear any difference in dynamic balancing as opposed to static balancing.

One may frequently read "arguments" for biasing output power tubes either "colder" or "hotter". There is no performance advantage for doing either. If the output transformer has been selected properly for the output tubes and operating conditions, then changing the bias in either direction from the designer's values will result in LESS power and increased distortion (reliability may also be degraded).

2.5 Matched Sets of Tubes

Those who have no means of characterizing vacuum tubes usually choose to purchase a "matched pair" or "matched quad" and that is a good idea. Tubes that have been "matched" usually will draw about the same amount of plate current at identical bias conditions.

As stated previously, this isn't necessarily the best condition for linearity but, as most of us cannot match tubes dynamically, it's about the best we can do.

Naturally, we are at the mercy of the person who is doing the "matching" (matched sets frequently originate in bedroom or garage operations, where the only test equipment available is an old tube tester – frequently a device of limited capability).

If the person who is performing the matching is technically adept, knows which model amplifier he is providing tubes for, and has the appropriate test equipment, it is possible to obtain a satisfactory set of replacement tubes. (The tubes should be tested under the same bias conditions as the actual amplifier.)

Unhappily, that may not be possible - some tube testers (especially smaller consumer models) might not have the capability of variable plate voltage and variable grid bias voltages, so it's difficult to replicate the actual amplifier operating conditions. And of course an assumption is that the tube matching service has exact information on your amplifier's grid bias and plate voltages and currents – that also may not be true.

Obviously, tube testers cannot replicate "driven" conditions. The parameters measured are quiescent - replicating typical peak driven conditions with a tube tester can result in tube failure or tube tester failure.

Despite the limitations of the tube matching process, matched sets are still a better choice than installing *unmatched* tubes, about which one knows little or nothing. You may get lucky and install a couple of unmatched tubes that sound great, especially if the amplifier is an older one that produces less than 30 watts. (This category increases the likelihood of cathode bias resistors being present.)

Amplifiers producing 40 watts and more tend to be more "finicky" about tube matching and balance. This is because there is little or no cathode feedback resistance. Cathode feedback minimizes differences in tube parameters, making tube replacement more forgiving.

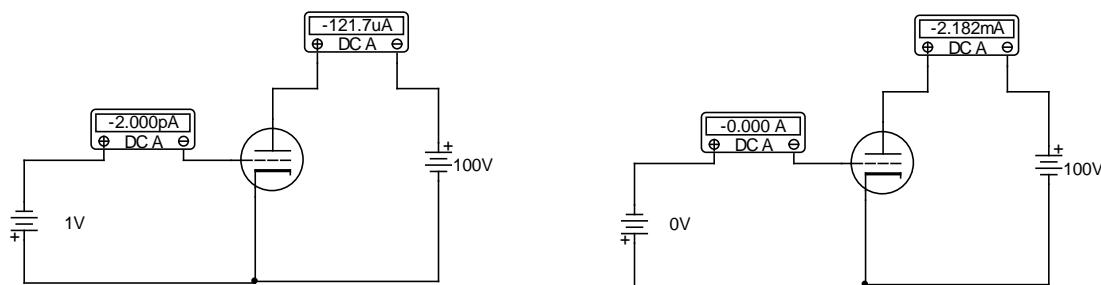
When one is designing an output amplifier stage, if the output power requirements permit, *use individual cathode bias resistors for each output tube*. The series feedback provided by the separate cathode resistors will provide a small amount of bias stability but - more - importantly, allow the bias current of each tube to be easily measured in a manner that will be discussed later.

3.0 How Vacuum Tubes "Amplify"

As discussed in Chapter 1.0, the addition of the control grid to the two-element vacuum tube made possible the first example of electronically-controlled current flow. We know that the filament (and later the filament-heated cathode) will emit electrons when it is hot and the electrons will flow from filament/cathode to plate (anode) if the plate is biased at a high positive potential with respect to the

cathode. At first, it was simply noted that when the grid was biased at a lower (less positive) voltage than the cathode, the cathode to plate current would vary as the bias voltage on the control grid was varied.

DeForest was able to demonstrate amplification with his "Audion" tube using the control grid as a "gate" or "valve" to adjust the flow of electrons from cathode to plate. The two circuits below are identical except for the voltage applied to the grid is -1.0 volts in one circuit while it is 0.0 volts in the other circuit.

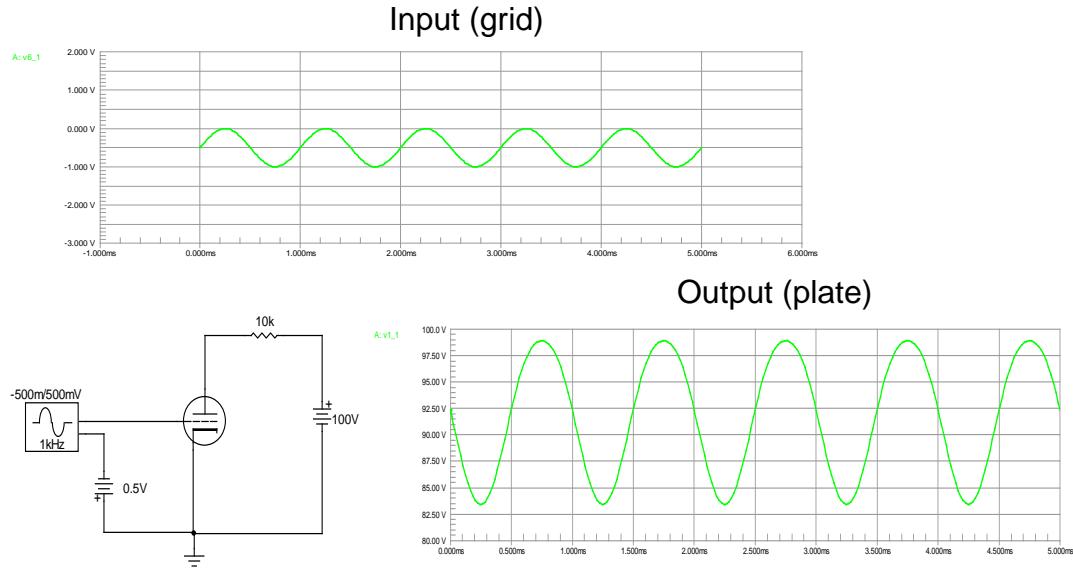


When the grid voltage is about -1.0 volts, there is a small amount of plate current flowing, about 0.122 millamps. If the grid voltage is reduced to 0.0 volts, as in the circuit on the right, slightly over 2 millamps of plate current will flow. If a resistor was placed in series with the plate (as in the next example), *the change in plate current would cause a change in the voltage drop across the plate resistor*. An observer would conclude that a change in output voltage has resulted from a change in input voltage. And if the output voltage change is greater than the input voltage change, *voltage amplification* has occurred.

Note that the current meter in series with the *grid* of the tube indicates little or no current flow. This is because the grid element is a very high impedance connection. All of the internal connections of a vacuum tube are of fairly high impedance but the control grid is the highest of any terminal. This is the most important characteristic of a vacuum tube.

De Forest demonstrated that an alternating voltage applied to the control grid could produce an alternating plate current change through a plate resistor and that the alternating voltage drop across the resistor could be larger than the grid voltage change.

An oscilloscope connected to the grid and the plate would display waveforms similar to those illustrated in the example that follows.



3.1 Definition of Voltage Gain

In the above example, the voltage applied to the grid varies from 0 to -1.0 volt while the voltage observed at the plate varies from about 83 to around 99 volts. The voltage gain in this example is

$$Av = \Delta V_{\text{output}} / \Delta V_{\text{input}} \quad \text{where "D" means "change in" or "difference between" ("input" and "output" terms are self-explanatory)}$$

The expression means that voltage gain is equal to the change in output voltage divided by the change in input voltage. For the example, the voltage gain is:

$$Av = (99 - 83) / (1 - 0) = 16 / 1 = 16$$

Several other observations can be made from this example. The first is that the output waveform is "inverted" compared to the input waveform. This means that the two waveforms are 180 degrees out of phase. The grid is "biased" with a 1 volt battery in series with the signal generator (which is adjusted to produce a signal of +/- 0.5 volt). The battery is used to offset the minimum and maximum voltages of the signal generator so that the range is within the linear range of the grid.

As a matter of interest, at the time that experiments like this one were being conducted, there was no electronic instrumentation with which to conduct the experiments: no electronic signal generators and no oscilloscopes. The signal generators were literally alternators, where a rotating coil was passed through a magnetic field and the alternating voltage extracted from the coil - amplitude was adjusted by a potentiometer (literally a variable voltage adjustment).

The coil (armature) was either hand or motor driven. The means of voltage detection was generally a mechanical voltmeter or mechanical pen-recording device, so the alternating current had to be fairly slow (low frequency) in order for the mechanical voltmeter to respond to the varying voltages accurately.

Once it was established that the "Audion" was capable of amplification, other experimenters (most notably Edwin Armstrong) saw possibilities that resulted in the development of oscillator circuits, mixer circuits, superheterodyne receivers and most analog communications circuits still in use today (except that the circuits were of vacuum tube construction).

3.2 Other Forms of Gain

Note that there are various definitions of the term "amplify" and "gain", such as in voltage amplification and current amplification. We'll use the term "voltage gain" throughout most of this discussion because it's convenient when dealing with high impedance devices, like vacuum tubes. Conversely, it is usual to refer to "current gain" when discussing solid-state, low impedance devices such as bipolar junction transistors.

A practical amplifier is one that provides power amplification. The transformation of voltage and current from one impedance level to another can produce power amplification. Current OR voltage gain doesn't have to be significant *as long as power gain is achievable*.

3.3 Vacuum Tube Configurations

There are three major circuit variations in which the vacuum tube may be configured. The above example is called the "common cathode" configuration because the cathode is "common" to both the control grid circuit and the plate circuit (the "input" circuit and the "output" circuit).

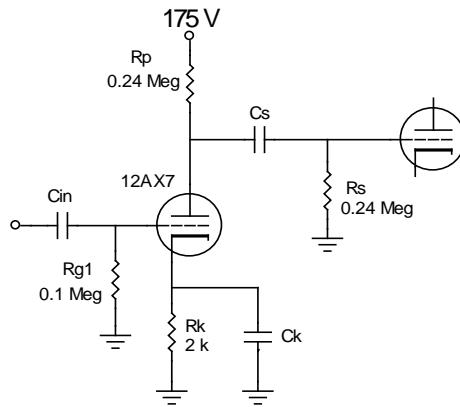
(A signal path is generally established by two conductors - but they do not necessarily consist of two wires. It is sometimes simpler to visualize the "common" electrode of a vacuum tube as being the second conductor in both the input and the output circuit.)

We will not discuss the other two configurations in the main text; they are not used in guitar amplifiers and are not topical. (Although a later chapter discussing "hybrid" combinations of vacuum tubes and transistors will illustrate an unusual topology using a common grid amplifier configuration.)

The other configuration commonly used in guitar amplifiers is the one called "common anode", or more familiarly "cathode follower". This circuit configuration is frequently used in "phase splitter" circuits and these circuits will be discussed in a separate chapter.

3.4 Common Cathode Configuration for Voltage Gain

The amount of gain that a vacuum tube triode can produce is predicated on internal characteristics and external component values. In chapter 2.2, when we discussed "biasing" a triode, we presented some manufacturer-published data, in conjunction with the following schematic:



If all component values in the above schematic are known, the tables can be useful to approximate voltage gain. Conversely, knowing the gain required for a specific application, one can select a set of component values from tabulated data that will provide the voltage gain and output voltage swing. These tabulations were published many years ago, when vacuum tubes were of consistent performance. The tables may not provide accurate performance when used with currently produced devices.

For simplicity, we'll assume that our tube is characteristic of those produced when the tabulated data was published. Examining the above schematic, we can find a condition that is representative of the component values:

R _p	R _s	R _{g1}	R _{total} (R _p R _s)	Eb = 90			Eb = 180			Eb = 300		
R _k	A _v	E _{o p-p}	R _k	A _v	E _{o p-p}	R _k	A _v	E _{o p-p}	R _k	A _v	E _{o p-p}	
100,000	100,000	100,000	50,000	1,700	31	14.1	1,000	40	42.4	760	43	84.8
100,000	100,000	100,000	50,000	2,000	38	19.5	1,100	46	56.6	900	50	113.1
240,000	240,000	100,000	120,000	3,500	43	18.4	2,000	54	50.9	1,600	58	104.6
240,000	510,000	100,000	163,200	3,900	49	24.3	2,300	59	67.9	1,800	64	132.9
510,000	510,000	100,000	255,000	7,100	50	20.9	4,300	62	53.7	3,100	66	110.3
510,000	1,000,000	100,000	337,748	7,800	53	25.7	4,800	64	67.9	3,600	69	130.1
240,000	240,000	10,000,000	120,000	0	37	25.2	0	53	42.4	0	62	90.5
240,000	510,000	10,000,000	163,200	0	44	15.3	0	60	53.7	0	67	115.9
510,000	510,000	10,000,000	255,000	0	44	14.1	0	61	48.1	0	69	99.0
510,000	1,000,000	10,000,000	337,748	0	49	18.1	0	66	59.4	0	71	115.9

(Note that capacitor values are not shown in the table above, although they will be presented in most tables furnished by manufacturers and in future chapters of this book. For signal analysis, we assume that capacitors are zero impedance at the signal frequency unless they are part of a filter - EQ - circuit.)

Observing the highlighted sections of the tabulations, it would be a good assumption that the voltage gain of this stage will be 54 and that the output peak-to-peak voltage will be very nearly 51 volts.

Other internal characteristics of the vacuum tubes are included in the calculated tables such as perveance, equivalent cathode resistance and internal plate resistance. These are other sources of uncertainty when applying the values to modern vacuum tubes. Also, up to this point, we are considering the tube cathode to be "grounded". The cathode capacitor in the above schematic blocks D.C. current flow but allows passage of signal current flow, hence "bypassing" the cathode resistor - the cathode of the tube is at "signal ground". We'll discuss this later and also make a differentiation between actual grounding and establishing the "signal ground" at the cathode.

Further, there will be occasions when the cathode is NOT at signal ground and has an intentional resistance inserted between cathode and ground. In those conditions, the published amplifier tables may not be useful for predicting static bias, voltage gain and output voltage swing. Other computational techniques need to be applied for amplifiers using series feedback (series feedback = resistor between cathode and ground). When this is discussed, we'll have to account for the internal characteristics of the vacuum tube, which we refer to as "parasitic elements".

As mentioned previously, performance predictions based on the use of tabulated data assumes the use of vacuum tubes with characteristics similar to those produced at the time the tables were published.

4.0 Amplifier Functions

By separating the different functions into easily understood "subassemblies" the chain of functions that make up an amplifier can be readily grasped. (The intent of this discussion is to eventually work through a typical design example of an amplifier.) When we progress to the design of the subassemblies and integrate them into the completed amplifier, we'll be able to evaluate each function of the amplifier as an individual circuit before chaining them all together. We won't build an amplifier but we can build a simulation of one, step by step, "testing" each part of the design as we go. Finally, we can integrate the various circuits into a complete amplifier.)

Vacuum tube amplifiers are frequently described by guitarists as having superior sound qualities compared to transistor (solid-state) amplifiers. Of course this is not an objective opinion, a lot depends on what a person is trying to accomplish and this short discussion won't get into that particular argument other than to mention that compression is the only observable difference between the two types. Harmonic differences are inevitably corrupted by output stage

configuration, transformer balance, tube bias and so many other considerations that plausible comparisons are not generally possible.

As a matter of interest, we note that the production of vacuum tubes is totally dependant upon the opinions of the people who believe that tube sound qualities are superior and are willing to pay HUGE sums of money to support those opinions. We mention this as an indication of how important opinions can be and how the market - especially influenced by the internet - influences opinions.

Large sums of money are spent, not on overt advertising, but on advertising disguised as user opinion. Take care, especially avoid "reviews" that state that differences in similar vacuum tubes produce audible differences in musical quality.

The human ear is the most unreliable, ever-changing, non-traceable, un-standardized measurement device one could devise. The adjectives that "reviewers" use to describe subtle differences in quality offer a clue to the substance of their observations.

Objective, double-blind scientific tests universally reveal that "reviews" are biased from the knowledge of what is being tested and why. In other words, there is probably an agenda. An interesting article, written by a highly respected designer can be found here:

<http://www.dself.dsl.pipex.com/ampins/pseudo/subjectv.htm>

Even if one's intention is simple repair - or modification - of existing circuits, it is useful to know how the old devices were designed. (As opposed to mindlessly copying the designs of earlier guitar amplifiers, which may have been mindlessly copied from other sources - all the way back to the invention of the beam power pentode.)

Designing a tube amplifier is both easy and difficult - mostly the design data required to produce reasonable performance are easily obtainable. These devices, despite what many might say, are *not* high-performance amplifiers. But even simple designs become difficult when working with inconsistent components that don't necessarily perform in a predictable way.

The typical transistor amplifier is about ten times more difficult to design but, in return, offers about ten times the performance of these tired old tube circuits with regard to fidelity of sound reproduction. The main reason that most guitarists prefer tubes is the compression characteristics, which are very different from compression characteristics of solid-state amplifiers.

Jazz musicians usually like clean “headroom”, implying that the amplifier is operated in a “linear” region and rarely compressed. That's not always possible, though, and one of the advantages of vacuum tube amplifiers is that they CAN operate outside of their linear region without producing obvious audible distortion (or at least not the unpleasant types of distortion products that transistor amplifiers produce).

This may be the source of confusion over what many call "tube watts" compared to "transistor watts" as we briefly discussed earlier. There is no difference in power levels produced by different types of amplification devices *provided that the same levels of distortion are being produced and that the amplifier architecture is similar*. Reviewing an earlier discussion:

As distortion increases, a "clean" sinusoidal waveform degenerates into something resembling a square wave. A square wave contains considerably more integrated power than a sine wave of the same voltage amplitude (about 41% more). So the difference in output levels from the two different types of amplifier is simply based on tolerable distortion - the human ear doesn't processes vacuum tube distortion in the same manner as it processes transistor distortion. The difference in perception could be interpreted as loudness.

“Linear performance” means that, if the input and output characteristics of the amplifier are plotted on graph paper, the result would be a straight line, hence “linear”. (Another way of describing linear operation is that the output signal is an identical copy of the input signal – the only difference being increased “amplitude” or volume level.)

Comparing vacuum tube devices to solid-state devices, as both devices deviate from linear performance, some would describe vacuum tubes as having a more “graceful decline” from linear to non-linear operation. If one were to examine both types of amplifiers on a wide dynamic range instrument like a spectrum analyzer, as the amplifiers were being overdriven, the description would seem appropriate.

4.1 Important Performance Parameters

Before discussing the individual subassemblies and their functions, we need to define the most important performance concepts: noise figure, gain, output power and distortion (distortion = lack of linearity).

4.1.1 Noise

This should be easy to understand but actually the subject is complex. There are various forms of noise (AM noise, FM or PM noise), there are various forms of noise distribution (white noise, pink noise, just about any color noise you might imagine) and there are different contributors that are mixed and summed to make

up a noise spectrum, thermal noise, flicker noise, random walk noise, and so forth.

We are going to keep things simple by considering only thermal noise. This can be justified by the fact that it's the major contributor to audible noise – noise that we can hear – in a guitar amplifier. This form of noise is universal, it's everywhere. We know this and we even know how *much* noise is everywhere. The expression that relates noise to some other factors of an amplifier is:

$$\text{Noise} = (4 \times K \times T \times B \times R)^{1/2}$$

Where noise is in volts, K is Boltzmann's constant (*a constant is an unchangeable number, frequently a physical limit, this one is named for the German scientist that first established it from his study of thermodynamics*), T is temperature in degrees Kelvin, B is bandwidth in Hertz and R is resistance in ohms. The $^{1/2}$ term means the same as taking the square root of the entire expression.

Let's not get bogged down in the mathematics, we'll simplify this term to represent AVERAGE noise at room temperature and over the loudspeaker frequency range, say 100 to 5,000 Hz (allowing for harmonic content) and assuming a nominal input (pickup) resistance of about 10,000 ohms. That leaves us with a simpler, more manageable expression:

Thermal noise at the input of an audio amplifier is approximately 1 microvolt.

This doesn't include noise contributed by the amplifier. This is important to understand: ALL AMPLIFIERS HAVE NOISE. A theoretically perfect noiseless amplifier would *still* have an audible "hiss" at the speaker output because the "hiss" results from universal thermal noise (remember that it's everywhere).

4.1.2 Noise figure

This term is used to describe the amount of excess noise in the amplifier. Since different amplifiers have different amounts of gain, the amount of noise at the OUTPUT of any amplifier can't be directly compared to another amplifier. In order to make comparison easier, noise figure is always referenced to the INPUT of the amplifier.

OK, let's make sure that we "get" this: the amount of noise emanating from a 100 watt amplifier will be much louder than the amount of noise emanating from a 5 watt amplifier even though the noise at the "inputs" of both amplifiers is equal. Intuitively, we should be able to understand that a 100 watt amplifier has a lot more *gain* than a 5 watt amplifier (at least 13 dB more, we'll get to that in a moment). Repeating: to make a "fair" comparison, we must always reference noise at the INPUT of an amplifier.

Noise figure does NOT include the thermal noise voltage that we discussed in the previous topic. Amplifier noise figure is determined roughly by three things:

The circuit resistor values, as modified by the amount of circuit gain at the resistor location.

The noise contribution of the gain devices in the circuit (whether tube or transistor) and where the gain devices are located in the circuit.

Location and turns ratio of transformers (more on this later) in the circuit. Some general rules for designing a low noise figure amplifier follow. Use low-noise tubes (or transistors), biasing them for optimum noise; noise is highly dependant on bias conditions. Keep resistor values low and distribute gain in the circuit so that gain always precedes resistance (or loss) if possible. We will discuss this in more detail as we get into preliminary design discussions.

Before we get *too* absorbed in trying to keep the noise figure low, there are practical reasons for not worrying excessively. If the object is to design a *guitar* amplifier, it is a low-fidelity, bandwidth limited and not very linear amplifier to start with. We don't require a high standard of performance from a guitar amplifier, so let's not obsess about noise for now.

4.1.3 Gain

Another easily understood concept, it's just the ratio of the output power divided by the input power. (Power is usually used because it is independent of input and output load/impedances.) The industry standard for power gain is the decibel. The name comes from inventor of the telephone, Alexander Graham Bell, an immigrant to the U.S. from Scotland. The definition of one Bel is an awkward unit of measure (it's not sensitive enough for most purposes) so we use 1/10 of a Bel therefore obtaining the "deci-bel" term. Determining the gain in decibels if the input power level and the output power level are known is as follows:

$$\text{Power gain, in dB} = 10 \times \log (\text{Pout}/\text{Pin})$$

$$\text{Voltage gain, in dB} = 20 \times \log (\text{Vout}/ \text{Vin})$$

$$\text{Current gain, in dB} = 20 \times \log (\text{Iout}/ \text{lin})$$

where the output and input power levels are in watts, the output and input voltages are in volts and current is in consistent units of amperes.

We divide the output power (Pout) by the input power (Pin) then take the logarithm (base ten - not natural log) of the result and multiply it by ten. Most pocket calculators can do this computation quickly and easily.

Let's work out an example ... suppose that we want to design a guitar amplifier that will produce 30 watts of power. Our input signal comes from a guitar that can typically produce 0.1 volt. We need to find the OUTPUT voltage level from the 30 watt amplifier. This is obtained from the simple expression:

$$P = E^2 / R \quad \text{and} \quad E = (P \times R)^{0.5}$$

where P is power in watts, E is voltage in volts and R is load impedance in ohms.

$$E = (P \times R)^{0.5} = (30 \times 8)^{0.5} = 15.5 \text{ volts RMS}$$

"RMS" means "root mean square" and is similar - although not identical - to "average". We need to make sure that our voltage units are the same for input level and output level before calculating the gain. The input level was 0.1 volts "peak to peak" - not the same as "RMS". To convert peak to peak to RMS volts:

$$V_{rms} = .354 \times V_{p-p} = .354 \times 0.1 = 0.035 \text{ volts RMS (35 millivolts)}$$

And the overall voltage gain required is

$$Av = 15.5 / .035 = 437.63$$

Av expressed in dB is $20 \times \log (437.63)$ or about **53 dB** (from the calculation of decibels described above).

4.1.4 Output Power

This parameter determines how "loud" an amplifier will be when connected to a loudspeaker (obviously the efficiency of the loudspeaker is also a major contributor). For optimum performance, the speaker impedance should be the impedance that the amplifier was designed for. Generally speaking, output power levels of a vacuum tube amplifier will decrease when the amplifier is connected to either lower or higher impedances than the design value. The amount the amplifier power will be reduced is approximately proportional to the ratio of the design impedance to the actual impedance.

For example, connecting a Fender "Bassman" designed to operate into a 4 ohm load, to an 8 ohm load will produce about 25 watts instead of the specified 50 watts. There are reliability considerations regarding incorrect impedances as well. Always try to operate the amplifier with a "matched load. If you're not sure what impedance your amplifier is designed to operate with, it's marked on the rear panel output jack to the speaker.

Over the years, and still continuing, manufacturers tend to pick their own definitions of output power, the reasons for this are usually apparent. All other electronic industries that specify this parameter do so in an unequivocal manner.

One can be assured that if a musical instrument amplifier is not specified the same way as all other equipment, there is a reason that the manufacturer doesn't want to specify it conventionally. The standard power measurement includes all of the following information:

- Power level, watts, RMS, continuous
- Percentage distortion
- Frequency range

The power level must be in RMS watts, continuous, to be meaningful and the other two parameters must also be specified for the measurement to be valid.

4.1.5 Distortion

Usually self-explanatory, except for the means of measurement. An easy way to visualize distortion is to observe (or imagine) the shape of the input waveform, on an oscilloscope for example, where voltage change is displayed as a function of time. If this can be visualized, then also imagine the output signal displayed on the same instrument.

Distortion is the difference between the *shapes* of the two sets of displayed information but NOT the difference in their magnitudes - which would be voltage gain (or loss). Observation of distortion on an oscilloscope is the grossest (least sensitive) form of measurement. Specialized instrumentation (distortion analyzers, spectrum analyzers) are required for determining distortion less than about 5 percent.

Recall that we described "linear" operation above? Here's a review: "Linear" means that, if the input and output characteristics are plotted on graph paper, the result would be a straight line, hence "linear". Another definition is now available to us: a linear amplifier produces an output signal that is an identical copy of the input signal, the only difference being "amplitude" or volume level.

Non-linear operation is distortion, whether it is caused by compression, clipping, phase shift/group delay variation, harmonic generation, whatever. *If the output signal is not identical to the input signal EXCEPT for amplitude, it is distorted.* Distortion and output power parameters must always be used together to convey or obtain meaningful information. *Describing a power level is meaningless unless the level of distortion present when the power was measured is also stated.* The converse is also true. (Industry standard for vacuum tube amplifiers seems to be the measured RMS continuous power level at 5% distortion and with a 1 kHz test frequency.)

4.2 Individual Circuit Functions

The vacuum tube amplifier configuration most commonly used today is made up of the following subassemblies:

Pre-amplifier (pre-amp)

Equalization (tone control, EQ)

Post-amplifier (post-amp, driver)

Phase Shifter/Power Divider (phase splitter, transformer)

Driver Amplifier (higher power amplifiers only)

Power Amplifier (power amp)

Output Transformer

Load (speaker)

4.2.1 Preamplifier

The main purpose of this circuit is to establish the noise figure of the complete amplifier. It is possible to eliminate this circuit - the tone control would then become the "front end" of the amplifier. We would lose gain from doing this but we could easily make up that gain somewhere else in the amplifier chain. BUT, if we did this, the overall noise figure would increase by about 13 dB.

Let's put that in perspective, if the preamplifier was removed, the detectable audio noise (that "hiss") at the speaker output would *be twenty times louder!*

Clearly we need the preamplifier to keep noise performance reasonable. Incidentally, the more gain in the preamplifier, the lower the amplifier noise figure will be but the price must be paid. Mother Nature cannot be fooled so the result of the increased gain is increased distortion. (Guitarists that play pop/rock music generally like amplifiers with lots of front-end gain. Jazz musicians generally do not.)

Typical gain values for a preamplifier might range from about 20 (voltage gain = voltage out / voltage in) up to 50. The gain, in decibels, of an amplifier or any stage of an amplifier is:

$$Av = \text{Voltage Gain} = 20 \times \log(V_{\text{out}} / V_{\text{in}})$$

where V_{out} is the signal output voltage (not the bias voltage) and V_{in} is the input signal voltage of the same stage. We then take the logarithm of the ratio of

output/input voltage and multiply by 20 to obtain decibels of gain. The expression is the same as the one for power gain except that we multiply by 20 (instead of multiplying by 10 for power gain).

4.2.2 Equalization

This circuit allows amplifier frequency response to be tailored for a specific guitar and to individual taste, more importantly to compensate for the frequency deficiencies of the magnetic pickup. There are various configurations of EQ circuits, but they will usually fall into four subcategories, with the Fender style being the most popular (and common) variety. Equalization always implies increased noise figure, there is no exception. Active EQ, touted by some manufacturers in the past, has excess noise - frequently noisier than simple potentiometer/capacitor circuits - and also reduced linearity (which means increased distortion).

The contribution of the EQ circuit to noise varies but generally will be in the region of 10 dB. Best noise performance is almost always obtained with the EQ set flat or in the position of least loss through the circuit. (A disadvantage that may result from flat EQ is that excess gain may produce distortion in the following stages.)

By now, you should be starting to get the impression that the design process involves trading off some good things with some bad things. Engineering is all about compromise: enhancing certain important areas of performance and degrading other not-so-important parameters.

The voltage loss of the typical EQ stage is around 0.1, in other words, the output voltage divided by the input voltage is 0.1. To get the gain in decibels, as we did in above examples, we take the logarithm of 0.1 and multiply by 20, thus finding that the EQ circuit has a typical "gain" of -20 dB.

The minus symbol will automatically result when you take the logarithm of a number less than one and it always signifies "loss" which is the opposite of gain. Gain is always represented with a positive number of decibels. No sign at all is always positive and is therefore represents "gain".

4.2.3 Post Amplification

The post amplifier contributes a large part of the overall gain of the complete amplifier (compensating for the loss of the EQ circuits in the process) and is usually implemented with/by the second element of the dual triode used for the pre-amp (the 12AX7 or 7199 families are universally used). The most important consideration of this stage is linearity. A general rule of thumb is that the higher the bias current, the better the linearity (less distortion).

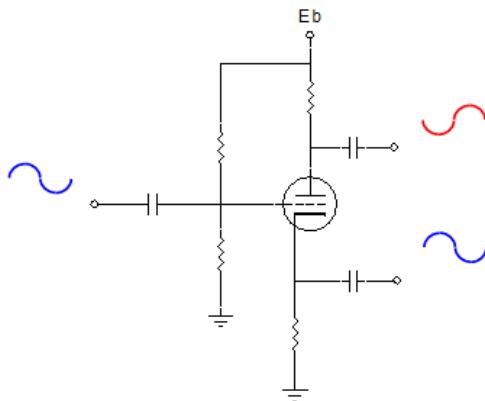
A “typical” gain value for a post amplifier used in an amplifier like the unit previously discussed can be from twenty or so up to nearly sixty (voltage gain) which is equal to a power gain of some 26 to 36 dB. (See above equations for calculating voltage gain and power gain in decibels.)

This is applicable to typical "triode" type vacuum tubes. Beam power tubes (discussed later) can have higher gain values than triodes.

4.2.4 Phase Shifter/Power Divider

The purpose of the phase shifter (sometimes abbreviated to phase-splitter from phase shifter + power splitter) is to convert a single signal into two separate signals that are 180 degrees out of phase. A simple transformer can easily accomplish the same thing. Many years ago, when tubes were inexpensive, it was cheaper to use a tube (or tubes) to perform the phase shifting function than to use a transformer. That practice has continued today, since almost all modern vacuum tube amplifiers are not designed, they are simply copied.

An example of a phase-splitter without gain, showing phase relationships between input and outputs:



There are two types of phase shifters, differentiated by whether they have gain or not. The common configuration in smaller, low-power amplifiers is a phase shifter that has no gain – it simply divides the incoming signal into two signals and inverts the phase of one of them. This type always uses a single tube - or else one element of a dual triode - and examples of this are seen in amplifiers like the small Fenders (e.g. the Princeton mentioned above).

Once we venture into power levels of 20 watts and greater, a different phase shifter should be used (or more accurately, should be used if one desires to maintain a low tube count in the amplifier). This phase shifter normally uses two tubes (usually in the form of dual triodes "packaged" in the same glass "envelope").

For our 25 watt example amplifier, the application requires that the phase shifter provide some gain so a two-tube design is necessary. Typical voltage gain for this example might be around 8.

4.2.5 Power amplifier

The universal configuration is two tubes, in push-pull configuration, operated in Class AB (or Class B, but this is rare). The two tubes are driven from the two different signals provided by the phase shifter which are always 180 degrees different in phase. This is equivalent to saying that the grid of one tube will always be positive when the grid of the other tube is negative and vice-versa. The result is that one tube supplies current while the other tube is turned "off", the tubes are alternately "pushing" and "pulling" current through the output transformer, hence the description "push-pull amplifier".

The "class" of the stage refers to a technical definition regarding current flow in the tube. Class "A" amplifiers have approximately the same amount of current flow throughout a complete signal cycle. Class "B" amplifiers produce current flow for about $\frac{1}{2}$ of a signal cycle in each of the two tubes of the output stage. Class "AB" is simply a combination of the two. Current flow occurs throughout the entire signal cycle but is not constant - the current changes in proportion to the input signal amplitude.

Beam power tubes are universally used in guitar amplifiers while high-power triodes are frequently found in high-fidelity equipment. Typically, tubes that are required to provide high levels of power have much less gain than those used in preamplifier functions (usually called "small-signal" applications). This is another reason why so much additional gain is required from the post amplifier stage, to make up for the *lack* of gain in the power amplifier.

Up to this point, we haven't discussed issues like maximum safe voltages and maximum power dissipation. These don't usually concern us for applications that don't provide more than say ... a watt or so of signal power. Working with power tubes, one must pay careful attention to the operating characteristics for several reasons:

Ignoring proper operating voltage and current will result in premature failure.

The operating voltage and current determine the amount of power that can be produced.

Operating voltage, current and tube parameters determine the output impedance of the amplifier; we will get into that in more detail in the next chapter.

The correct operating parameters would normally be found on the tube data sheet. Unhappily those data are not always accurate these days, since the tubes are produced in an environment that doesn't encourage consistency and quality. So, when one is using modern tubes, the data sheets are useful for *approximations* of the operating conditions, the safety notes and output impedance specification. If NOS (new, old stock) tubes are used – and there are still many available – the data sheets can be more useful and are easily located on the internet.

Note that I am not suggesting that the new tubes are necessarily inferior to the ones produced a half-century and more ago. I am suggesting that there is a lack of consistency but one might easily infer that this might suggest enhanced performance in certain areas rather than degraded performance (although it is not likely).

4.2.6 Output Transformer

One of the most misunderstood parts of the vacuum tube amplifier that performs two simple functions.

Vacuum tubes are high-impedance devices, unlike transistors (transistor audio amplifiers rarely require output transformers, they are found only in very old designs). The ultimate goal of a guitar amplifier is to drive a loudspeaker, which is a very low impedance device.

The most efficient power transfer occurs when the driver impedance (the amplifier) and the load impedance (the speaker) are equal. The transformer performs this simple but absolutely necessary function. A transformer is simply two coils of wire that are magnetically coupled (connected) by an iron or steel (magnetic) core.

A signal current passing through one winding of the transformer will induce a current to flow in the other winding of the transformer in direct proportion to the ratio of the number of turns of wire on the input coil to the number of turns on the output coil. Without going into the theory of these devices, it is adequate to state that the transformer can also change impedances from input to output. If the input coil has more turns than the output coil, the transformer will change a higher impedance to a lower one. The converse is true. The impedance transformation follows a simple mathematical expression.

$$N_{pri} / N_{sec} = (Z_{pri} / Z_{sec})^{0.5}$$

Where N_{pri} is the number of turns on the primary coil and N_{sec} the number of turns on the secondary coil, Z_{pri} is the impedance presented to the primary coil and Z_{sec} is the resulting impedance that will occur at the secondary coil. (The $^{0.5}$ superscript means that the square root of the expression in parentheses is

extracted.) This equation can be restructured in several different ways to achieve transformation information. During the design phase of the task, the above is the most useful of the equations.

A simple example: assume that an amplifier has an output impedance of 1000 ohms and needs to drive a speaker with an impedance of 4 ohms. That is a ratio of 250 to 1. If we take the square root of 250 we get about 16. So the number of turns on the primary coil needs to be 16 times greater than the number of turns on the secondary coil:

$$N_{pri} / N_{sec} = (Z_{pri} / Z_{sec})^{0.5} \text{ so } N_{pri}/N_{sec} = (1000/4)^{0.5} = (250)^{0.5} = 15.8$$

A transformer can also have voltage gain or voltage loss (or current gain/current loss). This is directly proportional to the ratio of turns on the primary and secondary coils, so:

$$\text{Voltage Gain} = N_{pri} / N_{sec}$$

and expressed in dB would be

$$20 \times \log (N_{pri} / N_{sec})$$

The other function of the output transformer is to provide the operating voltage and current to the two output tubes. This is accomplished by means of a "center tap" on the primary transformer winding. Each of the two primary winding inputs are 180 degrees out of phase with the other and the "center tap" is a virtual ground. This means that, in a perfect transformer, no *signal* voltage would be present at the center tap, therefore it is the ideal location for injecting the plate bias voltage since the primaries are connected to each of the plates of the output tubes.

Imagine the construction of the center tap as follows: an insulated wire is wound, say twenty times, around a magnetic core. At that point, the insulation is stripped from a small section of the wire before continuing to wind twenty more turns around the core. The ends of the wire constitute the out-of-phase inputs while the center, the portion with insulation removed, becomes the "tap", the point at which the plate voltage would be connected.

4.2.7 Load

This term is sometimes used in place of the loudspeaker normally connected to the amplifier output. This term is used because the loudspeaker may not be the component that is always connected to the amplifier, particularly during testing. The speaker is usually replaced with a large power resistor while testing – if the speaker was used, the test environment would be unbearable due to the high sound pressure level.

Additionally, every type of speaker has different, unintended "parasitic" characteristics, impossible to include in a representative speaker "model" and impossible to obtain meaningful measurements from. So we use "load" to mean anything connected to the amplifier *of the correct impedance*. To be accurate, we would use the expression "matched load".

(Note that, the more turns on a transformer (high turns ratio), the more resistance is introduced and the lower the efficiency. Considering these disadvantages, it is recommended that loudspeaker impedance be as high as possible provided that the loudspeaker performance is not also impaired by higher resistance and higher moving coil mass. This permits a lower turns ratio for the transformer, lower resistance and higher efficiency. *As with all design engineering, tradeoffs are possible and case-by-case analysis is recommended.*)

5.0 Amplifier Specifications

Many of the terms, parameters of performance and simple equations describing performance were previously described. Before starting any design, and especially the design of a vacuum tube amplifier, one must determine what the design is required to accomplish. In industry this is called many different things: preliminary specification, design goals, performance criteria and so forth, it makes no difference what we call it so let's just call ours "the specification".

This isn't an exercise; it's not efficient to attempt the design of an amplifier without first making a determination of what it must accomplish. A set of specifications can be as informal as simply "remembering" them or the usual practice of making written notes. If problems are encountered after the construction of an amplifier, it's not possible to effectively trouble-shoot and correct them unless one knows the required performance characteristics of each individual stage in the amplifier chain.

Remember that engineering is a series of compromises, both in the area of defining a specification and in designing the hardware. In general, it's best to make iterations and compromises on paper, rather than having to continually modify hardware. If one intends to copy an existing design, then reading this for reasons other than entertainment or to enhance understanding of individual circuits wouldn't be useful.

Introducing this subject earlier, we mentioned important performance parameters of an amplifier, such as noise figure, gain, output power and distortion. In addition to these four parameters, we also need to confirm the input and the output characteristics of the design. Let's establish a set of preliminary specifications to work from. In the table below, the abbreviation "TBD" means "to be determined" and we'll fill in the blanks as we work our way through the preliminaries.

Input impedance: TBD
Input maximum signal level: TBD
Noise figure: TBD
Gain: TBD
Frequency response: TBD
Power output: TBD
Distortion at rated power output: TBD
Output impedance: TBD

Some information required to fill in a few of the blanks is already known or can be determined. It's not too difficult to estimate the impedance of your guitar pickup and the maximum output voltage it can produce. Similarly, speaker configurations *might* be established now and the amplifier output impedance specified to be equal to the speaker impedance.

5.1 **Guitar Pickup Impedance and Signal Level**

Most jazz guitarists use some variation of the "humbucking" pickup patented by Gibson fifty years ago. We can use the characteristics of that pickup to suggest input specifications of our amplifier. A "humbucking" pickup can generate as much as 0.5 volts peak-to-peak as measured on an oscilloscope. The impedance of a pickup like this can vary considerably but a figure of around 10k ohms is probably adequate for our approximations.

Note that there are so many different pickup configurations available that an exact circuit equivalent is not possible - the two parameters above are suggested to be average but they do not include unintentional "parasitic" complexities present in most pickups. Chapter 6 provides a means of amplifier analysis using a spreadsheet, different impedances and output voltages may be substituted in the spreadsheet if desired.

Note that the output signal from a guitar pickup is not a sinusoidal waveform, it is complex and rich in harmonics. Because of the complexity, peak to peak voltage measurements are not the best way to measure output voltage. A better description of the pickup characteristics is to measure the RMS or the average output voltage of the pickup. (The two measurements are not the same but close enough to be adequate for our purposes.) It's also normal to specify the pickup impedance, if it is known, when describing the output characteristics.

5.2 **Amplifier Input Impedance**

In most applications, one must pay attention to impedance levels and voltage levels between *any* two circuits, we'll get into this in detail as we actually commence a practical design. For now, we'll keep things simple by noting that the guitar pickup has a "high" impedance. If we want to obtain the maximum

output voltage from the pickup, we need for the input impedance of the circuit that follows the pickup to be even higher. (The lower the amplifier input impedance, the lower the amount of voltage available from the guitar pickup.)

Let's use an example to illustrate this concept since it's an important one. Assume that one has a transistor radio battery, one of those small, rectangular nine volt batteries. Clip the test leads of your inexpensive digital multi-meter across the battery, first moving the selector switch of your DMM (digital multi-meter) to the "DC voltage" position and to the appropriate range to measure the 9 volt battery (for most DMMs, this will be the 20 volt range).

Assume that you have at hand a number of power resistors of many different values. With the DMM still connected to the battery and reading somewhere around 9 volts, carefully place the leads of several different value resistors across the battery terminal. "Carefully" because the resistors are going to get HOT if they dissipate any appreciable amount of power (which they WILL if low value resistors are placed across the battery terminals).

As different values of resistors are placed across the battery and as the battery voltage is measured, it will be apparent (and probably intuitive) that lower values of resistance produce lower values of measured voltage. The reason for this is that the battery has an INTERNAL impedance just like our pickup, it's usually a very low impedance, maybe an ohm or less. When the external resistor value is HIGH, compared to the battery impedance, the battery voltage will be unchanged as the resistor is attached to and removed from the battery terminals.

If the resistor value is LOW, compared to the battery internal impedance, then the measured voltage of the battery will appreciably diminish. As noted in several previous places the maximum POWER transfer occurs when the internal and external impedances are equal. What we're trying to achieve for our guitar pickup, however, is the maximum amount of *voltage* that we can obtain from it. We'll explore the differences and desirability of power transfer and voltage transfer as we start detailed design procedure.

Maximum output voltage implies, as we've just ascertained, that the input impedance of the amplifier has to be considerably higher than the impedance of the pickup. If we multiply the pickup impedance by a factor of 10, that would be reasonable for the input impedance of our amplifier. So $10 \times 10k$ (pickup impedance) = $100k$ so we can fill in our first blank on the preliminary spec sheet, "Input Impedance". (Note that this is a minimum value since an increase in the input impedance will not diminish the voltage produced by the guitar pickup.)

5.3 Output Impedance, Speaker Impedance, Output Power, Frequency Response

Here's one of the few places that one can exercise a little individual creativity - most of the other decisions that we make are going to have constraints imposed on us by Mother Nature. So how "loud" do we want this amplifier to be? And do we desire a single speaker or multiple speakers?

First of all let's make sure that we understand that "loudness" isn't directly related to amplifier output power but it IS proportionally related. To illustrate this point: an amplifier that produces 100 watts if connected to a speaker that is only 10% efficient is going to sound about as "loud" as a 15 watt amplifier that is connected to a speaker that is 67% efficient. Both are going to produce about 10 watts of "listenable power" after efficiencies are considered.

Generally loudspeakers are rated by a parameter called "sound pressure level" (SPL) instead of "efficiency". SPL is the amount of audible power measured by a calibrated microphone located 1 meter from the loudspeaker and with a constant drive voltage (usually 2.828 RMS volts for 8 ohm speakers) connected to the speaker terminals. This is an effective way of estimating the efficiency of a given speaker. SPL levels are expressed as a ratio, in decibels (dB), to a standard reference level of amplitude. As intuition suggests, the higher the SPL level the more efficient the speaker.

That's not the whole story, though, because the frequency response of the speaker must be taken into account. It's always best to look at the manufacturers plotted and specified performance of SPL as a *function of frequency* to make sure that there are no surprises (peaks and dips in the response).

But we're lucky in *this* application, which is not a high-fidelity amplifier. The guitar has a limited frequency response, around 80 Hz to just over 1 kHz. However, in order to perceive tonal variations, we need also to provide enough audible bandwidth so that the first few harmonics are also included in the frequency response of the speaker. In practice, if the speaker can respond to about the third harmonic, this will be adequate.

(In fact, providing a higher frequency response isn't even desirable - except for acoustic guitars - since pick noise, string squeaks and other miscellaneous accidental taps and scratches will be heard. And as described fully in the earlier "Introduction" chapter, too much bandwidth means additional noise. So we want the lower bandwidth to be the same as the lowest frequency of our instrument, or about 80 Hz. The upper frequency limit should be just enough to allow passing the first three harmonics, or thereabout.)

We can now define the frequency response of the speaker, and therefore the amplifier. For our purposes we will let the response be 80 Hz to 4 kHz, we can now fill in the appropriate blanks of our preliminary specification with this number and go on to select a speaker that meets our requirement.

It should be noted that, insofar as amplifier design is concerned, there is usually no penalty resulting from having excess bandwidth *provided that the loudspeaker filters out the excess noise associated with excess bandwidth*. In all cases, the active portions of the circuit provide much more high frequency gain than required, enhanced by the negative feedback applied (more on this later).

Generally, between the output transformer and especially the loudspeaker, the practical high frequency limit is established. If the speaker does NOT provide audible noise filtering, then we can add this within the amplifier, for example in the negative feedback loop (when we get to that part of our design). Just keep in mind, for example, that doubling the frequency response of an amplifier also doubles the audible noise.

At this point, another of the main performance parameters must be defined: output power. We have to do it now because we can't select a speaker unless we know that it is capable of handling our power level. This determination is going to depend on anticipation of the largest venue normally performed, weight, cost and a few other different issues.

We can point out that an observation of what works for others in the same/similar venue would be a good place to start. As an example, if a Fender "Deluxe" is working well for a guitarist in a group playing a 200 seat night club, and you anticipate similar size venues, then an amplifier of 25 watts would be a good selection. (Keep in mind when designing your amplifier any future needs. For example, it might be a good idea to provide a "preamp out" connection, allowing an option to patch into the house PA while using your amplifier tone controls.)

Remember from the "Introduction ... etc" chapter that distortion must always be specified with output power level? We'll use the industry standard of 5% for our purposes.

Specific recommendations for a speaker aren't appropriate here - there are so many criteria that must be considered. In addition to performance, power level, price, size limitations, personal opinions also influence the selection process. And if multiple speakers are considered, the problem becomes more complex. But we can make a few observations about selecting a speaker.

Speaker SPL is rated in dB, which is a logarithmic unit of measure. Human hearing is also logarithmic, so the system of SPL makes good sense from several aspects.

A difference of +3 dB or -3 dB in loudspeaker SPL, for example, is equivalent to either doubling or halving amplifier power - this is significant.

As an illustration, perhaps you have performed a preliminary analysis and identified two speakers as *suitable* for your needs:

Speaker "A" costs \$70, has a response of 100 - 6500 Hz and a SPL of 95 dB.

Speaker "B" costs \$95, has a response of 70 - 4000 Hz and a SPL of 98 dB.

Which is the logical choice ?

Spending the extra \$25 on speaker "B" buys an increase in SPL of 3 dB. As we've said, that's equivalent to doubling amplifier power. Could you achieve twice the power from your amplifier by spending another \$25? Probably not - there are many implications: twice the amplifier power means twice the current required, twice the amount of heat generated, larger power transformer, larger output transformer, more weight and so on.

In general, selection of the most efficient loudspeaker is ALWAYS a wise choice. (That's why so many JBL and E-V speakers were retro-fitted into so many Fender cabinets that originally were loaded with Jensens and similar low-cost products.) Another generality that is frequently useful: lower rated power speakers are *usually* more efficient than those rated at higher power levels. (A chapter concerning this topic is included later and may provide more insight.)

OK, let's assume that we've made the speaker selection and found that the speaker is available in 8 ohms impedance. This allows us to fill in more blanks on our preliminary spec sheet. Here's what we have, so far:

Input impedance: 100k ohms minimum
Input maximum signal level: 0.1 volt, p-p
Noise figure: TBD
Gain: TBD
Frequency response: 80 - 4000 Hz
Power output: 25 watts
Distortion at rated power output: 5%
Output impedance: 8 ohms

We can make a reasonable estimate for the gain, just as we did in the previous chapter, by using amplifier output power or voltage and determining the ratio to the input signal. (Note that a spread sheet will be described later that will enable a designer to determine many of these specifications quickly and also permit tradeoff analyses between parameters.)

Power measurements of musical instrument amplifiers are specified in watts "rms", which we discussed briefly in the previous section. Reviewing, "RMS" means "root mean square", it's an intimidating term but it means something close to "average" (although not quite). To convert peak to peak voltage to rms voltage, just multiply by .354 so our 0.1 volt peak to peak input signal becomes 0.035 volts (or 35 millivolts) RMS.

Now we'll use the same expression used previously to determine the RMS output voltage for any output power level.

$$P = E^2 / R \quad \text{and} \quad E = (P \times R)^{0.5}$$

where P is power in watts, E is rms voltage and R is resistance (or impedance) of the load (speaker). (The term "^{0.5}" means the "square root" value and most calculators include this function on their keyboard.) Let's determine the output RMS voltage for a 25 watt amplifier:

$$E = (P \times R)^{0.5} = (25 \times 8)^{0.5} = (200)^{0.5} = 14.14 \text{ volts RMS}$$

The overall voltage gain is the ratio of output voltage divided by input voltage and is:

$$14.14 / 0.035 \text{ which is about } 400$$

Big numbers and small numbers are usually hard to work with, so we convert them to decibels, as was described in previous discussions. The conversion is:

$$\text{Gain in dB} = 20 \times \log (\text{output voltage} / \text{input voltage})$$

If we take the logarithm of 400, the ratio that we just calculated, we get 2.6 and if we multiply that number by 20, the result is **52 dB**, the gain required for our amplifier to achieve full power from the pickup that we've selected as typical. We can add that number to our specification table.

The gain calculation is appropriate for the 25 watt output power that we've selected. It should be apparent that increasing the output power level will require more amplifier gain because the input signal level is limited to the guitar pickup capability. The practical inference is that, if more power than 25 or 30 watts is required, then more gain will be required. The additional gain may not be achievable using the architecture of the existing amplifier chain.

In Chapter 6.0, a spreadsheet will be described that is useful for estimating gain distribution and other performance parameters that are dependant on gain. Generally, when the output power requirement exceeds 25 watts, the additional gain required is provided by the phase-splitter stage. When we've progressed to

that point in the discussion, then decisions can be made regarding the configuration and design goals of the phase-splitter.

Additionally, negative feedback loop parameters need to be considered and we'll discuss that further in 8.44. The feedback characteristics of the amplifier chain are some of the most important aspects of the overall design: output power, gain and most importantly, distortion (or lack of linearity).

5.4 Noise Figure Considerations

This can be a tricky specification to determine or a very simple one, depending on the process used and the assumptions made. The *simplest* way to estimate the required noise figure is to make an assumption that (a) the noise of the vacuum tubes are what they are or (b) that they are noiseless. In either case, there's nothing a designer can do about the available tubes, so the noise figure specification is generally ignored.

That's convenient and easy but the decision *could* affect design decisions in unpredictable ways later and one *might* end up with a noisy amplifier as a result of making a hasty assumption. This, however, IS the way that the problem is usually approached. So for most people, the story of the preliminary specification table will end right here.

Input impedance: 100k ohms
Input maximum signal level: 0.1 volt, p-p
Noise figure: as the design permits
Gain: 52 dB
Frequency response: 80 - 4000 Hz
Power output: 25 watts
Distortion at rated power output: 5%
Output impedance: 8 ohms

5.4.1 Pragmatic Definitions of Noise Figure

There are differing opinions about how noise should be specified or how much can be tolerated. One opinion considers the amount of quiescent noise (the noise that is emitted from the amplifier when NO signal is present) and how that should determine the specification. Another opinion suggests that the RATIO of the noise to the FULL power signal should determine the specification. There are arguments for both opinion processes.

If one agrees that the noise limit should be based on *quiescent* noise, then other choices have to be made, e.g. does the maximum noise apply when the amplifier is in your living room (or other "quiet" environments) or when it is located in a

noisy nightclub? So, although this is the tempting choice, perhaps it's not the easiest one to make. Personal preference, personal opinions, and so forth ... and there may be a cost penalty if one insists on absolute silence from a vacuum tube amplifier.

The following represents an aggressive method for estimating noise figure but one that has *some* basis in common sense.

A SPL of 30 dB is generally acknowledged to be about the lowest musical level that is audible to the average human ear. If our selected loudspeaker has a SPL of 98 dB, can we use these two numbers to establish a noise figure specification? I think that we can, so let's determine how to do it ...

The SPL of 98 dB was measured at an input drive level of 1 watt (2.828 volts RMS driving an 8 ohm load = 1 watt). Let's use our predicted power level of 25 watts and estimate the new SPL level at full output power. It would be based on the original level of 98 dB SPL at 1 watt PLUS the difference between 1 watt and 25 watts expressed as dB:

$$98 + [10 \times \log (25 / 1)] = 112 \text{ dB SPL}$$

If we use the SPL of 30 dB (lowest music level detectable) then the signal to noise ratio at the speaker should be at least:

$$112 - 30 = 82 \text{ dB}$$

Applying the 82 dB (it's a ratio, not an absolute value) to the input, we can begin to approach the actual input noise figure specification. The maximum allowable noise would be 82 dB LESS than the guitar signal voltage of 0.035 volts rms.

Recall that voltage gain/loss, in dB, can be determined by $20 \times \log (V1 / V2)$, where V1 and V2 are the two voltages that we want to compare so we can rearrange the equation to determine the ratio between the two voltages

$$V1 / V2 = 10^{[(\text{gain or loss in dB}) / 20]}$$

The term "10^[and so forth]" means the "antilogarithm" of the term within parentheses and, as noted previously most pocket calculators make the determination of logarithms and antilogarithms very simple.

And our pocket calculator reveals that solving the equation gives a reduction of 12,589 from the original guitar signal, so the noise shouldn't exceed:

$$0.035 / 12,589 \text{ or } 2.8 \text{ microvolts}$$

Reviewing 4.1.1, we noted that thermal noise at any amplifier input is around 1 microvolt. The noise figure of our amplifier is the ratio of 2.8 microvolts to 1 microvolt, expressed in decibels or:

$$\text{Noise figure} = 20 \times \log (2.8 \text{ microvolts} / 1 \text{ microvolt}) = 8.9 \text{ dB}$$

NOTE: This is an optimal noise figure - many popular amplifiers will exceed this level and not be offensively noticeable. In point of fact, power supply noise (hum) will normally exceed thermal noise in vacuum tube amplifiers. (Solid state amplifiers are usually the opposite of this, generally having low power supply noise.) For our purposes, we'll just round this off to 10 dB and be done with it.

Input impedance: 100k ohms minimum
Input maximum signal level: 0.1 volt, p-p
Noise figure: 10 dB max
Gain: 52 dB
Frequency response: 80 - 4000 Hz
Power output: 25 watts
Distortion at rated power output: 5%
Output impedance: 8 ohms

The examples are somewhat simplified and practical designs frequently require a lot of iteration and revision of various specifications. Because there is interaction between various parameters and because performance/cost tradeoffs are normally being evaluated, it's sometimes helpful to have a tool to speed up the process of determining firm specifications for the various circuits within the amplifier.

6.0 Performance Estimates Using a Spreadsheet

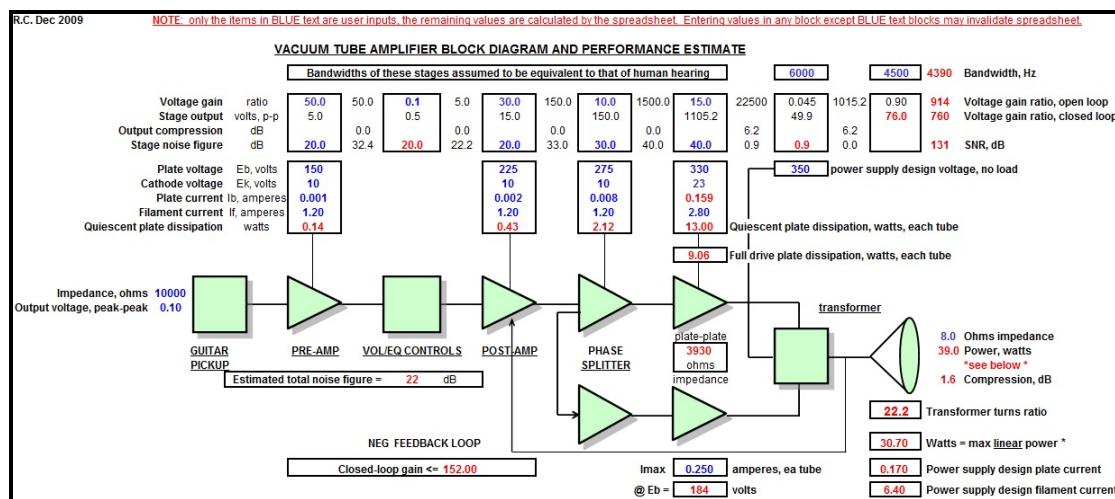
Shown below is a copy of one of the spreadsheets developed to make preliminary estimates of amplifier performance using Microsoft Excel. It's useful for checking noise, gain distribution, compression and the like. Output power prediction and output transformer characteristics can be estimated by another spreadsheet in the package. The spreadsheets are related and data is interchanged automatically between them as the design develops.

Spreadsheets are as ubiquitous as computers although they may not be used for purposes more sophisticated than balancing the family checkbook. A spreadsheet can be used to perform some of the functions provided by conventional "programming languages" (the closest one to spreadsheet language being "Basic").

Spreadsheets are simple to program, provide some sophisticated mathematical functions and can even be used in an optimization mode by employing internal "Visual Basic" capabilities that provide branching, looping, for-next and other techniques commonly associated with programming.

The spreadsheets generated for use with this book are specific to the manual calculations described in the appropriate chapters with a few enhancements. Various "what if?" scenarios can be quickly evaluated with these tools with little likelihood of computational error (once the spreadsheet is free of bugs).

The amplifier configuration shown in the spreadsheet below includes no "effects" such as reverb or tremolo, it's just an amplifier. A simple implementation can be made with as few as four tubes: one dual triode, a phase-splitter and two output power tubes (five tubes total if a tube rectifier is included in the power supply). It's possible to obtain up to 50 watts output power from this configuration. Up to 100 watts is obtainable if the output tubes are paralleled (four tubes total) and the gain of the phase-splitter is increased.



Blue text indicates user entries and red text indicates calculations performed within the spreadsheet. Some of the entries are also used for calculations that occur on other spreadsheets, accurate information must be provided. If you use placeholders for preliminary estimates, remember to replace them with accurate values before committing to a design. It's recommended that one accumulates the tube data sheets (easy to find on the internet) and makes copies of them before starting this exercise, as well as all other parts documentation that may affect performance. The documentation may be useful in the event problems are encountered when the design is proofed (the "breadboard" stage).

From the data sheets bias voltages and currents may be extracted as well as typical and maximum expected gain and distortion values. These operational requirements should be added to the spreadsheet as tube types are selected. Gain values entered in the spread sheet need to be *obtainable* - tube data sheets

will provide guidance on this subject. Incorrect (overly optimistic) gain entries will ripple throughout the amplifier chain and provide false results for many of the calculated performance parameters. It's best to use conservative (safe) values when developing a new design. Voltage gain values of around 40 for example, are suggested for the preamplifier/post-amplifier stages.

At this stage of personal design development, the spreadsheet that may be most useful is the "block diagram" shown above. This sheet is useful for estimating required stage gain and evaluating compression, for example. After we've developed the skills to determine how to do these things manually, we'll make more frequent use of these tools, speeding up the design process.

Several other spreadsheets will be available for similar purposes, there is one specific to the design of the output stage of the amplifier. Another spread sheet is useful for designing the power supply and filter circuits. As noted previously, the spreadsheets were created with Microsoft "Excel", therefore any spreadsheet program used must have the ability to read "Excel" files.

Use of the tools presumes basic background knowledge of the design process. If one is willing to work through all of the chapters on design, then the spreadsheets can be extremely useful and time-efficient. I don't recommend using them in a vacuum, however; a thorough understanding of how the results are obtained will result in confidence in the "answers".

We will work through a practical design example in chapter 22.0, indicating how spreadsheets can be employed to design the architecture, and many of the details of a medium power amplifier, in a matter of a few hours.

For now, in order to boot up our design exercise, we need to determine, as a minimum, the gain distribution throughout the stages that make up the amplifier. We need certain vital pieces of information such as input voltage level and output voltage level + output impedance in order to get things moving. If one hasn't a clear idea of what to do with this information at this point, it's a good idea to review chapter 5.0 and prepare a set of specifications. Once one is familiar with the design process and (especially) manipulating the spreadsheets, the procedure will become straightforward and simple.

Risking boring and repetitious suggestions that may cause lack of interest, I *strongly* recommend that the remainder of this text (excluding appendices and certain chapters that aren't directly related to design) be read and hopefully understood before starting a design, even as an exercise. As we've noted, all design is iterative but commencing a design without sufficient information to make informed decisions can be frustrating due to the large amount of repetitive work necessary. Frustration can stimulate a negative mental feedback process that results in poor decisions and poor implementation.

7.0 Precautionary Information

I assume no responsibility for the use of this material except as a learning tool. I strongly recommend that no circuits/circuit descriptions contained herein be copied from an assumption that these are successful and proven circuit designs. My intent is to make available, to those of you who are interested, some of the techniques used to design these circuits. I do not represent these discussions to be more than informed opinion - my intention is to stimulate the reader's curiosity to seek greater knowledge of the subject.

Potentially lethal voltages are present in equipment/circuits described here. **BETTER**, acquaint yourself of the hazards of constructing similar circuits before attempting to construct them yourself.

8.0 Designing the Output Power Stage

All of the design procedures described here are approximations. A more rigorous technique is not within the scope of this brief discussion, nor is it appropriate for the people that are anticipated to have an interest in this non-scholarly effort. I've used these simple approximations to design several vacuum tube amplifiers successfully, as have many others before me.

Performance variations from these approximations are almost always correctable when the amplifier is tested. Hopefully there is adequate information provided here for an inexperienced reader to perceive the relationships between various components and the performance parameters of the completed amplifier.

With ALL power amplifiers, solid-state or vacuum tube, the appropriate place to start is with the output stage. Most of the important performance parameters will be determined by the selection of these parts and the design of the output circuit. We'll spend more time discussing the output amplifier than the other amplifiers in the chain.

Issues of concern, regarding the output stage, are applicable to a lesser degree with other stages. The output stage is also the most challenging in terms of performance compromises and most of the cost drivers of a guitar amplifier are contained in (or heavily influenced by) the output stage.

The design process is based on the following:

A specified requirement

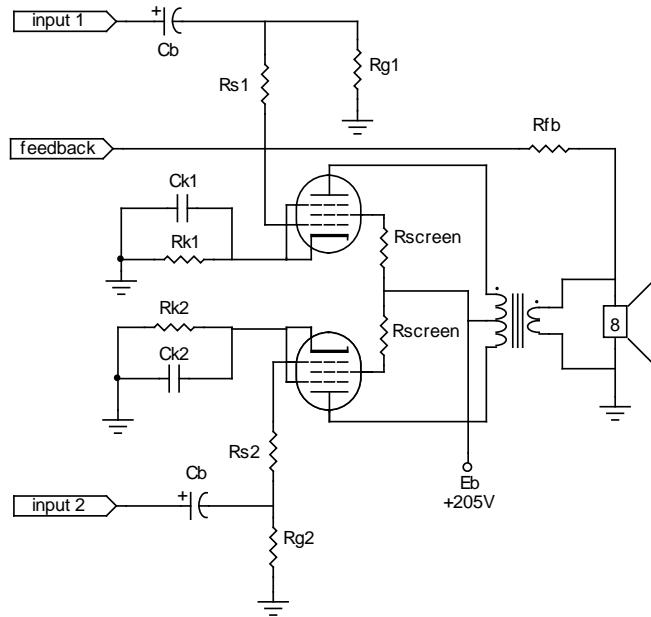
An architecture that is presumed to satisfy the requirement

Analysis supporting the performance of the architecture and justifying the selection of components that comprise the architecture

A lot of compromise

Many decisions, made during design, affect other areas of the circuit architecture and, consequently, other performance parameters. This always leads to reiteration: repeating a procedure previously performed. (Reiteration doesn't imply anything about choices previously made by a designer, it simply means that the design is evolving and hopefully improving.)

Below is a schematic representation of a typical power stage, transformer coupled to a loudspeaker(s). There are many variations of this circuit but the basic configuration is universal to all guitar amplifiers that produce more than a few watts of output signal power.



8.1 The "Push-Pull" Output Stage or "Balanced" Amplifier

Before we try to design an output stage, we must first have some understanding of why this particular configuration is universally desirable. The reasons are straightforward; listed in order of importance they are:

Efficiency (ratio of output signal power to D.C. power supply consumption)

Partial suppression of power supply ripple

Partial harmonic suppression

The first reason - efficiency - is by far the most important since it drives so many cost considerations in an amplifier. The two most expensive components in any guitar amplifier are the power supply transformer and the output transformer.

These two items comprise about 75% of the non-replaceable parts cost.

Because the cost of these critical parts is directly related to their current handling ability, and therefore their power capability, it should be apparent that the highest efficiency configuration is always preferred. This is why high-power Class "A" musical instrument amplifiers don't exist. (Affluent audiophiles sometimes rationalize the purchase of Class "A" stereo amplifiers for normal music reproduction - despite inefficiencies that might heat a small home during winter months.)

In addition to the previous points regarding efficiency, there is also the consideration of reliability. One expects to replace vacuum tubes from time to time, they are - by definition - "expendables". But "passive" components of the amplifier are expected to have a significant lifetime. It has been demonstrable for nearly a century that the biggest contributor to electronic part failure is heat. A circuit that has an efficiency of 50% will be significantly cooler than a circuit with an efficiency of 25% and have significantly longer component life.

8.1.1 Why Are Balanced Amplifiers More Efficient?

Balanced amplifiers are *not* more efficient by definition. Advantages of the balanced configuration apply in all cases *except* efficiency. One must look to the bias classification of the pair of amplifying devices in the balanced amplifier (whether solid-state or vacuum tube) to understand efficiency.

This is an appropriate time to define the differences between classes of amplifiers, about which there is universal misunderstanding among musicians. Here are some definitions, expressed in terms of current flow:

Class "A" circuits conduct current throughout the entire signal cycle and efficiency is around 25%.

Class "B" circuits conduct current for about 1/2 the signal cycle and efficiency is typically 50%.

Class "C" circuits (which are not used in musical instrument amplification) conduct current for less than 1/2 the signal cycle and efficiencies are better than 50%.

Class "D" circuits (solid-state amplifiers) conduct current only when a signal is present and efficiency may be as high as 70%.

If we desire the 50% efficiency of a Class "B" amplifier (Classes "C" and "D" aren't practical for vacuum tube audio amplifiers) then we have to wonder: what

happens to the other 1/2 of the audio signal when the vacuum tube is not conducting current?

That is where the "balance" part of the balanced amplifier configuration becomes important. If we could *select* which half of the signal cycle was being amplified by each tube in the output amplifier and then re-combine them ... ?

For example, if one of the output tubes amplified only the *positive* portion of the signal cycle and was idle during the *negative* part of the cycle.

And if the other output tube amplified only the *negative* part of the signal cycle and was idle during the *positive* portion of the cycle.

And if we could then COMBINE the two half-cycles of amplification to provide one *entire* cycle of amplification.

Then we would have a classic push-pull (balanced) output amplifier stage.

Here are the functions required for the push-pull power amplifier to function properly:

Divide a *single* input signal into two signals that are equal in amplitude but opposite in phase.

Send the divided signals to each of the two output tubes.

Each of the output tubes amplifies *half* of the signal while "idling" during the other half signal cycle (not drawing - or wasting - current).

Combine the two halves of the amplified signals into a single complete cycle.

Transform the very high output impedance of the vacuum tubes into the very low impedance required by a practical loudspeaker.

The schematic depicted at the end of chapter 8.0 accomplishes all of the above functions except for the very first one. That function will be performed by the "phase-splitter" which will be discussed in greater detail after we learn more about the output amplifier design.

8.2 Tubes, Transformers, Power Supply Voltage

These items are so interrelated that I've developed an Excel spreadsheet for the purpose of evaluating performance trade-offs between them. I'll make this available as well as other spreadsheets that may be helpful to those interested in this subject. As previously mentioned, my spreadsheets were created with

Microsoft "Excel", therefore any other spreadsheet program used must have the ability to read "Excel" files.

If you've waded through my previous meanderings, presumably you are sufficiently interested so that you may have a specific project in mind. I've frequently felt that a versatile configuration is a 25 watt amplifier with a single 8 ohm, 12 inch speaker. That's a convenient size amplifier with adequate power and not too costly. It is also a historically successful *product* - probably 80% of traditional jazz amplifiers during the seminal period of jazz guitar fit this description. Let's use that type of amplifier as a design example.

8.3 Selecting Output Tubes

We'll start the parts selection process by looking for a pair of appropriate output tubes. The selection process the industrial designer usually follows consists of establishing necessary performance criteria, cost goals, reliability considerations, long-term procurement availability - anything that would affect long-term production (whether favorable or adverse). We are fortunate in that we can reduce this to a few simple personal choices:

- Output power capability
- Availability of replacement parts
- Cost

The selection process is simplified by the fact that choice is limited - there are few manufacturers of vacuum tubes and associated/supporting components. But surplus, NOS (new, old stock) tubes, exist- and at attractive prices. The overwhelming temptation will be to select tubes of known performance - the ones that have been used in amplifiers for many years. (There's nothing wrong with that philosophy and most engineering managers would applaud taking this path if performance is adequate and cost-effective.)

There are, however, reasons for looking elsewhere for output tubes: cost and performance, for example. I don't feel limited by what mainstream amplifier manufacturers are using and, having some insight, I pass along to you my opinions. (Useful until the people who sell tubes "catch on" and raise their prices, perhaps.)

Tubes made for radio and television markets were the most volume-intensive ones produced. The 6L6, EL-34, EL-84, 7868 (and so forth) audio tubes used in guitar amplifiers never enjoyed production levels like this although the 6V6 did. Some applications in television receivers encouraged the design and production of tubes that can be very useful for audio application. Horizontal output circuits used beam power pentodes with high power dissipation and flexibility of screen bias voltage adjustment, suggesting performance enhancement tradeoffs (with supply voltage, for example).

These tubes are obviously no longer made but there are many thousands of them available - inexpensive too, compared with 6L6 tubes and the like. Many sources for these tubes can be located with a brief internet search.

For reliability, the power dissipation of the output tubes should always be greater than the amount of power that they must deliver to the load (load = transformer + speaker). For a 25 watt amplifier with two output tubes, we will need each tube to have a power dissipation greater than $25 / 2$ or 12.5 watts.

Looking over a list of surplus tubes, I see several of interest. Let's list some of these by part number, power dissipation and price to obtain a simple means of comparison. I'll start out by listing the EL-84, a pair of these will produce around 25 watts. They are commonly used for guitar amplifiers so they provide a good reference standard. (NOTE: estimated costs are those of the year of this discussion: 2009.)

Tube, Plate Power, Cost

EL-84*, 12 watts ea, \$15 ea

6BQ6, 11 watts ea, \$2 ea

6CW5, 14 watts ea, \$5 ea

6JN6, 18 watts ea, \$5 ea

6V6*, 14 watts ea, \$10 ea

6Y6, 12.5 watts ea, \$5 ea

One can easily see the trend: tubes indicated with the asterisk are commonly used in guitar amplifiers and cost a LOT more than surplus tubes primarily intended as horizontal deflection amplifiers (or other television circuit applications).

There aren't any subtle reasons for the continuing production of certain tubes and the obsolescence of others. Vacuum tube designs for the past several decades are an exercise in imitation - the same tubes are used now because they were used "then". Tube manufacturers simply provide the product that customers request; they do no development or research because there is no reason to do so.

My personal choice, given the above selections, would be the 6JN6, at 18 watts dissipation for each tube, the plates won't "glow" when used in a 25 watt amplifier. Additionally, from what I've observed, this tube is very well-

documented, having one of the most complete data packages available. It's also, by vacuum tube standards, one of the most modern designs.

Assume that we've made the tube choice, at least for now. The next step is to locate a data sheet (simple internet search) for the 6JN6. Print it out to include with the documentation that you will accumulate as you go through the design process. Looking over the tube data sheet, let's pick out anything that might affect our initial design decisions:

The tube is a fairly large one, in 12-pin configuration. It will require a 12 pin socket, looking up the socket, they are available off-the-shelf for about \$6 each.

Heater operates from normal 6.3 VAC and requires 1.2 amperes of current per tube (make a note, we'll need this information when we get to power supply design)

Plate supply can be as high as 500 volts, a nice safety margin for our application.

As previously noted, the tube can dissipate 17.5 watts each.

These tubes were still being produced by General Electric well into the 1960s, so they are likely some of the "newest" of NOS tubes.

Now let's get out the calculator and make a few approximations to help make additional choices easier. What we need to determine first is an estimate for plate voltage (output tubes require the highest D.C. voltages in the amplifier). Voltages required by other tubes in the amplifier will be discussed in the individual circuit design and power supply design chapters.

We also need to select an output transformer, one that will safely handle the 25 watt signal power plus the plate current that the output tubes will require. We need to find a transformer that will "match" the output impedance of the power tubes with the speaker impedance. The term "match" will be discussed shortly.

As previously noted, circuit design is about compromise, getting from "Point A" to "Point B" in a single straight line is unlikely. There will always be detours and some backing up required. As individuals desiring to learn about amplifier design, we don't have constraints that would be imposed on us if we were corporate engineers - there's no schedule pressure, no large cost constraints, no quality assurance requirements to satisfy, no reliability target, and so forth. All we need do is design and build ONE amplifier for our own education and enjoyment.

8.4 Estimating Plate Voltage and the Tube Data Sheet

Let's first make an initial estimate for the plate supply voltage required to produce 25 watts of output power (or more).

One of the most important parameters in power tube design is a parameter called "Imax". "I" is the symbol universally used to represent "current". **Imax**, defined here, is the amount of plate current measured for a vacuum tube operating at the following conditions:

Control grid 1 voltage (**Ec1**) is 0 volts ("E" universally denotes voltage)

Plate voltage (**Eb**) at 60% of normal or expected plate operating voltage

(In Chapter 17.0, we'll discuss in more detail the implications of **Imax** for beam power tubes, how the parameter affects circuit details and how it can be changed when a performance improvement or cost advantage is indicated.)

Let's use these terms to make some approximations of performance:

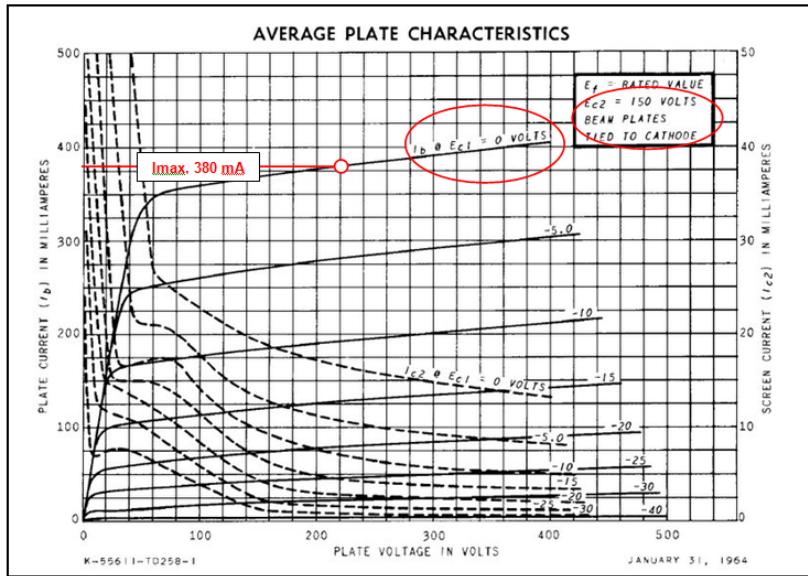
$$P_{out} = 0.32 \times Imax \times Eb \quad \text{or} \quad Imax \times Eb = 3.125 \times P_{out}$$

Pout is the required output power in watts, Imax and Eb are as described above. We can simplify and re-arrange the formula, replacing Pout with 25 watts to give us:

$$25 = 0.32 \times Imax \times Eb \quad \text{rearranging} \quad Imax \times Eb = 78.125$$

Now, refer to the data sheet "plate characteristics" curves (shown below), find the graph that represents "plate current" in the vertical (Y) axis and "plate voltage" in the horizontal (X) axis. Look at the various curves in that graph and find the curve that is labeled "**Ib @ Ec1 = 0**", which just means that the curve represents the *plate current when the control grid voltage is 0 volts* and at the screen grid voltage noted on the data sheet. The symbols used by the manufacturer who prepared the data sheet may vary slightly but you should be able to figure things out.

Examining the curve, you'll see that the variation in plate current is fairly small for large variations of plate voltage. Find the approximate center of the curve (from the "knee" of the curve to the end of the curve) and note the plate current value there ... for our 6JN6 tube, around 380 millamps or 0.38 amperes.

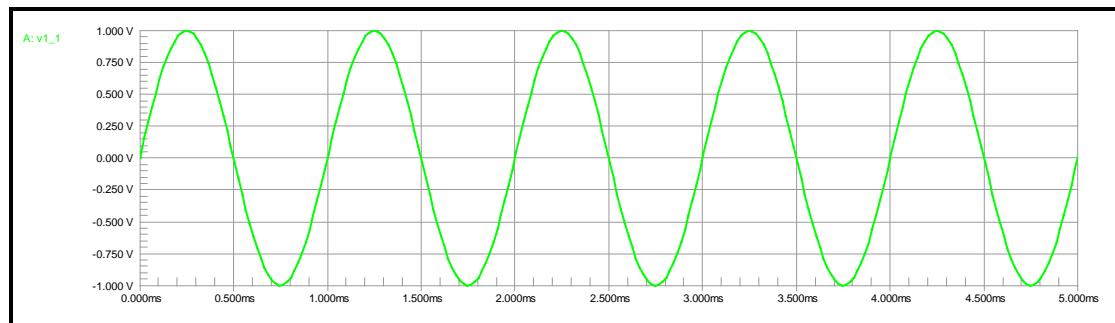


Now we can re-arrange/simplify our formula again, writing in .38 for "Imax":

$$0.38 \times E_b = 78.125 \text{ so } E_b = 78.125 / 0.38 \text{ and } E_b = 206 \text{ volts}$$

Trial and error approaches work well for vacuum tube designs, one needn't follow the above procedure to get from "A" to "B". The classical vacuum tube design procedures were usually graphical (and therefore intuitive). Reading through this discussion will hopefully offer some understanding of the techniques and allow one to develop a successful approach that is not necessarily identical to the one that I've taken.

An interesting fact that I've noted in reading old literature regarding tube design, is that the engineers rarely used RMS voltage or current terms; they preferred the use of average voltage and current. There's not a lot of difference between the two distinctions but there are places in this discussion where we will use both terms, so let's understand the difference. Consider a signal (alternating) voltage with an amplitude of + and - 1 volt. We would refer to the amplitude of the signal as 1 volt peak, or more commonly, 2 volts peak-to-peak.



If we want to convert a peak-to-peak voltage (or a peak voltage) to either RMS or average, the conversions are as follows:

$$\begin{aligned} V_{\text{peak}} &= V_{\text{rms}} / (2)^{0.5} & = .707 \times V_{\text{rms}} \\ V_{\text{peak-peak}} &= V_{\text{rms}} / [2 \times (2)^{0.5}] & = .354 \times V_{\text{rms}} \end{aligned}$$

$$\begin{aligned} V_{\text{peak}} &= 2 \times V_{\text{avg}} / p & = .637 \times V_{\text{avg}} \\ V_{\text{peak-peak}} &= V_{\text{avg}} / p & = .318 \times V_{\text{avg}} \end{aligned}$$

So the practical difference between the two terms is some 10% or so. Let's also point out that it's necessary to keep the various units of measurement consistent when making any computations. As an example, the following units are consistent:

Volts, amperes, watts
Millivolts, milliamperes, milliwatts
Microvolts, microamperes, microwatts

The following units are inconsistent and will result in computational error:

Volts, milliamperes, watts
Millivolts, milliwatts, amperes
Volts, amperes, microwatts

While we are examining the plate characteristics, there are a few other pieces of information that we should note. In the legend for the plate graphs, there are two provisions, one reads as follows:

Ec2 = 150 volts Ec2 is the technical abbreviation for screen grid (or grid 2) voltage.

The curves from which we extracted several items of data were measured with the screen grid biased at 150 volts, it follows that *if the screen grid is NOT biased at 150 volts, then the information is invalid*. Add the screen bias voltage to the information that we're accumulating regarding the design of this stage.

The other note in the plate characteristics legend reads: **Grid 3 tied to cathode**.

What this means is that the "repellor" (grid 3) must be electrically connected to the cathode to obtain performance similar to the data measured. Let's add that fact to our design information.

Now let's be clear on what the 206 volts represents. It's not actually the plate *supply voltage*, as one might logically infer, the 206 volts is actually the voltage *swing* at the plate. If our vacuum tube were perfect, 206 volts would be the amount of voltage deviation between plate and cathode. But if we spend a

moment looking at the plate curves, we see that they are linear over most of the plate voltage range but definitely NOT the entire range.

When the control grid voltage, I_{c1} , is 0, the plate curve starts to deviate from linearity at about 80 volts. The plate voltage, under maximum drive conditions, can't swing below 80 volts without severe distortion. So to insure linear operation, we should set the plate voltage to swing 206 volts *above* 80 volts. The plate voltage simply becomes

$$E_b = 206 + 80 = 286 \text{ volts}$$

This is a very conservative operating point - perhaps a greater change than actually required but let's use it for now. We can always reiterate the estimates based on a lower value of E_b if necessary.

At this point, a brief iteration is required to refine our estimate of I_{max} . Recall that I_{max} is the plate current for 0 volts grid bias and at 60% of plate voltage, so we the new value of I_{max} needs to be extracted from the plate curve at 60% or 286 volts or 172 volts. Referring to the plate curve, $I_{max} = 370 \text{ mA}$.

8.5 Estimating Quiescent Bias Current

We call the normal operating current of the tubes " I_q ", which comes from "I", the normal symbol for current and "q" which means "quiescent". This is the current that flows in the output tubes when NO signal is present. We also refer to this as the "bias current". It is related to I_{max} as follows:

$$I_q = I_{max} / p = 0.318 \times I_{max}$$

since we've established I_{max} as 0.370 amperes, then

$$I_q = 0.318 \times 0.37 = .118 \text{ amperes or } 118 \text{ milliamps}$$

Note that this is the value for BOTH tubes operating together, a single tube plate current would obviously be half, or **59 milliamps**.

8.6 Quiescent Power Dissipation

Before continuing, we need to confirm that the output tubes are within safe operating power dissipation limits. This shouldn't be a problem because we selected tubes that were rated higher than our application, but let's check to be sure.

During "quiescent" conditions (i.e. no signal), the power dissipation is:

$$P_{diss} = E_b \times I_q$$

and since we now know E_b and I_q then

$$P_{diss} = 286 \times .059 = 16.8 \text{ watts}$$

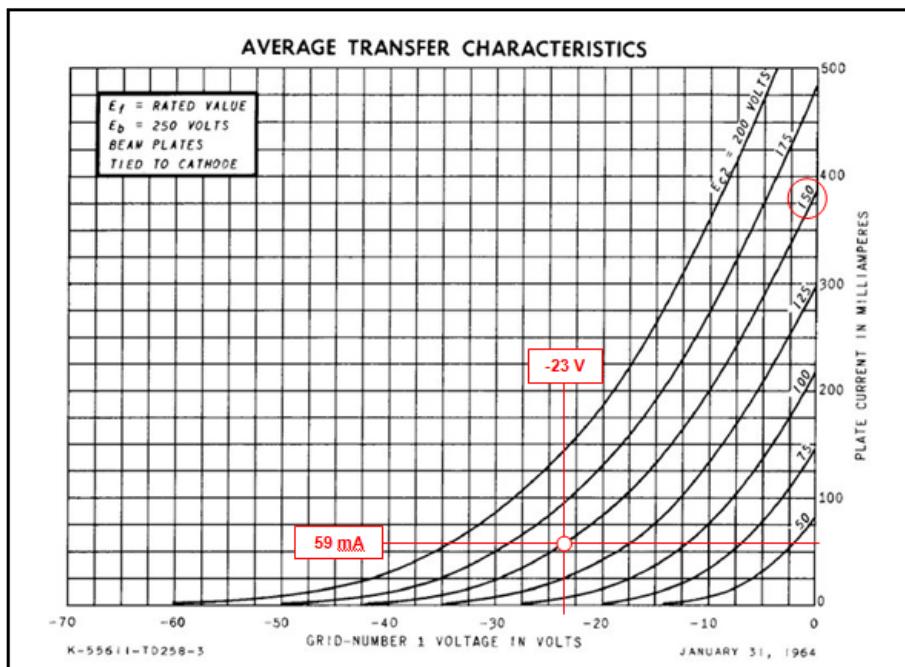
Comparing that with the allowable dissipation of 17.5 watts the tubes are operating within safe limits.

To avoid confusion in the future, let's define the voltage SWING (the peak to peak deviation) at the plate as E_o (as in E "out") to differentiate between the static plate voltage, E_b .

8.7 Determining Grid Bias Voltage

Now that we're satisfied that the tubes are operating safely, let's determine the grid voltage for each tube that will result in a plate current of 59 millamps per tube. We'll refer back to our tube data sheet curves to determine this, using a different graph this time (shown below).

Examining the vertical (Y) axis of the graph, which is plate current, we can draw a line representing the desired plate current of 59 millamps horizontally. Finding the curve representing a screen grid voltage of 150 volts, as discussed above, note the intersection of plate current and screen voltage (as shown in the example below), then draw a vertical line down from the intersection to the horizontal (X) axis that represents grid voltage (E_{c1}). The grid voltage that will produce a plate current of 59 millamps for screen voltage of 150 volts is -23 volts, record that parameter on our tabulated list of output stage design characteristics. Here's what the graph should show:



8.8 Screen Grid Current

Getting back to the screen grid, additional information is required so that we can properly bias the screen grid (which will be covered later). The information required is *screen grid current*, usually abbreviated as **Ic2**. Look through the various tube data until a second graph is found, one that includes screen grid current **Ic2** and grid 1 voltage (**Ec1**).

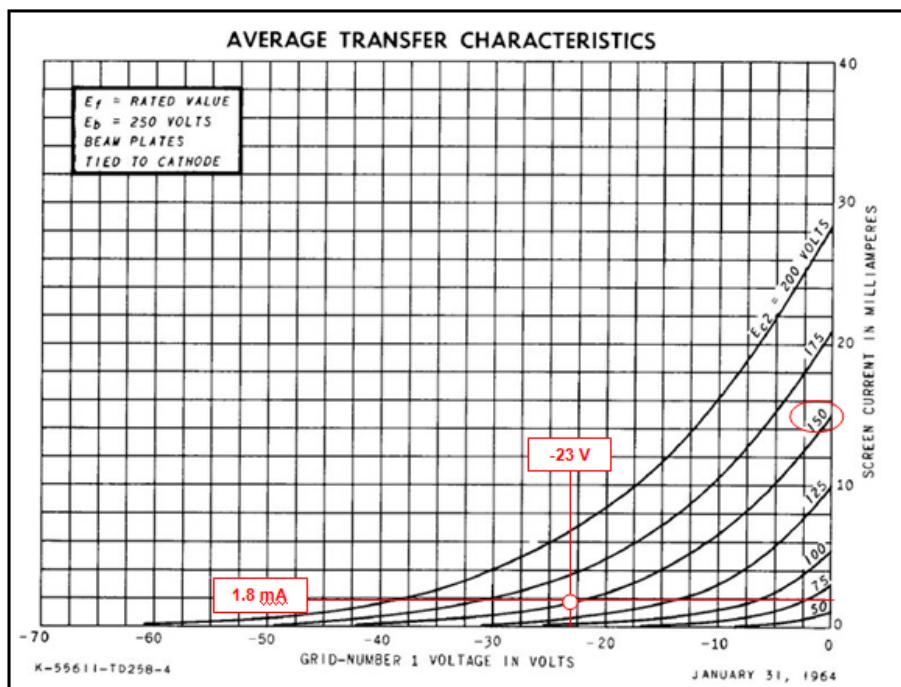
What's required is the screen grid current, **Ic2**, at the operating conditions of the output tubes, specifically at **Ec1** (grid 1 voltage) of -23 volts. Here's how we are going to find that information:

Examining the graph, we note that the horizontal axis (X) is grid 1 voltage (**Ec1**) so let's move along that axis until we find our operating grid bias voltage of -23 volts and draw a vertical line upward from that point. Now look for the curve that represents screen grid voltage (**Ec2**) and find our operating screen voltage of 150 volts.

Make a point where the vertical line of -23 volts grid voltage intersects the screen voltage curve of 150 volts. From that point, draw a horizontal line that intersects the screen grid 2 (**Ic2**) axis and from that axis and extract the screen current of about:

$$\text{Ic2 current} = 1.8 \text{ millamps (mA)}$$

Here's a copy of the curves with important points annotated:



Add a note of this current to the technical data that we are accumulating for our output amplifier design.

Up to this point, it wasn't too difficult to obtain the tube operating information. We MAY not want to use this particular plate voltage and we'll get into that later. Continuing our estimates and selection of parts ...

8.9 Estimating Output Impedance

Let's make an estimate of the tube output impedance now, so that we can look for a suitable output transformer. This information can be obtained graphically from the plate curves but it's useful to have an approximation (e.g. for spread sheets). **R_{pp}** is the symbol for plate to plate resistance (the resistance of both of the plates in the output stage) and is approximately:

$$R_{pp} = 4 \times E_o / I_{max}$$

E_o, I_{max} are known from the above as 206 volts and 0.37 amps (recall that E_o is the *voltage swing* at the plate, not the static plate voltage, E_b)

$$R_{pp} = 4 \times 206 / 0.37 = 2227 \text{ ohms}$$

8.10 Estimating Transformer Requirements

Now, as in earlier discussions, we can relate the impedance transformation required to transform 2227 ohms to 8 ohms, expressed as the turns ratio between the primary coil of the transformer and the secondary coil of the transformer as follows:

$$R_{in} / R_{out} = (N_{pri} / N_{sec})^2$$

where R_{in}, R_{out} are input and output impedances and

N_{pri} / N_{sec} is the turns ratio of the input and output coils of the transformer. The term "²" means that turns ratio must be squared (multiplied by itself). Re-arranging, simplifying and solving the equation:

$$N_{pri} / N_{sec} = (R_{in} / R_{out})^{0.5} \quad N_{pri} / N_{sec} = (2227 / 8)^{0.5}$$

$$N_{pri} / N_{sec} = (270.516)^{0.5} = 16.685 \quad (\text{note that the } "0.5" \text{ term means the square root of the number within the parentheses})$$

Let's select an output transformer to suit our requirement, one that will handle the signal power and the required plate current of the output tubes. Note that some manufacturers don't specify the allowable plate current because they have accounted for the bias current AND the maximum signal current. If there is no

specified bias current specification provided by the transformer manufacturer, it's a reasonably safe presumption that the transformer selection can be made based on maximum output power.

We need a transformer with a turns ratio of approximately 16.685 and that can handle at least 25 watts of power and at least 118 millamps of continuous current. *Note that it is always good practice to select parts that have a greater capability than the design requires.* Be assured, when selecting transformers, however, that there will be penalties in cost, size and weight if one is overly conservative. This may be the only exception to the general design rule of selecting parts that are capable of much higher stress than the design requires.

Looking over my spreadsheet of vacuum tube associated parts, I find the following transformer:

Turns ratio: 16.394

Power: 60 watts

Cost: about \$85

Availability: in stock

Although the plate current capability is not specified, the fact that the device will handle more than twice the required power indicates that it is a VERY safe selection, albeit a very HEAVY selection. A more practical design would be attained by shopping around for a transformer more closely rated to our output power requirement.

8.11 Re-checking Output Power

Since this isn't the exact turns ratio that we calculated (although it's closer than most situations that I've encountered), let's make a quick check to determine that we can still obtain the required output power with the turns ratio and the plate voltage selected.

The output power from a given plate voltage and load impedance is approximately:

$$P_{out} = (3.2 \times E_o)^2 / (8 \times R_{pp})$$

where the terms are as defined previously except that **R_{pp}** now becomes the speaker impedance transformed by the new turns ratio of the transformer (not the turns ratio that we calculated) and **E_o** is the voltage SWING

$$R_{pp} = R_{out} \times (N_{pri} / N_{sec})^2$$

where **R_{out}** is the speaker impedance and **(N_{pri} / N_{sec})** is the turns ratio of the transformer that we've selected. Substituting values for these:

$$R_{pp} = 8 \times (16.394)^2 = 2150$$

then substituting this value and the value for E_o

$$P_{out} = (3.2 \times 206)^2 / (8 \times 2150) = 25.3 \text{ watts}$$

This meets our design goal of 25 watts, although linearity considerations and approximations in the design procedure may slightly erode this estimate. A spread sheet has been developed to perform these routine calculations but it's important to know how to perform these estimates with no more than a calculator and the tube data sheets otherwise there is little or no understanding about how to achieve performance goals or what needs to be changed in order to modify performance.

It has been reasonably simple to define the parts that make up our output stage and estimate a few performance parameters. If we document what we've done so far (I'm going to use fictional part numbers for the speaker and transformer so that no specific brand or distributor is suggested), we might end up with something like this:

8.12 Output Stage Characteristics, Preliminary

Voltage gain: TBD*

Minimum output power: 25 watts

R_{pp} : 2150 ohms

Plate voltage required: 286 volts

Screen grid voltage required: 150 volts

Screen grid current: 1.8 mA

Grid 3 is connected to cathode

Grid 1 (control grid) bias voltage: -23 volts

Maximum current required: 118 mA (average, both tubes)

Maximum dissipation/allowable dissipation: about 94%

Speaker impedance: 8 ohms

Transformer turns ratio: 16.394

Filaments: 6.3 VAC @ 1.3 amperes each

Speaker, SPKR-123 1 each \$95 \$95 total

Transformer, TRF-456 1 each \$85 \$85 total

Vac Tube, 6JN6 2 each \$5 \$10 total

Tube socket, SKT789 2 each \$6 \$12 total

*still to be determined - this will become important when we design the stages that precede this one.

If this were a commercial project, at this point several more iterations would occur, the object being a cost/benefit analysis. For example, different

combinations of output transformer/plate voltage would be analyzed to see if a performance, cost or weight advantage might result.

(Since we are regarding this as an individual one-time project, probably the only thing we'd look at closely would be weight. A transformer with more than twice the power-handling capability that we require definitely suggests a heavier-than-necessary part. We'd want to look around for a lighter one.)

8.13 More On Transformers

So far, we've progressed smoothly through the various estimates and selections that led us to this point. But there are always problems and the way that transformers are specified is an important one. This hasn't been mentioned before because there were a couple of concepts that had to be understood first.

Distributors try to be helpful with the transformer selection process. They describe their product in terms of what buyers might be used to "seeing". There are several problems with that approach. Here's an example that I'm copying from a catalogue:

25 watts, primary 7,600CT, suggested tube types 6L6GC, 6V6, 807, 5881, EL34

A note in the catalogue states that: "*all units have secondary impedances of 4, 8 and 16 ohms*".

There's nothing wrong with the way that this transformer is described except that it is oriented toward those that *repair* amplifiers, rather than those that are attempting to *design* them. The first noticeable parameter after the power rating is the primary impedance of 7.6k and the "CT" notation that tells us that the transformer is "center tapped" and therefore appropriate for push-pull circuits.

If one isn't very experienced with the design process, it's possible to infer that *this transformer is only useful for vacuum tubes that have an output impedance of 7,600 ohms*. (And in fact, the transformer may be optimized for performance at this impedance but this need not be a limitation.)

As we have learned in previous chapters, the output impedances of vacuum tubes in an amplifier are not fixed, they are a moving target. Main contributors that define the impedance of a tube are Eb (plate voltage) and Imax (plate current at 60% Eb) and those easily change with screen grid voltage variation. So how useful is the "impedance" and "suggested tube types" in the catalogue description? Not useful for a designer. We utilize a transformer for one simple purpose: to "match" one impedance to a different impedance.

Example: assume that one has selected a speaker configuration that is equivalent to 2 ohms (four eight ohm speakers in parallel, for example) and that it

is desired to drive these speakers with an amplifier that has a calculated output impedance of about 2k (2,000 ohms), how would an output transformer be selected?

Unless the catalogue specifically stated a primary impedance of 2k and a secondary impedance of 2 ohms, which is unlikely, then as we've already learned we would have to calculate the turns ratio of the two impedances:

$$N_{in} / N_{out} = \text{turns ratio} = (R_{in} / R_{out})^{0.5}$$

(or) we could re-arrange the expression to give:

$$R_{in} / R_{out} = (\text{turns ratio})^2$$

In this case, we'll use the first form of the equation and calculate the following

$$\text{Turns ratio} = (2000 / 2)^{0.5} = 31.62$$

Now how do we apply this information to selecting a standard product from the catalog? Unfortunately it's not easy ... we have to extract the turns ratio from the sparse information that the manufacturer has provided.

Given the example above (2000 ohms primary transformer with 2 ohm secondary impedance required), we can use the above equations to determine the turns ratio for a transformer of 7600 ohms with multiple output taps.

Making a table of the transformer characteristics recalling that:

$$\text{turns ratio} = (R_{in} / R_{out})^{0.5}$$

<u>Primary</u>	<u>Secondary</u>	<u>Turns Ratio</u>
7600	4	43.589
7600	8	30.822
7600	16	21.794

Now that turns ratio is clear - and it's mostly independent of impedance - we see that the catalog transformer will work fine with the secondary 2 ohm load connected to the 8 ohm terminal of the transformer. That results in a turns ratio of 30.822 which fits our design requirement of 31.62 very well.

We need to apply common sense to our estimates, calculations and decisions about these things. Most of the inexpensive components that we purchase and use have a tolerance of +/- 10% on their values. So it's appropriate to allow that same tolerance in our initial design goals. (Iteration of these goals is always possible if the final performance calculations are unsatisfactory.)

I've made a spread sheet that calculates turns ratio of a number of common commercially available transformers from their catalog description. The information will form a part of this discussion and simplify the process of selecting an output transformer. This data is included in the Excel Workbook that can be downloaded along with this book whenever the two are referenced and linked on the internet. (There is no other source for this book - it is not commercially available.)

The utility of using turns ratio, rather than impedances, will also become obvious if one desires to use more than one speaker. When the resulting parallel or series combinations of speaker impedances is a non-standard value, then manufacturer's data sheets become less useful and turns ratio become even more important.

As discussed in an earlier chapter, the more turns on a transformer (higher turns ratio), the more resistance is introduced and the lower the efficiency. Considering these disadvantages, it is suggested that loudspeaker impedance be as high as possible provided that performance is not impaired. This permits a lower transformer turns ratio (which suggests lower resistance and higher efficiency).

As with all design engineering, tradeoffs are possible and case-by-case analysis is recommended. In this example, a higher loudspeaker impedance implies more coil windings, greater coil mass, reduced high frequency response and lower power rating - all other parameters being equal.

8.14 Input Drive Level, Estimate for Voltage Gain

This is a critical piece of information; we must know the peak to peak voltage at the input of the power stage that will drive it to full output power. (This is another way of expressing the voltage gain of the power stage if the output impedance is taken into account.) We can't determine the characteristics of the stages of amplification preceding the power amplifier until we have a good idea of the power amplifier gain.

One can usually obtain this information from the tube data sheet in the chapter that describes push-pull class AB1 configuration. BUT that information is available *only from data sheets of tubes that are commonly used as audio amplifiers*. If one follows the path that I've suggested here (i.e. ignoring the high-cost tubes normally used in guitar amplifiers in favor of better quality, lower cost tubes still available in large surplus quantities) then one needs a simple way to approximate the gain of the output tubes.

If a few assumptions are made, an approximation for voltage gain (A_v) could be:

$$A_v = 2 \times E_o / E_{c1}$$

where E_{c1} is the grid 1 voltage required to bias the tube at I_q . E_{c1} has been previously determined to be -23 volts, so, if we insert the known E_o (206 volts) and E_{c1} (23 volts) then

$$A_v = 2 \times 206 / 23 = 17.913$$

Lacking any better information at this time, we can insert this value into the characteristics of our power stage, with the realization that this gain figure is mostly for planning purposes. (For example, in an industrial situation if another designer was assigned to work on the driver stage while you were working on the power amplifier, he'd have to have a target value of gain for his design.) We'll look at another way to estimate the gain later, using the plate curves.

8.15 Biasing Power Tubes, Screen Bias

The most common and simplest way to bias the screen grid (grid 2), is to use a resistor. The screen voltage must **ALWAYS** be lower than the plate and this proviso suggests that the use of a resistor to bias the screen might be appropriate (more on this later). The value of the resistor can be determined by:

$$R_{screen} = (E_b - E_{c2}) / I_{c2}$$

where E_b is plate voltage, E_{c2} is screen grid voltage and I_{c2} is screen grid current. Replacing the symbols with the values determined above:

$$R_{screen} = (286 - 150) / .0018 = 75,555 \text{ or } 75\text{k ohms} \text{ (closest standard value)}$$

We need to determine the *power rating* of the resistor, this is given by either

$$P_{screen} = I_{c2}^2 \times R_{screen} \text{ or}$$

$$P_{screen} = (E_b - E_{c2})^2 / R_{screen} \text{ or}$$

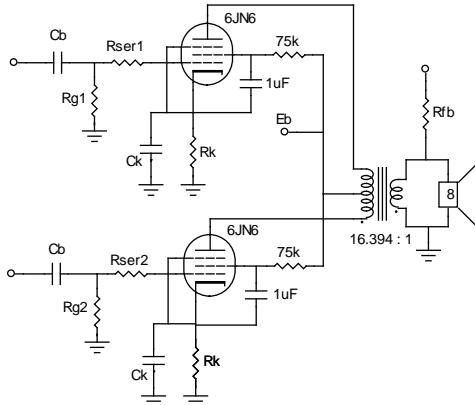
$$P_{screen} = (E_b - E_{c2}) \times I_{c2}$$

Any of the above equations will produce the same result. For simplicity, we'll use the last one and replace the symbols with the known values:

$$P_{screen} = (286 - 150) \times .0018 = .245 \text{ watts}$$

Resistors must always be de-rated which means they need to be operated with safety margin. This is necessary for long-term stability as well as for reliability. The general rule is to select a power rating that is about twice as high as the calculated dissipation. For the screen resistor we just calculated, normal practice

would suggest selecting a 1/2 watt (.50 watt) resistor. We can add the screen resistor value to our table and update the original schematic:



Note that the screen grids have been "bypassed" by connecting a capacitor between screen grid and cathode of both tubes. The capacitor provides a low impedance path from screen grid to cathode - if this isn't provided, the screen grid will behave like a smaller version of the plate, it will dissipate not only the bias voltage and current but the signal power will be added to the quiescent dissipation. Performance will be degraded and so will reliability.

If the screen bias resistors are small, less than 3k or so, then these capacitors are not necessary. A value between 1 and 10 uF is adequate to perform the bypass function, generally. The capacitor may be connected to ground instead of the cathode, if it's more convenient to do so but in some cases this may cause oscillation.

An exact determination of the screen grid bypass capacitance can be made by calculating the impedance of the circuit at the screen grid and selecting the bypass capacitor to be equal to that impedance at the lowest desired frequency to be reproduced. This entails collecting some data that is not always readily available (i.e. triode mode operating plate current) so it's useful to base the selection of the bypass capacitor on the rule of thumb described above.

If one does have access to the triode mode data, then by subtracting the plate current from the total cathode current, the screen current results. The effective screen resistance can be approximated from

$$R_{\text{eff}} = E_{\text{screen}} / I_{\text{screen}}$$

And this value, in parallel with the bias resistor network, is the total impedance at the screen grid, R_{total} . The value of the bypass capacitor, for a frequency of 60 Hz (lowest guitar frequency is 80 Hz) is then

$$C_{\text{bypass}} = 1 / (2 \times \pi \times 60 \times R_{\text{total}})$$

When one is attempting to squeeze every watt of power from an amplifier, especially a custom-designed, one-off item, it's common to optimize the screen grid bias voltage at test. The plate current and transconductance are quite sensitive to screen grid voltage and the match between power tubes and load can also be improved by varying this voltage.

If one chooses to do this, it's necessary to monitor the screen grid current while varying the voltage. Once the optimum value of bias has been established, confirm that screen grid dissipation limits have not been exceeded by multiplying screen grid voltage times screen grid current. Consult the data sheet for maximum screen dissipation rating.

8.16 Biasing the Power Tubes, Control Grid and Cathode

Earlier we determined that a grid voltage (E_{c1}) of about -23 volts will produce the desired quiescent current (I_q) of 59 mA for each output tube. There are several ways of providing bias for the output tubes:

Providing a negative adjustable voltage to the control grid that can be set to about -23 volts

Using a cathode bias resistor

A combination of the two methods

8.17 Biased Grid Configuration

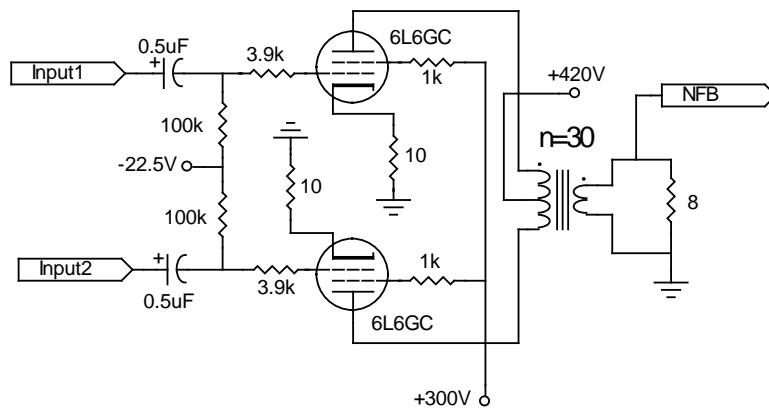
There are advantages and disadvantages for each technique. If a negative voltage can be made available to set the grids of the tubes to -23 volts, then the cathodes of the tubes can be grounded. This condition is the best one for maximum power gain and maximum output power. This configuration is also the most efficient one for power supply design and transformer selection. The power supply design needs only to produce the plate voltage of the power amplifier tubes. As we'll see later, cathode bias configurations require the power supply to have higher available voltage and be able to withstand higher electrical stress.

Disadvantages: this configuration requires "matching" the two output tubes when they need to be changed. (Matching means that the output tubes are selected so that their performance characteristics are as identical as possible.) Each time the tubes are changed, the negative grid bias voltage must be re-adjusted so that both tubes are operating at the same 59 mA of plate current.

Depending upon the way the circuit is configured, measurement of plate current can be very difficult and even inaccurate. Additionally, limitations in the power

supply design frequently suggest the use of a grid bias voltage - grounded cathode - configuration.

If the grid bias technique is desired, then an adjustable negative voltage needs to be designed into the power supply. We'll cover that later in the discussion on power supply design. Here's an *example schematic* depicting a power amplifier with grounded cathodes. This example uses a small amount of cathode resistance to stabilize the gain in the stage and permit measuring the current through each of the tubes (by measuring the voltage drop across the 10 ohm resistors):



Note that - even in the grounded cathode, negative biased grid - configuration it's always desirable to include a resistor in each cathode circuit. For the least amount of power loss, a 1 ohm, 1/4 watt, 1 % tolerance part is recommended. This enables accurate determination of the cathode current in each tube by measurement of the voltage drop across the 1 ohm resistors. The voltage drop measured will be exactly equal to the current flow (e.g. a measurement of .043 volts drop means that .043 amperes of current is flowing through the 1 ohm resistor).

Also, the above schematic indicates a single negative bias point for both control grids. This is not uncommon, especially in older amplifiers that used reliable, repeatable vacuum tubes. Our currently available tubes are not so consistent and it is recommended that each of the control grids be biased independently. The cathode currents in each tube can be then be adjusted so that they are equal, resulting in minimal distortion. In the later chapter on power supply design, a negative supply will be illustrated with the provision for adjusting the voltage for each control grid.

No screen grid bypass capacitors are included in the above schematic since the screen grid bias resistors are fairly small (they are less than 3k, which was the general limit that we set for unbypassed screen grids).

8.18 Biased Cathode Configuration

Using cathode resistors to develop the required bias voltage from grid to cathode provides some important advantages. Most significant is the fact that this approach adds a lot of low-frequency feedback to the output amplifier. In practice, what this means is that the output tubes don't have to be carefully matched, in fact when using older, higher quality tubes, no matching is necessary at all.

Another major advantage is that the bias current of each tube can be easily and accurately measured. Although this may not be very important after the amplifier has been designed and built, it is vitally important during testing so that all aspects of tube operation - especially power dissipation - can be confirmed.

This configuration does add to the parts count of the output stage, it requires two more resistors and possibly two more capacitors. The purpose of the capacitors will be discussed in a moment.

If cathode resistors are to be used, we need to calculate both their value and their power dissipation. But first we need to understand how a cathode resistor can provide the same function as applying a negative voltage to the grid of the tubes when the cathode is grounded.

The tube doesn't really "care" whether voltages are "positive" or "negative" when they are referenced to ground. What makes the tubes operate properly is *the polarity of grids and plate with respect to the cathode, not with respect to ground.*

For normal operation, the grid is always negative with respect to the cathode and the converse is true, the screen grid is more positive than the cathode and the plate is always positive with respect to ALL other tube connections. We can make the grid more negative than the cathode by using the plate current (which must also pass through the cathode) to generate a voltage at the cathode that is positive with respect to "ground". If we then ground the grid (through a high value resistor so that signal voltage is not reduced) then the grid will be negative *with respect to the cathode.*

It can be observed in many older schematics that the two cathodes were frequently connected and a single, common cathode resistor provided the bias resistance for both tubes. This was common and effective, given the fact that tubes were more consistent performers in past days. A better solution, for tubes with inconsistent parameters, is to separate the bias resistance and provide a resistor for each tube cathode, we'll modify our schematic accordingly.

To make that happen, we pass the plate current through a resistor that is connected from cathode to ground. Current passing through a resistor causes a voltage to develop across the resistor (it's usually called a voltage "drop") in

accordance with "Ohm's Law", which relates voltage, current and resistance in a simple form:

$$I = E / R$$

where I is current, E is voltage and R is resistance expressed in consistent terms.

If we select the resistor so that the voltage across it is exactly equal to the grid voltage (E_{c1}) required for desired plate current to flow, then we've satisfied the bias conditions and the correct amount of plate current will flow through the tube. Here's how to do that:

$$R_{cathode} = E_{c1} / (I_q + I_{b2})$$

where E_{c1} is the grid voltage required, I_q is the quiescent plate current and I_{b2} is the screen grid current, both of these currents have to flow through the cathode. Substituting known values for the symbols, we get

$$R_{cathode} = 23 / (.059 + .0018) = 378 \text{ ohms}$$

Let's change this to 360 ohms in order to accommodate the standard resistance values that are normally stocked. (We don't want to make too big a change to this resistor since it will determine the plate current, which affects other performance parameters.)

We can use any of the following to calculate the power dissipation of the resistor:

$$P = I_{c2}^2 \times R_{cathode} \text{ or}$$

$$P = (E_{c1})^2 / R_{cathode} \text{ or}$$

$$P = (E_{c1}) \times (I_q + I_{b2})$$

We can (arbitrarily) use the second formula to obtain power dissipation, substituting values for symbols:

$$P_{cathode} = (23)^2 / 360 = 1.47 \text{ watts}$$

As noted previously, we de-rate this resistor by selecting one with the same resistance and about twice the calculated power rating. Commercial resistors rated at 3 watts would be an appropriate choice.

8.19 Purpose and Selection of Cathode Capacitors

If the cathode resistors are small enough (in resistance - not size), they can be connected to the cathode and grounded ... done. If they are high enough

resistance to be an appreciable fraction of the plate impedance, then they affect the audio performance of the circuit. (This is because they form a series feedback circuit which tends to enhance bandwidth while reducing gain. Bandwidth is almost never a problem in guitar amplifiers, recall that we usually want to limit bandwidth, for noise considerations.)

What constitutes "an appreciable fraction of the plate impedance"? We can use an earlier approximation that we made for voltage gain, Av and the calculated plate impedance, Rpp, to determine when we need to add a "bypass" capacitor to our cathode resistor. If the ratio of plate impedance to cathode resistor is significantly less than the estimated gain, then a cathode bypass capacitor is recommended. Here's an expression that establishes the relationship:

If $R_{pp} / R_{cathode} < Av \times 3$ then use a capacitor

and inserting known values and solving:

$$2150 / 360 = 5.971 \text{ and } Av \times 3 = 17.913 \times 3 = 53.74$$

so obviously we need to add a bypass.

"Bypass" refers to installing a capacitor across the terminals of the cathode resistor so that the audio signal flows through the capacitor, thus "bypassing" the resistor. This effectively "grounds" the cathode for audio signals, although not for bias voltages and currents. The value of the capacitor is chosen as a function of the lowest frequency that we want the amplifier to reproduce, usually around 80 Hz (low "E" on the guitar, 40 Hz lower for bass). The two things that we need to determine are the value of the capacitor and the working voltage of the capacitor. Here's how to calculate the capacitor value:

$$C_{cathode} = 1 / \{2 \times \pi \times f \times [R_{pp} / (3 \times Av)]\}$$

where π is about 3.14, f is the desired low frequency (80 Hz), Av is the estimated voltage gain and Rpp is the calculated plate to plate resistance previously determined. Substituting our known values into the equation, we get:

$$C_{cathode} = 1 / [2 \times \pi \times 80 \times 2150 / (3 \times 17.913)]$$

$$= 1 / (20,110) = 49.73 (10^{-6}) \text{ farads (use the next larger standard value of } 51 \mu\text{F})$$

The working voltage of the capacitor is the voltage across R_{cathode}, or 23 volts, as we've previously discussed. But the capacitor voltage must be de-rated and the typical procedure for these low-voltage units is to double the operating voltage, so we would select a capacitor rated for at least 50 volts.

In areas of the circuit where very high voltages are present, cost considerations or size constraints might indicate less safety margin in the choice of the capacitor working voltage. Under NO circumstances should a capacitor with a working voltage equal to or less than the circuit voltage be used. When a capacitor fails due to overvoltage, the result is rather spectacular and dangerous.

The absolute minimum safety margin would be about 25% or 1-1/4 times the circuit voltage present. Further, note that most capacitors over the value of around 1 microfarad are polarized. That means that, like a battery, they must be connected so that the "positive" marking on the capacitor is connected to a point in the circuit that is more positive than the point to which the other lead of the capacitor is connected.

Before continuing to the next topic, it's worth noting that, as the remaining stages are designed, other choices of component values will need to be made that also affect frequency bandwidth. If we select the lower "cutoff" frequency to be 80 Hz for all of the other stages, the process of cascading the stages will modify the bandwidth (both lower and upper frequency limits). So we usually pick values that provide a little more bandwidth than we actually need, to allow for the gradual degradation.

We won't worry overmuch about the upper frequency limitations of the amplifier because the loudspeaker will be the dominant "filter" that determines maximum audible frequency. But the selection of the low frequency "cutoff" frequency is of some importance. If we make this frequency too low, we get more 60 Hz power supply ripple effect. If the frequency is set too high, we lose the authority of the bass notes of our instrument and get an effect that sounds like we are using too small a loudspeaker.

These effects are more pronounced as we work backward toward the preamplifier stages because of the much greater gains involved. For now, we'll keep our cathode capacitor at the value calculated but be aware that we may want to lower the cutoff frequencies slightly in the preceding stage amplifier designs.

8.20 Effect of Cathode Bias Resistors On Plate Voltage

In order of descending voltages, here are the voltage relationships for each of the beam pentode connections:

- Plate - most positive
- Screen grid 2 - next most positive
- Repellor grid 3 - connected to cathode
- Cathode - next least positive
- Control grid 1 - least positive

Note that, although we universally refer to the control grid voltage as "negative", it's also completely appropriate to consider the control grid voltage as "less positive", with respect to the other electrodes (connections) of the tube.

While there are many advantages to using cathode resistors to "bias" the output tubes properly, we must confront the major disadvantage. During our previous discussions, particularly those concerned with any parameter associated with plate voltage (E_b or E_o), we assumed that the cathode was grounded.

Thus, any calculation or discussion regarding the influence of E_b was actually referring to the plate-to-cathode voltage. Up to this point, we always made the assumption that the cathode was at 0 volts (grounded).

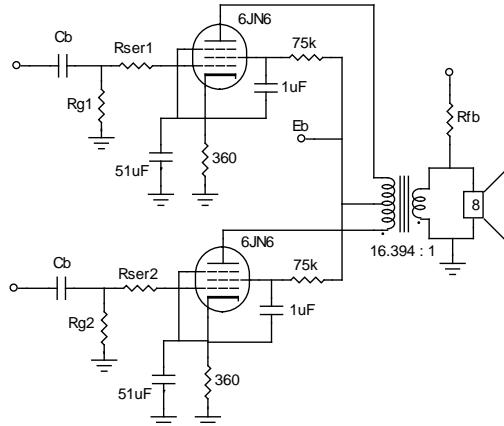
The addition of cathode resistors changes the relationship of plate voltage to cathode. We discussed the relationship of grid to cathode, stressing *that the polarity of the voltages with respect to ground were not significant*. The polarities and magnitudes of the voltages, with *respect to the terminals (and internal functions) of the vacuum tube ARE of significance*. That's why the voltage relationships were described in the first paragraph of this chapter.

Plate voltage, as it affects all characteristics of vacuum tube performance, is properly referenced to the cathode, not to the "ground" potential of zero volts, when cathode bias resistors are used. The cathode resistors that we included previously raised the cathode potential from zero ("ground" potential) to 23 volts, the value of grid to cathode voltage required to operate each tube at 59 mA quiescent current (I_q).

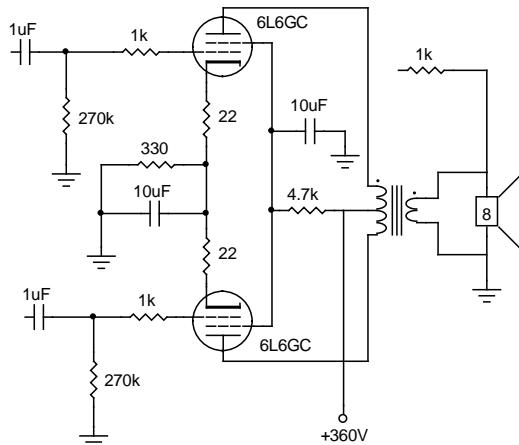
Taking into account the voltage drop across the cathode resistor, the plate to cathode voltage isn't 286 volts, it's 263 volts ($E_b - E_{c1}$). *This is a significant change and might affect output power, output impedance, transformer selection, voltage gain, etc, all adversely.*

Fortunately the solution is simple: we add the voltage lost in the cathode resistors back to the plate voltage so that the "new" plate voltage is now 286 + 23 or about 309 volts. Since we haven't designed the power supply, nothing is really affected - a "paper" change only. (But there is an obvious inference, suggested previously: the design process is iterative, changes in one minor aspect of the circuit ripple throughout the entire circuit. At the point where the power supply is being designed, cost or availability considerations might suggest revising the power supply voltage.)

We don't have to change the value of the screen resistor since the relationship between the plate voltage and screen voltage hasn't changed - they are still the same fixed voltage apart. Here's the new schematic, reflecting the addition of cathode bypass capacitors:



In past days, as we've noted previously, the practice was to connect the two cathodes together and then connect them to ground through a resistor of 1/2 the resistance of a single cathode resistor. This allows higher gain at the expense of bias stability. A good compromise might be to use three resistors, as shown in the circuit below, for cathode bias:



The value of the two un-bypassed resistors must not be too large or else gain will be lost. The advantage is the provision of some D.C. feedback so that unmatched output tubes will share current more equally than if a single cathode resistor, common to both cathodes, was used. (Design values will depend on the anticipated variation in output tube transconductance.)

8.21 Selection of Grid Resistors

This is a simple procedure and requires no calculation. The vacuum tube is a very high impedance device, so high that almost *no current flows in the grid*. We note that the grid must be negative with respect to the cathode for normal operation. (If we've used cathode resistors to make the cathode more positive with respect to ground, then nothing more needs to be done to the grid other than to "ground" it through a high value resistor.

Values from around 10k up to 1Meg ohm are commonly employed for this purpose, depending upon other circuit requirements (e.g. the previous circuit, the one that drives the output amplifier). For non-critical applications, a relatively high resistance may be used without particular regard to specific value. Most power tube data sheets will specify a maximum amount of grid resistance.

There is a practical limit for the value of this resistance - because the internal elements of the vacuum tube aren't operating in a perfect vacuum, impurities cause a small amount of current flow in the grid resistor. If the value of the grid resistor is too high, it's possible for a slight positive bias to develop. This leads to reliability issues due to tube overheating. The performance of the tube would be expected to be degraded as well.

According to the data sheet for our selected 6JN6 tube, the maximum value of grid resistance is 1 Megohm. We can pick a value of say 100k ohms, although higher values would be fine too. The selection should be made with some regard to the circuit that drives this stage. A typical plate resistor for a phase splitter (a typical driving circuit for an output amplifier stage) would probably not be greater than 100k so a grid resistor with a value lower than this wouldn't be beneficial for proper circuit function. Power dissipation is not critical, so a standard 1/4 watt resistor is appropriate.

There is a tendency for beam power tubes, like our output tubes, to "motorboat", which is a term used to describe low frequency oscillations that sometimes occur. Another undesirable characteristic is for the tubes to receive and reproduce radio signals from powerful transmitters that may be nearby (passing emergency vehicles and the like). These adverse aspects can usually be suppressed or eliminated by adding a series resistor to the grid of the tube.

This resistor is connected between the grid and the grounding shunt resistor, 100k in our case. The value is not particularly critical and is usually between one hundred ohms up to several k ohms. (Sometimes the value must be adjusted when the amplifier is being tested and a problem is observed.) Let's use a starting value of 1 k for our amplifier, at least for now. The power dissipation is not important because no appreciable current flows, a standard 1/4 watt value is acceptable.

(As a matter of interest, the addition of a series resistor to the control grid forms a lowpass filter that suppresses high frequency signals (that may inadvertently enter the amplifier chassis) to acceptable levels. That's because internal (and unintentional) characteristics of the tube result in a "capacitor" formed between the control grid and the cathode. A series resistor followed by a shunt capacitor is a lowpass filter. "Shunt" means a connection to ground.)

Since the input (grid) impedance of the output tubes is very high, the 100k resistors that are connected between the grids and ground *now establish the*

input impedance of each side of the amplifier. This should be added to our table describing the output stage. Although not of importance at this time, the input impedance of the stage will be necessary to complete the design of the *previous* stage.

There is one more situation in which the choice of the grid resistors is fairly critical. This situation occurs in the uncommon situation where an interstage transformer is used to couple a single-ended stage to a push-pull stage. For effective power transfer and predictable voltage exchange, both the source and load impedances must be established. At these low frequencies (audio) this is universally accomplished by the selection of appropriate plate and grid resistors.

An example of this particular configuration is depicted in chapter 19.1. In that chapter, a Fender "Champ" amplifier is modified to achieve the performance of a Fender "Princeton" amplifier. A transformer is used to perform the "phase-splitter" function as well as provide a slight amount of voltage gain. The plate resistor of the driver stage (the stage following the preamplifier) establishes the source impedance while grid resistors of the output stage complete the proper impedance and voltage transformation.

8.22 Blocking Capacitors

Now that we've defined the input impedance of the power stage, we can calculate the value of the input capacitors. These capacitors are necessary to block the high plate voltage (the previous stage plate voltage) from the control grids of the power stage. The procedure is the same as for determining cathode bypass capacitors except that we'll select the capacitor impedance to be proportional to the combined impedance of the input stage resistance and the previous stage output resistance.

An easy way of doing this is to pick the lowest of the two resistances (plate resistance of preceding stage or grid resistance of this stage) and determine a capacitance based on a ratio of this value of resistance.

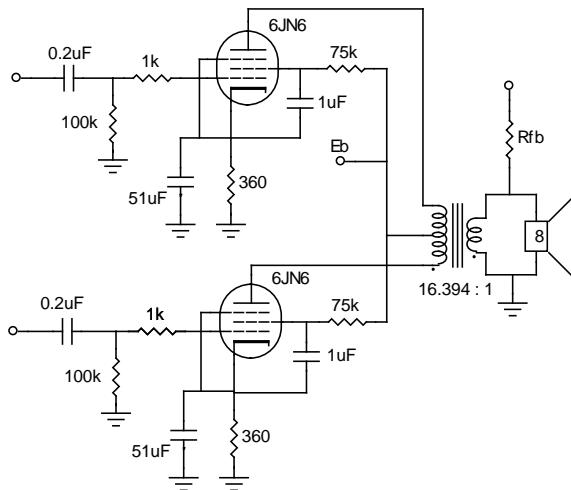
$$\text{Cblock} = 1 / [2 \times \pi \times f \times (R_{\min} / 2)]$$

Where π is about 3.14, f is the desired low frequency (80 Hz) and R_{\min} is the lowest of either grid shunt resistor value selected above or the plate resistor of the preceding stage. (We will arbitrarily divide this resistance by 2 to insure that the capacitor impedance is low enough.) If we assume that the preceding stage has a plate resistance of 22k, then clearly that value is lower than the 100k grid resistors, so substituting known values into the equation, we get:

$$\text{Cblock} = 1 / [2 \times \pi \times 80 \times (22,000 / 2)] = 1 / 12566371$$

$$= 1.809 \text{ (10-7) farads or } \mathbf{0.1809 \text{ microfarads}}$$

Use the next larger standard value of 0.2 microfarads, which will be the design value. The only remaining value left to be defined is the feedback resistance, R_{fb} . We will discuss this in a later chapter regarding feedback. Here's a schematic representation of our completed power amplifier output stage:



8.23 Graphical Estimates For Gain and Power

We've previously estimated the output stage gain at around 18, using an approximation. We can use the plate curves to make a more accurate estimation for the voltage gain and check the output power estimates at the same time. For simplicity, the following exercise will be performed for a single tube.

Referring back to the "plate characteristics" curves for our 6JN6 vacuum tube, let's recall that our operating conditions include plate to cathode voltage of 286 volts and quiescent current (I_q) of 59 mA. These are the conditions a single tube is biased under *no signal conditions*. Once an audio signal is introduced, at a level that can drive the output tubes to full power, these quiescent conditions are inapplicable. The output voltages and currents can vary (at the frequency of the input signal) as follows:

Plate voltage varies at TWICE E_o (E_o is the signal voltage swing of the plate)

Plate current varies at TWICE the signal plate current

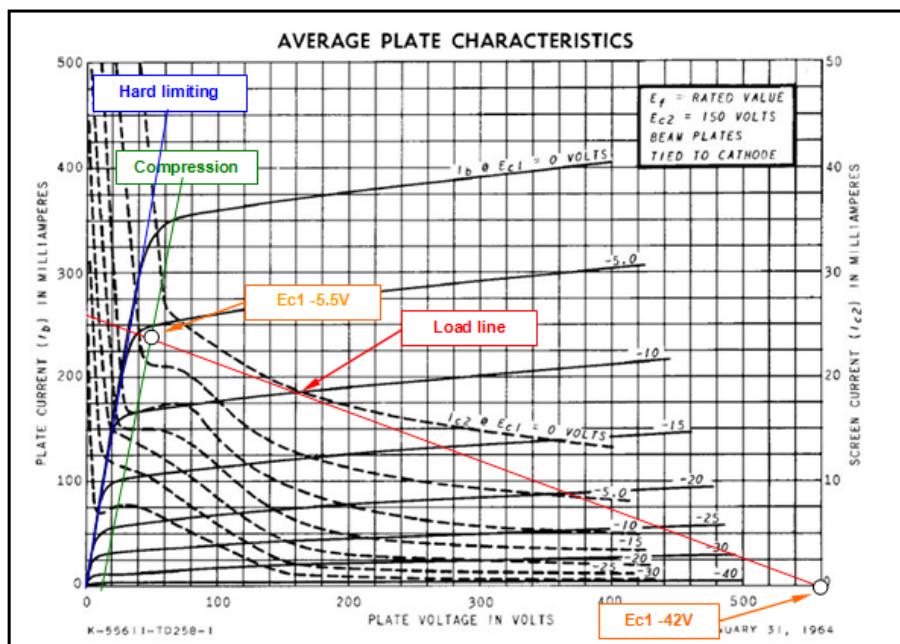
Considering this graphically, let's make a point on the horizontal axis that represents TWICE the plate to cathode voltage of 286 volts, which is 572 volts. On the vertical axis, let's make a point that establishes the load line. The load resistance is 2150 ohms, solving for I where

$$I = E / R = 572 / 2150 = 0.266 \text{ amperes}$$

Draw a load line through the two points at 572 volts and 0.266 amperes.

Although we covered compression earlier, reviewing the subject might be helpful for this exercise. Examining the grid voltage curves (E_{c1}) we see that, as the plate voltage gets lower, the grid voltage curves start to "bunch up" and then blend into a single line that is almost vertical. This is a graphic indication that the tube cannot operate in a linear manner when plate voltages are too low. As the grid curves begin to "bunch up", *that's an indication of compression*. When the grid curves have blended into a single line, *the tube is in hard limiting*.

NOTE: *Using the plate curves in this manner approximates the behavior of a single output. A more rigorous and accurate procedure would involve the analysis of two sets of plate curves combined (one being a mirror image of the other) representing the push-pull circuit operation of the output tubes.*



One can easily draw a line denoting the area where compression starts, by observing where the plate curves commence to deviate from a straight line. We can add a point to the load line (at $E_b = 50$ volts). Adding a point at the upper limit of 572 volts establishes the minimum and maximum limits of linear operation. We can also note that the grid voltage, E_{c1} , for these two conditions is -5.5 volts and -42 volts.

The audio voltage swing obtainable from a *perfect vacuum tube* would be twice the normal plate voltage, E_b , or about 572 volts. However for the tube that we've chosen and operated as we've determined, the maximum voltage swing is about 572 volts - 50 volts, or 522 volts.

This is important because it means that *the full supply voltage is not available to the tube, limiting both the output power and the gain.*

Given the established points, we can refine our predictions of gain and output power. First, for the two limits of plate voltage swing, I've noted the grid voltages (E_{c1}) associated with those plate voltages. The grid voltages are -5.5 volts and -42 volts. We can make a gain prediction using plate and grid voltages:

$$Av = (Eb \text{ max} - Eb \text{ min}) / (Ec1 \text{ max} - Ec1 \text{ min})$$

Which states that voltage gain is equal to the difference in output plate voltages divided by the difference in input grid voltages. Substituting known values and solving:

$$Av = (572 - 50) / (-42 - -5.5) = 522 / 36.5 = 14.3$$

Now let's use the plate voltage swing of 522 volts to predict the output power, using the transformer turns ratio and speaker impedance. The impedance of the speaker, transformed back to the plate of the output tubes is:

$$\text{Plate load} = R \text{ speaker} \times (N \text{ input} / N \text{ output})^2$$

substituting the speaker impedance and the transformer turns ratio we get:

$$\text{Plate load} = 8 \text{ ohms} \times (16.394)^2 = 2150 \text{ ohms}$$

Power can be determined by the expression:

$$P = E^2 / R$$

where E is the average signal voltage and R is load resistance of 2150 ohms. The average voltage is peak-to-peak voltage times .318 (or peak voltage times .636), so the average signal voltage is $.318 \times 522 = 165.996$ volts and substituting this into the power equation:

$$P = (165.996)^2 / 2150 = 34,147 / 2150 = 12.816 \text{ watts for a single tube}$$

double that for the entire output stage or **25.632 watts**.

Checking power dissipation for the output tubes, multiplying the average current times the nominal plate to cathode voltage gives

$$P_{\text{diss}} = 286 \text{ volts} \times (.637 \times .266) = 48.46 \text{ watts}$$

From which we subtract the output power since it will be dissipated *in the load*, not in the tube:

$$48.46 - 25.632 = \mathbf{22.828 \text{ watts}} \text{ for both tubes}$$

The dissipation for a single tube is one-half that value or over 11 watts, well within the rating of 17.5 watts for the individual tubes. Both quiescent and full-drive power dissipation are within the dissipation limits specified on the manufacturer's data sheet.

As a matter of interest, the efficiency of the output amplifier can be determined by dividing the output power by the total power dissipation which for this example:

$$\mathbf{\text{Efficiency} = P_{output} / P_{diss\ total} = 25.632 / 48.46 = 0.529 \text{ or } 52.9\%}$$

This is near the classical and common efficiency of 50% for a properly designed push-pull Class AB output power amplifier. If the calculated efficiency deviated more than say 10% from the normal 50% efficiency, it would be an indication that the design is flawed and the design process should be reviewed.

A last observation, regarding the particular pentodes selected for this application, it should be apparent at this point that these tubes would be appropriate for a power amplifier design of around 35 watts.

8.24 Output Stage Characteristics, Final

Input impedance: 100k, each input

Voltage gain: 14.3 (excluding transformer)

Output power: 25 watts minimum

R_{pp} : 2150 ohms

Plate voltage (E_o or E_b) required: 286 volts plate to cathode, 309 volts total*

Control grid (E_{c1}) voltage: -23 volts, grid to cathode, cathode is at +23 volts*

Screen grid voltage (E_{c2}) required: 150 volts, grid 1 to cathode, 173 volts total*

Screen grid (I_{c2}) current: 1.8 mA

Screen resistor: 75k 1/2 watt, single resistor bias (or voltage divider bias)

Grid 3 is connected to cathode

Cathode resistor: 360 ohms 3 watts

Cathode capacitor: 51 uF 50 volts

Grid series resistors: 1k 1/4 watt

Grid shunt resistors: 100k 1/4 watt

Maximum current required: 118 mA (average)

Maximum tube dissipation: less than tube rating

Speaker impedance: 8 ohms

Transformer turns ratio: 16.394

Filaments: 6.3 VAC @ 1.3 amperes each

Speaker	SPKR-123	1 each	\$95	\$95 total
---------	----------	--------	------	------------

Transformer	TRF-456	1 each	\$85	\$85 total
-------------	---------	--------	------	------------

Vac Tube	6JN6	2 each	\$5	\$10 total
----------	------	--------	-----	------------

Tube socket	SKT789	2 each	\$6	\$12 total
-------------	--------	--------	-----	------------

Screen resistors		2 each	\$0.10	\$0.20 total
------------------	--	--------	--------	--------------

Cathode resistors	2 each	\$1.50	\$3.00 total
Cathode capacitors	2 each	\$0.75	\$1.50 total
Grid resistors	4 each	\$0.10	\$0.40 total

* Referenced to ground.

8.25 Reliability and Performance Implications of Vacuum Tube Biasing

It's appropriate to point out at this time that much of the earlier knowledge regarding design, operation and reliability considerations of vacuum tubes has been lost. Unless one has access to historical technical literature, it's unlikely that the majority of us will ever have an appreciable amount of insight that a design engineer of the 1930s and 1940s possessed. What was routine practice to them - derived from experience - almost verges on witchcraft to those of us who do not possess the experience and knowledge of those engineers.

I must note that there is no implication that those designers were more capable than engineers of our time, actually the converse is true. Back in "the day" one needed to know about only one active device: the vacuum tube. Today's engineer must encompass enough practical knowledge to be able to design circuits using any combination of scores of different solid-state devices. And this doesn't mean different types of the same basic architecture of semiconductor, this means literally, physically and functionally different devices.

An engineer of average experience and education could probably seat himself/herself at a desk in Nutley, New Jersey (RCA in the 1930s) and be up to speed in a few weeks. An engineer of that period could not assume a design position in a modern electronics firm without a considerable amount of additional education. (There is no intent to suggest pejorative comparison; it's just the way that things are.)

In the normally available literature, very little can be found regarding reliability of vacuum tubes or how long they could be expected to last in service when compared with critical operating parameters. (Unlike semiconductors, whose limitations are well-known, well-documented and with well-established procedures for estimating their lifetime.)

We need to exercise common sense when considering some of these parameters:

- Plate voltage (E_b), maximum
- Plate current (I_b), maximum
- Plate dissipation (P_b), maximum

- Screen voltage (E_{c2}), maximum

Screen current (I_{c2}), maximum
Screen dissipation (P_{c2}), maximum

Control grid voltage (E_{c1}), maximum
Control grid voltage (E_{c1}), minimum

Interaction of these parameters probably isn't understood by most of us. Questions about vacuum tube operating conditions arise when schematics of musical instrument amplifiers are examined.

Excessive plate voltage: guitar amplifier manufacturers universally operate(d) tube plates *in excess* of the tube manufacturer's maximum rating.

Excessive screen grid voltage: amplifier manufacturers universally operate(d) screen grids *greatly in excess* of the tube manufacturer's maximum rating.

Screen grid bias method: amplifier manufacturers frequently bias screen grids in a manner that is not recommended by vacuum tube manufacturers.

Guitar amplifiers, as we mention often, are driven mostly by cost, rarely by performance *with the exception of output power*. A major cost indicator might be "tube count": the number of tubes in a given circuit. (At one time, this was also a selling point. Higher number of tubes was somehow interpreted as an advantage, despite the obvious fact that tubes needed to be replaced more often.) It should be intuitive that a cost-effective design is one with a low tube count. Manufacturers usually push performance as close to the envelope as possible to use the minimum amount of tubes and parts (which reduces assembly time as well as material cost).

An example of a common overstressed condition is the practice of operating the screen grid at the plate voltage. The main reason for doing this is because tube transconductance (and therefore gain) increases as screen grid voltage increases. It's desirable to obtain as much gain as possible from the output power tubes, reducing the amount of gain required from previous stages and perhaps thereby eliminating a tube.

There is no documented history explaining why consistent, deliberate overstressing occurred beyond the obvious points. My supposition is that the amplifier manufacturers made a conscious decision to trade off performance (in the form of output power) with reliability (the frequency that output tubes needed to be replaced). After all, during the 1950 era to the current period, it can be demonstrated that, if all other parameters are equal, a consumer will almost *always* choose the amplifier with the highest power rating.

Questioning the original design philosophy is pointless, it is what it is. Fifty years of utilizing this ancient technology successfully supports the early decisions of the amplifier marketing/management decision makers. The real question, buried in this history lesson, is this: what reliability criteria should we - designers of *personal* amplification equipment - be following?

I don't feel comfortable about exceeding data sheet maximum recommendations but that's a reflection of my solid-state design experience *and a clear and obvious contradiction to the success of many old amplifiers still in use today.* My preference is to use vacuum tubes that don't exceed manufacturer's maximum ratings in my circuits. What this means is, like our design example, I select tubes that operate at lower voltages than Fender (et al) would have selected.

Past experience and some of the few writings that I've found regarding the proper bias of the screen grid for power amplifier applications aren't consistent with the older practice although there are conflicts. Reliability concerns usually suggest some or all of the following:

Screen grids should not be operated at plate voltage potential, for reliability considerations.

Screen grid voltages should be de-rated to 60% of the plate voltage.

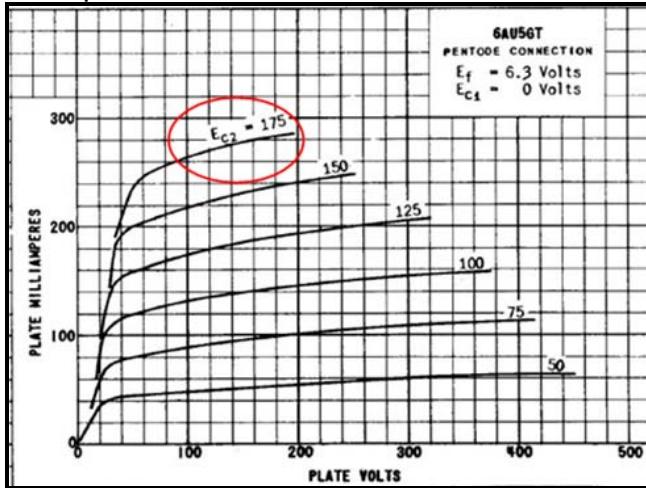
Screen grid voltages shouldn't be dependant upon plate voltage, i.e. plate dropping resistors shouldn't be used to bias screen grids.

But some readings suggest that the plate voltage should never be driven below the "knee" of the plate characteristics curves because this will overly stress the screen grid during operation. *This implies that the screen grid voltage NEEDS to be partly dependant on plate voltage, contradicting the previous statement.*

I usually make the presumption that the engineers who generated the ratings for vacuum tubes knew considerably more about their products than I can know. Having said that, let us recall that the military was the greatest consumer of these devices prior to the huge entertainment media expansion of the 1950s. One could make an inference that there were pressures placed upon tube manufacturers to establish practical limits on maximum operating parameters that would result in reasonable tube life.

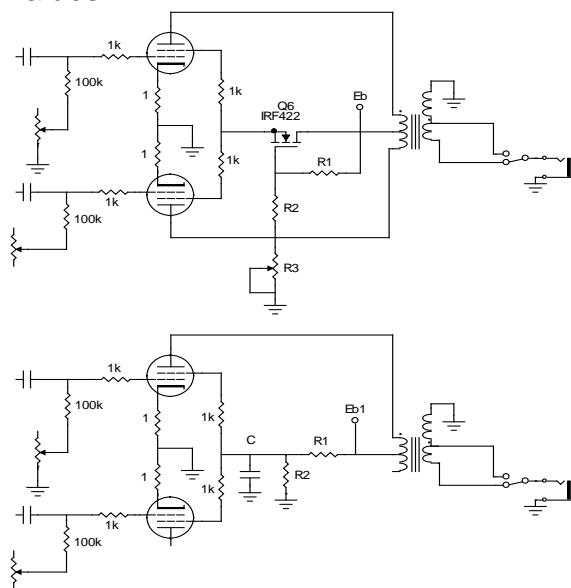
If that premise is accepted, we could adjust our comfort level to operating all bias conditions up to the data sheet maximum values, with the possible exception of the screen grid. I believe that the screen grid should be operated lower than the plate voltage. As a matter of fact, a good rule of thumb could be established from a simple glance at a particular tube's plate characteristics.

A set of plate curves based of screen grid bias is included in typical data sheets. I'd recommend that no greater voltage be applied to the screen grid than the maximum voltage that the manufacturer used to plot the plate curves. Here's an example:



For this particular tube type, the manufacturer plotted screen grid bias voltages from 50 to 175 volts. I would take 175 volts as the maximum allowable screen grid voltage for this tube.

Below is part of a schematic of an amplifier that I designed a few years ago. I include it here to point out different screen grid bias methods. In the topmost schematic, rather than using a dropping resistor from plate to screen grid, a circuit was employed that provides a more constant, semi-regulated voltage that is also adjustable. (This was included by specific request so that the operational screen grid bias voltage could be adjusted by the user.) Below that circuit is one that is simpler, less expensive and for which we can easily determine the required component values.



A "conventional" screen bias circuit is exactly like the one that we designed earlier, consisting of a single resistor that establishes screen bias voltage by dropping a predetermined amount of voltage from the plate supply. The amount of voltage that is dropped is dependant on the desired screen grid voltage and the resistor value, as we have previously discussed in our earlier design example. This has the advantage of being simple and cost effective. It's not the recommended configuration for improved distortion characteristics.

The earlier described circuit is adequate for musical instrument amplifiers where distortion is not of primary concern. The two circuits depicted above provide more constant screen grid bias than the single resistor design. *The disadvantage of these types of circuit is that during normal operation, it is possible for the plate voltage to swing lower than the screen grid voltage. If the duty cycle is high, the result would be excessive power dissipation in the screen grid* (because the screen grid would behave exactly like the plate for a portion of the signal cycle).

Addition of capacitor "C" tends to hold the voltage constant as plate current fluctuations cause the power supply voltage to fluctuate (can be omitted in practical circuits where the total parallel value of the screen resistors, including the voltage divider, is smaller than about 3k. Chapter 8.15 discusses the computational method for determining the value of the bypass capacitor.) A value of 0.2 to 10 μ F is generally adequate. Screen grid bypass capacitors need a working voltage significantly higher than the screen grid bias voltage, E_{c2} and, as noted previously, the proper polarity of the capacitor relative to the circuit voltages must be observed.

The procedure for designing this screen bias circuit is almost as simple as the single resistor circuit. As in that circuit, the plate voltage and the screen bias current at the desired screen bias voltage must be known and we've already established those, so referring back to our latest table of design information, we find that:

$$\mathbf{E_b} = 309 \text{ volts}, \mathbf{E_{c2}} = 173 \text{ volts and } \mathbf{I_{c2}} = 1.8 \text{ mA } (\mathbf{.0018 \text{ amperes}})$$

In the above schematic, it can be observed that there is a 1k resistor in series with each of the screen grid connections. We call this a "de-coupling" resistor, it minimizes the amount of audio signal leakage (coupling) between the two output tubes. The value is not particularly critical and if it isn't significantly larger than the 1k value shown, it can be ignored for our purposes.

The overall calculation becomes much simpler if we can pick one of the two unknown (R_1, R_2) values. If the voltage divider is to be effective, more current must flow through the divider than the screen grid current that the divider is supplying. If we make the ratio of divider current to total screen current about 4, then we can say:

$$R2 = Ec2 / (4 \times Ic2)$$

We can then substitute known values and solve:

$$R2 = 173 / (4 \times .0018) = 24,028 \text{ ohms}$$

We'll pick the closest standard value of **24k ohms** and the power rating required:

$$P = R2 \times (4 \times Ic2)^2 = 24,000 \times (4 \times .0018)^2 = 1.244 \text{ watts}$$

We should double the rating for safety and pick a standard power rating of **3 watts**.

Now we can solve for the remaining unknown, R1 by substituting known values into:

$$R1 = (Eb - Ec2) / (6 \times Ic2)$$

$$R1 = (309 - 173) / (6 \times .0018) = 12,593 \text{ (closest standard value is 12k)}$$

$$P = R1 \times (6 \times Ic2)^2 = 12,000 \times (6 \times .0018)^2 = 1.4 \text{ watts}$$

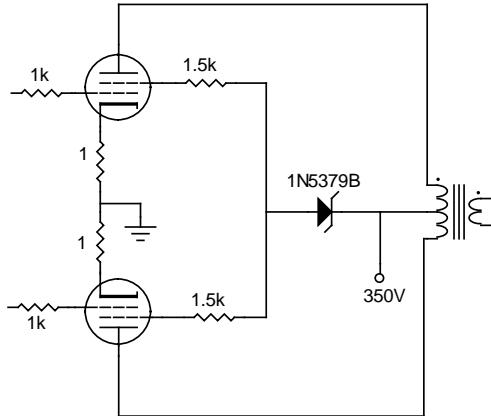
Doubling this value for safe de-rating, we select a standard rating of **3 watts**.

The circuit depicted below was used on an amplifier that I built a few years ago to bias the screen grids. To insure that the screen grid bias voltage can never rise to the plate voltage potential, a zener diode was added in series with the voltage supply to the screen resistors. A zener diode has a constant voltage drop across the two terminals, if the polarity is properly observed. (If the zener is reverse connected, it will behave like a normal diode and the voltage drop will be around 1 volt.)

Since there must always be a difference between the screen grid and the plate equal to the zener voltage, the screen grid can never be equal to the plate voltage. The zener value isn't critical, something around 10 to 20 volts would be representative. The zener voltage needs to be subtracted from the power supply voltage before calculating the value of the screen grid bias resistance. In the example schematic, the screen grid voltage could be calculated as follows:

$$V_{\text{screen}} = (V_{\text{supply}} - V_{\text{zener}}) - (I_{\text{screen}} \times R_{\text{screen}})$$

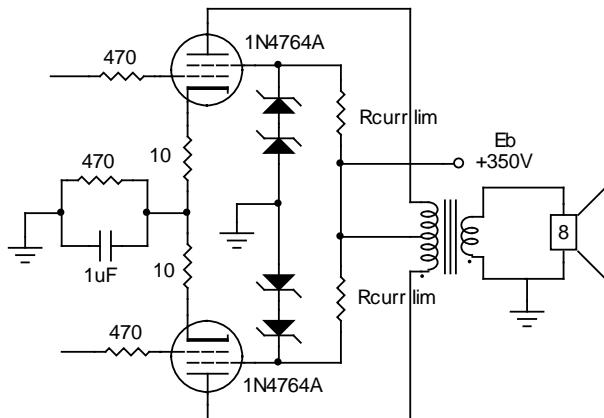
The expression can be re-arranged to calculate other parameters, such as the value of R_{screen}.



If the 1.5 k current limiting resistors were much larger in value, it would be wise to add a bypass capacitor between screen grids and cathodes. Note that the power rating of the zener should exceed the actual dissipation by a factor of 2. The dissipation is calculated as follows:

$$P_{\text{diss}} = \text{zener voltage} \times \text{total screen current}$$

The illustration below is an improved circuit, the screen grids are maintained at a constant voltage by the series-connected zener diodes. The diodes are selected so that the sum of their zener voltages is equal to the desired screen voltage. For the example shown, the 1N4764A diodes are rated at 100 volts and 1 watt so the screen voltage bias is simply the sum of the two 100 volt diodes or 200 volts.



The current limiting resistors are chosen to operate the zener diodes at a safe dissipation, which is usually about 1/2 the maximum dissipation. For the above diodes, rated at 1 watt, we need to operate each diode at 500 milliwatts or 0.5 watts. The current required for this dissipation is given by

$$I = P / E \text{ where } P \text{ is desired maximum power and } E \text{ is the zener voltage}$$

For the example shown, $I = 0.5 / 100 = .005$ amperes the value of the resistor can be determined from the maximum current, the screen bias voltage and the supply voltage as follows

$$R = (E_b - E_{screen}) / I$$

and for the example shown, where E is the screen bias voltage, 200 V

$$R = (350 - 200) / .005 = 150 / .005 = 30,000 \text{ ohms}$$

The power dissipation of the resistor can be determined by

$$P = I^2 \times R = (.005)^2 \times 30,000 = 0.75 \text{ watts}$$

Following safe de-rating procedures of 50% for resistors, this component should be rated at 1.5 watts, minimum.

It's a common practice to optimize screen grid bias conditions at test. If one chooses to do so, always measure the screen grid current, in addition to screen grid voltage. The product of the two is the screen grid dissipation and should be checked against the tube data sheet value to confirm that the maximum allowable dissipation has not been exceeded.

In summary there are references that cite the desirability of a constant voltage bias at the screen grid. For linear amplification there is little doubt that this is desirable. However guitar amplifiers are frequently operated in large-signal conditions, where the plate voltage swing may frequently force the plate voltage lower than the screen voltage. As mentioned several times previously, this poses a reliability problem because the screen grid may draw excess current and exceed its power dissipation capability.

It is difficult for the designer to know which screen bias technique is ideal for general applications. BUT the advantage of knowing how to design these circuits suggests that a clever designer can have the best of both worlds by designing a screen grid bias circuit that is switchable.

For example, one switch position might be the conventional single resistor feedback bias configuration. The other switch position could enable a voltage source bias configuration, like any of those in the above schematics. This could be an inexpensive and versatile feature in an amplifier that might require usage in both undistorted and slightly distorted modes.

8.26 Simple Forms of Distortion and Possible Improvements

I suspect that an audiophile might become uneasy with any design procedure that doesn't specifically address distortion and the means for calculating or

estimating it. We guitarists don't share this view, for the most part, we routinely allow distortion levels in our equipment and there is justification for this.

Distortion in a stereo amplifier, for example, would affect the characteristics of every single instrument in a recording playback. This might be an entire symphony orchestra, and the effect of that would be unacceptable due to an effect that we call "multi-tone intermodulation". (We don't want to get into that yet other than to state that multi-tone distortion is a far more serious problem than single-tone distortion and the fewer the tones, the better.)

If the playback, for example, contained only a single instrument, it's likely that the human ear wouldn't even detect the difference between the distorted signal and a perfect single-tone signal. (Differences, if detected, would probably be interpreted as "tonal".) When comparing high-fidelity amplifiers with guitar amplifiers, the same standards obviously do not apply. Most of the literature regarding the topic supports this statement.

So at this point, the obvious question would be how much distortion is permissible for our particular amplifier? There's no easy answer for this although many scholarly studies have been conducted on the subject. One clue might be that we seem to be happy with what the industry gives us and, as mentioned earlier, 5% distortion is the usual amount permitted during an output power measurement. (For multi-tone distortion, as would be experienced listening to orchestral music, a consensus suggests that levels as low as 1% can be audible to astute listeners.)

The calculation of distortion is not particularly difficult (at least for the first two orders of harmonic distortion) but requires a lot of work, plotting things on the plate curves and extracting graphical information is sometimes difficult and not all that accurate. Distortion characteristics are of more interest to the manufacturers that produce amplifiers. Their engineers typically use computer simulations or simply make measurements on a "breadboard" amplifier.

There are a number of causes for distortion, the major ones being the non-linearity of vacuum tube performance characteristics. These are simply inherent to the devices and it's frankly amazing to me that the long-dead designers of these tubes were able to obtain the level of performance from them that was established and documented. Having said that, there are a few practical suggestions to improve tube linearity:

Employing "ultra-linear" output transformers with special screen grid feedback connections (expensive)

Selecting a tube that has very consistent plate characteristics, specifically referring to the spacing between the control grid 1 curves (the more uniform, the better)

Careful design of screen grid bias circuit (e.g. "ultra-linear" output transformer configuration previously mentioned)

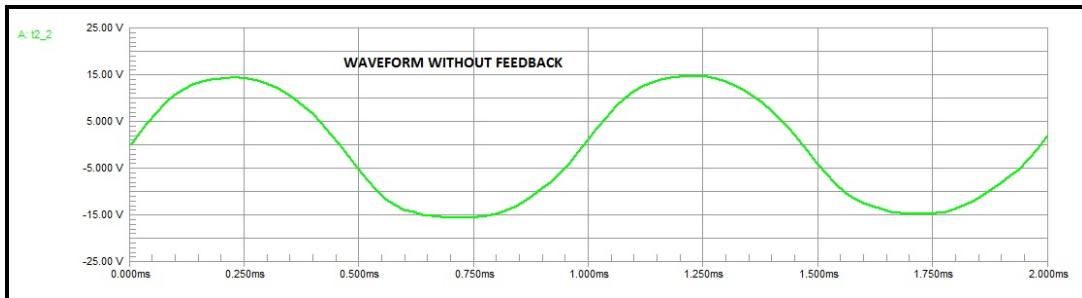
Utilizing the screen grid as the input rather than the control grid (screen grid draws current but the control grid doesn't, so this would require an additional power amplifier to precede the output amplifier)

Use of negative feedback (most common and cost-effective technique)

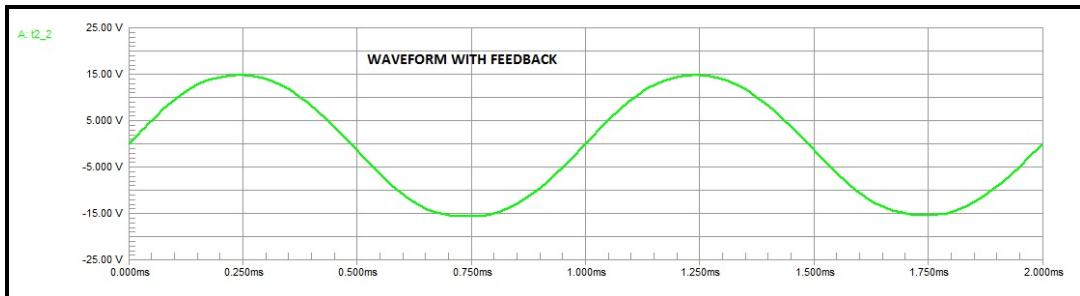
Very high performance high-fidelity amplifiers were developed and produced during the decade of the 1950s, using combinations of the above techniques. These produced very high quality music reproduction that, even today, rivals solid state amplifiers. They are very costly to produce and extremely impressive to look at.

The usual - actually the ONLY - method used in the majority of guitar amplifiers to improve distortion is negative feedback. We will discuss negative feedback in a later chapter; it's not a simple topic and deserves separate treatment. For our purposes, let's just leave it at this: negative feedback uses a small amount of the output voltage of the power amplifier to "pre-distort" one of the input stages further back in the amplifier chain; the result is that distortion is minimized appreciably.

Illustrating the concept, here are two computer simulations showing how distortion "looks" from the viewpoint of two different measurement techniques. The first simulation first shows the output waveform (at the speaker terminal) as it would appear on an oscilloscope. Examining the waveform, the bottom portion of the waveform is "flattened", compared to the top. This is exactly what our example amplifier, which had gain variations at the two extremes of the load line, would look like if measured:

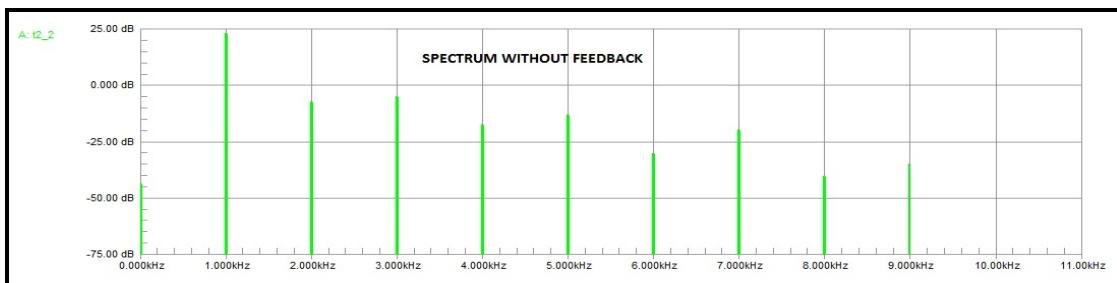


The second simulation shows the output waveform of the same amplifier WITH negative feedback included in the circuit, the waveform is now symmetrical.

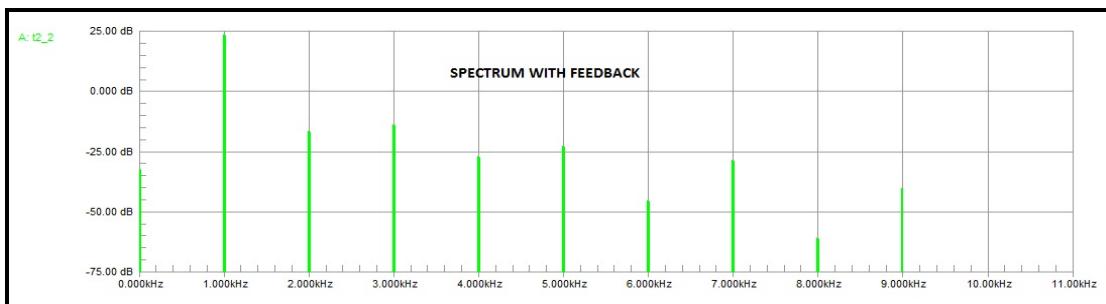


Oscilloscope waveforms are not especially useful for measuring small amounts of distortion, frequently an instrument called a "spectrum analyzer" is used. Very large signals can be displayed along with very small signals because the scale is logarithmic, it's calibrated in decibels (dB).

This is the same simulation as the first example except using a spectrum analyzer to examine the amplifier output instead of an oscilloscope. The largest signal is the desired signal tone - the single audio signal at a frequency of 1 kHz that is being introduced to the amplifier input. Note that harmonics occur at levels ranging from -30 dB to -65 dB from the fundamental signal. This would be representative of an output stage with about 5% distortion:



The next illustration is the output of the amplifier in the second example, where negative feedback has been added to the circuit. Although it looks similar to the above spectrum, the harmonic distortion has been substantially reduced when the individual signals are examined closely.



The reduction in harmonic content is 10 to 25 dB. In other words, if we use the conversions from relative power levels to dB ratios that we've learned from

previous discussion, *the power levels of the harmonics have been reduced by amounts ranging from factors of 10 to over 300!* This amplifier would have distortion around 1.5%.

As we noted above, addition of negative feedback is very cost effective (the addition of one resistor and sometimes a capacitor). The improvement in performance is almost amazing and is predictable. Negative feedback is also useful for enhancing certain performance parameters of other circuit elements besides the output tubes - the output transformer, for example.

If you're still with me at this point, you may have formed the conclusion that worrying about distortion at this stage of the design is not necessary. Negative feedback will correct most problems except those caused by defective parts or poor design. With the addition of one resistor, we've changed an amplifier with the distortion characteristics of a typical guitar amplifier to one that would be representative of a good quality vacuum tube stereo amplifier.

There are other ways of selecting a vacuum tube for power amplifier applications and there are other ways of biasing and configuring the circuit. The method described here is a simple one and an effective one but is not necessarily the best one. A more complete selection/design process could reveal a wider choice of output tubes. This method will be described later.

8.27 Negative Feedback

There is a complete discipline regarding the design of negative feedback loops, generally referred to as "Servo Theory". (This name dates back to some of the first applications of feedback: servo motors.) The subject is exceedingly complex and not remotely within the scope of a brief discussion like this one, so we will focus narrowly on the practical implementation of negative feedback, omitting some of the advanced mathematical concepts that a complete discussion would require.

Negative feedback is the most powerful tool the designer can employ in order to compensate for the many deficiencies in components that are available to us, especially in the power amplifier stage. The term "negative" doesn't have any "coloration" it simply refers to signals being out of phase (as in "positive" and "negative" battery terminals. Since the term "out of phase" suggests that another signal is present, with which the phase is being compared, let's clarify.

Feedback suggests that an output signal is re-introduced to another portion of a common circuit, usually an "input" (or at least a part of the signal chain that is at a lower level, preceding the point from which the "feedback" signal is derived). So phase difference refers to the difference in phase between "output" and "input" signals for our purposes.

The process has many advantages, some of which follow:

Stabilize gain at a fixed and predetermined value
Increase frequency response (bandwidth) to a fixed and predetermined value

Suppress distortion products of all types

Compensate for component deficiencies, particularly in output tubes and transformer

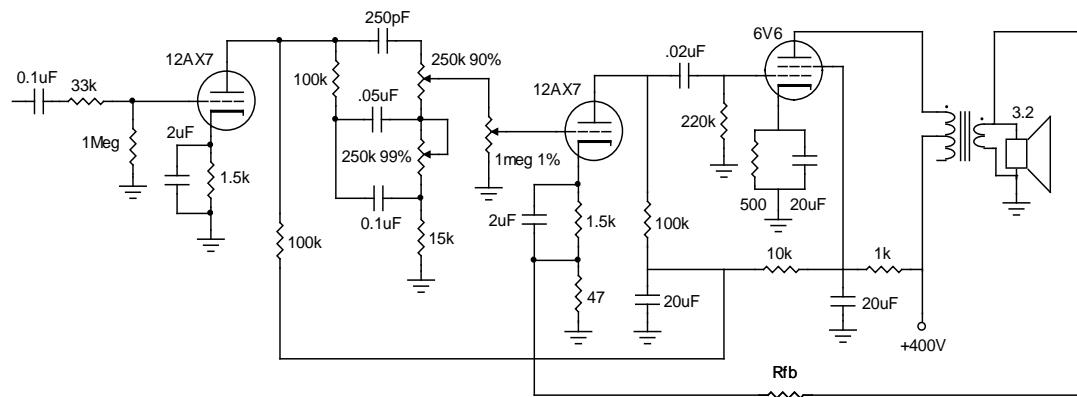
There are disadvantages to the process as well, generally minor ones:

A propensity toward oscillation if there is unintentional, excessive phase shift in any component within the feedback loop

A slight sacrifice in output power

Excess gain within the loop is always required

Here's a typical schematic of a guitar amplifier that employs negative feedback for all of the reasons described above:



This small, Class "A" amplifier has a feedback network consisting of a single resistor (R_{fb}) connecting the output of the amplifier (speaker terminal) to the cathode of the post amplifier stage following the preamplifier and EQ/volume control. The location of the feedback "input" signal is important; it needs to be far enough "back" in the signal chain so as to affect most of the components in the chain.

The signal phase at the location of the "input" must also be out of phase with the signal "output". This is generally not a problem, since we can pick either of the two loudspeaker terminals as our "output", one terminal will always be in phase and the other will always be out of phase with the input of the loop.

We can't place the equalization circuits within the feedback loop - if we did so, the feedback effect would try to flatten the frequency response of the amplifier and the tone controls would have little or no effect. So the traditional location is as shown above, just after the EQ circuit.

For the purpose of an example design, our design goal will be to establish a closed-loop gain of **20**.

Let's introduce the concept of "open loop gain" at this point which is simply the amount of gain in the circuit *WITHOUT* feedback (or as is commonly described: before the loop is "closed"). Let the open loop gain be called "**A_o**" and for the above circuit, the open loop gain consists of all of the gain and loss between the control grid of the post amplifier and the amplifier output at the speaker terminals.

The voltage gain of the above loop can be determined by the following:

$$B = R_k / R_f \text{ and}$$

$$A_v = A_o / [1 + (B \times A_o)]$$

Where B is the feedback ratio, R_f is the feedback resistor and R_k is the effective cathode resistance of 47 ohms. Note that the bias of the post amplifier stage is determined by the sum of the two cathode resistors, 1.5 k and 47 ohms. However, the 1.5k resistor is bypassed with the 2 uF capacitor so signal analysis dictates that the 1.5k resistor is "shorted" (because capacitors are assumed to have zero impedance for the purpose of signal analysis).

We don't know the open loop gain, A_o , of the circuit above so that's the first thing we need to establish. The simplest, quickest way of doing this is to refer back to the tabulated tables of gain that were used in previous examples, like this one:

R _p	R _s	R _{g1}	R _{total} (R _p R _s)	Eb = 90			Eb = 180			Eb = 300		
R _k	A _v	E _{o p-p}	R _k	A _v	E _{o p-p}	R _k	A _v	E _{o p-p}	R _k	A _v	E _{o p-p}	
100,000	100,000	100,000	50,000	1,700	31	14.1	1,000	40	42.4	760	43	84.8
100,000	100,000	100,000	50,000	2,000	38	19.5	1,100	46	56.6	900	50	113.1
240,000	240,000	100,000	120,000	3,500	43	18.4	2,000	54	50.9	1,600	58	104.6
240,000	510,000	100,000	163,200	3,900	49	24.3	2,300	59	67.9	1,800	64	132.9
510,000	510,000	100,000	255,000	7,100	50	20.9	4,300	62	53.7	3,100	66	110.3
510,000	1,000,000	100,000	337,748	7,800	53	25.7	4,800	64	67.9	3,600	69	130.1
240,000	240,000	10,000,000	120,000	0	37	25.2	0	53	42.4	0	62	90.5
240,000	510,000	10,000,000	163,200	0	44	15.3	0	60	53.7	0	67	115.9
510,000	510,000	10,000,000	255,000	0	44	14.1	0	61	48.1	0	69	99.0
510,000	1,000,000	10,000,000	337,748	0	49	18.1	0	66	59.4	0	71	115.9

Using the highest voltage in the table, as the one most closely approximating our circuit and the first example, let the open loop voltage gain, $A_o = 43$. We can justify this rough approximation since we are going to significantly reduce the open loop gain of the circuit, so the feedback ratio is probably going to dominate the gain calculations.

The estimated gain of the final stage can be approximated by consulting a data sheet for the 6V6 tube:

CHARACTERISTICS AND TYPICAL OPERATION			
CLASS A₁ AMPLIFIER			
Plate Voltage	180	250	315 Volts
Screen Voltage	180	250	225 Volts
Grid-Number 1 Voltage	-8.5	-12.5	-13.0 Volts
Peak AF Grid-Number 1 Voltage	8.5	12.5	13.0 Volts
Plate Resistance, approximate	50000	50000	80000 Ohms
Transconductance	3700	4100	3750 Micromhos
Zero-Signal Plate Current	29	45	34 Milliamperes
Maximum-Signal Plate Current	30	47	35 Milliamperes
Zero-Signal Screen Current	3.0	4.5	2.2 Milliamperes
Maximum-Signal Screen Current	4.0	7.0	6.0 Milliamperes
Load Resistance	5500	5000	8500 Ohms
Total Harmonic Distortion, approximate	8	8	12 Percent
Maximum-Signal Power Output	2.0	4.5	5.5 Watts

As previously discussed, the open loop voltage gain is:

$$A_o = V_{out} / V_{in}$$

which should be simple to ascertain since the above data sheet gives the input voltage (described as "Peak AF Grid-Number 1 Voltage", "AF" is the abbreviation for "audio frequency") as **13.0 volts peak**, as highlighted above.

Unfortunately, no output voltage is given, from which we could calculate gain. However the output power is given and the load resistance so we can determine the RMS voltage from:

$$P = E^2 / R$$

where P is output power, E is RMS output voltage and R is load resistance, re-arranging and substituting values we get

$$E = (P \times R)^{0.5} = (5.5 \times 8500)^{0.5} = 216.2 \text{ volts RMS}$$

We have the output voltage in volts RMS but before we can make the gain estimate, we need to convert the input voltage to the same units:

$$V_{rms} = .707 \times V_{peak} = .707 \times 13.0 = 9.2 \text{ Vrms and}$$

$$A_o = V_{out} / V_{in} = 216.2 / 9.2 = 23.53$$

The only other contributor is the output transformer and the voltage gain is:

$$A_o = 1 / \text{turns ratio}$$

We're assuming that the transformer turns ratio has already been determined and is 40 : 1, so:

$$A_o = 1 / 40 = .025$$

Summary:

Av of post amplifier: **43**

Av of output amplifier: **23.5**

Av of output transformer = **.025**

The total open loop gain is the product of ALL the above, or

$$A_o = 43 \times 23.5 \times .025 = 25.26$$

Recalling that our desired closed-loop gain is 20, we can re-arrange, substitute values and determine the value for the feedback resistor, Rfb:

$$A_v = A_o / [1 + (A_o \times B)] \text{ and re-arranging}$$

$$B = (A_o - A_v) / (A_v \times A_o) = (25.26 - 20) / (20 \times 25.26) = .010$$

$$B = R_k / R_f \text{ and re-arranging}$$

$$R_f = R_k / B = 47 / .010 = 4700 \text{ ohms or } 4.7k$$

Computer simulation of this circuit indicated a closed-loop gain of 21.4 compared with our design value of 20 - good agreement considering the approximations involved in the process. Although we didn't faithfully follow this practice the best procedure would be to determine the product of all the open loop gains before settling on a desired closed loop gain. If the open loop gain is not significantly higher than the desired closed loop gain, the loop will not be stable (it may oscillate) and the desired closed loop gain will not be possible.

For those who are inclined to investigate the loop in greater detail, it's fairly simple to perform a signal analysis on the loop, calculating the gain and the phase through the entire circuit. There are many free programs on the internet that will allow one to perform the analysis. It's also fairly easy to make the calculations on any pocket calculator that has complex algebraic functions (in the form: $r + jX$ or in the form of magnitude and phase angle).

A stable loop is indicated by noting the phase angle at the point where gain = 1. The phase angle must be greater than 135 degrees (recall that perfect out of phase condition is 180 degree phase shift). The difference between 180 degrees and the phase at gain = 1 is called "phase margin" and must always exceed 45

degrees. *Unless the transformer or other component within the loop has serious bandwidth limitation, this is unlikely to be a practical problem.*

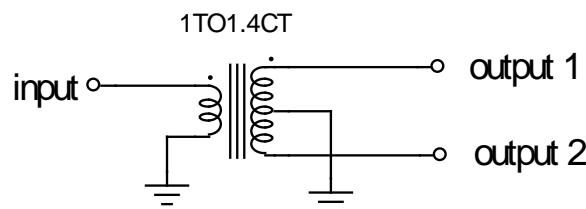
A historical note: Many - if not most - of the older Gibson amplifiers from the 1950 - 1970 era did not include negative feedback while all Fender amplifiers used negative feedback, down to the smallest "Champ" circuit. There are two schools of thought, obviously, and this is still reflected today. Many guitarists express a preference for "no feedback" amplifiers and have had their Fender amplifiers modified for that reason. I think that sufficient reasons for including feedback (predictable gain, limiting the noise bandwidth) exist to justify the slight additional design effort required to implement feedback.

On a practical note, when one first assembles an amplifier with negative feedback, one may inadvertently connect the loop to the wrong speaker terminal. This error will be immediately apparent when the amplifier commences to oscillate. The same problem may be experienced even when the loop is connected properly if there is insufficient "phase margin" in the loop. The universal cure for this problem is to narrow the loop bandwidth, eliminating or minimizing excess phase shift.

If the designer has performed initial amplifier analysis, such as completing the spreadsheet described in chapters 6.0 and 22.0, the gain distribution has been determined and a suggested value of closed loop gain has been calculated on the spreadsheet.

9.0 Designing Triode Phase Splitters

Previous chapters have described the function of this circuit. Briefly the phase-splitter divides the single input signal into two equal amplitude signals and then alters the phase of one of the signals by 180 degrees. All phase splitters share these characteristics and, as also previously mentioned, a simple center-tapped transformer can accomplish the same purpose at the cost of a little extra weight and the savings of a little current:



The two output signals are 180 degrees out of phase with each other. The amplitude of the output signals can be greater, equal to or less than the input signal, depending on the transformer turns ratio. Regardless of voltage and current gain/loss in this passive circuit, there is **ALWAYS** a power loss of 3 dB.

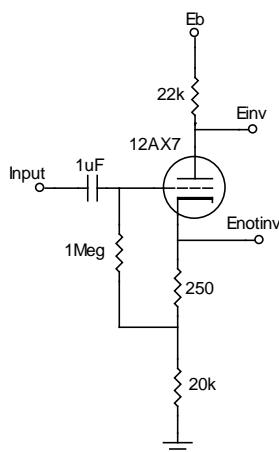
The use of vacuum tube phase splitters became common as an economic expedient, many years ago. Tubes were less expensive than transformers and cost always was the main driver of guitar amplifier design. The practice continues today mainly because of consumer pressure (the use of vacuum tubes throughout the amplifier chain is perceived to be "good"). There are two types of phase-splitters, differentiated by whether the circuit has voltage gain or does not.

9.1 Phase Splitter Without Gain

The simplest form is the circuit that doesn't have gain, functionally it consists of a single tube (or one-half of a dual tube) configured as a cathode follower but also including the normal plate output connection. A cathode follower, where the output is taken from the tube cathode, has no voltage gain. The gain at the plate of a common cathode vacuum tube amplifier is approximately proportional to the ratio of plate resistance divided by cathode resistance - if the cathode resistor is not bypassed with a capacitor.

If the same value of resistance is used for both plate and cathode, then *there is no voltage gain at the plate of the tube*, the signal voltage is approximately equal to the signal voltage at the un-bypassed cathode. But there is a phase inversion between the tube control grid and the plate while there is **no** phase inversion between the tube control grid and the cathode.

The single tube can divide the signal into two equal level signals, one being out of phase with the other. This type of phase splitter is limited to lower power level amplifiers, around 15 watts would be typical and an example amplifier would be a Fender "Princeton". Here's a typical circuit (the two output signals "E inverting" and "E not-inverting" indicate that the signals are out of phase with each other):



Note that the grid resistor is returned to a point between the two cathode resistors of 250 ohms and 20k. This establishes the bias condition of the phase splitter tube; the effective cathode bias resistance is 250 ohms NOT the series combination of 250 plus 20k. If the grid resistor were returned to ground through the 1 Megohm resistor, the cathode bias resistance would be 20,250 ohms and

the tube wouldn't operate properly at the extremely low plate current that would result.

Signal voltage output taken from the cathode of a tube is always slightly less than the input signal voltage. Since we need the output of the plate to be at the same level, we use the approximation for gain below:

$$Av = R_{plate} / R_{cathode}$$

we want unity gain, where $Av = 1$ so

$$R_{plate} = R_{cathode}$$

In point of fact, the voltage gain of a cathode follower circuit is actually slightly less than 1, due to losses within the tube. A reasonable estimate for the overall gain of a phase splitter of this type is about 0.9.

The sum of the two cathode resistors is approximately equal to the plate resistor, satisfying the requirement that the two outputs are the same signal level. The plate resistance is generally selected to be about half the value of the input resistance of the following stage for minimum signal loss. Our example power amplifier design, the stage that follows this one, has an input impedance of 100k so the condition is satisfied.

In order for a vacuum tube to operate properly, there must be adequate bias voltage between plate and cathode (unlike transistors, tubes don't function properly from low voltages). The phase splitter, in addition to the two functions earlier described, must be able to provide enough output voltage to drive the power stage to full output power plus have some operational margin so that the tube performs correctly.

If we refer back to our example power amplifier design, we find that the voltage gain of that stage is about 14.3 for the desired output power of 25 watts. Although we have previously calculated the voltage required for this power level, it's useful to review the process, starting at the load (the 8 ohm speaker). Solving for voltage across the load:

$$P = E^2 / R \text{ re-arranging}$$

$$E = (P \times R)^{0.5} \text{ and substituting values}$$

$$E = (25 \times 8)^{0.5} = (200)^{0.5} = 14.142 \text{ volts rms}$$

This is the rms voltage across the speaker terminals. Recalling that the turns ratio of the transformer is 16.394, we can solve for the RMS voltage across the two plates of the push=pull output tubes:

$$V_{\text{plate-plate}} = N \times V_{\text{out}}$$

where N is transformer turns ratio, substituting values

$$V_{\text{plate-plate}} = 16.394 \times 14.142 = \mathbf{231.846 \text{ volts rms}}$$

Using the output stage voltage gain of 14.3, the voltage at the input of the power amplifier stage is:

$$V_{\text{input}} = V_{\text{output}} / A_v$$

and substituting

$$V_{\text{input}} = 231.846 / 14.3 = \mathbf{16.213 \text{ volts rms}}$$

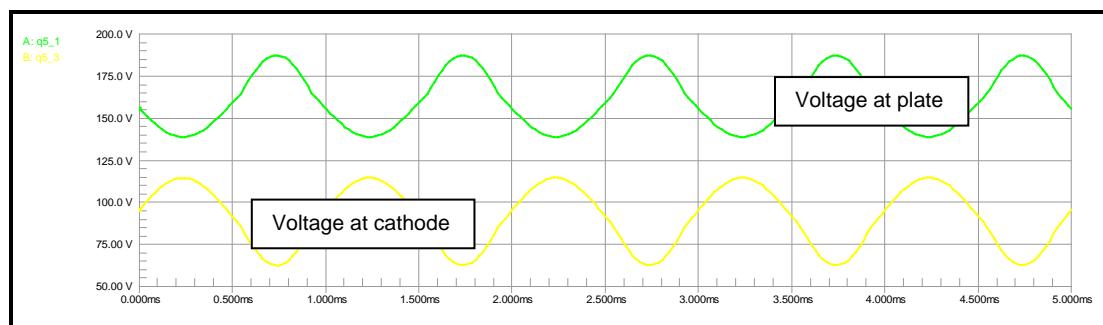
Let's convert this to peak to peak voltage, which we need in order to make a selection for the proper phase-splitter plate voltage:

$$V_{\text{pp}} = 2.828 \times V_{\text{rms}}$$

and substituting

$$V_{\text{pp}} = 2.828 \times 16.213 = \mathbf{45.85 \text{ volts p-p}}$$

The phase splitter must be able to provide two voltages, 180 degrees different in phase, of 45.85 volts, peak to peak. The tube plate to cathode voltage has to be more than twice the peak to peak output signal voltage of 45.85 volts. Two and one-half to three times the voltage swing would be a good choice to insure linearity. Three times the signal swing would give a plate to cathode operating voltage of around 138 volts and we used 150 volts in our example. The output signals would look something like this if we measured them at the plate and at the cathode:



Since the output voltages from plate and cathode are connected to the following stages with capacitors, the DC component of the signal voltages is blocked - only the AC signals are passed to the next stage. The signals, as was desired, are of equal amplitude and 180 degrees out of phase (which means that if one signal is

on the "positive" portion of the signal cycle, the other signal will be on the "negative" portion of the signal cycle - they are mirror images of one another).

The initial estimate of quiescent current was obtained as follows:

The supply voltage was selected to be 250 volts, the considerations for choosing supply voltages will be covered in the power supply chapter.

We picked the plate and cathode resistors to total about 50k so that they are about half the input impedance of the next stage.

We made an estimate for plate to cathode voltage of about 150 volts, in accordance with the reasons mentioned above.

The quiescent current was determined from "Ohm's Law" by substitution of the selected or existing conditions:

$$I = E / R \text{ and } E \text{ is approximately}$$

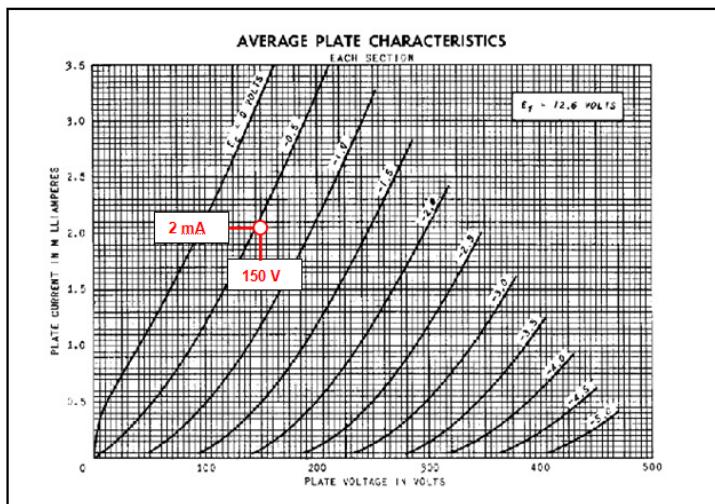
$$(V_{\text{supply}} - V_{\text{plate to cathode}}) \text{ or } (250 - 150) = 100$$

Substituting, we obtain

$$I = 100 / 50,000 \text{ or about 2 millamps}$$

(Round-off and replacement of calculated values with standard resistor values resulted in the values shown on the schematic.)

This phase inverter has no gain, so we don't need the plate characteristic curves to establish anything other than the quiescent bias condition. Referring to the plate curves, we can determine the value of the cathode bias resistor by first plotting the quiescent current of 2 millamps and the quiescent voltage of 150 volts (see below).



Examining the gate voltage curves (E_g), the gate voltage required to produce 2 mA current at 150 volts is about -0.5 volts.

The cathode resistor value can be calculated from "Ohms Law" which states that:

$$I = E / R \text{ rearranging}$$

$$R = E / I$$

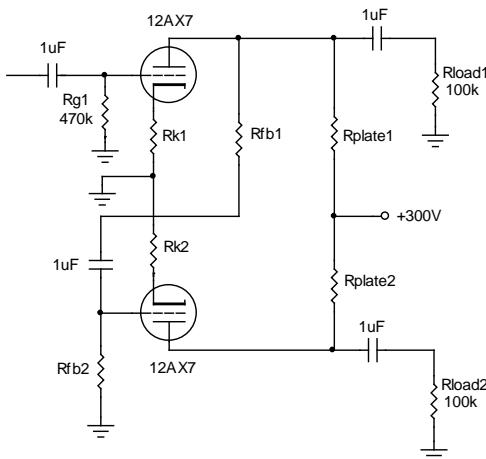
and substituting

$$R = 0.5 / .002 = 250 \text{ ohms}$$

This is the plate curve with the bias point plotted for the above example.

9.2 Dual-Triode Phase Splitter With Gain

The phase-splitter that needs to have voltage gain must have a different configuration than the single-tube version described in the previous chapter. Observing the circuit of a typical phase-splitter, it's immediately obvious that two tubes are required, or more accurately, two tube *functions* are required. Any of the small dual triodes can be used for this purpose, such as the archetypal 12AX7A, rated at one watt dissipation for each plate. Another commonly used dual triode, for higher power applications, is the 6SN7, rated at 5 watts dissipation for each plate and frequently used where large plate voltage swings are required. A typical schematic of a phase-splitter with gain is shown below.



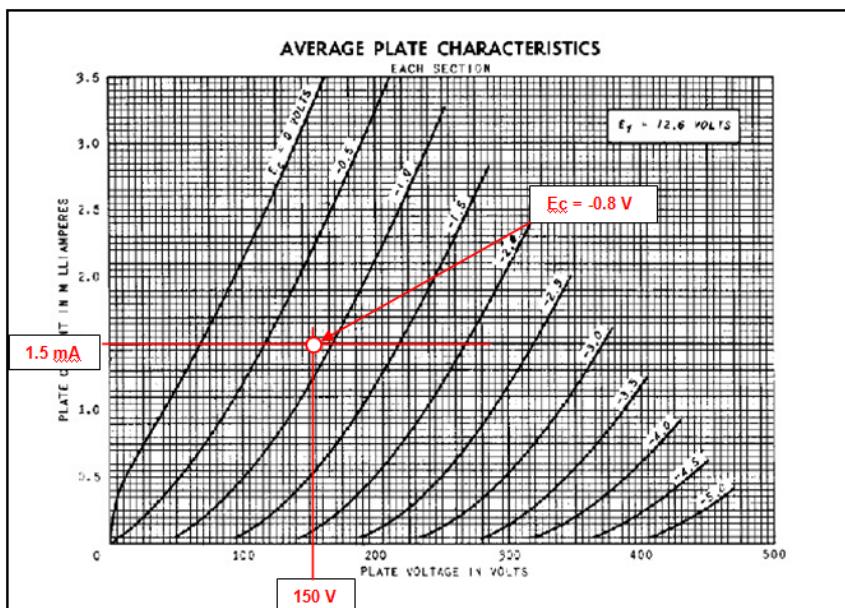
The input signal is applied to the control grid of the upper tube. After amplification (where the signal is also inverted), the output signal is applied to load 1. A sample of the output signal is applied through a feedback resistor network to the control grid of the lower tube. After re-amplification (where the signal is inverted a second time), the output signal is applied to load 2. Since the first signal was inverted once while the second signal was inverted twice, the two output signals are 180 degrees out of phase.

Let's make a presumption that the desired performance parameters of this particular circuit are as follows: Supply voltage is 300 volts, required gain is $A_v = 10$, we've chosen the quiescent plate current, I_q , to be about 1.5 mA each tube and we know the load resistor values of 100k (representing the input impedance of the next stage, which in our example would be the power amplifier stage). The output voltage required is 41.24 volts, peak to peak, as we determined in the previous example.

Note that the design process is an iterative one, as I've mentioned in other parts of this document. Normally we may have to perform some (or all) of the following calculations twice, once to make a good first estimate for some of the critical parameters and then a second time, inserting our estimates into the equations to determine final values for the components.

First Trial:

As in the previous example, we can use the 12AX7 plate curves to determine the cathode bias resistor value. As a first estimate, we make a point on the plate curves at the intersection of 1.5 mA and 1/2 of the supply voltage or 150 volts, we can estimate the control grid voltage, E_C , for this set of bias conditions. From the curve below, this voltage would be about 1.6 volts.



Using "Ohm's Law" $I = E / R$, we can re-arrange the expression to give

$$R = E / I \text{ and insert the known values to give}$$

$$R = 0.8 / .0015 = 533 \text{ ohms}$$

Or standard values of 560 ohms for each of the two cathode resistors.

Let's discuss how the various other component values are determined. The voltage gain, A_v , is approximately given by the following expression:

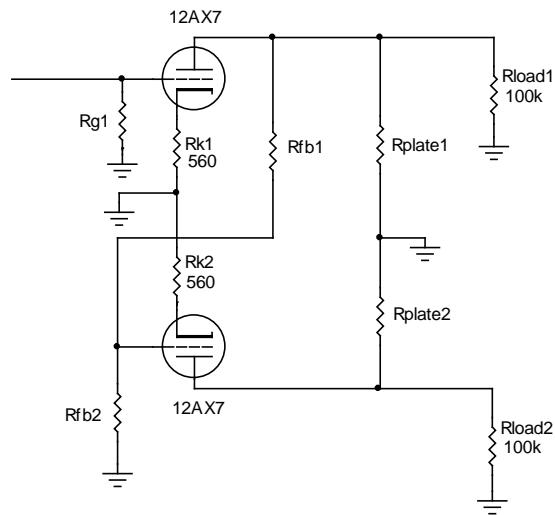
$$A_v = R_{load} / (2 \times R_{cathode})$$

(Note that the cathode resistors are not "bypassed" with capacitors - which would be the case if we were trying to obtain maximum gain.) The term R_{load} refers to everything that is connected to the plate of the tube. In the schematic above, the

load resistance consists of the load of the following stage (100k) AND the plate resistor.

When making a signal analysis, normal practice is to assume that all capacitors and all power supply connections are zero impedance (unless they are part of a filter or feedback circuit). For the example schematic, setting all the capacitors to zero impedance means that we can simply eliminate them from the circuit.

Setting the power supply connections to zero impedance means that we have "grounded" those connections, so far as the signal is concerned. We end up with this equivalent for analysis purposes:



Now that we know the value of the cathode resistors and given a desired voltage gain value of $A_v = 10$, we can re-arrange the expression for gain to read

$$R_{load} = A_v \times (2 \times R_{cathode})$$

and substituting values we get this expression

$$R_{load} = 10 \times 2 \times 560 = 11,200 \text{ ohms} \text{ (or standard value of 12k)}$$

this value, as noted above, includes the plate resistor - which is what we need to determine - and the 100k input impedance of the following stage.

The expressions relating the value of resistors in series or in parallel are:

$$R_{total} = R_1 + R_2 \text{ for series circuits and}$$

$$R_{total} = (R_1 \times R_2) / (R_1 + R_2) \text{ for parallel circuits of two resistors}$$

For parallel resistance in circuits of two and more resistors, the solution is:

$$1 / R_{\text{total}} = 1 / R_1 + 1 / R_2 + 1 / R_3 + 1 / R_4 + \dots \text{ and so forth}$$

To confirm our understanding of total resistance, let's use an example of two resistors, 1200 ohms and 3600 ohms and calculate the series total resistance and the parallel total resistance:

For the series case, $R_{\text{total}} = 1200 + 3600 = 4800 \text{ ohms}$

For the parallel case, $R_{\text{total}} = (1200 \times 3600) / (1200 + 3600) = 4,320,000 / 4800 = 900 \text{ ohms}$ or

$1 / R_{\text{total}} = 1/1200 + 1/3600 = 1 / .00111$ which is again 900 ohms

Referring back to the last schematic, it's clear that the plate resistor and the 100k input resistance of the following stage are in parallel. Let's re-arrange the parallel resistance expression so that we can account for the 100k parallel resistor and determine the value of the plate resistor:

$R_{\text{total}} = (R_{\text{plate}} \times 100,000) / (R_{\text{plate}} + 100,000) = 12,000$ and

$R_{\text{plate}} = (- R_{\text{total}} \times R_{\text{load}}) / (R_{\text{total}} - R_{\text{load}})$

$$=(-12,000 \times 100,000) / (12,000 - 100,000)$$

$$= (-12,000,000,000) / (-88,000)$$

= 13,636 we can select 15,000 or **15k** since it is a close standard value

Now we need to calculate the voltage drop across the plate resistors and determine the actual plate to cathode voltage. This voltage needs to be at least one and one-half the peak to peak signal voltage at the plate. Recall that the peak to peak signal voltage required is 41.24 volts.

Given the supply voltage of 300 volts, the plate resistance and the plate current, Ohm's law is also used to determine the voltage drop. First stating the expression and then substituting known values:

$E = R \times I = 15,000 \times .0015 = 22.5 \text{ volts}$

subtracting this from the 300 volt supply, we obtain the plate voltage of

$300 - 22.5 = 277.5 \text{ volts}$

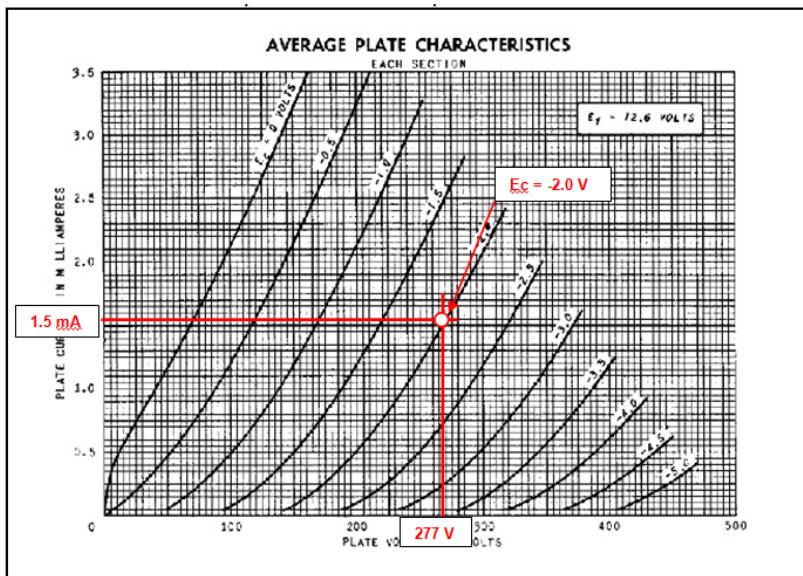
and from this value, we must subtract the voltage drop across the cathode resistor of 0.8 volts (this is the control grid voltage, E_C1) to get the actual cathode to plate voltage:

$$277.5 - 0.8 = 276.7 \text{ volts}$$

this value, compared to the peak to peak output signal voltage required, provides a ratio of:

$$276.7 / 41.24 \text{ or about } 6.7 \text{ to } 1$$

we stated that a ratio of 1.5 to 1 was required for the plate to cathode voltage and this condition easily satisfies the requirement. We can refer back to our plate curves at this point and refine the estimate for the cathode resistor, to see if a second iteration of the process will be required:



Second iteration:

Using the same procedure followed originally (Ohm's Law), we use the new value for control grid voltage (Ec1) of 1.9 volts and calculate a new value for the cathode bias resistor:

$$R = E / I \text{ and insert the known values to give}$$

$$R = 2.0 / .0015 = 1333 \text{ ohms}$$

We'll change the original value to 1200 ohms (1.2k) as it is a standard value, reiterating the procedure:

$$Av = R_{load} / (2 \times R_{cathode})$$

$$R_{load} = Av \times 2 \times R_{cathode}$$

and substituting values we get this expression

$$R_{load} = 10 \times 2 \times 1200 = 24,000 \text{ ohms}$$

$$R_{total} = (R_{plate} \times 100,000) / (R_{plate} + 100,000) = 36000 \text{ and}$$

$$R_{plate} = (- R_{total} \times R_{load}) / (R_{total} - R_{load})$$

$$= (-24,000 \times 100,000) / (24,000 - 100,000)$$

$$= (-2,400,000,000) / (-76,000)$$

$$= 31,579 \text{ we can select } 33k \text{ since it is the closest standard value}$$

All that remains is to determine the values of the feedback network, R_{fb1} and R_{fb2} . The resistor, R_{fb1} , can easily be determined by making it at least ten times the value of the plate resistance of 33k. (Making the value so high insures that it will have minimal "loading" effect on the plate resistor.) We can then determine the value of R_{fb2} because the ratio of the two feedback resistors determines the required voltage gain value as follows:

$$Av = R_{fb1} / R_{fb2}$$

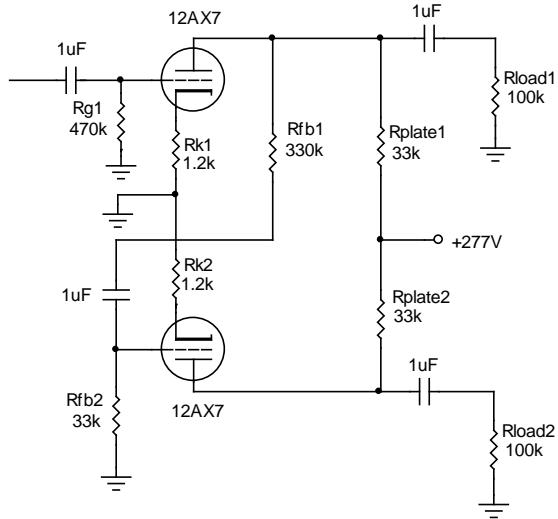
since we know that Av is 10 and that R_{fb1} is now 330k, we can calculate R_{fb2} as follows:

$$R_{fb2} = R_{fb1} / Av$$

and substituting known values

$$R_{fb2} = 330,000 / 10 = 33,000 \text{ (standard value)}$$

As a matter of interest, a computer simulation of this circuit predicted a voltage gain, Av , of 12, rather than the desired goal of $Av = 10$. After the circuit is constructed and measured, if it was determined to be of sufficient importance to obtain a voltage gain closer to 10, the two cathode resistors could be slightly increased (to the next higher standard value). The schematic diagram depicted below includes component values that we've calculated.



Running the circuit simulation a second time, using cathode resistor values of 1.5k instead of 1.2k predicted a voltage gain of 10.2. Note that the phase splitter will normally be inside the negative feedback loop so the gain of all stages within the loop will be externally established.

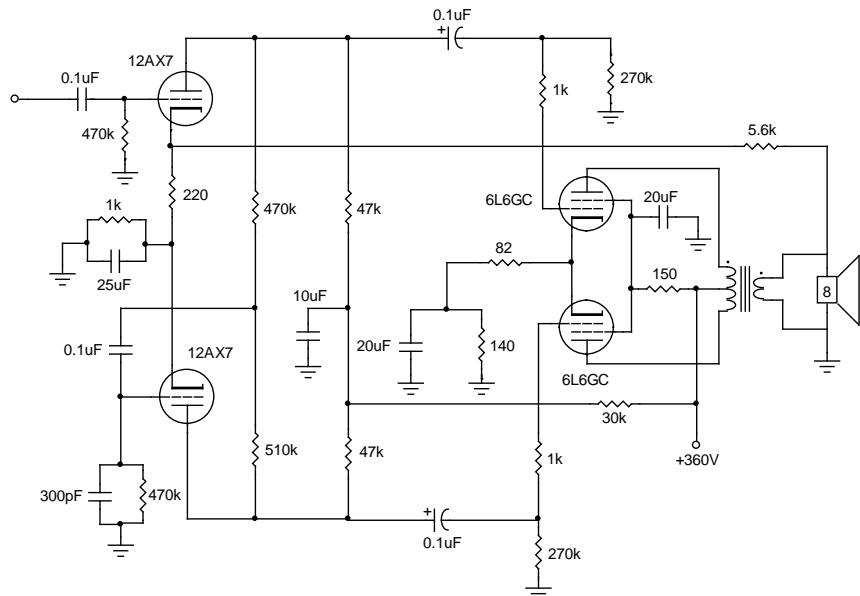
The clear implication is that it's *not* worthwhile to spend large amounts of time to achieve precise design goals for voltage gain. A tolerance of ten to twenty percent is acceptable in most cases. (The gain can always be adjusted by at least that much by making minor resistor changes when the circuit is constructed and measured.)

9.3 Practical Example of Phase-Splitter + Power Amplifier

I am fortunate to own an early 1960's "Ampeg Reverberocket" guitar amplifier, a model that is well-regarded and deservedly in my opinion. Ampeg vacuum tube amplifiers (and their modern solid-state models) have a reputation for reliability, based on conservative design practices and quality standards. This particular amplifier hasn't been played for more than ten or twenty minutes at a time in many years but I worked with it regularly at one time, 20 watts RMS power being adequate - even with a drummer - for jazz bands of the day.

An electrolytic capacitor in the power supply and the original Jensen speaker have been replaced and I've also added a "line out" option - the remainder is pretty much as shipped from New York. (As a matter of interest, I have the instruction pamphlet that was originally shipped with the amplifier.) The output tubes in my version of this amplifier are not easy to obtain (\$50 + for a pair of 7868 tubes) so I've allowed this amplifier to retire gracefully.

Reproduced below is a partial schematic for a more widely-produced version of this model (with readily-obtainable 6L6 output tubes). Only the phase-splitter and output power amplifier stages are shown.



The circuit is conventional in most regards but has some subtleties that are slightly more sophisticated than many amplifiers available at the time (e.g. Fender, Gibson). The conservative attitude of the original designers is apparent in the inclusion of multiple cathode bias resistors. Cathode resistors provide series feedback and enhance bias stability.

Feedback, as discussed in other chapters, is desirable for other reasons than bias stability, for example obtaining predictable linearity and gain. The Ampeg circuit above is a good example of the use of multiple feedback paths. The cathode resistors in the output stage provide both series *signal* feedback, from the un-bypassed 82 ohm resistor and series *bias* feedback, provided by both of the resistors in the cathode circuit. Additionally, the phase-splitter + power amplifier stages are contained within another feedback loop - the 5.6k resistor connected to the speaker output.

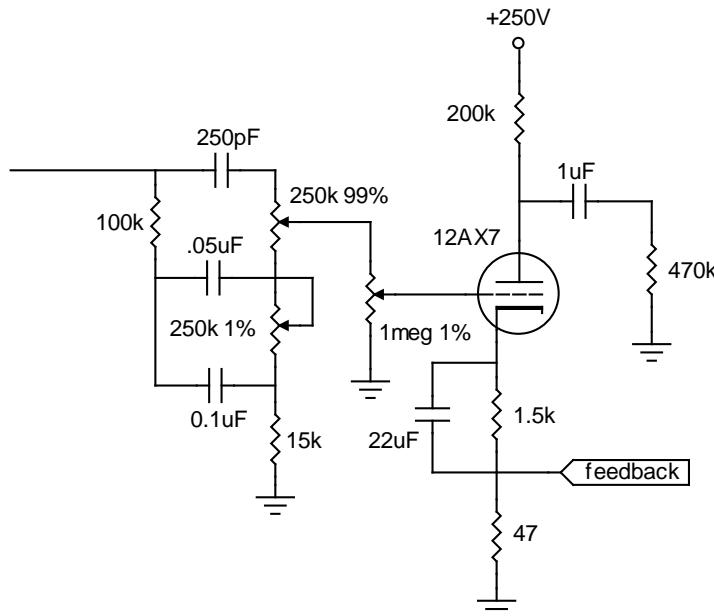
It's likely that few problems would be encountered if the output tubes were replaced with ones that weren't specifically matched for identical bias conditions in the output stage. The signal gain of this stage, equally, is predictable and also the distortion characteristics. But in addition to the cathode resistors, shunt feedback is provided by the 5.6k resistor, connected from the speaker output terminal to the cathode of one of the phase-splitter tubes. This establishes the overall gain and the linearity of both the output stages.

While the circuit configuration is not as common as the Fender version, I think that it's superior in linearity and makes matching of output tubes unnecessary. There is a slight penalty in output power resulting from the series feedback resistance but I'd recommend considering this configuration for new amplifier designs. (The equivalent Fender product to this amplifier was the "Deluxe Reverb", which produced an additional 2 watts RMS output power.)

An unusual - and less desirable - feature found in many Ampeg amplifiers from this era was the use of triple-triode tubes, such as the 6C10 and 6U10, found in preamplifier, post-amplifier, tremolo and reverb circuits. Although the usage of these tubes made sense from a packaging aspect (more functions available in the same packaging volume), finding replacements for some of these tubes isn't easy. The use of more commonly available dual triodes, especially from the 12AX7 family, is recommended for all "new" vacuum tube amplifier designs.

10.0 Designing EQ and Post Amplifier

Several good circuits are in common use for EQ (tone control) and volume control applications. For noise consideration, these circuits are always placed AFTER the preamplifier and immediately preceding the post amplifier. One of the most popular configurations is the circuit most frequently found in Fender amplifiers. The schematic below is a Fender derivation and is found in many amplifiers, with slight variations:



(The "%" symbol can be ignored in all of the following schematics, it's a residual artifact related to potentiometers in the circuit analysis program within which the schematic was created.)

10.1 EQ And Volume Control

A very useful, downloadable program for the analysis of various tone controls can be found on the internet. Many thanks to the people that wrote and made the following program available to us:

<http://www.duncanamps.com/tsc/download.html>

This very practical analysis application has four or five different configurations of tone control and allows substitution for all components in the circuit. Be careful, when using this program, to set the source and load impedances to accurately reflect the conditions of your trial circuit - *profound performance variation can result from incorrect source/load values.*

The particular circuit variation shown above has no midrange adjustment. If one were desired, the 15k fixed resistor on the bottom of the "totem pole" tone control could be changed to a potentiometer (25k to 50k typically). Making the assumption that a previous analysis has indicated the requirements for the EQ circuit (e.g. spreadsheet calculations in Chapter 6), the following are some typical preliminary specifications.

Volume control: 0 to maximum level, logarithmic response

Voltage loss: 0.1 voltage ratio (-20 dB) maximum loss adjusted for flat response

Frequency pre-emphasis: about 10 dB loss at midband (approximately 500 Hz) to compensate for guitar magnetic pickup response

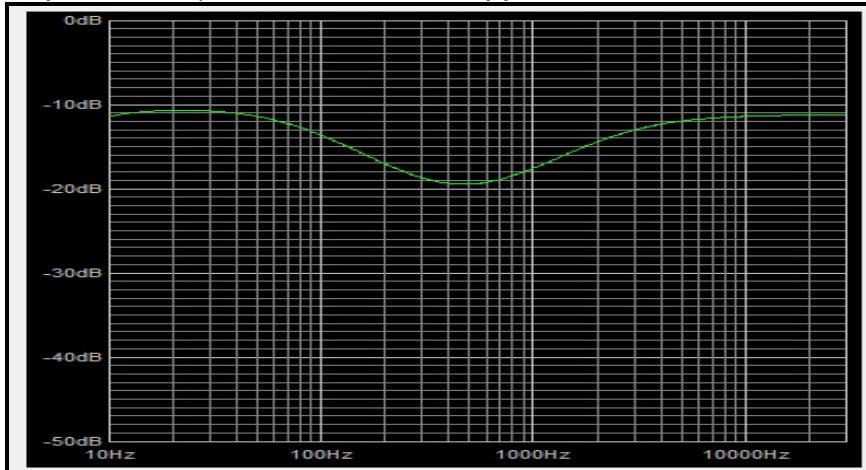
Frequency response adjustment: +10 dB at 80 Hz and 1 kHz (bass and treble adjustments, logarithmic response)

Input connection: electrically isolated from previous stage which may be at a potential up to +300 volts

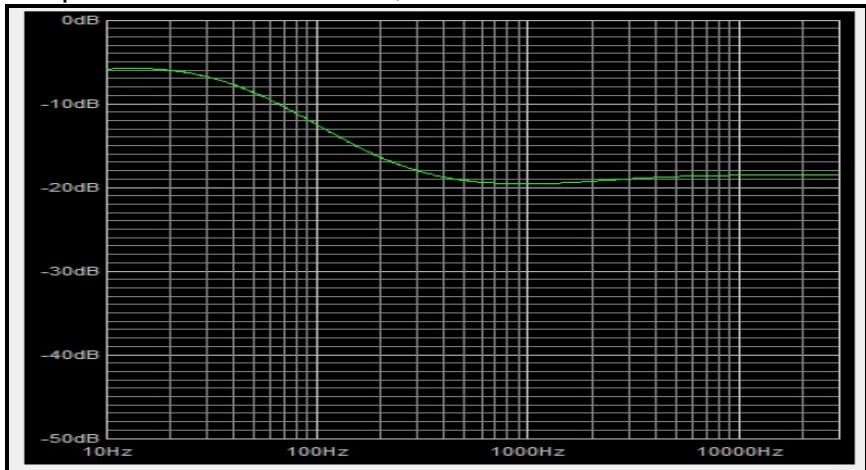
Output connection: resistively connected to ground through the volume control

All of the above requirements can be met using the Fender-style, two control circuit shown. Any number of means can be used to analyze the signal and determine that the adjustments are satisfactory. The downloadable program that is referenced above is very convenient for this purpose; let's use it to make a few swept predictions of the adjustment range.

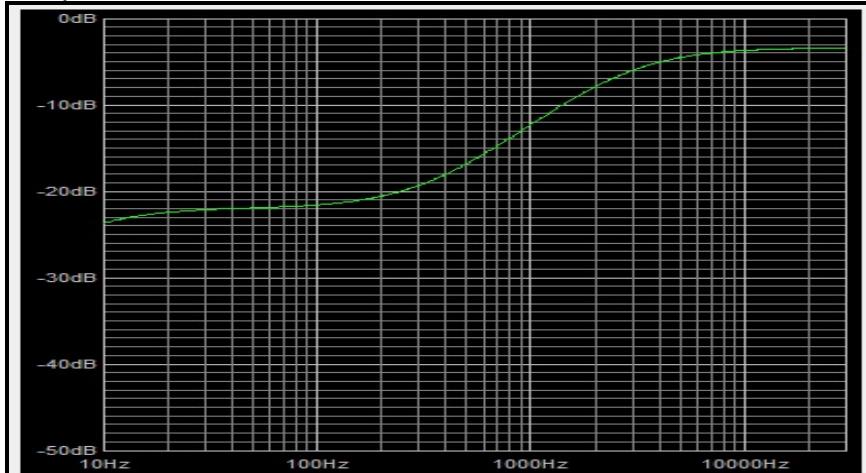
This is the response with both controls set to "5", the midband compensation (or "equalization") of about 10 dB is apparent:



Response with bass at "10", treble at "0" results in treble cut of about 10 dB:



Response with bass at "0", treble at "10" results in bass cut of about 10 dB:



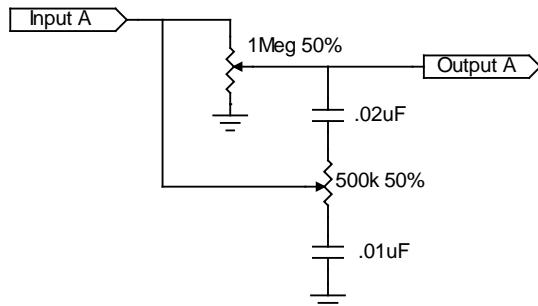
It is apparent that the adjustments are adequate for the typical requirement - the maximum loss is about the same as the specification and the connections are as described. The logarithmic response will be assured by selecting potentiometers that exhibit that characteristic. If a "mid" tone control is desired, it is usually linear response, not logarithmic.

As noted above, a midband adjustment - if desired - can be easily provided by replacing the fixed resistor with a potentiometer.

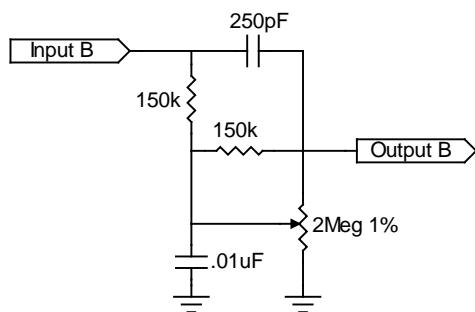
10.2 Other Commonly Used Tone Control Circuits

The following circuit is from the Ampeg "Reverberocket" amplifier discussed in section 9.3. It has the virtue of being very simple but that characteristic makes it unsuitable for certain pickup configurations. This circuit works well, for example, with Fender style pickups (single coil, bright) but not so well with pickups that have a lot of midrange response (humbucking styles).

The input impedance is about 150k when the output is terminated with a 1 Megohm resistor. Unlike most amplifiers, the volume control is placed ahead of the tone control circuit. It's not clear to me why the designer felt that this was desirable since there is interaction between the two controls.

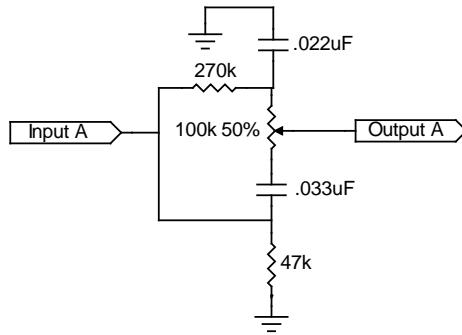


This is a tone control circuit from a Gibson amplifier of the same era, Model GA-79RTV. This circuit falls in between the Fender and Ampeg tone controls in complexity and provides acceptable performance for most pickups. The input impedance of this circuit is approximately 150k if terminated with 1 Megohm.



Here's another variation on the single-knob tone control, input impedance is about 30k when terminated in 1 Megohm. That's a little low for most preamps to

drive, we normally prefer an input impedance of 100k or greater. (This can be a very good circuit if driven from low impedance sources, like a cathode follower.)



There are many variations in tone control configuration. My personal preference has always been the circuit used in post-1960 Fender amplifiers. It seems to have the most flexibility, although parts count is higher than other tone controls.

10.3 Post Amplifier

Assuming that a previous analysis has indicated the requirements for this circuit (e.g. spreadsheet calculations in Chapter 6), here are some typical preliminary specifications:

Input impedance: 100k nominal

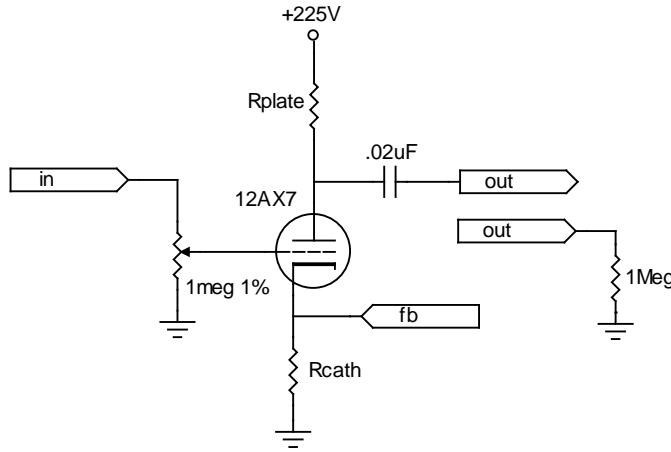
Voltage gain: 30

V_{out} p-p: 150

V_{supply}: +300 volts, maximum

Feedback connection: input connection located at the cathode, need not be D.C. isolated *provided that cathode bias current is not adversely affected by the feedback resistor*

If it were not for the feedback requirement, which mandates an un-bypassed cathode resistor as shown below, we could use the tabulated design data (from tube manufacturers) to design this stage (as in previous and subsequent examples). We can estimate the component values by using the plate curves. Here's the schematic for a basic amplifier with an ungrounded cathode:



(As noted previously, the "%" symbol should be ignored - it is a residual of the application used to create and analyze the circuit.)

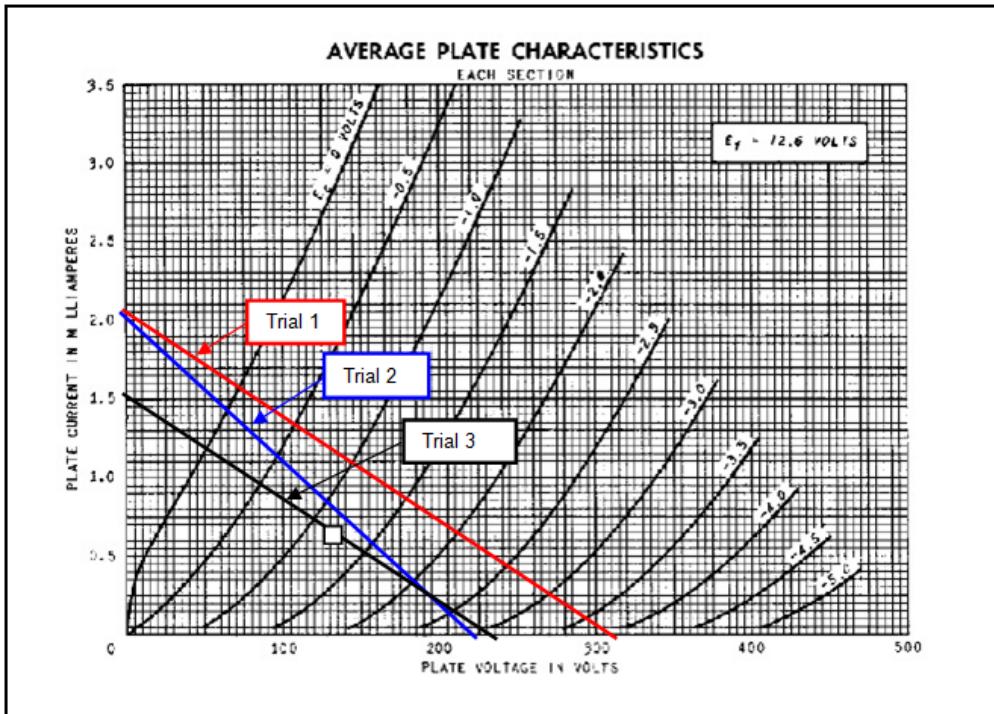
We can construct a load line (on the 12AX7 plate curves) to help determine component values for the post amplifier. We know that the output voltage swing must be at least 150 volts and we know that the gain must be around 30. As we've noted throughout the discussion, vacuum tube design is usually an iterative process, so let's make a trial load line, defined by 300 volts plate voltage and 2 mA plate current. After drawing the load line, we can make an approximation for the voltage gain:

$$Av = \Delta \text{ plate voltage} / \Delta \text{ control grid voltage} \quad (\Delta \text{ denotes "change" or "difference"})$$

the voltage extremes. Let's let the grid voltage vary from 0 to 3 volts and then determine the plate voltage at those points on the load line, which look to be about 60 volts and 265 volts (at $I_c = 0$ and -3 volts, respectively). So the gain for trial 1 is then:

$$Av = (265 - 60) / (0 - 3) = 68.3$$

Attempting to refine our estimate, we know that the maximum voltage swing is 150 volts and, from our above estimate, the minimum plate voltage is going to be about 60 volts, so let's set a new estimate for plate voltage at $150 + 60 = 210$ volts and adjust the plate current by changing the slope of the load line so that we can achieve 150 volt variation in plate voltage for a 3 volt variation in grid voltage, calling this load line "trial 2".



We can see that when the gate voltage is at -3 volts, the plate voltage probably cannot reach the supply voltage of 210 volts. Let's make one final revision and set the supply voltage (the maximum plate voltage) at 225 volts, then adjust the slope of the load line for a 150 volt plate variation when the grid voltage varies by 3 volts, call this "trial 3".

Checking the plate voltages along the trial 3 load line for grid voltage = 0 to -3 volts, we get a variation from about 60 to 210 volts, which gives $A_v = 30$, our design requirement.

Now lets make a point on the the load line *midway between the grid voltage of 0 and -3 volts*, shown by the small rectangle on the plate curve. This represents a grid voltage of about -1.2 volts at a plate current of about 0.6 mA. We can determine the cathode resistance from Ohm's Law, re-arranging and substituting values:

$$R_{cathode} = E / I = 1.2 / .0006 = 2000 \text{ ohms or } 2k$$

Examining the same load line, the plate voltage that corresponds to the bias point is around 120 volts. Subtracting this voltage and the grid voltage from the supply voltage and dividing the result by the plate current will provide the *total* plate resistance :

$$R_{total} = (V_{supply} - E_g - E_c) / I_q = (225 - 120 - 1.2) / .0006 = 173,000 \text{ ohms}$$

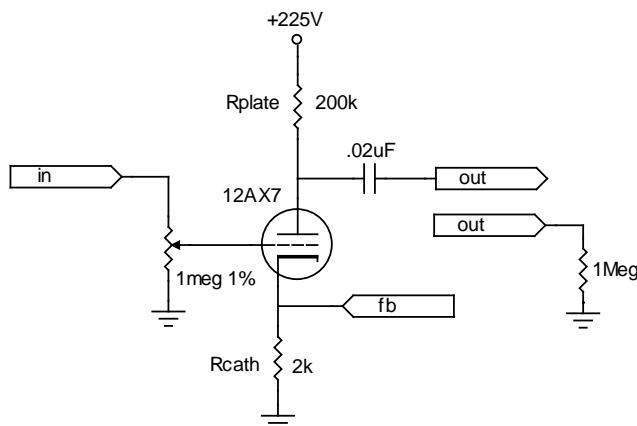
Recalling that when we analyze signal conditions, we make all capacitors zero impedance, so this total resistance *includes the 1 Megohm resistance of the grid resistor in the following stage*. We need to subtract that parallel resistance from the above calculated value to determine the actual plate resistance for our tube:

$$1 / R_{plate} = 1 / R_{total} - 1 / 1,000,000 = 1 / 173,000 - 1 / 1,000,000 = 4.78 \times 10^{-6}$$

and the plate resistance is

$$R_{plate} = 1 / R_{plate} \text{ so } R_{plate} = 1 / 4.78 \times 10^{-6} = 209,189 \Omega$$

We can use a standard value of **200k** and this is the finished circuit:



A computer simulation of this circuit indicated a voltage gain of around 35. The gain can be adjusted by varying the plate resistance if closer agreement is required. If an amplifier, when tested, exhibits low overall gain and the feedback gain (all of the gain in the stages that follow the post amplifier) is correct, then this stage would be the obvious place to make gain adjustments.

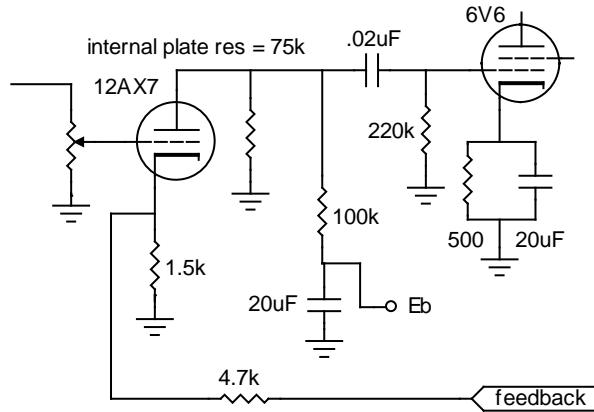
There is a "parasitic" element associated with the plate circuit of vacuum tubes that we haven't yet addressed. This element is called the internal plate resistance and shouldn't be confused with the external plate resistor. The internal plate resistance is readily visualized by assuming a separate resistor in parallel with the external plate resistance but only for A.C. signal purposes. It is not necessary to account for the internal plate resistance in bias calculations.

The voltage gain of small-signal vacuum tubes associated with the "front end" stages of an amplifier, because of the typically high impedances, are significantly affected by internal plate resistance. In power amplifiers, this is not normally of concern, because output impedances are relatively low and internal plate resistances large in comparison.

The internal plate resistance value can be found on the tube data sheet. The resistance isn't a constant value, it varies with bias voltage. Re-iterating: this

resistance appears in parallel with the plate resistor (and the grid resistor of the following stage) when the signal path is considered.

Here's a representation of a plate circuit, accounting for plate resistor, internal plate resistance and grid resistor of the next stage (NOTE: for signal analysis, we've shown the internal plate resistor as if it was grounded although it is obviously not):



There are *three* resistors now that make up the plate load, the internal resistance of about 75k, the 100k plate resistor and the 220k grid resistor of the following stage. The effective plate load is

$$R_p \text{ total} = 1 / (1 / R_p + 1 / R_{int} + 1 / R_{grid})$$

$$= 1 / (1/100k + 1/75k + 1/220k) = 35.87k$$

The effective cathode resistance is the cathode resistor in parallel with the feedback resistor and the total resistance is

$$R_k \text{ total} = 1 / (1 / R_k + 1 / R_f/b)$$

$$= 1 / (1/1.5k + 1/4.7k) = 1.14k$$

We can make an estimate for the voltage gain of the above stage as follows:

$$A_v \sim R_p \text{ total} / R_k \text{ total}$$

$$= 35.87k / 1.14k = 31.5 \text{ for the example shown.}$$

It's useful to make this simple gain approximation because it provides quick confirmation of the design goal (or perhaps an indication that a second look at the calculations might be appropriate). This is not an accurate prediction but it is useful for gain stages with un-bypassed cathode resistors.

11.0 Designing the Preamplifier

In previous chapters, the purpose of the preamplifier has been discussed. Briefly reviewing, the main purpose is to provide sufficient gain so that circuit losses (in the form of the EQ and volume controls) do not create a high noise figure. That would always be the case if there were no preamplifier and there would be an unacceptable amount of "hiss" at the amplifier output.

At this point in the amplifier design, a signal analysis has presumably been completed (Chapter 6) so that we have design goals for this circuit. Let's use the following preliminary specifications and schematic with the ubiquitous 12AX7 tube (or one of the lower noise, lower hum versions of this tube, if available).

Voltage gain, $A_v = 50$

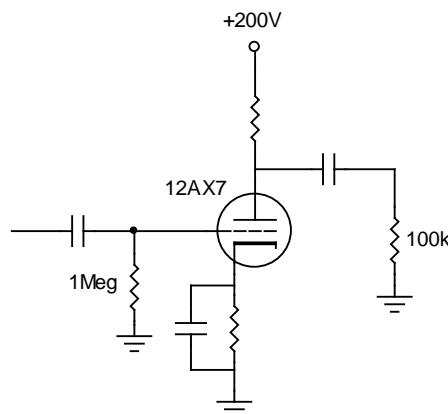
Quiescent current, $I_q = 1 \text{ mA}$

Power supply = 200 volts

Input signal is about 100 millivolts peak to peak

Guitar pickup impedance is about 10k ohms

Load impedance is about 100k ohms (the impedance of the EQ circuit)



Since the preamplifier is located at the lowest signal voltage point in the chain, there is little concern regarding linearity - the signal level at the input of the amplifier isn't adequate (unless boosted externally) to drive the preamplifier into compression or limiting. To alleviate any concern about non-linear performance, and for reasonable gain, make sure that the plate to cathode voltage is reasonably high (at least 75 volts) and that the quiescent voltage/current are established above the midpoint of the load line. A better quiescent selection is at the mid-point of the *linear* section of the plate curve.

As mentioned earlier (e.g. in the discussion about vacuum tube bias), the most commonly used vacuum tubes generally have prepared tables listing component values for various supply voltages, gain values and peak to peak output voltages. Another advantage of the published tabulations is that internal plate resistances

have been included in gain and signal output voltage calculations. Reproducing the one for the 12AX7 in the tube bias chapter:

R _p	R _s	R _{g1}	R _{total} (R _p R _s)	Eb = 90			Eb = 180			Eb = 300		
R _k	A _v	E _o p-p	R _k	A _v	E _o p-p	R _k	A _v	E _o p-p	R _k	A _v	E _o p-p	
100,000	100,000	100,000	50,000	1,700	31	14.1	1,000	40	42.4	760	43	84.8
100,000	100,000	100,000	50,000	2,000	38	19.5	1,100	46	56.6	900	50	113.1
240,000	240,000	100,000	120,000	3,500	43	18.4	2,000	54	50.9	1,600	58	104.6
240,000	510,000	100,000	163,200	3,900	49	24.3	2,300	59	67.9	1,800	64	132.9
510,000	510,000	100,000	255,000	7,100	50	20.9	4,300	62	53.7	3,100	66	110.3
510,000	1,000,000	100,000	337,748	7,800	53	25.7	4,800	64	67.9	3,600	69	130.1
240,000	240,000	10,000,000	120,000	0	37	25.2	0	53	42.4	0	62	90.5
240,000	510,000	10,000,000	163,200	0	44	15.3	0	60	53.7	0	67	115.9
510,000	510,000	10,000,000	255,000	0	44	14.1	0	61	48.1	0	69	99.0
510,000	1,000,000	10,000,000	337,748	0	49	18.1	0	66	59.4	0	71	115.9

Referring to the table, we can select the lowest plate voltage (Eb = 90 volts), although we have 200 volts available, 75 volts is adequate for this application. The configuration that provides a gain of 50 has been highlighted. The peak to peak output voltage is 20.9 volts - more than adequate for our requirement of 5 volts peak to peak (V_{out} = V_{in} x A_v). There IS a problem with this selection, however, because the load resistance isn't the same as the following circuit (EQ, volume control and post amplifier). We've estimated the input impedance of the next stage at about 100k.

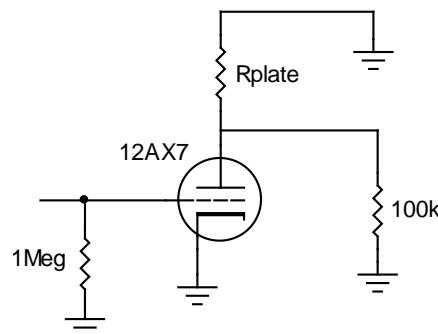
This would be a common problem with designs taken from a table, like the above. There are several ways of addressing this:

Use the table but modify the values as required to obtain the required gain

Make design estimates without considering the table at all

Use the plate characteristics to design for a different output load

All three of these approaches are feasible, the first is probably the simplest and quickest. Let's review an item from a previous chapter, related to the total resistance connected to the plate. As is the practice with signal analysis, all capacitors and power supplies are assumed to have zero impedance, so if we redraw the schematic to reflect this assumption:

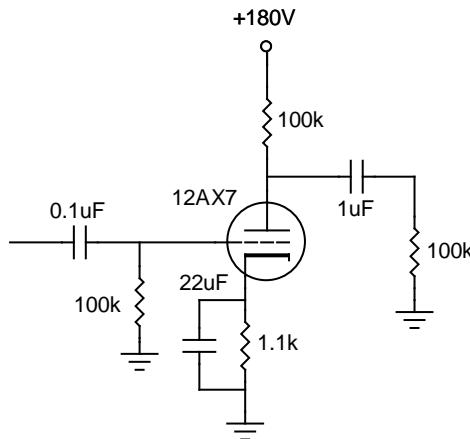


The total output load is the plate resistor in parallel with the 100k load resistance. Any value of plate resistance is going to make the total resistance lower than

100k. Let's look through the table of component values again, this time with the idea in mind that we need a total load resistance less than 100k.

R _p	R _s	R _{g1}	R _{total} (R _p R _s)	Eb = 90			Eb = 180			Eb = 300		
R _k	A _v	E _o p-p	R _k	A _v	E _o p-p	R _k	A _v	E _o p-p	R _k	A _v	E _o p-p	
100,000	100,000	100,000	50,000	1,700	31	14.1	1,000	40	42.4	760	43	84.8
100,000	100,000	100,000	50,000	2,000	38	19.5	1,100	46	56.6	900	50	113.1
240,000	240,000	100,000	120,000	3,500	43	18.4	2,000	54	50.9	1,600	58	104.6
240,000	510,000	100,000	163,200	3,900	49	24.3	2,300	59	67.9	1,800	64	132.9
510,000	510,000	100,000	255,000	7,100	50	20.9	4,300	62	53.7	3,100	66	110.3
510,000	1,000,000	100,000	337,748	7,800	53	25.7	4,800	64	67.9	3,600	69	130.1
240,000	240,000	10,000,000	120,000	0	37	25.2	0	53	42.4	0	62	90.5
240,000	510,000	10,000,000	163,200	0	44	15.3	0	60	53.7	0	67	115.9
510,000	510,000	10,000,000	255,000	0	44	14.1	0	61	48.1	0	69	99.0
510,000	1,000,000	10,000,000	337,748	0	49	18.1	0	66	59.4	0	71	115.9

The choice that is highlighted above comes close to satisfying our requirements. The *total* plate resistance can be as low as 50k. Note that the individual plate resistor is 100k and the load resistance is 100k - exactly the value we are trying to accommodate. The prices that must be paid are higher supply voltage and slightly inadequate voltage gain of 46 instead of 50. Here's the schematic of the preamplifier as designed with the tabulated values:



A computer simulation of the above circuit confirmed the gain of 46 in the table. If it were critical to obtain a voltage gain of 50 for the load and plate resistances in the previous example, one might increase the supply voltage up to 300 volts, as suggested by the tabulated component and performance values.

(Note that it is frequently practical to adjust the gain of a small signal amplifier stage during the breadboard/test process by adjusting the various resistances.)

In chapter 22.6.2, another possibility for increasing the gain of the first stage, the second stage or - more practically, the combined gain of the two - is described. Briefly, this involves the use of a dual pentode-triode tube in place of the typical dual triode-triode device normally used in guitar amplifiers.

11.1 Preamplifier Noise Figure Estimate

If it was desirable to compute the noise figure of this stage (*the preamplifier is the stage that is most critical to overall noise performance*), it's not that difficult provided that the vacuum tube itself is not considered. This is almost universally the case. Unlike solid-state devices, where noise parameters are well-understood and easily calculated, tube noise is rarely measured and data sheets do not contain noise data.

Noise contributions were previously discussed in chapter 4.1.1 but will require a slight amount of expansion at this point. The noise calculation includes "Boltzman's Constant", the absolute temperature and the circuit bandwidth, relating the noise at different points in the circuit to the circuit gain at that point.

As an example, the noise of the grid resistor would be amplified by the tube voltage gain while the noise of the plate resistor would not be amplified. The noise of the cathode resistor would be amplified ONLY if the "bypass" capacitor was omitted or on inadequate capacitance. (The amplifier analysis spread sheet in Chapter 6.0 performs a similar overall analysis but assumes that the user already has an estimate for the stage noise figure.)

The procedure for determining preamplifier noise is to calculate the noise of each resistor at the output of the circuit then combining the total noise voltages in a manner called "root sum square" method (RSS). The process can be just as well accomplished by summing the noise at the INPUT of the circuit.

The point is that the noise of every contributor must be modified by the gain or loss of the circuit where the noise "generator" is positioned. We'll perform a simple example to make the process more easily visualized.

The noise calculation for a single resistor is:

$$V_{noise} = (4 \times K \times T \times B \times R)^{0.5}$$

Where noise is in volts, K is Boltzmann's constant (*a constant is an unchangeable number, usually resulting from the study of physics, this one is named for the German scientist that first established it from his study of thermodynamics*), T is temperature in degrees Kelvin, B is bandwidth in Hertz and R is resistance in ohms. The $^{1/2}$ term means the same as taking the square root of the entire expression.

The term "4KTB" is usually simplified to 1.66 (10^{-20}) volts per Hertz at room temperature. If we also limit the bandwidth to about 10 kHz (average human hearing), we can simplify the noise expression to

$$V_{noise} = [1.66 (10^{-16}) \times R]^{0.5}$$

and if we apply this to the 100k grid resistor, multiplying the noise by the voltage gain, we get this voltage at the OUTPUT of the preamplifier:

$$V_{noise} = 46 \times [1.66 (10^{-16}) \times 100,000]^{0.5} = 187 \text{ microvolts (uV)}$$

Noise voltage is multiplied by the voltage gain of 46 to obtain output noise.

Now, performing the same analysis on the plate resistor - note that the plate resistor is located at the OUTPUT of the vacuum tube so *there is no gain applied to the noise*:

$$V_{noise} = [1.66 (10^{-16}) \times 100,000]^{0.5} = 4.07 \text{ microvolts (uV)}$$

The RSS calculation means squaring each of the noise voltages, adding them together and then taking the square root of the sum, as follows:

$$RSS = [V_1^2 + V_2^2 + V_3^2 \dots]^{0.5}$$

and substituting the above values

$$V_{noise} = [(187(10^{-6})^2 + 4.07(10^{-6})^2]^{0.5} = 187 \text{ uV at the output}$$

To obtain the input noise, divide the output noise by the voltage gain:

$$187 / 46 = 4.1 \text{ uV}$$

Note that the total noise - referenced to the preamplifier input - is about the same as the noise of the single grid resistor. This is a good intuitive example of how the preamplifier gain "masks" the noise of the following circuits.

If we want to determine "**noise figure**" we can refer back to chapter 4.1.2, where we estimated that the thermal noise at the input of any audio amplifier is approximately 1 microvolt. The noise figure calculation is then:

$$F = 20 \times \log (V_{noise} / 1 \text{ uV}) = 20 \times \log (4.1 \text{ uV} / 1 \text{ uV}) = 12.2 \text{ dB}$$

As noted previously, this doesn't include the noise of the tube itself, which isn't possible to determine without measurement. The measurement is not difficult but requires instrumentation that is not normally available. At this point, however, it's adequate to note that our 12.2 dB noise figure is close to the previously determined and desirable estimate of 10 dB.

Since this stage is at the most sensitive point in the signal chain, inadvertent coupling needs to be carefully considered and the layout accordingly planned. The preamplifier needs to be located as far as possible from the power supply, to minimize magnetic coupling of 60 Hz line voltage "hum". The high gain required

by this stage (to minimize noise contributions of the "lossy" tone control circuit) makes the stage susceptible to oscillation. Other stages in the chain with high signal levels need to be located remotely so that accidental coupling does not cause oscillation.

12.0 Designing the Power Supply

In the early years of high-volume guitar amplifier production (1953 - 1960), many performance parameters were not considered important - especially if significant cost savings resulted from their omission. Guitar amplifiers were never expected to replicate multi-tone orchestral music, even as high-fidelity equipment was being developed to perform that exact function.

Over time, as mentioned in earlier sections of this book, guitarists became accustomed to the shortcomings of products provided by major guitar amplifier manufacturers. At some point, we began to interpret the deficiencies as "desirable" and circuits originally designed eighty years ago are sometimes still used for guitar amplification.

An area that was usually treated casually, because of cost constraints and component deficiencies, was the amplifier power supply. Voltage regulation (usually provided by gas discharge tubes) while possible, was costly and generally excluded from the amplifiers of the time. Chapter 14.3 contains a hypothetical power supply design that discusses some of the typical engineering/cost tradeoffs that may have occurred fifty to seventy-five years ago.

Cost needn't now be a consideration for precise voltage regulation. Deficiencies of earlier amplifiers - that are now *desirable* - usually preclude the addition of voltage regulation circuits to "modern" vacuum tube amplifiers.

It's worth noting that jazz musicians normally don't welcome the types of performance deficiencies that other types of musicians might welcome or prefer. So if one *wants* to add voltage regulation to an amplifier power supply, it's simple and inexpensive. In a later chapter on test equipment, a variable output power supply is described. It's easy to incorporate this circuit into the amplifier power supply and achieve decent regulation (and the option of plate voltage adjustment, if desirable).

Although a typical vacuum tube power supply suitable for guitar amplifiers consists of only a few components, the design procedure can be deceptively complex. If all the various parts performed in an ideal manner, the process would be simple but unfortunately that's not the case.

12.1 Power Supply Functions

A power supply chain consists of three basic functions: transformer, rectifier and output filter. The following are brief descriptions of these circuits.

The *power transformer* converts line voltage (normally 120 V.A.C. in the U.S.) into several other voltage requirements:

Low-voltage, high-current required by the tube filaments

High voltage, moderate current required by the tube plates

Moderate voltage, very low current for control grid bias

The *rectifier* converts A.C. voltage into a single polarity: positive for plate bias, negative if the voltage is to be used for the control grid.

This voltage is not constant, it varies from the peak output voltage to zero, pulsing at a rate of *twice* the line A.C. voltage frequency (for full-wave rectifiers)

The *output filter*, comprised of a series resistance and a shunt capacitor, serves several purposes:

Filter the pulsating DC voltage so that "ripple" is reduced to an acceptable level and the DC voltage is relatively constant

Provide a current limiting function to protect the transformer and rectifiers from surge currents when the amplifier is switched "on"

The components within the power supply are more highly stressed than other parts of the amplifier with the exception of the output power tubes. One must exercise caution when selecting/specifying these parts; sometimes the stresses are not obvious from superficial examination of the circuit.

The power transformer, heart of the power supply, is the most difficult part to deal with because of a general lack of performance specifications from manufacturers. The transformer operates only with alternating current (A.C.) and cannot pass direct current (D.C.) which is what we want to obtain from the power supply. Internal resistances and leakage inductances make the performance of this part difficult to predict. There are two common configurations of amplifier power supply and they differ mainly in how the transformer is configured.

Normal power supply design consists of a multiple-part, trial and error procedure that makes use of several sets of complex curves that were generated in the U.S. by an engineer named O. H. Schrade during World War 2 (more below). Unless

one makes use of more sophisticated computer modeling procedures, this method is still the most effective way to obtain predictable power supply performance. Unfortunately, it's a tedious method to describe and teach, so I'm going to propose a simpler way - not as accurate, but adequate for our purposes.

The manner in which power transformers are specified varies from individual manufacturers. The most troublesome characteristic of these devices is the internal resistance of the windings (as well as some other parameters with lesser effect). This means that the winding resistance acts as part of a voltage divider, the load being the other part of the divider. The transformer output voltage therefore varies with the load (i.e. current consumption).

Manufacturers usually specify the A.C. (RMS) secondary voltage at a certain current consumption, a fixed load resistance in other words. If one's application corresponds with the voltage/current characteristics of the transformer then the performance of the rectifier is fairly predictable. If the current consumption is less than the specified current, then output voltage will be higher than that specified and the converse is also correct.

So the task isn't as simple as the circuit diagram might suggest. The most accurate method is the one mentioned in the introduction to this topic. It is available universally from texts on the subject (or internet search). The source paper is O. H. Schrade's "Analysis of Rectifier Operation", Proceedings of the IRE, volume 31 number 7, July 1943.

If one doesn't have access to the procedure suggested, then the spreadsheet provided is an alternative, although by no means an exact technique. One might be prepared to make alterations in the power supply circuit at test, as mentioned later on in this chapter.

12.2 Rectifier Configurations

The circuit that is probably most useful for vacuum tube amplifiers is called a "full-wave" rectifier. This is not an intuitive description but what it means is that the AC voltage (which alternates between negative and positive cycles) is rectified in such a manner so that the negative voltages become positive and add with the existing positive cycles. This circuit, which can be designed in two different ways, is universally used for obtaining plate voltages.

The other common rectifier configuration is called a "half-wave" rectifier, universally used for control grid negative bias application. This circuit is considerably less efficient than full-wave rectification and only useful for very low current requirements.

12.3 Rectifier Circuits

Two variations on the full-wave rectifier are shown on the spread sheet below, the main differences are the transformer configurations. The first example, case 1, uses a center-tapped transformer and only two rectifier diodes. It's a fairly simple circuit but has some disadvantages:

Higher secondary voltage required to obtain same DC output voltage as the case 2 circuit

Higher transformer power rating required than case 2

Higher voltage ratings required for diodes

Case 2, seemingly the more obvious choice, also has disadvantages:

Higher voltage filter capacitor required

Higher power rating required for current limiting resistor

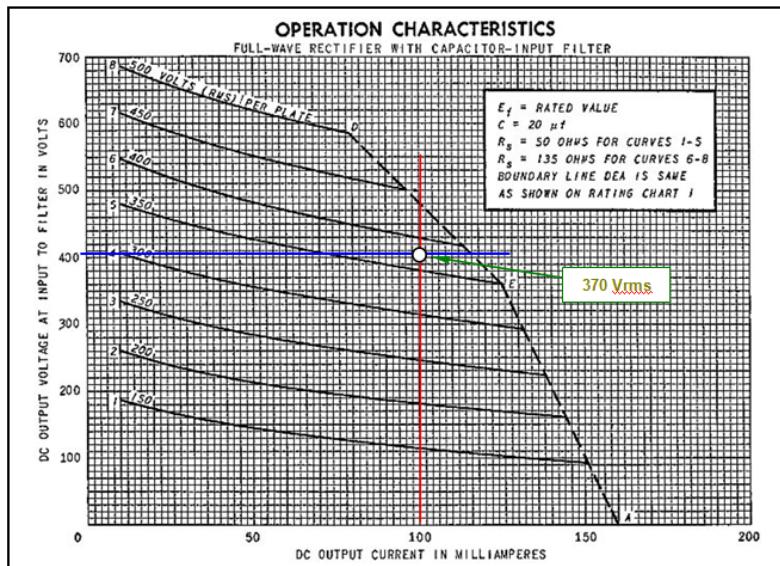
Since the design process is iterative, at some point it is likely that one of the two circuits will appear to have obvious advantages that suggest its selection. The spreadsheet below has been programmed to make some routine calculations that will allow the designer to try different component values and evaluate the effect on output voltage, percentage of ripple and stresses on all components for both of the circuit configurations that we've discussed.

		case 1		case 2				calculated parameters					
		PRI	SEC	PRI	SEC	FUSE		diode voltage rating:	900	volts, min	diode current rating:	0.15	amperes
								capacitor voltage rating:	450	volts, min	resistor power rating:	1.5	watts
								diode voltage rating:	400	volts, min	diode current rating:	0.15	amperes
								capacitor voltage rating:	500	volts, min	resistor power rating:	2.1	watts
<i>NOTE: only the items in BLUE text are user inputs, the remaining values are calculated by the spreadsheet. Entering values in any block except BLUE text blocks may invalidate spreadsheet.</i>													
select a transformer		voltage & current req'd		estimate the input filter values		calculated parameters				Secondary Voltage Too Low?	Transformer Secondary Rating Adequate?	Fuse Value Excluding Filaments	
Transformer Primary AC Voltage	Transformer Secondary AC Voltage	Transformer Rating VA	Required DC Voltage Emax	Required Current Imax, A	Current Limiter, ohms R	Filter Capacitor, uF C	Equiv Load Resistance	Estimated Transformer Resistance	Current Limiter Voltage Drop	DC Voltage	Ripple, %		
case 1 case 2	120.00 120.00	550.00 275.00	65.00 65.00	310.00 310.00	0.150 0.150	33.00 47.00	47.00 220.00	2067 2067	166 166	30 32	329 350	15 3	No No Yes 0.430 0.430

In order to simplify this process, a number of assumptions were made in the spread sheet calculations. They are all reasonable, in my opinion, but if one has the schematic of a similar supply (with similar voltage and current requirements and known transformer characteristics), then comparing circuits is a useful way to gain confidence in the design technique suggested here. The spreadsheet was created with Microsoft "Excel", therefore any other spreadsheet program used must have the ability to read "Excel" files.

The spreadsheet calculations assume that solid-state rectifiers are used, if vacuum tube rectification is desired for some reason, then the output voltages will be considerably reduced. To estimate the amount of reduction, the data sheet for the selected rectifier tube must be consulted. The data sheet will include a curve indicating the voltage drop across the tube as a function of load current. Subtract this drop from the voltage indicated on the spread sheet.

The inverse procedure would be to first determine the voltage drop across the selected rectifier tube and then add it to the desired power supply voltage. Use this corrected voltage as the input ("required D.C. voltage, Emax") on the spread sheet and manipulate the transformer parameters, current limiting resistance and filter capacitance to obtain the revised output voltage. Here's a sample data sheet for a 5Y3 rectifier tube:



To illustrate how the curve is used, let's use an example situation: a small amplifier requires 100 mA of current at 400 volts. First, draw a vertical line on the DC output current axis at 100 mA. Now draw a line corresponding to 400 volts. Examining the intersection, the RMS AC voltage required to obtain 400 volts at 100 mA is about 370 volts. If we compare this voltage to the peak D.C. voltage produced by a 370 Vrms AC voltage, we can estimate the tube voltage drop. The peak voltage conversion from RMS voltage is:

$$V_{\text{peak}} = [V_{\text{rms}} \times (2)^{0.5}] = 370 \times 1.414 = 523.2 \text{ volts peak}$$

So the voltage drop required by the 5Y3 tube is about

$$523 - 400 = 123 \text{ volts}$$

If we extend our example to a logical conclusion, we would use the 370 volt rms obtained from the 5Y3 curve above to make a transformer selection. We'd be

looking for a transformer that is center-tapped and has a RMS rating of 370 - 0 - 370 Vrms and a rating of 37 VA, minimum. The VA rating can be calculated from the product of the maximum current required and the RMS voltage of one leg of the transformer:

$$VA = I \text{ maximum} \times V \text{ secondary} = 0.100 \times 370 = 37 \text{ VA}$$

As mentioned frequently, very dangerous voltages are present in all vacuum tube guitar amplifiers, *especially in the power supply*. Unless one has experience dealing with high voltages, it's not a good idea to attempt to build these circuits or measure their performance. A more appropriate procedure might be to design the supply and then have the schematic reviewed by an experienced individual who might also be persuaded to build the circuit. (Similarly, unless one has a good understanding of the failure modes and safe ratings of the various components, it's not wise to make substitutions in values and safety ratings.)

12.4 Need For the "Bleeder" Resistor

NOTE: not shown in any of the above schematics is a component called a "bleeder resistor". This is a large value resistor that is placed directly across the terminals of the power supply filter capacitor; typical values range from 100k to several Megohms. The purpose is to gradually drain ("bleed") the charge from the filter capacitor when the amplifier is powered down. This lessens the chance of electrical shock when the amplifier chassis is opened for adjustment or modification.

This can occur even when the power cord is disconnected from the amplifier. Sufficient energy storage (in the filter capacitor) still exists to cause permanent neurological damage or death.

Bleeder resistor values can be determined by using an estimate for the desired amount of time required to drain most (63%) of the voltage from the filter capacitor(s). Knowing the time and the capacitance, the bleeder resistor can be calculated from:

$$R = t / C$$

Where R is in Megohms, t is time in seconds and C is the value of the filter capacitors in microfarads (uF). For example, if the filter capacitor is 330 uF and the designer estimates that it will require about 60 seconds to remove the chassis from the amplifier cabinet, then the resistor value required to drain the capacitor to 63% of the fully charged value is:

$$R = 60 / 330 = 0.182 \text{ Megohms or the closest standard value of } 180\text{k}$$

Should it be desirable to reduce the voltage charge further, as in the case of higher voltage power supplies, dividing the bleeder resistor value by two will result in reducing the charge of the filter capacitor by another 63%. Extending the concept, we can relate all of the variables and develop a generalized form of the discharge time/voltage of a capacitor/resistor combination as follows:

$$V_{out} = V_{supply} / \{e^{[t / (R \times C)]}\}$$

where V_{supply} is the maximum power supply voltage, V_{out} is the charge of the filter capacitor, C , after time in seconds, t , with a bleeder resistance, R . The term "e" is the base of the natural logarithm = 2.718. Re-expressing the equation:

$$V_{out} = V_{supply} / \{2.718^{[t / (R \times C)]}\}$$

From this expression, the resistance of the bleeder resistor can be determined for any combination of filter capacitance and time.

This is an appropriate time to again mention safety considerations. When one must perform measurements or repair on a high voltage powered device, the following steps need to be performed in exactly this order:

Disconnect the device from the A.C. outlet

Remove cover(s) to provide required access

Carefully, using insulated tools or insulated jumper wire, connect the power supply filter capacitor terminals together, discharging the capacitor. Don't do this momentarily - maintain the connection for a few seconds to insure that the filter capacitor is fully discharged. (Be prepared for the "snap" of the arc voltage and the visible spark that will occur.)

One last comment on the bleeder resistor: the current drain of this device must be added to the required load current of the power supply. If not and if the bleeder resistor is a fairly low resistance then the power supply voltage will be lower than the design value.

12.5 Designing a Negative Supply For Grid Bias

When the cathodes of an output power stage are grounded, a negative voltage is necessary to provide control grid bias. As previously discussed, control grids require very little current therefore no stringent requirements are placed on the negative power supply. All that's required is a relatively stable voltage of approximately -50 volts and a form of adjustment so that each tube grid in the output stage can be biased individually to provide equal cathode currents.

Note that some older amplifiers (e.g. Fender) provided a "balance" potentiometer for control grid bias adjustments. This was probably a cost-reduction measure implemented when vacuum tubes were consistent in performance, replacing the normal configuration of separate potentiometers for each control grid of the output tubes. The configuration is NOT a good idea - separate potentiometers are always desirable for balancing bias conditions of vacuum tubes commonly available at this time.

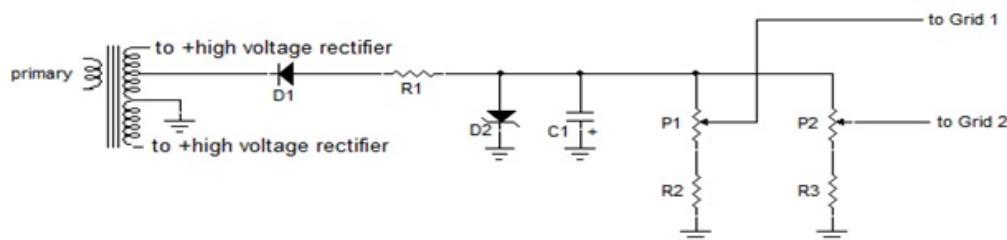
We've discussed the difference between "full-wave" and "half-wave" rectifiers previously, noting that full-wave rectifiers are universally used for plate supplies. Similarly, half-wave rectification is almost always used for grid supplies. When selecting a power supply transformer, it's customary to pick one that has a separate "tap" on the secondary from which to derive the negative voltage supply. (This isn't strictly required, but the "tap" is almost always available so we can base our design on that assumption.)

Before starting the design procedure, it's worth repeating previous discussion regarding cathode resistors in the output stage of the amplifier. Even in the grounded cathode, negative biased grid configuration, it's desirable to include a small value resistor in each cathode circuit. For the least amount of power loss, a 1 ohm, 1 % tolerance part is recommended. This enables accurate determination of the cathode current in each tube by measurement of the voltage drop across the 1 ohm resistors.

Grid bias "taps" usually provided by transformer manufacturers will provide a D.C. voltage of around 50 volts after rectification. It's not necessary to worry overmuch about the exact voltage provided as long as the transformer manufacturer notes that the tap is present. Older designs usually provided no regulation for the negative supply, even though plate voltage varied considerably under load.

This was for cost considerations - because an additional gas regulator tube would have been required at the time that these circuits were originally designed. Adequate regulation can now be provided by a single silicon diode at a cost of far less than \$1 U.S., so there's no drawback to regulating the grid voltage supply.

Here's a typical half-wave rectifier, using a "tapped" power transformer:



Note that the two diodes have markings that indicate specific polarization, as does the capacitor. These polarities must be observed when installing the components.

We can make an immediate selection for diode D1 by picking one that has a PIV (peak inverse voltage) rating of at least four times the estimated rectified DC voltage (about 50 volts), so we can select a rectifier diode rated at 200 volts or more. A common selection would be part number 1N4003. Additionally, given the minimal current requirements of this circuit, the filter capacitor value is not critical, any value over about 10 uF is acceptable so long as the voltage rating is twice as high as the estimated D.C. voltage, a voltage rating of about 100 V would be adequate.

Here's how the circuit functions: Alternating current (A.C.) passes through the current limiting resistor, R1, and is rectified by diode, D1. The combination of R1 and C1 filters the ripple component from the rectified voltage and the capacitor, C1, is charged to a D.C. voltage that is approximately:

D.C. voltage ~ 1.4 x A.C. input voltage

If the RMS A.C. voltage of the "tap" is known, and if the load current is small, a fairly accurate estimate for the D.C. voltage can be made from this expression.

The "zener" diode, D2, provides adequate voltage regulation for the negative supply. This diode maintains a fairly constant voltage across its terminals, provided that the input voltage exceeds the zener voltage. Zener diodes have a maximum power dissipation rating that must not be exceeded. Proper selection of current limiting resistor, R1, will limit dissipation to a safe level.

In order to keep the D.C. voltage high and the A.C. ripple low, as we've mentioned previously, very little current must be drawn from the rectifier circuit. The vacuum tube grids will present no problem since they are extremely high impedance. We can choose the values of the two potentiometers, P1 and P2, and the lower voltage limiting resistors, R2 and R3 to maintain a high impedance. The values are not particularly critical and 1 Megohm is a common potentiometer value, so we can select that value now without analysis or further consideration.

We can make other assumptions to simplify the design procedure. If we are using a transformer that has a traditional grid "tap", the rectified output voltage will be about -50 volts. It's not likely that any commonly used power tubes will require grid bias in excess of about -30 volts, so let's select a zener diode with a regulating voltage between those two voltages - a suitable part would be 43 volts with a power dissipation of one watt, a commonly available diode. If we allow the diode to dissipate 1/2 watt, it will be adequately de-rated. Calculating the maximum current for this dissipation:

$$P = E \times I \quad \text{and} \quad I = P / E$$

Where P is power (in this case 0.5 watt), E is current in amperes and E is the zener voltage, inserting known values and solving for I:

$$I = 0.5 / 43 = .012 \text{ amperes (12 mA)}$$

The maximum current that will be drawn by the parallel 1 Meg potentiometers will be (using Ohm's Law):

$$I = E / R \quad \text{and substituting values}$$

$$I = 43 / 500,000 = 0.9 \text{ milliamps}$$

500,000 is the parallel resistance of the two 1 Megohm potentiometers

The purpose of determining the current is *to verify that it is considerably less than the amount of current that we allow to flow through the zener diode* (the load current must always be much less than the zener current, for good voltage regulation). A rule of thumb is that load current should not exceed about 1/5 of the zener current, obviously that rule is satisfied in this case.

Let's now determine the value of the current limiting resistor, R1, using Ohm's Law and the known values. The voltage drop across the resistor will be the difference in the estimated *rectified DC* voltage and the zener voltage. The current will be the calculated maximum of 12 mA:

$$I = E / R \quad \text{and} \quad R = E / I$$

inserting the known values and calculating:

$$R = (50 - 43) / .012 = 583 \text{ ohms, we can use a standard value of 560 ohms.}$$

The power dissipation of the resistor is given by any of the following:

$$P = E \times I \quad \text{or} \quad P = E^2 / R \quad \text{or} \quad P = I^2 \times R$$

Since we already know the voltage and current, we'll use the first expression:

$$P = (50 - 43) \times .012 = 0.084 \text{ watts}$$

Which must be de-rated by a minimum factor of 2, which would result in 0.168 watts. In this case, however, since we don't know the EXACT rectified D.C. voltage, we'd like to be on the safe side, so instead of doubling the calculated rating, we'll select a higher standard power rating of **1 watt**.

The purpose of the two resistors, R2 and R3, is to limit the minimum voltage that the potentiometers can be adjusted to provide. If, for example, the voltage adjustment range was allowed to drop to zero volts, the output tubes would draw the maximum amount of current. The tubes, output transformer and the power supply transformer would all be highly stressed and possibly damaged. This could occur during the initial "turn-on" of the circuit or perhaps making a careless adjustment of the potentiometers.

We can make a reasonable estimate for the lowest adjustment voltage from the normal operating grid voltage, previously determined from data sheet plate characteristics. (This voltage, Ec1, in our power amplifier design example is -23 volts.) By allowing the lowest adjustment voltage to be about 2/3 the normal control grid bias, we will probably prevent inadvertent damage to the circuit components.

Using the -23 volt grid, the minimum voltage is:

$$2/3 \times 23 = 15.4 \text{ volts}$$

We know that the value of each adjustment potentiometer is 1 Meg and we know that the maximum voltage is the zener voltage of 43 volts. Using Ohm's Law, we can solve for the current through one potentiometer. The voltage drop across the potentiometer is 43 - 15.4 volts, so:

$$I = E / R = (43 - 15.4) / 1,000,000$$

$$= .00000276 \text{ amperes or } 27.6 \mu\text{A}$$

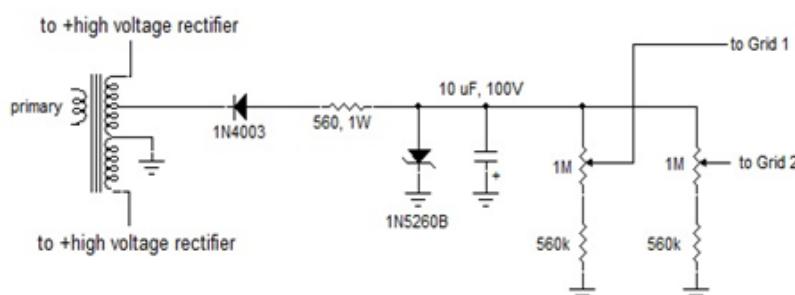
Again using Ohm's Law, knowing the current flow and the minimum voltage, we can solve for the value of R1 and R2:

$$I = E / R \text{ and } R = E / I$$

and inserting the known values we obtain

$$R = 15.4 / 27.6 \times 10^{-6} = 557,971 \text{ we'll use a standard value of } 560\text{k}$$

Here is the completed design of the negative bias supply:



12.6 Hum Reduction and Decoupling

An important consideration in amplifier design is "de-coupling" the individual stages from each other and also from the power supply. "De-coupling" is a common industrial term meaning "isolating". Isolating the amplifier stages, most especially the preamplifier and post amplifier stages (because they are located at the highest gain points in the amplifier) from the power supply is very important if one wants to keep "hum" at a minimum.

Selection of tubes for preamplifier application was once very important, since some tubes were specially selected for low noise, low hum applications. Many of the 12A - - 7 series are actually the same vacuum tube, made of the same component materials in an identical manner. During testing, should some of the tubes excel at certain parameters, they were set aside and marked with a new tube designation to denote superior features.

The 12AX7 family is probably the most frequently used dual triodes for preamplifier, postamplifier and phase-splitter applications. The low hum/noise version is the 12AX7A/ECC83/7025, the 12AT7 is sometimes used for low plate resistance applications, like reverb tank drivers.

Here's a table of inexpensive common dual triodes and their characteristics

Type	E _b Volts	P _b Watts	A _v @ R _p	Other
6BC8, 6BZ8	150	2.2	35 @ 5300	
6BQ7, BZ7, BS8	150	2.0	38 @ 5900	
12AT7/ECC81	300	2.5	60 @ 11000	
12AU7A/ECC82	330	2.75	17 @ 8000	
12AV6	100	0.6	100 @ 62000	
12AX7A/ECC83	330	1.2	100 @ 62000	1.8 uVrms hum
12AY7	300	1.8	40 @ 23000	Not for use in low hum applications
12AZ7A	330	2.5	60 @ 11000	
12BH7A	300	3.3	17 @ 5300	

It's not possible to rely on "hum" characteristics that appear on data sheets now. Vacuum tube hum is a matter of taking one's chances with the quality of an imported product or personally measuring performance. I performed a test several years ago, replacing the preamplifier tube in an amplifier with a serious hum problem. The amplifier had three 12AX7 tubes made by the same manufacturer - probably from the same production lot. I switched the tubes around in an attempt to find the quietest one. I found differences in the hum level of the three tubes of 20 dB!

Apparently the mechanical configuration (especially in the filament to cathode area) is not controlled as well as it should be - but that's not an informed opinion

since I have no experience with vacuum tube manufacturing. My point is that when all other options for reducing hum have been exhausted the limit of the particular tube type (especially the preamplifier tube) may have been reached.

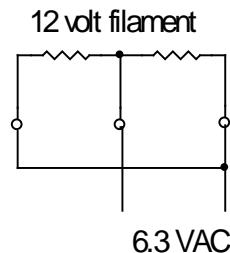
Not only is it critical to filter any direct path leading to the power supply, it's also important to minimize magnetic coupling to the high gain stages from the power supply transformer. The only way that this can be implemented, short of providing a separate power supply in a different chassis, is to locate the power supply transformer as far away from the preamplifier as possible. (Locating the power supply in a separate chassis was not uncommon for high performance equipment in past years.)

Another source of "hum" is the direct modulation of the cathode by the A.C. powered filaments in the preamplifier and post amplifier stages. I believe this to have been the problem with the new amplifier to which I referred above. Several methods have been used in the past to reduce this effect.

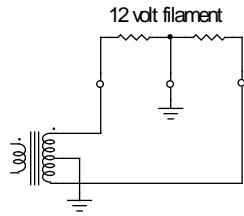
It's always helpful to keep the filament lead path (from power supply transformer to tube filament connections) as short as practicable and to twist the two leads together. Twisting the leads, so that there are four or five turns per inch of length, reduces the impedance of the wiring. (Low impedance lines have lower coupling than untwisted high impedance lines.)

Preamplifier tubes are almost universally provided with 12 volt filaments that are "tapped" at the filament center point. Filaments are tungsten alloy resistor elements that, like all resistors, produce heat when current is passed through them. As described in the initial chapter of this document, the heat is used to liberate electrons from the cathode, establishing current flow from cathode to plate.

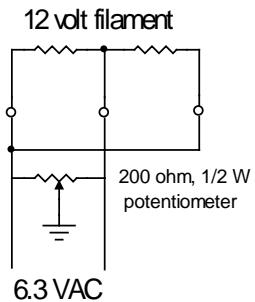
We normally operate the center-tapped filaments of 12 volt tubes at 6 colts by connecting them as shown below, letting the resistor designations in the schematic represent each half of the filament:



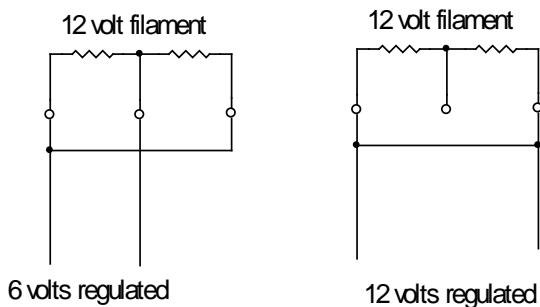
If the filament supply of the power supply transformer is center-tapped, then a better configuration is as follows:



Establishing a "real" chassis ground helps keep "hum" to a minimum. Usually, the best grounding configuration is the so-called "star ground", so-called because the various component leads are grounded to a common, central node. A variation on the first configuration that allows some adjustment to minimize hum is sometimes effected like this:



There are several more drastic "solutions" to the hum problem. Since the problem normally is caused by poor layout or poor grounding, these options won't help, as a rule. They have been used in the past for high-fidelity stereo power amplifiers, where power supply noise needs to be some 100 dB below the signal, for example. The simplest option is to operate the filaments from a regulated D.C. voltage:



The remaining technique is not easily implemented by inexperienced technicians (esoteric and rarely used). The technique consists of applying a negative bias to the cathode, when referenced to the filament, and adjusting all of the other bias voltages applied to the tube accordingly. An identical effect can be achieved by operating the filaments from a positive voltage, with respect to the cathode. The filament voltages, as referenced in RCA Receiving Tube Manual, RC-30, range from 15 to 40 volts more positive than the cathode bias voltage.

The purpose is to repel electrons emitted by the filament away from the cathode, thus preventing them from being consequently transmitted (and even amplified) to the plate. Except for some high power triodes, direct filament signal connection is uncommon. It's desirable for the signal path to flow from cathode to plate, modulated only by the control grid. (The filament - ideally - would have no influence on the signal.)

Some amplifier manufacturers (e.g. Gibson) went to great lengths to obtain hum reduction in their designs. Power supplies were housed in a separate chassis and located remotely from the amplifier chassis (usually in the bottom of the speaker enclosure). Shielded wiring was used for interconnections between amplifier and power supply.

There was an obvious audible performance improvement in these amplifiers, compared to the more popular Fender configurations. Although I've owned many vacuum tube amplifiers (still have seven of them), the most versatile amplifier may have been a Gibson GA-79RVT, purchased new in 1961. It was fairly loud (by jazz standards) at 35 watts and free of noise/hum when no input signal was present.

Incidentally, another major difference between the two amplifier manufacturers mentioned is that Fender universally used negative feedback in their amplifiers while Gibson did not.

Since all guitar amplifiers have high gain, another consideration of decoupling is to prevent a high level signal from leaking back to a previous point in the amplifier chain causing oscillation (feedback). As in preventing hum, circuit layout is important and a little common sense is helpful. A useful rule of thumb has always been to lay out a circuit in the same manner as a well-drawn schematic, without lines crossing one another, the signal following a defined path in a fairly straight line.

The path most likely to allow leakage is the positive plate supply line that is common to all of the amplifier stages. Successful designs employ a simple strategy that takes advantage of the fact that current consumption always increases as the signal path moves from preamplifier toward power amplifier. Additionally, so does the required plate voltage increase as the signal increases in gain. This allows the construction of a lowpass filter network, consisting of series resistors and shunt capacitors, placed in the plate voltage supply path. It's fairly easy to design such a network and analyze the effective decoupling.

The design goal of the power supply decoupling network is to reduce hum from the power supply, at each stage of the amplifier, well below the signal level. The gain of the stages must be accounted for when implementing the decoupling filter. In chapter 6, a spreadsheet was presented that performed many calculations related to amplifier performance. One of the parameters that the

spreadsheet calculates is the cumulative gain, at each stage of the amplifier chain. We can use the cascaded calculations to assist us in determining the amount of power supply filtering indicated at each stage.

The process consists of calculating the voltage dropping resistor between each stage and then using this resistance along with the calculated reactance of the filter capacitor at each stage to estimate the ripple attenuation. The approximation is as follows:

$$V_{\text{ripple out}} = V_{\text{ripple in}} \times jX_c / (R_s + jX_c)$$

where R_s is the series resistance and

$$jX_c = 1 / (2 \times \pi \times 120 \times C)$$

where C is capacitance in Farads ($1 \mu\text{F} = 1 \text{ Farad} / 1,000,000$) and π is about 3.14

As an example, for a series resistor of 10k, a filter capacitor of 1 μF and a power supply ripple voltage of 2 volts:

$$jX_c = 1 / (2 \times \pi \times 120 \times C)$$

$$= 1 / [2 \times 3.14 \times 120 \times (1 / 1,000,000)]$$

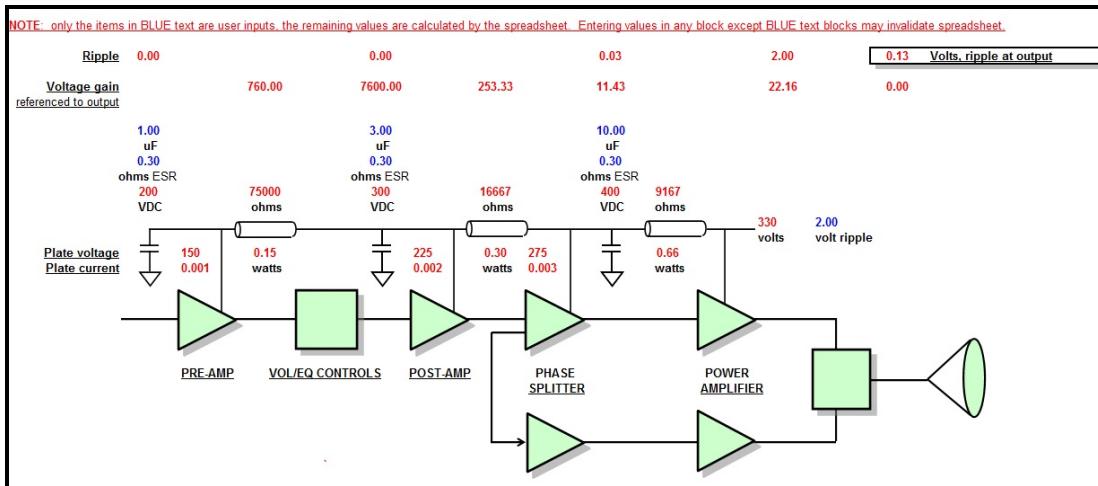
$$= 1 / [754 \times (1 / 1,000,000)] = 1326.3$$

$$V_{\text{ripple out}} = V_{\text{ripple in}} \times [jX_c / (R_s + jX_c)]$$

$$= 2 \times [1326.3 / (10,000 + 1326.3)]$$

$$= 0.234 \text{ volts}$$

note that voltage units are consistent, in other words, if the power supply ripple voltage is expressed in volts RMS then the output ripple is also in volts RMS. A variation of the spreadsheet used for general amplifier analysis can be used to calculate resistor values for each stage (this spreadsheet is included in the general workbook regarding vacuum tube amplifier design). The user can input the capacitor values for each stage and the spreadsheet will calculate power supply rejection throughout the chain, accounting for the gain of each stage in the process. Here's an example:



The user is required to enter a value for ripple voltage (obtained from the percentage ripple in the power supply spreadsheet. To use the ripple voltage from the spreadsheet, which is percentage, divide the percent ripple by 100 and multiply the result by the power supply voltage:

$$(\% \text{ ripple} / 100) \times \text{power supply voltage} = \text{ripple in volts}$$

After the ripple voltage is entered, various filter capacitor values may be tried until the desired level of ripple at the output is achieved. *All of the calculations, including resistor values and power ratings are based on accurate estimates of plate voltage and plate current, as they are entered on the amplifier performance estimate spreadsheet by the user* (see chapter 6). Chapter 22.5 discusses this topic in more detail.

12.7 Hum Issues Related to Parts Layout

Although this chapter is related to power supply design, most of the hum problems in an amplifier originate from power supply wiring and layout. Some important aspects of maintaining a clean signal path have been mentioned. Let's review them and add a few more as we consider how to keep hum, stray pickup and feedback to a minimum.

12.7.1 The Amplifier Schematic

A good amplifier parts layout commences with a good schematic - take your time making this invaluable piece of documentation. It doesn't have to be pretty but should reflect the concerns that you will later have to address when determining the configuration of your amplifier chassis.

As an example, a "clean" schematic would not include signal paths crossing over power supply wiring or over other signal paths - the drawing would look messy if multiple lines crossed one another. Equally, the actual layout of the circuit within

the chassis shouldn't include signal wiring or components crossing over power supply wiring - this allows accidental coupling. Accidental coupling can result in increased hum and noise or even feedback.

If one lacks experience in building these types of circuits, it's best to follow some fixed rules that govern how the schematic is created:

Signal flow goes from *left to right* (input to output)

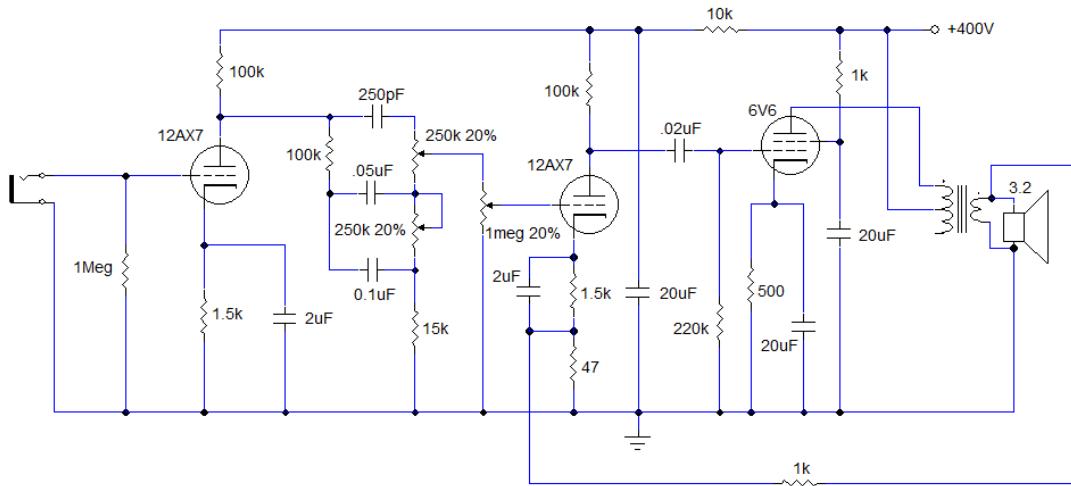
Ground connections are located *below* the signal path

Filament connections are located *below* the signal path

High voltage connections are located *above* the signal path

Power supply is usually (but not always) drawn separately from the circuit
- the bottom area of the drawing is typically used for this purpose

By following these conventions, it's easy to see - at a glance - all of the ground connections, the high voltage connections and the filament voltage connections. Here is an example of a schematic drawn in this fashion:



Note that most schematics don't strictly follow this format although it is a sensible one. With experience, one learns to intuitively avoid problems with the mechanical layout regardless of how the schematic is structured. For less experienced designers, I highly recommend the above practice.

12.7.2 Stray Coupling

After the design is established and the schematic diagram created, one should spend some time studying the schematic. Consider the two types of coupling -

electrical and magnetic - and how hum and external interference could be introduced into the signal path at various points in the circuit. (The high gain stages - preamplifier and post-amplifier - are the most critical areas.)

Electrical coupling is caused by the accidental creation of a capacitor between two different parts of a circuit. A capacitor is formed any time two conductors are placed near each other. The magnitude of the capacitance is proportional to the areas of the conductors divided by the spacing between them. Coupling is increased if there is insulating material between the conductors.

Impedance is proportional to the square root of inductance divided by capacitance, the larger the amount of capacitance between two conductors, the lower the impedance. This is why we tightly twist filament wiring, to lower the impedance - twisting the wires brings them closer together, increasing capacitance by decreasing the distance between them and increasing the area. Magnetic coupling is caused by a magnetic field that intersects conductors in the circuit which causes current flow - exactly like the operation of a generator. The magnetic field can be a result of the magnetism generated by either a transformer or the simple process of current flow in a wire. Magnetism is proportional to current flow - the higher current, the stronger the field. This makes filament wiring especially troublesome because of the large amounts of current required.

For either type of coupling, *parallel* wires provide the strongest coupling, which is undesirable. For all power supply voltage wiring, *high impedance* wiring is also undesirable. High impedance, for our purposes, means wires that are separated from the chassis by more than 1/4 inch and any *pairs* of wires that are not tightly twisted.

The exception to this is all of the wiring and components in the signal path where it is desirable to maintain relatively high impedance and fairly short lengths of wiring. The high frequency response can be impaired if the signal connection impedances are too low.

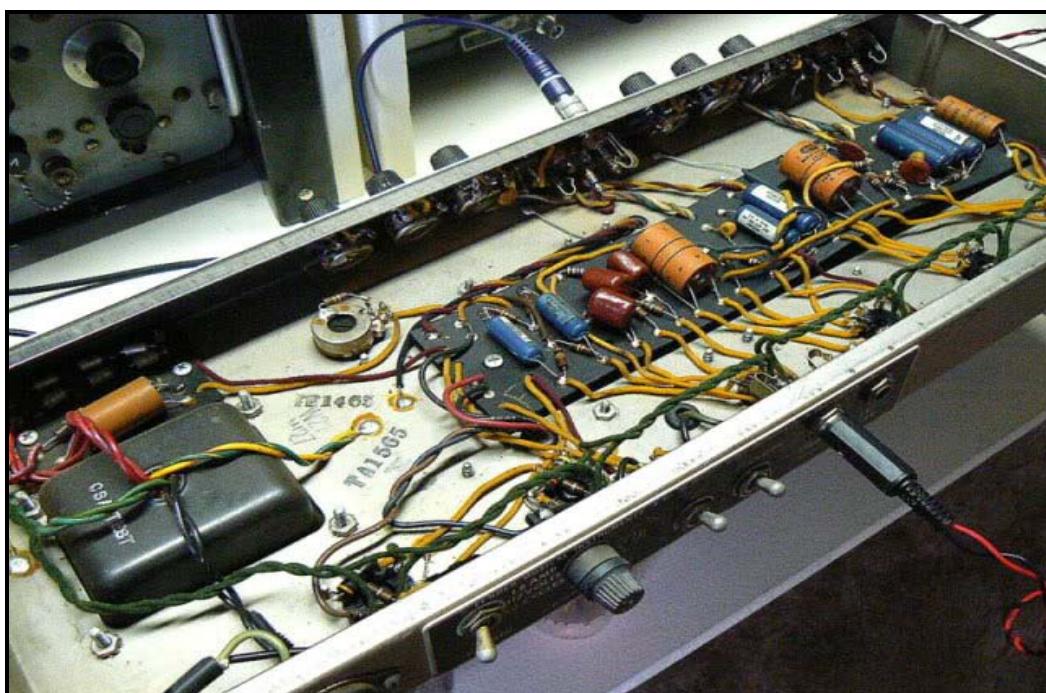
Ground loops can be a source of noise and hum (even feedback). A ground loop is created when all circuit grounds are not connected to a single point. Because the ground connections are made through the chassis material, a certain amount of resistance (and inductance) isolates the various ground points. This is because the steel chassis material has significant resistance (compared to copper wire, for example).

The term "star ground", used elsewhere here, refers to a single-point ground connection. This is recommended by many but there are drawbacks to the practice when long lengths of wire are required to establish a "star ground". Long lengths of wire create stray coupling, as previously noted and are generally to be avoided.

My personal practice is to use "star grounds" only where high currents are flowing. A good example would be the output stages of almost any amplifier. The preamplifier and post-amplifier stages - where small currents are typical - are uncritical and I recommend maintaining the shortest possible connection from cathode to chassis ground at these stages. The phase-splitter tube(s) are equally uncritical for "star ground" considerations.

12.7.3 An Example of a Successful Layout

The photograph below depicts a 1966 Fender Bassman chassis, removed from its cabinet for the purpose of adjusting the output tube bias conditions. We'll use this as an illustration of a good mechanical layout.



The power transformer is located at the left side of the chassis. Note that all of the wiring entering and leaving the transformer cover is tightly twisted. This lowers the impedance of the wiring and inhibits coupling AC hum to sensitive areas of the circuit. It is also good practice to "dress" the wiring close to the chassis. This also lowers impedance and prevents AC hum from entering the signal path. The sets of twisted pairs are all at *right angles* to one another - another good technique for reducing coupling.

The tightly twisted green wires at the lower edge of the chassis are the 6.3 volt AC filament supply. The wires are connected to all of the tube filaments, starting with the output tubes, then routed in sequence to the remaining tubes. The wires are twisted for the reasons described above.

The high voltage wiring is connected to the eyelet fiber board that is stuffed with the resistors and capacitors required by the circuit. All tubes are connected to the high voltage line through resistors except for the output tubes which are connected through the output transformer. Short lengths of yellow wire form the interconnections between the fiber board and the tube sockets. Note that these wires are at *right angles* to the filament wiring. This is the best way to minimize coupling the AC filament voltages to either the control grid or the plates of the individual stages.

From the perspective of this photograph, the input is at the *right* end of the chassis and the output at the *left* end. The power transformer is located at the output end of the chassis so that magnetic coupling is as far away from the high gain input stages as possible. *If the transformer was located close to the input stage, AC hum would be a serious problem, no matter how well other shielding and suppression techniques are implemented.*

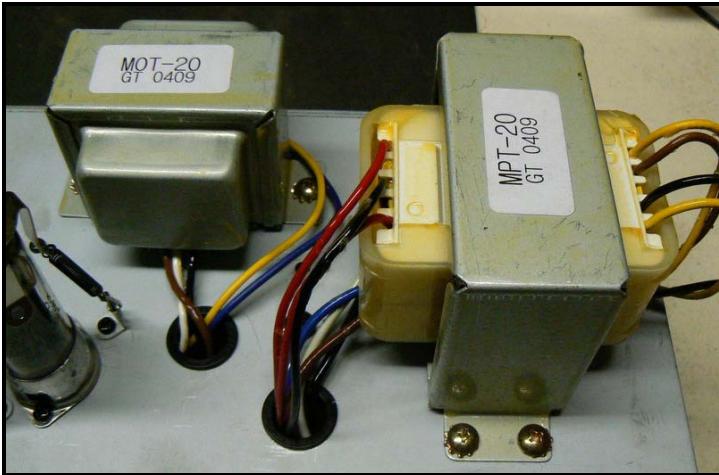
The output transformer, which is not visible in this photograph (it is on the far side of the chassis), is also located at the output end of the chassis. This is to prevent magnetic coupling from the output to the input, which would cause audio feedback.

12.7.4 Proximity and Orientation of the Transformers

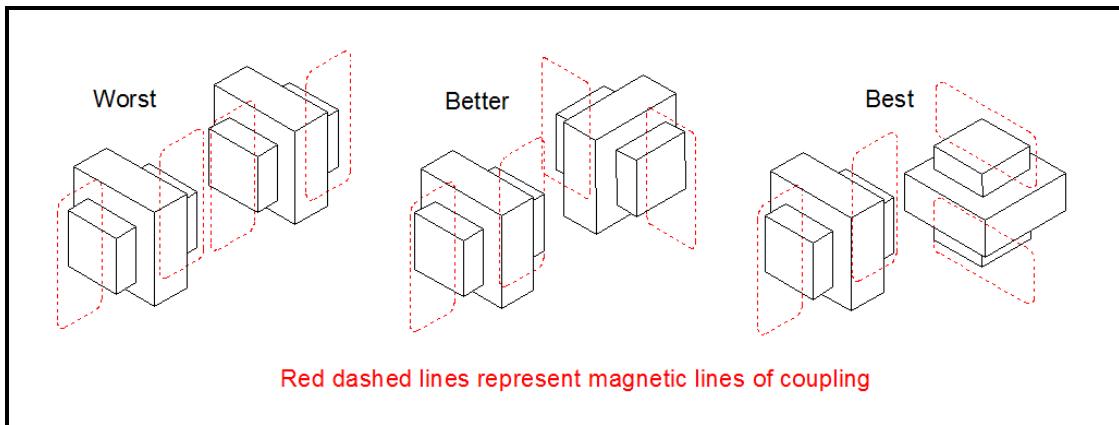
Since the two transformers are adjacent (so as to distance them from the amplifier input) a common precaution is to orient them at right angles to one another. This is to prevent coupling AC hum from power transformer to the output transformer. The coupling is magnetic and is weaker when the laminations of the transformers are not parallel with one another.

Below is a photograph of a small 25 watt amplifier that more clearly shows the orientation of the two transformers. Note that the output transformer is completely shielded with a steel shell to minimize coupling from the power transformer. Magnetic shielding can only be effected by magnetic materials, non-ferrous materials are not useful.

This orientation might be adequate but better still would be to mount the power supply transformer (on the right of the photo) in a manner similar to the Fender Bassman power transformer shown above. The Bassman orientation is such that no lines of stray magnetism (between the two transformers) are parallel (see "best" configuration in the sketch below).

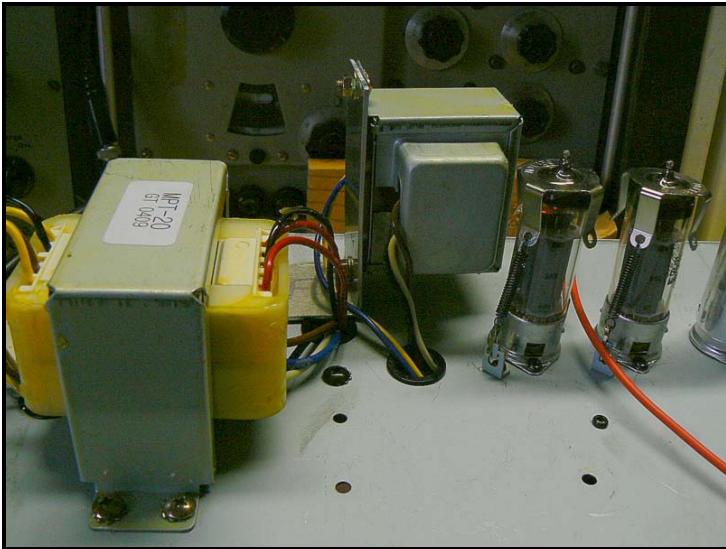


Here are possible configurations in orientation between the two transformers:

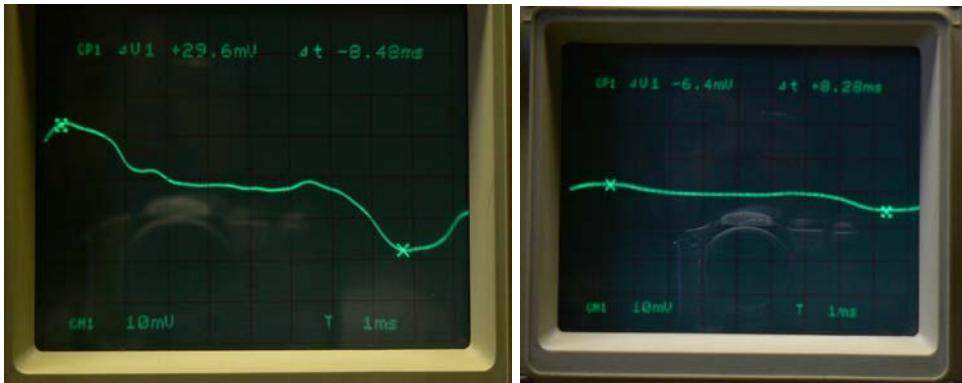


In the worst case, the two transformers are parallel to one another. In the next configuration, one transformer has been rotated 90 degrees in the "Y" axis. The best configuration results from the rotation of one of the transformers by 90 degrees in both the "Y" axis AND the "X" axis. In the "best" configuration, the power supply transformer is universally the device that is rotated as shown at the right side of the illustration. The magnetic lines of coupling are least parallel in this orientation.

The following is the same 25 watt amplifier depicted previously (photographed from the opposite side of the chassis). A temporary aluminum bracket has been installed to the chassis so that the output transformer can be rotated 90 degrees vertically from its previous position.



These are photos of an oscilloscope screen, displaying differences in hum level before and after the output transformer was rotated 90 degrees vertically from the initial position. The voltage scale is the same in both photographs. The photos were taken with the standby switch disabled - there was no high voltage applied to the amplifier and therefore no amplification.



The example is fairly simple, illustrating mutual coupling between two transformers. The output transformer is terminated with an 8 ohm load and the power supply transformer is loaded only by filament current.) Voltage magnitudes are not that important however the *ratio* of improvement IS important. Coupling was reduced by 14 dB in this experiment.

The effect of transformer orientation on magnetic coupling and power supply hum is critical. The possibility of power supply coupling to sensitive parts of the amplifier circuit, like the preamplifier, is especially important.

Summarizing this chapter, with a clean schematic and a little thought, a practical parts layout can be created. Tracing the signal and power supply flows in the diagram and observing a few precautions suggested by common sense will keep

noise and hum to a minimum. The Fender "Bassman" layout is a good one to emulate.

12.8 Testing the Power Supply Circuit

Before connecting the supply to the amplifier it has been designed to power, the output voltage must be measured under load. Power supplies are seemingly simple circuits but because of the "parasitic" elements associated with the power transformer, the prediction of accurate output voltage is not possible without having more information than the typical catalog provides. When ordering parts for the amplifier design, be sure and order a high power load resistor to test the power supply. The value of the load resistor is:

$$R = \text{Power supply voltage} / \text{current consumption}$$

The design values for Eb and the total high voltage current consumption (all plate currents and screen currents) are used in the calculation. Don't include filament current in this calculation. The power rating of the load resistor must exceed:

$$P > \text{Power supply voltage} \times \text{current consumption}$$

We don't need to excessively de-rate the load resistor for power supply testing because it will be under load for a very brief period of time - just long enough to make a voltage measurement. After first confirming that the power supply is functioning (providing a D.C. high voltage output), allow the bleeder resistor to drain the filter capacitor and then temporarily install the load resistor across the output of the high voltage power supply. Connect two DMMs across the load resistor.

Adjust one DMM to measure D.C. voltage in the correct range (usually the 1,000 volt maximum range). Adjust the other DMM to measure A.C. voltage in the correct range (either the 20 volt range or the 2 volt range, usually). Turn on the power supply long enough to note the D.C. voltage and the A.C. voltage, then turn the supply off. Compare the readings of both voltages under load with the design values to determine that they are acceptable. (The D.C. voltage is the high voltage supply for the vacuum tube plates and the A.C. voltage measured is the ripple voltage or power supply "hum" voltage.)

If the D.C. voltage is too high, the current limiting resistor value may be increased (this will also reduce the A.C. voltage (the "ripple" voltage)). If the D.C. voltage is too low, the current limiting resistor value must be lowered or the filter capacitor value increased. (This assumes that the correct power transformer has been selected and installed.)

Note that it is normal for the power supply voltage to be higher than the design value until the amplifier is driven at full rated output power. When driven at full

power, the power supply average voltage and current should be at or near the design values (+/- 15% wouldn't be uncommon).

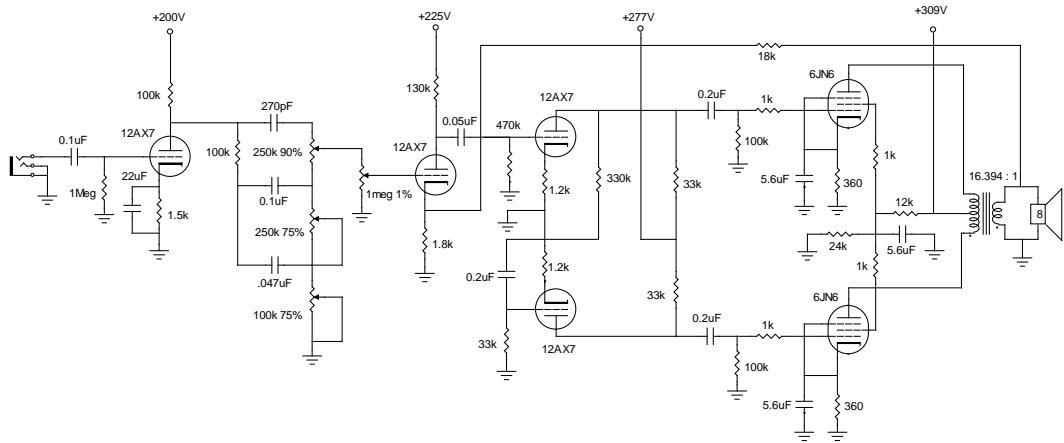
In chapter 22.0, we'll use several spreadsheets to assist in the design of the power supply and to aid in hum suppression. That chapter also offers other suggestions pertaining to the design process. Even if one chooses not to use the spreadsheet for power supply design, reading the applicable parts of chapter 22.0 is recommended. In chapter 26.0, an inexpensive high voltage supply is described that is appropriate for amplifiers in the 15 watt output power range.

13.0 Completed Guitar Amplifier Circuit

A guitar amplifier chassis in the process of assembly:



As we progressed through the various design exercises, we collected a number of individual circuits for each function in the amplifier chain. Except for the bias de-coupling circuits, and the high voltage supply line that connects the circuits, here's the signal chain, from preamplifier to speaker, for the medium power, four-tube amplifier.



Some of the coupling (blocking) capacitor values differ from those shown in the chapters discussing the design of the individual circuits. When the entire amplifier is assembled (at least schematically) the parallel load resistances of each stage becomes easier to determine and new values of capacitance determined.

In a high voltage circuit, like this one, it's always preferable to use the lowest value capacitor practical. This is because the cost of capacitors is related to their value and to their *direct current working voltage*, frequently referred to as "DCWV". The least expensive devices are universally those with low capacitance value and low working voltage.

Examining the above circuit, one may observe striking similarities with other popular amplifiers. If so, one message has been transmitted successfully: vacuum tube amplifiers are not particularly sophisticated devices and there are a limited number of ways to achieve a particular function without adding needless circuit complexity. A good design is one that simultaneously fulfills the goals of performance, cost and reliability. Therefore it's not surprising that functional designs tend to follow a similar pattern.

An expression commonly used in the defense/aerospace industries is "creeping elegance". The term referred to the inclination for engineers (frequently at the urging of marketing/sales personnel) to add features unnecessary to the basic function of the circuit. Some modern vacuum tube amplifiers are classic illustrations of this term, tending to be complex beyond the basic functions required.

The perpetual popularity of the old 1950s designs, however, suggests that many understand the basic function of a guitar amplifier *which is to accept the audio signal from a guitar and make it louder*. The Fender "Deluxe Reverb", mentioned frequently in the book, is a classic example of a good design: there are just the number of components required to fulfill the function of the amplifier and no more.

13.1 Test Strategies

The chances are good-to-excellent that if one assembles an entire guitar amplifier and then powers it up, it won't function properly. Despite the deceptive simplicity of the schematic representation, there are a number of solder joints, a number of potential component value possibilities, so there is a substantial probability of error.

An experienced technician, who carefully checks the assembly before attempting to apply power, might achieve reasonable success but the rest of us need to consider how to go about verifying proper performance without permanently damaging expensive components.

Before discussing testing and measurement, let's address safety - when one needs to perform measurements or repairs on a vacuum tube amplifier, involving making the circuit accessible, the following steps need to be performed in exactly this order:

Disconnect the device from the A.C. outlet

Remove cover(s)or chassis to provide required access after allowing at least two minutes to elapse (for draining filter capacitors).

Carefully, using insulated tools or insulated jumper wire, connect the power supply filter capacitor terminals together, discharging the capacitor. Don't do this momentarily; maintain the connection for a few seconds to insure that the filter capacitor is fully discharged.

The following thoughts are a result of evolving work habits and some years of experience on the test bench. Here's a brief summary of some thoughts that can be used to structure a test strategy after a project is completed:

Test each individual circuit - as it is assembled - rather than attempting to test the entire amplifier.

Check and re-check the wiring, component orientation and polarities of all components before applying power.

If an adjustable voltage power supply with current-limiting is not available, verify that the power supply to be used is fused or otherwise protected.

Collect all design paperwork, specifications, data sheets and calculations for ready reference.

Make sure that you know what you want to confirm with each test and what instrumentation will be required to confirm it.

Make a sketch of the test set-up. Compare the sketch with the circuit schematic and think about the test for a few moments. Try to estimate the effects of testing and confirm that neither the test instrument(s) nor the circuit under test will be damaged by the test process. (An example of potentially catastrophic damage might be the connection of a signal generator to a high voltage point in the circuit without adequate high voltage blocking.)

Clean and organize your work area before commencing testing so that distractions are minimized - dangerous high voltages will be present and your attention must be tightly focused on the task at hand.

Install test connections (clip leads for power supplies and test instrumentation) properly and then secure the wiring to the test bench with masking tape, so that the connections can't be easily dislodged or accidentally touched.

Remove all vacuum tubes from their sockets except for the circuit that is to be tested.

Before powering up any circuit, read over the specification for the circuit and carefully note the expected bias conditions.

Check the settings of all test instruments; verify that the proper range is selected and that no dangerous condition can result by probing a circuit point with an instrument that is not configured for the possible voltage/current stress it may encounter.

Apply filament voltage to the stage to be tested and observe that the filaments are glowing, wait a moment or so before further testing.

After you are assured that all safety precautions have been observed, instrumentation is configured according to your test diagram and that the filament has reached operating temperature, gradually power up the circuit to operating high voltage requirements.

This is best done with a variable power supply that is current-limited, to protect the supply and the circuit under test. As the supply is slowly brought up to operating voltage level, constantly monitor the current verifying that it never exceeds the design current for the stage.

If an adjustable supply is unavailable and the amplifier power supply has to be used to operate the circuit, it's recommended that a "Variac" (variable line voltage transformer) be employed to gradually apply voltage to the supply and consequently to the circuit under test. (NOTE: this will require a separate filament power supply - a 6.3 VAC transformer

operated from line voltage, this is an inexpensive and highly useful addition to your collection of test equipment.) As above, monitor the plate current of the stage as the Variac voltage is increased.

Once operating voltage has been reached, verify that the stage is biased correctly by measuring the plate current or by measuring the voltage drop across the cathode resistor, confirming that the measured voltage is the same as the design voltage in your design notes/calculations.

After bias conditions are determined to be correct, a test signal may be applied to the circuit input and measured at the circuit output. It's important that the signal generator be adjusted to provide the input level that will be present in the completed amplifier. Refer to your design notes, specifications and the amplifier spreadsheet values for individual circuit voltage gain and signal voltage levels.

Record ALL of the test conditions and all data obtained. Keep the test data with all of your design records. Carefully document ANY changes made to the amplifier by marking a copy of the schematic. Don't omit anything from this process. You may desire to make a similar amplifier in the future and all of this information will be helpful in making product improvements or lessening test and troubleshooting time.

Personal safety is the number one concern during all test activity, all else is secondary. If excessive current is measured during the power-up procedure, stop immediately, power down and resolve the circuit problem before attempting to power up again.

Testing is always easier with good equipment. An oscilloscope would be the preferred measurement choice in almost every situation but good work can be accomplished with limited tools. Inexpensive DMMs (at least two - and four are much better) can measure almost all of the electrical parameters that need to be determined. Remember, when making signal measurements, that DMM measurements of A.C. voltages are always RMS, never peak, or peak-to-peak, measurements. Refer back to the earlier chapters or to the Appendix to revisit the conversion from/to the various units.

The following is the order in which the various circuits should be tested:

Power supply

Output stage

Phase-splitter

Post amplifier

EQ and volume controls

Preamplifier

By structuring the tests from output to input, one can be assured that the chain is capable of driving the output to full power. As a minimum, the following data should be collected for each circuit tested:

Power supply conditions (voltage and current)

Load resistance

Source resistance

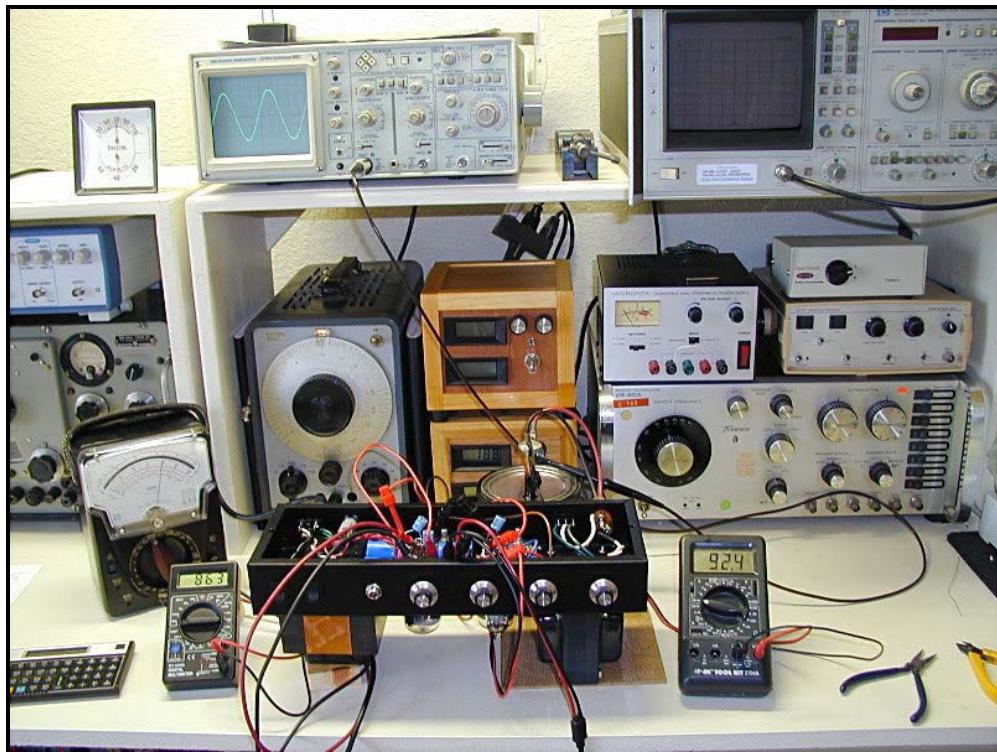
Output voltage measured across load

Input voltage from source for the measured output

If measured, the D.C. voltage at various points in the circuit

(Although some of this information may not be immediately useful, a future application might find the data helpful to the designer.)

A completed 40 watt guitar amplifier chassis under test driving a "dummy load":



Two DMMs are measuring the voltage drop across 1 ohm resistors connected to the cathode of each output tube, the voltage (in millivolts) measured across a 1 ohm resistor is equal to the cathode current (in milliamperes). An analog voltmeter (the one with the "needle" on the left) is connected across the terminals of the "dummy load", measuring the A.C. RMS voltage from which the output power level can be determined.

This amplifier, under full drive conditions, is drawing 86 millamps from one tube and 92 millamps from the other tube, after being adjusted for a quiescent bias current of 45 mA, each tube. The amplifier is being powered with the two high voltage adjustable power supplies (wood cabinets) in the background. The adjustable supplies provide current limiting, for safety reasons, in the event of wiring errors or defective components. After proper operation is assured, the internal power supply will be connected.

13.2 Problem Solving

The number one issue with equipment that doesn't function properly is human error - in either design or in execution of the design. An error at the design stage, an assumption that's incorrect, improper interpretation of data sheet parameters, all can lead to performance issues, down to the wrong bias voltages and currents in individual stages. The importance of reviewing all of the data sheets (and comparing with the schematic) and reviewing the schematic (comparing it with the assembly) cannot be over-emphasized.

A practice that I've found very useful over the years is to allow time to pass before reviewing a procedure previously completed. For example, after concluding a certain design procedure, allow a day or so to pass before re-checking calculations. Other related tasks can be performed during that time. Similarly, after assembling a "breadboard" circuit to be tested, leave it on the bench for a few hours and complete another task before checking your assembly against the schematic. (This practice tends to minimize overlooking the same error several times because of a faulty assumption.)

As mentioned above, testing should always be performed on individual stages, where problems are more easily detected and corrected. Test data obtained from individual stage performance can be used to help isolate problems when the entire chain is connected. As with the design process, the integration of the various stages should start with the output and "progress" backward toward the input of the amplifier.

After all individual stages have been performance-verified (and data collected) start chaining the stages together by connecting the "dummy load" to the power amplifier stage and connecting the phase-splitter to the power amplifier. Referring to design notes, determine the correct drive level to be applied to the phase splitter in order to achieve full power level at the amplifier output.

After thinking about the test setup, satisfying yourself that all measurements can be safely performed (and will verify the performance of the parameters that you're testing) slowly power up the two stages, constantly monitoring the current consumption. Allow the filaments to come to operating temperature (as described above) before applying plate power. Apply drive from the signal generator and confirm that the output power is correct.

If the output power does not reasonably conform to the design value, recheck for wiring errors, especially the interconnection between phase-splitter and output amplifier. Confirm that the D.C. blocking capacitor (the coupling capacitor) is of the correct value and not connected improperly and that the source and load resistances are proper. Confirm that bias voltages at the cathode and at the plate of the phase splitter are correct. At this stage of assembly/integration/test 95% of problems encountered will be the result of human error.

When a problem is encountered, it's NEVER wise to attempt problem solving by arbitrarily replacing components. Power down the assembly and think about what you've observed for a few moments. Hasty moves and decisions are dangerous when dealing with high-voltage circuits!

Make sure that you fully understand the function of each component in the circuit. If measurements with a DVM can't isolate the problem that you suspect, relax. Consider a few alternative test setups that might confirm your diagnosis. Even if the new setup(s) cannot provide confirmation, additional data may help isolate the problem perhaps suggesting a solution after considering the problem for a while.

When satisfactory performance has been established, power down the setup. The proper way to power down any vacuum tube circuit is to first reduce the plate voltage to zero and *then* turn off the filament voltage.

Connect the post amplifier to the phase-splitter. Verify that all of the connections are correct and connect the power supply. Note that the power supply de-coupling resistors, discussed in chapter 12.5, are present in the high voltage line. (If they are not present, the proper bias conditions cannot be obtained.) Measure all of the circuit voltages in the post amplifier and confirm that they are in accord with design values.

Connect the signal generator to the post amplifier and adjust it for the correct level so that full power output can be achieved. If the tone controls and volume control are in the post amplifier test setup, set all controls to maximum or to obtain a "flat" response. If the design value of peak-to-peak voltages from the phase splitter outputs are not being produced, perform troubleshooting procedures as discussed above. If careful thought doesn't suggest further tests that might assist problem solving then review the design documentation for possible errors.

There should be no architectural problems at this stage of the process, although (occasionally) small errors can produce large problems. Use common sense (and your calculator) to verify that the signal voltages throughout the chain WILL produce the design value of output power and that all potential circuit losses have been accounted for in the early design. Most likely, this won't be a problem, but it's helpful to perform the review as a sanity check and to get your attention relocated from the immediate problem (so that you don't overlook the problem again).

A designer can become too narrowly focused when experiencing problems. I've experienced many problems that had simple solutions ... obvious, after some time had elapsed. It's not uncommon to make the same incorrect assumption several times during a continuous period of testing. Observing the setup, the interconnections and re-thinking the difficulties after a period of relaxation may produce a different approach to problem solving.

Connect the preamplifier to the remainder of the amplifier chain and verify that the complete assembly functions as designed. Connect your guitar to the amplifier and take a "test drive". During this point of testing, it wouldn't be uncommon, for example, to be slightly dissatisfied with tonal characteristics or the way certain controls function.

Now is the time to make corrections of personal choice, rather than waiting until the amplifier has been installed in an enclosure. The modifications should be based on your understanding of how each circuit functions - not by arbitrary substitution of components until something "nice" happens. As simple as these circuits are, common sense and an analytical attitude will usually converge on a solution more quickly than intuition.

14.0 Fidelity Considerations

There are many different sources of distortion and noise in vacuum tube guitar amplifiers. Some are more important than others and, for the most part, the negative effects are controllable. The following briefly describes some causes and possible remedies.

14.1 *Compression, Single-Tone Distortion*

Chapter 8.26 discussed single-tone distortion without giving the effect a name. A simple review of "linearity" provides the basic understanding for compression, limiting and the generation of single-tone distortion. An amplifier, operating in a linear mode has the following transfer function:

$$V_{out} = A_v \times V_{in}$$

where V_{out} is output voltage, V_{in} is input voltage and A_v is voltage gain. At the point where the output voltage starts to deviate from this expression, the amplifier begins to "compress". When an increase of input voltage produces no change in output voltage, the amplifier has entered "limiting" (also called "saturation"). The transition from linear operation to subsequent modes of operation generates harmonics, at first gradually, then increasing rapidly as limiting occurs.

As discussed in 8.27, this effect isn't necessarily bad-sounding, providing that only single tones are being amplified. This can be easily tested by playing single note lines on a heavily distorted vacuum tube amplifier. The effect is almost violin-like, the harmonics providing a singing, melodic effect and the sustaining notes (caused by hard limiting) also resembling the bowed strings of a violin.

Unhappily, when additional tones are introduced, the situation is not so pleasing to the ear. Aspects of that situation were also discussed in 8.27 and will be explored further when "multi-tone distortion" is discussed.

14.2 Harmonics in Push-Pull Output Stage

Discussions about the advantages of vacuum tube amplifiers on the internet frequently opine why tubes sound "nicer" than solid-state amplifiers. Let's first review some ground covered previously.

The most commonly described characteristic of vacuum tube amplifiers is "compression". Compression, as previously noted, describes the point at which the output signal of an amplifier is no longer related to amplifier gain. In linear operation, the input level, output level and gain are simply related by:

$$\text{Output Level} = \text{Input Level} \times \text{Gain}$$

this is a similar expression to the one that describes voltage levels except that here we infer *power* rather than *voltage* levels.

If the output level no longer follows this relationship, it is compressed or limited. We sometimes value this characteristic because we may want our dynamic range to be limited (e.g. reduce the level of over-zealous pick attack on certain passages or notes). Compression adds sustain to individual notes as well, which some prefer in certain circumstances.

Vacuum tube amplifiers compress rather slowly ("gracefully") while solid state amplifiers compress quickly and, within a few dB, enter "limiting" or "hard saturation". This harsh limiting isn't a particularly agreeable sound, unlike the slower compression of a vacuum tube that doesn't hard limit until after 4 to 8 dB or so of compression.

Next to compression is the second most often heard reason that vacuum tube amplifiers sound "prettier" than solid state amplifiers. It's because they generate more second harmonic content, or even order content - rather than generating third harmonic content or odd order content which, it is said, sounds "not so good".

This reasoning - of psychoacoustic origin - isn't an area with which I am familiar. The focus of this part of the discussion actually is to point out an ironic aspect of the conventional thinking.

We know that almost all vacuum tube amplifiers over 5 or 6 watts are designed in the configuration that we call "push-pull" (or balanced), when describing the output stage of the amplifier. This material was previously discussed in chapter 4.2.5.

Reviewing, the typical output stage consists of two vacuum tubes, driven by a phase shifter/power divider (phase splitter). Each of the two tubes is driven 180 degrees out of phase (e.g. if the grid of one tube is "positive" then the grid of the other tube will always be "negative"). Each of the tubes, in turn, amplifies 1/2 cycle of the complete signal cycle.

The output transformer combines the two half-cycles of the tubes into a single complete cycle of the audio signal *then* transforms the tube impedance to the same level as the speaker that is to be driven. We learned this from the previously mentioned section.

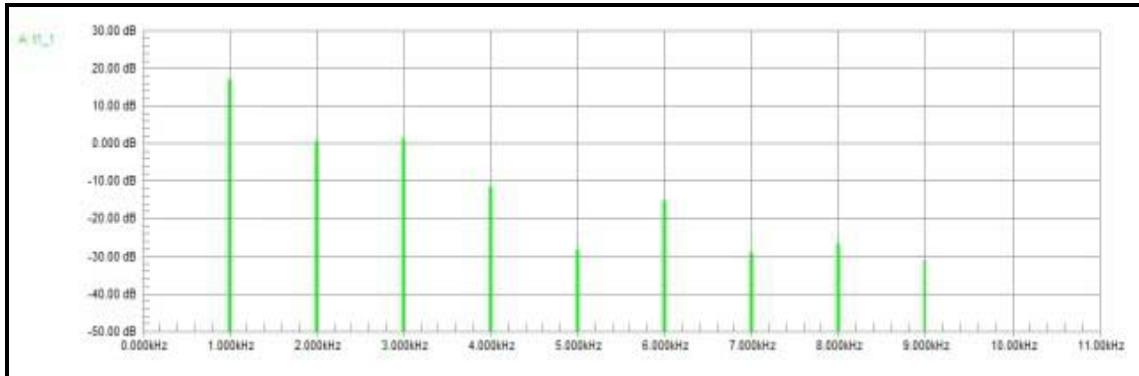
Not mentioned in that earlier chapter is another characteristic of push-pull or "balanced" output amplifiers: they suppress the second harmonic and all even order harmonics that are generated when the output tubes are overdriven. This is because, when the signals re-combine in the output transformers, the second harmonic and all even harmonics that are equal in amplitude are 180 degrees out of phase with one another when they combine in the output transformer.

That means that the voltages are algebraically added and cancel each other. Here's an example: assume that the even harmonic voltage originating from one tube is +1 volt. The even harmonic voltage from the other tube is out of phase so the voltage is -1 volt. The output transformer sums all of the voltages applied to it from the two output tubes, so the result of summing the two even harmonic voltage is:

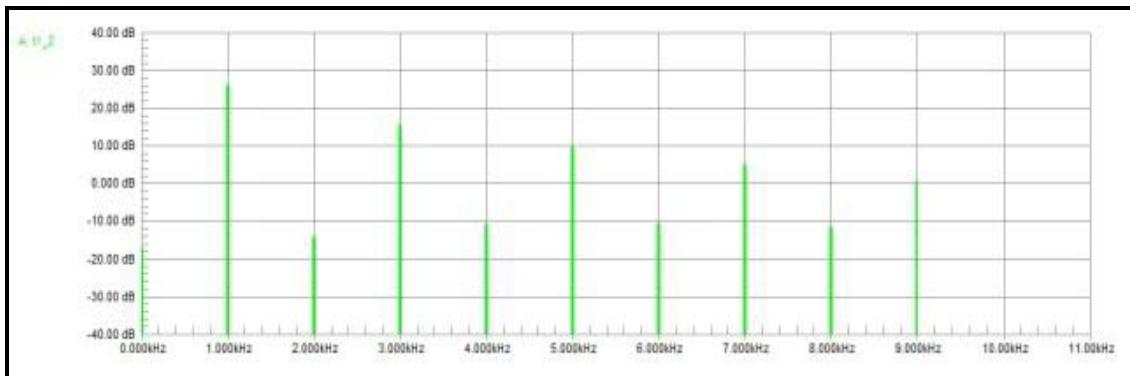
$$+1 \text{ volt added to } -1 \text{ volt} = 0 \text{ volts} \dots \text{no second harmonic is produced}$$

In reality, circuit imbalances, phase shift, differences between tubes and windings of the transformer preclude perfect even order harmonic suppression but most amplifiers do a good job of suppressing them to levels of a few percent of the desired fundamental signal level.

Here's a spectral plot illustrating what I've just explained. This plot is of a normal, single-ended (single tube, NOT push-pull) amplifier. The first line (far left) is the desired signal frequency, in this case it is at a frequency of 1 kHz. You can see numerous other signals extending to the right of the fundamental signal, these are all of the harmonics produced by the non-linearity of the amplifier. They are all lower in amplitude than the desire fundamental signal but they would be clearly audible as distortion. This is a computer simulation of a Fender "Champ":



Now here's a spectral plot of a push-pull amplifier, like all of the ones in current usage. The fundamental signal is visible at the left and, as above, the other lines to the right are harmonics. BUT, note that the even order harmonics are much lower than the fundamental signal frequency and also all of the odd order harmonics. This is a computer simulation of a Fender "Princeton" amplifier:



Note: this configuration suppresses the even order harmonics that are generated by the output tubes, NOT distortion that has been previously introduced further back in the circuit (including intentional distortion).

The irony is that, although almost all tube amplifiers exhibit this characteristic: the suppression of the second and even order harmonics, these are *exactly* the harmonics that are said to be desirable, psycho-acoustically.

14.3 Transient Responses

Transient response (or more precisely, the partial suppression of transients) are influenced by three mechanisms within the amplifier, all of which were/are unintentional:

- lack of bandwidth
- inadequate power supply design
- compression/saturation

The first "problem", lack of bandwidth really isn't an amplifier problem - most amplifiers have plenty of audible bandwidth. The problem is normally manifested in the loudspeaker selection. Since the guitar has a limited frequency range (around 80 to 1300 Hz or so) the selection of limited range loudspeakers wouldn't seem to be a problem. (And, in fact, if we were to connect our guitar amplifier to a wideband stereo cabinet, for example, we probably wouldn't *like* the full-fidelity response).

The fact that there is little effort to improve loudspeaker response (beyond approximately the third harmonic of the maximum guitar frequency) is related to cost and also with noise. From previous discussions, we learned that the amount of noise emanating from an amplifier is proportional to bandwidth (bandwidth = frequency response). So there is justification in selecting a limited range speaker: to minimize the amount of audible noise that would otherwise be apparent to the listener.

To give the idea of the relationship between transient response and bandwidth, a generally accepted engineering rule is that, in order to pass a transient with a time duration of "t", a linear system should have a bandwidth of at least:

$$\text{BW} = 5 / t$$

this implies that the reproduction of a 1 millisecond transient (.001 second), the bandwidth should be

$$5 / .001 = 5,000 \text{ Hz or } 5 \text{ kHz}$$

That's right around the upper frequency limit of many eight and ten inch loudspeakers- and the best 12 inch loudspeakers - to reproduce. Anything "faster" than a millisecond wouldn't reproduce with much fidelity and generally there is no need for reproducing transients of this nature.

The second problem is the lack of a power supply that can provide enough instantaneous energy to sustain a high-amplitude transient. As noted in previous

comments, engineering is all about compromise, one performance parameter being compromised to enhance the characteristics of another.

The main components in power supplies used in vacuum tube amplifiers are the power transformer, the rectifier (older designs used vacuum tube rectifiers) and the "filter" capacitors. When the original designs were created, the process went something like this:

I'll presume that I'm the engineer tasked to design the supply and I'm making mental notes about the requirements, OK?

Need about 100 millamps under normal drive conditions. Peak signal power would require at least double that current.

If I design the supply for 200 millamps, the cost of the transformer will double and I can't meet my price goal for this component.

It's unlikely that the amplifier will be used at full load very often, just at peak points of performance, I'll go with the 100 millamp transformer but use a large capacitor to store the energy required for those peak points.

So the compromise here was to use a smaller transformer than the one required and the justification was that the larger current wouldn't be used very often (remember that the "design" took place somewhere between 1925 and 1940 - everything that happened later was just copying the old stuff). And how could the engineer have predicted how "his" amplifier was going to be used, starting in the 1960's? Now the engineer had to make a decision about the filter capacitor that he had to select. Here are "his" notes about the capacitor.

To reduce AC hum to acceptable level, considering primary secondary resistance, I'll need something around 300 microfarads of capacitance.

That amount of capacitance won't store very much energy, for times when the output tubes require double the current. But if I select a larger capacitor based on double the current there's other problems.

The rectifier tube and transformer winding won't be able to charge the bigger capacitor without being over-stressed.

If my rectifier tube has to pass twice the current that it's designed for, the rectifier voltage drop will increase by ten or twenty volts and the output voltage is going to drop by that amount.

Big capacitor will cost more than I've budgeted for it. If it is to supply double the current for more than a second or two, it's going to have to be about 1,000 microfarads. No way will the rectifier tube charge up something that big, I'd have to replace the rectifier tube with a bigger, more expensive one

So, that's the way that these things go, the engineer may have presented the entire problem to his management, tossing the hot potato upstairs. Management, first and foremost being concerned about costs/profits would have made the obvious decision: don't design the supply for double the current, design it for the average amount of current.

(The configuration of most power supplies used now does not include "choke" input circuits. This was an older, commonly used design when efficient, low noise supplies were necessary (e.g. stereophonic amplifiers). Cost pressures in the musical instrument field quickly eliminated the relatively expensive power inductors used for this purpose with a few exceptions.)

And now, back to transients. When a guitarist makes a particularly vigorous attack on a string or chord, the filter capacitor remains fully charged at the power supply voltage for a brief period of time, so there is an instantaneous transient response that is faithful to the guitar's ability to produce it. But then, the capacitor starts to discharge, the transformer resistance and the rectifier tube "pervience" don't have the ability to quickly re-charge the filter capacitor so we have "sag".

To a lesser degree, this also happens in output transformers. These devices must conduct the quiescent plate current to the output tubes as well as passing the signal currents to the output load. Magnetic saturation sometimes occurs, which is similar, in effect, to "sag" or "compression".

Sag and compression are desirable to many; blues musicians seem to enjoy these effects, particularly. But they are just another term for imperfect transient response.

The last remaining cause of transient infidelity is compression within the vacuum tubes themselves. We've covered this in previous discussions but reviewing: vacuum tubes are not "linear" and at some point they will stop following the linear expression:

$$\text{Output voltage} = \text{input voltage} \times \text{gain}$$

When vacuum tube amplifiers deviate from that expression they first compress. Tubes exhibit more compression as input signal voltage rises until they can no longer produce a higher output voltage. This is known as "limiting" or hard saturation. It's not possible to reproduce ANY transient when saturation is reached.

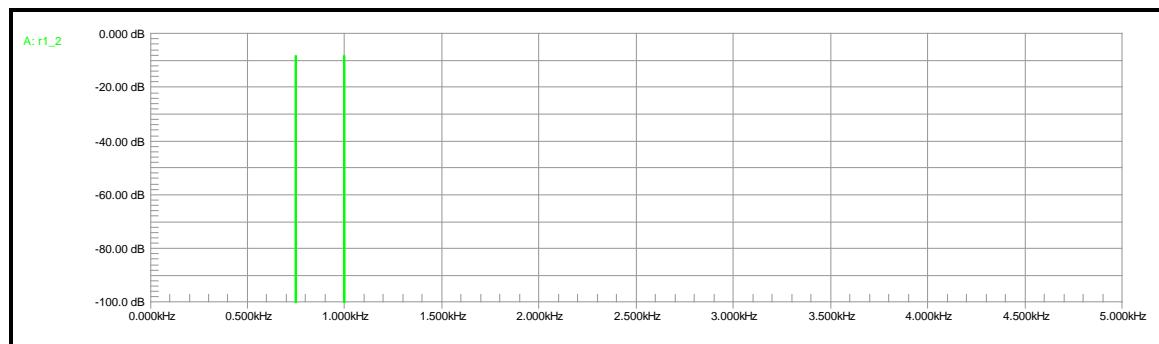
All amplitude information originating from the input signal is lost when processed by a saturated amplifier, only frequency information can be accurately reproduced by the amplifier. (Additionally, a certain amount of time is required to recover from a hard limited condition which suggests that transients are further suppressed by the vacuum tube.)

14.4 Multi-Tone Distortion

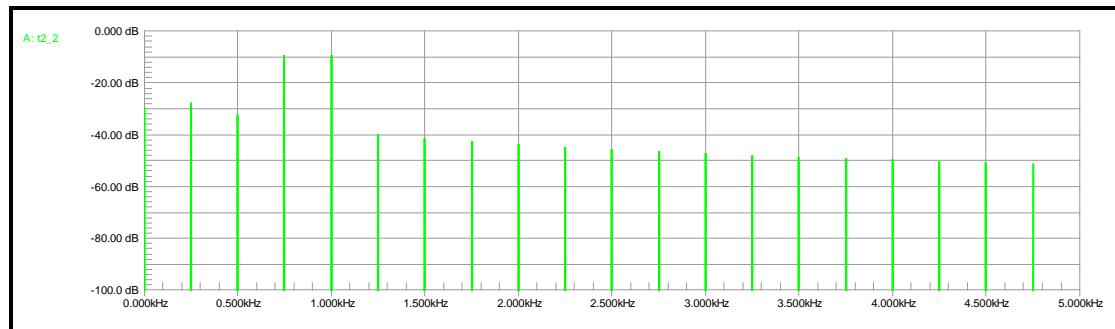
This is a fairly complex subject and to deal with it properly would require visiting first principals dealing with time domain waveform analysis, Fourier transforms, nonlinearities in active amplification devices - many topics that are not appropriate for the audience to whom this discussion is focused. The discussion will be limited to describing how the distortion is caused, visual examples of measured distortion and what to do about it.

Let's start out by considering the behavior of two tones, making things simple by assuming that the two tones are perfect sine waves and therefore distortion free. In general, it's possible to pass two-tone signals through any passive network without introducing *amplitude* distortion. That means that tone controls, coupling capacitors, even some transformers, will introduce no detectable amplitude distortion (although phase shift - group delay - is common). But when the two-tone signal passes through an active device, e.g. a transistor or a vacuum tube, distortion is always introduced.

Here's a spectral representation of two pure (unmodulated) sine waves:



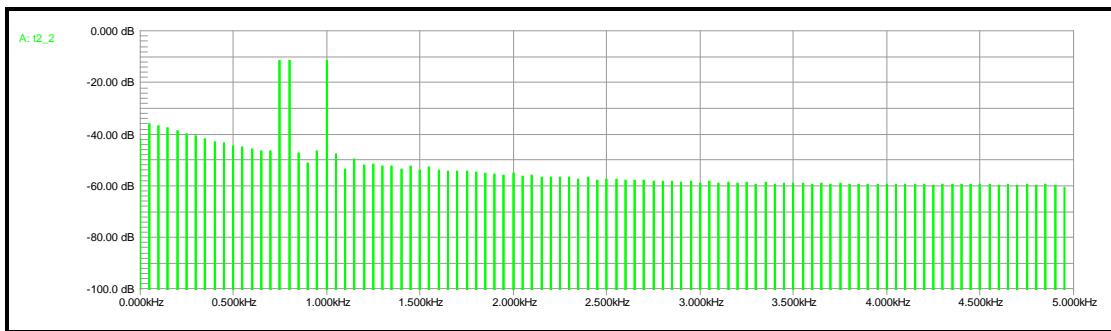
Visible are two signals, one at 750 Hz and a second one at 1000 Hz (1 kHz). Here are the same two signals passed through an amplifier and measured at the amplifier output.



Where did all those other signals come from ? The other signals are harmonic and "intermodulation products" - they occur whenever signals are passed through a non-linear network (ALL amplifiers). What happens is that the 750 Hz

signal and the 1000 Hz signal "mix" with one another and produce all of the sum and difference frequencies. A simpler description, although not mathematically accurate, would be to consider the difference between the two signals ($1000 - 750 = 250$ Hz) and imagine that ALL of the harmonics of 250 Hz are now produced by the amplifier.

Well, not really ALL of them, but as many as can be passed through the bandwidth limitation of the amplifier/speaker combination. This is a very undesirable situation and when the number of tones is increased beyond two, the situation becomes even more complex. For example, adding one more signal to the spectrum gives this:

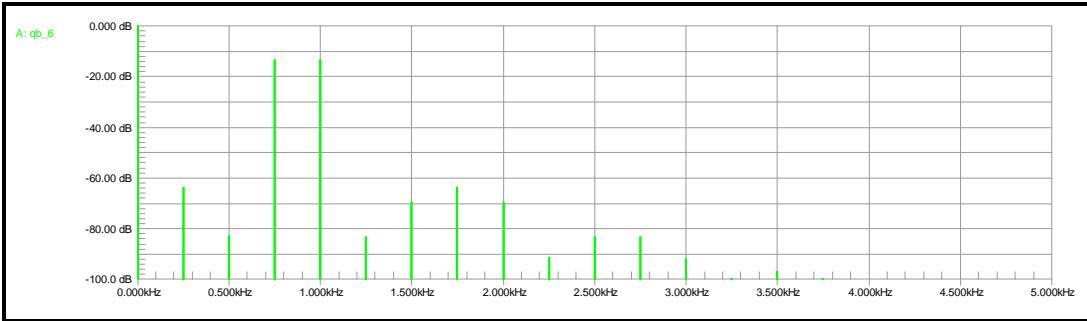


As more signals are added and the intervals between them decreased (such as would be the case in chords utilizing more than two or three notes) the distortion products increase and produce more audibly displeasing sounds. The obvious question is how to eliminate these undesirable products? That's not possible but there are ways to suppress them to a more tolerable level.

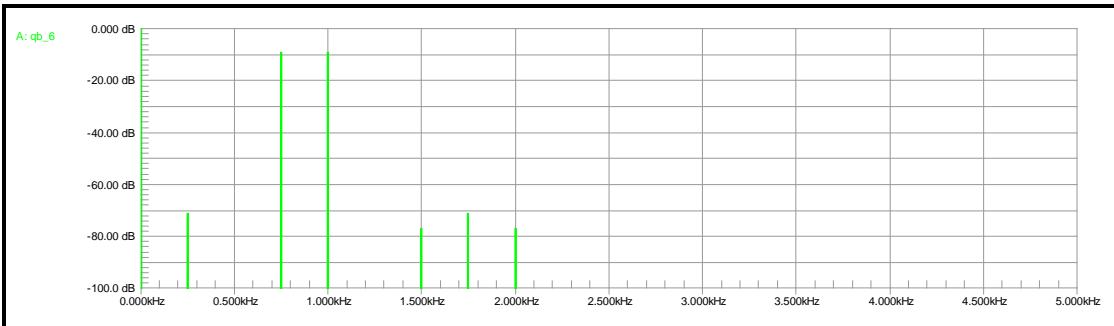
Since the problem is caused by non-linearity in the active devices, whether they are solid state or vacuum tube, this solution is suggested: make the circuits more linear. This is possible but there are undesirable consequences - remember that all design engineering is a series of compromises, as we've often noted.

One possibility would be to increase the amount of quiescent power dissipation in each of the amplification devices in the circuit. It's generally observed that linearity in active devices is improved with this practice, although not always. In certain situations, increasing the static dissipation will also compromise other performance parameters, noise figure for example.

Testing this suggestion, let's perform a computer simulation of a *phase-splitter* tube, applying the same two 750 Hz and 1000 Hz signals used above. Here's a simulation that results from a quiescent power dissipation of around 3 milliwatts:

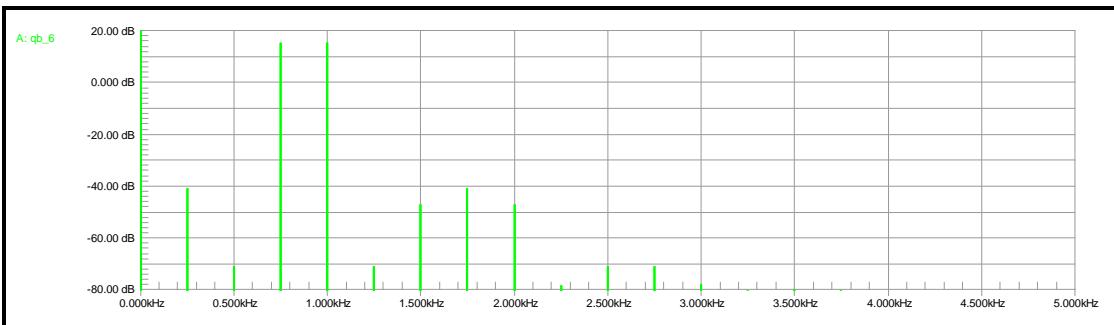


Another simulation, same as above, except that the dissipation is about 82 milliwatts:

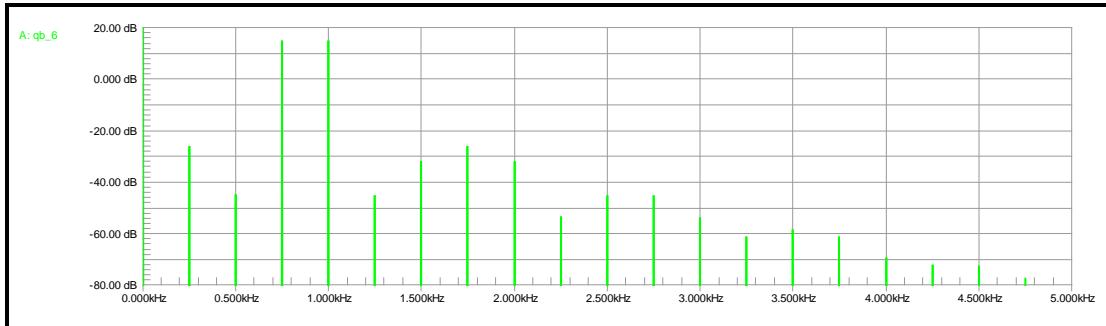


Very little difference is noted in the spectrum of the phase-splitter but then *this phase splitter has no gain*, and in fact can have a fair amount of loss, depending on the configuration. These simulations were performed on a phase-splitter that had 3 dB of loss.

Let's perform the simulation again, except this time on a *preamplifier* tube with a voltage gain of about 24 dB and dissipating 200 milliwatts:



Here's the same preamplifier tube, about the same amount of gain but with dissipation reduced to about 20 milliwatts:



There is significantly more distortion in this simulation. In fact, if we were to calculate the difference between the two cases, we'd find that the first simulation had a distortion of about 1.6 % while the second case distortion was around 7.7 %, that's very significant ! So we were able to achieve an improvement in distortion of about 5 : 1 by increasing the power dissipation by a factor of 10. That's not a bad tradeoff.

We might make a general presumption that the improvement in intermodulation distortion is related to the ratio of quiescent power dissipation to signal power. In preamplifier tubes and phase-splitter tubes, the signal power is usually much lower than the quiescent power dissipation - if our presumption is correct then we can expect fairly linear performance (provided that these stages are not overdriven).

This presumption suggests that controlling distortion can be easier although it's not easy to predict distortion levels beforehand. We can infer that adjustment of the bias conditions of tubes that do not produce appreciable amounts of signal power can resolve *some* distortion problems. This can easily occur at the "breadboard" phase of amplifier construction.

Generally speaking, for guitar amplifiers, distortion products will be a problem only at higher power levels: in the output stage or possibly the driver stage of the amplifier chain. That's because the ratio of quiescent power dissipation to signal power is always quite high (20% - 50%) in these stages. It's possible to reduce distortion by increasing quiescent power dissipation, as in small signal stages. However, output tubes are normally operating at (or near) maximum allowable dissipation so that option is unavailable.

So why do we not hear all of these undesirable tones at our amplifier output? *Because of the effect of negative feedback.* Distortion products that exist in the output stages of the amplifier can be suppressed to near inaudibility IF they do not also exist OUTSIDE the feedback loop.

The importance of this discussion is that distortion must be controlled in stages that are *outside* the feedback loop. The feedback loop includes the stage where the negative feedback connection is introduced throughout the chain to the output speaker connection. Feedback is usually - but not always - introduced in the second stage or post amplifier. If excessive multi-tone distortion is noted after the amplifier is constructed, it can usually be suppressed to acceptable levels by increasing the quiescent power dissipation in the preamplifier tube(s) or by increasing the amount of negative feedback.

14.5 Noise and Hum Resulting From Excess Gain

Generally, degradation in noise and hum is equal to excess gain (or gain compression) in preamplifiers. The following is an example of this premise. Note that practical circuit elements can add as much as 10 dB to the noise figures described.

A typical two-stage vacuum tube preamplifier/post amplifier (12AX7 or similar tube family) might have 20 dB gain per stage. The two stages are usually separated by a tone control circuit, which has a loss of about 10 to 15 dB, depending upon the way the controls are configured.

Noise figure and hum suppression of vacuum tubes varies but if we assume a noise figure of about 5 dB (a very optimistic estimate for modern tubes) a linear preamp would have a gain of about 25 dB and a noise figure of about 6.2 dB.

If the same amplifier was configured for distortion, the first stage gain might be increased to about 30 dB, insuring that the second stage is driven into compression. The tone control circuit will still have about 15 dB of loss and the second stage, no longer linear in operation, will have a large-signal gain of perhaps 15 dB and a similar noise figure.

The overall gain in the second example will then be about 30 dB, so a master volume control needs to be added, adjusted for about 5 dB of loss (if the volume is to be maintained at the same level as in the first preamp example). Because we have added more gain than loss, the resulting noise figure will be about 6.2 dB, the same as in the first example.

Everything looks OK for the second case *at this point*; the gain and noise figure are the same in both preamp cases. Of course the linearity of the second preamp is highly degraded but that was the goal, to produce harmonically related distortion.

14.5.1 Quiescent Noise (No Input Signal)

The noise output (the “hiss” that comes from your amplifier, especially noticeable when volume is high) can be predicted by

$$V_n = F \times A_v \times (4 \times K \times t \times B \times R)^{0.5}$$

where **V_n** is noise voltage, **F** is the noise figure of the circuit expressed as a ratio (not in decibels) and **A_v** is the voltage gain of the circuit expressed as a ratio (not in decibels)

$$(4 \times K \times t \times B \times R)$$

is the thermal noise existing at the input of the circuit and the $^{1/2}$ symbol indicates that we must extract the square root of the expression within the parentheses.

We don't need to pursue the mathematics of noise any further. Earlier, when this topic was introduced, we simplified the input noise voltage to approximately 1.0 microvolt (uV) by making some reasonable assumptions. So the noise voltage at the output can be simplified to

$$V_n = F \times A_v \times 1 \text{ microvolt}$$

In the case of the first preamplifier, the calculated noise output is about 36 microvolts.

In the second case, with no signal present there is no compression, the second stage gain reverts to its small-signal level of 20 dB, the overall preamplifier gain is 30 dB and the noise figure is around 5.1 dB. Although the noise figure is lower than the first example, noise at the preamp output is increased to 57 microvolts, a difference of 4 dB.

The actual numbers are not very important to us but the ratio between them is. The ratio represents the degradation in performance between the second preamplifier example and the first one. Audible noise will be even higher in the second example because the plate resistor in the first stage has to be increased in order to increase the gain from 20 to 30 dB.

The higher plate resistance causes overall noise to increase. (Remember that the above noise equation included resistance - noise increases by the square root of resistance.) Adding other practical circuit elements - like bias resistors - will further increase the noise of both examples well beyond these simple calculations.)

For these examples, the degradation in noise is about 4 dB and, because the second preamplifier example has 5 dB of excess small signal gain, the increase in 60 Hz hum is also 5 dB.

Note that some older designs (e.g. "Music Man") attempted to avoid these drawbacks by applying compression to the output stages of their amplifiers rather

than the more sensitive preamplifier stages. Presumably they also felt that the overall sound quality was enhanced by doing so. I infer this because the "Music Man" configuration is considerably more expensive to produce than the recent trend of employing a vacuum tube preamplifier to drive a solid state power amplifier.

So-called "noise gates" cannot suppress noise originating from an amplifier with an excessive noise figure or excessive gain. Noise gates are helpful only in reducing noise that occurs BEFORE the amplifier input (i.e. "ahead" of the noise gate in the signal chain).

15.0 Design and Parts for Specific Performance Goals

The following are suggestions that may be useful if a particular type of performance is desirable. This material originated from a website in Australia, I think - I copied the material some time ago and have made slight alterations.

15.1 Bright, Clean, Minimally Distorted Sound

Screen grids in output tubes should be supplied from a reasonably constant voltage power supply. In simplest form this can be a large filter capacitor, around 100 uF, after a filter choke (not a dropping resistor) from the plate supply. Preferred is a separate lower-voltage power supply, as previously discussed in the reliability considerations chapter.

If plates and screen grids must be supplied from a common power supply then the filter capacitors should be as large as practical - the bigger the better, provided that voltage ratings are safe.

NOTE: some state that, for reliability considerations, screen grid voltage should never exceed plate voltage. If the screen grid voltage is fixed, as in the above suggestions, then the A.C. signal swing of the plate WILL drive it below screen voltage. I cannot find an authoritative reference that reconciles these apparent opposing opinions.

It is worth noting that the so-called "ultra-linear" output transformer configuration easily satisfies the "requirement" of screen grid always being at a lower potential than the plate. Ultra-linear output transformers are expensive, compared to conventional guitar amplifier transformers but the low-distortion advantage of these transformers is time-proven.

Silicon rectifiers, implemented as a full-wave bridge, in the plate and screen grid power supply are preferred over tube rectifiers.

Plates of the output tubes should be supplied from a power supply capable of providing the maximum current and maintaining operating voltage under heavy transient loads. Use high-quality industrial grade (low "ESR" - equivalent series resistance) high value filter capacitors in the plate supply - at least 100 uF for each pair of output tubes (double that value for bass guitar amplifiers).

The power transformer continuous current rating should be double the maximum output stage cathode/plate current (to improve regulation and reduce the "sag" caused by transformer winding resistance).

Tube filaments are best supplied from a separate transformer (to prevent voltage drop during peak power output and consequent "sag").

Output transformer should have low D.C. resistance (to prevent voltage drop during peak power signals), remember that all of the output plate current flows through the output transformer.

Output transformer can be low inductance (to roll off low frequencies and prevent low frequency overloading of the loudspeaker - especially in an open-backed cabinet). This is a good suggestion for cost reduction.

Inter-stage coupling capacitors should be selected to provide about -3 db rolloff at around 50 Hz.

Negative feedback loop from output tubes is suggested (rather than from the loudspeaker terminals, as is normal practice). Note that this is a typical characteristic of older Ampeg vacuum tube amplifiers.

Output tube grid 1 (control grid) bias voltage should originate from a regulated, well-filtered voltage source.

Output tube grid 1 ground or shunt bias resistors should be as low a value as is practicable - e.g. 100 k Ohms per tube.

Output stage bias circuits (grid bias or cathode resistors) need to be optimized for minimal crossover distortion.

15.2 Jazz, Standards, Clean Rhythm

All of the above recommendations should be implemented although perhaps larger inter-stage coupling capacitors would also be useful (decrease the low-frequency cut off to 30 Hz).

Ultra-linear output stage and parallel-push-pull tubes configuration is essential for bass guitar (to reduce output impedance).

Highest practicable loudspeaker impedance - e.g. series connected multiple loudspeakers (to reduce output transformer turns ratio), as mentioned in previous sections dealing with speakers and transformers.

Silicon rectifier diodes (implemented as a full-wave bridge) are recommended for the plate and screen grid power supply.

15.3 Distortion

Directly heated tube rectifiers contribute to high voltage drops on transient peaks (e.g. 5AS4, 5U4G/GB, 5Y3GT, 5Z3)

Smaller filter capacitors, around 10 to 20 uF (provide poor regulation, allow "sag"). This will be a trade-off with the A.C. ripple voltage "hum" audibility.

High D.C. resistance filter choke (provides voltage drop on transient peaks).

Plates and screen grids should be supplied from a common power source and operated at the same D.C. voltage.

Output transformer should have high DC resistance (to reduce plate voltage on transient peaks, allow "sag").

Pentode output tubes, EL34, EL84, rather than tetrodes (6V6, 6L6) are suggested.

Cathode bias, possibly even Class "A" operation, if power output and efficiency permits.

All of the above characteristics are representative of the low-cost amplifiers available to musicians (and to the recording studios) of the influential guitar periods, 1948 - 1962.

15.4 Bass

Screen grid voltage should be provided from a reasonably constant voltage power supply. In simplest form this can be a large filter capacitor, at least 100 uF, following a filter choke (not a dropping resistor) and derived from the plate supply. A preferred implementation might be a separate lower-voltage power supply, previously discussed in the reliability considerations chapter.

If plates and screen grids must be supplied from a common power supply then the filter capacitors should be as large as practical - the bigger the better provided that the voltage ratings are safe

Silicon rectifiers, implemented as a full-wave bridge, for the plate and screen grid power supply are recommended.

Plates should be supplied from a power supply capable of providing the maximum current and maintaining operating voltage under transient loads. Use high-quality industrial grade (low "ESR" - equivalent series resistance) high value filter capacitors in the plate supply - around 200 uF for each pair of output tubes.

Power transformer continuous current rating should be double the maximum output stage cathode/plate current (to improve regulation, reduce "sag" because of transformer winding resistance).

Tube filaments preferably supplied from a separate transformer (prevents voltage drop during peak power output and consequent "sagging").

Output transformer must have low D.C. resistance (to prevent voltage drop during peak power signals), remember that all of the output plate current flows through the output transformer.

Inter-stage coupling capacitors selected to provide -3 db rolloff at 30 Hz.

Negative feedback loop driven from output tubes, rather than from loudspeaker terminals, as is the normal practice.

Output tube grid 1 (control Grid) should be supplied from a well-regulated supply if negative grid bias is used. Also, grid 1 ground or shunt bias resistors to be as low a value as is practicable, around 100k maximum for each grid.

Output transformer should have grain-oriented silicon steel laminations and high inductance (to ensure high power transfer at low frequencies). Preferably the windings will be interleaved for good frequency response, stability and low leakage capacitance.

Ultra-linear transformers are always recommended for bass amplifiers (they are expensive).

16.0 Versatility of the Beam Power Tube Screen Grid

The tetrode and pentode are the most useful vacuum tubes of the family. These tubes have two variable connections - the control grid and the screen grid - as opposed to the single control grid contained in triode tubes. There are advantages/disadvantages between the triode configuration and the younger, more sophisticated vacuum tubes that followed the triodes. Generally, triodes provide better noise performance (when properly designed and manufactured).

16.1 Adjusting I_{max} By Varying Screen Bias

In chapter 8.0, some mention was made of the importance of the parameter " **I_{max}** " and its relationship to the maximum linear output power. Since many other characteristics of the amplifier are also related to output power level, they are also related to I_{max} . *In fact, it's easy to make the case that I_{max} is the most important tube parameter in the power amplifier design process.*

Although the selection of vacuum tubes for high power application is limited, there are three or four tubes that generally fulfill most amplifier requirements. These tubes have different power dissipation capability, power *output* capability and, obviously, different I_{max} specifications. One disadvantage pointed out earlier is that they are expensive and not necessarily reliable performers. (The adjective "reliable" refers to tube lifetime.)

Also mentioned previously is the large quantity of NOS television horizontal output tubes that are still available. These tubes have good power dissipation ratings and can be operated at very high plate voltages. They don't get used in guitar amplifiers very often however. The reason is that most have I_{max} ratings that are simply too HIGH for amplifier application, or at least that might be the theory.

Recall that it is general practice for guitar amplifier manufacturers to operate the screen grid at the same potential as the plate voltage, or to drop that voltage somewhat by using a voltage dropping resistor. From the aspect of reliability, neither of the practices is ideal. Many sources recommend biasing the screen grid with a voltage source, implying much better regulation than a screen voltage dropping bias resistor can provide. This can be disputed, however, by the fact that so many old vacuum tube amplifiers are still functional despite the lack of a true voltage source to bias their screen grids.

Although this seems to be a diversion from the topic of I_{max} , it's really not. By selecting a regulated screen grid bias voltage, *one can set I_{max} to almost any value one desires*, within certain practical limits. So a power tube that has I_{max} around 500 mA - much too great for practical guitar amplifiers - can be adjusted to a lower, practical value of I_{max} by setting the screen grid voltage to an appropriate value.

Referring to the data sheet of the 6JN6 tube that we selected as the power tube for our example 30 watt amplifier, let's look at a set of curves that we didn't consider when we selected this tube originally. These curves show the normal plate current and plate voltage characteristics but with two important differences:

Control grid voltage, E_{c1} is set to 0 volts

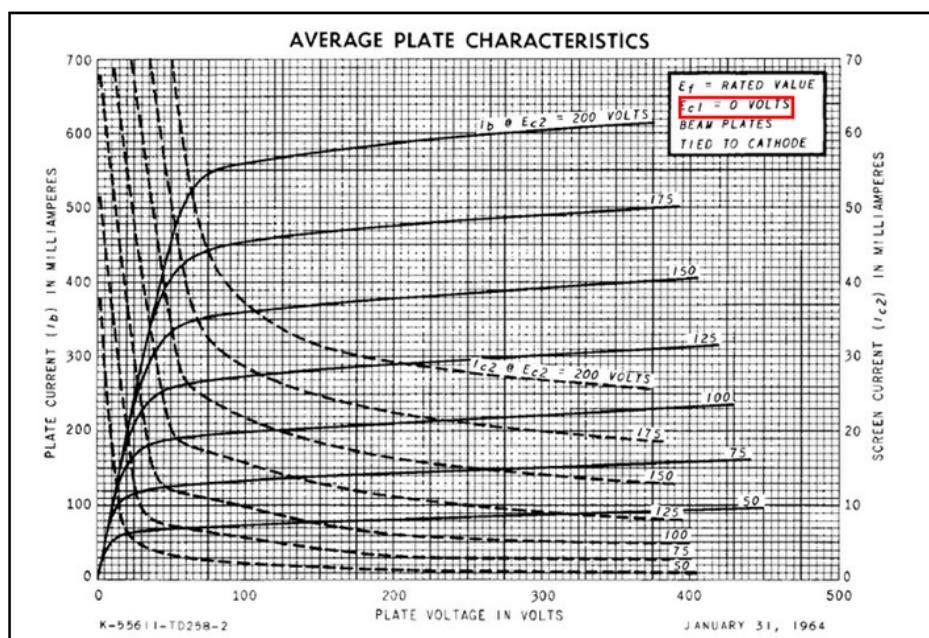
Instead of the variable parameter being E_{c1} , the plate curves are plotted using the variable E_{c2} , the screen grid bias voltage

We originally defined I_{max} , in the chapter dealing with power amplifier design, as the plate current when control grid voltage, E_{c1} , is 0 volts and plate voltage is 60% of the normal operating voltage, E_b . Looking at the curves below, it is obvious that we can define I_{max} at any screen grid voltage shown because E_{c1} is set to 0 volts and that satisfies our definition of I_{max} .

Think about this for a minute --- selecting a tube for a specific value of I_{max} isn't necessary. Provided that the desired value of I_{max} falls within the safe operating limits (plate voltage and power dissipation) of the tube, we can ADJUST the value of I_{max} by adjusting the screen grid voltage.

Therefore a single beam power tube can be used for any output power level up to its maximum rated power dissipation.

The practical variation of I_{max} , as shown in the curves below, can be *any value from 60 mA up to 600 mA*. Allowable plate voltages range from around 175 volts up to 400 volts provided that power dissipation is not a problem.

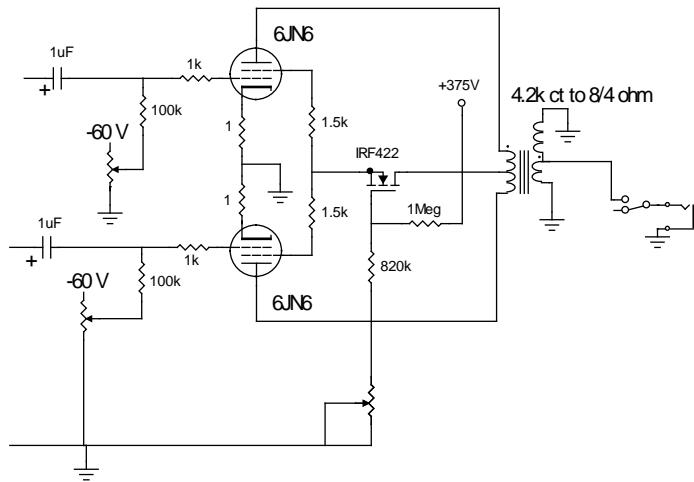


16.2 Provision for Variable Screen Grid Bias

Getting back to the I_{max} characteristic and the implication of adjusting screen bias voltage, several years ago I used this in an amplifier design to vary the operating conditions and intentionally operate the tubes so that they were non-linear. This permitted achieving output tube compression and distortion at lower levels of output power than normal. In other words, an effect similar to that

obtained with a master volume control but not having to pay the price of increased noise that the master volume control always suggests.

I should note, before continuing, that I wasn't happy with this arrangement. I also should note that I spent no time optimizing the circuit for more satisfactory performance. The function was "satisfactory" but the compression characteristics didn't sound particularly good to me. I used this circuit previously as an illustration in the subsection regarding screen bias which is contained in the power supply design chapter. Here's the circuit again, or actually the output power stage of the amplifier:



A high voltage MOSFET (heat-sunk to the amplifier chassis), referenced to an adjustable voltage divider, provides screen grid bias for the output tubes. The values selected gave an output power adjustment from about 25 to 45 watts, and the ability to achieve intentional power tube non-linearity as previously mentioned.

This feature provides an *adjustable* constant voltage source for the screen grid. The constant voltage is a feature that some writers claim to be desirable for "clean" amplification. Other writers dispute the desirability of a constant screen voltage source and note that the screen voltage must always be less than the plate voltage, including the voltage swings during large signal operation. Based on the sample of *this* single amplifier, I'm inclined to agree with the latter opinion. (Whether this is a considered, scientific opinion or a conditioned response is questionable.)

16.3 Accommodating Existing Transformers

In Chapter 8.0, we designed our example output power stage in the following sequence:

Select a tube for I_{max}

Establish power supply voltage

Determine plate-plate resistance and the ratio to speaker impedance

Select an output transformer

That procedure is fine, for the most part, but does have some problems, for example:

Suppose that a specific power supply transformer does not allow the design plate voltage to be obtained

Lower plate voltage = less output power

Higher plate voltage = excessive power dissipation

Can't obtain the required turns ratio for the output transformer to match the plate to plate resistance to the speaker impedance

Less output power

The implication of a pair of tubes with *adjustable* I_{max} is that either of those problems can be mitigated or solved. (Another implication *might* be that unmatched output tubes can be matched effectively if screen grid bias voltages are independently adjustable.) We can obtain a power supply transformer or output transformer, as an example, based on price/availability, rather than to suit a specific tube type. Tube parameters can be optimized to suit the transformers, rather than the other way around.

Let's test the concept with a couple of examples. Suppose one found a bargain price on a small power supply transformer (as I did recently) with an A.C. output voltage of 240 volts and a rating of 75 volt-amperes (75 VA) and wanted to use it in an amplifier power supply.

Using the power supply spread sheet described in chapters 6.0 and 22.0, we can insert the 240 VAC output voltage and the 75 VA rating and determine the items that interest us. (Incidentally, to estimate the allowable current for the transformer, divide the rating by the voltage or $75 / 240 = 313 \text{ mA}$.)

Manipulating the resistor and filter capacitor values in the spread sheet, we find that a power supply of about 305 volts at 150 mA is obtainable from our bargain transformer without too much A.C. ripple at the output. Suppose that we desire an output power (from a pair of tubes) of 22 watts (e.g. Fender "Deluxe" power level), can our power supply support this and what would be the transformer turns ratio and I_{max} for the output tubes?

We could readily rearrange some of the equations in chapter 8.0 and calculate the items that interest us or simply go back to the spread sheet described in chapter 22.0, that covers the design of push-pull power amplifiers, insert the known values for the power supply then manipulate I_{max} and the transformer turns ratio to see if we can obtain the 22 watt desired output power. Here's a suggested sequence:

Set the speaker impedance value to 8 ohms on the spreadsheet.

Set the plate voltage, E_b , to 305 volts in the appropriate spreadsheet cell.

Observing the operating current, $I_{max\ sig}$ (maximum signal current), adjust I_{max} in the spreadsheet cell to obtain an maximum just under the 150 mA max rating. Setting I_{max} at 230 mA will produce an operating current of 146 mA. Make a note of the quiescent current "I_{q approx}" on the spread sheet (88 mA).

Look through the available transformers listed on the same spreadsheet and find one that has a turns ratio close to the calculated optimum of 23.774. Look ONLY at the transformers that are *rated at greater than 22 watts*.

An obvious choice is the transformer with a ratio of "25" and rated at 30 watts. Inserting the transformer turns ratio into the spread sheet, we get a predicted output power level of "22.7" watts.

That's it, we've accommodated a power supply transformer that came our way for a good price.

Now let's approach the problem from the opposite direction, we'll pick an inexpensive output transformer from the table, perhaps the one with a turns ratio of 28.7 that is about half the cost of the above transformer. Then we'll estimate the plate voltage required and select a power supply transformer.

Inserting the turns ratio into the appropriate cell of the spreadsheet, we can iterate as follows:

Allow I_{max} to remain the same (because it determines the quiescent current which has to be less than 150 mA).

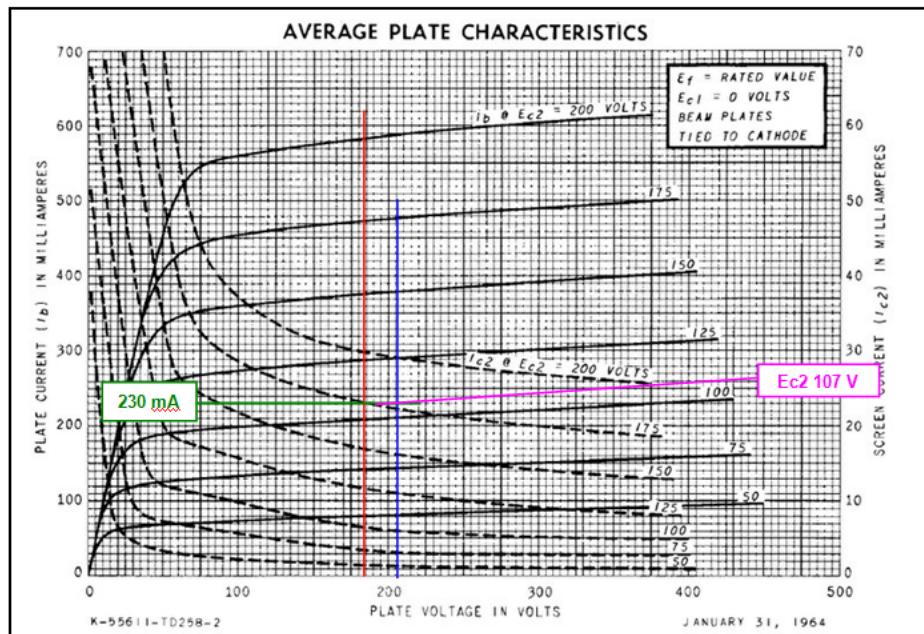
Change the value for plate voltage, E_b , in the appropriate cell and the plate voltage swing, E_o , until the predicted output power level is 22 watts or greater. A power supply voltage of 330 volts with $E_o = 285$ volts will produce a predicted output power level of 22.5 watts.

Note, as mentioned in other discussions, the full power supply voltage is not useable by vacuum tubes. A value between 30 and 60 volts, depending upon tube type, must be subtracted from the power supply voltage to obtain the operating voltage, E_0 .

Now, using the power supply transformers listed on the push-pull tube design spreadsheet and the spreadsheet for power supply design, we can iterate design parameters until a satisfactory compromise has been reached between power supply transformer/operating voltage and output transformer turns ratio. One would expect to make a couple of iterations to select an optimum power supply transformer.

16.4 Obtaining Specific I_{max} Values

In both of the previous examples, for two different reasons, we have changed the value of I_{max} - at least from the value as we originally defined it back in chapter 8.0. Now let's talk about how we achieve the I_{max} value of 230 mA. Referring back to the new set of plate curves, where the control grid voltage, E_{c1} , is 0, let's pick two plate voltages that represent about 60% of the operating voltages we've already selected (305 volts and 330 volts).

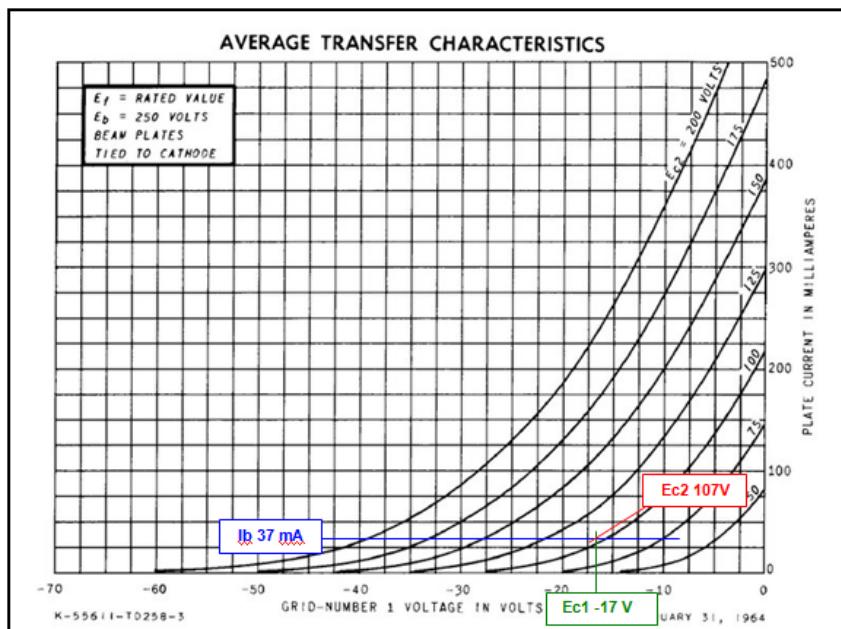


Now make a line at the I_{max} selected value of 230 mA. From the intersection of the I_{max} line with the two plate voltage lines, we can estimate a value of screen bias voltage, $Ic2$, at about 106 - 108 volts. This value should be adequate for either plate voltage application.

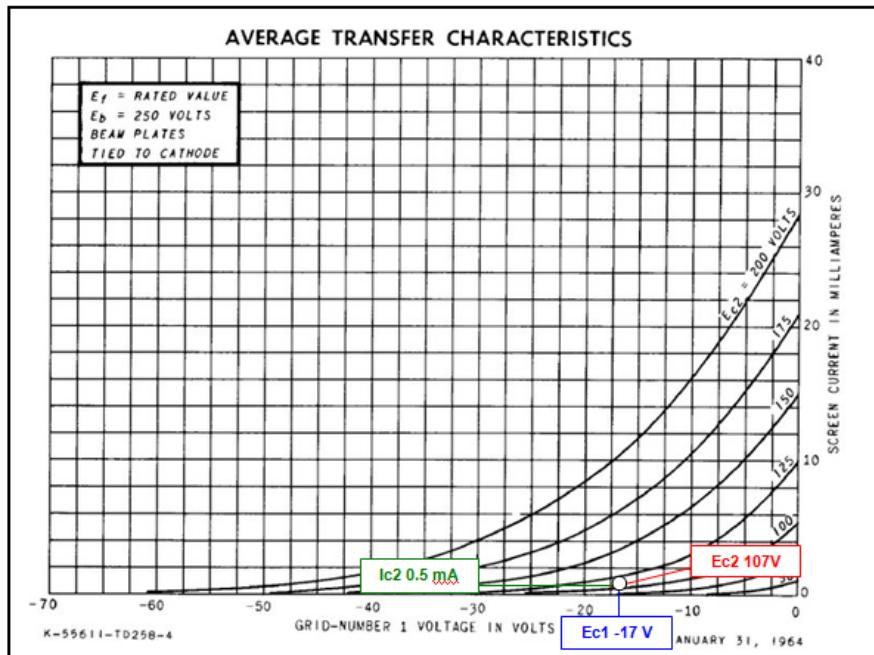
Recalling that our quiescent plate current is 73 mA and using the above information, we can design a voltage divider network to provide screen bias voltage for our output tubes. We need a few more pieces of information first, such as the screen grid current when it is operated at 107 volts.

We need the control grid, Ec1, voltage to adjust plate current to 73 mA when the screen grid is at 106 volts and the plate voltage is at operating potential (either 305 volts or 340 volts). We'll do this exactly as we did previously in chapter 8.0 by using other curves contained in the tube data sheet.

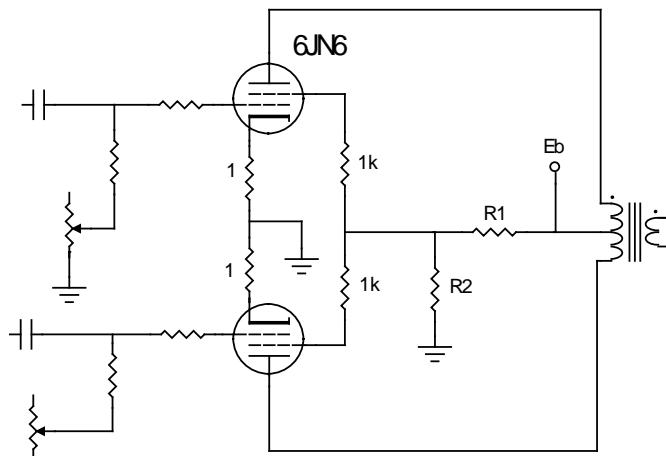
Here's a curve that relates plate current, control grid 1 voltage and screen grid 2 voltage. First, the plate current, Iq, is the current for BOTH tubes. To use this curve we need to divide the plate current by two to get the current for one tube. Establish a point that represents an intersection of 37 mA plate current and 107 volts screen 2 bias voltage:



Noting that the control grid voltage is -17 volts, we move on to the next set of curves. We can draw two lines representing the control grid 1 voltage and the screen grid voltage of 107 volts, then find the screen current from the left vertical axis of the curve:



We now have all of the information required to design a voltage divider network for biasing the screen grid. The method discussed in Chapter 8.25 can be used, using the schematic below:



$$E_b = 305 \text{ volts} \quad E_{c2} = 107 \text{ volts} \quad I_{c2} = 0.5 \text{ mA} \quad \text{or } .0005 \text{ amperes}$$

In the above schematic, it can be observed that there is a 1k resistor in series with each of the screen grid connections. We call this a "de-coupling" resistor, it minimizes the amount of audio signal leakage (coupling) between the two output tubes. The value is not particularly critical and if it isn't significantly larger than the 1k value shown, it can be ignored for our purposes.

The overall calculation becomes much simpler if we can pick one of the two unknown (R_1 , R_2) values. If the voltage divider is to be effective, more current

must flow through the divider than the screen grid current that the divider is supplying. If we make the ratio of divider current to total screen current about 2, then we can say:

$$R2 = Ec2 / (2 \times 2 \times Ic2)$$

and substituting known values then solving, we get

$$R2 = 107 / (4 \times .0005) = 53,500 \text{ ohms}$$

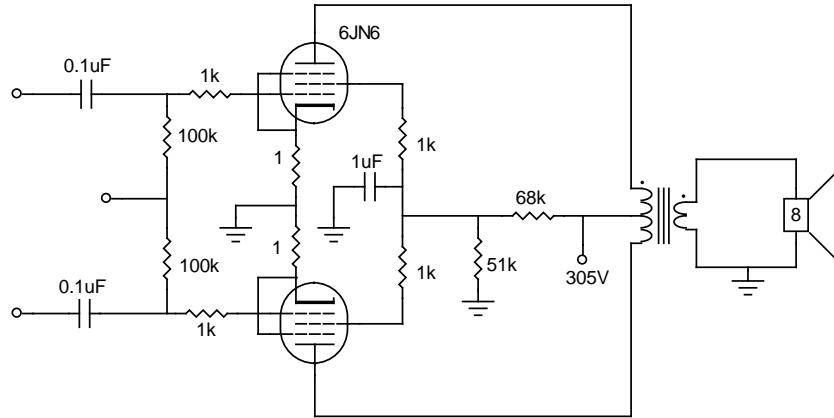
we'll pick the closest standard value of **51k ohms**

Now we can solve for the remaining unknown, R1 by substituting known values into:

$$R1 = (Eb - Ec2) / (6 \times Ic2)$$

$$R1 = (305 - 107) / (6 \times .0005) = 66,000$$

The closest standard value is **68k** and here is the completed schematic. (The second example would be designed similarly except replacing Eb with 340 volts.)



Summarizing, the screen grid bias voltage can be adjusted to vary other power tube characteristics, permitting performance optimization when other important parameters need to be fixed at a certain value. Using the spreadsheets described in chapter 22.0 can make the process quick and simple.

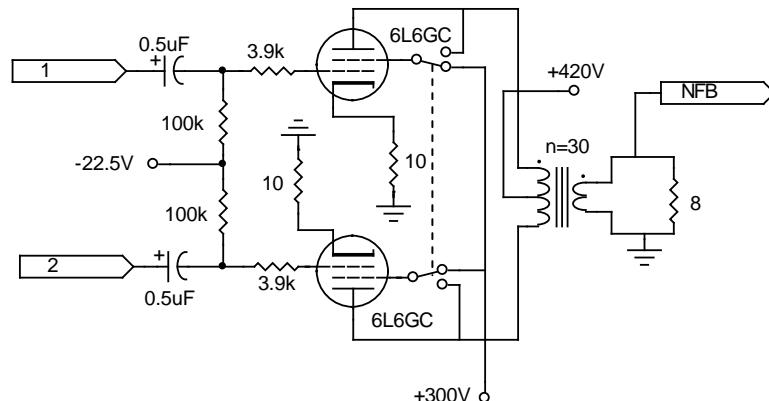
17.0 Using Beam Power Tubes As Triodes

In recent years a convenient circuit modification has been introduced (I *think* by the Carvin Company, San Diego, CA). A switching circuit added to the output power stage allows the normal beam power pentode to be operated in "triode"

configuration. Many believe that triode vacuum tubes provide a more pleasing sound than pentodes. Observation of the plate curves of both tube types suggests that triodes can have a fairly wide compression range, which is generally held to be a desirable characteristic.

One disadvantage - and this depends on ones viewpoint - is that output power is considerably reduced in triode configuration. However, this allows the operation of the amplifier at a lower volume level, still producing a slightly compressed sound that is attractive to guitarists, many jazz guitarists included. Manufacturers represent this as being a practical alternative for a "practice" amplifier. (As with almost all aspects of music, this is based on opinion.)

Another disadvantage of "triode mode" is operating the screen grid at a higher potential than the tube manufacturer might normally recommend. It's an individual decision as to whether a potential reliability consideration (lower tube life) is a worthwhile tradeoff for the flexibility of triode operation. Manufacturers that provide the option have obviously reached the conclusion that the feature is worthwhile, possibly on the assumption that the triode mode of operation will not often be used, compared to normal operation. Here's an example of this configuration (*note that for simplicity, screen grid bias resistors are not shown*):



The double-pole-double-throw (DPDT) switch connects the screen grid either to a separate screen bias voltage or directly to the plates of the two tubes. The output power varies from around 30 watts to about 5 watts in the two switched states of operation above.

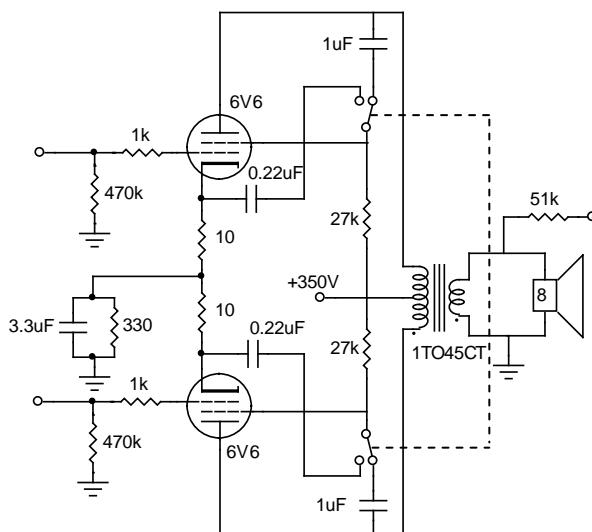
My inclination, should I desire the triode option, would be to design a mode switching circuit that operates both plate and screen grids at the maximum rated SCREEN voltage, rather than the maximum rated plate voltage.

If an amplifier designer thinks that triode operation would be useful, it's a fairly simple feature to add to the amplifier circuit, provided that bias currents are considered when switching from one mode to the other mode. Generally, the required is that the screen grid(s) of the output tube(s) be disconnected from the normal screen bias circuit and reconnected directly to the plates of the tube(s).

A precautionary note: the switch is at a VERY high potential (the highest voltage in the amplifier is universally the plate voltage). The switch must be selected so that the DC voltage rating exceeds the maximum power supply voltage with safety margin. A preferable design would be the use of a solid-state device (high voltage BJT, MOSFET or SCR) that can be enabled by a lower, safe voltage.

If the amplifier designer has already established the screen grid bias voltage at (or very near to) the plate voltage, there is little problem in the implementation of the switching circuit. If the screen grid bias voltage has been established at a substantially lower potential than the plate voltage, however, then the quiescent and operating bias conditions can change substantially, possibly in the direction of an unsafe, unreliable condition.

Here's a simple way of switching to triode mode when screen bias has already been established, in this case by means of screen grid resistors connected to the power supply voltage. The triode mode is enabled by inserting two capacitors which complete a *signal* path between plate and screen grid with a double-pole-double-throw (DPDT) switch:



The screen grid bias will not change when switching modes. This configuration provides a power reduction from around 18 watts down to about 3 watts. (Recall that we normally consider capacitors to have zero impedance when considering signal conditions in the amplifier chain.)

18.0 Hybrid Configurations (Solid State + Vacuum Tubes)

The term "hybrid" as used here refers to the use of vacuum tubes and solid-state active devices in the same circuit. In recent years an attitude has developed that

suggests vacuum tube circuits must contain only vacuum tubes, that there is no reason for inclusion of more modern devices and, in fact, that circuit performance will be degraded as a consequence of their inclusion.

This opinion has no basis so far as I can determine and seems almost "Luddite" in oversimplification and exclusion of fact. As I've frequently stated, throughout this book, design engineering is an iterative and compromising process, we balance one requirement against another and attempt to configure a circuit that is optimum in the fulfillment of all the requirements that we are required to consider.

Excluding certain components that are ideally suited for certain applications isn't sensible. We select vacuum tubes for a specific characteristic that can be described in many different ways but, simply stated, is related to non-linear (compression and limiting) characteristics. These characteristics do extend to other components than the vacuum tubes and the output transformer is frequently mentioned in discussions of this nature.

As an example, selecting a vacuum tube as the major element in a fast, pulse-generating circuit would be ... well, inappropriate. No parameter of vacuum tube performance suggests that this application would be suitable for this component. Other applications are equally implausible; all high-frequency circuits - for example - are the exclusive domain of solid-state devices, for many, many reasons.

Some have recognized this for years, others have attempted to over-complicate simple circuits by designing around the inherent limitations of vacuum tubes. Most "modern" vacuum tube amplifiers (if "modern" can be applied to this old technology) have features that are so far beyond the basic purpose of an amplifier that they would be unrecognizable (and perhaps regarded with amusement) by engineers and performers of an earlier era.

Perhaps we've lost sight of the function of an amplifier. The intent, originally conceived, was to reproduce the sound of an acoustic instrument but with adequate volume to be an equal performer with horns, strings and keyboard instruments. Clearly there are limitations on the transducer (pickup) which earlier amplifiers attempted to correct. This is the classic frequency response pre-emphasis sometimes referred to as "scooped". We've discussed this previously in the chapter that discussed EQ/post amplification.

Without attempting to proselytize, my personal inclination is to make (typical) engineering compromises, using the most suitable parts for the required function. Excluding certain parts because they were not available during a particular time period isn't particularly important to the design process and hinders the ultimate goal. One must take care to "play to the strengths" of each type of component while remaining aware of their respective weaknesses.

(For example, semiconductors are not as thermally stable as vacuum tubes, therefore any performance parameter that is critically sensitive to temperature change needs to be analyzed to determine if there is a negative effect on circuit behavior.)

There are several amplifier configurations that combine semiconductors and vacuum tubes. This is a typical engineering compromise where certain parameters are traded off to enhance other parameters. Since the two technologies are extremely dissimilar and require different operating conditions, there should be good reasons for the additional complexity required to combine two differing types of active devices in a single product.

To date, few of the *linear* hybrid designs have been a commercial success. From a personal viewpoint, it has not been clear to me that combining the two technologies produces a measurable benefit in audio performance. But other reasons for combining the technologies exist such as cost, reliability, maintainability and so forth. In a later chapter, modifications will be discussed for reasons of "improving" existing circuit performance, improvements are not necessarily related to audible differences in sound quality.

Some of the notable successes in recent hybrid designs include the use of digital technology, rather than conventional analog circuits. We frequently find, in a modern amplifier some or all of the following: microprocessors, digital-to-analog and analog-to-digital converters, digital signal processing circuits (and the consequent emulation of various non-linear characteristics as well as special effects such as echo, chorus and harmonic modification). The inclusion (and economic advantages) of digital circuits seems sensible, at least to me.

I can also accept the dichotomy - exclusive as it may seem - of a philosophy that suggests that the following signal chain is all that is required to produce an excellent sound for a jazz guitarist:

Guitar
Cable
Amplifier

In musical expression, as with design engineering, there's no reason to be exclusive of the benefits of recent technology. My most important influences (jazz guitarists) used the simple signal chain described above. I like the sound, I'm conditioned to that sound, but I like the purity, linearity and flexibility of other options too. My personal compromises have evolved to the use of a small vacuum-tube amplifier that I can route to a more powerful, clean, solid-state power amplifier (or the house sound reinforcement system).

18.1 Solid State Front End Vacuum Tube Power Amplifier

An early attempt - and I thought a successful one - was the amplifier structure introduced by "Music Man" (a company that marketed the image of Leo Fender) that no longer exists. This circuit employed the typical push-pull power tube output circuit, including output transformer. Most of the remainder of the amplifier was solid-state and the usage of integrated circuit operational amplifiers was generous - highly unusual at the time.

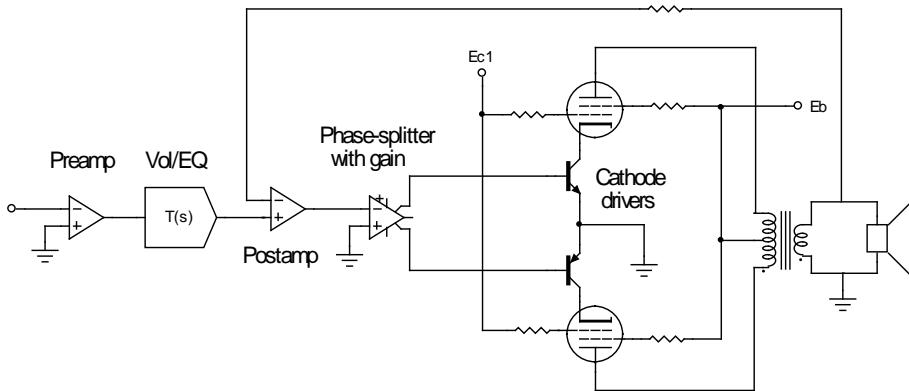
The design goal was to include any desirable compression/distortion characteristics that resided in the output tubes/output transformer while "updating" the remainder of the amplifier, especially with regard to "hum" and noise performance. (The intentional inclusion of excess gain to obtain compression and distortion is always accompanied by higher noise. The designer of the Music Man amplifier series obviously knew that because he placed the compression near the end of the amplifier chain, where the least amount of excess noise is generated.)

I assume that cost considerations also entered into initial design discussions and the use of semiconductors must have seemed attractive. However the amplifier schematics indicate that they were complex and high in parts-count (like most quality solid-state amplifiers).

In an implementation virtually unknown in the history of *guitar* amplifier manufacturing, the amplifier output tubes were operated in *common grid* configuration rather than the universal *common cathode* circuit. In terms of sound quality, it's uncertain whether Music Man design engineers had a specific goal in mind that this configuration might achieve.

From an engineering aspect, the configuration was elegant - the interface between the low voltage transistors and the high voltage tubes was optimal. (The transistors drove the cathodes of the output power tubes, minimizing stress on the transistors, since the cathodes are operated at low voltages.) However, the common cathode configuration would not have been very much more difficult to implement currently - perhaps the parts or the cost were more prohibitive at the time these amplifiers were designed.

In my opinion, Music Man produced a fine product (I don't know about their price goals); the amplifiers sounded very good. Needless to say, they were still quite heavy due to the necessary inclusion of the output transformer. It's not clear why the market did not accept these improvements, my surmise is that the timing was not right. At the time Music Man introduced these designs, music still was characterized by heavy distortion (the "Marshall" sound). Here's a representation of the Music Man amplifier architecture:



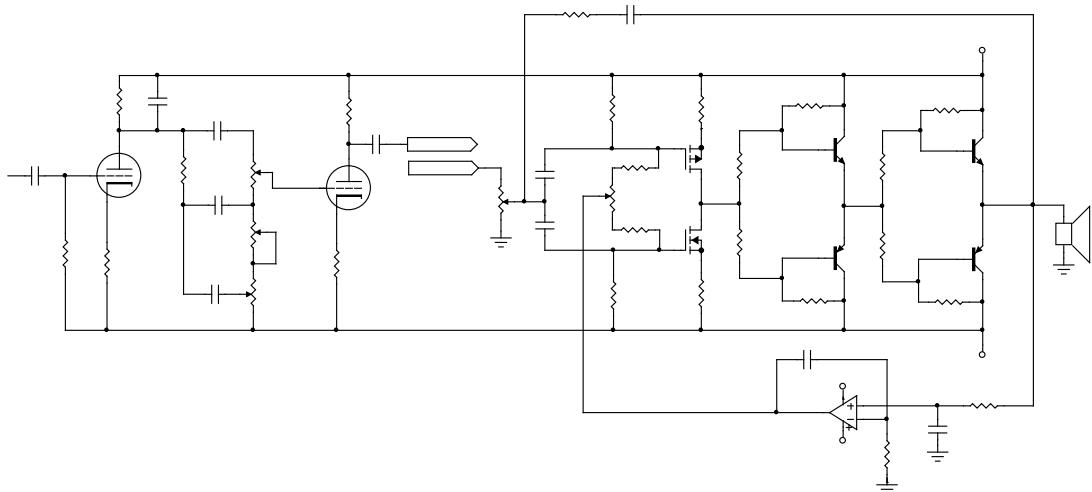
The control grids of the output tubes are effectively "grounded" through the control grid power supply. The cathodes are the "inputs" and the plates are still the output terminals. Note the use of negative feedback, from the speaker terminals back to the post amplifier that follows the volume/EQ control circuits.

18.2 Vacuum Tube Front End, Solid State Power Amplifier

A configuration that seems to gaining current popularity is vacuum tube preamplifier and post amplifier followed by all solid-state circuits. (A number of manufacturers of "pedal effects" also offer vacuum tube preamplifiers which can be used to drive any kind of power amplifier.) The main reason for this type of hybrid is marketing appeal, in my opinion.

The preamplifier and post amplifier stages can be biased from low-voltage, inexpensive power supplies and *might* offer some compression advantage if the post amplifier is slightly over-driven. It has to be noted that this can occur only when the post amplifier plate voltage is low enough to cause "clipping" - which is virtually guaranteed because the tubes share the low voltage transistor power supply. (There isn't enough gain available in the 12AX7 type tube to overcome EQ losses and drive the post amplifier into "normal" compression.) The clipping effect really isn't the same as the gradual compression experienced in conventional tube circuits.

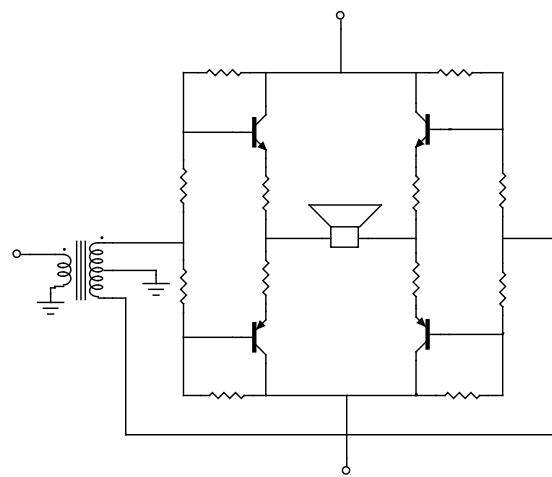
Economies of scale will always favor the use of solid-state components as opposed to vacuum tubes, even though solid state amplifiers are typically more complex. Here's the architecture of an inexpensive circuit that I designed about ten years ago but have never built:



The ubiquitous 12AX7 front end was intended to be operated from the transistor power supplies of about +48 and -48 volts. This amplifier was intended to produce an output power level of 100 watts. I did make some experiments by "breadboarding" the tube part of the design, which is a configuration based on almost all vacuum tube amplifiers, regardless of manufacturer.

I used the tube circuit to drive several solid state amplifiers and found the sound to be pleasant. However, I couldn't actually determine that the "sound" was superior to the solid state amplifier alone, it's all very subjective. I was never enthusiastic enough about this concept to actually build it because I didn't see any performance value to offset the additional cost of the tube front end.

It's possible to achieve much greater power levels by modifying the output stage, here's an example of a "bridged" configuration (where the speaker terminals must not have a ground return). The speaker terminals are driven by two amplifier stages that are 180 degrees out of phase, the effect is that of doubling the available power supply and the output power. The preamplifier and post amplifier stages, as in the above example, would be vacuum tubes.

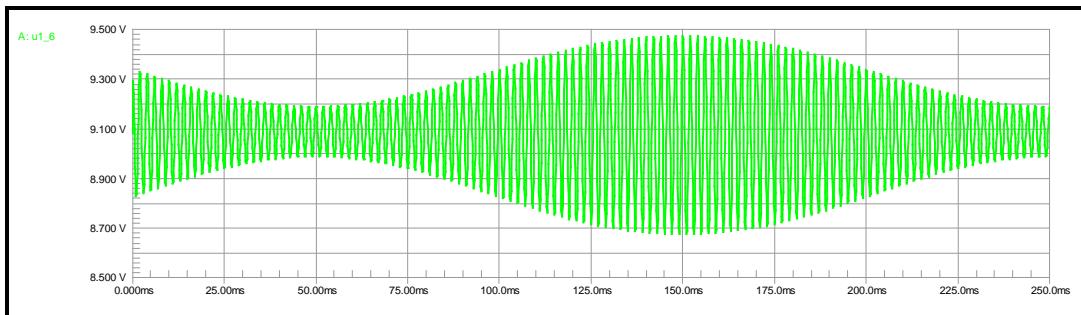


The drawback with this, and most configurations of solid-state amplification, is removing the heat from the semiconductors. Unless efficient heat exchangers are used (and surface temperatures isolated from human contact) the maximum output power of most transistor amplifiers will be limited to less than 100 watts without forced-air cooling. For this particular configuration, another disadvantage is that the speaker connections are both "hot" - this configuration cannot be used with a speaker cabinet in which one of the speaker terminals is returned to "ground".

18.3 Tremolo Circuits

These circuits (sometimes called "vibrato") are simple amplitude modulators; a low frequency signal, from about 1 to 5 Hz, is used to modulate the normal audio signal in a manner similar to a vocalist modulating the ending note of a phrase. One circuit, introduced by Fender, used a light-bulb coupled to a "photo-resistor". The circuit sounded fine but was notoriously unreliable.

Other manufacturers developed different variations on the circuit, with similar sounding results, one amplifier to another. Here's the waveform of a typical "tremolo" equipped amplifier, the main tone, at 500 Hz, is modulated at a 5 Hz rate, to produce the classic "warble" of this effect:



All that's needed to produce this effect is a low-frequency oscillator, whose frequency and amplitude are variable and the proper place in the amplifier chain to insert the signal. We want to use the output voltage from the low-frequency generator to vary the bias at a point in the chain that will have a significant effect on the signal level.

Although this sounds like a simple proposition - and it should be - we need to account for the effect of any feedback loops that may be present in the amplifier chain. The universal negative feedback loop, the one that corrects distortion and frequency response in the output stages, won't be a problem, it's configured mainly to correct distortion in the audible frequency range - the very low frequencies produced by the tremolo oscillator won't result in any form of correction from the loop.

In the next chapter (chapter 19.4), we will encounter another loop that is designed specifically for low frequency corrections. This loop maintains the bias conditions of the output tubes at an optimum current level. This type of loop will definitely interact with a tremolo effect in a negative manner. The bias loop will attempt to "level" current variations in the output tubes caused by the varying signal drive amplitude. If ones amplifier is equipped with a bias circuit of this nature, the addition of a tremolo circuit becomes more complicated. (We'll ignore that situation for the present except to note that it's practical to modulate a closed loop from an external reference source, such as a tremolo oscillator circuit.)

*It will have occurred to most people that an **external** tremolo effect could be a good alternative to constructing the onboard effect. I certainly would not disagree with that, in fact most modern signal processing effects would offer many more effects, not just the simple tremolo. The tremolo topic is introduced mostly as a stimulus for considering how hybrid circuits can be effective for cost/performance goals.*

Although the modulating signal can be applied at several different points in the amplifier, the usual location is at the plate of the post amplifier or at the grid of the output amplifier tubes. In both cases, the amplitude of the modulating signal is required to be fairly high. It's fairly easy to estimate the level required, it will be approximately double the control grid, E_{c1} , voltage, peak to peak. As an example, our earlier example amplifier required the power tubes to be biased at a control grid voltage of -23 volts. This implies that the tremolo bias signal must be 46 volts, peak to peak.

Consideration of the available supply voltage(s) is important before deciding on how and where to implement a tremolo circuit. Here are some thoughts regarding planning for a tremolo modification:

If the modulation is applied to a plate circuit, then the modulating circuit should be capable of varying the plate voltage from the nominal power supply voltage to a point where the tube is effectively "cut off". (Examination of the plate curves will determine the voltage at which this occurs, usually around 10 to 40 volts.) The circuit must also be able to pass the required plate current, at least the RMS current level.

If the modulation is applied to a grid, then the modulating circuit should be capable of varying the grid voltage from the quiescent voltage to the point where the tube no longer conducts significant plate current.

The peak to peak voltage variation for grid modulation is generally equal to the magnitude of the nominal grid voltage operating point. In other words, if the grid voltage is -18 volts, then a peak-to-peak modulating voltage of 18 volts is adequate for satisfactory tremolo operation.

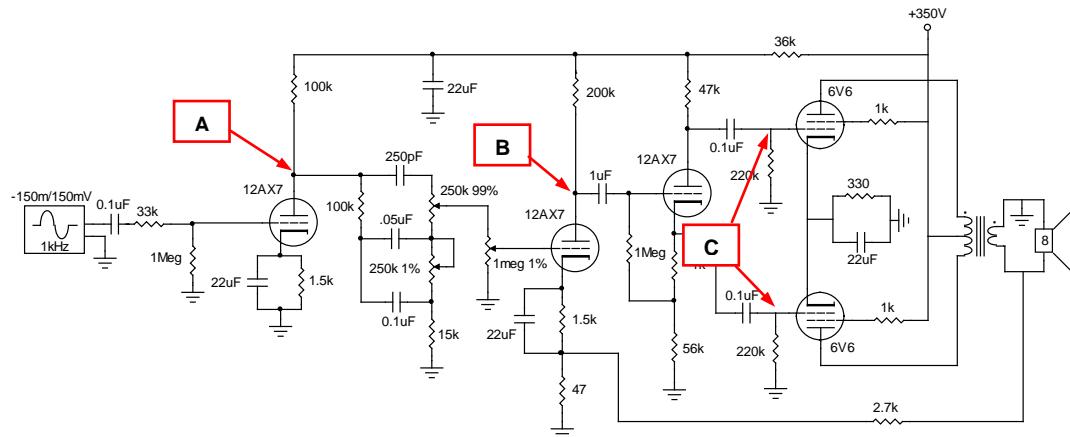
A previous statement suggested that twice the previous value is required. This is because practical circuits for modulation will usually require some form of resistive injection - a resistive voltage divider will be required. It's therefore prudent to plan for the voltage reduction that the divider will cause and design the tremolo modulator circuit for excess capability.

An important consideration with grid modulation is that *the polarity of the modulator power supply is usually opposite from the plate supply*. A negative power supply is frequently unavailable (although it may be possible to generate a negative supply by modifying the power supply circuit).

If there is an unused dual triode section available in your amplifier, it may be possible to implement the tremolo circuit, maintaining an all-tube lineup, if that's of importance. Most dual triode tubes are quite limited in maximum current levels and that *may* preclude their use for this application.

High-voltage, high current, inexpensive MOSFETs are readily available in a tiny package. They are efficient replacements for many vacuum tube applications and require no filament voltage/current. These devices are an obvious choice for implementing a tremolo circuit and we'll explore this further in a moment.

This is a schematic of a Fender "Princeton" amplifier, it's a prototypical configuration and a good example of a simple vacuum tube amplifier; I've indicated three locations where a tremolo modulation signal could be injected:



"A" and "B" are injection points to the plates of the preamplifier and post amplifier, respectively. Typical requirements of modulation injected at these points are that the output impedance of the modulator be five or ten times greater than the effective plate resistance (so that the stage gain is not degraded) and the peak-to-peak modulation current is adequate to cause a voltage change

across the plate resistance that allows normal operation or near complete cut-off of the tube.

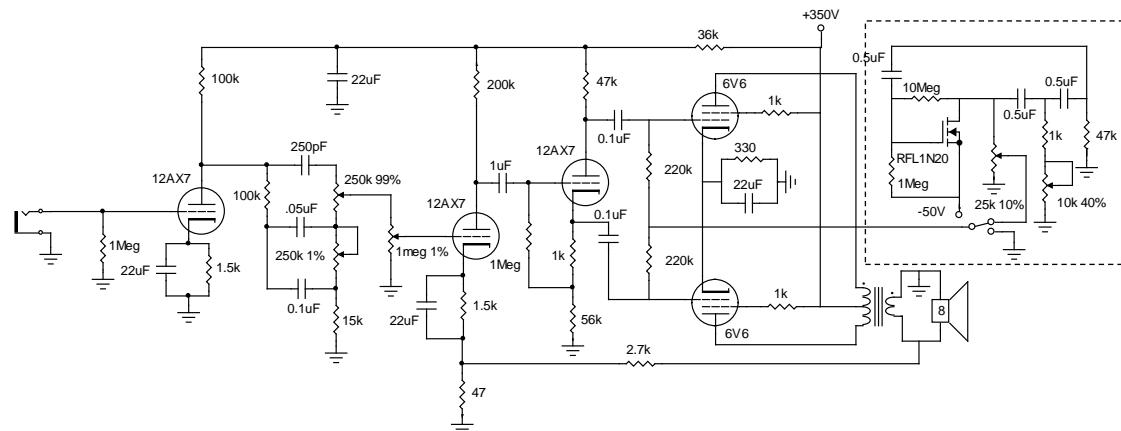
Because of these simultaneous requirements, tremolo modulation is rarely introduced at these points. Grid modulation, due to the very high impedance at that circuit node, is more practical (as in the output stage above, indicated by "C"). One might wonder why grid injection wouldn't be desirable in the preamplifier or post amplifier stages and it's a good question.

We don't apply modulation to the preamplifier because additional circuitry or signals applied will contribute to overall noise; this is the highest gain point in the amplifier, it's not a good idea to introduce even small amounts of noise here.

The post amplifier is a more convenient insertion point but the volume control is a complication for modulation injection at the grid. Similarly, negative feedback (and possibly a "line out") connection complicates injection of the tremolo modulation to the cathode of the post amplifier.

The grid of the output amplifier is a frequent application of the modulation voltage, although there are negative aspects, as noted in some of the above notes. If one doesn't mind designing a low-current, high voltage negative power supply (as described in chapter 12.5), this is the simplest way to include a tremolo circuit.

Here's the same Princeton circuit as above with a tremolo circuit added:



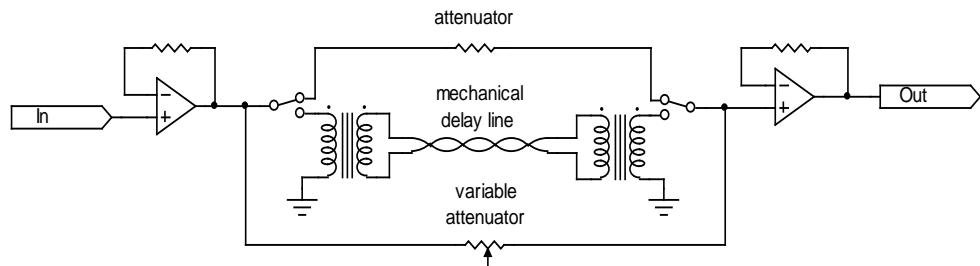
An additional advantage to implementing the tremolo circuit in this manner (adding the -50 volt supply) is that the output amplifier can be converted from cathode bias to grounded cathode configuration, if it's desirable to obtain slightly more output power from the output stage. This is not actually recommended, tube matching would be more critical and bias adjustment provisions would have to be added to the circuit, but it is possible. If this is desirable, be sure to calculate the increased power dissipation as a result of plate-to-cathode voltage

increase and possible increased plate currents (in addition to likely screen grid dissipation increase).

18.4 Reverberation Circuits

Adding spring reverberation to an amplifier is a challenge, not only does the fairly large spring unit need to be accommodated but the unit sometimes must be oriented in a specific manner that limits design and mechanical layout possibilities. Modern DSP (digital signal processing) circuits do a fine job of emulating this function and they are not costly - one should consider that option before undertaking a modification of this level of complexity. Adding the tremolo circuit, above, would represent a level of complexity of about 15%, compared to adding a mechanical reverb effect.

This is a block diagram of a typical reverberation circuit:



The two triangular symbols represent amplifier stages (the resistors above them infer feedback so gain can be "set" to a fixed value). A mechanical delay line, consisting of a set of steel springs (usually three) provides the echo effect. The transformer symbols in the schematic are actually transducers. A transducer is a device that translates mechanical motion to electrical modulation, or vice-versa. (Loudspeakers, microphones, piezoelectric pickups, headphones, speedometers, magnetos, generators, solenoids and so forth are examples of transducers.)

Electrical variations at the input transducer are translated to mechanical motion exactly like a loudspeaker. The mechanical motion produced by the input transducer is coupled to the "springs" of the delay line. After a delay determined by the overall helical length of the springs, the mechanical movements are coupled to the output transducer. The output transducer converts the mechanical motion to electrical modulation. The resulting signal resembles a low-fidelity version of the input signal and all of the reflections that have traveled - back and forth - along the delay line.

The delay line is inefficient, that's why it's necessary to drive the input transducer at an optimal level and then "recover" the loss of the delay line after the output transducer. "Recover" obviously is simply a term describing amplifying the signal to make up for the loss experienced during the path from transducer-spring-

transducer. The resultant signal can be equivalent in amplitude to the input signal but, unavoidably, is much noisier.

A bypass path is provided by switching the signal path around the delay line as shown above. An attenuator is required in the bypass path, roughly equivalent to the loss of the delay line so that the following stage is not driven too hard. The attenuator adds some noise, although not as much as the delay line. The variable attenuator depicted in the schematic provides a means of "leaking" the direct signal around the delayed signal, adjusting the mix to a ratio adjusted by the user.

The "Accutronics" web site includes a great deal of technical information regarding the use of their products. There are recommendations for reverb circuit implementation using both vacuum tube and solid state circuits. I recommend that anyone considering adding reverb spend a little time on research and I can't think of a better place to spend it than here:

<http://www.accutronicsreverb.com/rvbapps.htm>

I enjoy a small amount of reverb when practicing at home but have never used the effect when performing. As a consequence, I've had little interest in adding reverb to the old amplifiers that don't include it. I have an old "Echoplex" that can serve the purpose, if necessary, although frankly it's been rarely used it for guitar. (The device has seen many hours of service in the effects loops of sound reinforcement systems.)

Many guitarists insist on reverberation circuits that include vacuum tube implementation. Frankly, I don't understand that preference - reverb is an extremely low-fidelity, low-level effect (compression has no contribution) as it is commonly implemented. There's no reason to design a circuit that is overly complex and wasteful of power for reasons that are - at best - insubstantial.

18.5 Building a Vacuum Tube Preamplifier

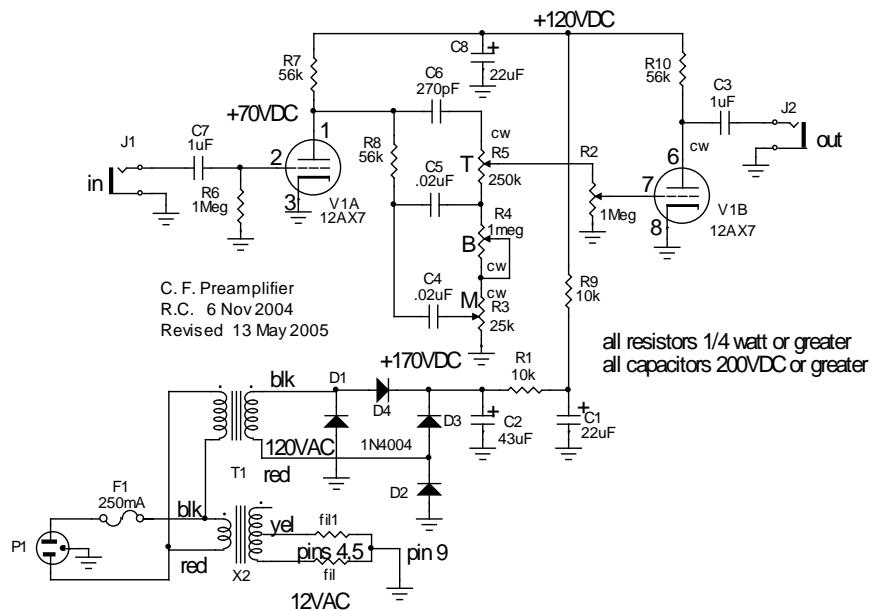
The following circuit is a "hybrid" only because of the solid-state rectifier diodes in the power supply. The low operating voltage and the small size precluded the use of even the tiniest of tube rectifiers. A slight change, described at the end of this section adds a single transistor to the circuit. This improves the overall performance and makes the circuit a true hybrid.

Vacuum tube preamplifiers are useful for various purposes, such as the amplification of acoustic instruments that employ piezoelectric (crystal) pickups or microphones. Folk instruments, dulcimers, ukuleles, autoharps and the like can benefit from the use of a low gain preamplifier with full tone control.

The major disadvantage of most preamplifiers is the need for the high plate voltage, E_b . However, the 12AX7 family - the family most frequently used for

high-gain, low level applications, will operate at relatively low voltages provided that signal voltage levels are not too high. A simple line transformer, 120 VAC to 120 VAC, can be used as the basis of the tube power supply.

Depicted below is the schematic of a circuit that I constructed a few years ago, housed in a small Hammond die-cast enclosure, 3 x 4 x 2 inches. The circuit is the universal preamplifier configuration used in vacuum tube amplifiers for the past fifty years. The user requested a "Fender-type" vacuum tube preamplifier to slightly modify the sound obtained by his Fender solid-state amplifier.



The circuit was extremely compacted, because of the limited volume available from the small enclosure; there was no room even for a fuse holder. The fuse was hard-soldered into the circuit, accessible by removing the four screws that secured the cover. The dual triode, inserted into a conventional 9-pin socket, was installed on an "L" bracket inside the enclosure. (Above the dual triode tube, the enclosure incorporated a series of slots - milled through the cover - allowing air circulation and heat exchange.) The power supply transformer had two 120 VAC windings (one of which I used for the plate supply) and a 12 VAC winding, which was used for the filament supply.

There is little notable about the circuit except for the lack of cathode resistors. These were omitted primarily for size considerations (because the resistors would have required fairly large bypass capacitors) and also for maximum gain. Since the tubes are operating at very low plate voltages, the plate current for 0 volts grid bias is also quite low - around 1 mA. This allows the use of fairly large plate resistors (56k) to obtain reasonable voltage gain while still maintaining adequate plate-cathode voltage. There is no concern about linearity in this

circuit, despite the low voltage/current operating conditions, because the signal levels are so low.

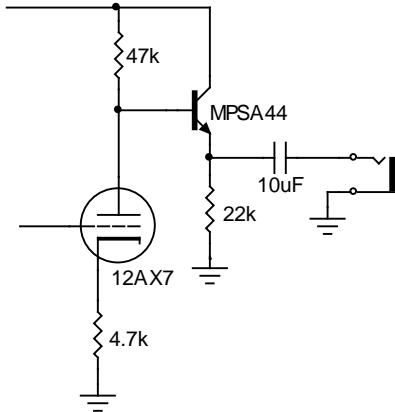
The packaging density was tight - even the power supply transformer had to fit inside the small aluminum enclosure. This device had a full EQ circuit: bass, mid-range and treble in addition to the normal level control. Adjusted for a flat tonal response, the preamplifier had a nominal gain of 20 dB, adequate for most, if not all, acoustic pickups. Despite the circuit density, no inordinate problems with hum and stray pickup were experienced (the power supply filtering was aggressive, as can be noted in the above schematic).

Here's a photo of the slightly homely, five year old preamplifier, parked on top of a small solid-state amplifier in the owner's studio. This guitarist used the tube preamplifier to "warm up" the tone of the small amplifier. The input/output jacks and power connections are located on the rear of the small enclosure.



A preamplifier like this one is a useful addition to one's collection of cables, instruments, microphones and tools and can be built compactly enough to fit in a guitar case (or in the bottom of a combo amplifier). Note that the output is high impedance, however, and therefore long cables from the preamplifier are not recommended.

The addition of a single high-voltage, inexpensive transistor (such as type MPS A-44 or equivalent), configured as a voltage follower, provides a more versatile output configuration. Here's the modification to the second stage; this modified circuit will drive any practical impedance one is likely to encounter. (Adding the transistor will not affect the tonal qualities of the preamplifier, the transistor is configured to have no gain.)



19.0 Modifying Amplifiers

One of the most practical applications that results from learning about vacuum tube amplifiers is the ability to make useful modifications to your equipment. Instead of blindly making component substitutions, not knowing what to expect, a little education can provide the basis for intelligent experimentation and product improvement.

It's possible to tailor individual performance parameters to suit one's individual taste (keeping in mind that the manufacturer has already optimized the amplifier for best *general* performance). Most of the time, it's possible to modify an amplifier in such a manner as to allow complete restoration of the original configuration. This is the most desirable situation and one toward which the modification should always be planned.

If one owns an older vacuum tube amplifier and wants to extend its life, there are modifications that can achieve this purpose. As with all engineering, compromises are a part of the process. If one is willing to relax the requirements of a particular performance parameter, then enhanced performance may be achievable in another area. As always, when considering trade-offs, it's helpful to make a list of positive and negative aspects of the work effort.

Sometimes, a desirable modification might entail the use of unconventional circuits/components (or at least unconventional at the time the original design effort transpired). Please review the introduction to the previous chapter regarding mixing technologies, if personal ambivalence exists. There are sensible engineering reasons for mixing technologies; there are also considerations involving personal taste, history, continuity and other intangibles that influence or suggest an adherence to the technology of the day.

Fortunately, unlike guitars, modifications to most older amplifiers don't require careful consideration of the resale value. Seemingly the most valued characteristic of the older models is the exterior appearance. It's also quite

practical to locate and re-install components that were manufactured fifty and sixty years ago. Returning an amplifier chassis to original configuration is simple and inexpensive, provided that one does not require that the original circuits are as untouched as the day they left their manufacturer's workplace.

19.1 Making A Practical Amplifier From the Fender "Champ"

Here's an interesting project focused on the Fender "Champ", I'm a fan of these amplifiers, especially the silver-face variety with better EQ and more output power than the original. The advantage of this amplifier is small size/portability. Disadvantages are inadequate power (and headroom) coupled with the small, limited range loudspeaker of non-standard impedance (3.2 ohms).

Since the Champ is so easy to work with (there are large amounts of unused real estate inside the chassis), I've tinkered with mine often. It's been thoroughly manipulated and revised with little discernible performance improvement. Some improvement, however, resulted from replacing the speaker and the output transformer with a larger 10 inch speaker of normal impedance (8 ohms), as in a Fender "Princeton".

NOTE: the output transformer does not need to be replaced if a 4 ohm speaker is used. I chose to replace the transformer so that I could evaluate various 8 ohm speakers (there is a greater variety available in this impedance than in other impedances).

Replacing the speaker wasn't a trivial task, although replacing the transformer was simple. Here's a photo of the Champ with the back removed. The outline of the old 8 inch speaker cut-out is visible, replaced with a piece of plywood cut to fit the hole, epoxied into place and painted. (Note that the speaker was moved off-center in the enclosure to clear a large electrolytic capacitor on the chassis.)



After replacing speaker and transformer, a listening test left me desiring a more balanced response - the Champ always lacked bass. I measured the frequency response of the new configuration then experimented with partly covering the back of the enclosure until I arrived at this variation, which has a satisfying amount of bass and midrange. One of these days, I'll get around to painting that piece of cedar.



The amplifier as modified was fine for practice and for duo performance (no drummer) in small venues. Once a drummer is added, the Champ is impractical for *any* venue. Appreciating the convenience of this small unit, I wondered about increasing the output power. I listed the implications of bringing the Champ up to "Princeton" performance levels (around 12 watts) from the measured 5 watts produced by my Champ at 5% distortion.

The Champ is a single-ended amplifier - one could double the power with a push-pull output circuit, requiring two output tubes and a phase-splitter tube

Extra tubes = increased current consumption = bigger power supply
transformer = increased \$

Need to cut holes in chassis for additional output tube and phase-splitter tube

Extra output power means that extra gain is required (increasing the power level from 5 to 12 watts means about one and one-half times more voltage gain)

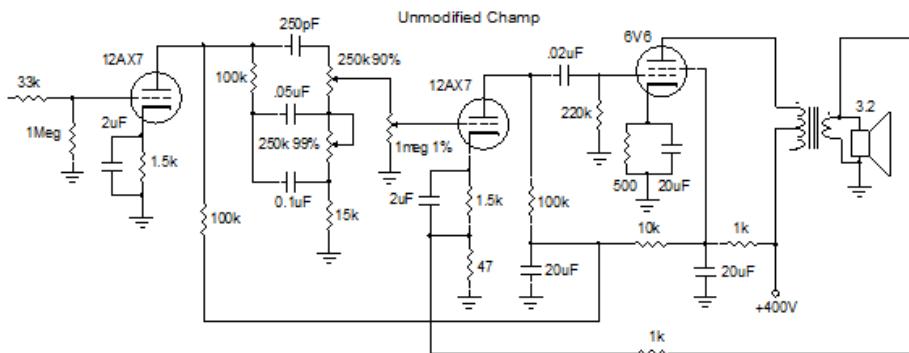
I thought about these things, I wasn't in a hurry and wanted a simple, easy implementation (and one that was readily reversible to restore the original configuration). First, addressing the increased power consumption, I looked up the Fender "Princeton" schematic on the internet and found that the power transformer was the same used on the Champ! That was a step forward - no expensive power transformer to buy, mount and re-wire.

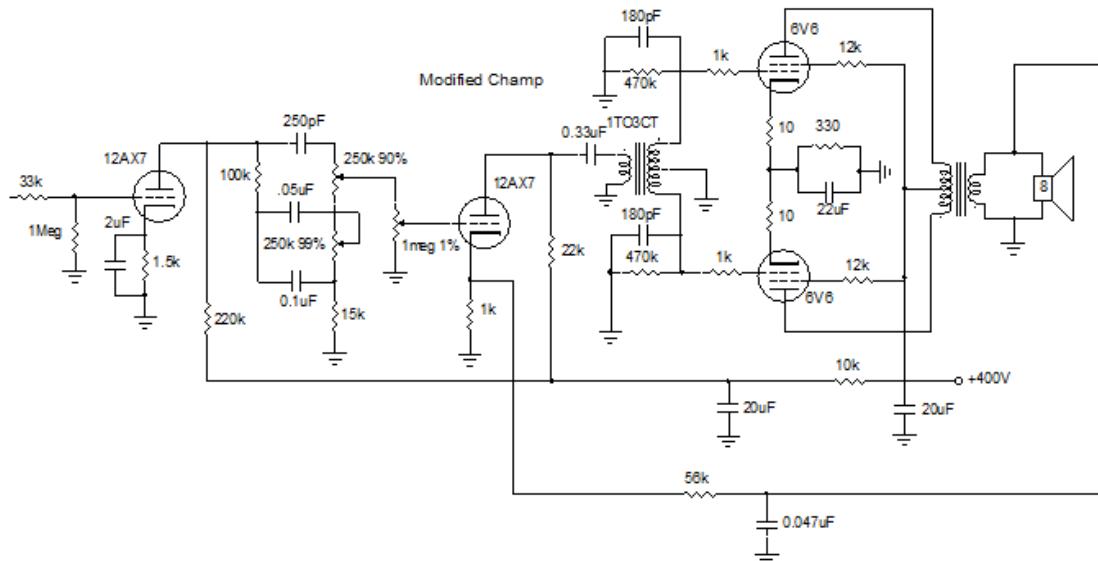
Then I considered addition of the phase-splitter tube. The purpose of the phase-splitter is to divide a single signal into two signals with opposite phasing. It's not necessary to use a tube for this, a small transformer can provide this function. Since no appreciable power is being transferred, the "interstage" transformer is inexpensive and easy to obtain. I found a "1 : 3" center-tapped transformer for a few dollars. Only two small holes need to be drilled in the chassis to mount the transformer. The transformer eliminates one tube AND the filament/plate current required to power it.

Addressing the power amplifier stage, I *could* have replaced the single 6V6 with a single 6L6 plus a few component changes to increase the output power of the single-ended design but that approach would need about 120 millamps plate current, requiring a power supply redesign and more \$. (Push-pull Class AB and B amplifiers are much more efficient, using a pair of 6V6 tubes requires about 70 millamps which is within the capability of the Champ power supply transformer.)

Some design changes are required, adding more voltage gain for example. Happily this isn't too difficult and accommodated by changing a few resistor values. Adding the extra transformer in place of the phase-splitter tube does create a disadvantage. Transformers have limited bandwidth and consequently more phase-shift than any other circuit in the amplifier except for the tone controls. And excessive phase shift can cause oscillation if one doesn't account for it.

The solution for this problem was fairly simple. Frequency compensation was added to the two output tubes and to the feedback loop, three more capacitors added to the circuit. Here are the schematics (excluding power supply) of the unmodified Fender Champ and the "Princetonized" version.





The modification was fun, informative and successful. The measured output power was about 12 watts at 5% distortion. Computer circuit simulations suggest that the output stage is capable of almost 20 watts (if cathode bias resistors are eliminated and a negative voltage is used to bias the grids of the output tubes, as in the Fender "Deluxe Reverb"). *It is unlikely that the original power supply transformer or output transformer would last long at that power level, however.*

That happens to be my personal recommendation for the minimum amount of amplifier output power required to play with a jazz drummer. The project was "bread-boarded" to the extent necessary so that no actual chassis modifications were required. After modifying and measuring the performance, I restored the Champ to original configuration except for the speaker and output transformer.

I'm not necessarily recommending this project for anything but reading entertainment and illustration of some of the concepts discussed previously. Most people would find it far simpler to buy an amplifier with adequate output power. I was interested, however, in how far I could extend the Champ concept and the result was satisfying.

19.2 Adding "Line Output" or "Preamplifier Output"

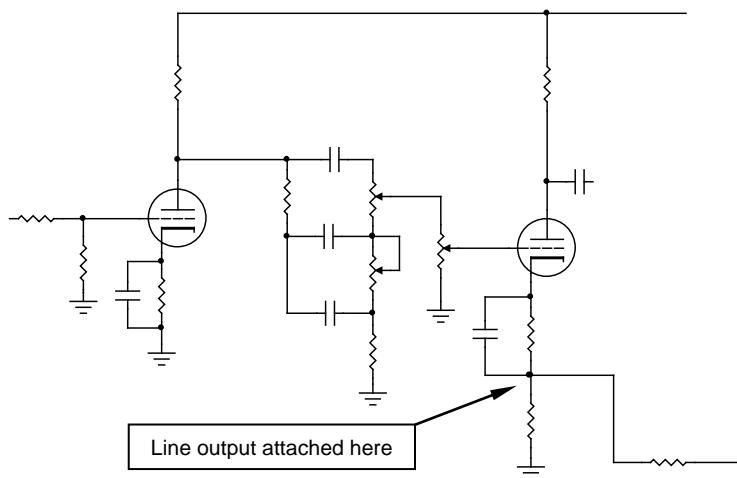
There are several reasons for adding a "line impedance output" to an amplifier, especially a small amplifier. The major reason is to be able to transmit the audio signal from your guitar appreciable distances without losing amplitude or high-frequency response. This is not possible by connecting a long cable to the normal high-impedance guitar pickup. Examples of situations where this is desirable would be to run the guitar directly to the mixing board in the house P.A. system, for sound reinforcement.

This allows using an amplifier as an individual stage monitor but not requiring much volume from the amplifier - just a few watts. Small, light amplifiers can then be used to play large venues without the problems associated with placing a microphone in front of your amplifier (feedback, incidental acoustic pickup). Another desirable feature is that, if the line output is appropriately placed, the preamplifier "sound" (the "warmth" that some believe is added by vacuum tubes) is provided by the low impedance output.

Other uses for the line output include routing the audio signal to recording devices, computer sound cards and additional high-power onstage amplification (it is possible to obtain a slight overdriven characteristic from all vacuum tube preamplifiers when the line output is connected to another guitar amplifier).

The correct placement for the line output in most vacuum tube amplifiers is in the post amplifier. This permits using the normal volume and tone controls of the preamplifier to "color" the line output. The cathode of the post amplifier tube provides a low impedance point with some voltage gain. There is no problem connecting a line output to this point provided that the other functions of the post amplifier are undisturbed.

One of the most common preamplifier/volume/EQ/post amplifier configurations is the one used universally by Fender. It's a simple, easily understood circuit and we have used variations of this circuit throughout this book. Here's a typical circuit:



There are several things to note about the point in the circuit to which we want to install the line output:

A series feedback resistor - from the output amplifier stage - is connected to that node

The shunt resistor for the feedback loop is also connected to this point

The cathode bias resistor is connected to the same point

A simple blocking capacitor will prevent disturbing the bias currents flowing through the resistor network but won't protect against the affects of external connections upsetting the feedback loop and causing oscillation. Happily, the shunt feedback resistor is universally a low value, typically less than 100 ohms. So if we attach a series resistor in the line output circuit, of a value about 5 to 10 times higher than the shunt feedback resistor, we can be assured of minimal effect to the feedback loop.

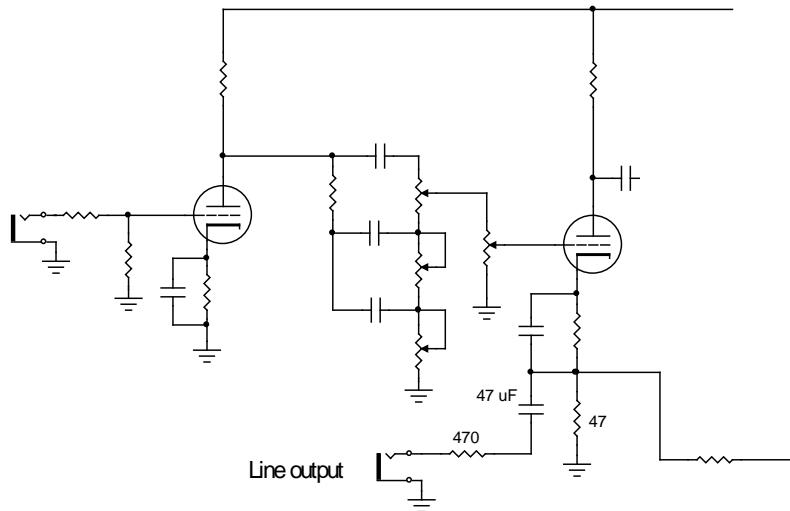
In the above circuit, the shunt feedback resistor was 47 ohms, so we could add a series resistor of about 470 ohms to the line output. The blocking capacitor must be large enough to pass the lowest frequency the guitar produces (80 Hz). We can use the value of the shunt feedback resistor to estimate the blocking capacitor value:

$$C_{\text{block}} = 1 / (2 \times \pi \times R_{\text{shunt}} \times 80 \text{ Hz})$$

$$C_{\text{block}} = 1 / (2 \times \pi \times 47 \times 80) = 1 / 23,625$$

$$= 4.233 (10^{-5}) = 42.33 \mu\text{F}$$

We can use the next larger standard value of 47 μF at a working voltage of about 12 volts. The line output circuit looks like this:



In order to avoid drilling a hole in the chassis of the amplifier (and to allow restoration to original configuration, if desired), the line output is usually wired to an input jack that is not normally used. It's helpful to mark this jack, even if the label is no more than a hand-written scrap of paper, taped near the jack.

19.3 Reducing the Power of Large Amplifiers to a Practical Level

Sometimes it may be desirable to reduce the output power level of certain amplifiers, such as the Fender "Twin", "Showman" or other older vacuum tube amplifiers designed in the manner of the Sunn "Concert" or Marshall 100 watt series. These amplifiers have four output power tubes, the circuit is the standard push-pull configuration but each half of the circuit includes two tubes, wired in parallel. This allows twice the normal operating current and results in roughly twice the signal power. Most of these amplifiers use multiple speaker configurations, as well.

Very high power amplifiers are sometimes difficult to adjust for the "right" sound in smaller venues. Halving the output power may result in a more manageable configuration; the process is simple and readily reversible. Removing one tube from each half of the output amplifier stage reduces current consumption and power output. Stress levels on the output transformer and the power supply transformer are proportionally reduced.

By removing two of the tubes the output impedance is doubled because the load line has changed. In certain "combos", such as the Fender "Twin", it's very easy to accommodate this change in impedance by simply disconnecting one of the two speakers (changing the speaker impedance from 4 ohms to 8 ohms). Doubling the speaker impedance transforms back to the output amplifier and accommodates the new load line, not to mention reducing the SPL (sound pressure level) to a more friendly level.

Other amplifiers, such as the Fender "Single Showman", don't allow such an easy impedance accommodation (this model was designed for a single 8 ohm, 18 inch speaker). The basic circuit is the same as the Twin except for the lack of reverb effect and the output transformer. By removing two tubes from this particular model, the new speaker impedance becomes 16 ohms, instead of the designed value of 8 ohms. This suggests using a dual speaker cabinet, two 8 ohm speakers wired in series, rather than the normal parallel configuration. (A more manageable combination might be a cabinet with dual 10 inch speakers.)

An amplifier, modified as above, will not sound very different than the original configuration but will have more practical (less sensitive) level adjustment. The enhanced reliability obtained by reducing the current by a factor of two will allow the two most expensive components in the amplifier (the transformers) to be stressed at a level such that their lifetime will be virtually indefinite. (It's good practice, when making this slight modification, to replace the output fuse with one that is about 2/3 the original current rating.)

19.4 *Eliminating the Need for Matched Output Tubes*

In Chapter 1.0, we talked about some of the design compromises that have been made in these amplifiers. Cost drove the design and as vacuum tube manufacturing capability was lost and tubes became less consistent, other circuit changes were made to the original designs.

Never very sophisticated, these circuits became even simpler in response to the need for maintaining (or increasing) output power levels while accommodating component inconsistencies. Feedback, in the form of cathode resistance, disappeared and adjustable grid bias networks became universal for amplifiers over 25 watts or so.

As jazz guitarists, we are less inclined to purchase an amplifier based on the amount of output power it can generate. Our selection criteria tend to be focused more on size/weight, fidelity (headroom) and then on power. A frequent favorite is the Fender "Deluxe Reverb", an amplifier with modest output power (about 22 watts). Replicating the performance of this iconic amplifier without the need for matched output tubes or "fiddly" bias adjustments would be a worthwhile project.

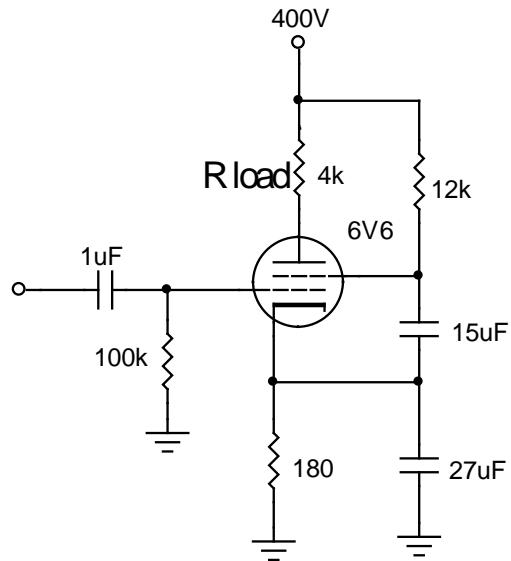
The most important aspect of this goal is achieving "matched" performance from unmatched output tubes. There are two separate issues involved, the static (bias) issue and the dynamic one involving performance under driven conditions. Unhappily, there is little to be done about the latter problem, we tend to ignore it since the ability to test and match under dynamic conditions is unavailable outside of a well-equipped electronics lab. The "matched" tubes that we purchase today are matched only from the bias aspect and they seem to function satisfactorily so our efforts will be limited to achieving that same level of performance.

As in many other performance variation problems, the key to obtaining consistent results from unmatched tubes is the use of a feedback system. We won't use the vacuum tubes themselves as the feedback network but they will form a *part* of the system. In Chapter 18.0 the use of mixed active devices was discussed and some justification was provided for the use of solid-state components within a system that is largely dominated by vacuum tubes. The current topic suggests an ideal application for the use of inexpensive power transistors to achieve the goal of consistent (and automatic) biasing of the output tubes.

First a brief review of the operation of tetrode and pentode beam power tubes. As in any vacuum tube, primary current flow occurs between cathode and plate, the amount of current being determined by the applied voltage on control grid 1. Tetrodes and pentodes have an additional grid, the screen grid (also referred to as grid 2) which *also* controls the current flow from cathode to plate.

The general configuration of a beam power tube circuit is with fixed screen grid bias voltage, adjusting the plate current by varying control grid 1 voltage. Here's

an example circuit, in this case a Class "A" power amplifier circuit designed for a 6V6 tube with a load resistance of 4,000 ohms, driven at 1 kHz with a signal of 16 V p-p (8 volts peak):

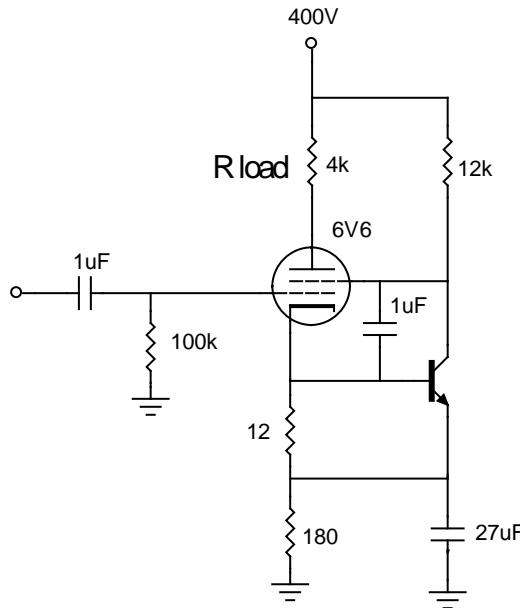


Assume that we now remove the 6V6 from its socket and replace it with a 6L6 and then a 6JN6, measuring the circuit performance for each tube and then summarizing.

Note, we are NOT suggesting this as practical exercise! It's a worst-case computer simulation intended to show how profoundly the use of feedback loops can influence stability and performance.

Tube Type	Plate Current	Voltage Gain	Output Power	Distortion
6V6	56 mA	18.0	10.4 watts	4.3%
6L6	76 mA	16.2	10.6 watts	28.0%
6JN6	91 mA	0.9	1.06 watts	25.6%

It is obvious - and expected - that the measured parameters vary considerably as plate current changes, most especially the distortion. Now let's add feedback to the circuit by including an inexpensive, high-voltage power transistor:



We'll repeat the same tests performed above, substituting the three tubes and recording the measurements.

Tube Type	Plate Current	Voltage Gain	Output Power	Distortion
6V6	46 mA	17.0	8.8 watts	4.4%
6L6	51 mA	20.7	10.9 watts	5.1%
6JN6	53 mA	25.2	11.6 watts	9.5%

Note the consistency of all of the measured data resulting from the feedback loop. Without diverging too far from the subject of vacuum tubes, let's explore the function of this circuit just enough to understand how this dramatic improvement in circuit consistency occurred.

The bias current, as we mentioned previously, is the critical performance parameter. It should be apparent that if the bias current can be stabilized for each of the different tubes used in the above analysis, then stabilizing two of the SAME tubes with slightly different characteristics will clearly be within the capability of our simple feedback loop. This is how the feedback circuit functions:

The control grid must have a negative bias that exceeds the peak input signal voltage

The bias is the voltage drop across the cathode resistance - in the above circuit, the current is about 50 mA and the resistance is 192 ohms total, so the grid bias is $.050 \times 192 = 9.6$ volts which exceeds the peak input signal voltage of 8 volts.

If the input signal level exceeds the control grid bias voltage, severe distortion will result and positive grid current will flow, affecting the long-term reliability of the tube.

A 12 ohm resistor has been added, in series with the cathode resistance. This resistor will establish the plate current through the tube.

The current will be about: $I_b = 0.65 / R$ and for $R = 12$, the plate current is about **50 mA**.

The transistor added to the circuit controls the current flow through the screen grid bias resistor. If the current increases, the voltage drop across the resistor will lower the screen bias voltage, if the current decreases then the screen voltage will increase. Altering the screen grid voltage will cause the plate current flow to change.

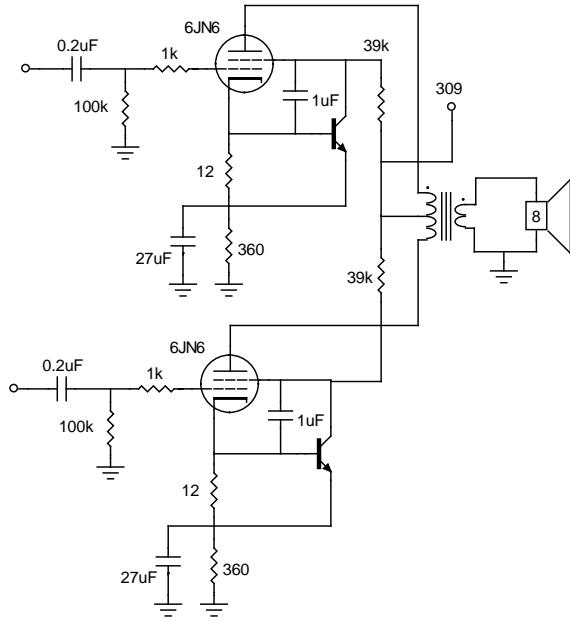
The amount of current that the transistor allows to flow through the screen resistor is dependant upon the plate current flowing through the 12 ohm resistor in the cathode circuit. The transistor starts to turn "on" when the current flow through the 12 ohm resistor causes the voltage to be 0.6 volts.

If the plate current causes a greater voltage drop, the transistor will cause more current to flow through the screen resistor, thus reducing the screen voltage and diminishing the plate current.

The "system" is a closed feedback loop, each element affecting the next, stability is achieved when the voltage drop across the 12 ohm resistor is just adequate to "turn on" the transistor (about 0.6 volts).

At this point, one might question whether the feedback loop has an effect on any other characteristics of the amplifier (or more accurately, a *negative* effect). Provided that the circuit is properly implemented - both capacitors shown must be included - no problem will be experienced. The loop bandwidth is very low; it has a very slow response time so normal audio signals are unaffected.

Here's the implementation in a conventional push-pull amplifier output stage. A stage that includes these feedback loops will not require matched output tubes.



Note that the above schematic is an example of the bias feedback technique described - it's not intended to be a functional circuit to be copied and used in any arbitrarily selected amplifier.

Solid-state design is a complex process and does not generally resemble the rough approximations we've been using for these vacuum tube designs. There are serious thermal and reliability considerations associated with this circuit that need to be taken into account as well as considerations of loop stability. Although the task is not a difficult one for an engineer accustomed to designing solid state circuits, it's probably not a project that one would normally expect amateur experimenters to undertake without supervision from an experienced engineer or technician.

As previously stated, we substituted various audio power tubes in our example circuit only to illustrate the concept of incorporating feedback into biasing techniques. *There is no recommendation or implication that arbitrary tube substitutions provide any performance advantage.* The point of this particular discussion is that shortcomings of current vacuum tube technology could be alleviated by solid-state devices. This isn't recommended for experimenters without knowledge of solid-state design.

19.5 Replacing Tube Rectifier With Solid-State Rectifier

Several advantages result from this routine upgrade. The (expensive) rectifier tube will never require replacement and higher plate voltage will be available for the amplifier circuit. An analysis needs to be performed before making this change, since an increase in power supply voltage of from 20 volts to over 100 volts might be expected!

Examination of the manufacturer's original schematics usually will provide enough "clues" to determine operating voltages/currents (and of course these parameters can always be measured, provided that a signal generator and a high power load are available). Adding the expected supply voltage increase to the existing voltage will yield the new operating voltage. The screen voltage of the output tubes will most probably increase too, so plate current will increase and the power dissipation will doubtless be excessive.

Here's a suggested procedure to perform before replacing a tube rectifier with a solid-state rectifier:

Examine the data sheet for the existing tube rectifier. Based on the total high voltage power supply current consumption, estimate the voltage drop across the rectifier from the data sheet curve. Note that this voltage will be ADDED to the power supply voltage.

Make your own assessment of power dissipation in the output tubes - don't assume that the manufacturer has properly de-rated these parts (especially if the output tubes have a short replacement history).

Examine the plate curves for the output tubes and determine the additional current that will flow as a result of the increase in screen grid bias voltage. It's virtually certain that the maximum power ratings for both the plates and the screen grids of the tubes will be exceeded.

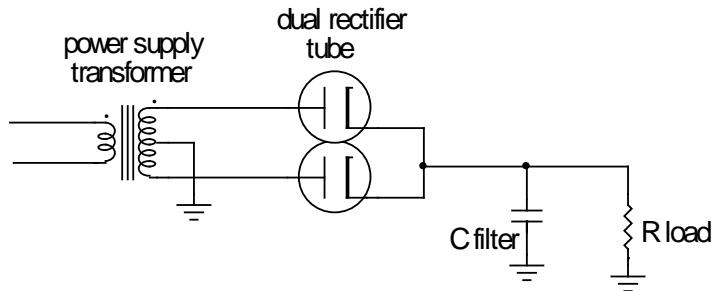
If the tubes are conservatively de-rated in their current bias configuration, you may gain some desirable headroom improvement by increasing the operation voltage to a SAFE maximum. By adding a voltage dropping resistor to the new power supply voltage, as described below, dissipation can be kept within allowable limits.

Estimate the effect on each of the tubes throughout the amplifier chain, based on the new predicted power supply voltage. Insure that no dissipation ratings are exceeded. If tetrodes or pentodes are used, be sure to account for the increase in screen voltage and the consequent increase in plate voltage and power dissipation.

If the amplifier chain can be safely operated at the higher supply voltage (especially the output tubes), then one can determine a practical implementation for the rectifier replacement then proceed with the project.

Assuming that all is well except for the output tubes, one can refer to Chapters 8.0 and 16.0, using the procedures described there to adjust bias conditions for safe operation while still maintaining the load line appropriate for the existing transformer and speaker configuration.

The easiest solution, if one is interested only in the reliability aspect of vacuum tube rectifier replacement, is to add a voltage-dropping resistance to the solid-state rectifier. This is the practical equivalent of the tube rectifier and nothing further in the amplifier chain need be modified if this approach is implemented. Here's a typical schematic and we'll determine how to make the modification:



Avoiding reference to specific amplifiers, let's assign some values to the rectifier circuit. Assume that the RMS output voltage of the transformer is about 400 volts, center tapped (200 volts RMS at each "leg" of the transformer). The maximum, unloaded D.C. output voltage available will be:

$$(2)^{0.5} \times V_{rms} \text{ or } 1.414 \times 200 \text{ volts} = \mathbf{283 \text{ volts}} \text{ (the "peak" voltage)}$$

Assume that the required current is 250 mA and that the original rectifier tube is a type 5U4.

From the RCA "Receiving Tube Manual (RC-30)", the 5U4, operated at a load of 250 mA, for a CAPACITOR input filter - the most common configuration - with an RMS input voltage of 283 volts per plate will produce a D.C. output voltage of 270 - 280 volts, we'll use 275 volts.

Assume that, by measurement or by schematic deduction, the actual circuit voltage under load is 265 volts. Then the estimated voltage drop across the 5U4 rectifier tube is

$$\mathbf{275 - 265 = 10 \text{ volts}} \text{ at a current of 250 mA}$$

which would represent a dropping resistor value of

$$R = E / I = 10 / 0.25 = \mathbf{40 \text{ ohms}}$$

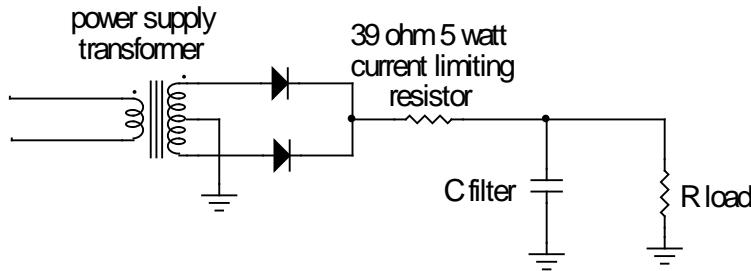
and the power dissipated by the resistor is

$$P = E \times I \text{ or } P = I^2 \times R \text{ or } P = E^2 / R$$

any of these will work; using the first expression we get

$$P = 10 \times 0.25 = 2.5 \text{ watts}$$

Applying a de-rating factor of 2 and picking a standard value, we get a 5 watt current limiting resistor of 39 ohms (a standard value). We can insert this value between the solid state rectifiers and the filter to obtain the following schematic:



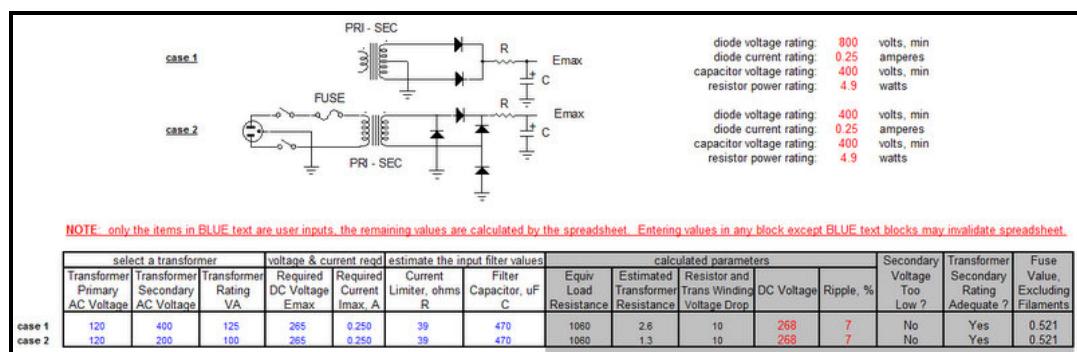
The diodes can be selected by their minimum PIV rating. PIV means "peak inverse voltage" and the rating must be greater than the peak voltage for the transformer. For the above example, the peak voltage is twice the peak voltage of the individual windings and is given by

$$V_{\text{peak inverse}} = 2 \times 1.414 \times V_{\text{rms}} \text{ or } 586 \text{ volts}$$

To assure safe de-rating, we'd select a diode with a PIV about double the required value. The diode must also be able to pass one-half of the required current of 250 mA. Since we usually de-rate solid-state devices by a factor of two, the diode must be rated at 250 mA. A 1N4006 diode, commonly available and inexpensive, would be a likely choice, it is rated at 1 amp and 800 volts.

Note that the power supply voltage will not be constant, the voltage will be greater under no-signal conditions, it's wise to re-compute the output stage power dissipation based on the new quiescent voltage and current.

The spreadsheet discussed in chapters 6.0, 12.0 and 22.0 for designing power supplies can be used to make many of the routine calculations used in this exercise. Here's a screen capture of the spreadsheet (case 1) after entering the appropriate information:



19.6 Tube Substitution

This topic was once a familiar one to virtually any consumer of electronic entertainment devices. It was possible to buy paperback tube substitution guides (\$0.25 sell price) from the local grocery store at one time (self-help tube testers and stocks of common vacuum tubes were also to be found in grocery stores). The substitution guides were helpful if the limited stock didn't happen to include the particular tube from your television that was possibly defective.

That's not a common situation now, most people that are interested in substitution have a specific goal in mind - perhaps performance enhancement, cost reduction or maybe finding a tube that is no longer commonly available (owners of early sixties *Ampeg "Reverberocket"* amplifiers will identify with the latter). In point of fact, it's not practical to find exact pin-compatible replacements for any vacuum tube products that are in current manufacture (although one may locate a NOS source).

Readers of this book, almost by definition, won't trouble themselves about locating pin-for-pin replacements anyway. Readers are assumed to have the knowledge required to make informed decisions about tubes (and the implementation of a practical solution) if a substitution seems desirable. Regardless of what one's goals are in making a substitution, here are some obvious considerations:

Filament voltage (and current)

Plate current, voltage and power dissipation

Screen grid current, voltage and power dissipation

Control grid voltage (with special regard to whether an existing circuit will overdrive the replacement tube into a positive-bias situation)

Maximum control grid resistance (from data sheet), if exceeded positive grid current can flow - potential reliability problem

If the replacement is a triode, be aware that many triodes are limited in amplification factor - 40 is a typical maximum value

Effective parallel plate resistance (lower plate resistance will reduce gain)

For single tube phase-splitter applications, examine the plate curves to confirm that the tube will function properly for the low plate-to-cathode voltages that may be encountered in this application

Having considered the obvious, here are some slightly more subtle concerns that may require some circuit modification.

Plate curves - can the existing circuit accommodate the bias requirements of a different tube?

Make a prediction for the bias current and plate voltage to be expected from the substitution or plan for the inclusion of a circuit adjustment component (to accommodate a different tube type).

If the existing circuit plate voltage chain is the typical resistor-capacitor-resistor-capacitor configuration, seemingly minor changes in screen and plate currents may cause fairly significant changes "upstream". Estimate the voltage drops across the various resistors in the plate voltage chain.

Consider replacing a triode with a low-power tetrode, they are available NOS (cheap) in dual configurations that usually include a triode. The beam tubes can provide a LOT more gain than the typical triode which may be advantageous.

Unless one has the "Luddite" mindset then consider the inclusion of solid-state devices - even complete circuits - if performance or cost advantage is suggested. Inexpensive, high-voltage MOSFETs allow a painless way of replacing a tube function with a more reliable, inexpensive solid-state device.

Occasional suggestions are oriented toward cooling a vacuum tube amplifier. I have made thermal estimates of the result of forced air cooling and concluded that there is little advantage to be gained *for the tubes*. But the other components contained in the amplifier assembly may benefit from this "upgrade" - transformers especially. Since transformers are costly, cooling might be a good idea.

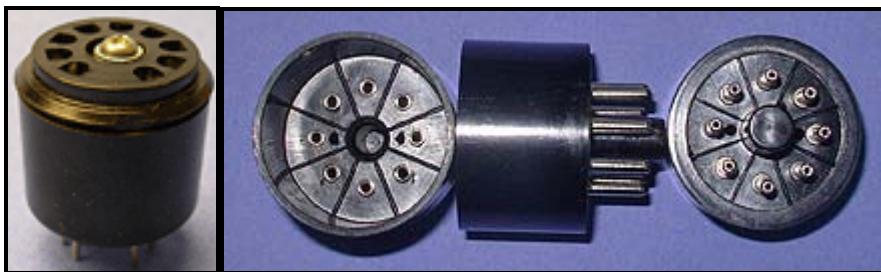
Sometimes better sound reproduction can be obtained by partly sealing an open-back cabinet (may be noticeable after a speaker replacement). In this case, forced air cooling might be recommended, giving a LOT of thought to the ingress and egress of air flow and the impact on the sealed speaker enclosure.

Note that it's NOT a good idea to locate small fans or blowers near the loudspeaker(s). The powerful magnetic field of the speaker(s) will impair proper operation of the fans, sometimes to the point that the motor will not rotate.

If the amplifier construction permits (physical space above the tubes), it's frequently possible to package small circuits (like the bias feedback circuit

described previously) in such a manner as to provide a plug-in "adaptor". The package would consist of a tube base and a tube socket, with a section of plastic "pipe" that joins the two and contains any circuits that are desirable for inclusion.

Representative of this concept are the various adapters that allow replacement of output tubes with those of different pin configurations or socket types. Solid-state rectifier replacements for vacuum tube rectifiers are also packaged in this manner. The parts are readily available, all that is lacking is the ingenuity of a circuit designer and the skill to assemble the "adaptor". The photos below show an assembled adaptor and the individual parts of the assembly:



20.0 Test Equipment, Build or Buy ?

The equipment required to make routine measurements and performance evaluations of vacuum tube circuits is not particularly expensive or hard to find, with the exception of high voltage adjustable power supplies. "Ebay" and other online sources offer the opportunity to obtain old, but high-quality, instruments at bargain prices.

Many of my test instruments function as accurately as when new (but cost 5% to 10% of their original value). It can be rewarding and cost-effective to design and build one's own equipment. It's also useful for educational purposes and in some cases may actually be the primary means of obtaining specialized items like the high voltage lab supplies mentioned previously.

20.1 Mandatory Test and Measurement tools

In addition to a modest collection of hand tools and a soldering iron, the following items are required for even modest electronic projects:

Inexpensive digital multimeter (DMM), at least TWO of these (\$20 ea U.S.)

Audio generator (\$150 U.S.) but also see chapters 20.3 and 20.6

High power speaker load (\$10 U.S. in parts)

Plate voltage power supply (can be eliminated if one first builds the power supply for the intended project and the supply is protected from routine mishaps)

Miscellaneous: scraps/blocks of wood (to support an upside-down chassis or enclosure), small vise, collection of connectors/adapters/test leads

The list of *desirable* test equipment and supplies is unending, it should be noted. For years, I got along pretty well with a "milk crate" (stored in a closet) that contained a Triplett Volt-Ohmmeter (VOM), soldering iron, hand tools, test leads and a small, multi-drawer parts cabinet with a collection of components. I don't know how I got along without at least an oscilloscope but I did and you can too.

At the next tier of equipment, in addition to the above, I'd include:

Oscilloscope, single channel analog (about \$200 U.S.)

Additional power supplies, up to about 350 volts both positive and negative (not commonly available, about \$100 U.S. parts cost to build)

Additional signal generator (actually, a "function" generator with the capability of providing square and triangular wave outputs in addition to sinusoidal, about \$200 U.S.)

Software for a "laptop" computer (with sound card), to emulate various pieces of test equipment (e.g. spectrum analyzer, oscilloscope, audio generator, distortion analyzer)

And if you have room on your workbench, you may end up with something like this:



Much of this equipment in this photograph is old and out of date. Much of the gear is actually vacuum tube powered. In other words PERFECT for my design, breadboard and test needs.

20.2 High Voltage Power Supplies

A pair of high voltage power supplies is shown below, the upper one is rated at 0 to +355 volts (fully adjustable) at 180 mA for plate supply application and includes forced-air cooling and current limiting protection as well as a filament supply of 6.3 VAC at 6 amperes.

The lower supply is a dual polarity medium voltage supply for screen grid and control grid bias application. Voltages available are 0 to +250 volts at 50 mA and 0 to -100 volts at 20 mA (both fully adjustable), forced-air cooling and current limiting is included. The cabinets are made of wood for insulation purposes and because wood is cheap and easy to work. All voltage terminals are mounted on the back panel so that they are not accidentally "accessible" to the user.



We can use the material contained in Chapter 12 as the basis for an adjustable power supply. The negative half-wave adjustable supply described in that suggestion is adequate for most purposes when bench-testing, all that would be

normally necessary is to "package" the supply in an appropriate enclosure and provide a fuse, output terminals and an adjustment knob.

20.2.1 Regulated Adjustable Plate Supply

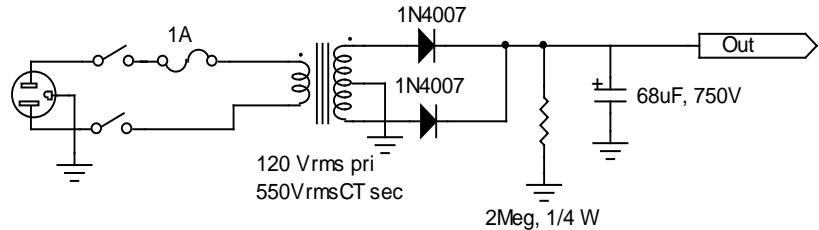
All that's required for a positive high voltage adjustable plate supply is to add "adjustability" to the basic plate power supply circuit in Chapter 12. The addition of a "pass transistor" (so called because the transistor "passes" all of the current required by the load) and some control circuitry will allow the adjustment of the output voltage.

A typical and convenient selection for a high voltage pass transistor is a high voltage MOSFET (metal-on-silicon field-effect-transistor). These are inexpensive and readily available, the control circuit is simple and the main problem is heat dissipation. Although power MOSFETs have data sheet ratings that suggest very large amounts of power dissipation capability, these ratings are made with the transistor "case" (the mounting surface) maintained at a low temperature.

The power limitation of the MOSFET is determined by the amount of heat generated internally and thus, the more heat we can extract from the device, the more power the transistor can dissipate. Sometimes this can be implemented by using multiple transistors but usually forced-air cooling is required in applications like ours. This isn't particularly complicated, just a matter of obtaining a small, readily available fan, providing the operating voltage and then efficiently coupling the fan thermally to the MOSFET mounting surface. The uppermost power supply in the previous photograph was designed to provide adequate plate power for amplifiers up to about 50 watt capability

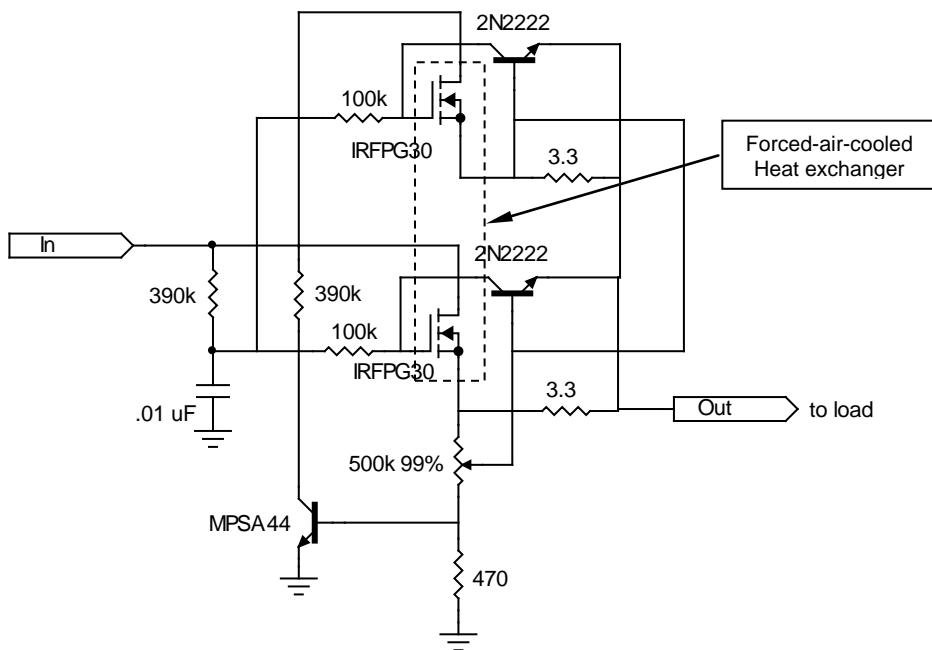
(An interesting note about this type of power supply is that it is most LIGHTLY stressed when providing maximum output voltage. This is because when the supply is operating with low output voltage, the maximum transformer voltage has to be absorbed somewhere and that "somewhere" is the pass transistor of the supply. If the power supply transformer can produce a rectified D.C. of 400 volts but the supply has been adjusted to provide a 200 volt output at 100 milliamps, then the pass transistor has to dissipate $(400 - 200 \text{ volts}) \times 0.1 \text{ A} = 20 \text{ watts}$. When the supply is adjusted to provide 375 volts at 100 mA, the pass transistor only has to dissipate $(400 - 375) \times 0.1 = 2.5 \text{ watts}$.)

Here's a schematic for the simplest plate power supply circuit, first starting with the basic rectifier and filter (the configuration is "case 1" as described in the power supply design chapter 12.0):



An alternative fixed power supply is described in chapter 26.0, that is less expensive (although more complex) and can provide modest voltage levels (about 275 volts in basic configuration, up to 400 volts is possible) and a practical current level of 200 mA. The circuit can provide A.C. and D.C. filament voltages of around 2 to 3 amperes. The example supply circuit would be suitable for an adjustable configuration if modified as in the following paragraph.

Here's a schematic for the pass transistors and control circuit (this simple circuit is designed for a maximum input of 375 volts and maximum current of 400 milliamps):



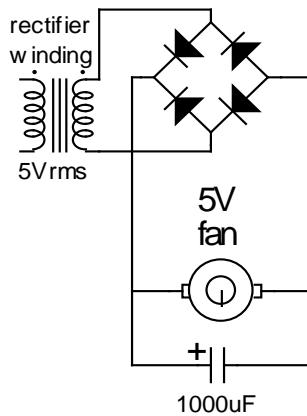
No metering circuits are shown above, they may be added as the user desires. I chose low cost digital panel meters for my power supplies that included a circuit schematic (which made installation fairly simple). Analog meters would have been even simpler and they require no external power supply. From a human engineering aspect, note that analog meters are almost always preferred to digital ones; variations in voltage and current are easily recognizable by movement of an indicator needle, rather than by the slowly changing digits in a display.

Note that two pass transistors are used, they are mounted to separate heat sinks that are placed directly in the ducted air flow path of a small 5 volt fan. Some thought is required when mounting the MOSFET pass transistors to the heat sink (because the mounting surfaces of the MOSFETs are an electrical connection as well). The circuit also uses current limiting (the 2N2222 transistors and the 3.3 ohms resistors) to prevent overstressing the pass transistors and to force them to share the load current.

It's well beyond the scope of this document to discuss in detail the design and function of the power supply, although it's actually fairly primitive. Those who are serious about the design of vacuum tube amplifiers and desire to construct a fully adjustable power supply are recommended to expand their interest to include solid-state semiconductor design. This class of experimentalist may well be familiar with the design of power supplies, so additional explanation might well be unnecessary and/or even confusing since it is not our primary purpose to design/build solid-state test equipment.

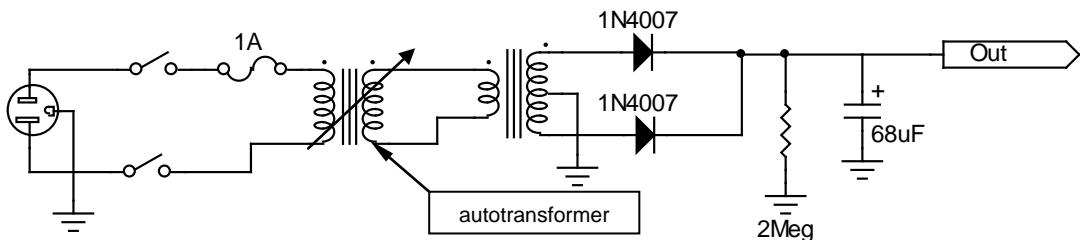
There are simpler alternatives to this type of power supply, one of which will be discussed in a moment. Regarding this circuit, it's fairly well regulated but doesn't include a precision voltage reference so the output voltage will vary somewhat with temperature. All adjustable supplies will have the same design problems that this one does - regardless of how the supply voltage is controlled. Care must be exercised in the selection of parts (voltage and power rating) and removal of heat from the pass transistors.

The fan which cools the pass transistors is powered from an unused winding of the power supply transformer (the five volt winding normally provided for tube rectifiers). A simple rectifier/filter circuit is adequate for the minimal amount of current required by a small fan, the following circuit was adequate for my supplies. Note that there is no ground connection to rectifier or fan and that the wires should be twisted, like filament wiring.



20.2.2 Unregulated Adjustable Plate Supply

A far simpler alternative (although more expensive, without regulation and requiring a separate filament transformer) can be configured using a variable "autotransformer" or "Variac" as it is sometimes called. The design procedure is also quite simple: an autotransformer of the correct rating (volt-amperes) is selected and then configured to precede one of the plate voltage supply designs from Chapter 13. Here's the simplest adjustable plate voltage supply:



Autotransformers function **ONLY** with alternating current (A.C.) - never connect an autotransformer input to any circuit other than the household "line" voltage outlet. Always provide protection for excessive current draw (as in the accidental "short circuit"), to prevent this costly part from being destroyed by a wiring error or the like, if the autotransformer is not fuse-protected.

Autotransformer construction is similar to a normal transformer except that the secondary winding number of turns can be varied mechanically. This is accomplished by sliding an electrical contact (called a "wiper") over the secondary winding so that the wiper contacts first one turn and then another as it is moved across the winding contact area. (This is the same principle as the "potentiometer" except that the potentiometer uses a resistive element rather than a transformer secondary.)

The result is that the autotransformer A.C. output voltage is varied from 0 to the full rated output voltage. This is a very convenient way of adjusting the rectified D.C. output voltage of the power supply connected to the autotransformer, with disadvantages as described above. The rating of the autotransformer should be at least the same as the power supply transformer, preferably greater.

A separate filament transformer is required when using a Variac-controlled power supply, otherwise plate and screen voltages will be applied to the tube well before the filaments have reached operating temperature. (This condition has reliability implications.)

The cost of these devices is generally around \$100 U.S. Here's a photograph of a typical autotransformer, this one a low-cost Chinese-made version:



Observe that a line cord is required to connect the autotransformer to the power supply. This particular model is rated at 500 VA (volt-amperes). To be safe, it's normal to de-rate these devices by 50%, therefore the operating conditions shouldn't exceed 250 VA at the input:

$$\text{VA rating} = E \text{ input} \times I \text{ input}$$

where the two variables are input voltage and input current
250 VA rating and normal line voltage, substituting and re-arranging:

$$I \text{ input} = 250 / 120 = \mathbf{2.1 \text{ amperes}} \text{ maximum input current}$$

which suggests that the value of the fuse used for the above autotransformer would be the next greater standard size, around 2.5 amperes.

Regardless of the configuration and type of high voltage supply constructed, be sure to include a bleeder resistor, as discussed in chapter 12.0. Another convenient feature is a "standby" switch, as is usually included on higher power guitar amplifiers. In case of a fault during test or repair, it is usually quicker to switch the standby switch off rather than rotating the adjustment knob to its minimum setting.

Powering the supply "off" is not useful since the filter capacitor stores a great deal of energy that requires time to "bleed" off.

20.3 Signal Generator

The signal levels at various points in a vacuum tube amplifier chain can vary from as little as a few *millivolts* to *hundreds* of volts, peak to peak. This implies that a

signal generator is required that can generate hundreds of volts (with an attenuator to reduce the output to millivolt level) or conversely, one that can generate low voltage levels at fairly high current levels which can then be *transformed* to high voltage levels. The latter approach is the most economical and safest, although not necessarily the highest performance. Fortunately, testing and trouble-shooting these amplifiers doesn't require equipment with laboratory-standard performance.

There are numerous simple circuits that will provide the performance needed for most applications, available from internet search, examining electronic hobbyist magazines, textbooks and the like. However, the purchase of a low-frequency generator with lots of capability isn't a major expense.

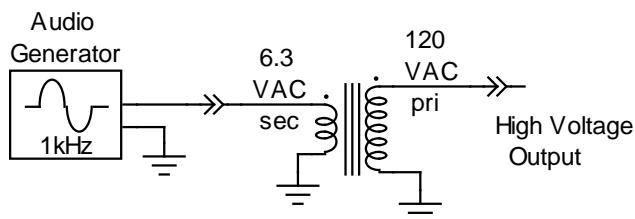
An internet search for "hobby audio generator" or the like should turn up numerous commercial instruments more than adequate for our requirements at modest cost. Many electronic distributors offer new instruments with impressive performance for far less than \$200 U.S. These are sometimes also offered in kit form, which can be useful as a training exercise in workmanship and schematic reading.

We actually require very little from an audio generator, any circuit that can achieve the following performance would be satisfactory:

Frequency: 100 Hz to 10 kHz, adjustable
Output level: 0 to 10 volts peak to peak, adjustable
Waveform: sinusoidal (sine wave)
Output impedance: 50 to 600 ohms

The need for high output voltage occurs when testing phase-splitters and power output stages but we don't need a separate generator for this. We can use any low impedance audio generator capable of providing about 10 volts peak to peak output and drive an inexpensive transformer, obtaining a high output voltage.

All that's required is a 0.25 amp (or greater), 6 volt secondary transformer with a 120 VAC input. This would normally be operated in a "step-down" configuration (output voltage is lower than input voltage), however if we reverse the configuration, connecting the 6 volt leads of the transformer to the output of our signal generator, we can extract a high-voltage output signal from the 120 VAC leads:



Some low-cost power transformers that I have measured have a useable bandwidth of about 50 Hz to 10 kHz, adequate for our needs. One would have to repeat the measurement on candidate transformers to verify that they are adequate for your own requirements since some are not at all suitable. The cost is about \$2 - \$3 U.S. and the transformer can be mounted either internal to a home-built generator or in/on any convenient external manner (block of wood, metal plate, purchased chassis box, etc) with the appropriate connectors soldered to the transformer terminals.

Used high output voltage audio generators may often be found (e.g. Ebay), the older Hewlett-Packard HP-200 series are laboratory standard instruments sometimes available at low cost. Although 1940-1950 vintage, these instruments compare favorably in most respects with modern, synthesized generators and have the additional advantage of high output voltage capability.

20.4 High Power Loads (Speaker Simulators)

This is a necessary item for testing amplifiers of even modest power level. Simply clipping a pair of test leads across a couple of 25 watt resistors doesn't approximate a resistive load very well, considering other variables.

First, it's well known that loudspeakers are not purely resistive and that they don't represent constant impedance as a function of frequency. Nevertheless, resistive loads are necessary for several reasons:

It is impractical to fabricate and present a load to an amplifier that represents ALL loudspeakers under ALL conditions

resistive loads are the ONLY practical means to measure power because reactive loads do not absorb power, they reflect power (or at least, substantial amounts of power, which makes accurate power determination impossible)

without knowing exact real and imaginary (i.e. $r + jX$) components of the load, the determination of power could not be consistently determined (and agreed upon) from one measurement site to another measurement site

Power resistors are readily available in the ranges required for simulating the impedances of commonly available speakers. There are several problems associated with using a standard power resistor for power determination, as an example.

resistors get HOT, they must be placed in a location that doesn't permit inadvertent contact

resistance values change with temperature - may or may not be critical, depending upon what measurement is being made and what accuracy is required of the measurement

Numerous practical solutions are available, such as mounting resistors on passive heat exchangers, air cooling and so forth. Individual requirements might well allow a single power resistor, or a pair, to be located in such a manner as to allow convection/radiation cooling without the possibility of contact.

My personal requirements for power measurement included a load capable of dissipating at least 50 watts indefinitely and 100 watts for prolonged periods of time (up to 30 minutes). Given these requirements, I determined that a simple, inexpensive, liquid-cooled load could meet my requirements.

My "dummy load" consists of two 16 ohm 25 watt resistors, wired in parallel. The cooling technique was implemented as follows:

obtain empty 1 quart paint can from local hardware store

punch two 3/8 inch holes in lid of the paint can

paint the outside of the can "flat black" to radiate heat (important)

install and secure two insulated binding posts through the holes in the lid

solder the pair of resistors to the insulated binding posts, first trimming the resistor leads so that when the lid is attached to the paint can, the resistors will be located centrally within the volume of the can
(my curiosity compelled me to drill another small hole in the top and install a tiny rubber grommet, through the center of which, I forced the shaft of a meat thermometer so that I could monitor internal temperature)

fill the can with cooking oil

lower and seat the lid (plus the resistors and the thermometer) into the can and seal tightly

As a matter of interest after several hours duration, powered at 50 watts, the meat thermometer indicated that temperature of the cooking oil was about 210 degrees F (100 degrees Celsius). The power ratings/resistance values of the resistors are stable and predictable at this temperature. (For 100 watt dissipations, constant amplifier operation should be limited to less than 30 minutes unless the load is allowed to cool or is cooled by forced-air.)

Obviously other combinations of resistors (switches too) can be included to make the load suitable for the normal speaker impedances likely to be encountered.

The power rating of the resistors used for the load should be AT LEAST as great as the maximum amplifier power level to be measured if liquid cooled and twice as great if air-cooled.

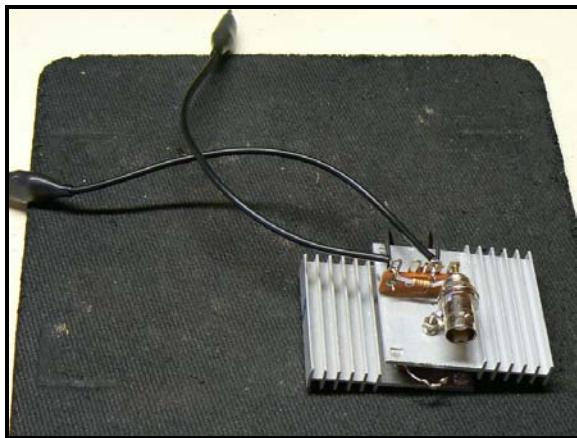
The cooling provided by the cooking oil in the sealed container will stabilize resistance values for prolonged periods of time - long enough to make adjustments and corrections within the amplifier circuit. *Adding to other cautionary notes included here and elsewhere, continuous operation of high power amplifiers WITHOUT MONITORING PERFORMANCE is NEVER recommended. Large amounts of heat can be generated over long periods of time. Power supplies must always be switched "off" when the amplifier is not under direct examination.*

Here's a photo of my version of a high power load, the "meat" thermometer is visible between the binding posts:

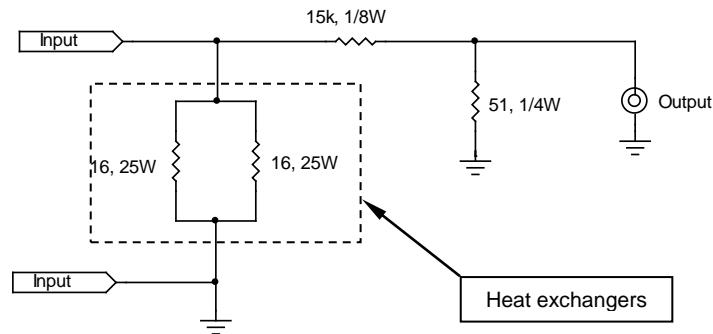


This is another variation of the high power load, this one is rated at 25 watts RMS continuously. Two 25 watt 16 ohm resistors, wired in parallel, are "sandwiched" between aluminum heat exchangers. Connected to the resulting 8 ohm load is an attenuator and a "BNC" connector (normally used for precision test equipment).

The attenuator reduces the full rated input power level to about 1 milliwatt so that a variety of test/measurement equipment may be connected. Many pieces of test equipment are far too sensitive to be able to directly measure high power levels (e.g. spectrum analyzers and some analog/digital converters, as in a computer "sound card") the attenuator allows sensitive measurements while the amplifier is still being driven fairly hard.



The two "alligator" clip leads allow connection to a convenient point where a measurement is desired. NOTE: No blocking capacitor for high voltages is included in this particular device - it can only be connected to parts of the amplifier circuit which do NOT have a D.C. potential. Connecting the device to points within the amplifier that are not at zero volts D.C. will damage either the load or the amplifier itself. The primary intent of this device is connection to the output of the amplifier for distortion measurements under drive. Here's the schematic for the above load/attenuator:



The output attenuator is configured to suit instrumentation with an input impedance of 50 ohms. If other impedances are desired (600 ohms is another "standard") then the 15k and 51 ohm resistors can be scaled by the ratio:

$$R_{desired} / 50$$

(It should be noted, however, that for impedances higher than 50 ohms, the series resistor (15k in the above circuit) can become so large that it starts to become inductive in nature and may require compensation in the form of a small value shunting capacitor.)

For increased sensitivity, the 15k resistor can be reduced in value. If one chooses to do this, the maximum input level of the instrumentation connected to the output of the attenuator needs to be carefully considered. Expensive

damage could result from over-driving precision test equipment. (Normally an output power level of 1 milliwatt is suitable for conventional test instrumentation.)

A later chapter discusses using a computer sound card as a signal generator. A protective circuit will be shown which *does* provide the protection against DC voltages that the circuit above does not provide. The reason that the above circuit doesn't have a blocking capacitor is that the low speaker impedance (8 ohms) combined with low test frequencies (down to about 40 Hz) would require an expensive high capacitance, high working voltage capacitor. Typical values would be about 1,000 microfarads (μF) and 500 volts - that's not an easy part to find.

In addition to the previous, it would be extremely unlikely to encounter a situation where high voltage DC would be present at the output of an amplifier. The implication would be that the primary winding of the output transformer (the winding through which the high voltage plate current is passed) has become shorted to the secondary winding (speaker winding) of the transformer. Should this have occurred, some indication would already exist (e.g. catastrophic damage to the speaker and/or blown power supply fuse).

Note, since I constructed the high power load described previously, I've located lower cost, higher power 8 ohm resistors. There are still concerns with the heat generated by these devices however. If the designer chooses to use a configuration that is not liquid-cooled, it is highly recommended that the resistor be located remotely, so that it cannot be inadvertently touched. Or perhaps the load can be cooled with a small muffin fan or the like; these are increasingly inexpensive and readily available in several A.C. and D.C. voltage options.

20.5 Pulse Testing

Pulse testing is a technique that allows operation at high power levels for very brief periods of time. We do this to alleviate a number of difficulties:

power supply voltage, current and dissipation inadequacies

tube current and dissipation problems

output transformer dissipation problems

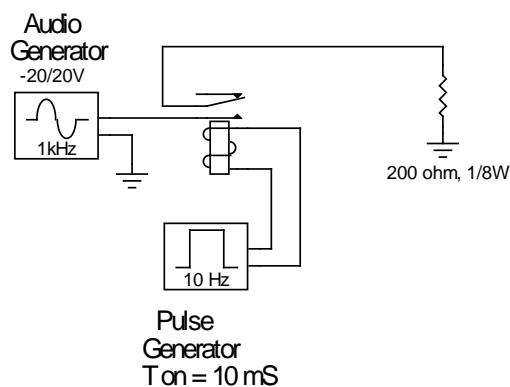
output load limitation

We can illustrate how the technique works with a brief example. If I were to quickly pass a finger through the flame of a candle, I might feel a slight sensation of warmth, no more. Clearly, if I hold my finger in the candle flame, intense pain would result as well as tissue damage. The difference between the two extremes is the duration of exposure to the heat source. The exact same technique is

used in pulse testing of electronic equipment to obtain results from high power, brief duration test exposure.

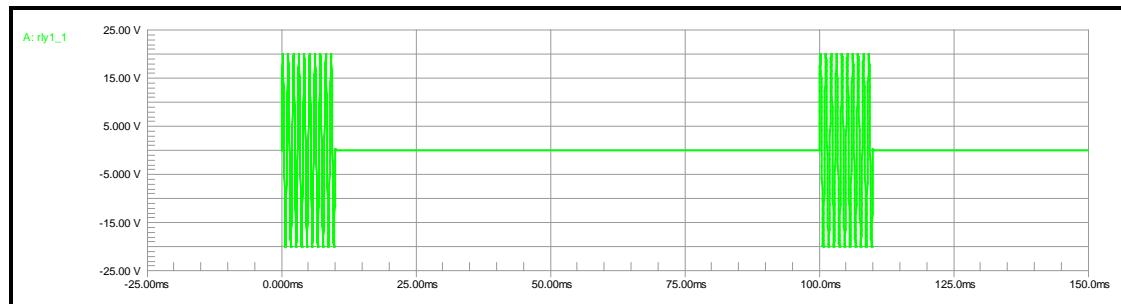
In chapter 20.9 a simple universal "breadboard" will be described that is useful for many things, including trying out power amplifier designs using an output transformer that is capable of providing many different plate impedances thus accommodating many different vacuum tubes. *The problem with this useful transformer is that it is only capable of 15 watts output power, or about the power capability of a Fender "Princeton".* Nevertheless, by pulse testing techniques, I've used the breadboard with its limited power output transformer to test 50 watt circuits many times.

Here's a schematic representation of the pulse testing technique:



The audio generator is the test source and is adjusted to provide the output signal frequency and desired level. In series with the audio generator is a relay (either mechanical or a solid-state switch) that can be opened and closed by another generator (a square wave generator is best but not absolutely necessary).

The pulse generator frequency and pulse width can be adjusted to allow the audio signal to pass through the relay at a rate and duration determined by the pulse generator settings. (The amplitude of the pulse generator is adjusted to properly operate the relay.) For example, here is the output waveform of the audio generator when the relay is switched according to the above pulse generator settings:



The audio generator is turned "on" for about 10 milliseconds every 100 milliseconds as can be determined by examining the waveform.

In the schematic above, we aren't testing an amplifier, just a single resistor. Assume that it is necessary to drive the resistor with a 1 kHz audio signal of 40 volts peak to peak. The power dissipated by the resistor under normal circumstances could be determined from any of the following expressions:

$$P = E \times I$$

$$P = I^2 \times R$$

$$P = E^2 / R$$

Where P is in watts, E is volts RMS, R is resistance in ohms and I is current in amperes. To calculate resistor power dissipation, we first need to convert the 40 volt peak to peak signal to RMS volts which is:

$$V_{rms} = .354 \times V_{p-p} = .354 \times 40 = 14.16 \text{ volts RMS}$$

and the power, using the last expression is

$$P = E^2 / R = (14.16)^2 / 200 = 1 \text{ Watt}$$

Unfortunately, our resistor is rated at only 1/8 watt, so it would be overstressed by a factor of 8 under these conditions. Normally this would result in failure or serious damage. But by making the pulsed measurement described above, we can still test the resistor at 40 volts peak to peak without overstressing the resistor. We do this by momentarily applying the 40 volt peak to peak signal then removing it and allowing the resistor to "rest" for a period of time.

If we select the "on" time and the "off" time appropriately, we can operate the resistor at the required test voltage and at the safe rating of the resistor. The ratio of "on" time to the time interval between "on" times (the "period" of the pulse interval) is known as the "duty cycle" of the test.

$$\text{Duty cycle} = t_{on} \times f$$

where t_{on} is on time in seconds and f is frequency in Hz

Referring to the above schematic, the frequency of the pulse generator is 10 Hz and the on time is 10 milliseconds or 0.010 seconds, so the duty cycle is

$$\text{Duty cycle} = t_{on} \times f = 10 \times 0.010 = 0.1$$

(or 10% if expressed as percentage)

The power dissipation of the test resistor can now be re-defined as the steady-state dissipation times the duty cycle or

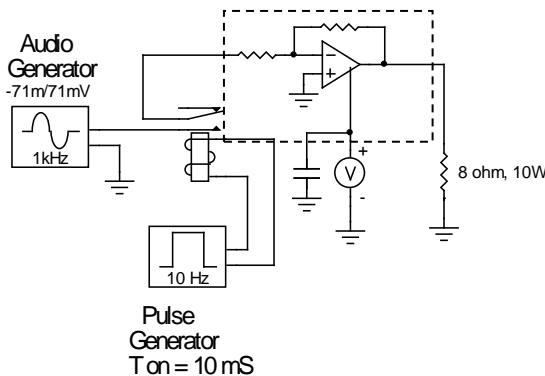
$$P_{pulse} = P_{rms} \times d$$

where P_{pulse} is the pulsed power dissipation, P_{rms} is the RMS power calculated above and d is duty cycle, substituting known values:

$$P_{pulse} = P_{rms} \times d = 1 \times 0.1 = 0.1 \text{ watts}$$

which is less than the rating of the resistor ($1/8$ watt = 0.125 watts) and the resistor is not overly stressed.

This is an important concept - we are able to drive the test device at a power level *8 times greater than the rated power*, simply by "pulsing" the test signal at a low duty cycle. The same exact procedure can be used to test other circuits, such as a guitar amplifier. Here's a typical test setup for measuring an amplifier that is being pulsed:



The amplifier under test is represented by the simple schematic within the dashed lines; it has a voltage gain of about 500. If the amplifier were driven by the audio generator in a steady-state condition (not pulsed), the 8 ohm output load would be required to absorb **80 WATTS**. The 10 watt resistor wouldn't last long at that level. But since the pulsed duty cycle is only 10%, the actual dissipation of the load is only **8 watts**. (The RMS power times the duty cycle.)

The plate power supply shown in the schematic can also benefit from the pulsed test in some cases. If we place a large capacitor across the output of the power supply (as an energy storage device) then the current limitations of the supply are improved. *The implication is that a test power supply, for example, that is not rated to drive a high-power output stage can be completely capable of driving the same stage under pulsed duty-cycle conditions.*

If the output waveform of the amplifier was observed on an oscilloscope, a similar trace display to the waveform shown at the beginning of this chapter would be observed. From this waveform, the peak to peak voltage could be measured,

converted to RMS voltage and, knowing the load resistance, the output power could be calculated.

A less accurate but still useful measurement can be made with a simple DMM by adjusting the instrument to measure RMS voltage across the load. The measured RMS voltage can then be corrected by the pulse duty cycle to obtain steady-state RMS voltage from which power output can be calculated. As an example, assume that the RMS voltage measured was 2 volts, the steady-state voltage would then be:

$$V_{pulse} = V_{rms} \times d \text{ re-arranging } V_{rms} = V_{pulse} / d$$

and substituting values

$$P_{pulse} = P_{rms} \times d = 2 / 0.10 = 20 \text{ volts RMS}$$

and the power can be calculated from

$$P = E^2 / R = (20)^2 / 8 = 50 \text{ watts}$$

A more accurate estimate can be obtained by first making a brief steady-state measurement then comparing it with the measurement made under pulsed conditions. This procedure can provide a correction factor that, when applied, can provide accuracy of pulse measurements within a few percent or less.

20.6 Virtual Signal Generator

In some cases, where a signal generator is unavailable for some reason, it is possible to use a computer to provide a test signal. A sound card (D/A and A/D converter) is necessary with appropriate software. As mentioned in earlier chapters, I use an open source sound editing program called "Audacity", available at no cost from the internet.

This is a powerful editor and special effect processor and can be used as a signal generator within the limitations of the computer sound card. There are many different software packages that provide the same or more advanced features; I'm not familiar with them so the features and measurements discussed here are in the context of "Audacity".

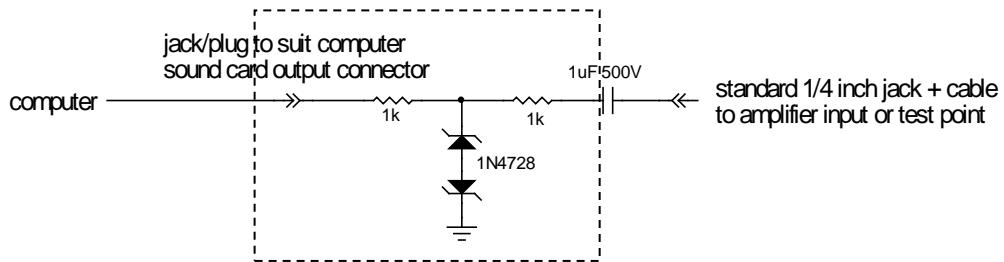
"Audacity" can provide fixed and swept audible sinusoidal frequencies at variable levels up to 1 volt, peak to peak, as measured at the headphone output of my laptop computer. This is more than adequate to drive any guitar amplifier to full rated output power. Other waveforms and options are available from the software.

As an example, a convenient signal generator feature is the "chirp" generator. This allows swept measurements of the amplifier to be performed by setting the minimum and maximum frequencies to be measured and a sweep time. Once enabled, the computer soundcard output will sweep the signal from the frequency limits specified in the time specified and at the voltages specified. If a DMM, for example, is connected across the amplifier output while connected to a "dummy load", a good indication of frequency response can be obtained.

The sweep time obviously has to be set low enough so that the voltage measured on the DMM can be monitored accurately. (A better representation of the frequency response could obviously be obtained by monitoring the output voltage with an oscilloscope that is synchronized with the sweep time. In this case, the sweep time can be much faster and therefore easier to observe in real time.)

If higher signal voltages are required, an external transformer, like the one discussed in the earlier chapter on signal generators, can be used. Output voltages up to ten volts or so can be obtained with this option, perhaps even higher depending on the transformer primary/secondary ratio and the impedance of the sound card. The software can be set to provide any voltage desired provided that it is less than the maximum output voltage.

If the soundcard on the computer is used as a signal generator, care must be exercised to prevent damage to the computer. This can occur when the computer sound output jack is connected to a load that is too low (in impedance) or from inadvertently connecting the output to a test point with voltage present. These mishaps might be avoided by constructing a simple adapter circuit such as the following one:



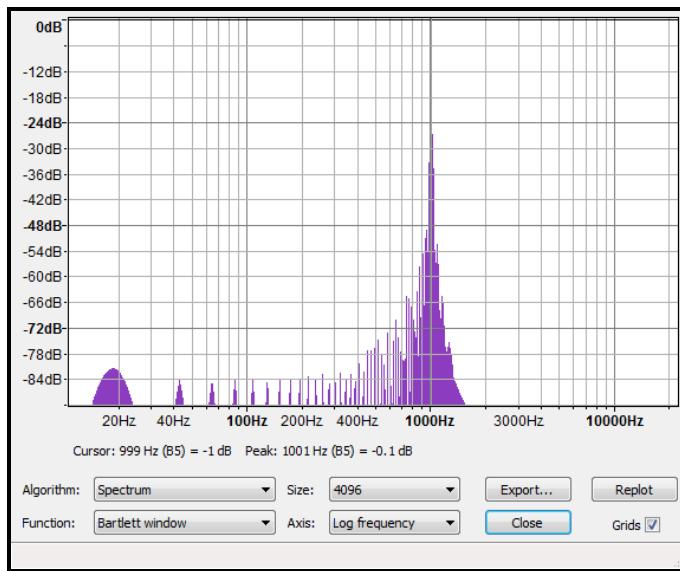
This circuit will protect the sound card from high DC voltages or AC voltages but the virtual signal generator shouldn't be intentionally applied to a test point where dangerous voltages may be present. Additionally, the virtual generator can't be used to drive low impedance loads. This will never present a problem when working on vacuum tube guitar amplifiers.

The protection circuit may generate harmonics under certain conditions which may impair measurement accuracy. Generally this won't be a problem when making relative measurements on low level signals. High power signals are best

treated with a voltage divider circuit such as the one described in the following section.

20.7 Virtual Spectrum Analyzer

Here's a spectrum analyzer display from "Audacity". The depiction is a 1 kHz tone at a magnitude of 1 volt peak to peak (corresponds to "0 dB" magnitude in the Audacity program). The very low-level "spurs" that appear randomly on the display are a result of "sampling" - the process that permits the conversion of analog signals to digital signals and processes the result:



It is very important that the input level to the computer sound card A/D converter should never exceed one volt.

The same software package used for the virtual signal generator can also be used as an audio spectrum analyzer. A protection circuit must always be used to prevent damage to the computer sound card. If a high degree of accuracy is required, a calibration procedure can be devised to determine the maximum power level indicated by the software.

This is fairly simple and requires only the measurement of the voltage across a "dummy load". Calculation of the power level (load resistance and RMS output voltage are known) and then noting the level displayed on the spectrum analyzer will provide a calibration difference that needs to be applied to future measurements.

For example, assume that a measured output power level is obtained from an amplifier producing 30 watts. Connecting the protection circuit to the output and measuring the level on the spectrum analyzer might produce a maximum signal

level of -6 dB. The -6 dB point is now representative of a power level of 30 watts and any other measurements can be determined by noting the decibel difference between the measured level and -6 dB.

One can avoid calibrating if the protection circuit is designed as a fixed voltage divider, with a simple division ratio, e.g. 100:1. The second load circuit in chapter 20.4 is an illustration of a fixed voltage divider that can accommodate large input power levels without exceeding the 1 volt maximum output level.

Note that spectral measurements are universally about ratios, rather than absolute power levels, so decibel units are perfectly suitable. A typical measurement might be the determination of harmonic distortion. A measurement of the amplifier output has indicated that the second harmonic is 20 dB below the desired output signal and the third harmonic is 30 dB below the signal.

Making an adjustment to the bias conditions of the post amplifier (as an example), one wants to know whether the distortion characteristics have improved. Re-measuring the spectrum, the second harmonic is now 23 dB below the desired output and the third harmonic is 35 dB below the fundamental signal. This represents a significant performance improvement (remember that decibels are logarithmic, a change in level of 3 dB represents a difference in power of two, either one-half or twice the original level).

When making measurements like this one, note that one is actually using the software, the sound card and the computer to *record* the signal briefly and then processing the recorded signal to display a spectral "picture".

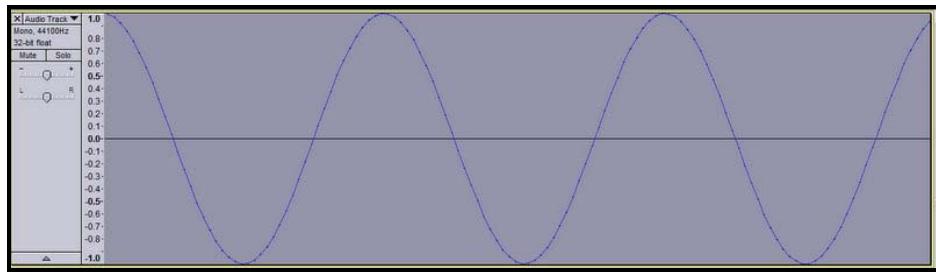
It is possible to measure noise figure using a spectrum analyzer. The validity of the measurement depends on the sound card being used, sampling rate, number of effective bits in the A/D converter, output level and a few other considerations.

A simple test for validity is to connect the output of the amplifier under test to the computer through the protection circuit and examine the spectrum. Note the "noise floor" level observed and then disconnect the test amplifier. If the displayed noise level drops significantly (say 6 dB or more) then a noise measurement made at this point is reasonably valid. If the noise floor doesn't diminish, the measurement is meaningless.

To obtain noise figure, convert the output noise level to voltage and then divide the voltage by the total voltage gain of the amplifier. The resultant input noise voltage compared to 1 microvolt (uV), in dB, is the noise figure. (This is only applicable to guitar amplifiers.)

20.8 Virtual Oscilloscope

Using the same process as described above, it's possible to display the waveform of the measured signal using the "Audacity" recording software as a virtual oscilloscope. The protective circuit must always be used between computer and the device being measured so that the sound card is not damaged. The measurement won't be in "real time" because a brief segment of the test signal needs to be "recorded". Observing the waveform of the recording will present the same display as an oscilloscope. Here's an example of an Audacity oscilloscope display, 1 kHz tone at 1 volt peak to peak:



20.9 Universal "Breadboard"

The photo below shows a useful home-built "breadboard" used for testing tubes and evaluating various circuit configurations and modifications.



The breadboard includes the following "universal" components:

Input ("RCA") phono jack

two large tube sockets, suitable for octal or 12-pin configuration

One nine-pin socket

Universal output transformer with 8 impedance configurations

Center-tapped 1:3 inter-stage transformer

Simple high voltage MOSFET adjustable internal power supply (controlled by the external potentiometer shown in the photo) for screen bias adjustments and the like

External terminals for filaments, cathodes and plate voltages

Internal terminal strips allowing up to fifty solder connections, in addition to the terminals on the tube sockets

The aluminum plate, to which the components are mounted, serves as a ground plane. It is secured to a wooden "frame" made of wood scraps from my shop. I've found this little breadboard chassis useful for many different experiments. (In fact, this small package at one time was used to breadboard the entire "Princeton" guitar amplifier circuit described in chapter 19.1, excluding speaker and tone controls.)

Although the output transformer is rated at only 15 watts, by using "pulsed testing", it is possible to make measurements up to 100 watts output power on circuits contained in the small chassis. (The pulse testing technique is described in chapter 20.5.)

21.0 Amplifier Cabinets and Chassis

Many of those wishing to build their own amplifier may be intimidated by the craft involved in building a chassis or a cabinet to house their design. I don't want to encourage anyone to exceed their personal tool manipulating abilities and especially engage in any activity that may be unsafe. I would like to point out, however, that one doesn't have to be a skilled machinist or a cabinet maker to produce an amplifier that not only performs well but looks professional.

21.1 *The Enclosure or Cabinet*

Various materials have been used to construct amplifier cabinets, speaker enclosures and combinations of the two, ranging from lighter woods (pine, alder, poplar) to manufactured materials (particle board, MDF). With the exception of enclosures that are designed to house speakers, nothing special is required other than the ability to work the material with traditional tools and to bond it with traditional or modern adhesives. Reasonable strength is necessary and, for

loudspeaker applications, a self-damping property is sometimes useful (particle board, for example).

If traditional construction is to be employed (finger-joints or box-joints), then natural wood is the preferred material. The wood must be dry and free of resin (sap) residue; poplar is very useful since it is a straight-grained material, easy to work and inexpensive. Poplar is readily available and can be assembled with any commonly-used adhesive. White glue is useful for bonding strong joints. It has reasonable open cure time, so that joints can be aligned properly before clamping.

Plywood has even been used in the past, always with cleats securing the corner joints (internally) in addition to glue. This type of construction is universally finished by the application of "contact paper" which is a form of adhesive-backed, textured material never actually made from paper. A form of inexpensive vinyl is the material of choice and the color can vary from the tweeds of early Fender and Gibson amplifiers to the darker hues most frequently used today. These materials are available from many sources. If purchased from outlets that deal in vintage amplifier parts, the material cost is usually marked-up considerably.

A trend in "boutique" or custom amplifiers is the use of more costly, attractive hardwoods finished with clear polyurethane or a heavy grade of cabinet lacquer. Amplifiers and enclosures made in this manner can be very attractive although they are more vulnerable to scuffing and scratching than vinyl material coverings.

One can obtain custom amplifier and speaker cabinet work, made from virtually any type and grade of wood available, from fine woodworking and cabinet making suppliers. Chrome or nickel-plated stamped corner protectors are available to prevent splintering of vulnerable areas, although they are not inexpensive.

Obviously the wide variety of available materials and the manner in which they can be configured is advantageous to an amplifier designer/builder who also has woodworking skills. There is quite a lot of pride engendered by the knowledge that the entire device - electronic design, assembly and test - as well as the craftsmanship required to fabricate the cabinet has been performed by one individual.

21.2 Making Finger Joints

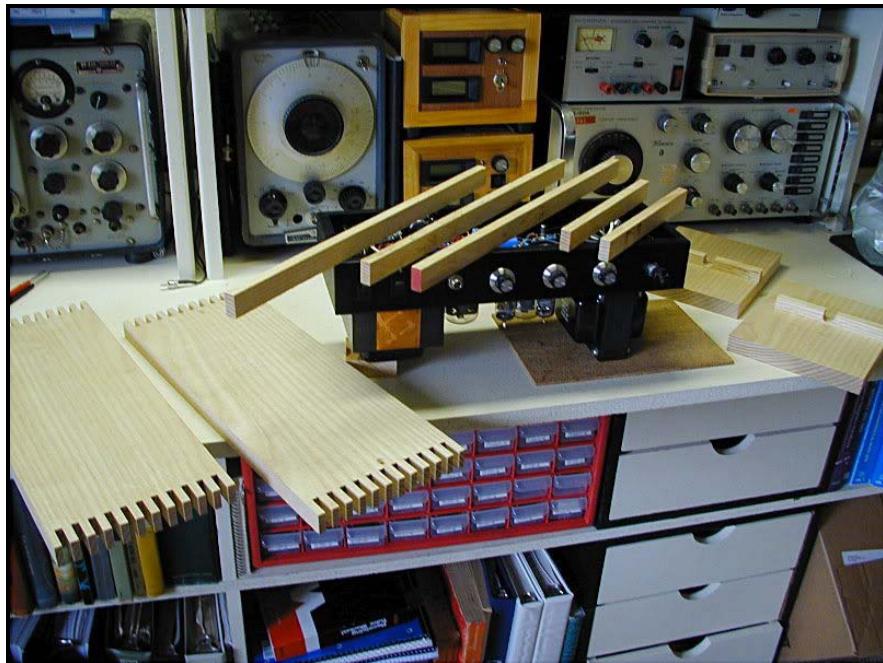
A Finger joint is the traditional joint used in making chests, toolboxes and other rectangular enclosures that are required to contain heavy weights safely while presenting an attractive appearance. The joint is named from the appearance of entwined fingers and offers superior strength due to the large amount of gluing surface (and no reliance on end-grain joints). Finger joints can be made on a

number of common woodworking tools including table saws, radial saws, routers, jig saws, bandsaws - and by hand, using a small backsaw and chisel.

The easiest method for making these joints is to first build a simple fixture for your table saw - if you have one. There are numerous plans available for these effective jigs, make a search on the internet for "box joint jig" or "finger joint jig". I haven't yet taken the time to make a finger-joint jig (although only a few hours work is required).

I usually install a 1/4 inch dado blade on a long arbor then mount the arbor in a vertical milling machine. The milling machine has the capability of indexing to great accuracy, so no jig is required. I keep telling myself, however, that the next time I do this, however, I WILL make a jig - they are considerably faster and worth the time to build.

Here's a photo depicting finger joints cut on the top and bottom elements of an amplifier cabinet. To the right are the side plates that have not yet been finger-jointed:



After all joints have been machined, the enclosure is assembled, two pieces at a time. Glue is brushed into the finger joints and the two parts are tapped together using a soft-faced mallet. The pieces are clamped into position while the glue is curing after first assuring that they are square to one another by checking with a carpenter's or machinist's square.

Shown in the photo above are the top, bottom and side plates plus four strips that will be used to "picture frame" the enclosure front after it is completed. (This

covers any unattractive joinery that might be visible from the front of the cabinet.) The following photograph is of the complete assembly, after glue has cured. The cabinet has been sanded and wiped clean and is ready for finishing.



Note that to maintain the clean lines of the design, this cabinet has hand holes routed in the side instead of a standard handle secured to the top. (The hand holes also permit the circulation of cooling air within the cabinet.) The lower front panel of the enclosure is a small sheet of 1/4 inch mahogany plywood - the type is not critical.

This panel will normally not be visible, it will be hidden by textured cloth, for example the same material used on the speaker cabinet grill. The cloth is folded over and stapled to the back of a thin sheet of plywood or fiberboard - cut slightly smaller than the recessed opening - and then forced tightly into the recess against the plywood backing. (In the photos below, a piece of black felt was used for this purpose since the speaker enclosure had not yet been built and no grill cloth had been selected.)

The photo depicts the completed cabinet with amplifier chassis installed. The custom amplifier is positioned on top of a Fender "Bassman" to give a relative idea of size with respect to performance - the amplifier is comparable to the "Bassman" in performance and specifications. This amplifier has additional features such as 4 ohm and 8 ohm selectable output impedance and, as mentioned in an earlier chapter, screen grid bias adjustment of the output tubes. (A control is provided on the rear panel of the amplifier for this purpose).



21.3 The Amplifier Chassis

Many people avoid working with metal, lacking the specialized tools and perhaps some rudimentary knowledge of the craft. It's not necessary to be a machinist to fabricate the minor amount of metal parts that go into a guitar amplifier; basically all that's needed is a prefabricated sheet metal chassis. Blank chassis boxes of various dimensions are available from any large electronics supplier. The preferred material is steel from the reasonable assumption that a certain amount of magnetic shielding is provided from the ferrous (magnetic) material.

Most of the work required to modify a blank chassis for use as an amplifier can be performed on a small bench drill press equipped with a few inexpensive hole-saws, several sheet-metal drills and a normal set of twist drills. Here's a photo of a hole-saw being used to produce holes for tube sockets on an amplifier chassis:



The drill press, when used for metalworking, needs to be adjusted to turn at the lowest speed (typically about 1/10 - or less - the revolutions one would use for machining wood). The cutting edge of the tool, whether it be a hole-saw or a drill, should be occasionally brushed with a lubricant to cool the cutting edges and make metal removal more efficient. (A useful lubricant, obtainable at any hardware store, is pipe-threading oil. Any oil, however, is preferred over "dry" cutting - even cooking oil.)

CAUTION: There is a very real danger, when machining sheet metal, that the cutting tool will "catch" in the metal, spinning the workpiece and causing serious injury. The workpiece must ALWAYS be clamped securely to prevent this occurrence as shown in the previous photograph. Many say that the simple drill press is the most dangerous of all power tools. The reason for this statement is that there is a tendency to treat the tool casually, without properly securing the workpiece.

The large cut-out at the left of the chassis is for the power supply transformer. I performed this operation on a vertical milling machine using a small "end mill" but the operation can also be done manually. A common technique for removing large amounts of metal from a chassis is to first drill a series of holes around the outline of the desired cutout and then use a saw, a "nibbler" or a file to cut, straighten and smooth the edges. (A "nibbler" is an inexpensive tool that is very useful for working sheet metal. It is hand-operated and, as the name suggests, "nibbles" away small amounts of material until the desired configuration is obtained.)

It is worth noting that metalworking with hand tools is always safer than metalworking with power tools.

Many hobbyists have metalworking shops in their garage; perhaps one of these hobbyists might be an acquaintance. Home machinists *welcome* the opportunity to use their tools to do "real work". Once you have determined where all of the parts need to be placed on/in the chassis, make a sketch of the placement and provide the sketch, the parts and the chassis to the machinist-friend. Generally, the work required will cost no more than a couple of beers (NOT consumed until after the machine work has been completed).

People willing to help with metalworking can be located fairly easily - perhaps with a "machinist's help needed" post on your employer's bulletin board or a search on the internet for a local metalworker hobbyist group. Another possibility might be a local trade school or high school if the institution has a metalworking program. As a last resort, the project could be turned over to a local machine shop but be prepared to pay a premium for their services - shop rates will be at least equal to the average auto repair shop.

Here's a photo of a finish-machined chassis, before the burrs have been removed and black touch-up paint applied to exposed metal surfaces:



All of the work performed on this chassis can be accomplished in an afternoon with a drill press, the tools mentioned above and a "nibbler". It's even possible to do this work with a hand drill (drilling a sequence of holes that are almost tangent to one another, breaking out the waste material and then filing edges smooth) and I have often done so, when more suitable tools were lacking.

Don't forget, when layout out the chassis for machining, to provide a few extra small holes for the purpose of mounting internal terminal strips. Although the tube sockets in an amplifier provide convenient "tie points" for interconnections, there is always need for additional connection points, especially on a new, unproven design.

22.0 Using Spreadsheets, from Concept to Design

Beginning with the initial concept of the amplifier application, this chapter will review previous work and suggest how to streamline the process by using some application-specific spreadsheets. It will take longer to read this chapter than it requires to implement a trial design, including reviewing past work. We hope to compress some of the required design time with these application-specific spreadsheets.

22.1 The Starting Point

A designer needs *some* specified parameter(s), fixed reference points from which the design may commence. It's not possible to design in a vacuum (pun intended) because all parameters, all selection of parts, cost, schedule and so forth would then be open-ended. It might be useful to review chapter 5.0 before proceeding any further.

The current chapter will focus on the use of design expedients - means of employing tools and published data to shorten the front-end design process. The presumption regarding usage of these tools (mainly the spreadsheets referenced in chapter 6.0) is that the designer has an understanding of the manual design procedure, the calculations, iterations, decisions that must be made. Experience in this process develops intuition too - good designers frequently rely on intuition provided that it is backed by experience.

If only a single specification is available, other parameters will soon follow, as described in chapter 5.0. Most amplifier designs start with the determination of a sound-reinforcement requirement, which may be as simple as evaluating the typical performance venue (e.g. area, seating capacity, musical genre, size of performing group) and estimating the sound pressure level required to fulfill the requirement. *As I've mentioned, the selection of a loudspeaker may be the most important consideration in the process.*

The SPL of the loudspeaker is a measure of its efficiency, related to frequency. We've discussed the frequency needs to reproduce the guitar, adding the second and third harmonics which influence the "tone" of the instrument. Generally, the frequency range 80 to 4,000 Hz is adequate to reproduce the tonal requirements of jazz guitarists. (The construction of guitar + pickup and the ability of both to produce tonal variation is the limiting factor here - it's unnecessary to design an amplifier that has superior performance than the instrument that the amplifier is intended to amplify.)

A means of evaluating the "sound" of various loudspeakers with measured sound pressure levels has been described in chapter 17.2. That's a useful means of comparing speakers but, of course, the comparison is valid only to the listener and the designer ... it might be useful to consider the audience as well.

A thoughtful designer *might* take into account all of the above-noted factors and maybe even the audience's ability to "appreciate" the frequency response of the music to be played. That's rarely the case, though, and would result in a "one-trick-pony" amplifier. It's sufficient to include adequate variation in amplifier EQ to accommodate combinations of guitars and musical genre. Discussions in 4.2.2 and also in chapter 10.0 covered some of these issues.

Once the loudspeaker has been selected, the output power level of the amplifier can be estimated. Other specifications can be ascertained from this "stake in the ground". This is described in chapters 4.0 and 5.0. For our example design, let's assume that we want a power level equivalent to a Fender "Princeton", or something around 10 to 15 watts (we'll use 15 watts as a design goal).

22.2 Power Supply Comments

One of the most expensive circuits in the amplifier is the power supply, due to the transformer cost. It's frequently useful to determine what voltages and currents can be made available to the output stage as a function of cost. This was mentioned in an earlier chapter (16.0), where I described the purchase of a low-cost transformer, around which a power supply was designed and the output stage tube performance optimized for the power supply voltage/current.

Some intuition/experience is useful here - simple observation of amplifier schematics reveals that higher output power amplifiers require higher plate voltages. This has a lot to do with increasing the efficiency of vacuum tubes, recall that earlier estimates of output power, plate-to-plate resistance and so forth, required the knowledge of the parameter E_0 , rather than E_b . A brief review: E_b is the power supply voltage available to the tube plate while E_0 is the maximum peak-to-peak voltage swing of the plate. E_0 is simply E_b minus the perveance potential of the particular tube, this value can be approximated at around 50 volts.

So 50 volts of the power supply can be considered to be "wasted", as no useful work can be obtained from the low end of the power supply voltage. Generally, the approximations in other sections of this book take that into account, as does the spreadsheets that address output stage. Some thoughts regarding trade-offs between output power levels and power supply voltages can quickly reveal a first approximation for power supply voltage and we can use the power supply spreadsheet ("Power Supply") to make some trade-offs and design decisions.

Here's the first "intuitive" decision that a designer needs to make: Class AB1 amplifiers (99% of vacuum tube amplifiers with output power levels over a few watts) **have an efficiency of about 50%**. That is, the output power produced by a pair of tubes in the normal push-pull configuration will require twice the available power from the supply.

So if we have determined that we desire an amplifier with the capability of 15 watts, then we'll need a high voltage power supply capable of supplying about 30 watts. (Actually, just a bit more than 30 watts, since the preamplifier/post-amplifier/phase-splitter tubes must also be supplied. Note that this does not include the filament voltage requirements.)

In chapter 16.1, the tradeoffs between power supply voltage and critical tube performance parameters (e.g. output power) were discussed. The parameter **I_{max}** and the relationship to screen grid 2 bias (**E_{c2}**) was noted as being of particular importance. Keeping in mind that this very important beam power tube characteristic, I_{max}, can be adjusted (within limits) by screen grid 2 bias, then we have some freedom to trade off power supply voltages and the I_{max} parameter of the output tubes in order to achieve a reasonable compromise between output power and power supply *transformer* cost.

Recently, I've become aware of some low-cost, low-voltage transformers, mainly intended for use in solid-state circuits. With some manipulation, I've decided that it's possible to use a pair of these small, inexpensive transformers to construct a useful power supply, capable of providing enough voltage/current capability for a more powerful amplifier, such as one in the 30 watt range. (The same power supply would be perfectly appropriate for the "Princeton" level amplifier mentioned previously.)

I'll discuss the use of these transformers and a developed power supply in chapter 26.0, an Appendix. The configuration is more complex than the ones described on the power supply spreadsheet and therefore inappropriate for this discussion. The spreadsheet can be used for the more complex configuration with some modification to the procedure, which will be described. (The low cost of the supply is always of interest to the home-builder and experimenter. The configuration would also be a good basis for a medium power, adjustable laboratory type supply if some precision regulation is added.)

22.2.1 Using the Power Supply Spreadsheet

R.C. Dec 2009

ESTIMATES FOR TWO HIGH VOLTAGE, FULL-WAVE POWER SUPPLY CONFIGURATIONS

case 1

case 2

diode voltage rating:	0	volts, min
diode current rating:	1.00	amperes
capacitor voltage rating:	#NUM!	volts, min
resistor power rating:	2.0	watts

diode voltage rating:	0	volts, min
diode current rating:	1.00	amperes
capacitor voltage rating:	#NUM!	volts, min
resistor power rating:	2.0	watts

NOTE: only the items in **BLUE** text are user inputs. The remaining values are calculated by the spreadsheet. Entering values in any block except **BLUE** text blocks may invalidate spreadsheet.

select a transformer		voltage & current reqd		estimate the input filter values		calculated parameters						Secondary Voltage Too Low ?	Transformer Secondary Rating Adequate ?	Fuse Value, Excluding Filaments
Transformer Primary AC Voltage	Transformer Secondary AC Voltage	Transformer Rating VA	Required DC Voltage Emax	Required Current Imax, A	Current Limiter, ohms R	Filter Capacitor, uF C	Equivalent Load Resistance	Estimated Transformer Resistance	Resistor and Trans Winding Voltage Drop	DC Voltage	Ripple, %			
case 1	1	1	1	1	1.000	1	1	1	0	#NUM!	#NUM!	#NUM!	No	0.625
case 2	1	1	1	1	1.000	1	1	1	1	#NUM!	#NUM!	#NUM!	No	1.250

This is the appearance of the power supply spreadsheet before any entries have been made. Note that two different rectifier configurations, as discussed in the power supply chapter, are included - case 1 and case 2. Also included on the spread sheet (*but not shown here*) are a number of candidate transformers of various capabilities and costs. One can select those parameters that seem appropriate for the amplifier design and insert them into the spreadsheet.

Please note that only the "blue text" boxes of the spread sheet are allowable user entries. Including values in ANY other box of the sheet may invalidate the sheet completely. It's always recommended to re-name the spreadsheet with your specific circuit need (e.g. "my_15W_amplifier") so that the original file is never altered.

Let's start the process by using our example 15 watt amplifier. From the above comments on Class AB1 amplifiers, we know that we actually require a power supply that provides 30 watts of plate dissipation. It must also provide enough filament current for all of the tubes, unless a separate filament transformer is to be used (which would be my personal choice). Looking through the listing of transformers, an obvious choice might be one with the following characteristics:

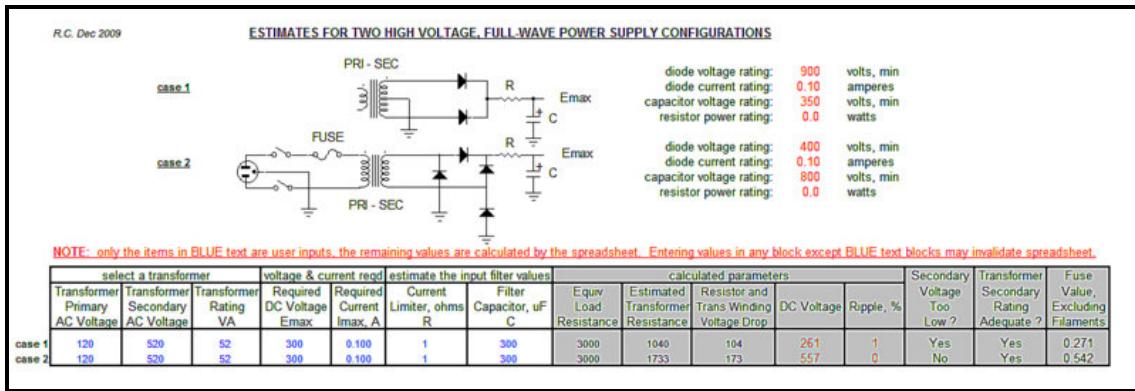
Primary is 120 VAC, secondary is 260 - 0 - 260 VAC @ 100 mA

The secondary rating means that 260 volts is available from each leg of the center tapped (that's the indication of "0" between the two "260" volt legs) output winding. We could select case 1 in the spread sheet, and enter the input voltage, the secondary voltage and the VA rating. The input is 120 volts, the secondary needs to be 2×260 volts or 520 volts (remember that each leg is 260 volts, so there is 520 volts potential difference between them), insert this value into the secondary block of the spreadsheet. The VA rating is simply the secondary voltage times the secondary current or $520 \times 100 \text{ mA} = 52 \text{ VA}$, add this to the appropriate block.

Although not exactly equivalent, the VA rating of the transformer and power output of the amplifier, measured in watts, are comparable for evaluating whether the transformer is appropriate for the application. In this case, the 15 watt amplifier output power suggests a power supply with the capability of 30 watts. The 52 VA rating (roughly equal to 52 watts) is more than adequate for our need, (although we'll also need to take into account the VA rating required by the tube filaments, thus far ignored).

We can also insert the values of the transformer into the "case 2" section of the spreadsheet, as a comparison (which will be dramatic, as we'll see) and in the event that we may have required a higher voltage supply. We don't use the center tap of the secondary in case 2 (in practice, the lead would be clipped and insulated with electrical tape).

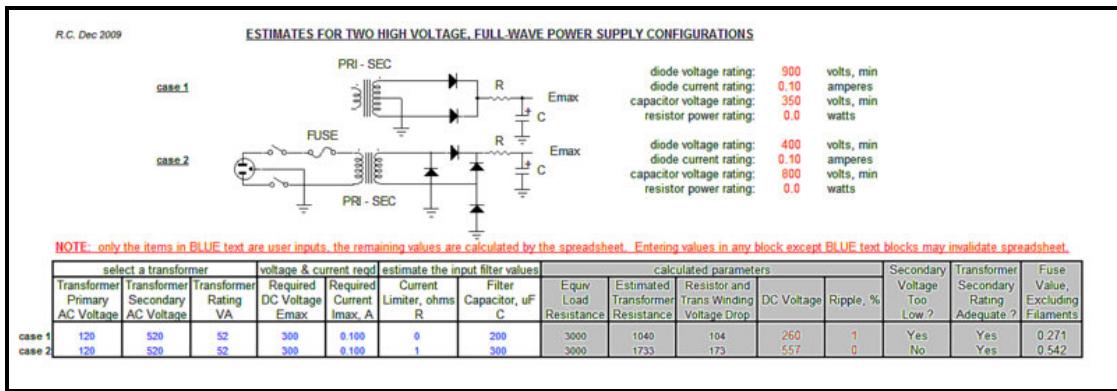
We can add some place-holder values in the blocks for the current limiting resistor and the filter capacitor as well as the desired output voltage and current. At this point, these values don't have to be meaningful since we're just beginning the process of optimizing the power supply. Here's the spreadsheet after adding the above values and some place-holders:



At this point, we're interested only in the D.C. output voltage and the ripple for case 1 and case 2. We can fairly quickly rule out case 2 as a practical power supply since the D.C. output is 557 volts - well beyond what most components we'd like to use would be rated. Looking at case 1, we see that the predicted output voltage is 261 volts and the ripple is 1%. That's a very good start for our power supply, in fact let's delete the current limiting resistor completely by setting the current limiter value, R, to "0", while examining the spreadsheet, it's interesting to vary the value of the filter capacitor, C, while watching the effect on output voltage and ripple%.

You'll see that lowering the capacitance of the filter also lowers the D.C. voltage while increasing the ripple voltage, neither of which is desirable. As the capacitor value is increased, the ripple diminishes but, beyond a certain point, the D.C. voltage DOESN'T increase. There's a point of diminishing returns, and obviously quickly reached. A good compromise would be a capacitor value that corresponds to ripple percentage of 1% or less. A value of 200 uF is appropriate, we will find.

Why do we perform this exercise? Because the cost of the filter capacitor is related to capacitance and voltage and it is an expensive component. We attempt to find a compromise of cost versus function. here's the new spreadsheet display:



If we look at some of the information in the upper right corner, the critical component *selection* criteria are denoted. We're interested only in case 1 and we can determine that the two rectifier diodes required need to be rated at 900 volts minimum and 0.1 amperes minimum. The filter capacitor, 200 uF minimum, needs to be rated at least 350 volts, as shown. This is our first iteration of the power supply design and, to me, it looks pretty good, so we'll move on to the next step.

Note that there is other useful information provided on the power supply spread sheet. I didn't reproduce the information here but it does supplement other discussion on power supply design covered in chapter 12.0. There are technical notes and suggestions regarding the information depicted on the spreadsheet.

Also, much of the calculations performed in the spreadsheet are estimates/approximations. There are many component variables in a power supply and it's pointless attempting to make exact calculations of inexact parts. For the most part, any deviation between prediction and requirement can be resolved at the "breadboard" stage of amplifier development.

22.3 The Output Stage

Chapter 8.0 describes the design of this stage, now we'll attempt to enhance the efficiency of the (frequently laborious) process by using one of the spreadsheets entitled "P-P Output" (Push-Pull Output). This is what the spreadsheet looks like with some "place-holder" information included:

Some beam power tube approximations for push-pull Class AB1 amplifiers			
tube type	tbd	% of Class "A" current:	20%
R speaker	1	Iq approx =	0.000 both tubes
E _b	1	I max sig =	0.001 both tubes
max available supply voltage		E _o =	1 volts
turns actual	1.00	max supply voltage - pereance drop	I _{max} = 0.001 amps
R p-p est	840000	E _{c1, V_{g1}} =	-1.0 volts
Vin(max.)	2 volts, p-p	E _{c2, V_{g2}} =	1 volts
turns est	916.515	Distortion can't be predicted from the information contained in this table	
Av =	1.0	P actual =	0.0 watts
excludes output transformer			

We can immediately add some input to the "blue text" blocks, the only blocks with user-required information. Do not modify or add anything to any other part of the spreadsheet except for the blocks with "blue text" except as noted later in one specific spreadsheet dealing with parts selection.

Since we know that we require 15 watts from TWO output tubes and we know that the efficiency will be 50%, then each tube has to have a plate dissipation of

15 watts, minimum. Let's assume that we want to drive an 8 ohm speaker, so we can add that information. We now know our power supply voltage, Eb, which is 260 volts and we know that the maximum current our power supply can provide is about 100 mA.

We also know because of the "perveance" of vacuum tubes, that most beam power tubes are not useful for the first 50 volts of their plate voltage supply. So we can establish a first estimate for Eo (the peak available supply voltage) by simply subtracting 50 volts from Eb. After inserting these values, our spreadsheet looks like this:

Some beam power tube approximations for push-pull Class AB1 amplifiers			
tube type	tbd	% of Class "A" current:	20%
R speaker =	8	Iq approx =	0.000
Eb =	260	Eo =	210
max available supply voltage		volts	amps
turns actual=	1.00	Ec1, Vg1 =	-1.0
R p-p est =	840000	volts	Ec2, Vg2 = 1
turns est = 324.037		ohms	volts
Vin(max) = 2 volts, p-p		P max = 0.1	watts
Distortion can't be predicted from the information contained in this table		P actual = 0.0	watts
Av = 260.0 excludes output transformer			

Now comes the iterative part of the power stage design. We need to find the right tube and the right output transformer that can produce 15 watts of output power given the supply voltage and current restraints that we have to work with.

On the spreadsheet is a number of commercial output transformers, their turns ratio, power rating and cost. Let's start this very iterative process by picking a 15 watt output power transformer (greater than 15 watts is OK but more costly and heavier), inserting the turns ratio into the input block noted "turns actual". One must develop a little experience to shorten up the optimization process.

That is what the spreadsheet is intended to do: provide a quick means of evaluating various parts without going through laborious calculations, as we did in chapter 8.0. In general, lower plate voltages require low output transformer turns ratio. My intuition suggests that a starting choice might be this one, selected from the list of available transformers on the spreadsheet:

p/n 1615, 15 watts, turns ratios of 17.7, 25.0 and 35.4 at a cost of about \$40

Let's insert the turns ratio of 17.7 into the appropriate block and then start the iteration process by varying the parameter "I/max" in the spreadsheet. The purpose of this exercise is to narrow down the selection of tubes that may be appropriate for our purpose. We can start with a value of say, .05 amperes,

increasing the value a bit at a time while observing "Pmax" and "Pactual". Pmax is the maximum output power the tubes are capable of delivering while actual is the power the tubes can deliver using the transformer that we selected.

After obtaining the best impedance match, using this technique, which means that the tube impedance, as modified by the output transformer, is equal to the load impedance, we end up with this:

Some beam power tube approximations for push-pull Class AB1 amplifiers			
tube type	tbd	% of Class "A" current:	20%
R speaker =	8	Iq approx =	0.128 both tubes
Eb =	260	I max sig =	0.213 both tubes
max available supply voltage		Eo =	210 volts
turns actual=	17.70	Imax =	0.335 amps
		Ec1, Vg1 =	-1.0 volts
		Ec2, Vg2 =	1 volts
R p-p est =	2506	R p-p actual =	2506 ohms
turns est =	17.700	P max =	28.2 watts
Distortion can't be predicted from the information contained in this table		P actual =	28.2 watts
Vin(max) =	2 volts, p-p	Av =	260.0 excludes output transformer

This looks impressive, the predicted output power is 28.2 watts, instead of the 15 watts we're trying to obtain! There are some problems, however... Look at the predictions of plate current, at the top of the spreadsheet. The two parameters "Iq approx" and "I max sig" both exceed our allowable power supply current of 100 mA. (Iq is the quiescent current and I max sig is the current at the maximum output power.)

This is a useful exercise that indicates the tradeoffs between low voltage and high voltage power supplies and the output transformer turns ratio. The value of the spreadsheet for evaluating "what if?" scenarios will be VERY important in the next exercise.

Recall that the same transformer selected has three different turns ratios (available from different "tap points" on the secondary), let's replace the turns ratio of 17.7 with the next higher one, which happens to be 25, then perform the same exercise as above (varying the I max current, starting at around .05 amperes):

Some beam power tube approximations for push-pull Class AB1 amplifiers					
tube type	tbd	% of Class "A" current:	20%	Max diss =	15.0 watts ea
R speaker	8	Iq approx =	0.064	both tubes	Static diss = 8.3 watts ea
Eb	260	Eo =	210	volts	Imax = 0.168 amps
max available supply voltage		max supply voltage - perveance drop			
turns actual=	25.00	Ec1, Vg1 =	-1.0	volts	Ec2, Vg2 = 1 volts
R p-p est =	5000	R p-p actual =	5000	ohms	P max = 14.1 watts
turns est =	25.000	Distortion can't be predicted from the information contained in this table			P actual = 14.1 watts
Vin(max)	= 2 volts, p-p	Av =	260.0	excludes output transformer	

This is a nice scenario, the maximum current is now 107 mA, there is a good output match and the estimated output power of 14.1 watts is very near the required 15 watts. At this point, if the designer required safety margin and the output power specification was REALLY important, the power supply spreadsheet could be revisited and a slightly higher current transformer substituted. Repeating the brief power supply optimization (a couple of minutes work with the spreadsheet) would give the designer more current with which to work.

Assuming that the above scenario is satisfactory, we'll need to fill in the rest of the user-defined data in the spread sheet, which are the grid 1 voltage, Ec1 and the grid 2 voltage, Ec2. Both of these must be obtained from the plate curves of the beam power tube that is selected. These, as we know from previous chapters, are available on the internet for various tube types.

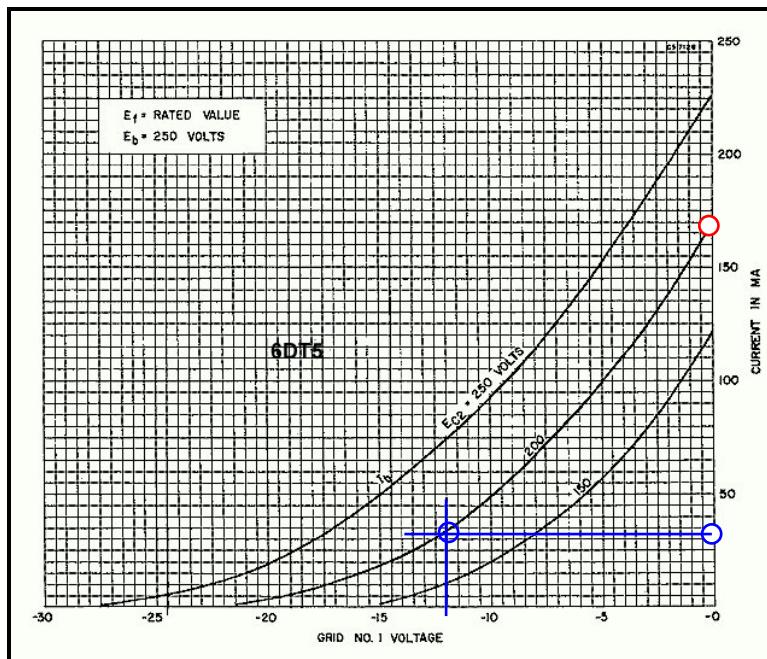
All that's required to finalize a selection is to find a tube that meets our cost and availability goals and can be adjusted to the I_{max} value that we optimized above (168 mA). That's fairly simple - any tube with I_{max} GREATER than 168 mA is acceptable because we can adjust the screen grid bias voltage, Ec2, to produce a *lower* value.

As a matter of interest, taking a look at the predicted power dissipation of the output tubes, both static dissipation and full drive dissipations are less than 9 watts. We can relax our power dissipation requirement at this point and safely use tubes that are rated at greater than 9 watts.

The spreadsheet package also includes a sheet that lists useful parts (other than transformers) for amplifier applications. It's intended that the user add to this sheet by including part numbers, cost, important specifications and recommended vendor. Referring to this sheet now, I find various tubes that can readily meet our requirements so I'll select an output tube based on price and availability: 6DT5, available NOS at a price of \$1.60 each, rated at 9.0 watts each and with I_{max} of 200 mA.

We'll add in the part number, power rating and then take a look at the tube data sheet to get the rest of the required information. At the right vertical axis of this curve ($E_{c1} = 0$ volts), locate plate current = 168 mA and estimate the screen voltage, E_{c2} . The screen bias voltage is 200 volts, add this to the spreadsheet. Now, referring to the spreadsheet, find I_q , which is about 64 mA, *divide this by two since the spreadsheet value is for both tubes*.

Locate a point that represents 32 mA on the plate current axis and draw a horizontal line that intersects the screen grid bias voltage, E_{c2} , of 200 volts. Strike a vertical line from that intersection downward to locate grid 1, E_{c1} , voltage of about -12 volts.



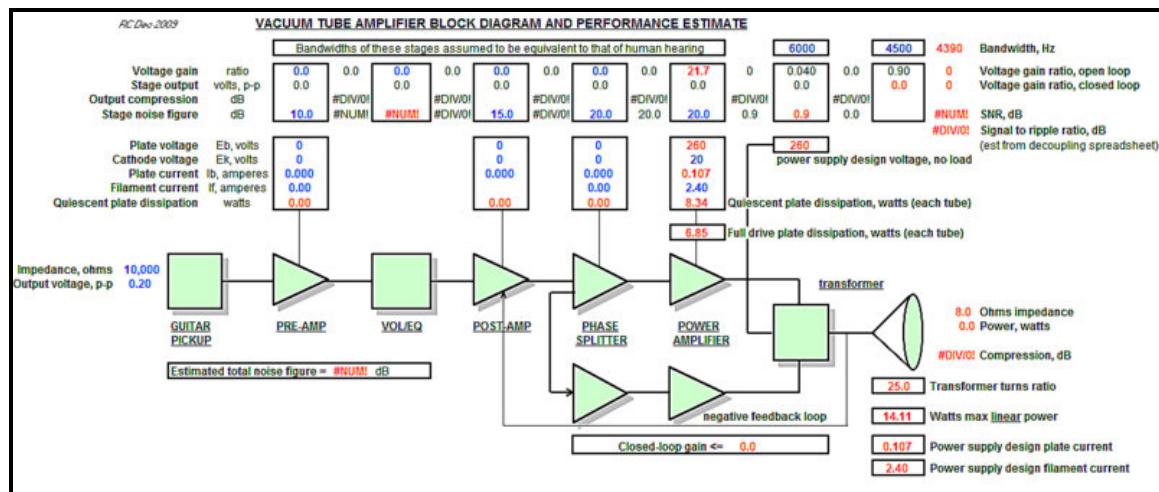
Enter all of the above information on the spreadsheet, it should now look like this:

Some beam power tube approximations for push-pull Class AB1 amplifiers					
tube type	6DT5	% of Class "A" current:	20%	Max diss =	9.0 watts ea
R speaker	8	I_q approx =	0.064	both tubes	Static diss = 8.3 watts ea
		$I_{max\ sig}$ =	0.107	both tubes	Drive diss = 6.8 watts ea
E_b	260	E_o =	210	volts	I_{max} = 0.168 amps
max available supply voltage		max supply voltage - pereance drop			
turns actual	25.00	E_{c1}, V_{g1} =	-12.0	volts	E_{c2}, V_{g2} = 200 volts
R_{p-p} est	5000	R_{p-p} actual	5000	ohms	P_{max} = 14.1 watts
turns est	25.000	Distortion can't be predicted from the information contained in this table			P_{actual} = 14.1 watts
$V_{in(max)}$	24 volts, p-p	A_v	21.7	excludes output transformer	

And while we're looking at the 6DT5 data sheet on the internet, let's make a note of the filament current and the screen grid current . The filament current is 1.2 amperes for each tube and the screen grid current looks to be about 5 mA per tube.

22.4 The Overall Amplifier Block Diagram

Some of the information that has been entered or calculated in the previous spread sheet has automatically been transferred to the "block diagram" spread sheet. Here's what it looks like, including some place-holders and including the filament current (times 2) for the output tubes:



Note that there are some error messages displayed on the spreadsheet, these are a result of incomplete data entry. They will disappear as we gradually add the remaining data. Also note that all of the performance characteristics of the output stage already exist on the spreadsheet, they are automatically transferred, as the information on the power stage spreadsheet is updated.

This is the point at which the remaining amplifier decisions are made. At this stage, one wonders exactly how to start the process ... the beauty of automating some of the tedious computations required to develop the performance of the entire chain is that a trial/error approach is surprisingly successful.

We can start by filling in some of the voltage gain information for the various stages. The EQ stage, as mentioned many times, will normally have a loss of around -10 db, which is a voltage gain of 0.1. The normal phase-splitter without gain will have a voltage gain of about 0.6. (We have the option of choosing a phase-splitter WITH gain and this will be discussed shortly.)

We can insert these two numbers into the appropriate "blue text" blocks. That leaves the establishment of voltage gain for the preamplifier and the post-amplifier. We'd like to distribute the gain throughout the chain so that the

estimated output power (in the block beside the loudspeaker symbol) is about the same as the block below it labeled "Watts max linear power" and there is some compression (at least three dB at full power). The amount of compression is also a decision made by the designer - the spreadsheet allows one to observe the compression in any stage in the amplifier chain. This allows the designer to distribute the compression, throughout the amplifier, in a manner that it felt to be most useful.

The amount of gain and the amount of compression is predicated on the characteristics of the input device (guitar pickup). Once reasonable performance is predicted by this spreadsheet, it's wise to change the input characteristics to reflect pickups that can be used with the amplifier. Some re-optimization can result in an amplifier that is useful for a number of input devices. It's possible to measure the output voltage of any guitar pickup with the ubiquitous digital multimeter, if one is aware of the measurement parameters and can intelligently extract useful information from this simple measurement.

Examples of parameters that need to be accounted for: pickup impedance, contribution of volume/tone controls, whether the measurement is of a single-string (recommended) or chordal, in nature, intensity of string attack and so forth. The waveform shape is not known therefore neither is the power level of the pickup output, as measured by this simple technique. Nevertheless, I recommend it - a useful sanity check against published guitar pickup specifications (which may not be accurate).

Measurement techniques and definitions vary, especially when considering the many "hand-winders" that manufacture after market pickups. The characteristics of the pickups may differ and rarely are any more sophisticated measurements made to individual pickups than the D.C. resistance, as measured by the DMM and perhaps an inductance bridge ... the manufacturer provided information may not be useful if one wants to ascertain frequency response, for example.

For now, using the spreadsheet input default values of 10,000 ohms input impedance and 0.2 volts, peak-peak, let's insert the above values and work out trial values for the front end gain stages. We need to also input some estimates for all of the plate voltages in the chain, since some of the calculations require them.

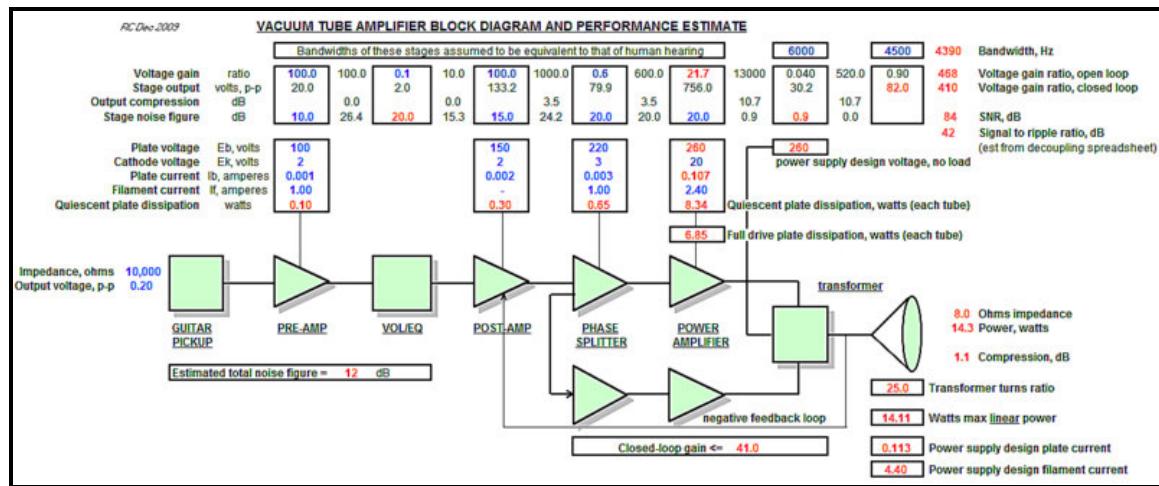
For reasons having to do with de-coupling the A.C. ripple from the power supply (discussed earlier and covered again in a moment), it's normal practice to lower plate voltage successively, from the output stage (full power supply voltage) "down" to the preamplifier stage. Recall that there is a certain minimum amount of plate voltage, beyond which these tubes cannot function properly.

Ninety volts might be a convenient minimum voltage starting point because many tube manufacturers provided tabulated component values for this voltage level.

Other "steps" in power supply voltages that are commonly specified in tabulated amplifier design data are 180 volts and 300 volts. The 90 volt and 180 volt design tables are the most useful for small signal applications (preamplifier and post-amplifier).

It's useful to note that the de-coupling of stages with low voltage gain is not as critical as de-coupling stages with high gain. The phase-splitter, for example, needn't be operated at a plate voltage too much lower than the output stage since it has a gain of 0.6. A little intuition can also be useful in assigning preliminary values to the various stages. Remember that these values are easily changed at any time - that's the whole point of the spreadsheets - optimization is a simple matter and new performance estimates are available instantly. We'll get into the power supply de-coupling in more detail shortly ...

We know that the preamplifier, post-amplifier and phase splitter tubes require only a few milliamperes of plate current and a few volts for the cathode bias, so we can fill in some estimated bias values in the blue text boxes too. If one has a pretty good idea of the preamplifier, post-amplifier and phase-splitter tubes, it's also useful to fill in the known filament currents at this point. Here's a first approximation of the chain, using estimated values:



I'd be fairly satisfied with this approximation - satisfied enough to take a look at potential "hum" problems, produced by power supply ripple. Take a look at the upper right corner of the above spreadsheet, where signal-to-noise ratios (SNR) are estimated. SNR, refers to thermal noise, discussed in other parts of the book - which ALSO can be optimized in the spreadsheet provided that one has adequate information about the noise contributions fo the various tubes.

Below that parameter is "Signal to ripple ratio" which refers to the power supply A.C. ripple, as it is processed through the amplifier chain. In the above case, the ripple ratio is fairly high (42 dB) and we'd like to lower it, most tube amplifiers might regard 60 dB ripple rejection as good performance. The ripple ratio is

calculated from the spreadsheet entitled "decoupling" and results from the calculations in that sheet. This parameter is strongly dependant upon the potential usage of the amplifier and individual opinion.

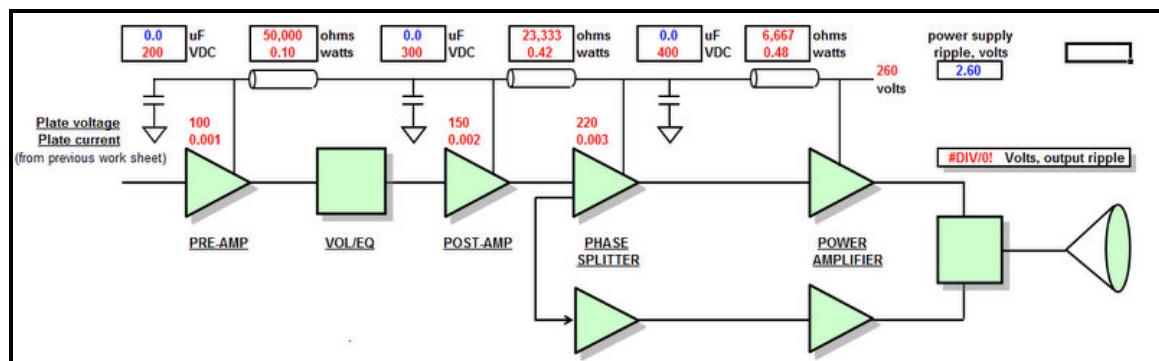
All of the performance parameters are related, as we know. That was the motivation behind creating these spreadsheets, so that the various relationships could be investigated and optimized. Let's move on to the next topic.

22.5 Decoupling Power Supply "Hum"

It may be useful to review chapters 12.6 and 12.7 in conjunction with this topic. Minimizing power supply hum involves a combination of electrical *and* mechanical precautions and practices - ignoring either will result in an unsatisfactory design.

Any power supply noise that is introduced into any stage in the complete amplifier chain will be amplified by subsequent amplifier stages, so we need to pay attention to leakage paths, particularly magnetic coupling as discussed in the chapter on power supply design. *Because the phase, amplitude and the insertion points of ripple voltages aren't predictable, estimates are subject to large uncertainties.* Assumptions made when preparing the spreadsheet may not be valid, however the sheet is useful for determining decoupling resistor and capacitor values and ratings.

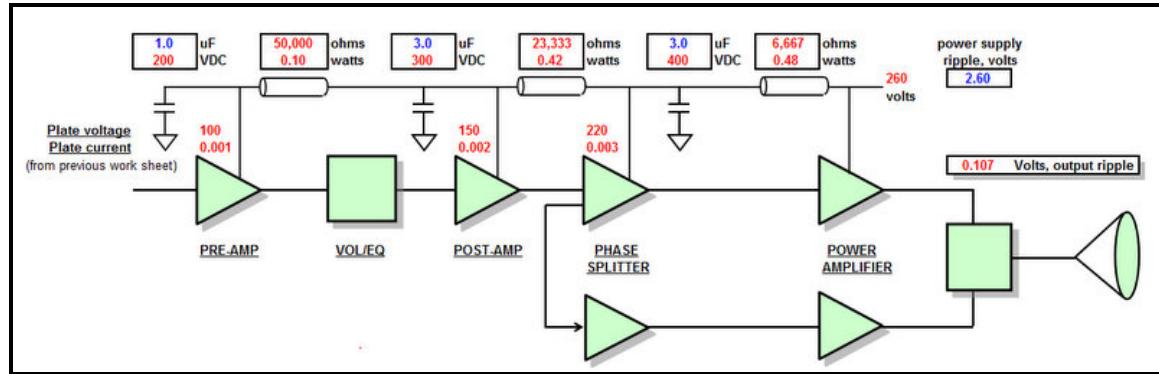
Here's a spreadsheet with some values already entered, the plate voltages and currents are automatically transferred from the block diagram spreadsheet and the output stage spreadsheet. The first entry that needs to be manually entered is the power supply ripple voltage. Referring back to the power supply spreadsheet, multiply the D.C. output voltage (260 volts in our case) by the ripple% (1% in our case) to obtain the ripple voltage or $260 \times .01 = 2.6$ volts. Here's what the spreadsheet looks like with nominal inputs:



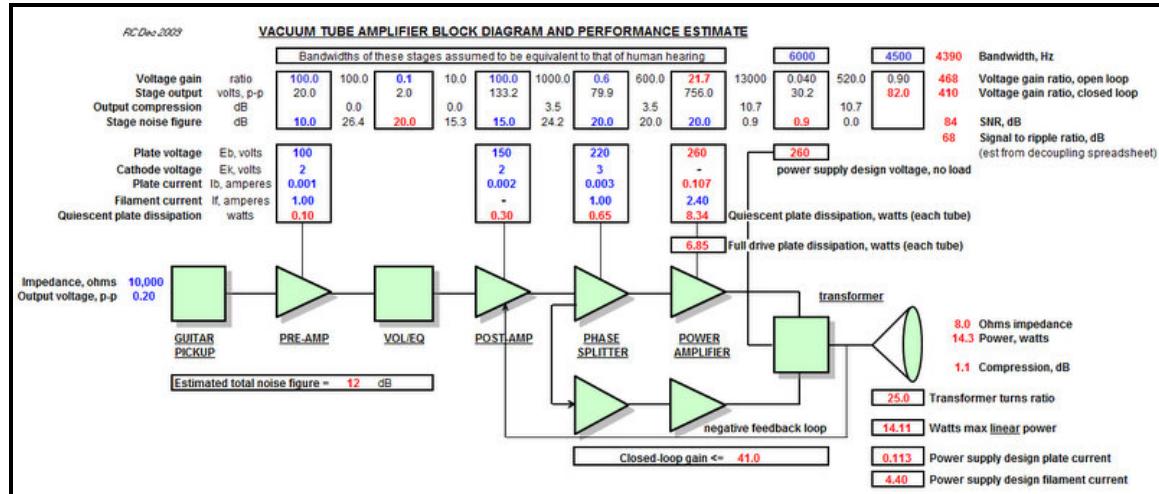
Error messages on the spreadsheet are a result of insufficient data entries. The object of this spreadsheet is to calculate the values and power ratings of the stage de-coupling resistors and allow the user to determine the value of de-

coupling capacitors. Recalling that capacitor cost is related to value and to voltage, one can optimize various capacitor values to produce the lowest output ripple voltage. A reasonable process is to increase/decrease in half-decade increments, such as 1 - 3 - 10 - 30 or 1000 - 300 -100 -30, for example.

Here's an example, after optimization. Note that every stage capacitor reaches a point of diminishing returns very quickly, this exercise takes only a minute or two.



And since this spreadsheet "reports back" to the block diagram spread sheet, we can take a look at THAT sheet and see the new estimated signal to hum ratio:



The signal to ripple (or hum) ratio is about 68 dB now and that's generally a tolerable level for a vacuum tube amplifier. These amplifiers are always noisier with a higher hum level than solid-state counterparts. Because of the difficulty in suppressing magnetic coupling, it's one of the prices we pay for a simple, inexpensive design.

(It's possible, as is commonly done in high-end vacuum tube stereo amplifiers, to reduce this level considerably. Further reduction requires more complexity, expense and more sophisticated packaging, not usually justifiable for the cost most guitarists are willing to pay for an amplifier.)

22.6 What's Left?

An old engineering adage states that "90% of the design takes 10% of the time - the remaining 10% of the design requires 90% of the time". It's not likely that those percentages will change too much by using the collection of spreadsheets previously described BUT hopefully the overall time required to implement an initial design will diminish. Decisions will still be required and the detailed design work still needs to be accomplished but some of the grinding, repetitive and boring up-front work and planning will be aided by the use of the spreadsheets.

At this point in the process, the overall design architecture has been established, the most critical stage (the output power stage, along with output transformer) has been largely determined and so has the power supply. We can extract an existing EQ circuit, from a previous chapter (or design a new one), most of what remains is determining component values for the preamplifier and post-amplifier. That too, can be as simple as reading information from a chart. Recall that we used manufacturer's published data in several parts of the discussion, in chapter 2.2, for example, to extract the values to bias a specific tube to required plate voltage and plate current.

We also used these data when designing the preamplifier, in chapter 11.0. We can replicate that procedure for our current preamplifier and post-amplifier requirements. We've defined the required voltage gain and the power supply voltages for these stages, all we need do is refer to one of the appropriate design charts for the tube selected (we'll probably use a dual triode to perform both functions). In review, here's a typical design chart, available from many sources on the internet, this one is for a 12AX7 tube:

R _p	R _s	R _{g1}	R _{total} (R _p R _s)	Eb = 90			Eb = 180			Eb = 300		
R _k	A _v	E _{o p-p}	R _k	A _v	E _{o p-p}	R _k	A _v	E _{o p-p}	R _k	A _v	E _{o p-p}	
100,000	100,000	100,000	50,000	1,700	31	14.1	1,000	40	42.4	760	43	84.8
100,000	100,000	100,000	50,000	2,000	38	19.5	1,100	46	56.6	900	50	113.1
240,000	240,000	100,000	120,000	3,500	43	18.4	2,000	54	50.9	1,600	58	104.6
240,000	510,000	100,000	163,200	3,900	49	24.3	2,300	59	67.9	1,800	64	132.9
510,000	510,000	100,000	255,000	7,100	50	20.9	4,300	62	53.7	3,100	66	110.3
510,000	1,000,000	100,000	337,748	7,800	53	25.7	4,800	64	67.9	3,600	69	130.1
240,000	240,000	10,000,000	120,000	0	37	25.2	0	53	42.4	0	62	90.5
240,000	510,000	10,000,000	163,200	0	44	15.3	0	60	53.7	0	67	115.9
510,000	510,000	10,000,000	255,000	0	44	14.1	0	61	48.1	0	69	99.0
510,000	1,000,000	10,000,000	337,748	0	49	18.1	0	66	59.4	0	71	115.9

Reviewing the chapters related to the design of these two circuits is useful but basically all that's required is to find examples in the table that most closely reflect our required supply voltages and voltage gains from the block diagram spreadsheet. This will frequently be yet another iterative process, pointing out the convenience of the spreadsheet approach to design.

Referring back to our block diagram spreadsheet, we had entered voltage gains of "100" for both the preamplifier and the post-amplifier. Looking at the above table, we note that none of the configurations will produce voltage gains, A_v, of that magnitude (the highest one shown is "71"). The entries of 100 were

intentional, so that novice designers can become aware of how to address gain distribution problems throughout the amplifier chain. (And to - yet again - be reminded that one size does NOT fit all.)

Two obvious choices are available. One is to use a different phase-splitter configuration, one with gain, making up for the inability of the first two stages to provide our selected voltage gain. Here's how we would implement that approach.

22.6.1 Revisiting the Phase-Splitter with Voltage Gain

Using the existing block diagram preamplifier and post-amplifier gains of "100", they have a combined voltage gain of:

$$\text{Av total} = \text{Av1} \times \text{Av2} \text{ and for the above example, } 100 \times 100 = 10,000$$

We used a voltage gain estimate for our phase-splitter tube of 0.6, we can multiply this voltage gain by the above combined gain to obtain the gain of all three stages, so

$$0.6 \times 10,000 = 6,000 \text{ which is what we require from the three stages combined.}$$

Looking at the tabulated design values above for the 12AX7 tube and reflecting that our plate voltage estimates for the preamplifier and post-amplifier (the estimates that we used in the block diagram spreadsheet) were 100 volts and 150 volts, respectively, let's select the closest plate voltages to those values and extract the voltage gain. The closest values would be the 90 volt values in the table and the 180 volt values in the table.

From the 90 volt columns, we find that a voltage gain of around 50 is reasonable. Likewise from the 180 volt columns, a reasonable voltage gain would be about 60. Let's use those two gain values to estimate what the desired gain of the phase-splitter needs to be. We determined that we need a total gain of 6,000 for all three stages, we can insert our two known gain values and solve for the phase splitter gain as follows:

$$6000 / (50 \times 60) = 2$$

which is the gain required by the phase-splitter if we choose that particular option. Design of the phase-splitter with gain was covered in chapter 9.2. Knowing the voltage gain required and the power supply voltage available, reviewing that chapter should result in a practical phase-splitter design. There is another possibility, however.

22.6.2 Increasing Preamplifier and Post-amplifier Gain

An easier way to achieve the performance that we predicted on the block diagram spread sheet might be to expand our selection of tube choices for the first two stages of the amplifier. It's traditional to use dual triodes for amplifier "front ends" and there was good justification when tubes were carefully manufactured and carefully tested.

At that time, lowest noise performance could generally be achieved with the triode; specific families were screened for noise and hum and marked with a different designation to differentiate them from similar tubes. This was mentioned in chapter 12.6 where certain tube types were identified in a table of tubes used for low noise applications.

Currently there's no indication that any particular tube is better than another or that any two tubes from an identical lot share identical noise characteristics, so we don't need to follow conventional wisdom. Another point is that we don't require state-of-the-art performance for our amplifier. We can examine another reasonable alternative that is cost and performance effective: beam power tubes in one of the first two stages. We could use a dual tube that includes both triode and pentode.

Why would we want to include a pentode in the first two stages? Pentodes have considerably more voltage gain than triodes. For those who prefer a more traditional triode front end a good compromise is offered by the 7199-type tube (there are many variations of this device - it's still available NOS but quite expensive).

This class of tube is readily available, frequently for lower cost than the dual triodes commonly used in amplifier front ends. Another advantage of this tube is that it has tabulated component values as a function of performance, similar to the dual triodes discussed in earlier portions of the book. Here's an example of tabulated data provided by RCA for the 7199:

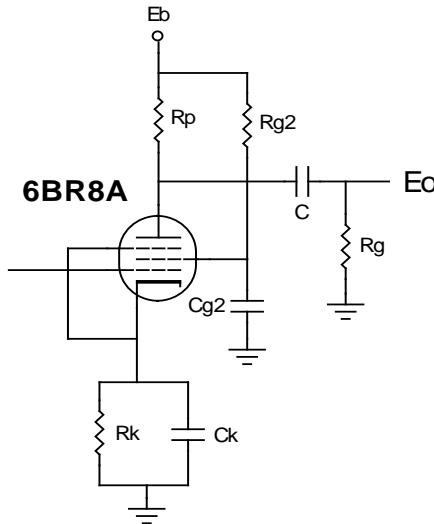
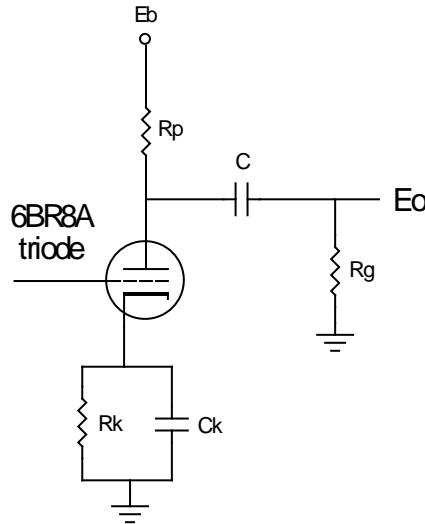
7199 (Pentode Unit)									
See Circuit Diagram 2									
E _{bb}	R _p	R _x	R _{g2}	R _k	C _{g2}	C _k	C	E _o *	V.G.
90	0.22	0.22	0.560	3700	0.046	4.50	0.0090	12	73
	0.22	0.47	0.600	3900	0.043	4.30	0.0055	17	95
	0.22	1.0	0.640	4200	0.039	4.00	0.0033	19	109
	0.47	0.47	0.870	6000	0.036	2.70	0.0046	16	95
	0.47	1.0	0.980	6700	0.044	3.00	0.0030	22	113
	0.47	2.2	1.09	6700	0.043	2.80	0.0020	25	131
180	1.0	1.0	2.00	12200	0.021	1.44	0.0028	15	119
	1.0	2.2	2.20	12800	0.024	1.74	0.0016	21	167
	0.22	0.22	0.530	1570	0.069	7.50	0.0088	32	82
	0.22	0.47	0.600	1730	0.064	7.40	0.0084	38	164
	0.22	1.0	0.650	1820	0.061	7.30	0.0084	45	190
	0.47	0.47	1.12	3200	0.053	5.30	0.0046	35	147
300	0.47	1.0	1.40	3500	0.042	5.10	0.0028	40	209
	0.47	2.2	1.57	3740	0.040	5.40	0.0019	45	250
	1.0	1.0	2.50	6500	0.039	2.80	0.0024	34	179
	1.0	2.2	3.40	7500	0.025	2.30	0.0015	39	277
	0.22	0.22	0.600	9200	0.086	11.2	0.0085	52	182
	0.22	0.47	0.670	1010	0.076	10.5	0.0052	65	236
90	0.22	1.0	0.720	1100	0.076	10.0	0.0033	77	257
	0.47	0.47	1.25	1950	0.060	7.0	0.0044	41	221
	0.47	1.0	1.43	3210	0.053	6.4	0.0027	72	296
	0.47	2.2	1.45	2200	0.055	6.3	0.0019	82	345
	1.0	1.0	3.00	4100	0.040	4.2	0.0022	57	295
	1.0	2.2	3.30	4340	0.037	3.6	0.0016	74	378

7199 (Triode Unit)									
See Circuit Diagram 1									
E _{bb}	R _p	I _{R_x}	I _{R_{g2}}	R _k	C _{g2}	C _k	C	E _o *	V.G.
90	0.047	0.047	—	1292	—	3.3	0.060	8	12
	0.047	0.1	—	1401	—	2.8	0.032	10	13
	0.047	0.22	—	1470	—	2.4	0.016	11	13
	0.10	0.1	—	2630	—	1.60	0.029	9	13
	0.10	0.22	—	3090	—	1.24	0.015	12	13
	0.10	0.47	—	3440	—	1.10	0.008	14	14
180	0.22	0.22	—	6550	—	0.70	0.015	12	12
	0.22	0.47	—	8270	—	0.51	0.0077	15	12
	0.22	1.0	—	9130	—	0.44	0.0045	18	12
	0.047	0.047	—	723	—	4.0	0.061	16	14
	0.047	0.1	—	836	—	3.5	0.032	20	14
	0.047	0.22	—	948	—	2.9	0.015	24	15
300	0.10	0.1	—	1543	—	2.0	0.031	17	14
	0.10	0.22	—	2002	—	1.5	0.016	24	14
	0.10	0.47	—	2522	—	1.2	0.0082	30	13
	0.22	0.22	—	4390	—	0.79	0.015	24	13
	0.22	0.47	—	6122	—	0.57	0.0078	33	12
	0.22	1.0	—	8060	—	0.47	0.0046	41	12

* Peak volts.

* Peak units.

Use of these tabulated data to determine component values has been discussed previously, here are schematic representations of the pentode and triode amplifier circuits with reference designations adjusted to conform to the tables. Note that all capacitor values are in microfarads (uF), resistor values are in Megohms except for cathode resistor values which are in ohms.



(Remember that this is a dual tube - both the triode and the pentode are in the same package.) The values in the tables are specific to the 7199 tube but I've also used the tabulations for the inexpensive 6BR8 tube. It's always wise to find tabulated data for the specific tube under consideration or design each section conventionally (graphically). If one has the time to breadboard the circuit, I'd suggest using the tabulated values and then optimizing the values at breadboard stage.

The above schematic suggests that the triode precede the pentode but that's not absolutely required. As will be explored next, there are reasons for exchanging the positions of the two stages, the designer must make the decision as to the best compromise. The most usual configuration of this tube, in stereo tube amplifiers, is that the pentode is used for gain and the triode follows the pentode, configured as a phase-splitter.

22.6.3 Preamplifier/Post-amplifier Gain Distribution

If it seems reasonable to use a beam pentode in the front end of the amplifier, with greatly increased associative gain, where should the pentode be placed: preamplifier or post-amplifier? The question addresses multiple performance issues that are characteristic of vacuum tube amplifiers (and others too), such as:

- Microphonic response
- Hum (A.C. ripple)
- Noise
- Compression (and distortion)

Because the interaction between these parameters is not always simple and easily predicted, an argument can be conveniently made for the dual-triode configuration (gain of both stages being approximately equal). The following truth table suggests the influence of placing the pentode in the preamplifier position or the post-amplifier position, related to overall amplifier performance.
Pentode in preamplifier stage: Microphonics \geq Hum \geq Noise \leq Compression \geq
Pentode in post-amplifier stage: Microphonics \leq Hum \leq Noise \geq Compression \leq

The symbols refer to the performance of the parameter described increasing or decreasing, depending on where the high-gain pentode is placed in the chain. For example, with the pentode in the *preamplifier stage*, overall noise of the amplifier will decrease. Note that these are general trends, individual stages designed for extremes in gain (more gain or less gain) may exhibit trends that aren't compatible with the above predictions.

High fidelity amplifiers universally place the pentode in the preamplifier position, presumably for best noise performance. Usually, the pentode is followed by a triode phase splitter - that's a nice compromise since the pentode can obtain the gain normally achieved by a dual triode and making the triode (in the same "package") available for phase-splitter duty. Unhappily, this configuration may not be ideal for a guitar amplifier, especially in the "combo" configuration with speaker and amplifier included in the same mechanical enclosure. The vibration environment is "unfriendly" as compared with a stereo amplifier.

This suggests that the pentode might be better placed in the post-amplifier position for musical amplifiers, compromising noise performance with microphonic performance. That is not a predictable compromise, usually, since

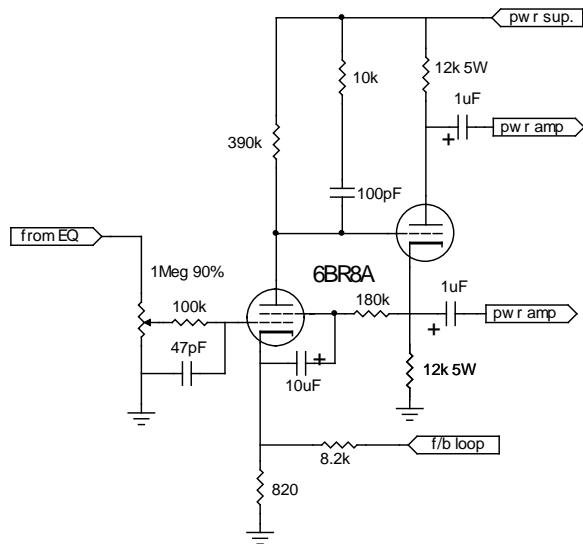
microphonic performance isn't available on a tube data sheet (with a few exceptions for old tubes). It's an unfortunate fact of engineering - even at these very low frequencies, that sometimes performance must be confirmed by prototype construction and performance measurement.

22.6.4 Deciding on Gain Distribution

We're still left with the decision as to how to distribute the gain. Noise and hum, although important, are usually influenced by the performance environment. This was described in chapters 4.1.1, 4.1.2, 5.4 and 5.4.1 in a superficial manner. We discussed the "irritant level" of noise and hum in one's living room opposed to practical performance levels - where noise and hum of the most primitive amplifier designs would be unnoticed. That's a consideration that the user - not the designer - (unless user = designer) must determine.

Microphonics are a more complex topic because the possibility of feedback exists. From that aspect alone, I'd be inclined to make the gain distribution throughout the amplifier chain fairly uniform. Observation of guitar amplifier schematics dating back fifty years seems to roughly support that inclination.

So what's the best decision for our current example: high-gain pentode front end or phase-splitter with gain? I've used both configurations successfully. For example, below is the schematic of a post-amplifier + phase-splitter using a 6BR8 tube, which is a dual pentode/triode like the 7199. The circuit worked well - no microphonics - when used in an amplifier *head* but I've not tried it in a combination amplifier/speaker cabinet configuration. (Incidentally, the 6BR8 tube cost is about \$2 as of this writing and can replace the popular 7199 tube in many circuits.)



The conservative choice would be the phase-splitter with gain, since the pentode circuit hasn't been tried in a combo amplifier (at least, not by me). Guitar amplifiers universally use the dual triode configuration - high-fidelity amplifiers

almost universally use the pentode configuration. I believe that the reasons for both choices are fairly apparent. Above all, a guitar amplifier must be rugged and resistant to feedback mechanisms while the environment of a stereo amplifier is benign. The design of the phase splitter with gain is covered in chapter 9.2.

In the event that the pentode seems to be a better choice, the tabulated design tables provided by tube manufacturers and reproduced above are the easiest way of determining component values. One could also use the same procedures described in chapter 8.0 (power stage design) to design a small-signal pentode amplifier. Chapter 8.0 emphasizes power design but one can design for gain using the graphic procedures discussed in 8.23.

22.6.5 Putting It All Together

As we progress, stage by stage, through the design process, through the breadboard process, it's always helpful to keep the spreadsheets updated. Most especially, the "block diagram" spreadsheet should be kept current: because that sheet estimates virtually all of the critical performance parameters of the amplifier chain. It's also important, as the preamplifier, post-amplifier and phase-splitter are designed, to update the block diagram spread sheet with correct plate voltages and currents otherwise the dropping resistor values on the "decoupling" spreadsheet will be incorrect.

If, for example, we determine that there is a limitation that prevents us from obtaining a certain amount of gain - or there is excessive compression in a certain stage of the chain - then re-evaluating the block diagram can offer important insight into what OTHER stages can be modified to compensate for these limitations.

The advantage, if not already obvious, is that spending an hour or two with a computer can save DAYS of workbench optimization, making changes, making measurements, trying to intuitively decide what the next step should be when a limitation somewhere in the chain is observed. Even if the computer model is not completely accurate (and it never will be), the critical value is the **trends** that can be observed by manipulating performance parameters in the spreadsheets.

It may not be possible, for instance, to increase amplifier output power from 23 watts to 27 watts (just as an example) by manipulating figures on a spreadsheet. But by manipulating figures, one is able to see an improving trend, a diminishing trend or one that has almost no effect. I've found that computer predictions frequently allow me to return to the workbench with a clear correction strategy in mind - this also gives me a break to clear my mind of thoughts that are not success-oriented. *It's helpful - no necessary - for a designer to be completely optimistic, always assuming that the design goals can and WILL be met.*

Making a circuit change, making a measurement to see the effect, adding observations to the spreadsheet; consider the advantage of this type of methodical work effort as opposed to the "easter egg hunt" strategy. Trial and error parts replacement, unclear measurement tactics and so forth - attempting to correct a problem that isn't clearly understood - just isn't productive.

No process will ever replace human ingenuity (plus knowledge seasoned by experience) in creating productive, effective designs. If we can reduce some of the repetitive burdensome calculations, minimizing the possibility of errors, then the process becomes not only more productive but more enjoyable ... freeing up our brain to use for more inventive purpose than repetitive calculations. Let's end this chapter on that note rather than attempting to completely design an amplifier based on five simple spreadsheets.

Having said this however, a continuing process should be to create and add other automated tools that will assist the designer. A clever designer will note areas in which excessive time is expended and develop personal tools to alleviate this inefficiency.

23.0 Appendix: Abbreviations and Relationships

A = amperes, unit of current

Average current = $.636 \times I$ peak or $.318 \times I$ peak-to-peak

Average voltage = $.636 \times V$ peak or $.318 \times V$ peak-to-peak

B = bandwidth, Hertz (maximum frequency - minimum frequency)

C_b = Capacitor, blocking

C_c = Capacitor, coupling

C_k = Capacitor, cathode

dB = decibel, a logarithmic ratio of power, voltage or current

E_b = Plate voltage

E_{c1} = Control grid 1 voltage

E_{c2} = Screen grid 2 voltage

E_k = Cathode voltage

E_o = Plate voltage signal swing, peak to peak

ESR = equivalent series resistance of large value capacitors

DMM = digital multi-meter

f = frequency, Hz

I_{avg} = Average current, see above

I_b = Plate current

I_{c2} = Screen grid current

I_k = Cathode current

I_q = Quiescent plate current (current with no signal present)

k = 1,000 units (e.g. 1k ohm = 1000 ohms, 1kV = 1000 volts)

mA = millamps, unit of current, 1 mA = 1 ampere / 1000

Meg = 1,000,000 units (e.g. 1Meg = 1,000,000 ohms, 1 MV = 1,000,000 volts)

mV = millivolts, unit of voltage, $1 \text{ mV} = 1 \text{ volt} / 1000$
mW = milliwatts, unit of power, $1 \text{ mW} = 1 \text{ Watt} / 1000$
N = turns ratio, primary to secondary (transformer)
Peak voltage = $1/2$ the peak to peak voltage
Peak-to-peak voltage = the difference between the highest and the lowest voltages of an alternating signal waveform
R = unit of resistance, ohms
Rg1 = Grid 1 grounding or bias resistor
Rg2 = Screen grid bias resistor
Rk = Cathode resistor
RMS voltage = $.707 \times V$ peak or $.354 \times V$ peak-to-peak
Rp = Plate resistor
Rs = unintentional series resistance (as in transformers or chokes)
T = temperature, degrees Kelvin (deg K = deg Celsius + 273)
t = time, seconds
uA = microamperes, unit of current, $1 \text{ uA} = 1 \text{ ampere} / 1,000,000$
uV = microvolts, unit of voltage, $1 \text{ uV} = 1 \text{ volt} / 1,000,000$
V = volts, unit of potential
Vavg = average voltage, see above
Vp = peak voltage, see above
Vp-p = peak-to-peak voltage, see above
Vrms = RMS (root mean square) voltage, see above
W = watts, unit of power

Current Gain:

$I_{\text{output}} / I_{\text{input}}$
 $20 \times \log (I_{\text{output}} / I_{\text{input}})$ in dB

Current Transformation:

$I_{\text{input}} / I_{\text{output}} = 1 / N$

Impedance Transformation:

$R_{\text{input}} / R_{\text{output}} = N^2$

Noise Voltage:

$V = (K \times T \times B \times R)^{0.5}$

Ohm's Law:

$I = E / R$ $E = I \times R$ $R = E / I$

Power Gain:

$P_{\text{output}} / P_{\text{input}}$
 $10 \times \log (P_{\text{output}} / P_{\text{input}})$ in dB
Power relationships:
 $P = E \times I$ $P = I^2 / R$ $P = E^2 / R$

Voltage Gain: $V_{\text{output}} / V_{\text{input}}$ $20 \times \log(V_{\text{output}} / V_{\text{input}})$ in dB**Voltage Transformation:** $V_{\text{input}} / V_{\text{output}} = N$

24.0 Appendix: Setting Up A Guitar

Many, many articles have been written about this subject. I've read a lot of them and have yet to find one that is "wrong". All will accomplish the desired results, so why am I going to add yet another one to the mix? Because it seems to me that there are two distinctly different problems in setting up a guitar and that solving the first is the key to solving the second (achieving a satisfactory setup).

Most of what I've read doesn't adequately *quantify the measurement technique so that it is 100% repeatable* (in fact it's not even 50% repeatable for me *within a reasonable length of time*). We're all somewhat familiar with the basics, what the various adjustments accomplish and how to perform them. But even before attempting to make the adjustments, one must be able to establish a standard of satisfactory performance (or compare against one). Perhaps an example will illustrate my point better.

24.1 Measuring String Clearance/Relief

You've just changed your strings to a set that you suspect is exerting different (makes no difference whether more or less) tension on the neck of your guitar and you think that a new setup is in order. Or you think that the instrument doesn't play as "easily" as it used to play. Or it fell off the stand, or any combination of these. How do you actually confirm that the instrument is playing as nicely as it used to play?

It's simple isn't it? One measures the clearance between the strings and the fret (usually first and sixth strings) at an arbitrary known point (usually the twelfth fret). The helpful folks who talk about these things – even factory technicians – will generally give guidelines to assist you in determining what satisfactory clearance should be (it's an individual thing). Clearance is usually given in terms of the distance between fret and strings, almost always expressed in units of 1/64 inch. If the measurements are the same as the last time you made them, the guitar has not changed.

Here are my problems with that technique:

It's difficult to make a measurement with readily available instruments, like a 6 inch machinists rule (e.g. individual vision quality, parallax error, reading and counting those tiny little tick marks).

Can't see small differences in measurements - 1/64 (approximately .016 inch) is actually a fairly significant amount, in my opinion. I can "feel" differences much smaller than 1/64 inch EASILY with my left hand when playing and so can most players. (As a point of reference, experienced machinists routinely feel errors of .002 inch in the alignment of two surfaces by touch.)

It takes an appreciable amount of time to make a measurement that is not very precise - that's unsatisfactory to me.

Some have proposed incrementally improved techniques, like substituting feeler gauges for measurement scales. This has the possibility of improving accuracy but a lot of experience is required to develop a "feel" technique that assures repeatability – successful, repeatable measurements with feeler gauges consistently elude me.

More sophisticated ideas have been tried for obtaining accurate measurements while eliminating "operator error" from the process. One that I've admired is a special-purpose device consisting of a stand that supports the guitar neck firmly while holding a dial indicator, calibrated in .001 inch increments, rigidly above the strings.

The indicator plunger is used to depress the string from normal position until it's resting firmly on the fret. The string clearance is then read from the indicator. It's accurate, too – I'd say at least five times more accurate than visually reading a precision machinist's scale.

(An electrical circuit is sometimes used with this method to assist in determining when the string is seated against the fret, an indicator light or buzzer is included in the circuit.)

There are multiple variations on this theme but a big problem with this type of instrument is that it's unavailable to the average guitarist (it's costly). Maybe the repair tech at the music store has one, but who wants to have to pay to have their action checked two or three times a year?

Another problem with this technique is that the plunger applies enough unintentional spring pressure to the strings to deflect them slightly, even when one is not applying intentional force to the plunger. This introduces error into the measurement that may be considerable, possibly as much as 30% and the amount of deflection is inconsistent between the first string and the sixth string.

(To be fair, since the measurement only has to be relative, this isn't a point of real concern.)

OK, maybe it's not so simple - What I wanted was a simple procedure, using *no special instruments* - that determined string-to-fret clearance (action) and was *consistent and repeatable*. I wanted to be able to write down the string clearance of the top/bottom strings and place that information in my guitar case. When the season changed, my strings changed, or I suspected something was amiss, I wanted to be able to *QUICKLY* re-check those measurements and *compare* against the last set of measurements.

And it turned out that, after fifty years of messing around with various other methods, the answer was simpler than anything I'd ever read: just use a guitar pick.

What?

Yep. First making sure that your guitar is in tune, stand it up vertically and slip a heavy pick (most jazz guitarists use them) between the first string and a fret, say at the fifth fret for example. Let go of the pick; gravity wants it to slip out and fall to the floor, right? But string tension - in a guitar with low action - constrains the pick between fret and string.

To make the measurement, one slips the pick between different frets, starting at a low fret (around 3 to 5) then moving up the neck, one fret at a time, until the string tension can't restrain the pick and it falls. At that point, obviously the clearance between fret and string is just slightly greater than the pick thickness. Write down the number of the fret. Repeat the process on the sixth string. (Make sure that the pick is constrained ONLY by the string being measured, don't allow the pick to contact anything other than the string and the fret.)

OK, got it, but how is that a "measurement"?

It's a "relative" measurement (although it CAN be calibrated to yield "real" results), a comparative, qualitative measurement - not quantitative. If you use the same pick each time then the thickness is constant. And by noting the fret number at which the pick "slipped out", you have a record of a known clearance at a known location which can be re-checked in a matter of a few seconds.

Do this on the first string and on the sixth string, write down the fret numbers, slip them into the guitar case and you're done. With just these two facts, one has established reference points that can be used to evaluate configuration differences as adjustments are made.

What is the significance of which fret the pick fell from or was constrained by?

If, during the course of your adjustment, you are able to secure the pick in a higher fret than when you started, you are lowering the action (e.g. the pick used to remain in fret 5 without falling, you made some adjustments and now the fret will stay in fret 7 without falling. The action has been lowered about .005 inches – that is a significant amount and you will feel the difference when playing)

But if you were trying to raise the action (and were successful), the pick would “stick” in a lower fret (e.g. you started out with the pick being constrained by fret 5 and now it falls from fret 5 but is constrained from falling at fret 3. You’ve raised the action by about .007 inches.)

Well, what constitutes a “good” setup, where should the pick be constrained on the first and sixth strings?

It is personal preference: defined by your attack, whether you play with a pick, your fingers, fingerpicks, and so forth. Don’t get bogged down in the detail of what the “right” measurement might be, the benefits of this measurement method: it’s FAST, reliable and invites easy comparison. You can try different adjustments and evaluate the results (allowing adequate time for your guitar to settle into a new mechanical configuration, obviously).

Let’s say, for example, your buddy just bought a new ES-175, you like the way that it plays and want to replicate the neck performance. Well, you sure can’t change the shape of the neck or your scale length, but you can easily determine the measurements of his action, using your pick, and then set up your guitar to those measurements.

You can even do this in a music store and make meaningful comparisons between guitars. BEWARE, however, those instruments are probably not set up as well as they could be. The concept – at least to me – FINALLY allows quantification of the phrase “low action”. (For what it’s worth, I set my guitars up so that a .038 thick pick can be constrained at the seventh fret of the first string, this results in a clearance at the twelfth fret of about 3/64 inch. No “buzzing”.)

Accepting this - so far - is this really an accurate method? Yes. The average jazz guitar has a slope between neck and strings of approximately:

$$\frac{(\text{string clearance} @ \text{13th fret}) - (\text{string clearance} @ \text{1}^{\text{st}} \text{ fret})}{(1/2 \times \text{scale length})}$$

which for a 25-1/2 inch scale guitar with string clearance of about .010 at the first fret and about .045 at the thirteenth fret gives a slope of about .0027.

Using a pick and inserting it between each fret in turn until it slips out will allow the detection of a string-to-fret height difference of [fret distance x slope]. For

example, using an average fret distance of around an inch, one can easily detect fret height differences equal to the above slope, or less than .003 inches (almost as accurately as the dial indicator setup discussed above).

What if I have really low action, will this still work?

Yes, just use a thinner pick, if necessary. The pick should be of an approximate thickness so that it just slips between string and fret about midway up the neck. But it's not at all critical – we don't care a piddle for exact measurements, we want a measurement that is repeatable. If you use the same pick each time you make a measurement, it will be repeatable. Just record the fret number at which the pick could no longer be constrained by string pressure, when the guitar is tuned to pitch.

What if I like really high action? Use a thicker pick or some other object (credit card or other “light” object with the right thickness - don't use coins or heavy items) and re-read the last paragraph. We want the *friction between pick* (or whatever you use in place of it) *and fret* to establish the point of slippage ...

Accurate, consistent measurement of “relief”, “clearance”, “action”, whatever you want to call it is CRITICAL to setting up a guitar properly. *If you have no standard against which to compare, there's no way to determine whether your adjustments are better or worse except by “feel” and you have minimal chance of making a good setup in a reasonable length of time.*

Note that this technique presupposes that the nut is in proper adjustment. As a general rule, the first priority in instrument setup should be to address the nut, adjusting if required.

If you cannot grasp this technique at this point, it's not your fault – it's mine for not making it clear enough in writing. It would take about fifteen seconds to actually show how to do this and explain why it works (and works well).

24.2 Guitar Adjustments

I believe that the reasons for setup/adjustment/maintenance are well understood but reviewing, we need to apply compensation to fragile wooden structures that have temperature stability and moisture absorption characteristics that cause movement (but have sonic characteristics that we value).

Earlier we established an efficient method for measuring neck relief . We can use measurements obtained in this way to evaluate adjustments that we make to our instrument. Guitar setup usually goes something like this:

Define what's to be accomplished by setup/adjustment.

Measure baseline configuration

Make first set of adjustments.

Allow guitar to achieve dimensional stability.

Decide if performance goals have been met

Iterate, if required.

Quality instruments usually require only “maintenance” of the setup configuration after a satisfactory initial baseline has been achieved. (Instruments of lesser quality may need some fiddly work on a frequent basis if they are to perform continuously at reasonably high standards.)

Defining the goal, what do we hope to accomplish? This is a “fuzzy” area and it may be helpful to document our goal(s). We may need to clarify our thinking on the subject (determine what we really want/need) also because the order in which we perform adjustments might depend on what we want to achieve if, for example, modifications are to be performed too. Some typical goals:

Play “easier”

Raise or lower the action

Stop string “buzzing”

Adjust “simple” intonation (making open string harmonic frequency = octave frequency)

Improve intonation over a broad range (difficult/time consuming)

No alignment should be attempted before verifying that the nut is properly adjusted with the strings anticipated to be used. If you think that you must make this adjustment, it's wise to anticipate problems by buying one or two unfinished nuts. (The parts are inexpensive and available from many distributors that are easily located with a routine internet search.)

24.2.1 Nut Adjustment

Strings should have *just enough clearance* over the first fret so that they don't “buzz” when playing open chord configurations/open strings. Many sources advocate a clearance of around .005 inches and that seems adequate to me. I certainly wouldn't make the clearance any greater and, in fact, I would try for a smaller gap - around .003 inches. This is because strings don't readily buzz on

the first fret, but an implication is that the nut adjustment is finicky - to get a tight clearance without buzz – and it is.

Recall when working on the nut, that the nut broadly affects the guitar's performance, especially in the area of the first four or five frets. Poorly adjusted nuts result in small difficulties in intonation and larger problems in "action".

Nut adjustment is easily made with a set of "jeweler's files" - tiny files of various configurations - supplemented with a few "jeweler's saw" blades for narrow slots. In a packaged set, available from normal tool sources, these cost a few dollars U.S. To make adjustments, one lowers string pitch until strings, with a little effort, can be lifted from their saddle notch and pushed off to one side of the slot, making the string slot accessible for modification (by filing).

It's a trial and error procedure: remove small amounts of nut material then replace the string in the saddle, tune the string to pitch and check for "buzzes" on first fret. As optimum configuration approaches, it takes only a stroke or two of the jewelers' tools to remove a significant amount of material - take care. Repeat the process for each string until all have about the same amount of clearance, string to first fret, and none of them buzz.

This is a situation where one CAN use feeler gauges with some assurance that the clearance measurement will be meaningful (unlike measuring clearance at the twelfth fret). This is because there is minimal deflection of the string, so close to the nut, and it is possible to get a good "feel" when slipping various thickness gauges between string and nut to find the right one.

If you go too far, no matter, start over again with one of the extra nut blanks. The work actually goes fairly quickly but even so, it's worth spending some time here. (If your string selection doesn't include wild variations of diameters, you'll only have to do this work on the nut once. So take some time and get it right.)

When you're satisfied with the nut configuration, you may want to use a larger file and reduce the overall height of the nut (i.e. reduce the amount of material that extends above the guitar strings). Many recommendations suggest taking the nut down to about the string center line level. I don't have strong feelings about it but I feel more comfortable with adjusting the nut height so that it is approximately even with the top of the strings.

24.2.2 Baseline Configuration

It's time to record the baseline configuration. If the term isn't obvious, what it means is a "snapshot" of the configuration of the guitar before we start making adjustments - with as many variables fixed as possible. We compare this baseline with measurements made during the adjustment process to monitor

progress and determine if the adjustments are producing improvement or degradation and how much.

Generally, a baseline configuration consists of your written notes confirming that the guitar is tuned and the pick/fret location (first and sixth strings) that we described earlier in the discussion. This is necessary to establish that improvements in action have been made - not only to the technician but to the customer who expects to detect a difference in guitar performance. Documented measurements are useful for resolving disputes.

Our work on the nut may have affected the action of the guitar slightly, so we should make a preliminary bridge adjustment. Because of the different behavior of the large and small diameters of the top and bottom strings, relief between string and frets will also be different. The first string can always be adjusted to have less clearance than the sixth string, all other considerations being equal.

The alignment process, whether it is bridge adjustment, truss rod adjustment (or even nut adjustment) requires that the instrument be brought to pitch each time an evaluation of performance is to be made (even if the evaluation is such a simple one as checking for buzzes). Progress compared to baseline performance cannot be evaluated unless the instrument is tuned to the same pitch as it was at baseline. And that's exactly what we need to do at this time ...

24.2.3 Bridge Adjustment

Bring the guitar to pitch and play every single-note position on the neck, trying to use an attack that you think represents your normal one. Reduce the height of the bridge on each side gradually (first string side and sixth string side), using the bridge adjustment thumbwheels, until you start to experience "buzzing". Note that the instrument must be brought to pitch after each bridge adjustment. An inexpensive electronic guitar tuner is invaluable for this and also for intonation work.

If you notice that one string is consistently buzzing while the rest are not, this may be a good time to replace that string with one by the same manufacturer and type but with a slightly larger diameter OR consider adjusting the depth of that particular string slot on the bridge. Be sure to make a note of the change.

(Conversely, if most of the strings are buzzing and one is not, it may be desirable to replace the non-buzzing string with one of slightly smaller diameter or lower the bridge slot for that particular string.)

Once the bridge has been adjusted to your satisfaction (bridge adjusted just high enough to be buzz-free across the normal playing area of the neck), it's time to make the measurements that establish baseline configuration, even if the current "action" is not acceptable. Using the technique described in detail previously,

with the guitar in proper tune, document the fret positions of the first and sixth string as your reference configuration.

If you are VERY lucky, you are now satisfied with the guitar in current configuration but it's never happened to me. I keep tweaking and twiddling because I like low actions. I suspect that frequently I have adjusted my guitar necks back to their original configuration - but at least I know that I have investigated the practical range of adjustment and satisfied myself that no "improvement" is available.

24.2.4 Truss Rod Adjustment

NOW comes the tricky part, the truss rod adjustment, the one that seems to cause the most mischief and misunderstanding. The truss rod serves one simple purpose: to compensate for the "bow" that string tension imposes on the guitar neck.

Remove the small cover on the headstock that covers the truss-rod adjustment nut. I strongly recommend that, before making any adjustments to the guitar, you remove the nut and clean the entire area carefully, using a soft brush, compressed air or both (both available from camera stores at modest cost). Lightly lubricate the threads of the truss rod with petroleum jelly (if you don't have this, use a light lubricant, even cooking oil is better than nothing) before replacing the nut. Screw the nut over the rod until it is snug (finger tight).

Measure the distance *across the flats* of the adjusting nut and confirm that you have a hexagonal socket wrench that *exactly* fits the nut. Not "almost fits" and not any other kind of wrench (e.g. 12-point), you really need a wrench that not only fits closely but completely confines the length of the adjustment nut for best results.

(The socket must also have an outside diameter that will fit within the confines of the pocket routed in the guitar headstock. If you can't buy a socket that meets these requirements, select one that fits the nut precisely and then have a machinist friend turn down the external diameter of the socket so that it fits into the neck pocket without friction.)

The socket isn't costly – about the price of a fast-food meal, so don't cut corners. The adjustment nuts are universally made from brass and easily damaged. That's intentional - it is preferable to damage the nut rather than the truss rod !

Using the "optimum" socket, adjust the truss rod (1/4 turn increments only – smaller is better if you have lubricated the adjustment nut as suggested), making the pick/fret measurements described above, until the baseline configuration has

been restored. Set the guitar aside for 12 hours in an environment representative of normal temperature and humidity.

Make the fret/string measurements again, after the stabilization period, and compare with the measurements on the first and sixth strings that were previously made. Don't be surprised if they differ by a fret or two ... The purpose of making the measurements now is to begin the process of "intuiting" the behavior of your guitar and to familiarize you with the measurement process. Remember to write everything down – keeping good records is the best way to understand what is happening and why (after you've LONG forgotten the adjustment details).

Now repeat the bridge adjustment process described above, checking each string for buzzing at every playing position until you are assured that the neck is buzz-free (but *just barely* if possible). Bring the instrument up to pitch as you make the bridge adjustments and confirm that the final adjustment is made with the guitar tuned.

Measure fret/string configuration again, compare with earlier measurements (again accumulating intuitive information about how the guitar is responding/moving with your adjustments). Play it for a while, up and down the neck, handling the instrument in a way that is representative of your normal manner. Make a determination as to the performance of the instrument as adjusted, if it's satisfactory, set it aside, let it "rest" for a day and make one more measurement to make sure that nothing has changed.

When the guitar is optimum for a particular set of strings, slightly reducing the height of the bridge will cause a buzz throughout normal playing areas – this is the classically desirable adjustment scenario. From this point, one should need only to adjust the bridge height in order to make the action as "stiff" or as "low" as one desires. (Obviously, there is a limit as to how "low" you can go.)

Assuming that you've adjusted the instrument for optimum performance with the strings that are on it and don't LIKE the combination, you might consider replacing the strings with ones that have different characteristics and start over again with the optimization process. Your intuition will suggest whether heavier or lighter strings will be more suitable.

(As noted earlier, sometimes one finds that just one string is a problem ... for example, everything plays great except for the SECOND string, which buzzes when the other strings sound fine. Try replacing the second string with the next larger diameter, it's not hard to find the right one, provided that you are not using an esoteric set, made from some abnormal material. Just write down what you've done so that you'll remember the next time that you buy strings.)

It's worthwhile to point out that by now most guitars should have reached a point of diminishing returns and you will notice only very slight improvement for very large amounts of time expended. Decisions about further work on the instrument need to take this into account.

If you have a "young" guitar (perhaps from a manufacturer that doesn't observe careful materials selection and aging) it's possible that the guitar is still moving around. There's not much to do about this except to allow time to pass ... years may be required, who knows ?

Summarizing the adjustment process:

Adjust the nut

Adjust the bridge

Adjust the truss rod

24 hour stabilization

Iterate as required

After one is satisfied with performance and stability, intonation can be addressed.

24.2.5 Intonation Adjustment

Intonation is a term that describes the ability/inability of an instrument to produce the tones of a specified scale. Western instruments usually are tuned to the "even-tempered scale". (Please run both terms through an internet search to obtain specifics in the unlikely event that you are unfamiliar with them.)

We don't have very much latitude in the correction of intonation problems. Usually, the only adjustments made during setup/maintenance are those that are harmonically related. The process is simple but requires the use of an electronic tuner. Many years ago, this was a drawback but tuners these days are almost throwaways, in terms of cost.

The adjustment procedure that I'll describe doesn't require an instrument of any particular accuracy, since we are only interested in the relative difference between two tones and not their absolute accuracy. This is a well-discussed procedure and should be familiar to all.

(Although it is frequently claimed that a high degree of tuner accuracy is required for intonation adjustment, this is simply not correct. Without diverting TOO much from the main topic, here's why. All electronic tuners use a crystal oscillator to establish a reference tone, they employ digital frequency dividers to process the

reference frequency into those tones to which we want to tune. We can't depend on absolute frequency accuracy but we CAN depend on the fact that the tones are integer divisors of the reference frequency, so if we want to compare harmonics, ANY electronic tuner – regardless of frequency accuracy – is adequate for the purpose.)

After tuning the guitar to pitch, the procedure requires striking a harmonic on each string and examining the indication of frequency on the tuner. (Generally, if one places a finger lightly right ON the twelfth fret wire – not centered between frets as we usually finger the instrument – it's easy to produce a harmonic with reasonable amplitude.)

Moving from string to string, play a harmonic, carefully noting the indication of pitch (actually pitch error) on the tuner. Then fret the twelfth position and play the note, monitoring the tuner and noting any difference between the *fingered* note and the *harmonic*. Then the bridge is adjusted to reconcile differences between the two notes. For TOM (tune-o-matic, Gibson TM) one uses a small screwdriver to adjust individual string saddles until the difference between harmonic and fretted tone is negligible.

For fixed, wooden bridges, a compromise is required. Generally, one makes an adjustment to the bridge (by moving it forward and back) so that the first string and sixth string harmonics are equal to their fretted notes. If the bridge compensation is carved correctly for the set of strings you are using, and you're satisfied with the performance then you're done.

Direction of bridge movement (or saddle movement) is predicated on the pitch error of the fretted note. If it is "sharp", move the bridge (or saddle) toward the tailpiece of the guitar. For "flat" errors, the bridge (saddle) is moved toward the headstock of the guitar.

If you want to take the correction further (which suggests that your bridge isn't compensated properly for your particular strings), my suggestion is to play all of the strings and play their harmonics, monitoring the tuner and recording the errors. For convenience, devise a table and chart the magnitude of the errors between tones, noting whether the error is sharp or flat.

Examining the chart can reveal that sloping the bridge at an angle will minimize errors across the entire range. In other words, instead of a perfect high "E" and a perfect low "E" with a "G" that is flat by a magnitude of ... say $\frac{1}{2}$ division (on whatever tuner you're using), you could slope the bridge to obtain high and low "E's" that are $\frac{1}{4}$ division sharp and a "G" that is $\frac{1}{4}$ division flat.

The point is to equalize the errors across the entire scale instead of absorbing the error into one or two strings that will be blatantly obvious. (Having said that,

the “blatantly obvious” will probably only be apparent to you – the guitarist – and not to an audience.)

(Do an internet search on the “Buzz Feiten method”, you’ll find mostly discussion of the compensated nut – which may not be of particular interest. But if you read the discussion of the entire method carefully, the implication is similar to what I’ve proposed above: that the frequency errors are distributed across the entire scale instead of being concentrated in the second through fifth strings.)

I haven’t furnished any detail about how the slope of the bridge can be mathematically determined, to equalize pitch errors. That’s because I think that those who are interested in this method will have the capability to easily do it themselves. However, an empirical adjustment, using a tuner and a chart, is within anyone’s capability.

It should follow that, once one has carefully positioned the bridge for intonation, changing strings implies taking a little care to prevent moving the bridge. Most do this by replacing the strings one at a time, depending on remaining string tension to hold the bridge in place. Taping the bridge to the guitar body at both edges is not uncommon if the entire string set needs removal.

In summary, once a guitar has been properly adjusted and assuming that the guitar has been carefully manufactured from quality, aged materials and that the strings intended for use will be similar (from set to set), nothing much in the way of maintenance need be anticipated other than $\frac{1}{4}$ turn or so on the bridge adjustment screws once or twice a year with the change of seasons. (Naturally travel, changes in altitude, changes in temperature and humidity, will complicate adjustment procedure.)

25.0 Appendix: Comparing Loudspeakers Using SPL Data

The following is a scenario that many have experienced. Needing to replace a loudspeaker, how does one make a sensible selection when it’s not possible to make a side-by-side comparison between two different speakers driven by the same amplifier? Please note that this procedure is appropriate for open-back configurations only. (Sealed or ported enclosures can also be modeled in a similar manner but the technique is slightly more cumbersome to employ and to explain and is omitted, at least for the moment.)

(Note that this technique is intended for linear musical reproduction - the area in which most jazz guitarists are interested. It is possible to devise methods that would incorporate distorted sound in the evaluation but at the moment, I am not motivated to do this.)

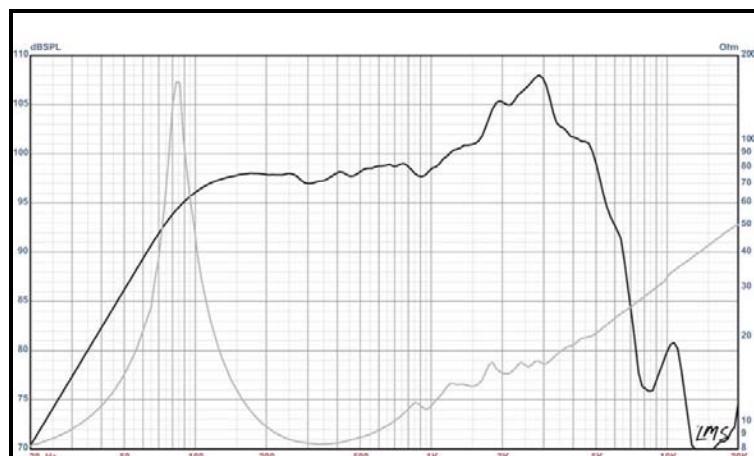
Unhappy with the performance of the loudspeaker in his small amplifier, Tal would like to replace it. He's found the description of his original speaker, which the manufacturer describes as:

"Very loud, touch sensitive and responsive with nice bell-sounding top end and a little bite."

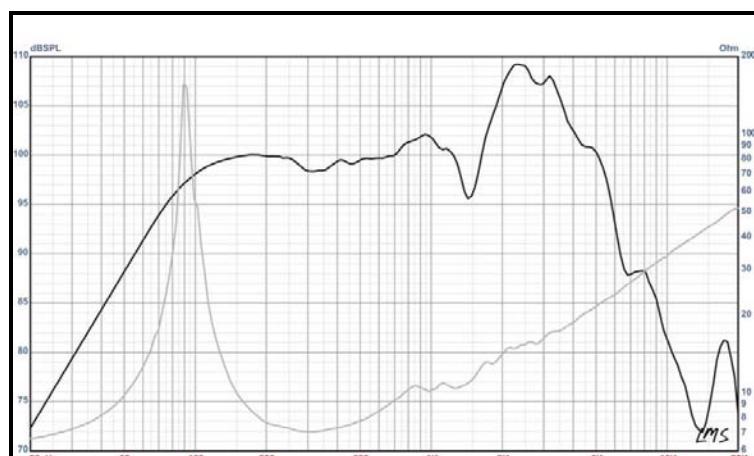
He is considering a speaker recommended by a friend that has *this* description:

"Performance optimized for lead guitar. Also suitable for bass guitar, vocal P.A., keyboards, club music systems and stage monitors."

Unsure of the meaning of the market-speak, and slightly troubled by the fact that the same manufacturer actually makes BOTH speakers, our man decides that a comparison between the two should be based on measurements of the two speakers. A side-by-side comparison being impossible, he makes a brief search of the products on the internet and finds these sound pressure level (SPL) curves, one for his existing speaker and one for the new speaker being considered:



Yankee's Best Loudspeaker



Empire Ruler Loudspeaker

Looking at both performance curves carefully, Tal observes that "his" loudspeaker has higher SPL (it's louder) but less bandwidth (total frequency response) than the "Empire". He makes what *he* believes is a reasonable assumption: that the *total sound pressure level* each loudspeaker projects is actually about the same. He believes the difference is in the power *distribution* across the frequency range.

Tal wants to make this speaker exchange just once and he's careful when making financial decisions. He feels that the speaker decision needs to be based, NOT on performance curves but on how he's *always* made decisions about music: with his ears. But how can he do this?

Technological advances in digital signal processing (the same technology represented in low cost CD players) have progressed logically toward sophisticated computer recording and editing programs. These are universally available at modest cost - sometimes free. These programs have an impressive array of post-processing features, some of which are overlooked due to their seeming simplicity of purpose.

Tal has downloaded an, open source recording/editing application called "Audacity". As his intuition has suggested, Tal opens the program and discovers that there is an "EQ" effect included. He notes that the equalization response can be adjusted to emulate an SPL curve with any practical level of accuracy (depending upon how much time he wants to spend equalizing the curve).

(Note that "Audacity" is not claimed to be superior to any similar commercial programs but it is *free*.)

Re-examining the original "Yankee's Best" SPL curve, Tal makes some observations. He notes that the lines depicting measured speaker SPL are not smooth, there are lots of "squiggles" in the frequency response. Most can be ignored because differences in levels of less than 1 or 2 dB or so are not distinguishable by the typical human ear.

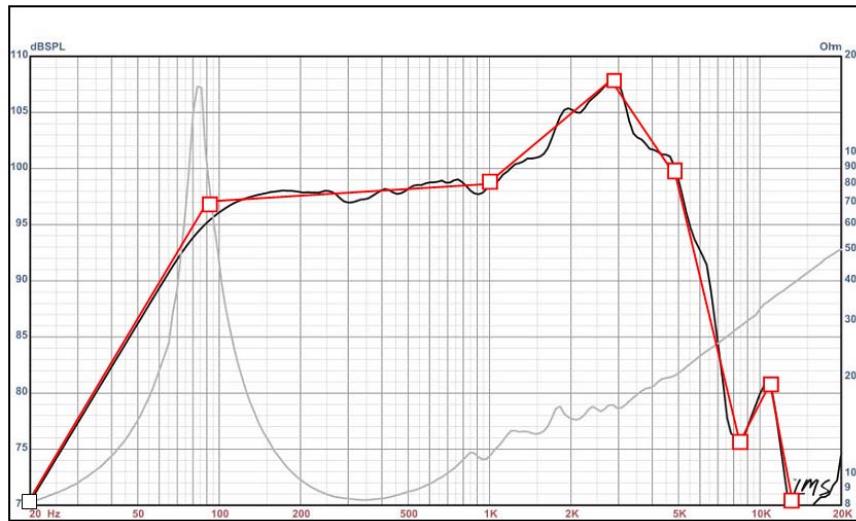
Tal starts to draw some straight lines through the erratic responses of the SPL curves, approximating the slopes and magnitudes of the measured response. He realizes that if he WANTED to closely replicate the nuances of the squiggly variations in the SPL curves, he could do so, but it would require considerable time. (The point is to define the important points in the EQ curve that replicate the sound of the speaker without expending an inordinate amount of time.)

There are many ways to perform this exercise, one is to click on the SPL curve displayed on the internet by the speaker manufacturer, "copy" the information, then "paste" it somewhere convenient, like in this "Word" document that I'm typing. Then "draw" EQ lines with available drawing tools. I used the tools

available from the "Word" drawing toolbar. Microsoft "Paint" also works; use whatever is familiar.

If you happen to have the catalog with the SPL curve displayed, you can draw in the straight line approximations with a pencil. The goal is to define the SPL curve with a relatively simple series of frequency and sound pressure coordinates.

This is Tal's approximation of the SPL curve after he drew his lines on it:



After sketching the EQ characteristics, Tal wants to use "Audacity" to set up identical equalization (in the "effects" section) to duplicate his speaker SPL curve. (*Each of these recording programs has its own peculiarities and preferences, since "Audacity" is the one that I have been using so it will be the reference for our purposes.*)

Tal needs to establish a maximum signal level and for Audacity this is a reference level of 0 dB, signal levels beyond this point will result in clipping, which is undesirable. (*Note that the SPL curve, properly, is calibrated in a scale that is appropriate for the human ear, but not necessarily for recording purposes.*) Tal decides to modify the SPL level curve by "normalizing" it to "0 dB".

This is accomplished by subtracting the maximum SPL (in dB) from the SPL of each point on the curve.

The first point on the curve, 70 dB, would become $(70 - 108)$ or -38 dB.

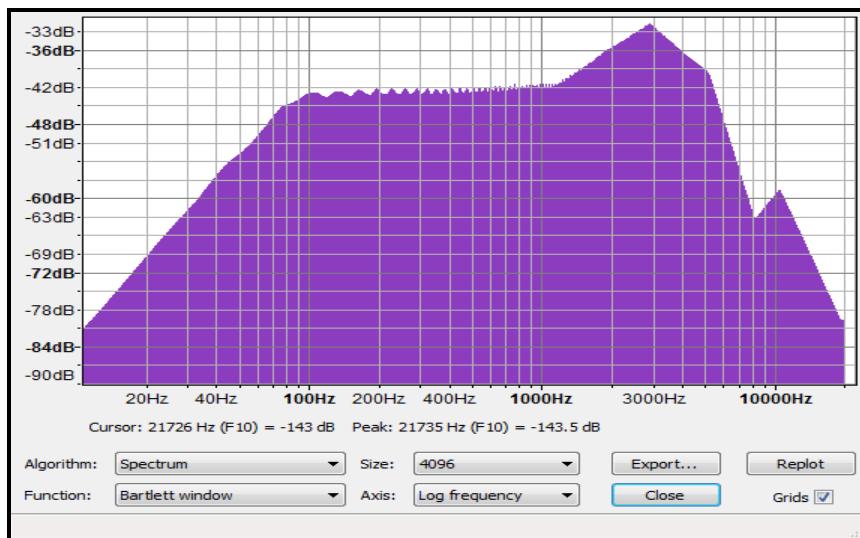
The next point on the curve, 96 dB becomes $(96 - 108)$ or -12 dB.

The peak reading of 108 dB becomes $(108 - 108)$ or 0 dB.

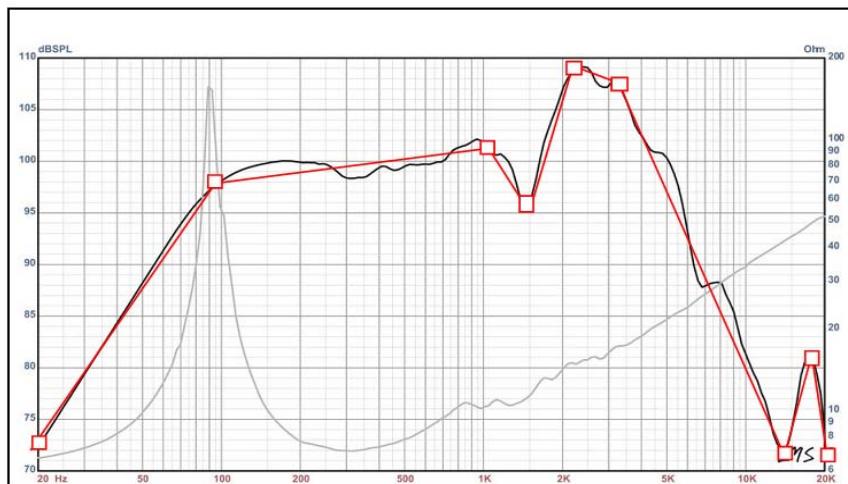
And so forth.

(Note that the SPL curve characteristics have NOT BEEN CHANGED, just re-referenced to a level that the recording program can use. Attempting to input a SPL of 108 dB would not have been permitted by the program.)

After Tal adjusted the Audacity equalizer, he gave it a name (the loudspeaker name) and saved it in the program. Tal used the Audacity internal white noise generator, passing the noise through the EQ curve to "record" the response. Then he used the Audacity spectrum analyzer tool to examine the equalizer frequency response confirming that the response appears the same as the loudspeaker SPL curve.

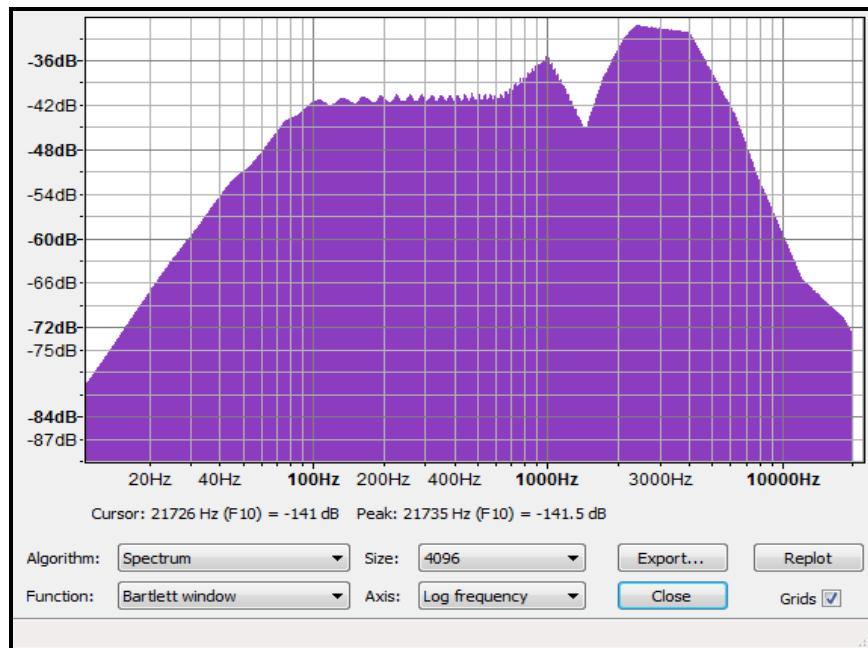


He then repeated the exercise, using the SPL curves from the "Empire Ruler" speaker that he is considering, first drawing straight lines to approximate sections of the SPL curve that are linear and using the intersections of the lines to define the points that will be used for equalization. After spending a few moments, Tal's results produce the following :



As in the first exercise, Tal normalized all of the sound pressure levels by subtracting the maximum SPL rating from the nine points shown on the above graph. The maximum SPL is 109 dB, so "109" is the value that must be subtracted from the other eight points on the curve.

After adjusting the "Audacity" equalizer for the new SPL points and saving the EQ curve, internally generated white noise (from Audacity) is passed through the equalizer to confirm that the shape is the same as the shape of the SPL specification for the speaker. Here's the spectrum analyzer display of the result:



Tal is almost done ... he's created and stored the SPL equalizer curves of the two different speakers, now all he need do is play some music through the two different equalizer curves and LISTEN.

The best possible tool, since his intent is to install the speaker in a *guitar amplifier*, Tal reasons, is probably his *guitar* recorded directly to "Audacity" with no "color" added to the sound of the instrument. A complication is introduced, at this point. All guitar amplifiers have internal equalization that accommodates magnetic pickups commonly used in electric guitars but Audacity has no intentional equalization.

Since we can create any EQ curve desired in Audacity, let's create an approximation of "normal" guitar equalization. Set the 500 Hz EQ for -6 dB loss and adjust the 400 Hz and 600 Hz EQ for about -3 dB loss; all other frequencies to be adjusted for 0 dB loss.

That should be an adequate representation of most amplifier pre-emphasis scenarios. In order to avoid confusion, this should be saved as a separate EQ curve and applied to guitars that are to be recorded directly to Audacity.

The following is a suggested procedure when recording a guitar directly to the computer for speaker evaluation:

Guitar, recorded straight into Audacity

Guitar track filtered by the pre-emphasis EQ curve

The pre-emphasized guitar track filtered by the "Yankee's Best" EQ curve

The pre-emphasized guitar track filtered by the "Empire Ruler" EQ curve

For the purposes of evaluation, the computer output is connected to studio monitors or high quality headphones and the last two tracks are "panned" between left and right to get an idea of the tonal qualities of the two speakers being compared.

If the two different speakers have different maximum sound pressure levels, then the final two tracks should also have their levels altered to represent the difference in efficiency between the two speakers. The "Yankee's Best" loudspeaker was 1 dB higher in SPL than the "Empire Ruler" loudspeaker therefore the last track - the one recorded with the "Empire Ruler" EQ curve - needs to have its level reduced by 1 dB. This is easy to accomplish by clicking on that track and then on the "effects" toolbar, selecting "amplify" and then entering "-1dB".

Another useful means of evaluation is to use pre-recorded music, that is exclusively (or mostly) guitar music. All that is required to make this evaluation is to pass the pre-recorded track through the two loudspeaker EQ curves and then listen to the results.

Many are thinking at this point that this is a LOT of work. But it's not, it takes about five minutes to EQ a SPL curve, start to finish. Add five or ten minutes to record your instrument and another ten minutes to set up your software, EQ and playback ... the entire process should take way less than an hour. And once the source material (guitar) has been recorded, the comparison of other speaker SPL curves will require only the time required to create an EQ curve and a few moments listening time.

I've previously pointed out the importance of speaker SPL, noting that a difference of 3 dB in loudspeaker SPL is *the same as either doubling or halving amplifier output power*. That should resonate rather emphatically with most - like moving up from a Fender Princeton to a Fender Deluxe Reverb ! But I've also

observed that human hearing can't detect small differences in SPL. (Many that study the subject of audible sound suggest that the minimum detectable change in power level is on the order of 1 to 3 dB, depending on the frequency of the sound and the age/gender of the listener.)

There's a dichotomy - but it's a psychoacoustic issue, not an engineering issue.

Although making side-by-side comparisons of different speakers with the same amplifier would be desirable, a method has been described here by which open back cabinets loaded with various loudspeakers can be evaluated quickly.

I believe this technique to be useful for screening to individual taste and might be particularly useful for reducing a number of loudspeaker candidates to a manageable two or three.

One can even expand the technique by creating yet another equalization curve that is exactly suited to ones personal hearing characteristic. There are online applications that can quickly and easily measure your personal hearing range.

I've used the following tool satisfactorily:

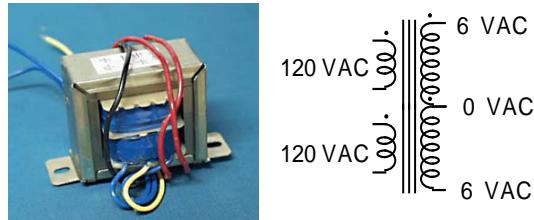
<http://www.phys.unsw.edu.au/jw/hearing.html>

This evaluation procedure is limited to "clean" (undistorted) performance, which may not be useful to some. I've employed the technique with success and passed along pre-equalized material to many people who were interested in comparing different loudspeaker responses. The open-back cabinet limitation similarly does not have to be restrictive since that is the configuration most frequently used for jazz guitar amplifiers.

26.0 Appendix: Inexpensive H-V Power Supply

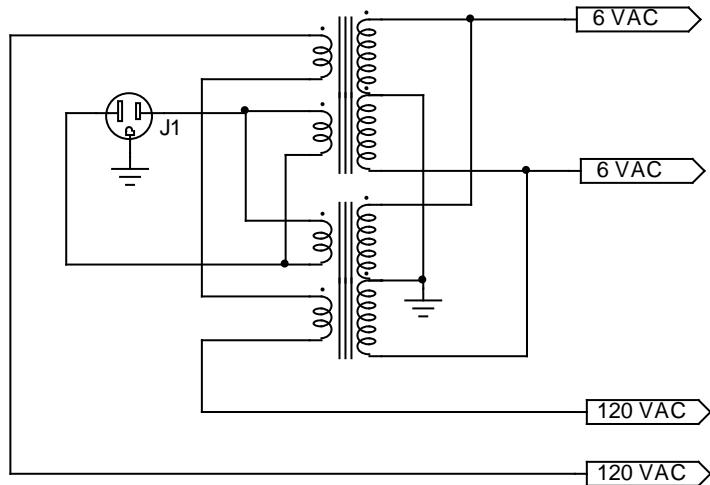
The power supply transformer is one of the most expensive parts in a vacuum tube amplifier and also establishes the maximum output power level of the amplifier. Because there is not much demand for high voltage power supplies, there are no economies of scale - this drives the cost of the transformer. The demand for low voltage transformers used in solid state equipment, however, continues to grow, reducing prices accordingly. Noting this, I inferred that a useful power supply could be constructed by pairing transformers with some useful characteristics.

Specifically, what is required is a transformer with 120 VAC/240 VAC primary voltages. This type of transformer is usually constructed with separate 120 VAC primary windings as shown below:



The transformer is used for 120 VAC inputs by using one of the primary windings (or both, connected in parallel). For 240 VAC input, the two primary windings are connected in series, the center connection point must be insulated so that it cannot contact other circuit nodes. These transformers usually are limited to a rating of less than 50 VA, so they wouldn't be useful for many vacuum tube circuits, where filaments require large amounts of current. (The vacuum tube preamplifier described in chapter 18.5 was powered by a similar transformer.)

We can configure a *pair* of transformers as follows, producing interesting possibilities:



Two transformers, connected as shown above, have the following characteristics:

A single primary input winding from one transformer is wired in parallel with the other transformer, doubling the current carrying capability of the primary windings.

The other primary input windings of the two transformers are connected in series, these windings are now available as secondary windings, they have double the input voltage and the same current capability as the input winding

(The common connection between the two primary input windings must be insulated)

The secondary windings of both transformers are wired in parallel with one another, all voltages remain the same but current capability is doubled

Instead of two separate transformers we have what is, in effect, a single transformer of enhanced capability. If each of the transformers is rated at 48 VA, then our new "transformer" will have the following characteristics:

Primary voltage: 120 VAC, max current consumption of 0.8 amperes

Secondary voltage 1: 12 VAC, center-tapped, up to 8 amperes

Secondary voltage 2: 240 VAC, up to 200 milliamperes

Note that the current suggested by the VA ratings of the transformers applies to the sum of both secondary windings, if in use simultaneously. The maximum current available from any combination of the two must be less than the sum of the two transformer ratings or $2 \times 48 = 96$ VA, this isn't a very restrictive limitation. As an example, a selection of four vacuum tubes with 6 volt filaments that require a total filament current of 4.5 amps requires

$$6 \times 4.5 \text{ VA} = 27 \text{ VA}$$

This value must be deducted from the total rating of the transformers before estimating the plate current available from the power supply. Subtracting the filament power consumption, the resulting transformer power rating is:

$$96 \text{ VA} - 27 \text{ VA} = 69 \text{ VA}$$

And the available current from the 240 VAC secondary is then

$$69 \text{ VA} / 240 = .288 \text{ A} \text{ (at full drive - maximum signal - condition)}$$

The advantage of using two transformers for the power supply:

Cost is considerably lower than a typical 240 VAC transformer with filament windings (about 1/4)

Weight and size, compared to typical H-V transformer, are diminished (about 1/ 2)

An additional possibility that is attractive: converting the 12 VCT secondary voltage into a D.C. supply of about 5.5 volts - a very desirable feature for reducing A.C. ripple (hum) emanating from tube filaments, especially those in the preamplifier and post-amplifier stages of the amplifier. A collateral benefit is the extension of tube life.

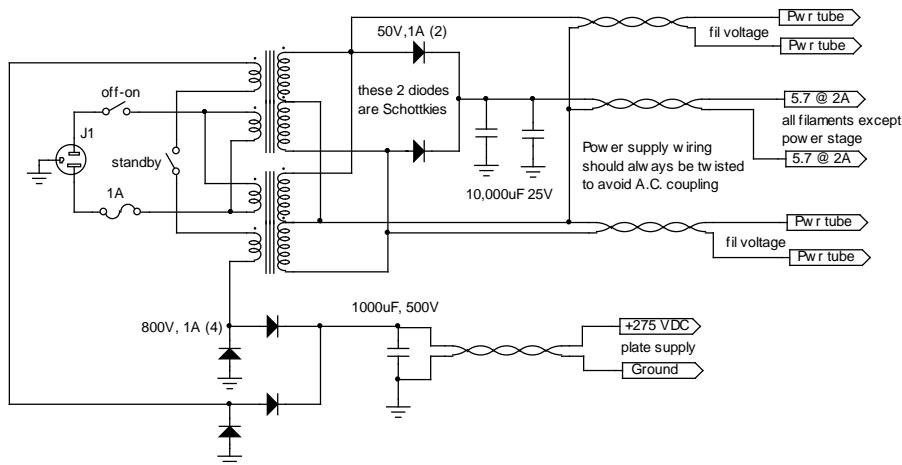
Output tubes have low *voltage* gain, compared to other stages of the amplifier, so it's not necessary to operate their filaments from D.C. We can operate the output power tubes filaments directly from each leg of the 12VAC transformer output which is equal to 6 VAC. Another reason for operating the power tube filaments from the A.C. voltage is to avoid the voltage drop caused by the rectifier diodes, approximately 0.5 volt (the small-signal stages won't be affected by this drop).

Transformers for this example were intentionally selected for moderate rating (not a high VA rating); they are inexpensive to purchase and replacements - if needed - will be easy to find. Replacement transformers for high-voltage applications are generally teeth-clenchingly expensive. (This was in my mind when an alternative occurred to me (NOT that it's a unique idea). This configuration, it should be noted, isn't helpful for replacing existing transformers unless one wants to perform fairly extensive metal-work modifications.

If we return to our "power supply" spreadsheet and use the 240 VAC, 288 mA secondary obtainable from the above circuit, we find that the "case 2" circuit with some iterations will provide a plate power supply of about 280 volts at 200 mA with a low-ripple (hum) output. Recalling that output stage power efficiency is about 50%, the power output available from an amplifier supplied by the circuit described previously could be conservatively rated at 25 watts.

That's a nice supply capability - selecting a pair of surplus television horizontal output beam power tubes (inexpensive) for a 25 watt amplifier wouldn't be at all difficult. Adding a few embellishments to the new transformer circuit (filtering, circuit protection, switching) this could be a compact, lightweight power supply design at an estimated cost of about \$40.

Depicted below is a schematic representing a power supply with the performance that we've discussed. This supply would be a basis for an inexpensive amplifier using NOS 18 watt power tubes (with I_{max} greater than 400 mA - many are available for less than \$6 U.S.) and an output transformer with turns ratio of about 17:1.



A few comments regarding the power supply: This concept can be extended to three transformers for greater voltage capability. Schottky rectifier diodes are required *in the filament circuit* to minimize voltage drop (about 0.3 volts for the Schottky diodes versus 0.8 volts for standard rectifiers). The above component values and ratings are minimum - substitutions of greater values and greater ratings are perfectly acceptable.

The standby switch is optional, I don't feel strongly about this option for plate voltages less than 300 volts so long as the screen grid is biased consistent with previous comments on this topic. All of the voltage drops and output voltages are based on approximations of the internal transformer resistances and may vary. The transformer type that I had in mind for this supply configuration would cost about \$8.

With that modest cost in mind, I *strongly* suggest that a breadboard exercise might be appropriate (before building up a complete supply only to find that it falls short in plate voltage by twenty volts or in filament voltage by a volt or so). A quick exercise, substituting power resistors for the various load voltages/currents can be performed on ONE transformer, using load resistances that are twice those actually required and capacitors that are half the value of those required. This will provide a go/no-go decision as well as some useful insight into transformer voltage drops.

Recently, I've observed some interesting U.S. made transformers entering the market at very attractive prices. These fall into the range of voltage conversion transformers whose primary intention is to allow 120 VAC devices to operate from 240 VAC and the converse. I'm looking at one of these transformers rated at 150 VA and a cost of \$13 U.S. This small (4 x 2.5 x 2.5 inches) transformer can easily provide the high voltage power supply for an amplifier of up to 70 watts RMS with appropriate output tube/output transformer selection. As noted previously, a separate filament transformer would be required (which is always my personal preference). The information for this transformer has been included in the parts selection sheet of the vacuum tube amplifier workbook described in chapters 6 and 22.

A pair of these transformers would make an excellent , small laboratory supply, providing up to 600 volts at 400 mA ! Variable output voltage features with current limiting can be added similar to those in the examples of chapter 20.2.1.

The circuits shown in that chapter are not adequate for a power supply of the magnitude suggested above (240 watts) but mentioned only as an example of possible means of attaining adjustability. In addition to the very high power circuit design problems, the heat exchanger for the pass transistors would be a formidable design project and should not be considered as an amateur-capable project. Only highly efficient forced-air cooling is capable of keeping pass

transistor junction temperatures at reasonable levels and this part of the project would be beyond most experimenters' capabilities.

However a *fixed* power supply, using a single or dual transformer like the one above, is *definitely* within the capabilities of those who understand the concepts discussed in this book and follow good safety practices regarding high voltages.

If one has a requirement for performance in the 15 to 20 watt output power range with "turnkey" convenience, I've noticed recently that open-frame, linear power supplies around \$40 U.S. are available (265 volts, 100 mA). Note that "standard" (as in low power Fender amplifiers) vacuum tubes wouldn't be appropriate for this low-voltage requirement.

One would have to select tubes with a much higher I_{max} than EL-84 or 6V6 tubes. An additional filament transformer would be required at modest cost, \$3 to \$5, perhaps. This might be a good design decision; these supplies are fully regulated with current limiting and so forth - very good value/performance package for low-power applications.



27.0 Appendix: Audio Impedance, A Small-Signal Overview

Placing "something" between the electric guitar and the amplifier is a very different proposition than the choices available to me when I obtained my new 1961 Gibson Les Paul guitar. At that time, the only choices available were:

6 foot or 10 foot cable from perhaps two (?) manufacturers

Black Diamond strings or Gibson strings

Round-wound or flat-wound were the only choices, by the way - string gauges were established by the two manufacturers and it didn't occur to anyone to

question the choice (but we DID remove the wound third string and replace it with a second string, just like Chuck Berry did).

I'm consistently astounded by the choice of "pedals" available today - about 70% of them seem to be about different types of distortion, 20% about delay and the remainder related to phase-shifting and chorus functions. Choices of these products is related to taste, preference, opinion, individual hearing ability and the way that one chooses to modify the sound of one's instrument.

Recall that the definition of the ideal amplifier is:

$$\text{Output signal} = \text{Input signal} \times \text{gain}$$

Any deviation from that description approaches either "distortion" or "signal processing" (whether the implementation is digital or analog). I wanted to speak to the complexity of choices before discussing the effects of inserting *anything* between guitar and amplifier.

Let's consider how impedances of the various devices in the signal chain - musical instrument, cables, processors and amplifier - interact. Let's start the discussion by describing some transducers (mechanical-voltage converters) that we refer to as "pickups". Transducers that approach an impedance of 1 megohm are characteristic of "crystal pickups". This configuration is rarely used (in the past, the most common usage was phonograph cartridges). Piezoelectric pickups are also high impedance; they are used mostly for acoustic instruments such as guitar, violin, dulcimer, autoharp, accordion and other instruments that produce sound by vibration of a surface.

Pickup impedance is complex (literally "complex" by mathematical definition) because impedance consists of "real" and "imaginary" parts - resistive and reactive - a boring topic for most of us. Generally we refer only to the "magnitude" of the complex impedance, about 5k - 10k ohms for guitar pickups.

"Line" impedance (in the U.S.) is usually understood to be 600 ohms, derived from traditional telephone engineering practices. More recently, the term is applied to impedance levels ranging from a few hundred ohms up to 2k ohms or so.

Small-signal impedance interfaces normally affect two significant performance parameters, signal voltage loss (attenuation) and frequency response.

27.1 Attenuation From Impedance Mismatch

It may be helpful to review chapter 5.2 at this point. The example in that chapter, describing the interaction between a battery and a resistor may provide an intuitive understanding that can be useful as an introduction to this topic.

Here's a description of signal voltage, related to the interface characteristics, and a simple mathematical description of the effect:

Higher impedance sources (guitar pickups and many microphones) are "loaded" by lower impedance input impedances. "Loaded" refers to signal voltage reduction of the pickup or microphone. The signal voltage of these devices is reduced by the following relationship:

$$V_{\text{load}} = V_{\text{source}} \times [Z_{\text{load}} / (Z_{\text{source}} + Z_{\text{load}})]$$

where **V source** is the signal voltage (voltage produced by the guitar pickup) available to the mixer, amplifier, processor, **V load** is the *actual* voltage available to the mixer, amplifier, processor (the load)

Z source is the input (guitar) impedance and **Z load** is the input impedance of the mixer/amplifier/processor and so forth.

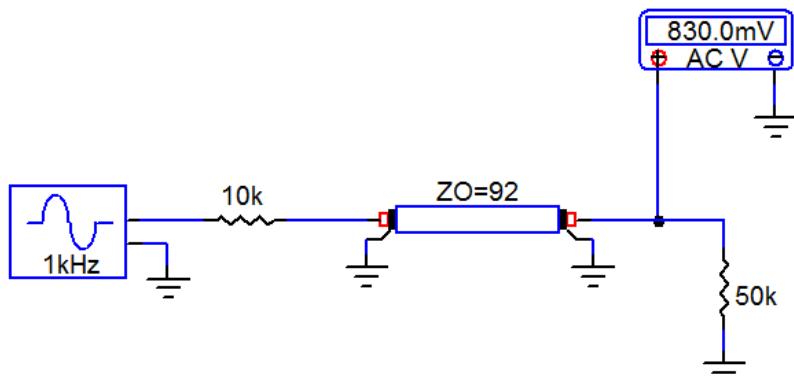
Manipulating the above equation, one can conclude that the *load* (the processor or amplifier, not the guitar, which is the signal *source*) should have the highest impedance possible. Good intuition, if you concluded that. But, Mother Nature must always be paid - compromises are always required. Let's create a simple, practical circuit example that demonstrates the mathematical expression:

Source impedance (a guitar pickup, for example) = 10k ohms

Load impedance (an amplifier input, for example) = 50k ohms

Connection between the two is a coaxial cable with an impedance of 92 ohms and a length of about 15 feet

(For the moment, this cable isn't important but we'll understand why it is included in a moment.)



The schematic is a screen capture of a computer circuit analysis simulation. In circuit analysis, a signal generator (or a battery) is always assumed to be zero impedance so the series resistor of 10k ohms represents the source impedance (generator impedance of zero ohms + 10k ohms = 10k ohms). The output voltage of the signal generator has been fixed at 1 volt RMS in this example.

The 50k resistor represents the amplifier input impedance. I've placed a "simulated" RMS voltmeter across the load to illustrate how load impedance affects signal voltage. In the above situation, we see that the 1 volt source signal voltage has been reduced to about 0.83 volts. Checking this simulation by solving the above mathematical expression:

$$V_{\text{load}} = V_{\text{source}} \times [Z_{\text{load}} / (Z_{\text{source}} + Z_{\text{load}})]$$

and inserting values:

$$\begin{aligned}V_{\text{load}} &= 1 \text{ volt} \times [50,000 / (10,000 + 50,000)] \\&= 1 \text{ volt} \times (50,000/60,000)\end{aligned}$$

Or about **0.833 volts** (note that round-off errors in the simulation cause a slight disagreement with the calculated value).

If we change the amplifier input impedance to **10k**, the expression predicts that the signal voltage at the input of the amplifier will then be:

$$\begin{aligned}V_{\text{load}} &= 1 \text{ volt} \times [10,000 / (10,000 + 10,000)] \\&= 1 \text{ volt} \times 10,000/20,000\end{aligned}$$

Or **0.500 volts** and if we repeated the simulation with the new value we would obtain the identical signal voltage.

27.2 Frequency Limitations Resulting From Impedance Mismatch

In the above example, an arbitrary amplifier input impedance of 50k was used. Some amplifiers (e.g. solid-state devices) might represent this magnitude but vacuum tube audio amplifiers always exhibit *much* higher levels. High impedance levels require more attention to the methods of interconnection.

Dimensional and material characteristics limit impedance levels obtainable in co-axial cables. The range of useful impedances is about 30 to 120 ohms - quite low compared to the impedances discussed above. There are limitations to the frequency response of the system due to the capacitance of the cable length (which is directly related to impedance). *Despite the claims of costly cable*

manufacturers, their products do not substantially differ in any way from the performance obtained by standard, low-cost cables so long as they are similarly constructed.

99.9% of guitar amplifiers exhibit an input impedance between 50k and 1 Megohm. This allows *most* of the input signal to arrive at the amplifier input without attenuating high frequency products (as a result of the low impedance of the connecting cable).

Designers of signal processors intended to be used between guitar and amplifier aren't always consistent in the implementation of good interconnection practises. Intuition, considering the performance penalties/tradeoffs, might suggest that devices placed in the signal path have the following characteristics:

- high input impedance
- low output impedance
- low noise figure
- unity gain (unless the device is intended to overload the amplifier)

(Impedance mismatches, *in small-signal conditions - low voltage/power levels* - won't cause damage to interconnected equipment - just degradation in performance. For higher power levels, as in the amplifier-to-speaker interface, it's very important that the impedances are similar. That's the reason for the output transformer.)

Various aftermarket pickup manufacturers have attempted optimization of signal path deficiencies by lowering pickup impedance. The optimal configuration that results in acceptable output signal voltage, noise figure and low impedance - apparently hasn't been achieved, based on market acceptance.

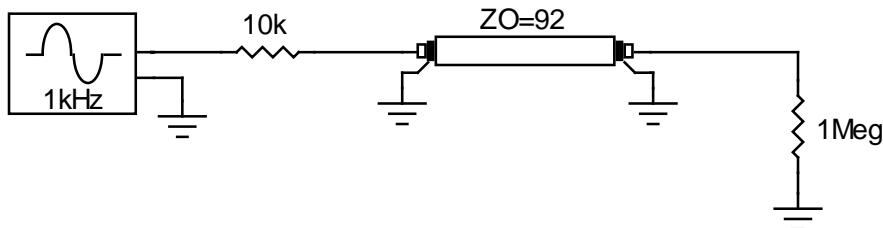
Some guitar manufacturers, both in the U.S. and offshore, have produced instruments with low impedance capability (powered active pickups, usually). Some prefer this type of pickup for the ability to interface effectively with any type of amplifier. I have no direct experience with these types of devices and cannot offer an opinion about sonic qualities. I *can* state that - properly designed - an on-board active pickup could provide solutions to *some* difficulties associated with conventional pickups while creating other difficulties that don't exist with conventional pickups (additional noise, for example).

It's not constructive to overly expand this topic but it is worthwhile to point out simple problems with devices connected between a guitar and an amplifier. Let's employ the same simple example used above: guitar + cable + amplifier. A variation: the source (pickup) impedance will remain the same but we'll change the amplifier input impedance to 1 Meg - typical of all vacuum tube amplifiers.

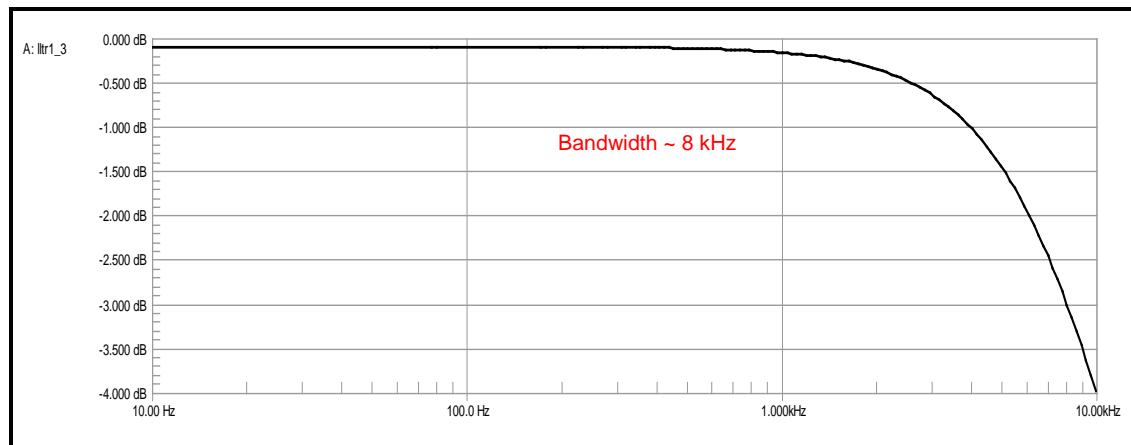
The expression for signal voltage that we evaluated previously will tell us that - because of the large ratio between the source and load impedances - there will

be *almost no signal loss* in the interconnecting cable *if the transfer is independent of frequency*. That's not very helpful since we desire to amplify an instrument that has a fairly wide bandwidth.

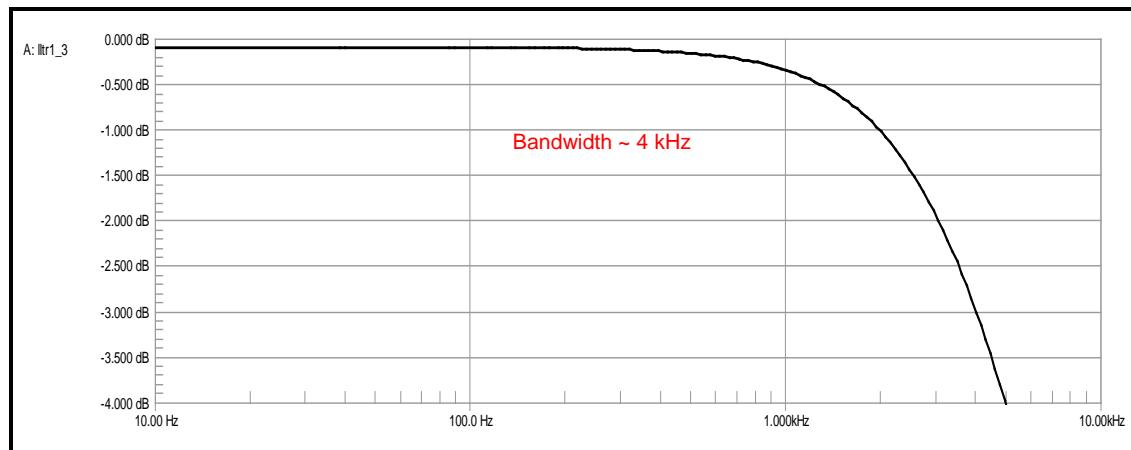
There *is* a consequence to the frequency response of the system, depending upon cable impedance and cable length. Frequency response (bandwidth) is *diminished* by low cable impedance and long cable length. Frequency response is *enhanced* by higher cable impedance and shorted cable length. Referring to the example schematic:



This is the frequency response of the network:



And if we double the length of the cable, the frequency response is degraded by the ratio of the two cable lengths:



Although we've not explored the impedance topic in detail, the relationship of causes to effects should be apparent or at least intuitive. We've discussed impedance effects on input signal loss and frequency response, at the amplifier-to-guitar interface. We observed these effects by use of a simple model: a coaxial cable of varying length.

Inserting more sophisticated circuits than a length of cable has complex overall performance effects. BUT, as we've noted frequently in this book - a guitar amplifier is not a high-fidelity piece of equipment, as a rule.

When large signal behavior is considered - as in the impedance transformation from the output tube stage to the loudspeaker - the effects of impedance mismatch have more serious consequences than those that result from the mismatch between guitar, cable and amplifier.

Increased signal gain (if required by input mismatch) is fairly easy to attain (at the price of increased noise) but large signal power loss affects the most expensive and least reliable parts of the guitar amplifier: power supply transformer, output transformer and the output tubes.

28.0 Appendix: Guitar Trivia

This section is an afterthought - a collection of general thoughts and observations that may be of interest to those who have an abiding interest in our common instrument.

28.1 Binding on Guitar Bodies

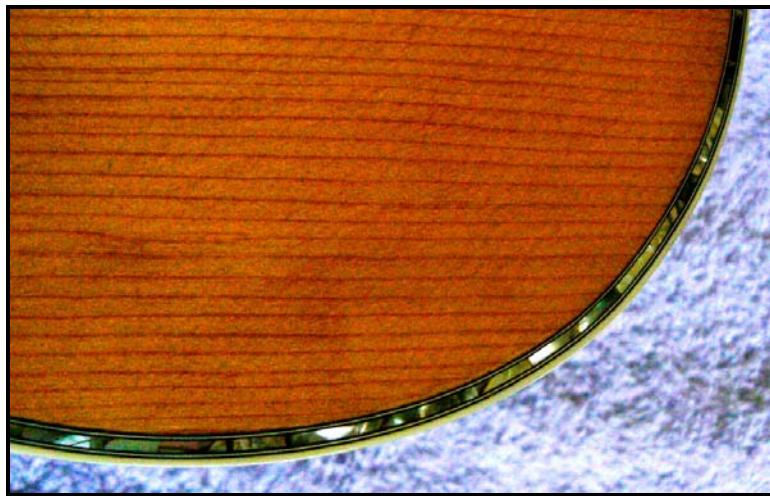
A subject simple as the binding on an acoustic or semi-acoustic guitar shouldn't require more than a few paragraphs to discuss. But there is a history to the evolution of any musical instrument, however simple, that becomes more complex as one considers how the instrument came to its present form.

Each time I write about topics like this one, I find that I become bogged down by detail - that's the disadvantage of "remote" communication. Face to face discussion and the nuances of body language quickly establish the levels of experience and knowledge of the participants. But when one writes, it's safest to assume that readers will range from those with far more competence than me to those who have little knowledge of the topic.

If I have made a relatively simple discussion overly complex, I apologize. I'm proceeding under the presumption that too much information is better than too little information.

Much of what a guitarist perceives as visually "attractive" in an instrument may be the result of conditioning and experience. Tradition drives our perception of music and the instruments that produce the music and tradition is based on practicality: form follows function, life imitates art and other expressions that we have come to accept as truths.

I like the elaborate binding used on some acoustic instruments produced in the 1930's (typical examples would be "herringbone" Martins). Custom builders frequently incorporate complex body binding styles and some mass-produced instruments exhibit very attractive binding. This is a photo of a 1976 Ovation Custom Legend with a body binding of eight plies. The showpiece is the abalone layer that forms a beautiful frame for the aged spruce top, emphasized by the thinner multi-ply binding strips on either side.



Body/neck binding, prior to mass production of guitars, could be an accurate representation of the quality of craftsmanship. Inexpensive instruments had no neck binding and a single ply body binding. Substantially priced guitars manifested the time and craftsmanship invested in them by the usage of multiple-ply bindings and pick-guards.

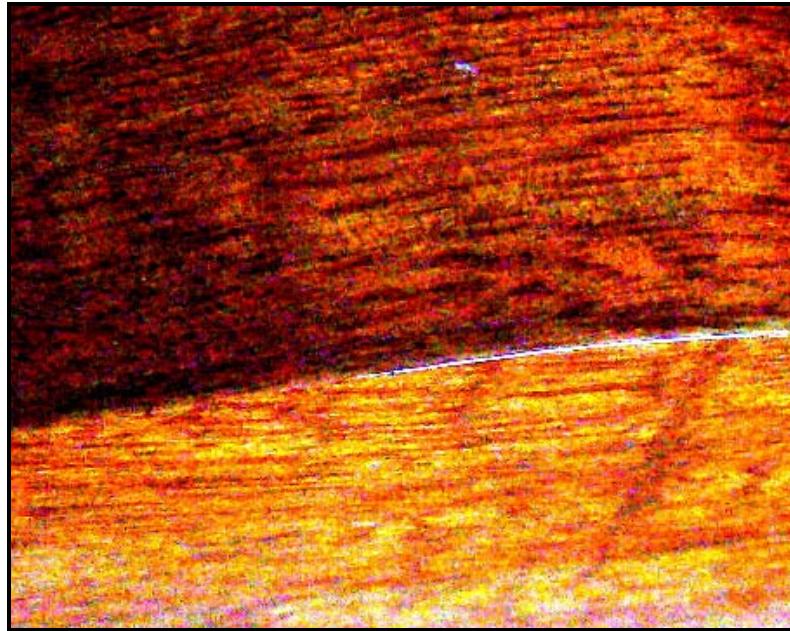


Five and seven-ply body bindings were common in top-grade professional instruments while single, two and three-ply bindings decorated workman-level instruments. Neck bindings were typically limited to a single ply material except for the higher cost guitars which were decorated with three to five-ply bindings, sometimes incorporating a "sculpted" appearance at the end of the neck like this L-5CES (note the five-ply binding on both neck and pickguard):



Different terms are used for the decorative portions of the binding strips and I'm uncertain that I apply them with accuracy, since I do not make guitars professionally. I generally categorize all of the external strips as binding but I believe that luthiers would define "binding" as *only* the strip that covers the exposed edge grain of the top or bottom of the instrument. The additional plies that are largely decorative are referred to as "purfling", I believe.

Here is an example of an unbound guitar body, a 1928 Weissenborn guitar, valued for lap steel and bottleneck styles:



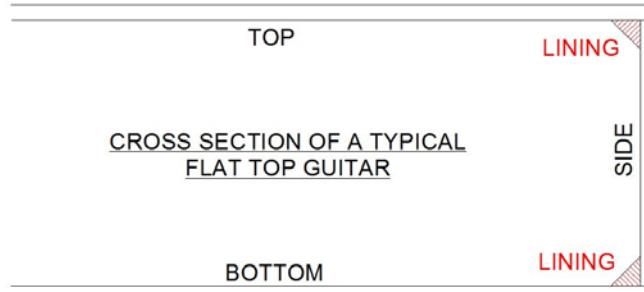
Although I am not very familiar with classical guitars, it's my impression that body binding - unlike most American and European archtop guitars - universally consists of hardwoods (rosewood, maple and ebony are frequently mentioned). The following photograph depicts an inexpensive classical guitar of customary construction (cedar top, rosewood sides) but with plastic body binding - the color match between binding and sides is very good and the binding seam is apparent only in strong light.



There are several practical manufacturing (and structural) reasons for the application of binding materials around the perimeter of a guitar body and none are related to the appearance of the instrument. One might think that this is a simple subject but it's not. Experienced woodworkers will grasp the concepts

immediately but those without that experience may have to consider the reasons for a few moments.

The interface between the top and sides of a guitar is a critical area. The top must be free to resonate freely when excited by the plucked string and yet be securely attached at the edges to a fairly rigid structure. Because the materials involved are quite thin, there is not much gluing surface available for securing the top/bottom to the side. The traditional solution to this problem is to employ a thicker section of wood, usually triangular in cross section that is called the "lining". The lining is first glued to the sides then the top and bottom are attached.



Various styles of lining have been used to allow the internal strips to follow the contours of the guitar sides closely. The most common technique is that of "kerfing" the lining wood. Kerfing consists of making thin saw slots spaced at close intervals along the triangular-shaped lining strips. If the slots are spaced appropriately, the strip can easily be bent and glued to follow the curves of the side. After clamping in place, the glue joint is allowed to cure. After curing, the lining is leveled (usually by sanding) and the top is then glued and clamped to the sides. After attaching appropriate internal top bracing (if this has not already been done) the bottom is glued and clamped to the sides.

One might expect the process to be almost done at this point but there are a few problems. For one thing, the top and bottom of the guitar has exposed "edge" grain (or "end" grain). This is the part of any piece of wood that is the least attractive (it has no "figure") and is difficult to work because of splintering. The edge grain can be visualized by imagining a large number of drinking straws that are glued together along their sides. The edge grain presents a series of very small tubular openings that are difficult to glue and difficult to finish.

In fine furniture and cabinet work, the deficiencies of edge grain are usually hidden by design or "edged" with solid wood trim. The same technique can be used to hide the exposed edges of the guitar top and bottom. The joint described above is first "rabbeted" (usually with a router) which cuts through the top and part of the sides, removing a small amount of wood and leaving a "ledge" cut away from the assembly. This is the space required for the binding.

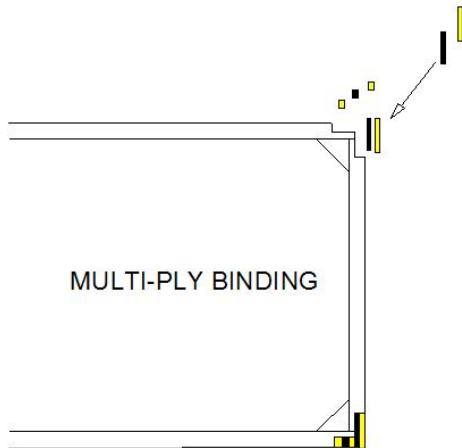


As shown in the second sketch, the binding is glued into the rabbets at the top and bottom of the guitar. The thickness of the individual binding strips is determined by the desired number of plies and the rabbet dimensions.

The binding solves an unattractive problem resulting from exposure of the edge grain of the top and bottom of the guitar. But there is another reason for binding - of especial importance to the individual guitarist. We've all accumulated the requisite amount of dents and scratches on our instruments (the first one is the one that really hurts)!

The tops of high-end jazz and acoustic guitars are usually made from spruce which is NOT a hard wood. Spruce deforms easily and the edges of the guitar are susceptible to damage since these are the areas that usually come into contact with other (harder) surfaces in an undesirable manner. The binding is the first-order protection of the fragile joint between the sides and the top/bottom of the guitar. This protection is afforded by materials selection, either plastic or sometimes a hardwood like ebony or rosewood.

An alternate technique for wider decorative binding strips might require thicker sides or perhaps two passes of the router as in the following sketch:



By making a second, shallow pass of the router and using two different thicknesses of binding, a wide decorative pattern can be achieved. A single router pass would cut through the side and into the lining, not a desirable situation. This is the technique employed on the 1976 Ovation illustrated at the beginning of this chapter. (The Ovation has the widest binding of any of my guitars, including the L-5 and the Golden Eagle.)

As a matter of interest, body binding width of various guitars frequently reflects the method of construction. Here are typical binding widths for some of my archtop guitars:

Ovation Custom Legend:	0.20 inches wide, 8-ply
Gibson L-5CES:	0.18 inches wide, 7-ply
Heritage Golden Eagle:	0.18 inches wide, 7-ply
Gibson L-4CES:	0.16 inches wide, 3-ply
Epiphone Emperor Regent	0.12 inches wide, 5-ply
Gibson ES-135	0.09 inches wide, 3-ply
Guild SF-IV	0.09 inches wide, 3-ply
Heritage H-575	0.07 inches wide, 1-ply
Gibson ES-330	0.07 inches wide, 1-ply

28.2 Neck Binding

Many guitars do not have neck binding. It has traditionally been applied to more expensive models because the binding process is hand work and therefore costly. With the huge influx of Asian guitars into America and Europe, we see binding applied on even budget guitars - there is almost no cost impact because factory labor is about one hundred times less expensive in Asia.

There is no edge grain to disguise on a guitar neck but there is an interface between the fingerboard and the neck materials on most guitars. Depending on the consistency of the manufacturing process, the appearance of the joint might

be enhanced by edge binding. (Note that solid maple neck guitars rarely have neck binding because there is no separate fingerboard.)

However there is an argument for including neck binding if one is concerned about protection. The edge of the fingerboard is an area prone to damage - it's likely that this area will come into aggressive contact with microphone stands and the like. On a crowded stage, a brief body rotation could create one's worst nightmare: a confrontation between guitar fingerboard and a cymbal.

The more durable plastic binding is better able to withstand this type of contact while permanent damage would be likely to occur with an unbound neck (in the form of a gouge or dent). In the worst case, where the binding is seriously deformed, it would be far less expensive to replace the binding than an unbound, complete fingerboard.

Installing frets on an unbound neck is a very simple process. One taps (or uses a mechanical press to force) a few inches of a long length of fret wire into the pre-cut slot on the fingerboard then snips off the excess. This procedure is repeated until all frets are installed. Then the excess at each end of the fret is carefully filed smooth and flush with the neck.

For bound necks, this process becomes considerably more complex. As with body binding, the process requires rabbeting the previously glued neck/fingerboard joint to create a ledge for the binding material. The binding material is glued into the rabbet and scraped or sanded flush with the neck and fingerboard.

Unlike the unbound neck, where a length of fret wire can be tapped (or pressed) into place into the fingerboard slot and snipped off ... the remainder of the length of fret wire tapped into place in the next fingerboard slot and so on. The bound neck requires that each individual fret be first cut to length.



The "tang" of the fret wire (the portion of the fret that is forced into the fret slot of the fingerboard) must be removed from the fret at each end. In the above photo, a close look reveals that the tang has been clipped and filed or otherwise

removed from the ends of the fret so that the tang will not interfere with the four-ply neck binding.

This fret has been cut to rough length and will be filed and sanded so that it is just slightly longer than the neck width. Each fret will differ in length due to the gradual taper of the guitar neck. It's important to leave a slight excess at each end so that after installation, the fret can be carefully filed/sanded flush with the binding.

This is a *very* finicky operation and explains why bound necks are uncommon in budget-priced instruments except in guitars made in Asia, where labor cost is not a significant contribution to the selling price.

Some guitarists consider bound necks to be indispensable - *I always thought that I did* until I realized that many of the guitars that I play frequently do NOT have them. In fact after counting, I find that there are seven guitars that I routinely play that have unbound necks. There are two reasons for preferring bound necks over unbound necks: appearance and the protection binding offers to the edge of the fingerboard.

28.3 Headstock Binding

The application of binding to the headstock of a guitar is a protective measure, to prevent the sharp edges of the headstock from inadvertent damage, as described previously. However this is probably secondary, the primary purpose being an indication of the quality and workmanship presumed to be applied to a high standard instrument model. Here are two examples, the first is a Heritage "Golden Eagle" and the second is a Gibson "Wes Montgomery" L-5CES.





Note that some of the decorative aspects of binding, especially the abalone inlays shown in the first illustration of body binding, is not noticeable more than a few feet of distance from the guitar. Although this might seem to be the application of considerable effort for very little reward, embellishment of costly instruments was never really intended to visually impress an audience.

The main reason for these visual features was to provide an intrinsic reward to the artist that purchased the instrument: pride of possession. It is an indisputable fact that better work is produced with the use of finer tools. Part of this is the functional aspect - the "utility" of the tool. But much is also attributable to the value and pride that the musician takes in his/her instrument.

This may not even be detectable by an audience; anecdotally it's frequently suggested that great artists can make junk instruments sound like those costing many, many thousands of dollars. I wouldn't care to argue this point as a listener but as a guitarist, I can state without equivocation that my best efforts are concurrent with the quality of the instrument that I am playing. That has been my observation from age 15 to age 65.