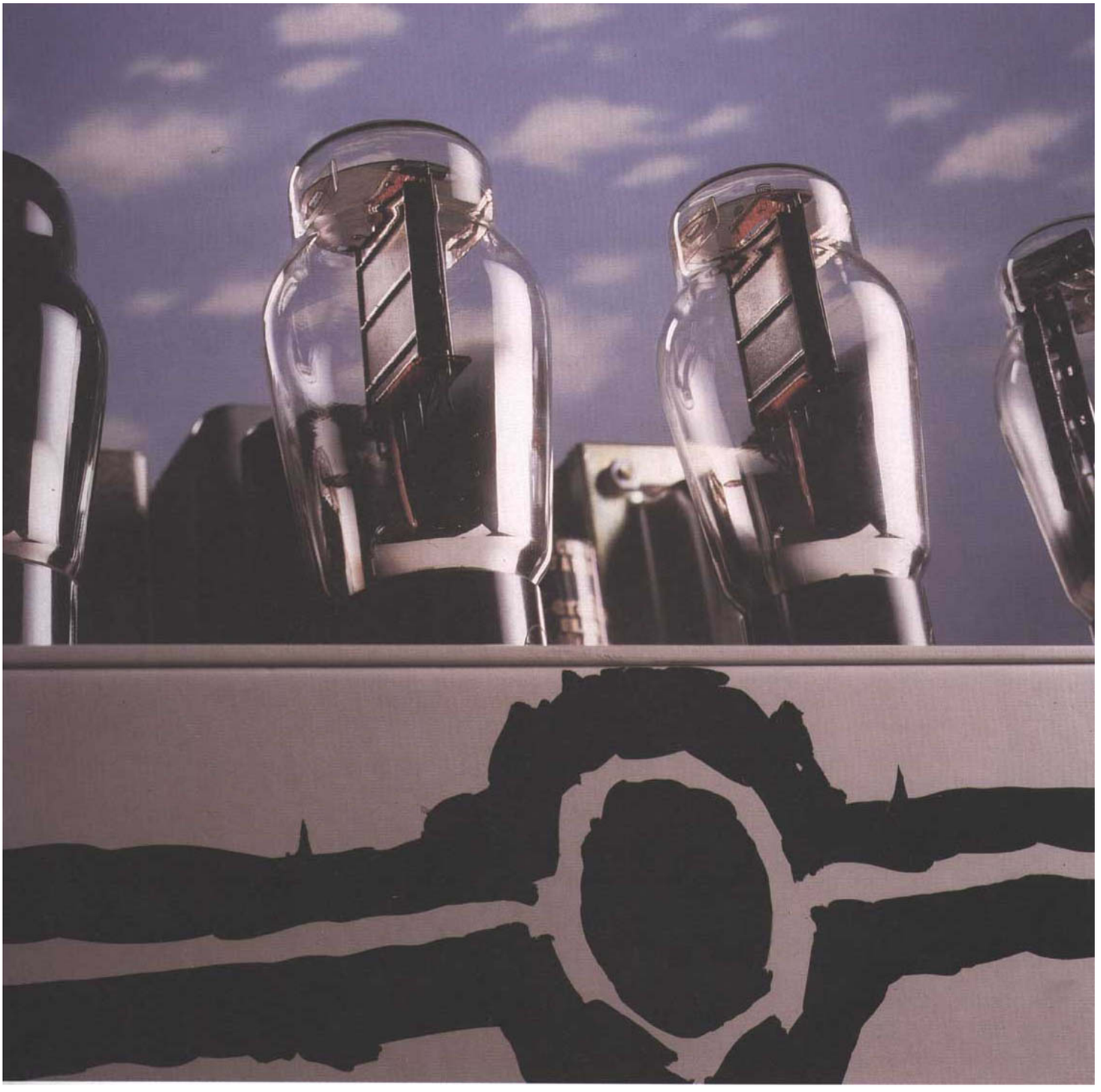
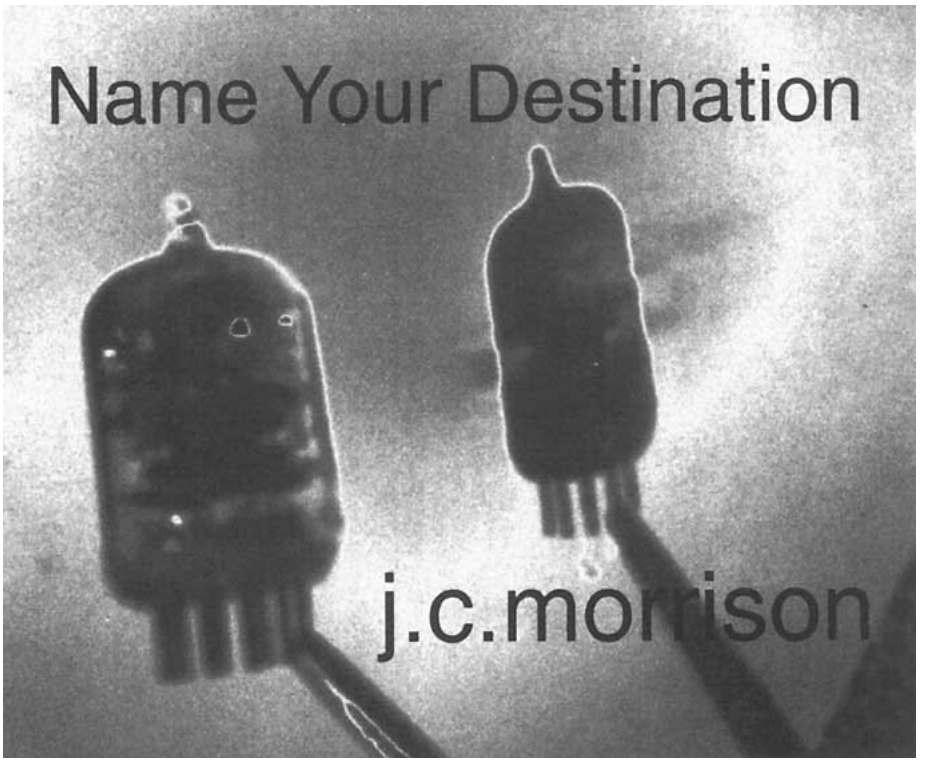
SOUND PRACTICES

THE JOURNAL FOR AUDIO EXPERIMENTERS FIVE DOLLARS



Innocence and Experience Issue 14



I'd like to begin with a minor confession: I am a cynical bastard. Its a full time job, let me assure you. The extra work comes about because of the effort required to navigate around the most fundamental suspicion: that humans fu*k everything up, especially the remarkable. Now, I have not simply jumped to some arbitrary conclusion here! And it is not only because of my tiny mind, just a natural order of things in a human world dominated by fear, ignorance, and heavy competition. Who can actually afford to deeply care about anything extra right now, other than James Watt's big three important necessities? Some of us, perhaps, are actually forced by circumstance to place the pursuit of "Pampers" or similar other goods and accessories, ahead of a beautiful ethical life...

Whatever the root causes might be, the result is the same. Just as you are becoming aware of the mysterious and terrible beauty of the universe, some Joe-neckbone is going to throw his leftover Bob's Bigboy out the window of his rapidly moving Camaro in such a way that the most spectacular light you have ever seen in August is permanently darkened. So when something magnificent comes along, whether or not it has anything at all to do with making the rent, or binging and purging, it takes work, at least on my part, to make sure I am present for it. And I do generally try, because that kind of experience is precious, utterly valuable.

In light of this, I would like to suggest that the audio recording medium is an amazing thing. It is possible to bring home, right now, from both the Sistine Chapel and any one of its many charming present day counterparts in Los Angeles, the ghostly echoes of the two great castrati of our century: Allesandro Moreschi and Michael Jackson. The immense distance between the two contexts, the 1902 passing of a strange and decadent tendency of "classical" European culture and the equally complex pop construction that has replaced it, doesn't present any obvious difficulty for us "listeners" today. Most of us just don't care about why or what these traditions mean in any original sense, we just use the music for our own base motives. We use it to eat to, to seduce our lovers, we collect it as objects, as a soothing emotional wallpaper, or to make us seem sophisticated or fun-loving to those we hope to impress, maybe even dance to, blah, urp, etc....

Delivered under the proper circumstances, however, many of us are still able to find something remarkable, perhaps a spiritual thread, in recorded music that can occasionally lift us above the pressures of ordinary existence. Just don't assume I mean to use "spiritual" in a religious sense at all. Charles Manson's ego deflating "I am we and we am I" is close enough. That or, "the things that make you go hmmmm... (reprise)".

As a side-note, it is still surprising, even for a jaded punk such as myself, that actual contact with a musician has fallen so out of favor, in comparison with the more easily handled electronic surrogate. That extra "whatever" that differentiates musical magnificence from normal routine is apparently more easily consumed at home. But, the topic for today is the amazing EP/LP/CD/tape/DAT/Minidisk/Laserdisk/8-track(!). What the hell is it really?

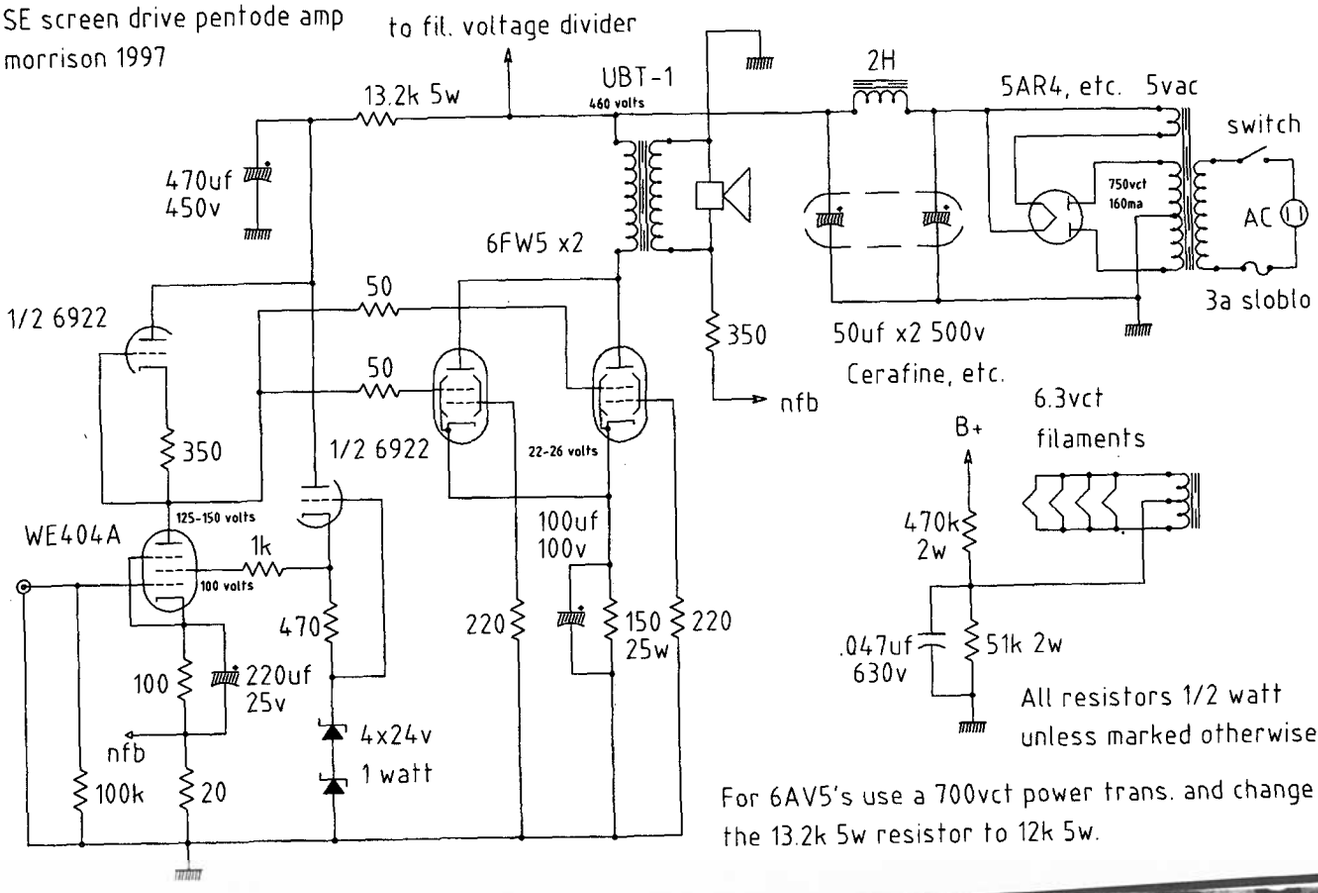
Recorded music is fundamentally different from live music. The core difference, as I see it, is its virtual nature. Music has done gone vaporized! The very word "music" no longer means "that which is performed by a musician", actually most people think immediately of a compact disk. It is no longer necessary to dress up for the opera or have a duchess aunt to get down with the male sopranos. In fact, there is no need to know anything or anyone to immerse yourself in the sacred music of the tribes, violin or accordion virtuosos. You can even splice it into your own funky slice (I do love the Beastie Boys...) with a minimum of musical know-how, equipment and taste. We can, in fact, beam down to any salable musical ethic or tradition at will. The only thing one actually needs is a spare $16.99.

This virtuality has obviously changed the way we use music, but it has also changed the way we judge it, make it, market it, buy it, and HEAR it. The problem is that the hardware/software we rely upon for "playback" and the context in which it is employed today underlines the experience with such complete transparency, most people have forgotten that it's there.

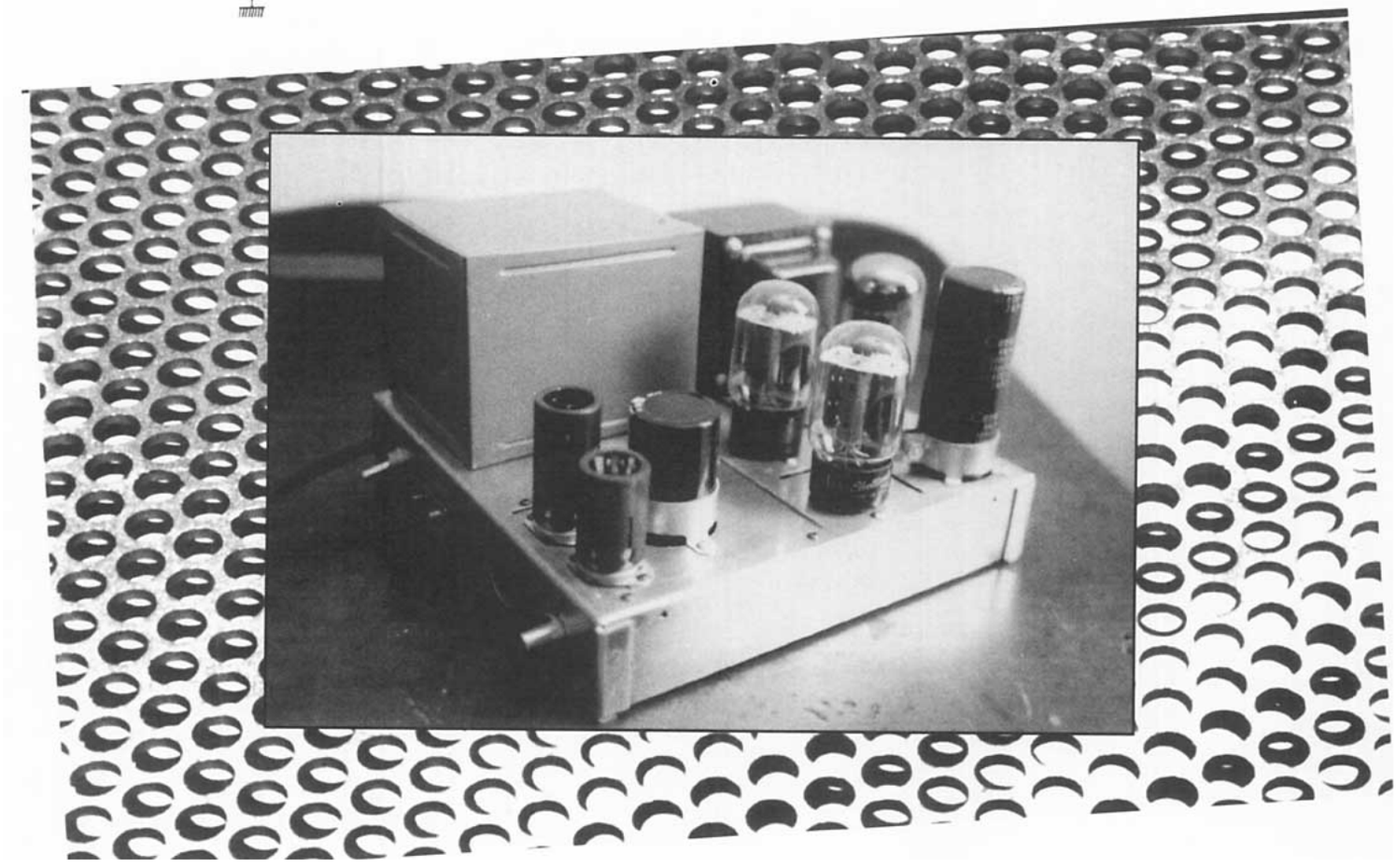
As I mentioned above, for most people, the "ware" IS music now. But, there's a lot more! The "space" of the recording is no longer any place, it is all places and any time. Name your destination!

Actually, none of the musicians on any given recording need to see or even hear one another and the place they perform is a small beige and blue foam lined cubby. Let me try again.... Audio space is a movie set for a fantasy fiction action hero, a "transformer". The recorded elements are discretely equivalent animated props, tape splice, sequencer and warm body alike. Add your favorite recordings to a ghetto blaster or "Walkman" and you can also transport it anywhere and into just about

SE screen drive pentode amp
morrison 1997



For 6AV5's use a 700vct power trans. and change the 13.2k 5w resistor to 12k 5w.



any context. Bring the "Metallica" into the most menial workplace, or to Mecca (all roads...) or how about the High Sierras? Any experience, sacred or profane, can be accompanied by your most beloved chamber orchestra.

Without making any sort of value judgment about this, it is interesting to me that, unlike film (which has experienced a parallel history in many ways), either the suspension of disbelief has been total or very few have thought to question it at all. The initial use of recording was very much like taking a picture with a camera, or making a newsreel. Listening to the Moreschi at the Vatican (old eunuch singing into the horn of an Edison cylinder recorder) you will hear this clearly. Even I would think of it today as "bush recording" (recorded in the "bush"—as Don Van Vliet dubbed it), as opposed to the studio stuff, which is somehow more plebeian. *Audio verite* vs. Disneyland. In any case, the site at which we meet our music is a slippery place.

Another facet of this concerns recorded time. Whereas we easily enjoy, expect, swallow or sleepily ignore all sorts of spatial audio special effects, there is today one homogenous monolithic concept of timekeeping. The time in which music, as a whole, has become measured by is 4/4 (four beats to one measure). Even if someone were to make a record of waltzes today, its commercial novelty would be derived from the inability to count "1..., 2...,3..., 4...". At ninety beats per minute, by the way. The smallest deviation from this formula banishes a recording to a fringe market, or the cut out bin, pronto. Even the most viciously angry gangsta rap and thrash conforms.

All this is because American pop music, the dominant form worldwide today, which exists in that form because of the recording process, has overwhelmed all other traditions to a level where anything else is forcibly manufactured and listened to in relationship with it, whether intentional or not. Pop really does rule.

Back in the early days of recording, because the mechanical 78 fit just 3-1/2 minutes of "music" on a side, popular forms and dance music adapted much more rapidly to the new "reality". The technology gave an edge to fast entertainment, not because of any obvious idealism or value judgment, but simply because of the limits of what could technically be done at the time. The classical "hit" actually dates from the beginning of the century where a short popular sequence of operatic showmanship could sell a million copies. Popular stylings still had an advantage over classical, "modern" or ritual music. By the late 1950s, Buddy Holly had a "gold" record with 22 "tunes" on one LP! It's also a great record, milestones notwithstanding. Today, the arrangement, techniques and tools developed for the pop music market have affected all recording regardless of intent. A short attention span helps.

Multitrack recording, for example, has become a formula for nearly all music. Many engineers and many musicians could not imagine doing without it. This one tool has had important effects on the way live music is perceived, although I don't mean to imply that these are entirely negative. It does seem that live is too "unreliable" for many people, both for industry types and the average listener (including many audiophile reviewers!). It "never sounds like it's supposed to", like it does on the record or on MTV.

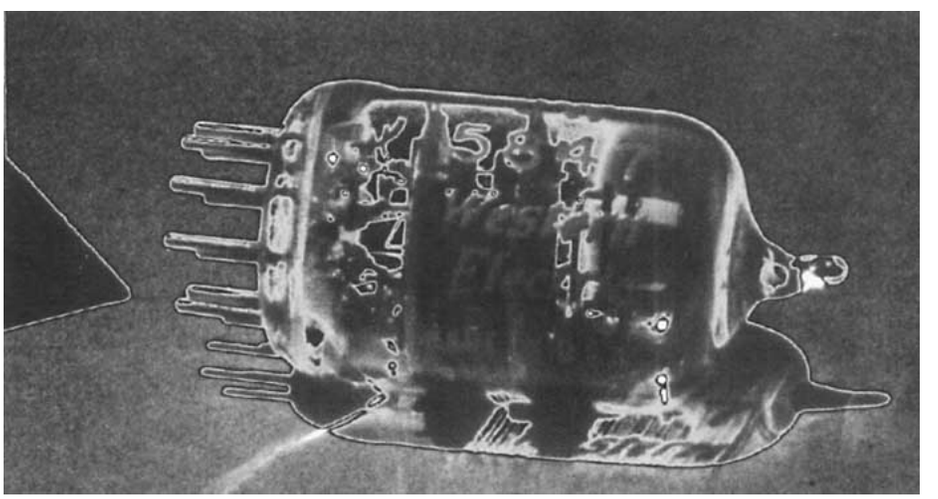
Not to mention, musicians are such a debauched and irresponsible collection of exploitable trash anyway, and why should record company people have to worry about them showing up late, intoxicated, or not at all? Or, why should a recording company, who pays for an accomplished classical orchestra at Carnegie Hall, have to put up with super expensive mistakes on the part of the talent?

Thanks to the multitrack studio, it is not at all necessary for the musicians to be able to play their instruments, much less "perfectly", or show up on time. Fix it in the mix! Absolute "perfection" and control can be had over the musicians and their efforts, to the point where the producer has become the key creative node in the music network. Remember the "Monkees"?

This is not simply a bad thing either. It has also led to a new category of composer who needs nothing but a multi-channel digital sampler, a sequencer and a mixing board. It has contributed to a condition of general mediocrity, but there are those who have become real lo-fi savants, like Beck, or Sir Mixalot, who can't "play" for shit but splice up some really good music anyway. And there is a whole mature genre of *musique concrete* that relies not on the individual virtuosity of musicians w/instruments, but upon the editing of pre-recorded sounds and textures. The hardware floats underneath someplace. I want to be fair here, the trend, however, is that it always sounds the same even when it's totally different! Amazing!

Like I said, I'm a cynical bastard.

Having said all this, and despite the fact that the recording has changed our relationship with musicians, with musical tradition, with the architectural space where it all happens, with the quality and quantity of it, and with the clothing we wear and the positions it assumes in our domestic and sexual lives, and how much we pay and who gets it, it is possible to get connected to the vital element we believe music contains with recordings. Occasionally, we even experience something remarkable.



One might think this a strange way to begin an article about amp design, but I never expected anyone to take me seriously in the first place!

This particular project will challenge some currently fashionable design dogma that I am in part responsible for. That's key. I would like to present a pair of designs that use common as dirt beam tetrodes and are parallel single-ended arrangements. Oh yeah, these amps will also incorporate the use of negative feedback. They both use inexpensive stuff and I have gone out of my way to eliminate as many opportunities for tweaks as I am able (direct coupling and minimal circuits). These amps also have shocked me, and others too, for how good they work and sound. You might consider these little guys as an elegant sequel and a performance upgrade to the "morrison micro", which fulfilled a very similar musical compromise.

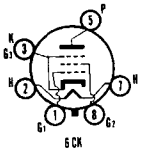
I became intrigued by screen-drive tetrode/pentode operation several years ago while I had the good fortune of borrowing one of David Berning's PP amps from my friend, Mike Trei. I used this amp to drive my bass cabinets (because I sold off my own bass amps to make the rent...). Despite the class B operation (which I remain underwhelmed with...), it worked pretty good on the low end! It kind of sucked on complex material through the midrange, so I never seriously considered using it full range, but even so, it was a sweet sounding forgiving amp with great bass. I understood that this amp used a screen driven pentode output stage and that perhaps this contributed some of its positive aspects.

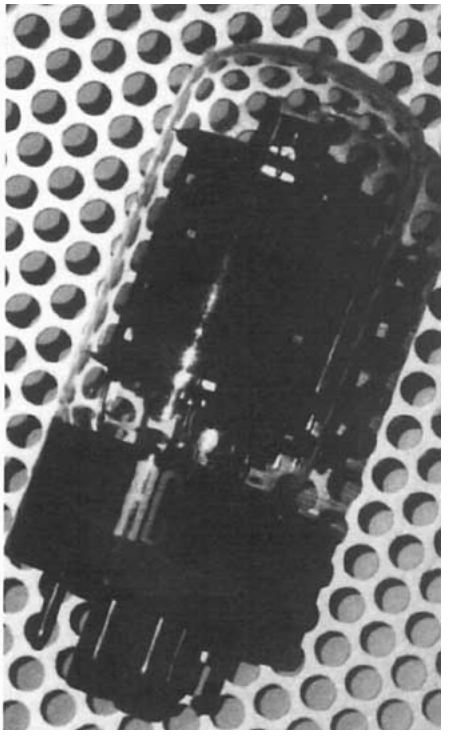
One can actually drive any and all of the control elements in a multi-grid tube. It changes the transfer characteristic radically and not always for the better. "Screen-drive", or the method of applying a control voltage to the screen instead of the control grid, is an old trick that RF engineers used as a high quality method to modulate a carrier signal with CW or voice. It's basically a form of quasi-triode operation with some options. I'm not entirely sure how Berning got a patent on it back in the day, because there is a large amount of prior art in this case. However, no question at all that he was the first to specifically apply it to commercial audio amplification. With the right tube the advantages are, 1) extremely linear operating points w/o feedback, 2) ease of use in simple dc circuits, and 3) unusual flexibility in modifying the performance of an active stage (lots of opportunities to splice in feedback or filtering for specific purpose with the extra grids and cathode).

My main gripe about the way Berning applied this arrangement was the class B operation and an overly complex driver stage (that relied on feedback to get its reward), which in my opinion was responsible for the less than remarkable performance of this amp in the middle and top. Lots of feedback made the bass excellently tight. I think Berning did a very good job (all those many years ago...) at a very reasonable price. However, I was interested in whether or not "screen drive" might be pushed beyond this amp.

Just to touch on a subject that I have harped on about in other articles, I would reiterate that you can't fix the past. Feedback, not anywhere near as evil or as useful as folks generally seem to spout on about these days, is only really helpful (in sonic terms) in those amps that don't really require any to work well. It has a vital place as an impedance transforming technique (for pentodes, this is crucial) and as an aid to retaining ac/dc balance over a period of time in a PP or DC coupled amp, both of which are important applications. Its usefulness in reducing amplitude distortion and phase shift is completely overstated, and the cost of performance complexity and radically worsened high order distortion spectrum and poor high frequency stability is too great, especially in an amp that clips all the time. Those things all relate to good sound.

SYLVANIA TYPE 6AV5GA
12AV5GA
17AV5GA
25AV5GA
BEAM POWER AMPLIFIER





Now, all power amps clip constantly and NFB makes it BAD. Especially very fast clipping transients. Better to design right for the open loop pre-feedback condition. If feedback is to be used it is NOT primarily for error correcting, but to alter the functional behavior in a useful way (such as output Z or for filter networks).

It has not been my experience that 3 to 6 dB ever hurt a well designed triode amp, and did offer modest improvements. Plate-loaded pentode output stage amplifiers must use NFB, but even here there is a proper amount, NOT the rote 20db most tube-amp designers rely upon.

In any case, a severely bandwidth limited triode amp, with a shitty power supply or output transformer, almost always sounds better without any, than with, although why anyone wants such an amp is a good question on its own (I sense the Asano ninjas drawing their weapons...)??? The standard EE argument about efficiency of operating point and use of feedback to error correct can be thankfully ignored in a DIY situation, so I won't waste any more of your time.

The first amp circuit I want to introduce you to (schemat. #1) is a dc screen-drive amp built around the 6AV5 or, its slightly bigger sister, the 6FW5. These octal tubes are some of the most unheralded groovy little power tubes and I just can't imagine why. Triode wired, they compete with anything. Running as beam tubes, they kick butt. The only reason I can imagine why the cheapskate audio world hasn't latched on to them is that they have "TV tube" stigma. I don't have any problem with that! And, because they were commonly used as horizontal amplifiers from the black and white age on, there are droves of them at very fair prices. General Radio, by the way, recognized a top performer and extensively used them in their test equipment.

I am so exhausted from the sound of politically correct propeller-heads expounding upon the virtues of single-ended amps, that absolutely must employ a directly heated triode w/o feedback, etc., that I feel like vomiting. I know that I am in part responsible for this and I will continue to build amps around them no matter what. But still, I feel that the point was missed

AMPEREX TUBE TYPE 5847

The 5847 is a high-gain, miniature pentode designed for applications in broad-band amplification where its high figure of merit is required. It is a plug-in replacement for the type 404A. The 5847 is intended for replacement purposes only in existing equipment. For new equipment designs, the Amperex type 6688 is recommended. The 6688 has a different base pin arrangement and higher transconductance.

5847

GENERAL CHARACTERISTICS

ELECTRICAL

Heater Voltage		6.3 volts	
Heater Current		300 mA	
Direct Interelectrode Capacitances	Without External Shield	With External Shield (EIA #315)	
Grid to Plate (max.)	0.05	0.04	μμf
Input	7.0	7.1	μμf
Output	2.5	2.9	μμf

Characteristics & Typical Operation

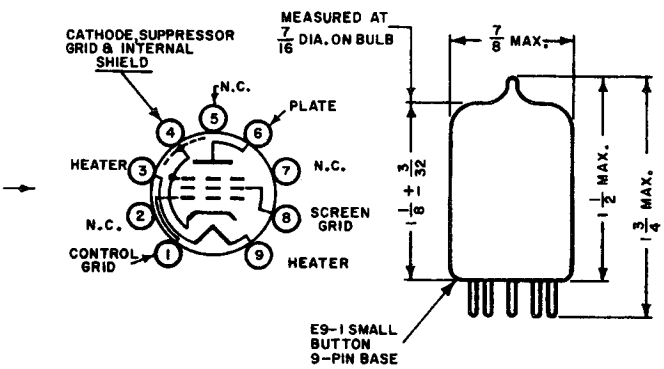
		*	
Plate Voltage	150	160	volts
Screen Grid Voltage	150	160	volts
Cathode Bias Resistance	110	600	ohms
Control Grid Voltage	—	+8.5	volts
Plate Current	13.0	13.0	mA
Screen Grid Current	4.5	4.5	mA
Transconductance	12500	12500	micromhos

MECHANICAL

Base	Small button, 9 pin
Max. Overall Length	1¾ inches
Max. Seated Height	1½ inches
Max. Diameter	7/8 inche
Mounting Position	any

MAXIMUM RATINGS (Design Center Values)

Plate Voltage	180	volts
Plate Dissipation	3.0	watts
Screen Grid Voltage	180	volts
Screen Grid Dissipation	0.75	watt
Cathode Current	35	mA
Heater-Cathode Voltage	50	volts
Grid Resistance	0.1	megohn



* Operating conditions to minimize gain variations due to supply voltage fluctuations.

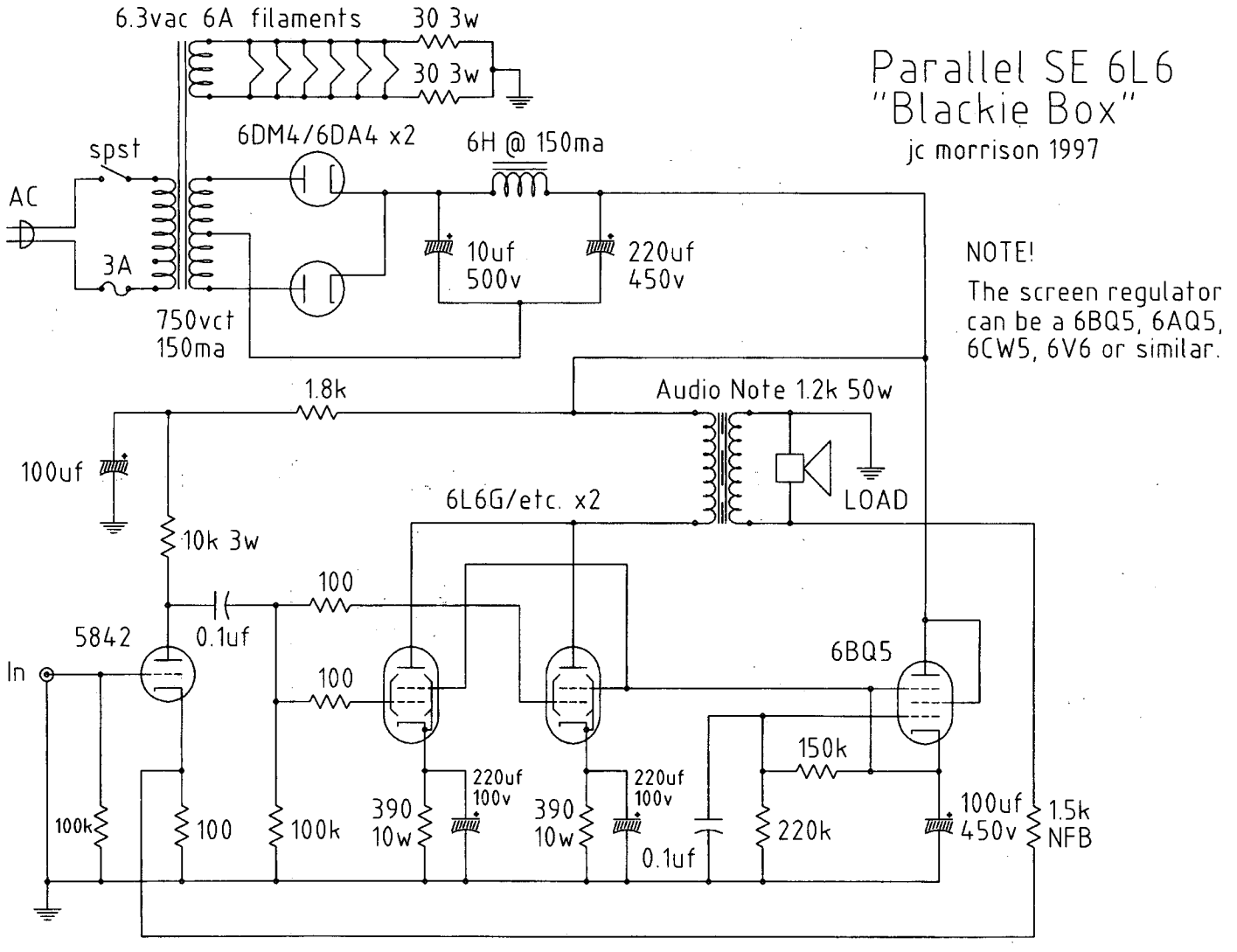
somewhere along the way, namely that it isn't the stuff, the tubes, the circuits... its the attitude! Well, just for you SP readers, I had in mind to come up with some smoking good little multi-grid amps that didn't cost much or require a degree to complete. Brown clothing for mono operation is optional.

The output transformer is the One Electron "UBT-1", which has to be the under $100 red hot deal at the moment. There are no coupling caps so I don't need to hear any crap about which ones are better or worse. In my experience, caps never made as big of a difference as did many other things. And there are so many good caps now! Besides my beloved REL polystyrenes, there are Hovland and Audio Note and MIT. As far as power supply caps go, I used a Cerafine dual because it was the only 500v cap I had handy and I couldn't fit stacked Panasonic TSHA's in the Dynaco Mk III chassis I gutted for use with this circuit. I absolutely recommend Panasonic TSHA's for all power supply applications. You can get them from DIGI-KEY. I do not bypass large caps, if they don't do what I want at hi freq., I don't use them.

The most important tube in any amp is the first one, and in this case I chose the Western Electric 404A/5847. My brother, Herb Reichert, introduced me to this tube 8 years ago when he built a phono stage around it. It was too microphonic (rattling screens) and he gave it up eventually, but not before we had heard this magnificent sound! I have been wanting to do a higher level circuit with a 404A ever since. These tubes were used in the famous TD-2 VHF repeater (along with the 417A), which guarded the free world from the red menace prior to the satellite age. They are meant to be run at high currents (read wide bandwidth) and can work well enough open loop in no-feedback audio amps. That is unusual for pentode voltage amps. If you have trouble locating these tubes, other candidates are the 6AC7, 6688, 7788, 6CL6 and 12BY7. Any one of these can be adapted easily to this amp, and while different, all will certainly sound great.

The first stage uses 1/2 half of a 6DJ8/6922 as a plate load for the 404A. I used the smallest cathode resistor I could get away with (without losing all gain from the lower half) on the 6922. It occurred to me that a 6AQ8 or a 6BK7 might allow the same current, at a higher impedance and at roughly the same capacitance, for more gain from this stage. Since the amp worked out, I didn't spend the time for checking it out. This would be the one change I would look into now because there are a few more watts to be squeezed out of the amp and the sensitivity is on the medium to low end of the scale. For a CD only system, the amp works really great as shown.

You can see that the stage looks vaguely like an "SRPP" arrangement. The reasoning for this was to keep the first stage output impedance as low as possible so as to keep



the bandwidth up. The Z out is the plate resistance of the 404A in parallel with the total source impedance of the 6DJ8 + cathode resistor combo. In comparison, a 47k plate resistor knocked nearly an octave off the bandwidth (although the gain was higher...).

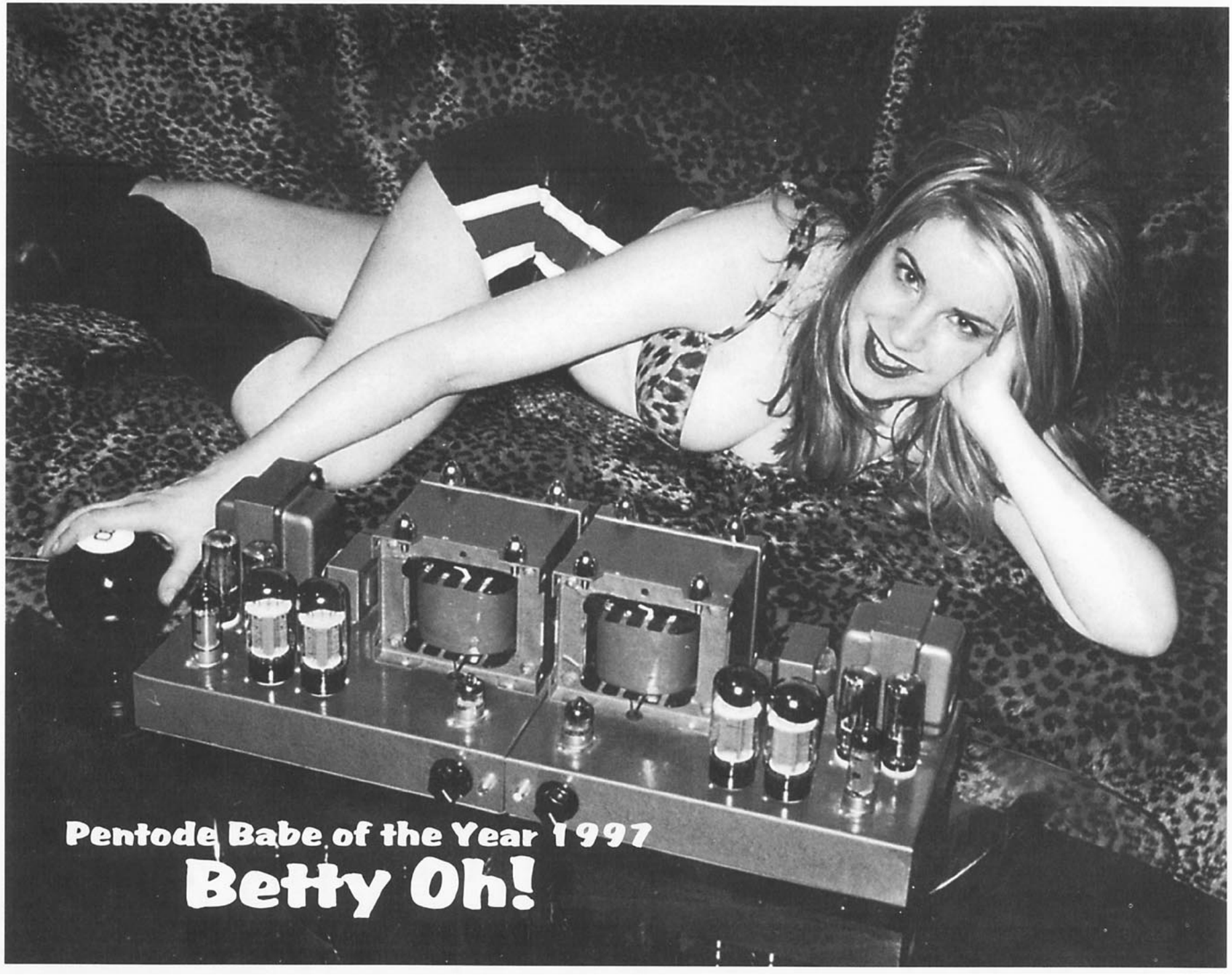
I adjusted the biasing of this first stage to get the best X/Y phase trace for both stages ahead of the tranny at 30 kHz (a modest broadening, at most a 1/16"). I take this to mean that the driver stage behaves itself until a bit less than 300khz (one order of magnitude). The output transformer then represents the primary high and low frequency poles in this amp and, for the most part, characterizes the behavior (because the voltage amp and output stages are much wider bandwidth).

The UBT-1 is remarkable for such an inexpensive unit and the small amount of feedback used causes no significant problems until 160khz. The small amount of overshoot seen on higher frequency squarewaves is due to the phase turning around up there. There are amps for sale for big dollars that are much worse.

Now, I don't really care for "SRPP" both for how it generally sounds and from my own engineering aesthetics, but it accomplished what I needed here and I admit it. I used the other half of the 6922 to regulate the screen voltage on the 404A. The grid-stopper is to bandwidth limit the regulator voltage so I didn't need to bypass with a cap (zener crap). Regulating or separately supplying the screens is always a good idea with a pentode in my experience. The point here is to get 125-150 volts on the plate of the first stage, 100 volts on the screen, and use the plate potential to correctly "bias" the screens of the 6AV5/6FW5s. One of the many excellent features of this tube family is that it will draw current at low screen voltages. A 6L6 type tube will need something more like 225-250 volts on the screen to pull enough current.

The control grids of the 6AV5/6FW5 are simply grounded and the cathode resistor is calculated (in this case...) to drop the intended bias voltage for safety. One note about this: I am over-dissipating the tubes quite a bit. The normal Pd of a 6FW5 (the tube used in the amp) is 18 watts. I am running them at 25 watts each. No problem. At roughly $3.50 a tube, I do not lose sleep. Nearly a year later, no problem. But I held back from grounding the cathodes directly, which can be done in a screen-drive circuit with properly adjusted screen voltages, because I intended from the start to direct couple and burn the tubes up for my own twisted reasons.

If you use 6AV5's you will have to limit the B+ to 400 volts instead of the 490 I used here. I would limit the dissipation of 6AV5's to 20 watts each. Among the other great things about this tube is the low grid to plate capacitance. This makes parallel


Pentode Babe of the Year 1997
Betty Oh!

operation much less problematic than say a pair of bi-plate 2A3's. Parallel operation of vacuum tubes does NOT have to be a problem if you pay attention to the way you drive them.

The amp put out 10 watts at 5% THD, 8 watts at 1% (HP 334A). My crappy Heathkit signal generator is responsible for some of that. Obviously, this could be improved upon with a little more gain out of the first stage, mostly just for more power. But the sound of the amps was gorgeous. Very un-electronic. I brought these amps to the Philadelphia Triode Show back in the early spring, they didn't get much attention (they were auditioned briefly at the end of the day) but I was told, by those that heard them, that they were real contenders against the many 845 amps that were there. Obviously, I thought that they blew away everything except, of course, for big daddy Komuro's dc 300B (w/permalloy outputs) amps, still one of my all time favorite designs.

The next amp is a parallel single-ended 6L6GC type amp, which means that you can use any of that family, even 807's with a socket change. This amp has a good story behind it. I have a friend named Blackie, who is one of the top guitar amp technicians in NYC. He serves your favorite rockers, all of them. Anyway, we've been hanging out for some time and at one point, the topic of building an amp for his home came up. Blackie asked me, "What would be the most unpopular and out of fashion hi-fi amp I could build?" That was easy, thinks I, "A parallel SE amp with guitar tubes for output devices." So we designed it around the Western Electric 417A/5842 (again, from the cold war arsenal...) and 6L6 family. Other features include a 6BQ5 screen regulator and TV damper diode rectification (my fave...). This amp sounds so good, it continues to impress. It absolutely has enough guts to drive most real world speakers. The schematic (schemat #2) is very straightforward without tricks. Blackie built it into a lean green hammertone machine. His wife, Betty, approves and insists upon single-ended pentodes for all her serious playback.

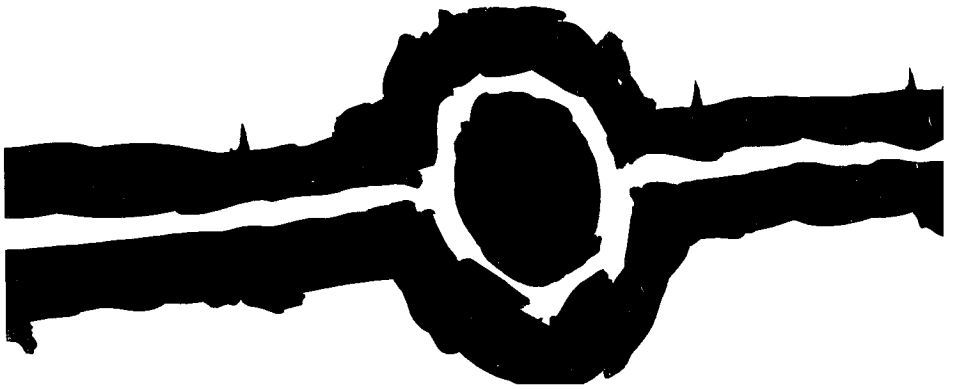
Blackie, by the way, makes a mean guitar amp. It's a Class A PP 6550 head, with glow tube regulation for screens (dual OD3's), a trick tone section, and a unique appearance (Plexi-glass cage and bottom cover that allows you to see the regulators and his gorgeous point to point). He can be found at "Tubesville" 153 Ludlow St. New York, NY 10002 tel. 212-529-7345.

Back to the amp, we selected the Audio Note 1.2k 50w single-ended output trannies, which turned out to be excellent performers. The power transformer and the chassis are Hammond units all purchased from Angela Instruments. The result is compact and practical. The regulated screen supply also did make a big difference in the low end performance. NOS 6L6GC's are not all that easy to track down these days, but apparently it is still possible to get 7581A's, which are the same thing. These are 35 watt dissipation tubes and you can bias them all the way up. The cathode resistors I specified are really conservative. They could be reduced to 250 ohms if you are daring and have the "GCs" (forget the Chinese 6L6's for this amp). Sovtek 5881's, as well as the rest of the 6L6 family, can also work well here but I would stick with the 390 ohm cathode resistors. Great tubes for hi-fi audio, I don't know why they have not caught on again.

Of the two amps, the "Black(ie) Box" is simpler to make and more punchy and powerful. It is a good performer with impressive measurements. The screen drive amp is more subtle and delicate, but also a bit more work. Both are solid workhorses that sound great. Betty and Blackie are hard-core daily listeners. My friend, Kato Hideki, uses the screen drive amps full-time with Tannoy monitors in his recording setup.

In closing, one of the really positive things that can be said about the hi-fi hobby is that it encourages one to pursue the hardware/software thing with a demanding attitude. "I want to love it!" Building the gear or making your own recordings is a radical statement of this, and re-involves us with the quality of the experience. Audio design is not easy and requires one to incorporate a number of seemingly contradictory forces, but really good work also needs this: an individual effort to go after the remarkable.... How do we nail this down into an approach?

In my own experience that is sort of like asking some big fat French guy to pass the potatoes, but he hears it as "Your mother turns tricks for centimes." It's just fraught with blind turns and misunderstanding. But, I do believe it amounts mostly to making the attempt, even though we know it will slip away.



The Feral Eye

by Herb Reichert, Audio Note NYC

I have a confession to make. By any rational standard, I am not qualified to design an audio amplifier. I have built every feasible circuit. I have read dozens of books and even studied in the classroom of my mentor Arthur Loesch. Still, after almost twenty years of trying, I feel radically unprepared for the chore.

I am a slow learner and I have the worst memory in the world, which may be an asset but it sure is embarrassment. I sound like a darn fool when discussing mathematical problems. I continue to ask the same dumb questions and rely on the advice and consul of guys like Steve Berger, J.C. Morrison, Komuro, Andy Grove, Kondo and Nobu Shishido. Without their help this amplifier would not exist.

On close examination, the only real talents I can muster are a basic ability to paint pictures and a potential to (sometimes) be quiet and listen. My (imagined?) capacity to glimpse a painting, read poem or visualize the forces of nature is the source of my belief that I can indeed design audio amplifiers.

The secret is forgetting

In order to make anything of exceptional value the maker must be untethered. As long as we truly believe that there are definite rules or concepts that govern our endeavors, as long as we are conscious of any notion that says we can't do something, we will mostly fail. To be free and create something original, we must embrace purposelessness. In fine art, when purpose is too much in evidence, the ART is no longer there. The creation becomes a machine or an advertisement. Beauty is replaced and the not so beautiful hand of the artist becomes altogether too obvious.

In art, the simplest ink sketch — even if seemingly very crudely executed — can stimulate deep feelings and engage the focus of our whole being. Amplifiers must do the same and present music as if we were facing our own personal nature directly. When we listen to music in the home we should feel every pulsation of it as if it belonged to us. Reproduced music attracts our attention most easily when we can "identify" our own nature in it. Therefore we should let our own character resonate in the design of the amplifier.

When the new amplifier is very good, the music comes from the speakers like "the breath of life". Music reproduction should feel direct and be perceived as having a similar substance and material character as a live musical event. The presence of the musician should appear before us.

Working in the lab

At the work bench, I feel like a student working on a science project in Dr. Zarkov's lab. The solder fumes enhance this vision. Sometimes I imagine that a tube is like a lens and the signal is a delicate energy form, like pulsing light. I picture a field of flickering, throbbing light in front of me. Looking at it, I ask myself, what is the simplest, most effective way to build a projector. How can I make the energy bigger, how can I stimulate the loudspeaker, without losing the signal's proportions and intensity?

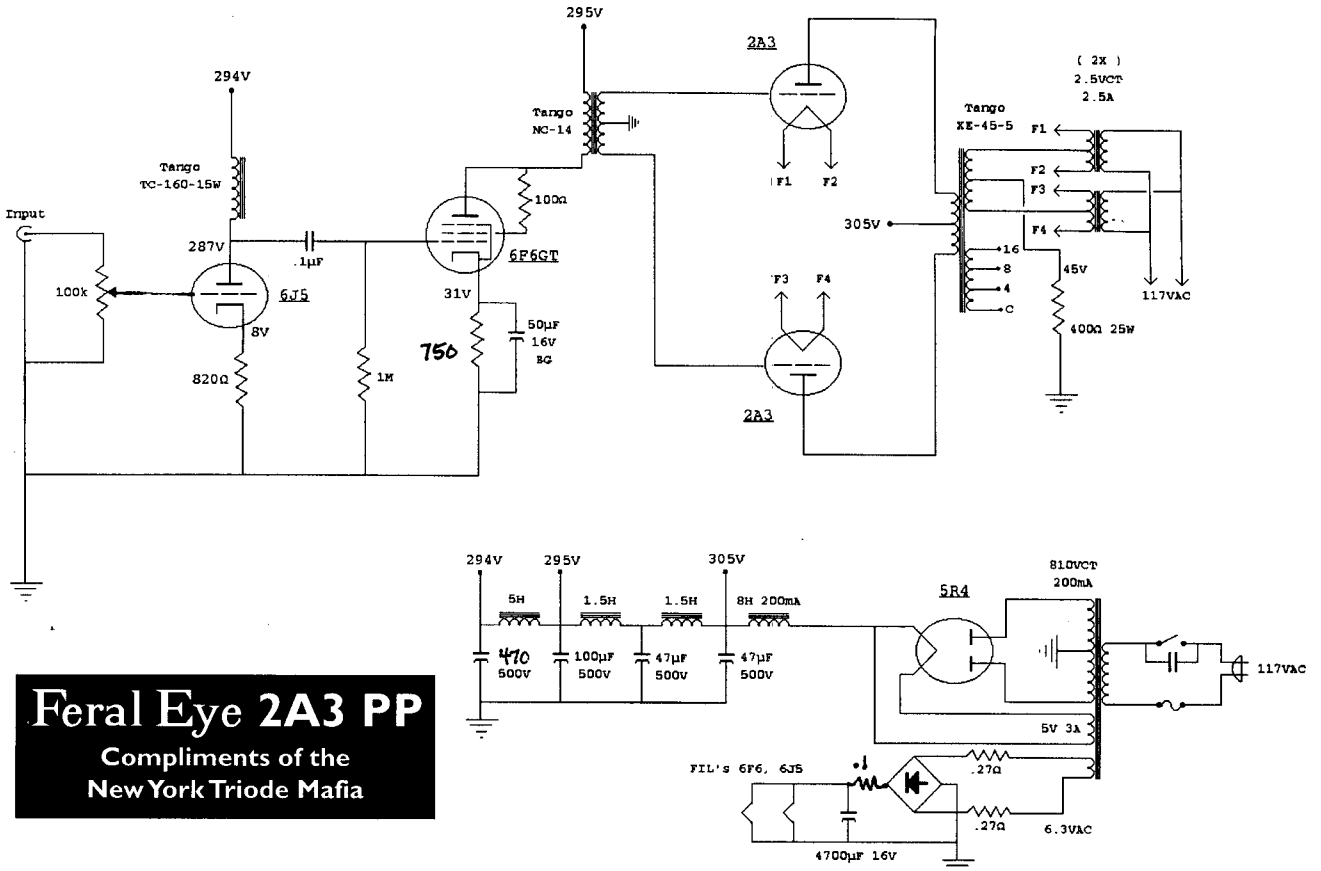
For me, the process of designing an amplifier starts with making pictures and setting my mind at ease. I need to feel I can do this. I imagine a nearly invisible, vibrating, electromagnetic slipstream forced to pass through a series of lenses and I suppose that I can compare the image of the original on one side to the thinner, bigger expanded version appearing at the other side. I try to keep a dynamic energy model in my head. I think power and energy not sinewaves. I struggle to visualize the effects of a series of partially mirrored pieces of glass (filters) on the energy. I picture a river of energy impressed on the load. I try to invent lattices of components that do the least damage to the original proportions of the energy stream. The components, the tubes and the transformers must be in good alignment. They must harmonize. I look for an ecology, a synchronous structure, a simple mathematical model and the most benign arrangement of all the physical aspects. This kind of mental imaging is, I believe, critical towards making quality amplification.

Blink, blink, blink...

The Feral Eye is being designed as a archetype amplifier for *Sound Practices* readers who want to experiment with push-pull. I hope you don't find it too strange to try yourself because this design has a few excellent qualities that are seldom found in push-pull amplifiers. The goal was to maximize PPs strengths and minimize its weaknesses.

The Feral Eye is a concept amp. It was conceived by me but several of my friends helped me bring it to fruition. *Feral Eye* is really an amplifier format designed for home builders as an inspiration towards their own creation. Look at its architectonic aspect. It represents my thoughts about what it takes to make a push-pull amp sound less shattered and artificial. It comes out of a desire to look again at an underdeveloped technology and see if something better can be done with center-tapped primaries.

Before I ever built a SE amp, I had a "paradigm shift" while listening to records at a friends house. This unusual friend had an AR turntable, Decca cartridge, RCA type preamp and a little "Bud box 2A3" amp driving Western Electric 755As. When I asked him why I loved the music so much today he said, "The only natural way to split phase is with a transformer or a split-load inverter!". Looking closely at the Bud box I noticed the UTC LS-55s and a UTC interstage transformer.



Several years later...

My best amp design ever was a transformer coupled parallel single-ended 210/310. Nobu's designs were the inspiration for this amp. This topology made for an even more unconstrained sounding amp than the *Flesh and Blood*. Tone character and dynamic contrast were extremely natural. At the time I designed this amp I was reading a lot of Norman Crowhurst. Since then, I remain convinced that a tube-iron-tube-iron lattice can reveal subtleties of musical expression that more conventional R/C (or direct) couplings miss.

The Feral Eye amplifier is a topology that can be used with almost any Class-A power tube. Believe me, you have not heard how psychedelic and colorful the 6L6 or the EL-34 can sound until you try them with this configuration. Using the "cathode feedback" secondary coil hookup with the above pentodes (in triode) will give you an amp with an extraordinary high frequency integrity and the kind of bass you never thought was possible with tubes.

Summer of '97

Most of my current amp design belief system can be seen in the schematic.

The following are some of the poetic ideas that informed this design:

1) Keep resistance values low. If you can replace a resistor with a piece of iron and a coil...do it! Resistors dramatically effect tone character and resolution. The worst offenders are high value plate load and cathode resistors. This is not simply an issue of metal film vs. carbon composition. It is the value! Using plate resistors of greater than 2xRp, despite issues of measured THD, make triode gain stages generalize and sound rough and jumpy. The trick is to reduce the value of plate resistance to the point where the transconductance (and probably THD) are maximized and then reduce it a touch further.

Choke is good

The same principle applies to the A/C impedance of the choke in the first stage. use as little inductance as necessary and adjust the plate current of the stage until it "opens up" sonically. Radical "henry changes" in amplifier plate loads almost always require adjustments in overall amp tuning. This kind of work can be over my head so I asked J.C. to check my arrangement on his bench. With a stage like this at the beginning of a three stage amp, you must be certain to provide adequate decoupling or you will experience some low frequency motorboating.

2) Use low mu triodes. (This should have been number one.) Low mu and plate loads between 1x and 2x Rp give whole, complete, unedited type reproduction.

3) Avoid differential amplifiers. My friend was right. Transformers and for those on a budget, split load inverters, give a push-pull amp a more "together", less shattered sound character. Transformer coupled amplifiers, with quality iron and well calculated time constants, sound considerably more intact and continuous. The most noticeable difference between live and reproduced music is that the real thing doesn't sound like it has been taken apart and put back together. Hi-Fi as most of us know it has that "Humpty-Dumpty" sound.

4) Keep the parts count minimal. The transformer as phase-splitter and driver serves several very useful ends. In this design it replaces at least four resistors and two capacitors. It allows for greater volt-

age swing and extremely gentle clipping.

5) Because of the zero crossing inherent to PP designs I probably should have called this amp *The Blinking Eye*. My basic feeling about push-pull vs. single-ended is that a PP amp designed by a perceptive, discriminating designer will always sound better than a SE designed by Mr. Radiotron. Bottom line, I am certain that SE has certain very tangible advantages when listening pleasure is the main criteria. PP has a slight edge when it comes to adaptability.

6) Use choke input filters. To me they give a sense of relaxation to the music making. They take the virtues of tube rectification one step further.

Courage under fire

Many amps with fixed bias and global feedback have really violent and annoying clipping characteristics. This amp, with the single-plate 2A3s and RCA 5R4GY rectifiers has what I call a "lubricated" clipping characteristic. It has a sort of James Bond like poise when it runs out of power. *The Feral Eye* was built by the hands of Steve Berger (Aprilsound). He laid out the chassis, bolted it together and soldered it with care and good jazz insight. Berger, Komuro and J.C. all participated in the design of the first two stages and the power supply. The creation of this amplifier was precipitated by the design team's love of, and experience with, the three pieces of Tango iron employed. I designed this amp because I have a special affection for the results I achieve when I use the Tango XE-45-5 and its center tapped secondary feedback coil.

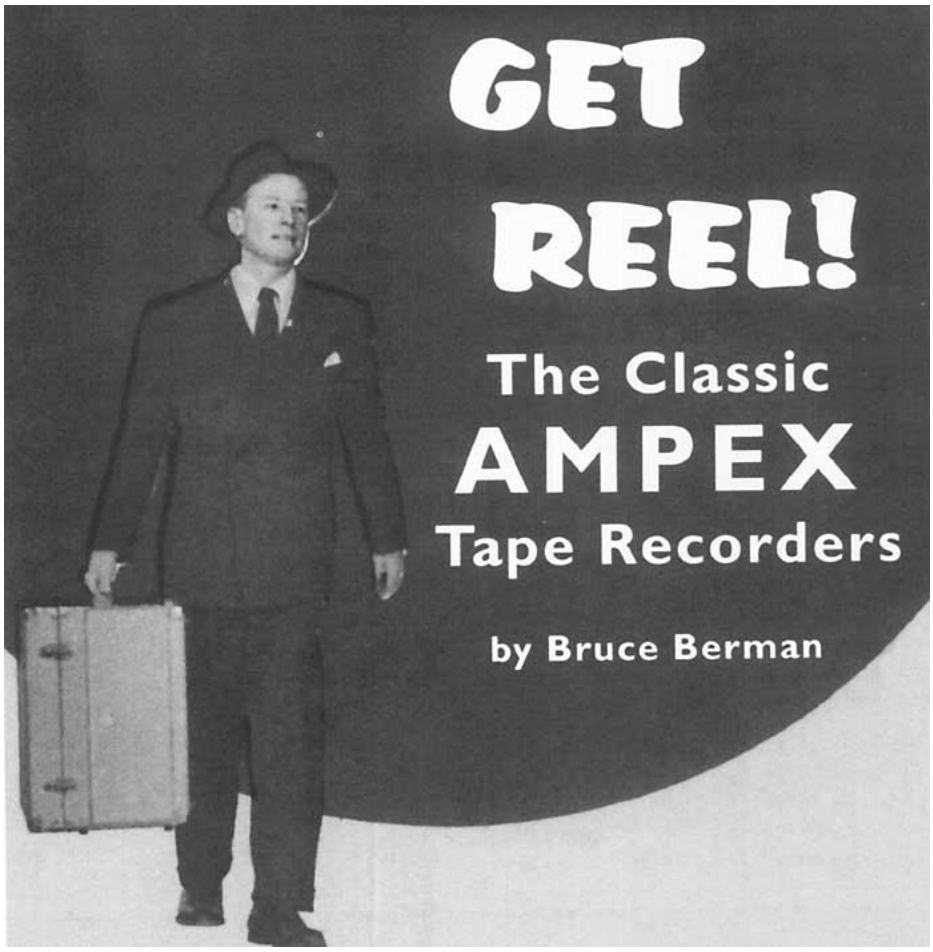
Connecting the filament transformer center taps to this coil not only reduces distortion at the frequency extremes and lowers the output impedance but it also allows the designer to use A/C filaments in push-pull with very low values of A/C ripple left on the output.

To hook them up correctly it is important to measure the gain of the amplifier with the heater trans CTs connected to alternate sides of this coil. The connection with the lowest gain is the correct wiring and the one with the higher gain is the positive feedback version.

The driver stage, with the Tango NC-14, can alternately employ the 6L6 or 6V6 tubes. My personal favorite is the 6L6. Go mad. Try the 6L6 driver stage and a 6L6 (ultra linear) output with your Quad ESLs. Do not let the current get too high on this stage or the NC-14 core will saturate on climaxes. The 6J5 is perfect with the TC-160. I love this stage! It is so good you can use it to improve almost any amplifier.

You can use other brands of iron. But! If you lose the secondary feedback coil you might as well switch to single-ended. The beauty of this design is in the use of iron.

I promise you that, if you build an amp, with this arrangement of metal, you will experience a new type and quality of music reproduction. *The Feral Eye* plays music very differently than most of us are likely to have heard before. It sounds more full, whole and complete than any traditional PP amp I have experienced. It is very rich sounding. It loses a bit to quality SE amps on pure magic and texture but makes up for the loss by driving a broad range of loudspeakers. Regard it as a modest gift from the New York City Triode Mafia.



It never ceases to amaze me how far audio technology had actually been perfected by the early 1950's. Companies such as RCA, Western Electric, Fairchild, Altec-Lansing, Langevin, among others, had made enormous strides in terms of quality sound reproduction. One company above the others always seems to be the unsung hero for also advancing the audio state of the art: the Ampex Corporation of Redwood City, California. It is still difficult to believe that nearly all of those treasured, highly praised (and now reissued) RCA Living Stereo and Mercury Living Presence recordings were recorded on Ampex Model 300-3 3-track tape machines during the Eisenhower era.

Ampex deserves more recognition for bringing about the tremendous technical and social impact that audio, video, and data magnetic tape recording has had upon us all. High quality stereophonic and multi-channel sound, modern LP recording techniques, Cinema-Scope sound, and videotape recording, all owe their development in whole or in part to Ampex. The professional versions of their machines exemplified the superb quality of American audio engineering and manufacturing know-how that was the envy of the entire world.

Indeed, Ampex deserves most of the credit for perfecting and marketing modern magnetic tape recording technology. Prior to the wide-scale introduction of tape recording equipment in the U.S. in 1947, 12 and 16-inch 33-1/3 and 78 RPM disk transcriptions were the primary storage media for professional sound applications, along with optical recording techniques for film sound. During World War Two, the Germans, through the efforts of AEG Telefunken, had developed the first high quality audio tape recording system using high-frequency AC bias, known as the Magnetophon. As a spoil of war, the U.S. Army Signal Corps captured and evaluated a number of these machines, one of which was subsequently demonstrated at an Audio Engineering Society (AES) meeting in San Francisco. Ampex Electric, who had manufactured motors and generator units under government contract for the military during the war, was looking for a new product to carry them through the post-war period, and upon hearing the startlingly realistic high fidelity sound reproduction provided by the Magnetophon at the AES demonstration, quickly sensed that there would be a tremendous future for

The Groundbreaking Ampex Model 200 Magnetic Recorder

this new technology.

Short on cash, Ampex found a willing investor in Bing Crosby. Crosby, who apparently had been something of a stickler for the audio quality of his popular radio show, was looking for a recording system which offered superior fidelity and ease of editing compared to the 16-inch transcriptions then in use to record his show for rebroadcast. He personally bankrolled the fledgling Ampex Electric Corporation to a large extent during the company's early tape recording years, and for many years thereafter was an enthusiastic proponent and supporter of the company as well. An early Ampex advertisement shows Crosby happily using one of their portable 600-series machines.

Alexander M. Poniatoff, the president and founder of Ampex, along with his team of engineers, (his initials plus "EXcellence" were the basis for the company's name), carefully studied the shortcomings and merits of the German Magnetophon design, and in 1947 with Crosby's financial backing, introduced the landmark Model 200, with the first production units delivered to the American Broadcasting Company (ABC) Radio Network.

To some extent, the Model 200 was to establish the basic mechanical design of the tape transport system for all future Ampex professional recorders. Using three motors, with one each functioning as torque motors for the take-up and supply reels, and one driving the capstan, and with solenoid-actuated braking, these units were immediately recognized within the audio industry as offering the finest sound recording technology available to date. To Ampex collectors, the 200 is analogous to a Western Electric 91A amplifier; extremely scarce and desirable, as only 112 of these machines were built.

Unfortunately, I wouldn't even bother looking for one, as most were scrapped out long ago or are now in the caches of collectors. Really overdesigned for the task at hand, with oil-filled capacitors in the record/reproduce electronics power supply, and a beefy push-pull 6L6G bias/erase oscillator, these units were truly the cutting edge in the late 1940s for high-fidelity sound recording and reproduction. With a single tape speed of 30 ips, the Model 200 was the only Ampex unit manufactured in which the tape wind was "oxide-out" like the Magnetophon. All subsequent machines (beginning with the Model 201) were built with the conventional "oxide-in" format.

In 1949, Ampex introduced the Model 300 as a lower cost and more versatile version of the 200. The 300 was an upgraded machine compared with the 200, featuring redesigned tape heads, new electronics, and a massive, improved two-speed tape transport mechanism. The 300 was the machine that really launched Ampex into the recording studio, cinema sound recording, broadcast, and telemetry/data recording markets. With monaural console-mounted versions vaguely resembling a washing machine with a hamper door in appearance, this was the first tape machine to be configured for multi-track recording, and 1/4-inch half-track, full-track, two-track, and 1/2-inch three-track versions were built in large quantities. The 300 was also the basis for the impressive-looking Ampex 3200-series tape duplicating systems, introduced in 1954.

The first Model 300 was delivered to Capitol Records, and all of the major

recording labels quickly followed suit. Monaural versions of these machines used the series 554 electronics, which were relatively similar to the electronics employed in the Model 200, but with much improved distortion performance.

Multitrack units used the 350-series electronics which were to become the Ampex standard for record/reproduce electronics until the late 1950's. Using a combined pentode/triode record/playback circuit topology and vacuum tube rectification, these machines are capable of absolutely stunning sonics, as RCA and Mercury well knew, and if a 55 to 60 dB signal-to-noise ratio doesn't bother you.

Model 300s are still around, although their quantities are dwindling, and transport parts have become somewhat scarce. Incredibly reliable and smooth-acting self-energizing drum braking was pioneered by Ampex for the 300, and was always regarded as a major hallmark of the Ampex design.

One problematic feature of these machines is the indirect capstan drive assembly, which used a large rubber-tired capstan-flywheel assembly, friction-coupled by a solenoid to the pivoted hysteresis-synchronous capstan drive motor. Over time or through improper storage, the rubber tire lining the capstan-flywheel can develop hard or flat spots which will severely degrade the wow and flutter performance of the machine. Replacements for this part are in very short supply and expensive, although I have acquired junked 300s which had perfectly good tires. When buying any 300-series machine, always be certain to check the integrity of this tire, as it may be a long time before you can locate a good spare. Perhaps a vendor exists who can resurface or replace the rubber on this component to Ampex specifications; if anyone is aware of such a service, please let me know.

Ampex used this indirect capstan power transmission scheme to obtain very low tape flutter and precise tape speed adjustment by varying the motor pulley pressure on the rubber tire and observing the results with a strobe disk. However, the long-term speed stability was never really that good with these machines, and a retrofit kit was later offered by Ampex to convert them to a less troublesome direct capstan drive system. Nevertheless, the 300-series units are a real classic in the history of high-fidelity sound recording, very cool looking in a 1950's industrial sort of way, and well worth owning and maintaining if you have the space.

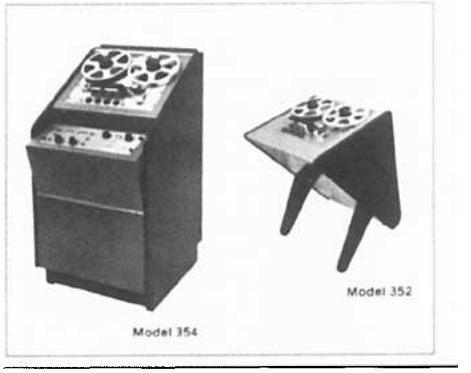
Model 351

SPECIFICATIONS:

(Note: specifications apply to all three models except where noted)

Speeds	Dual speeds: 7½ and 15 ips or 3¾ and 7½ ips
Frequency Response	± 2 db, 30-18,000 cps at 15 ips
Signal-To-Noise	55 db at 7½ and 15 ips*
Flutter and Wow	Below 0.15% rms at 15 ips
Timing Accuracy	± 0.2%
Crosstalk Rejection	Model 354 or 351-2 : 65 db at 400 cps

* 60 db at 7½ or 15 ips for Model 351 full-track



Model 352

Model 354

350 SERIES

. . . a complete family of rack-mounted, console, and heavy duty portable recorder/reproducers designed primarily for the broadcast and recording industries. The 350 Series is accepted as the world standard of the broadcast industry. Thousands of units are in professional use throughout the world. Includes Ampex "Four Star" one year warranty.

FEATURES: The basic tape transport is common to all three models — 3 motors including direct drive by hysteresis synchronous motor — has pushbutton, relay-solenoid operations for remote control — reel-size selector switch automatically adjusts tape tension for various reel sizes — non-wearing tape lifters, fast wind and rewind (rewinds 2,400 feet in just 60 seconds) — all controls interlocked to prevent jamming — basic electronics with separate record and reproduce amplifier electronics permit easy monitoring at input or playback.

ADVANTAGES: Meets broadcasters requirements for long-life, ruggedness, and dependability. Wide variety of models, each designed to meet specific professional requirements of monophonic or stereophonic recording and reproduction in broadcasting, recording, education, business or research.

MODEL 351 . . . a monophonic recorder/reproducer available unmounted, console-mounted or portable — two-channel stereo model (351-2) available unmounted or as a portable.

APPLICATIONS: For monophonic (351) or 2-track stereophonic (351-2) recording and reproduction — designed primarily as a broadcast recorder for heavy, continuous duty operation requiring exacting performance characteristics — has also found wide use in recording studios, education, business and research.

MODEL 352 . . . a tape playback-only unit available unmounted for rack use or in its own floor console — two-channel stereo model (352-2).

APPLICATIONS: Identical playback functions as 351 recorder/reproducer — the playback-only function reduces cost, eliminates accidental erasure of valuable tapes — lower cost, without sacrificing performance, makes it ideal for: broadcasting and recording industry, commercial background music for businesses and industry, dance studios, education, etc.

MODEL 354 . . . a monophonic and stereophonic recorder/reproducer available unmounted, in floor console or portable — records and reproduces two-track stereo with simplified electronics and controls.

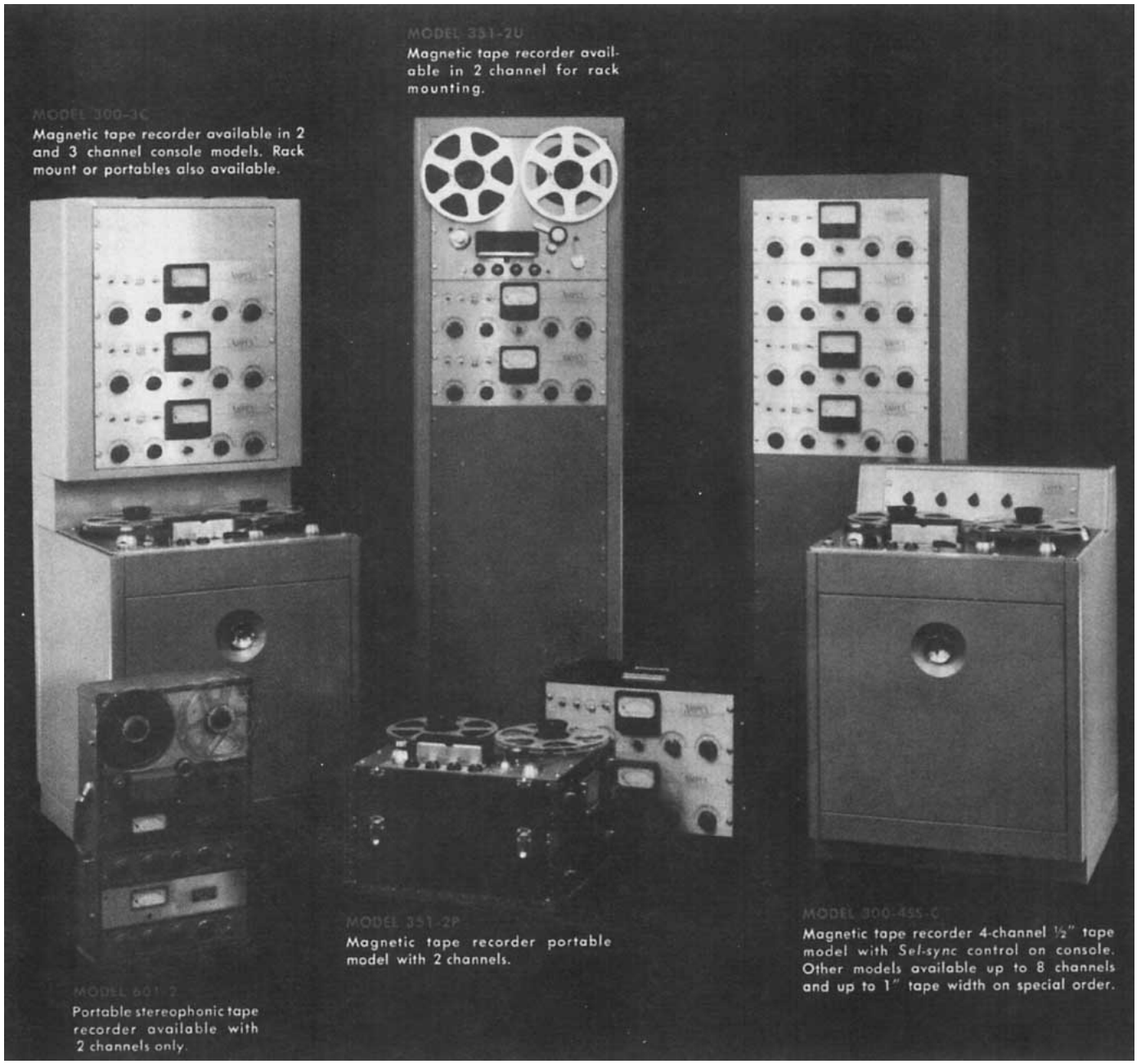
APPLICATIONS: Has same applications as Model 351-2 — two-channel electronics in Model 354 are on one panel, occupy the same space as a single-channel panel, thus reducing space and weight, providing easier operation and control. Ideal for stations considering FM Multiplex stereo in future, permitting use as monophonic recorder now, stereophonic in future.

In 1953, the series 350 was introduced as a lower cost alternative to the Model 300. Utilizing a lighter, smaller, somewhat less robust and simplified version of the 300 transport, such as a hysteresis-synchronous tape drive motor with a direct-drive capstan which eliminated the troublesome capstan drive rubber tire, this transport essentially established the mechanical design standard on which nearly all of the subsequent Ampex professional 3-motor machines were based.

Like most Ampex recorders, the 350 was also available as a portable version for recording in the field, wherein the transport and electronics were mounted in individual transit cases. The 300 was also available as a similar portable unit, but required two persons to carry the transport assembly. Most 350s were sold to radio stations, educational institutions, and lower-end recording studios, primarily due to their lower cost, and relatively large numbers of these machines still exist.

The 350 record/reproduce electronics, a pentode/triode design using octal-based tubes, point-to-point wiring, and a separate outboard tube-rectified power supply unit for each electronics assembly or channel, are considered by many enthusiasts to be the best sounding units ever made by Ampex. The separate power supply should be considered as a definite asset, as it provides significant electrical isolation between channels. However, these units employed a shock-mounted and selected 6SJ7 pentode as the first gain stage in the reproduce section, and these tubes were extremely prone to noise and microphonics.

For this reason, many users may prefer the 351-series electronics introduced in 1958,



Classic Ampex machines of the vacuum tube era

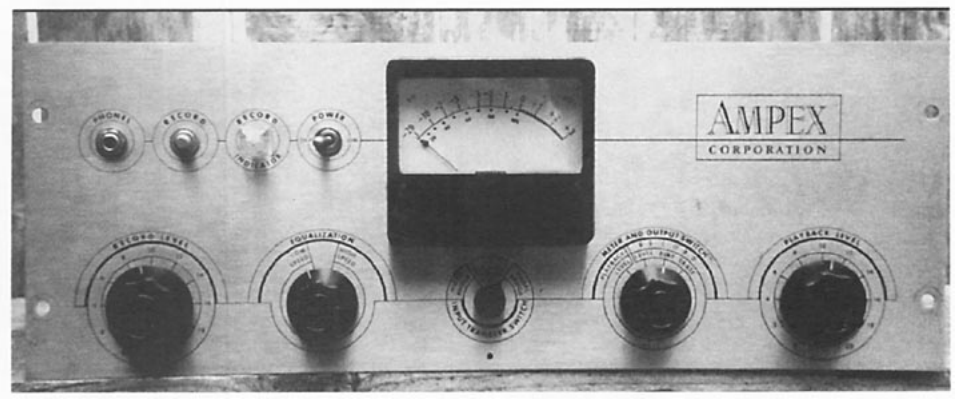
which used miniature triodes (12AX7s, 12AT7s, and 12AU7s, all Telefunken as installed at the factory!), tube rectification, printed circuitry, and have more of a "triode sound". However, 351-series machines do not employ the outboard power supply design approach; the power supply is integral to the electronics. The 350 and 351-series electronics each have their sonic strengths and weaknesses; if you can, try to audition both and draw your own conclusions.

The really adventuresome experimenters will want to design and build their own tube record/reproduce electronics, perhaps modeled after Ampex's best efforts. One day, I intend to do just this.

For the serious amateur recording enthusiast, the 1/4-inch stereo 2-track 350 or 351-series recorders are recommended. Replacement parts are still fairly common for these units if you know where to look, and these machines are of extremely robust design and construction, and easy to work on as well. These recorders are absolutely gorgeous sounding when properly set up and aligned.

Ampex professional machines were designed for demanding 24 hour a day continuous operation in broadcast or professional recording service. They are workhorses in the strictest sense of the word, and set the stage for the legendary Ampex reputation for quality and reliability. For home use, they should last nearly forever.

Drive motor and brake assembly rebuilding services are available, as is tape head relapping if the heads in your machine are not too far gone. Be warned that replacement new old stock Ampex heads are now hard to come by and expensive.



One Channel of Model 351 Electronics

Rear View of Model 350 Electronics

Rear View of Model 351 Electronics

Record/reproduce electronic parts are all fairly standard and readily available, and many possibilities exist for modifying and improving the sonics of these units, such as the usual parts upgrades, improved power supplies and the like, but I'll leave that up to you.

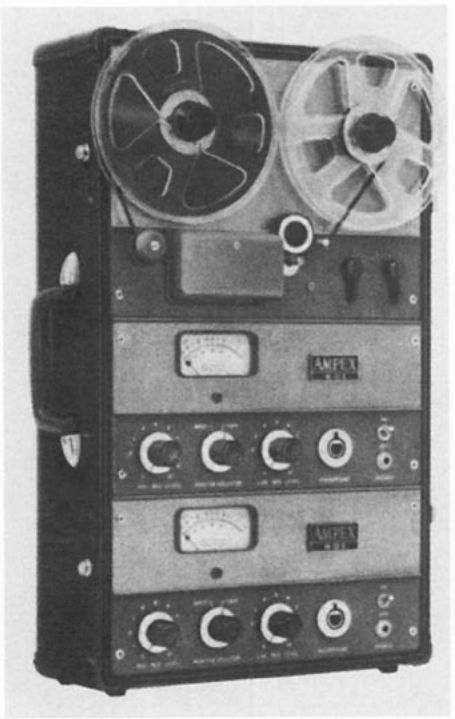
Around the same time that Ampex introduced the 351-series units, the stereo 2-track Model 354 was brought to market. This machine used the 350 style transport with an electronics package incorporating two of the 351-series electronics within a single enclosure with an internal shared power supply, in an effort to reduce the size for console mounting. Tiny VU meters, poorly ventilated hot-running electronics, and a notoriously unreliable record/reproduce slide selector switch which is no longer available, make three good reasons to avoid these machines, and more importantly, the sonics of the 354 are not on par with the 350 or 351-series equipment.

Early 351 and 354 record/reproduce electronics units used phenolic printed circuit boards, which held up very poorly. I find it surprising that Ampex resorted to the use of this material, usually seen in low-end consumer electronics products. Try to find the electronics which incorporate the later glass-epoxy boards which were essentially trouble-free, and save yourself a lot of grief downstream.

Over the years, Ampex manufactured many versions of truly portable professional single-motor machines with 7 inch reel capacity, designed largely for high quality recording in the field, such as the venerable 600-series and the PR-10 (Portable Recorder) series. Although the tube versions of these units are also capable of superb sonics, their transport mechanisms age poorly, and replacement parts are now scarce. Avoid them unless you are certain that the machine you are looking at has been well-maintained and is in good working order.

By the mid-1960s, Ampex began gradually phasing-out the vacuum tube recorders in favor of the solid-state units that the recording industry wanted. The AG-300 and AG-350 (for Audio General) were introduced as the transistorized successors to the 300 and 351-series tube machines. Featuring a dressed-up 300 or 350-style transport, and solid-state electronics, they were the evil-sounding versions of their tube predecessors.

They were discontinued in late 1966, and



601-2

PR-10

AG-600 and AG-600-2

AMPEX

Portable Recorders

ATR-700

replaced in early 1967 with the much improved AG-440 series of product, which ran successively through the early 1980's as the AG-440, AG-440B, and AG-440C units, until Ampex was forced, due to severe competitive price pressure, to resort to offshore-manufactured equipment beginning with the ATR-series recorders. The AG-440C was the last domestically manufactured Ampex audio machine to roll down the assembly line. I believe the AG-440s have one of the best transport designs Ampex ever used. A significantly more robust version of the 350-series transport, 440-series machines feature automatic tape lifters and, in my judgement, the most attractive styling Ampexes' industrial designers ever came up with.

Actually, the AG-440 series recorders sound surprisingly good, considering their transistorized electronics. I run a stereo 2-track AG-440B-2 and a mono full-track AG-440C, in addition to my console-mounted 300C-2 stereo 2-track and 351-2 and 602-2 portable stereo 2-track tube machines. I find that the AG-440s have a very full-bodied sound given their silicon-based design, although some of that solid-state coarseness is apparent upon critical listening. 440-series machines are still very common and easy to get running, and were built in numerous configurations, so you may wish to consider one of these units as well. However, like myself, I'm sure most of you are hung up on getting the old tube machines, so.....

Over the years, Ampex also made a number of machines aimed primarily for the consumer market. Generally speaking, these ranged in quality from fairly good to atrocious, so you would do well to avoid them.

And the best Ampex audio recorder ever made? Most enthusiasts agree that it was the Model MR-70 (Mastering Recorder), introduced in 1964, and manufactured in very limited quantities. I seem to remember reading somewhere that only 70 or so of these fine machines were actually built. An expensive tube design employing extensive mu-metal shielding, metal film resistors, film capacitors, low-noise nuvistors in the front-end of the reproduce electronics section, and a modified version of the 300-series transport, the MR-70 was probably condemned to fail from it's inception. It was brought to market just as Scully was introducing their solid-state model 280, and most recording engineers of the day simply couldn't wait to get rid of the noisy, troublesome, microphonic tube gear used in their studios. Remember, these guys did it for a living, and like anyone else, wanted their jobs to be as easy and nuisance-free as possible.

Unfortunately for us, sonic considerations were and still are almost always of secondary importance in the recording industry.

Internationally, Ampex was the undisputed leader in professional audio tape recording well up to the mid-1960's, and for many years the name was literally synonymous with the technology.

Companies such as Magnecord, Fairchild, Presto, Rangertone, and other manufacturers of professional recording equipment came and went, largely due to their inability to overcome quality, performance, or customer support issues, which were Ampex strengths. Ampex only really began experiencing serious competition in the mid-60's from Scully with their introduction of the model 280, the first high quality solid-state professional studio audio recorder. From that point on, Ampex slowly began losing market share to Scully, Studer-Revox, Otari, and others.

AG-440C-2
(late 70s–early 80s)

AMPEX

Late-Model Solid State Professional Audio Tape Recorders

ATR-100

When you find the Ampex of your dreams, you absolutely must get the original Ampex technical manual for your machine, which are still available from the sources indicated below. Carefully set the reel tensions, head alignment and azimuth adjustments, clean and demagnetize everything in the tape path, lubricate the capstan and/or drive motor, and verify that the electronics are in good working order. Select a brand and grade of recording tape to use, purchase 7.5 and 15 ips NAB or IEC alignment tapes, and align the electronics to Ampex specifications.

Whether tube or solid-state, these machines really come alive at 15 ips. The tape head structure and design, another Ampex hallmark, the so-called "heart of the Ampex" in their early 1950's advertising efforts, and so critical to high quality, distortion-free, wideband magnetic recording, was really optimized for a tape speed of 15 ips. If you want to experience what real stereophonic reproduction sounds like, forget about Audioquests, Mobile Fidelitys, and other audiophile-grade recordings. Make your own. Beg or borrow master tapes and studio out-takes from recording engineer friends. If anyone out there can help, I would love to score an analog master copy of one of Garrison Keilor's "Prairie Home Companion" radio shows, for playback through my system.

We homebrew our own audio reproduction systems; perhaps it is now time for us to begin a revolution in high quality analog vacuum tube homebrew tape recording as well. Our Japanese friends have been doing this for many years now, and you'll never have any shortage of amateur or semi-professional musicians eager to be recorded.

With a pair of good quality, properly positioned microphones feeding straight into the 600 ohm balanced "MIC" inputs of a 351-2 stereo 2-track, and a reel of Ampex Grandmaster 456 recording tape ripping along at 15 ips, you will make recordings with such incredible depth and spatial perspective, clarity, naturalness, and low distortion, that you will probably hear for the first time what stereophonic sound reproduction is truly capable of. Prerecorded tapes and LPs have so many generations between the master tape and you, along with the deleterious sonic effects of signal processing such as gain limiting and compression, etc., that so much information intrinsic to the original musical performance is stripped away that we never get to hear.

When your Ampex is played back through a pair of single-ended directly-heated triode amplifiers with zero feedback and a nice pair of horns, you will be experiencing a totally new meaning to the well-worn phrase, "high-fidelity". And from analog tape recording equipment that was largely perfected before the dawn of the Space Age!

AMPEX Resources

The following sources are recommended for obtaining service, replacement parts, technical manuals, and alignment tapes, as Ampex is no longer supporting their now-discontinued audio product line:

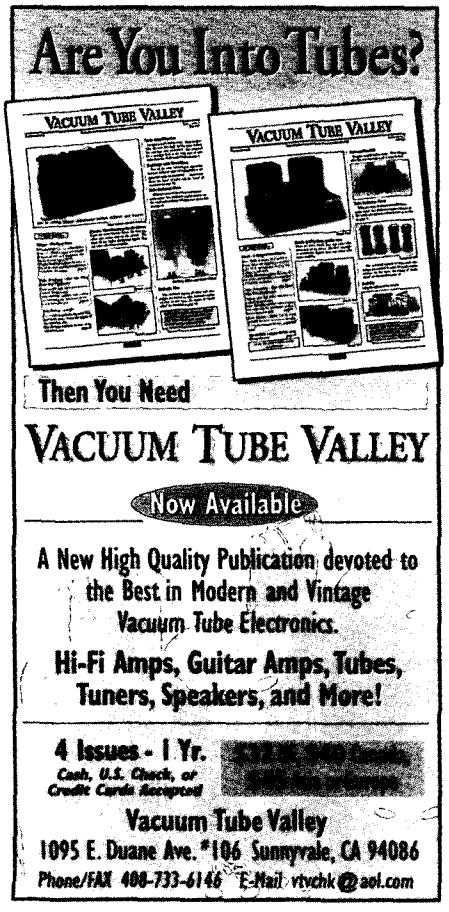
AMPEX Mailing List online FTP Archive

ftp.soundpractices.com
UserID — Ampex
Password—guest

Mr. Vern Sauer
TSI/Technical Services, Inc.
P.O. Box 765
Arlington Heights, IL. 60006
Phone: (847) 392-2958

A genuinely nice guy with a veritable wealth of knowlege about virtually every Ampex audio recorder ever built, Vern is a former employee of Ampex and has serviced these units since the early 1950's. An excellent source for spare parts, service, and technical manuals. Highly recommended, but please don't waste his valuable time with trivial questions about these machines.

(continued on page 44)



(continued from page 24)

Sprague Magnetics
12806 Bradley Avenue
Sylmar, CA. 91342
Phone: (818) 364-1800

When Ampex stopped supporting their audio products in the early 1990's, Sprague acquired all of their remaining spares inventory, engineering documentation, etc., and has reintroduced some parts which Ampex had discontinued. Consider them for parts for mostly the AG-440 series and ATR-series machines, tape heads, technical manuals, etc., but be prepared to spend some bucks.

VIF International
P.O. Box 1555
Mountain View, CA. 94042
Phone: (408) 739-9740

High quality after-market pinch roller assemblies.

IEM
350 No. Eric Drive
Palatine, Il. 60067
Phone: (708) 358-4622

Replacement tape heads and offers a good relapping service.

JRF Magnetic Sciences
249 Kennedy Road
Greendell, N.J. 07839
Phone: (201) 579-5773

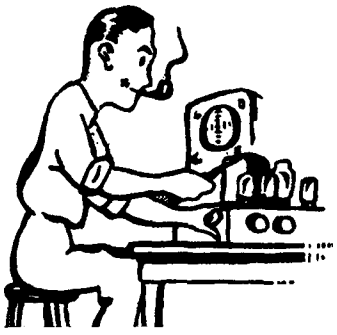
Replacement and reconditioned tape heads, and also offers a relapping service.

Premium Parts & Electronics Company
Route 3, Highway 59
Whitewater, WI. 53190
Phone: (414) 473-2151

Replacement belts and idler wheels for Ampex portable professional machines, such the 600-series and the PR-10. Also offers an idler wheel rebuilding service, and stocks some replacement tape drive parts for the Ampex consumer machines. They cannot, however, rebuild the rubber-tired capstan-flywheel assembly used on the 300-series recorders.

Mr. Jay McKnight
Magnetic Reference Laboratories
229 Polaris Avenue
Suite 4
Mountain View, CA. 94043
Phone: (415) 965-8187

High quality alignment tapes, in NAB, IEC, AES, and CCIR equalization. Mr. McKnight is a former long-time employee of Ampex.



Homebrewer of the month

Ed Billeci of Portland, OR

When you step into audio craftsman Ed Billeci's listening room, you know you are in the presence of a maniac homebrewer playing in the far outer ionosphere of Planet Audio, out where there are no maps and inspiration is the only guidepost.

Ed's system has the unique charm that only one-of-a-kind and built with love audio gear has, but this set-up is so *extreme* that it stands apart from the crowd. Very few searchers are hard-bitten enough to stretch out quite so far in pursuit of their personal audio vision.

Really, only a handful of audio hobbyists progress to the point where they would even know they wanted something like this—let alone wanting it bad enough to make it a reality? No doubt about it, Ed Billeci is a homebrewer of the first order.

The centerpiece of this breathtaking installation is a pair of monster 212 single-ended amplifiers, constructed on three industrial-strength chassis per side: one each for the HV supply, LV supply, and audio circuitry. Every audio tube is provided with its own separate supply chassis and, to give an idea of the over-the-top beef factor of Ed's mighty 212, the mono PS for each WE 437A input tube weighs 75 pounds! The HV supplies are 130 lbs. a side!

Ed says that he got hooked on the rare, ancient, and huge 212 triode about five years ago when he needed a tube amp to drive some hoggish two-panel Acoustats. A prototype 212 SE amp he built as an experiment impressed him to the point where he decided to undertake this no-limits project, so he must have been pretty damn impressed!

The eye-popping 212 amps in the pictures took the man over three years to build, including two years selecting and collecting parts. The final push to get the amp "on the air" required six months of what the builder called "intensive" application of time and labor.

Due to the level of build quality he just had to have, poor Ed was forced to blow out a big chunk of his vintage audio collection in order to finance construction, but so what?— a whole warehouse full of junky old-fashioned tube amps doesn't even equal one channel of this phenomenal 212 amplifier! Real excellence never comes cheap.

Each chassis is fashioned from aerospace aluminum I-beams heliarc welded into frames that hold a total of 30 individual polished brass framed subassemblies. The intent of the packaging design is to have the ultimate triode amp breadboard. The modular construction streamlines the process of continual experimentation and refinement by swapping out subassemblies.

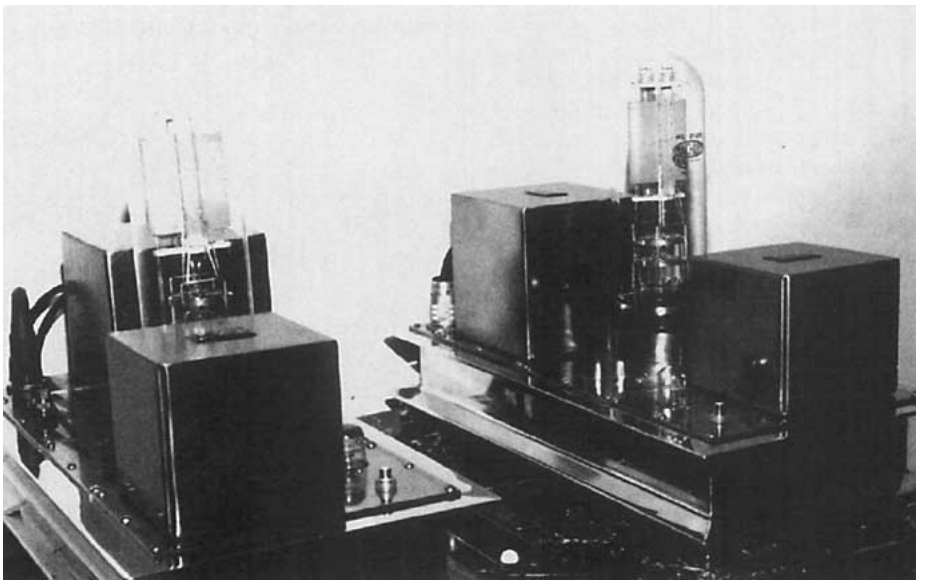
One would guess that the other intention of the builder's packaging concept was to blow the minds of all who gaze upon the creation.The silvery aluminum bar stock, golden brass top plates, and shiny brass hardware speak pure elegance to the eye.

If you got to look underside, what you would see is a large collection of transformers and chokes. According to Ed, there are 32 pieces of iron in these babies, counting the four Bartolucci audio transformers.Twenty-four of the transformers are in polished steel cans mounted above the chassis.

In addition to lots of chokes, all power supplies utilize mercury vapor rectifiers, 866s on the HV and 816s on the low voltage supplies. Weighing in at 700 pounds, including the custom-made aluminum and polished marble racks, these three-box 212 mono amps are not very well suited for portable applications. Ed did drag the amps up to the recent VSAC show in Washington State where he handily swept away first prize in the homebrew amp craftsmanship competition.

Although these 212s put out 50 watts easy, Ed likes to think of the amplifiers as "twenty-watters with lots and lots of headroom," although they can crank out a lot more juice on the bench. Since he converted from electrostats to horns a few years ago, twenty watts is enough, thank you.

The current loudspeaker setup, subject to continual change, is a TAD 2001 compression driver on an RCA 300 Hz fifteen-cell horn. Ed prefers the sound of the fiber material used in these three-row RCA horns over what he was getting from the usual metal multicells. A modified JBL adapter mounts the TAD to the RCAs, originally supplied with hardware for screw mount drivers instead of bolt-ons.



Audio circuitry for Ed Billeci's three-chassis monoblock WE 212 amps

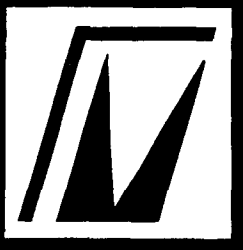
The bass cabinet uses a *Western Electric* 728B 12" in a modified horn salvaged from a discarded Altec A-7 cabinet and the driver is loaded by a rear-firing bandpass ported system on the low end. Ed made his own foil inductors from copper flahing and polypropylene film..He rolled the stuff up then took it to a shop to have it cut into slices. Signal cable is custom 21 ga. and 23 ga. silver wire, made from ingots at a local refinery., insulated with fiber tubing, sonically superior Ed says, except in the HV lines where he uses teflon tubing over the fiber for safety's sake.

Amazingly, Ed is happily married, living proof that a big horn system and marriage are not entirely exclusive. Ed says his wife, Chi, is the most patient woman in the world. Well, that's good because Ed is now working with John Camille on a high frequency AC filament supply for the 212s, since DC doesn't sound as good as AC. Hey Ed, looks like you'll need another shelf on that rack!

Check out Ed's homebrewer page!
http://www.teleport.com/~tube

Phono setup includes an idler-wheel turntable built from a recycled broadcast turntable and a mono Denon DL-102 cartridge on a 50s vintage 16" Gray professional transcription arm. The tonearm is mounted on a separate 20 pound copper base which can be moved around for set-up adjustments. Spin those mono LPs and 78s, dude!

The φ42
A Phono Preamp for "WOT" and Whatnot



by Diego Nardi
Audio Note Italia

Promises are promises. When, at the end of my article about the "WOT" line preamp in *SP #10*, I said "A true phono stage for use with the WOT I will design when I have time," I was meaning it. Here's a design for a phono stand-alone unit which may partner either the WOT or virtually any other existing tube or solid-state line preamp or integrated amplifier and which, given a competent line stage, may even readily become a one-chassis complete preamp.

It is the latest development of my split-RIAA Phono circuit, of which I have already given you an anticipation in the above mentioned article. I have been working on it for the last two years and I think it has now reached a very satisfactory level.

GENERAL CONSIDERATIONS

I reckon the Phono preamplifier is one of the most interesting and challenging topics an audio gear designer may ever decide to face. It is a deadly cocktail of massive gain, accurate frequency response shaping, low noise, and vast headroom demands. Designing a power amp, even a relatively sophisticated one, looks nothing but a primary school test of arithmetic by comparison, and this is probably why you don't see too many Phono circuits in audio magazines.

The lower the operating level, the more difficult it is getting low coloration AND low or no dynamic compression AND high resolution at the same time.

1. DESIGN CHOICES.THE GAIN

First of all, it is required to set a gain target. The RIAA spec sets the 0dB reference of the de-emphasis curve at 1 kHz, therefore it is standard practice to consider the gain at 1 kHz as the "nominal" gain, while at the low end of the curve (which is specified at 30 Hz, below which the response should more or less flatten out;(see RIAA spec for further details) the output level shall be +18.61 dB.

The actual inherent gain of the circuit will then turn out to be the 1 kHz nominal gain, plus 18.61 dB, plus the loss of the equalizing network.

The necessary gain at 1 kHz depends on the input level, i.e. the cartridge output voltage, and the sensitivity of the rest of the chain.

The output voltage of cartridges is expressed by most manufacturers at 1 kHz, 5cm/s, and we may use this value as a "nominal" one. You all know that today's cartridges may be broadly divided into two categories: "traditional" moving coils, and all the others including moving magnets, variable reluctance and a few moving coils of the so-called "hi-output" variety.

The nominal output of moving coils may vary between, say, .05mV and .4mV (-84 up to -66 dBm), yeah that's desperately low. The "others" show outputs nearly always falling between 1.5mV and 7mV (-55 to -41 dBm), with the great majority between 2.5 and 5mV.

Let's now pretend we have a 5mV magnetic screwed to our beloved tone arm, a WOT preamp or something with a similar gain (a bit more than 20dB into 100k ohms) and a power amp demanding 0dBm into 100k ohms to put out full power. Question: how much amplification do we need in the Phono preamp?

It is reasonable to assume a few more things:

a) The program material is likely to contain peaks about 10dB above the nominal level, and we don't want these to overdrive the power amp;

b) The program material might VERY SELDOM contain peaks up to 20dB in excess of the nominal level; we accept to take the risk these will clip the power amp but we DON'T want them to clip the preamp; the preamplification must have tons of leeway under the most tragic circumstances;

c) We don't want to be forced to turn the volume full up to entirely exploit the power amp with "average" records. We want at least 20dB volume leeway.

The above leads to a required level, at preamp input, of about -10dBm (line gain 20dB, average amp drive -10dBm, amp clipping 0dBm therefore amp leeway 10dB). Our moving magnet cartridge puts out 5mV, that's about -44dBm.Therefore, minimum required gain 34dB.

Hey, folks, that's already 52.61dB at 30 Hz, PLUS the EQ loss. Serious amplification indeed!

Yet this is the very least required for a Phono stage, and, in commercial products, a Phono nominal gain in the 34-38dB range has become a de facto standard. Only problem is, this sort of gain does not leave any margin and will give weak sound if the cartridge only puts out 2mV and/or the power amp is only 2V sensitive and/or this 2V sensitivity is referred to just...three watts (the typical tragedy of the usual super-uninspired, oversimplified 2-stage single ended lice so fashionable today...).

If there's one thing that I really hate, that's "insensitive" systems. Therefore, the target I chose was higher gain at no sacrifice of dynamic margin. The nominal gain of this preamp is 42dB, and you need to throw HALF A VOLT RMS to approach saturation. 42dB above 500mV makes for about 63V RMS -180V p/p- and this is more or less what the circuit can put out at less than 5%THD.

This preamp is therefore NOT of the MC direct plug-in variety. The reason why I think this is a loser strategy is explained in the next paragraph. Nonetheless, the gain is high enough to rock and roll your system even with the most uninspired "not-too-high-output" 1mV coils—and get a quiet background from this level up.

As regards the dynamic headroom, I chose to keep it so high in order to get super-super low distortion at ordinary operating levels, but avoiding, at the same time, to have insufficient drive signals anywhere in the circuit. In other words, while high gain is helpful in keeping the signal far above thermal noise and low level "dirt", large dynamic margin is an insurance against dynamic compression and non-linearity. This is not to get academic nice-spec results, but for very positive sonic purposes —explosive dynamics, smooth distortion-less presentation.

Proper component choice also contributes to this, but you get your money's worth of high-quality components only if the design you start from is itself capable of worthwhile performance. Even the very best component, be it active or passive, is wasted money if the equipment it is used in is not up to it. A shrewd design with inexpensive components, cleverly used, will most easily outperform a modest, trivial design even if the latter is made with high quality parts costing several times as much.

Sorry for all you Silver Transformer WE-91 Loftin-Whiters: a side valve engine is always a side valve engine—regardless of magnesium cylinder blocks or titanium rods.

2. DESIGN CHOICES. MC OR NOT MC? THIS IS THE QUESTION...

What to do if what we are trying to pick up any music from is not a "vigorous" 5mV magnetic, but a shy .1mV moving coil? Let's go back to the previous example and pretend for a moment (just pretend, and just for a moment!) we try and add a further, flat response active stage in front of our MM preamp. .1mV is ANOTHER 34dB below 5mV. Nice, isn't it? In order to get the same level as before, it takes a 34dB preamp working between -78dBm and -44dBm.

It's IMPOSSIBLE to make it with no feedback AND decent S/N ratio—let alone sound resolution. If you have just one miserable microvolt equivalent thermal noise at the input—that's nearly -118dBm, and you might already need to be taken to a psychiatric hospital after you've been looking for a device that quiet—you're STILL getting an accordingly miserable 40dB S/N ratio, or, to put it another way, you're getting a heck of a noise out.

There's just one way to (apparently) get rid of this noise: F-E-E-D-B-A-C-K. Design a nice strong 74dB stage and stick 40dB feedback around it. With this foxy strategy, the input stage of this fantastic preamp of yours will be driven by just one microvolt out of the hundred your bold cartridge is so generously delivering, as the other ninety-nine are being, well, fed back.

With one microvolt you won't even convince a flea choir to sing "Wimoweh". Forget about driving any audio amplifying device.

The only thing that can successfully handle the signal from MC cartridges is a TRANSFORMER. If properly designed and made, it will introduce much less crap into the signal than any electronic means, and will provide excellent impedance matching. There's no point in wasting money on an expensive MC cartridge if you're going to mix mud with its signal.

By the way, beware of cheap MCs, most of them are over-resonant, over-distorted, are hard on records and sound compressed and metallic. If you're on a budget you'll be better off with a good magnetic. Try out the recent Audio Note IQ series, for example. I may look partial but I don't demand you take my word on this—try one yourself. These cartridges offer a much better combination of dynamics, low surface noise, and low overall distortion than any MC/transformer combination at comparable price. I myself have gone for an IQ II after ten years of MCs.

3. DESIGN CHOICES. EQUALIZING.

There are two main solutions to the problem of RIAA equalization: there can either be a passive network between two stages, typically where there is an RC coupling, or there amy be a selective feedback loop. They are called respectively—and quite intuitively—"PASSIVE RIAA" and "ACTIVE RIAA".

From the commercial manufacturers' standpoint, the "active" solution offers the important advantage that the shape of the response suffers less of the sample to sample variability of the active devices: this is why the great majority of preamp designers have gone for it in commercial products.

Fine for them, not for us. Obtaining RIAA equalization by selective feedback requires applying a feedback factor increasing nearly 36dB from 30Hz to 15kHz, whatever the starting value (i.e. the value at 30Hz). Not only do you get the loss of resolution and dynamic compression that go together with high feedback, but this effect is also frequency dependent.

I reckon the consequences this fact has on music reproduction are truly devastating. I'm not surprised so many people definitely took to CD and threw away all LPs in the Eighties, claiming they were hearing lots of distortion from LPs; in some respect, they were probably right. A great many of them had always been listening to LPs through active RIAA preamps.

Well, let's be sincere: active RIAAs were certainly not the ONLY cause for this. Over-resonant over-compressed cartridges, big under-resolving over-distorted amplifiers, horrible no-dynamic over-coloured speakers ALSO played their role in LP misappreciation—and still sadly do in modern "hi-end," look at how many spend megabucks in expensive D/A converters...

Nonetheless, LP playback through feedback EQ preamps remains a most frustrating experience regardless of the quality of the rest of the system. So, as far as I am concerned, active EQ can be thrown into the sewer—finest brown rat food—where it rightfully belongs.

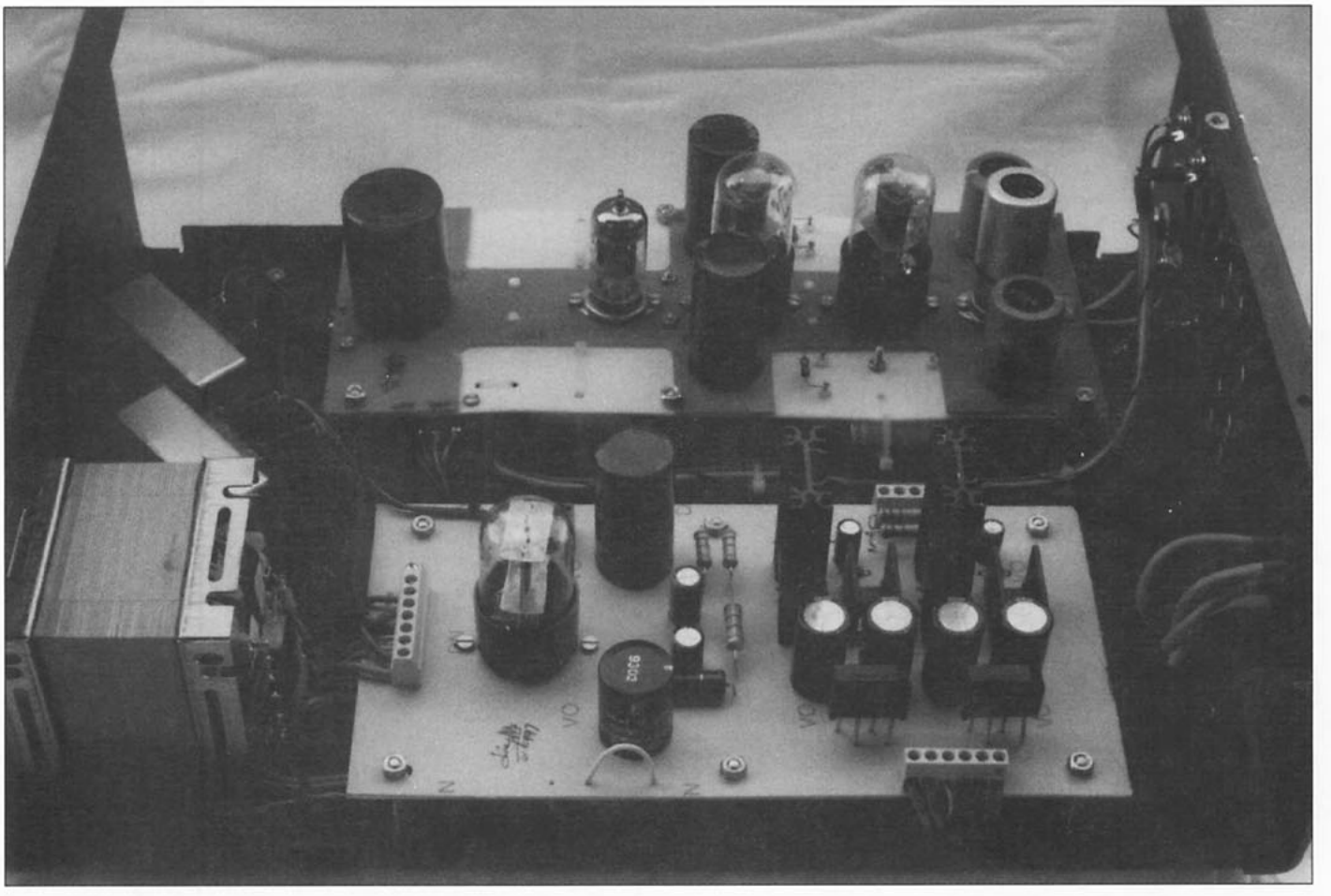
Attempts of making semi-passive EQs (usually two time constants in feedback, the third one passive) yield semi-results. Brown rat food with some sugar on. Not very worth wasting any time on, really.

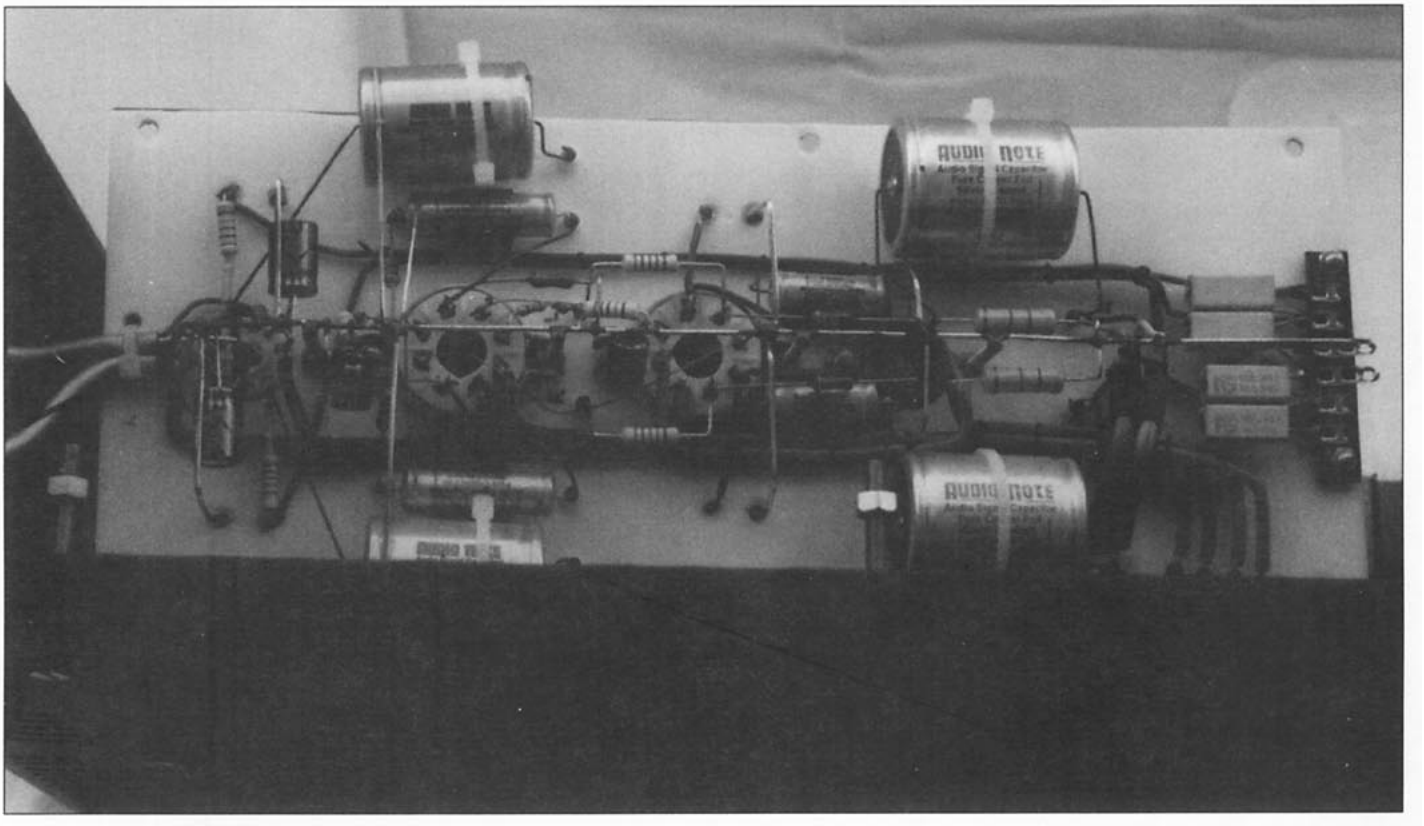
Entirely passive equalization is not an "alternative". It is the ONLY possible solution if you are even marginally concerned with sound quality.

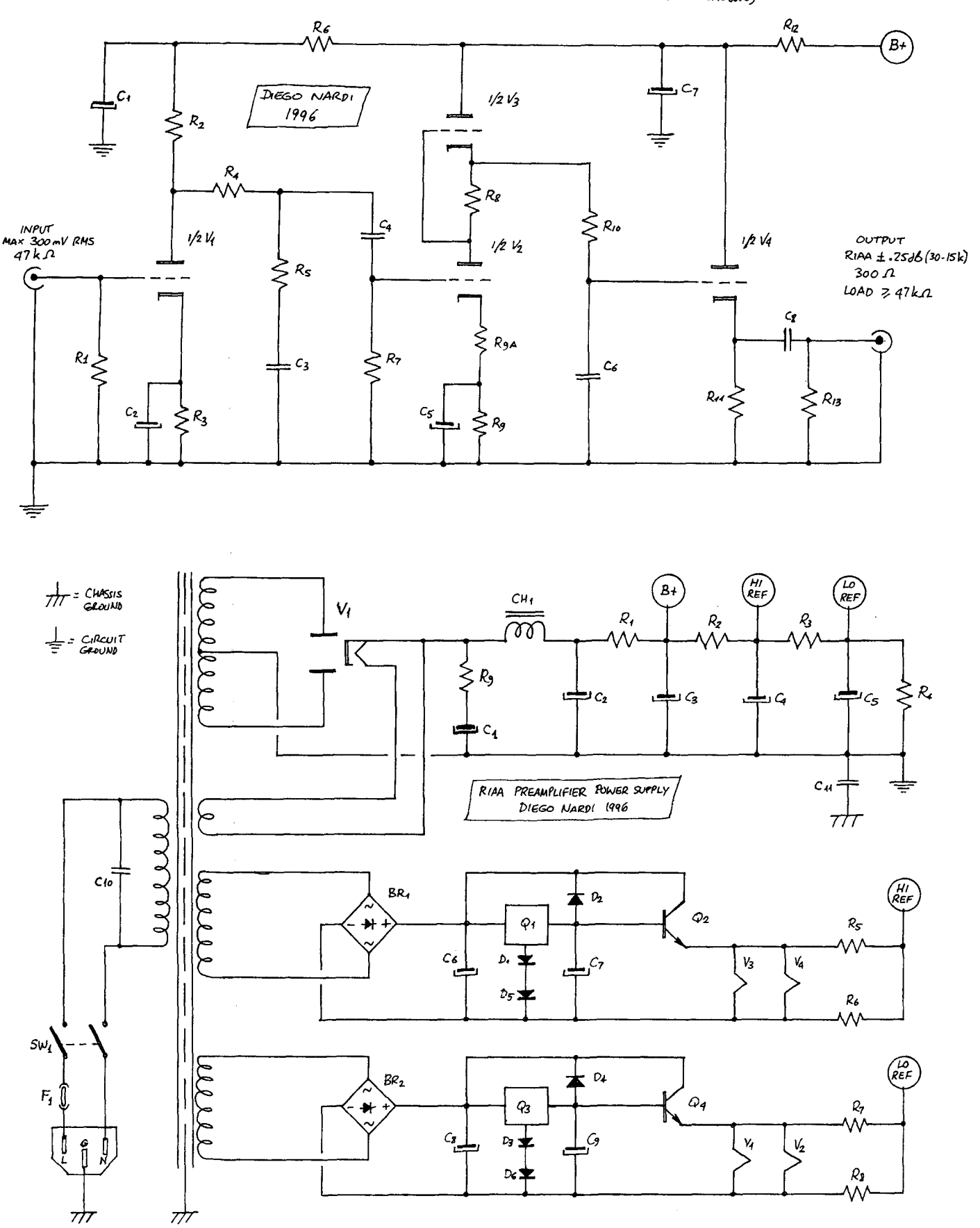
As I have anticipated, passive RIAA is generally obtained by placing a suitable network, implementing all the three required time constants (for further details, see again RIAA spec), plus, if desired, the fourth one provided for by the RIAA/IEC (basically a subsonic filter with -3dB point

Above–
General Layout of preamp

Below –
Audio circuit module featuring super-compact point-to-point construction technique






3-STAGE RIAA PREAMPLIFIER - AUDIO CIRCUIT (ONE CHANNEL SHOWN)
DIEGO NARDI 1996
INPUT MAX 300mV RMS 47kΩ
OUTPUT RIAA ±.25dB (30-15k) 300Ω LOAD ≥ 47kΩ
B+
1/2 V1
1/2 V2
1/2 V3
1/2 V4
= CHASSIS GROUND
= CIRCUIT GROUND
RIAA PREAMPLIFIER POWER SUPPLY DIEGO NARDI 1996
HI REF
LO REF

AUDIO CIRCUIT

Position	Standard		SL	
R1	47k		47k	○
R2	220k 1W		100k 1W	
R3	1k		680R	
R4	392k		150k	○
R5	27k 4		15k	○
R6, R12	1k 5W	●	1k 5W	●
R7	820k		470k	○
R8	2k 2		2k 2	
R9	not used		2k 2	
R9A	2k 2		330 R	❑
R10	75k		47k	○
R11	47k 2W	❑	47k 2W	❑
R13	2M2	❑	2M2	❑
C1	47uF 400V		47uF 400V	
C2	100uF 6.3V	■	220uF 6.3V	■
C3	.012uF 400V	▼	.022uF 630V	▲
C4	.022uF 400V	▼	.22uF 630V	▲
C5	not used		100uF 6.3V	■
C6	820pF 630V	▼	1000pF 630V	▲
C7	220uF 400V		220uF 400V	
C8	.33uF 400V	▼	.33uF 630V	▲
V1	12AX7/ECC83/7025		12AY7/6072	
V2,V3	6SL7GT		6SL7GT	
V4	6SN7GT		5687	

Notes:

Unmarked resistors are metal film, 1/4W except where noted.

○ = tantalum film 1/2W

● = wirewound, vitrified or ceramic base

❑ = low noise carbon film, 1/4W except where noted

Unmarked capacitors are electrolytic

■ = Black Gate "F" type

▼ = Audio Note paper in oil, aluminium foil

▲ = Audio Note paper in oil, copper foil

POWER SUPPLY

R1	2k2 5W	C7, C9	470uF 16V
R2, R3	33k 2W	C10	.1uF 250V
R4	1k5 2W	C11	.01uF 1000V
R5, R6,		CH1	25H@30mA
R7, R8	100R 2W	V1	6X5GT
R9	470R 5W	D1, D2, D3	
C1	47uF 400V	D4, D5, D6	IN4003
C2	220uF 400V	BR1, BR2	80V@4A
C3	470uF 400V	Q1, Q3	7806
C4	10uF 350V	Q2, Q4	BDV 65
C5	10uF 63V	SW1	DPDT switch
C6, C8	4700uF 16V	F1	T315mA for 220-240V, T630mA for 110-120V

Notes:

Unmarked resistors are carbon film.

● = wirewound ceramic base

★ = class X2 surge suppressor cap. Example: RIFA type PME 271 M. On the SL version, R1 is replaced by another choke, same as CH1. C11 may need to be replaced with a short—depends on how the mains grounds of the other equipment is connected.

at 20Hz) between two RC-coupled stages.

These three time constants are virtually always obtained by means of resistors and capacitors—no inductors. Inductors are not used not certainly because they don't work (they would, and very well at that), but mainly because of the inordinate inductance values necessary to get the required time constants with manageable overall impedance values. The only recent example of LCR RIAA Phono stage that I know of is the Audio Note M10...(of course next morning you'll go and buy one, won't you?)

Back to our tried and true RC EQ network. There's a big problem in designing in the "traditional" way, i.e. with the whole curve obtained with a single network. If the equalization is to be really "passive", it should have negligible effect on the inherent gain of the stage driving it, i.e. the input impedance of the network should be much higher than the output impedance of the active stage across the whole audio band. A network to comply with this has a very unpleasant side effect that the designer simply can't avoid: high insertion loss. On balance, a network having limited insertion loss would overload its driver stage at the higher frequencies. There's no real way out and all the designer can do is settle for a compromise, typically ending up with some 6dB attenuation at the beginning of the curve, and easily more than that.

If you privilege constant loading, you'll get smooth sound, but dynamics will inevitably be somewhat sacrificed. If instead you prefer low insertion loss, you'll get a dynamic but somewhat uneven and aggressive sonic presentation, and you might find that some dynamic compression is better tolerated after all.

A countermeasure that effectively mitigates this problem consists in splitting the RIAA network into two separate ones, and this is precisely what I did.

There has been another split-RIAA preamp described in SP, it was J.C. Morrison's "Siren Song" in #3, although, apart from this basic idea, it doesn't share any other solution with the φ42, even the order in which the time constants are placed in the circuit is different.

With this technique, I was able to limit the insertion loss of the first network to only 2.5dB, while the second one is placed in a direct coupling and its passband insertion loss is practically negligible.

The load variation on the first stage is such that the max. inherent gain variation is about .5dB between 50Hz and 2kHz, where the RIAA drops about 18dB.

ACTUAL CIRCUIT DESCRIPTION

I'm actually proposing two versions of this preamp. They differ in circuit parameters, and choice of both tubes and passive components; consequently, COST.

Both sound clear and intensely contrasted, with strong and neat dynamics and almost total lack of strain; the SL (Special Luxury) version however features better focus and a finer grain, tauter sound overall. Neither version sounds "mellow" and syrupy however. If you're looking for romantic, adulterated sound, don't go for this preamp as this is not what I mean by "lack of strain". I do not want any of my designs to have any sonic signature themselves; what I try to get is equipment that sounds different with every record. The φ42 is very much this way.

Sorry for this brief introduction that is not relevant to circuit description directly. It is meant to help you decide to read on or turn the page...

If you want to fully appreciate the above characteristics, however, I have to warn you that you MUST follow my recommendations regarding component choice as strictly as possible, and you MUST NOT mix up elements of the standard version with elements of the SL. The two versions have independently undergone a very painstaking fine tuning procedure and, if you want to re-start from the beginning by modifying anything, you can do it at your own risk but, as Herb Reichert says, it'll be your own preamp design, not mine. I won't accept any responsibility about the sound you might end up with. Don't forget that both circuit parameters and component choice do contribute to the final result.

The first stage is made of a high-mu, high impedance triode in a conventional plate follower arrangement, driving the network which implements the first "step" of the de-emphasis curve: R4-R5-C3 form a reactive divider which is substantially in parallel with the load resistor. C4 is chosen to complete the action of the divider in proximity of the low-frequency flattening-out pole (the 3,180 microsecond time constant), and start a subsonic filtering rolloff around 30Hz.

Please note that the low-end frequency characteristic of the SL is somewhere in the middle between RIAA and RIAA/IEC specs, i.e. it is subsonic filtered but slightly less than RIAA/IEC. The standard is very close to RIAA/IEC.

The standard and SL have different input triodes (12AX7/7025 on standard, 12AY7/6072 on SL) with matched loads and EQ networks.

The second stage of both versions is made of a 6SL7, SRPP connected and DC coupled, via the 75 microsecond network, to the cathode follower output which is based on a 6SN7 in the standard, a 5687 in the SL.

The 6SL7 stages of the standard and SL differ in cathode degeneration, DC operating point and EQ network. Apart from the split RIAA, which is not very conventional, the circuit is therefore quite conventional and its main feature, which you can't tell from the circuit diagram alone, is the care taken in the choice of loading/operating conditions of every part of the circuit.

The input impedance is 47k ohms. The great majority of hi-output cartridges and MC matching transformers will work fine into this load. Input capacitance is about 120pF plus wiring capacitance in the standard version, 60pF in SL.

Resist the temptation of building those wicked 1980s style DIP-switched "load matching" R-C banks. This stuff is as good as active RIAAs (sewer rats'll love it).

The output impedance is low, about 300 ohms in standard and 200 ohms in SL, but the output stage is not designed for very low loading. Optimum load impedance is 47k ohms or higher—25k ohms is tolerated. Overloading the output will narrow the available dynamic headroom and, although this may seem quite immaterial, you run the risk of getting a "tired" sound. It should be noted that no sensible line stage has less than 47k input impedance, so this is no limitation in fact.

THE POWER SUPPLY

Power supplies are always very critical to overall performance, and this case is no exception. The specific demands of this circuit are quite different from those in the WOT: there I had a single stage working at a rather low voltage, which I wanted to be able to modulate relatively significant amounts of current; here there are instead three stages, all of them running into high controlled impedances and modulating modest currents. As a consequence, very much smaller values of capacitance, compared to those in the WOT, are sufficient to provide a "quasi battery" operation of the HT supply. It is, on the other hand, vitally important to obtain a clean, low ripple HT source to feed the circuit from, and that's why I went for a twin-pi preliminary filtering, which is then split into two separate decoupling rails, one for each channel.

The rectifier is again a 6X5GT, connected in the same "pseudo-direct heating" way already used in the WOT.

The resistor in series with the filter input capacitor has the sole purpose of limiting the current surge through the rectifier at switch-on, which would otherwise slightly exceed the peak current rating for 2-3 seconds about 15 seconds after switch-on. This may occasionally result in some sputtering which, although not destructive, is unpleasant to see. When the power supply has settled, about one minute later, the rectifier runs under very relaxed conditions, with a 60mA repetitive peak current per plate and a 50% overall duty cycle.

It is absolutely necessary to provide two completely independent heater supplies, in order to ensure correct heater to cathode voltages within all tubes. I decided not to make dual supplies, as I did in the WOT, partly because they should have been +/-3V instead of +/-6 (6SL7 can't light up on 12V), therefore difficult to regulate, and partly because they would have taken too much room and would have complicated the power transformer too much. By the way, the latter is a proprietary, custom unit, specifically designed for this preamp. You don't find it in parts catalogs by any supplier.

The heater regulators are transistor buffered 7806s for maximum noise rejection. Both supplies have "artificial" center taps connected to an auxiliary HT divider supplying the relevant reference voltages. All of these measures contribute to avoiding any heater-induced noise.

It is sad to notice how underestimated the problem of correct heater arrangement has been, and still is. There are a lot of preamps in the market, even very expensive ones, that have "structural" noise problems due to incorrectly organized heater supplies. A very typical mistake is connecting the two sections of the same twin triode as an SRPP stage or as a direct-coupled voltage amplifier/cathode follower pair. In this way you will almost invariably get "whistling" or "hissing" type parasitics, regardless of the potential of the common heater. This is due partly to the cathode/heater capacitance and leakage current, unduly and uncontrollably connecting together two cathodes supposed to be independent (electrical coupling) and partly by the two triodes ping-ponging any mechanical vibration between each other (mechanical coupling).

It wouldn't be that difficult to use the two triodes within each envelope as the same section of the two channels, instead of as the two sections of the same channel. If the two co-tenant triodes are made to work at the same potential,they practically stop fighting with each other, because the grid-to-grid capacitance is fractions of a picofarad, and if there is little or no potential difference across the K1/H/K2 impedance, little or no current will flow between the two cathodes! In addition, if two cathodes working at largely different potentials are heated by independent heaters, you can separate the latters' power supplies thus making it possible to set correct heater/cathode voltages in each section...simple, isn't it? But no manufacturer seems to care, really...and the user has to live with his whistles.

Reacting with the decision of not using SRPP or cathode followers or direct couplings would be an ultimately stupid way of sidestepping the problem in my view.

COMPONENT CHOICE

After having spent long, sleepless nights studying and experimenting circuit solutions and parameters to give the least distorted and most convincing sonic results, it would be utterly foolish to spoil all this hard work with an incorrect component choice. I repeat once again: NO component, be it a tube, a capacitor or a transformer, of whatever finest quality, will ever turn a toad into a prince: you need an inherently high performance design FIRST. But then components must be competent with it if you are to exploit its possibilities.

A very critical thing here is capacitors, especially those in the circuit. If you want to avoid the risk of bitter disappointment after building this preamp, you need to follow my advice and go for Audio Note's paper-oils. Standard version is intended for being equipped with the alu-foil type, SL instead requires the copper foils.

If you say these caps are madly expensive, I will answer I also think so, but you're stuck with using them if you really want lack of distortion and dynamics, and at

Audio Note they know this full well; so all that's left for you to do, whatever you are planning to build (not only this preamp), is resign. Can't get Audio Note caps? No problem, please feel free to contact me personally and I will be glad to help you on this.

The cathode bypass caps are also critical. The φ42 was developed with, and specified for, Black Gate F, an affordable version for your joy and delight. Since the complete set of cathode BGs costs less than a single alu-foil paper oil, it is totally senseless to resort to normal electrolytics, with, maybe, some fancy horrible sounding French style set of bypasses across them...

Can't find the required values in Black Gate F? Please read above.

Filter and decoupling electrolytics are all normal aluminium types in both versions. These also must NOT be bypassed in any way for any reason. As I have already written and will keep on repeating, bypassed electrolytics sound like cat piss and are to be avoided like AIDS.

Please note that all capacitors, and very especially Black Gates, demand some run-in time for their dielectrics to mechanically and chemically settle. Remember there is a varying electrical field, and consequently mechanical forces, between the foils of a capacitor. Therefore, you should never attempt critical listening of any brand new piece of electronics, or of a used piece of electronics after a cap change if the cap under test is brand new—this also applies to tubes, transformers, coils and speaker drivers—or you will easily end up with grossly wrong conclusions!

Resistors in the standard version are all normal 1% metal films except for a few low noise carbons, and some, largely derated, wire wounds in decouplings. The SL features some tantalum films in critical spots.

Now to the delicate topic of chassis signal wiring. Input wiring must, first of all, be as short as possible in order not to uselessly increase the input capacitance; output run may be longer, within reason, depending on chassis shape and size.

If you decide to go for Audio Note wiring, as I not surprisingly do, recommended cable types are AN-S for standard, AN-V for SL. It is not cost effective to equip the SL with the more expensive AN-Vx; with the same extra money you get more sonic benefit by upgrading all cathode caps to Black Gate FK or by using a silver foil for C4. How much the extra expense would actually be obviously depends on the length of your cable run.

The AN interconnects are themselves balanced. To use them as internal wires in this preamp, connect them as follows: all input and output RCA sockets must be insulated from chassis; on both input and output cables, use the red internal conductor as "hot", the white one as ground return; on input cable, connect the copper screen at circuit ground end; on output cable, instead, separately connect the screens at RCA connector collars.

CONSTRUCTION

This preamp is best built on two separate modules, but NOT in separate chassis. The first module should support the heater supplies, the HT rectifier and the first two HT filter caps; the other one will then support the circuit with all HT decouplings. This eases handling and rationalizes wiring. I recommend the first module to be a true PCB with the heater supply routes etched on (transistors ain't very suitable for hardwiring). The best material is the good ol' smelly Bakelite if you can still find some.

The second module is best point-to-point hardwired with bus grounding—avoid star grounding. An alternative to bus grounding consists in using another PC board. Etch a ground plane of suitable size on it, then hardwire the circuit making all grounds to the ground plane, which will then be connected to the power supply ground through a single path.

Be neat, don't extend component legs—you won't need to do it if your layout is correct—and make provision for all solderings to have a sound mechanical support. (On the pictured board, there are a couple of points in which I had to join component legs "on air"; that was a "last minute" mod, don't take this technician's shortcut as an example!)

Diego Nardi FREE BONUS 6SN7 SRPP Line Preamp

Parts List

R1–R2	1k 1/2W
R3	470R 1/4W
R4	1M 1/4W
R5	33k 1W
R6	1k 5W wirewound

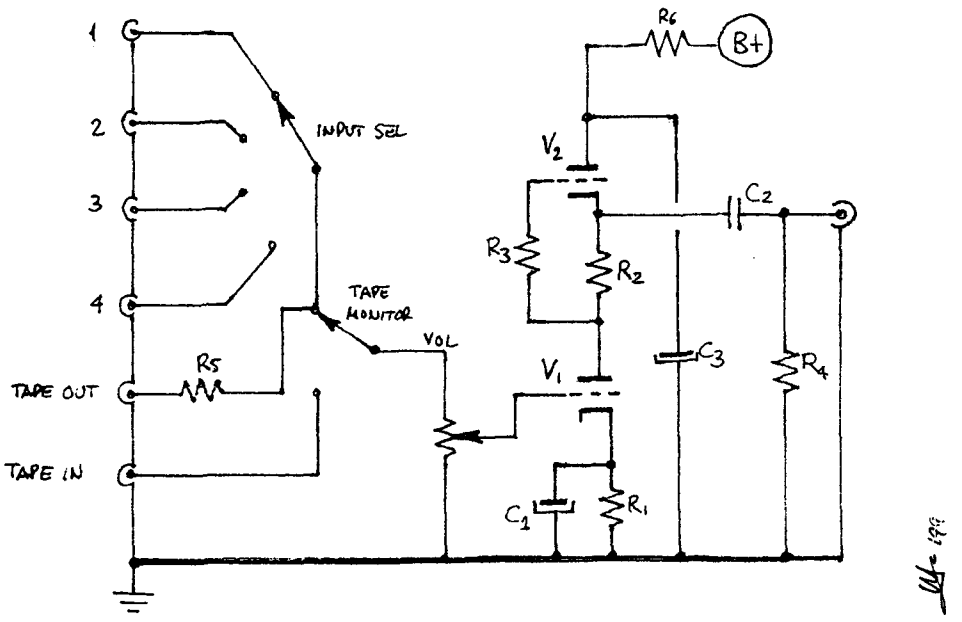
All other resistors carbon type

C1	220 mF/16V electrolytic or Black Gate type
C2	0.47 – 1 mF 400V Audio Note paper/oil (copper or alu foil)
C3	220 mF 400V electrolytic

Volume	50k or 100k B Taper
V1–V2	6SN7 GT/GTA/GTB

Power supply

Refer to phono schematic. B+ is taken from the positive terminal of C2.V1 heated by LOREF supply,V2 heated by HIREF supply.



There's no need for many soldering stand-offs in this circuit, most joints can be made directly at socket pins, decoupling cap pins and at ground bus with the exception of very few points; the use of tagstrips or other similar devices can, and should, be reduced to a strict minimum. This can be done at no sacrifice of order and reliability.

BONUS LINE PREAMP!

Aha. This is for you, Mr. Igot Nomoney. You have just made a list of the parts you need to build yourself a φ42, and you discovered you are ALMOST going to run out of money? And, by chance, you ALSO need a line preamp?

No problem. Here's a suggestion for a super-simple, very good sounding line stage you can very readily stick to your α42 using its in-board power supply.

All you have to add is a pair of 6SN7s, a volume pot and a few other components. The 6SN7s, arranged in a classic SRPP mode, form an excellent line amp, yielding a gain of about 23dB into 100k ohms with around 2k5 output impedance and a max out level of over 60V RMS across a quarter megahertz bandwidth. No, it doesn't sound quite as good as a nice new WOT, but it's still better than nearly everything you can find at audio stores regardless of price, and it's cheap, WOT the hell!

It takes both heater and HT power from the existing supply and you can also run "backwards", i.e. first you build yourself the power supply and line amp, then you add the φ42 Phono board. The 6SN7 SRPP line stage is in fact an old favorite of mine, and I have proposed it several times in the Italian press, the last time was in the September 1995 issue of *Costruire HiFi* magazine (the first one dates back to 1991). It always receives great acclaim after people try it out, and an earlier version of the φ42 was in fact published in the same magazine as a Phono stage to go with the SRPP line! Therefore, the two are tried and true and go great together. Have fun!!!

Contact Info

Diego Nardi
Via.Volterra 12
20146 Milano, ITALY

Costruire Hi-Fi
Mozart Editrice S.r.l
Voc. La Castagna
05024 Giove (TR), ITALY
FAX 0744/99.40.34

Design and Easy Calculation of a Regulated Power Supply for a 300B Stereo Amplifier

J.C. Verdier

Laboratoire JC Verdier

The job of the power supply in a push-pull amplifier is facilitated by a symmetrical arrangement of the output devices that eliminates residual hum by cancellation. Moreover, if the push-pull amplifier operates in true Class A the current flow is constant and the power supply voltage does not fluctuate with the rhythms of audio modulation. Class A push-pull amplifiers present a relatively simple challenge from the standpoint of power supply requirements.

It is a completely different story with the single-ended triode amplifiers currently in fashion. In these amps, the residual hum is applied to the output transformer through the internal resistance of the tube (see Fig.1). If we presume that the residual hum is 100 mV, a hum voltage on the order of 4.5 mV will appear at the output of the amplifier. Of course, this is not tolerable, especially when high efficiency loudspeakers are employed.

The classical cure is to add more filter capacitance, which can work well, or to add more inductance in the choke, which could be bad if it raises the internal impedance of the power supply. The consideration of internal supply impedance leads us to the second point of the argument, by far the most important.

Let us look at the operation of a single ended output stage when driven almost to the point of clipping by a large sinusoidal signal. The positive half of the signal applied to the grid increases the current in the tube and creates a condition of maximum power dissipation in the supply. Then the negative signal causes the current to decrease until it is nearly cut off and at this moment the power dissipation is again at maximum.

Where does the power come from? Obviously, the power is taken from the energy stored in the primary inductance of the output transformer.

$$Q = 1/2 \; LI^2$$

There is an analogy in the work of the coil ignition of an automobile. In order to do a good job, the output transformer must have a substantial inductance, a well-known fact, but the power supply must also be able to reload the primary inductance effectively.

In your car, no problem. The battery is husky and the charging current gushes forth. In an amplifier, it is easy to be far from the mark. For instance, a reservoir capacitor of 100 microfarads will have an impedance of 80 ohms at 20 Hz. That is *not* negligible. (see Fig.2, the measured impedance of a 100 microfarad capacitor).

The regulated power supply can cure both of these problems, by decreasing the hum below the thermal noise level and providing an internal impedance lower than 1 ohm at all reproducible audio frequencies.

Principle of the Regulated Supply

In our regulated supply, the rectification and filtering are achieved in the classical way but the high voltage current is supplied to the amplifier through the regulator tubes, called the *ballast*. (see Fig 3).

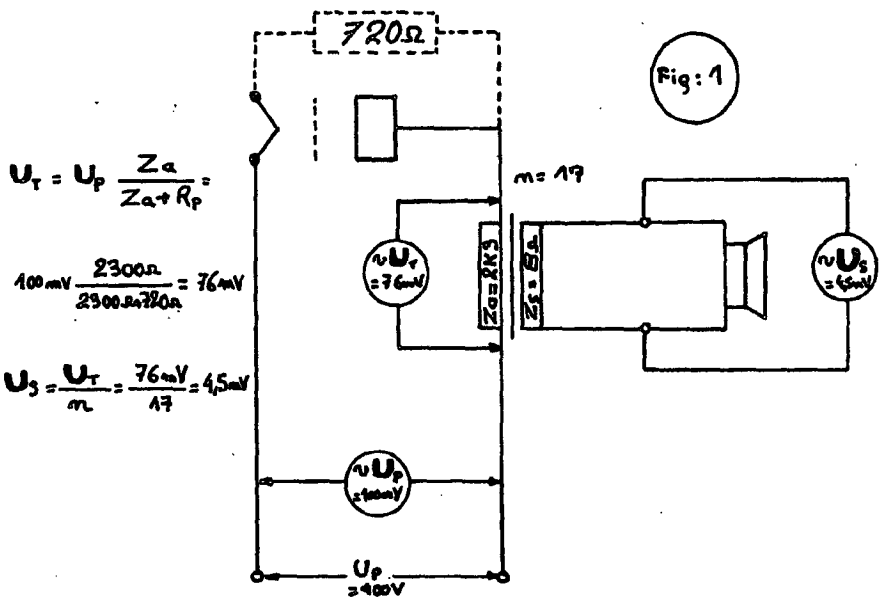
The ballast in the regulator under discussion consists of several low plate resistance triodes (6080, 6AS7, 6336) connected in parallel. The triode grids are controlled by a differential amplifier which compares the instantaneous value of the high voltage to a reference voltage provided by zener diodes or a voltage regulator tube (85A2, 0A2, 0B2, etc..

From moment to moment, the variations of the high voltage supply are corrected through the action of the regulator. Since nothing is perfect, there is always a residual hum and some voltage variation dependent on the load. Let us examine how we might keep these inevitable failings as negligible as possible.

Design of a 400V—250mA Supply

Figure 4 gives the diagram of a simplified regulator minus the various artful devices which we will add to achieve the desired quality. The parts shown in the block diagram of Figure 3 are replaced by electronic components:

The ballast tube T1 is the 6080 dual triode, a tube type specifically designed for



this application.

The T2 tube, EF 184, is a sharp cut-off pentode normally used as an RF amplifier, but also perfect for the job here.

Resistors R2 and R3 feed the screen grid of the pentode and provide the current for the zener diode voltage reference source. R1 is the load for the pentode.

Potentiometer P1 acts as a voltage divider to supply a part of the high voltage to be compared with the reference voltage.

Let us move on to a practical exercise, a regulated power supply to power a stereo 300B amplifier with a required voltage of 400V and a current delivery of 250 mA.

Calculation of the Ballast (T_1)

Figure 5 shows the basic system with one triode section of the dual triode 6080 illustrated. Since these tubes have a control grid with a tight pitch placed very near the cathode, the consequent grid current can be very inconvenient. This is especially a problem with old tubes.

Therefore, it is prudent to maintain the instantaneous voltage of the grid very negative and that prescribes the first limitation: The bias will have to be above –40V. That means that the voltage between plate and cathode will have to be no less than 100V in the worst case, that is to say when the line voltage is decreased by 10%.

The necessary voltage after rectification and filtering, therefore, must be:

$$U_F = U_R + U_{T1} =$$

$$400V + 100V = 500V$$

Now we find that, when the line voltage is increased by 10%, the rectified voltage will reach

$$(500V \times 120\%)/100\% = 600V$$

In this case, the voltage between plate and cathode of the ballast tube will be:

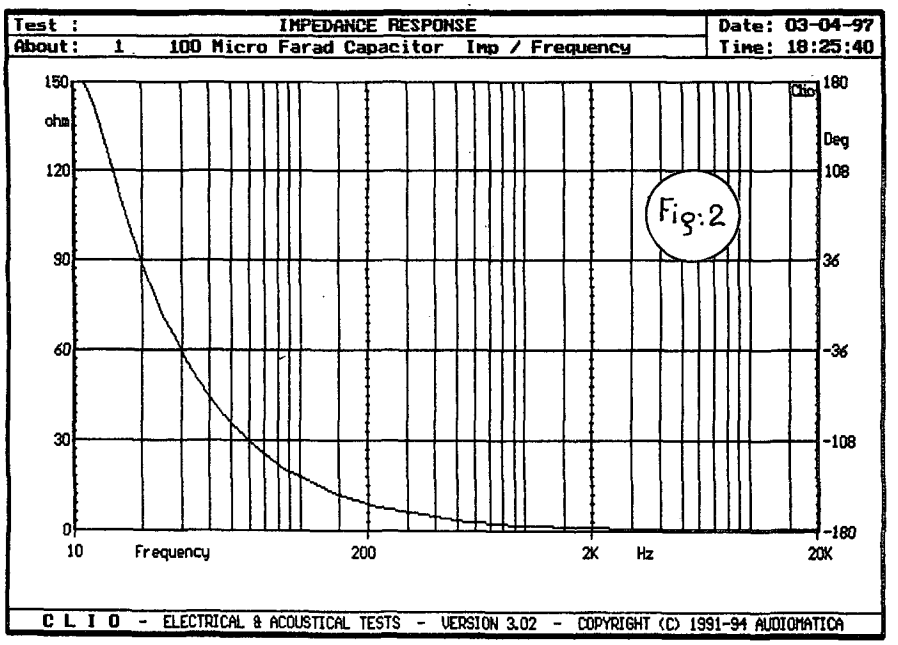
$$600V - 400V = 200V$$

That gives a dissipated power of

$$200V \times 250\ mA = 50W$$

for a current consumption of 250 mA.

We can see immediately that this power requirement, although a rarely seen extreme case, forces us to use several 6080 tubes in parallel because each half of a 6080 can only dissipate 1 W. Two 6080s (4 triode sections) are good for 52W. This is just enough, without any margin of safety.



An improvement to the basic scheme of Fig. 4 consists of fitting a resistor in parallel with the 6080 to handle part of the current (see Fig 5).

The choice of value of this resistor must be made with the 250V plate-cathode voltage limit of the 6080 in mind. A higher voltage can arc over, causing damage to the cathode coating.

Note also that the 300B tubes the supply is powering are quick-heating filamentary types and the 6080 are slow-heating cathode types. Without the parallel resistor R4, the voltage after turn-on across the ballast tube would reach the totally unacceptable value of 600V in the worst case situation of 10% high line voltage.

Since the normal current flow of a pair of 300B tubes is about 150 mA, R4 will have to drop 250V maximum at this rate of current flow.

$$R4 = 250V/150mA = 1k\Omega7$$

The current through the 6080 will be notably decreased and will go down as the line voltage goes up, thereby helping to regulate the power dissipated by the ballast tubes.

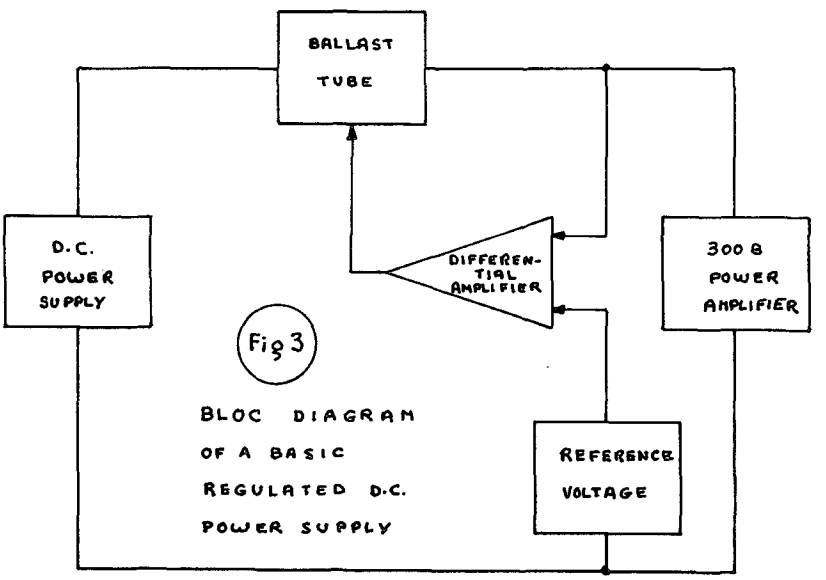
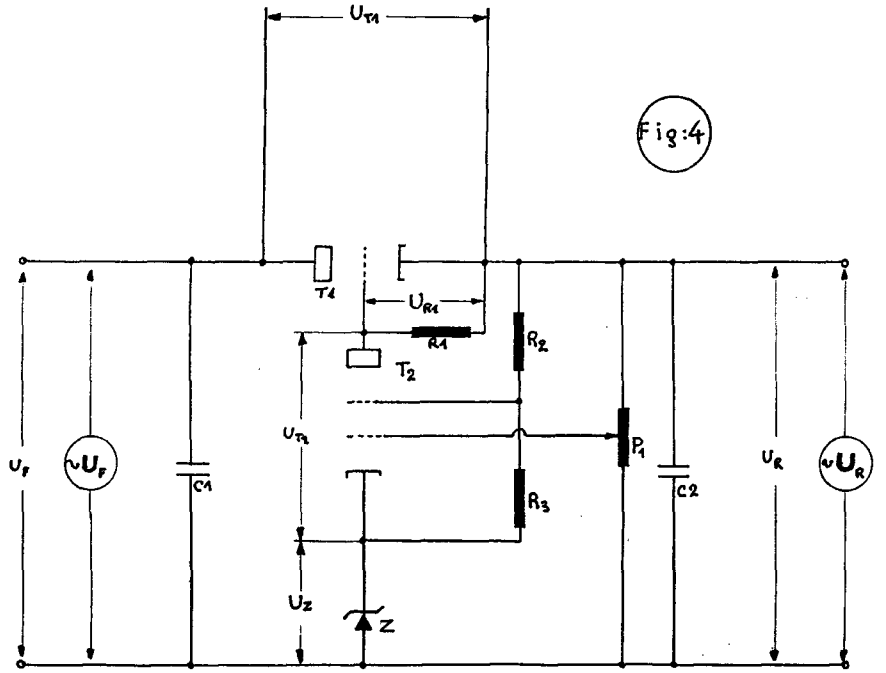
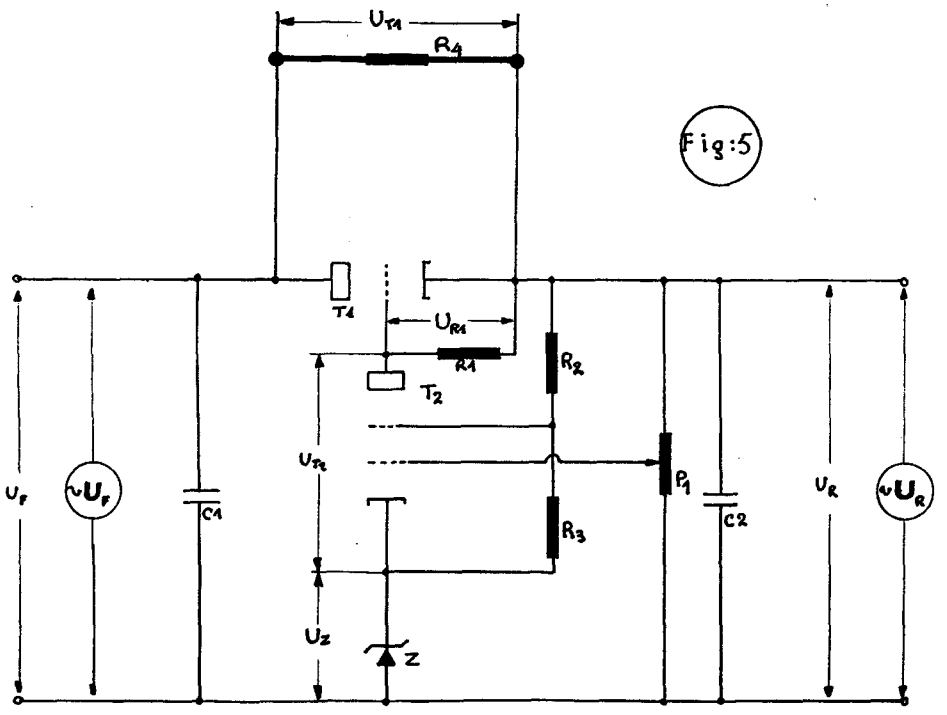


Fig 3

BLOC DIAGRAM OF A BASIC REGULATED D.C. POWER SUPPLY





So for a minimum line voltage (-10%)

$I_{R4} = U_{T1}/R4$

$= 100V/1k\Omega7 = 59mA$

and the dissipated power in the tubes will be

$P_{T1} = 100V \times 191\ mA = 19W1$

for a nominal line voltage.

$U_F = 550V$, $U_{T1} = U_F - U_R =$

$600V - 400V = 150V$

$I_{R4} = U_{T1}/R4$

$= 150V/1k\Omega7 = 88mA$

and the dissipation of the 6080 will be

$P_{T1} = 150V \times 162mA = 21W3$

For the case of maximum line voltage (+10%):

$U_F = 600V$, $U_{T1} = U_T - U_R$

$= 600V - 100V = 200V$

$I_{R4} = U_{T1}/R4$

$= 200V/1k\Omega7 = 117mA$

the current in the 6080 will be

$P_{T1} = 200V \times 133mA = 26W6$

It appears that on the basis of power dissipation alone, one could be content with only one 6080. However, prudence commands to use two tubes for security's sake and for two other reasons.

1—Two tubes improves the regulation from the standpoint of residual hum and lower internal impedance of the power supply.

2—The variation between 6080 tubes is astonishing. It is an idle dream to find two tubes that are matched between sides and between tubes. Therefore, we must also wire in balance resistors in series with the cathode of each tube. The chart in Fig.6, published by SOVTEK gives useful information. For four triode sections in parallel in the absence of precautionary measures, the maximum power handling of each tube is not more than 7W4 and 29W6 for the quartet, thereby erasing our security limit entirely.

Installing 100 ohm series resistors in series with each plate improves the situation such that the power handling becomes

$9W5 \times 4 = 38W$

This arrangement is the second improvement we propose (see Fig.9).

Design of the Differential Amplifier

The transconductance of the EF184 is 15000 micro-mhos in normal use. Here, the transconductance is highly reduced owing to the very low current flowing through the pentode in our special connection. Since the differential amplifier escapes calculation, it will have to be estimated empirically, but we are mainly interested in the gain (and regulation factor) we obtain. Maximum gain will be determined empirically by varying R1 using a 1MΩ potentiometer.

On the other hand, The values of R2, R3, and the Zener reference voltage can be determined by calculation, considering the current through the pentode as negligible and without any practical effect.

Zener Reference Voltage Choice

It is a compromise. The higher the voltage, the more of the potentiometer will be in circuit.

Since $U_R = U_{R1} + U_{T2} + U_2$, the Zener voltage is a compromise between the voltage drop across R1, the voltage between the plate and cathode of T2, and the voltage on the Zener diode. Given the realities of the components that we are using, we have to provide a reasonable operating voltage for the pentode. In view of this requirement, the error voltage applied to the grid of T2 through P1 must be limited to a certain value, even if the efficiency of the regulation is thereby decreased.

What is the voltage on the plate when the power supply is at nominal line voltage? We previously determined that in this condition, the current through the 4 x 6080 would be 162 mA (40.5 mA in each triode) and the voltage between cathode and plate would be 150V.

Referring to the curves of Figure 8, it is easy to determine that the grid voltage should be –70V. therefore, the plate voltage of the EF 184 is

$$U_R - 70V = 400V - 70V = 330V$$

If we consider that it is wise to keep about 200V between plate and cathode on the EF184, we have

$$330V - 200V = 130V$$

for the reference Zener voltage. Since 62V Zener diodes are readily available we can use two in series to obtain

$$U_Z = 62V \times 2 = 124V$$

The values of R2 and R3 will determine the current through the Zeners. It is necessary to be very prudent in this, because zeners definitely don't like heat. The wattage rating given by the manufacturer must be severely derated.

We use B2X 85C diodes and they are specified for 1W3 or about 2.5 W for the two diodes. Taking a factor of 5 as a security coefficient, we would only dissipate 0.5W for the two diodes and 0.25W per diode. For a half-watt dissipation, the current should be

$$P/U_2 = 0.5W/126V = 3mA9$$

$$R_2 + R_3 = U_R\text{-}U_Z/3mA9 = 70K\Omega$$

We can choose to use 2 resistors of 39KΩ each, yielding a power dissipation of 0.22W per diode at 3.5 mA.

Resistance in cathode circuit of each triode separatly, Ω	Number of triodes at parallel operation											
	1		2		4		6		10		over 10	
	I_A mA	P_A W	I_A mA	P_A W	I_A mA	P_A W	I_A mA	P_A W	I_A mA	P_A W	I_A mA	P_A W
0	130	13.0	93	9.3	74	7.4	68	6.8	64	6.4	56	5.6
50	130	13.0	101	10.1	87	8.7	82	8.2	78	7.8	72	7.2
100	130	13.0	106	10.6	95	9.5	90	9.0	87	8.7	82	8.2
150	130	13.0	109	10.9	100	10.0	96	9.6	94	9.4	89	8.9
200	130	13.0	112	11.2	104	10.4	101	10.1	98	9.8	94	9.4
250	130	13.0	114	11.4	107	10.7	104	10.4	101	10.1	99	9.9

I_A — anode current of one triode
P_A — anode dissipation of one triode

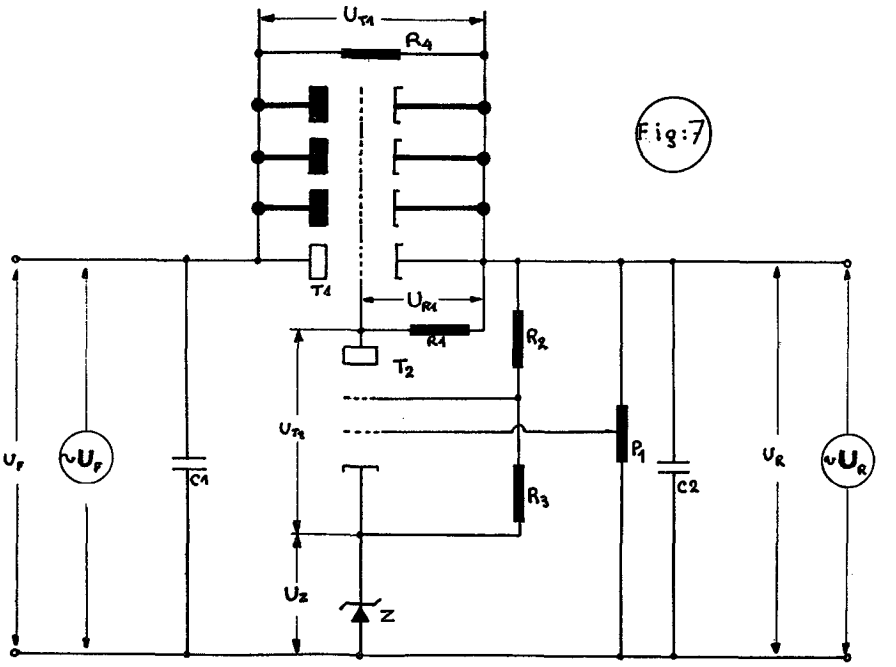
Fig:6 REPRINT FROM SOVTEK 6AS7 DATA SHEET

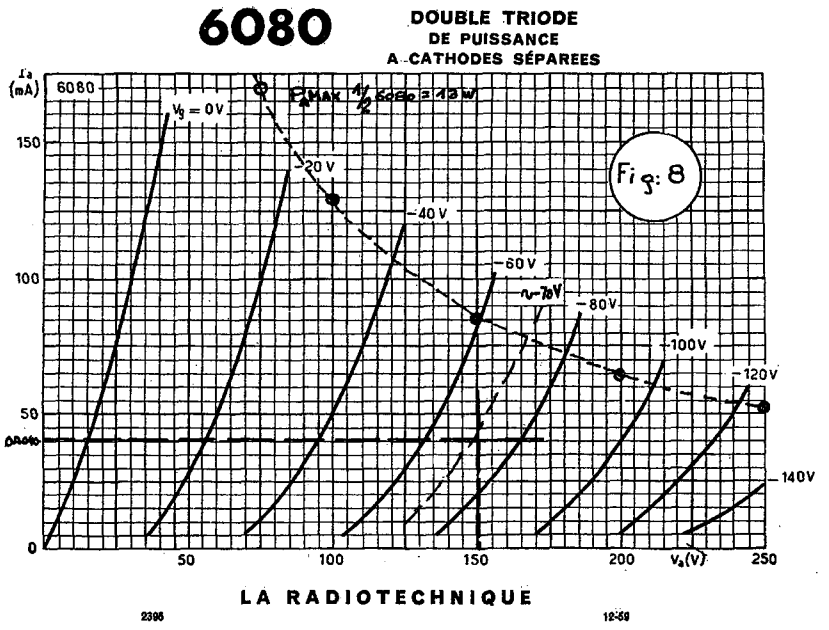
I believe that modern Zener diodes are better than glow discharge VR tubes in terms of noise and internal impedance. However, it is absolutely necessary to bypass them with a high value capacitor. We use a 100 microfarad electrolytic capacitor (C4) paralleled with a 0.47 polystyrene. This is the third important improvement on the basic circuit of Fig. 4, as pictured in Figure 10. The bypass capacitors eliminate residual hum on the regulated output and should be considered absolutely essential. It is not possible to adequately bypass a glow discharge VR tube because a VR tube and capacitor in parallel has a tendency to provide an excellent generator of sawtooth waves!

Choke Filter Elimination

The above discussion does not address rectifier and filter problems coming upstream of the regulator circuit we describe. A good regulator allows us to forego a choke and use only one capacitor after the rectifier.

The rectified and filtered voltage U_F is wired to Grid #2 of the EF 184 pentode



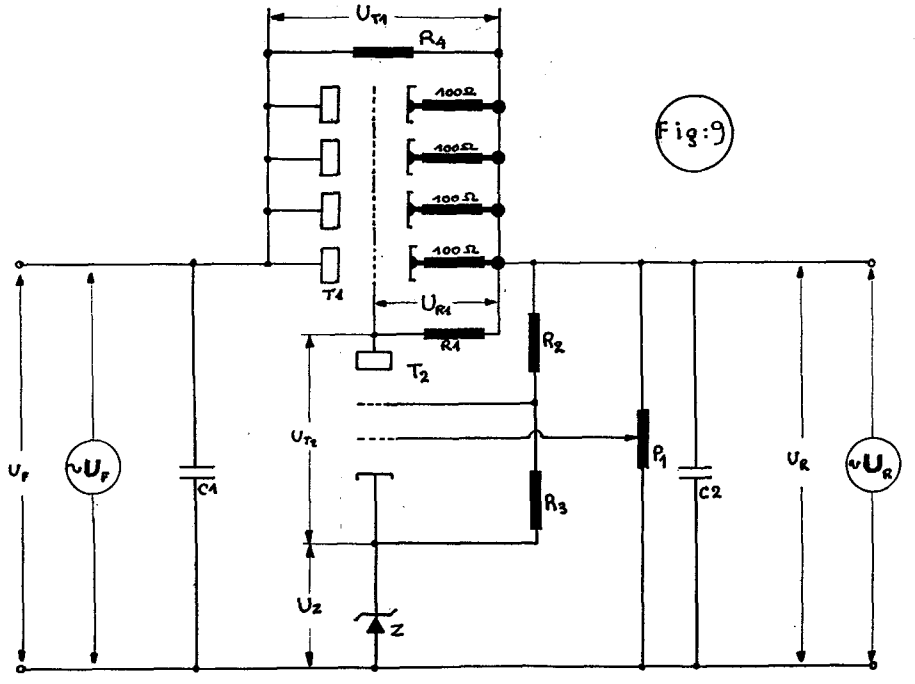


through C5 and adjustable resistor R5. This capacitor should present a negligible resistance at 100 Hz (120 Hz for the USA) We use a 0.47 for C5. There is no sensible progress in increasing the value. In any case, R5 allows a fine adjustment of the compensation. This is the fourth improvement toward making the best of electronic regulation, shown in Fig 11.

Lastly, there is another simple way to improve the regulation. We install a capacitor C6 between the upper terminal of potentiometer P1 and its moveable contact so that any variations in U_R are conducted directly to the grid of T2. This represents the fifth and last improvement we propose (Fig. 12).

Results

All those efforts to put this regulated supply into place truly pay rewards. The benefits are clearly evident in the measurements, taken in actual conditions of use powering a stereo 300B single-ended amplifier and 2x 6BQ5 in the input stage.The actual current flow is 250mA with the output voltage at 400 V.



Controversy on R1 Disposition

Some writers have indicated other solutions to power the load resistor of the pentode tube, R1.

1st Variant: The resistor is connected before the regulation on U_F.

In this case, the voltage U_F is always more than U_R, therefore the current is increased through the pentode and, consequently, its transconductance is increased. And, the efficiency factor is increased also.

This is an advantage, but unfortunately, there is the problem of hum which does not exist in the U_R connection.

2nd Variant: The resistor is connected to an increased external voltage source.

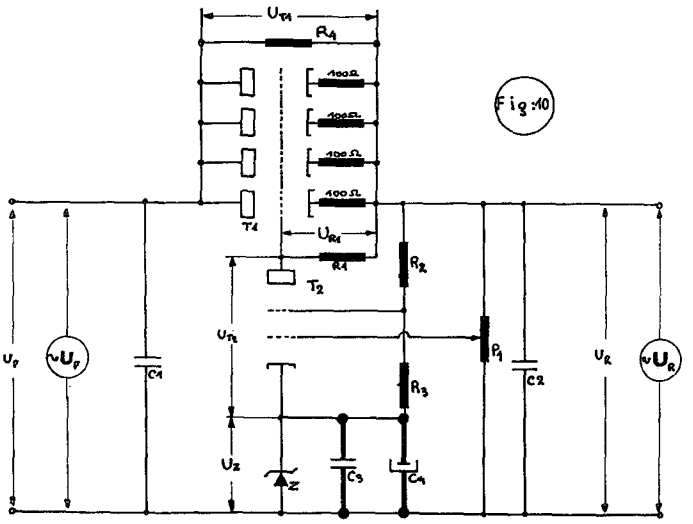
If the auxiliary voltage is very well-filtered, the result is indeed improved. However we have to pay for this improvement with a substantial increase in complexity if the auxiliary supply is not already in place. Therefore, we chose not to adopt that strategy.

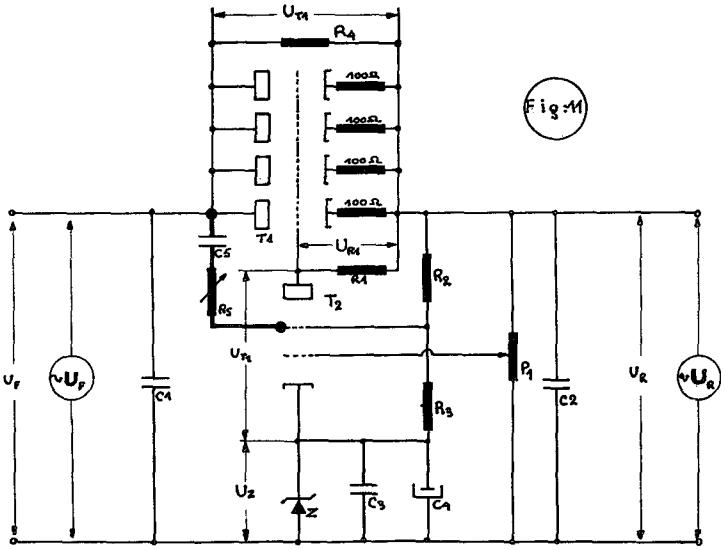
The curve shown in Figure 13 indicates the regulation area. The voltage on the primary of the power transformer is varied with an auto-transformer. The voltage is stable at 400V under line voltage variations between 180V and 250V (and probably more, but 250V is the limit of our auto-transformer). This 20% security zone is very comfortable, even though we may never use it in practice.

Experience shows that when the hum stays lower than 10 mV, the signal to noise ratio of the amplifier is not seriously affected. This hum level becomes evident with a line voltage decrease of about 12%. Taking this as the limit of the regulator, the security zone of the regulator is still quite considerable.

The practical results demonstrate the reliability of the calculations and suggest that the method outlined above would be useful for the design of other power supplies.

Finally, the measurement of internal impedance as a function of frequency given in Fig. 15 is most pertinent. The





amplifier is driven to maximum power by a sine wave generator between 20 Hz and 10 kHz. The millivoltmeter is connected to U_R through a 100 microfarad capacitor, large enough not to affect the frequency response of the test setup.

At 20 Hz, the internal impedance of the power supply is less than 0.35 ohms. To obtain the same result with a capacitor would require 22,000 microfarads. This demonstrates that a regulated power supply offers a large improvement and at a cost that is practically the same (considering that the filter choke is removed). The result is near perfection.