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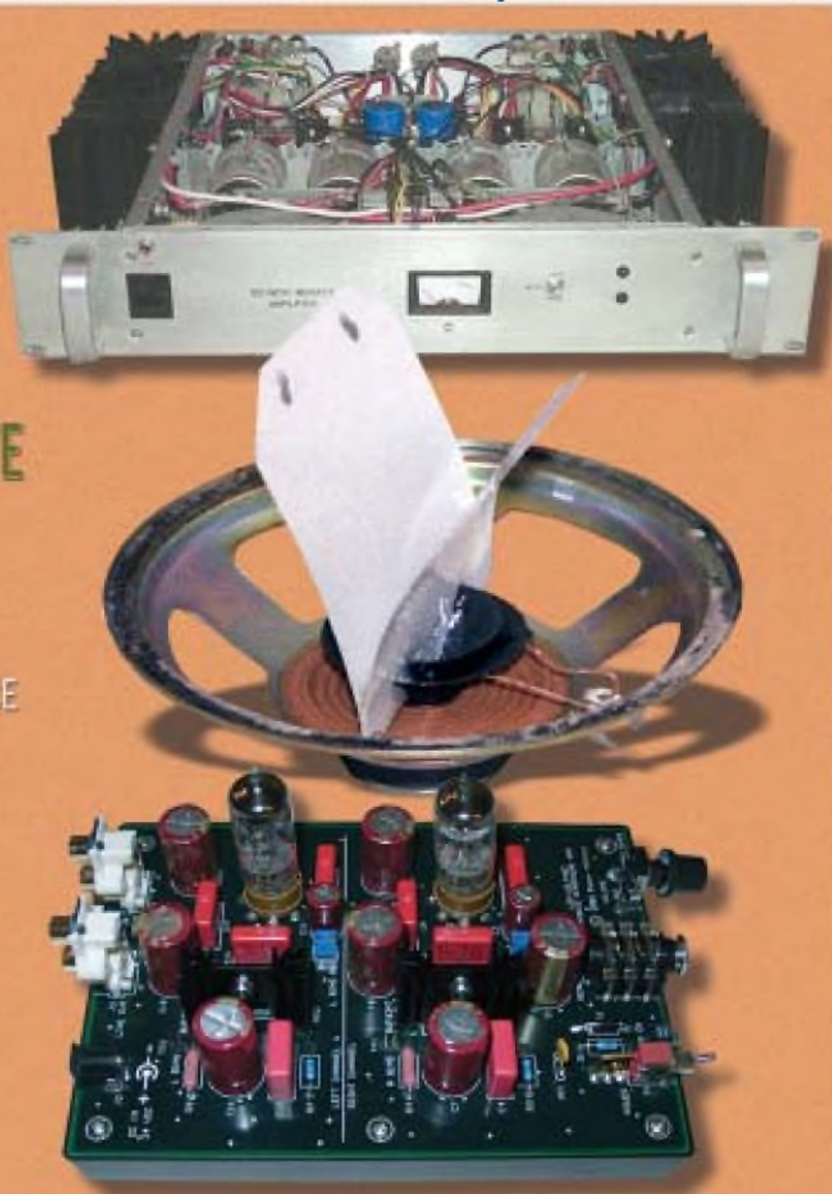

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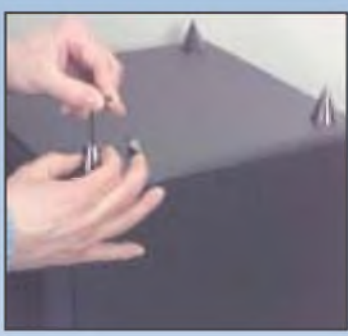
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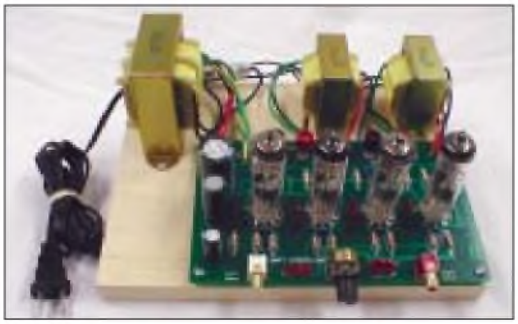

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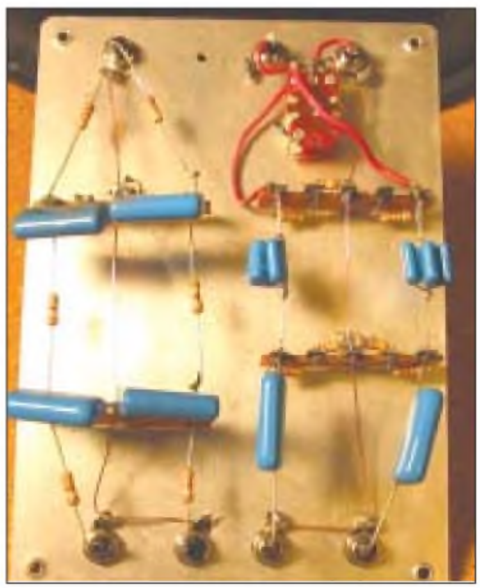

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.

JOHN STUART MILL

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Audio News

■ KTS ACOUSTIC INTERFACE PADS

KTS introduces four types of Acoustic Interface Pads, each type utilizing a special polymer, and each with a specific application. KTS Acoustic Interface Pads (AIPs) are designed to damp and absorb distortion vibration energy in speaker stand applications—along with general supporting surfaces—and are normally placed between the speaker cabinet/enclosure and the top plate of the speaker stand. They are adhesive, producing solid coupling between surfaces. Mass loading can be accomplished by placing the AIP atop the enclosure. Detailed data and specifications sheets are available for each AIP, and the KTS website www.kleintechsys.com offers detailed specifications along with graphic dimensional information and a special section for retailers. For more, contact Klein Tech Systems Corporation, 14th Lane, Building 1406, Lake Worth, FL 33463.

■ MOSFET POWER AMP

Marchand Electronics introduces the MB401 Monoblock MOSFET Audio Power Amplifier. It is rated at 300W RMS continuous operation (into 8Ω), and has both a balanced (XLR) and single-ended (RCA) input. The output uses heavy-duty five-way binding posts, while the power supply uses a heavy-duty 500VA toroidal power transformer and large capacitors, totaling 85,000μF. The MB401 construction is housed in a strong aluminum chassis. A large heatsink on the rear of the cabinet dissipates the heat the amp generates, and a thermal cutout mounted on the heatsink shuts off the amplifier in case it becomes too hot. The MB401 is available in two versions, class A and class AB; the normal class AB version provides the highest power, while the class-A version delivers less harmonic distortion with only 40W RMS output. Manufacturer's warranty is for three years, parts and labor. For more, visit www.marchandelec.com.

■ POWER FOR A/V SETUPS

Audio Line Source, LLP announces the addition of the Richard Gray's Power Company™ Models 1200S and 600S, which feature 12 20A Hubbel outlets to accommodate up to 12 components with no current limitation. The 1200S overcomes lapses in an AC line by working cycle-to-cycle to fill in the cyclic dips in the line when demand momentarily overcomes supply. It also incorporates a quality surge suppression system, which absorbs surges and protects the onboard MOV (metal oxide varistor). Like its predecessors (400S and 1200S), the 600S uses an "electric flywheel" effect to store and release energy on demand. The 600S is wired in parallel to the source of power and does not limit valuable current necessary to satisfy power transients required of the AC line. For more, visit www.castorcomm.com.

■ GUITAR AMPS

Maven Peal® Instruments, Inc. introduces a new pricing structure for the Zeeta amplifier designed to make Sag Circuit Technology affordable. The new pricing involves a two-level gold and silver platform that brings the Zeeta into the same price range as other boutique amplifiers. The Maven Peal Sag Circuit gives guitarists control over power amp distortion, and the wattage control helps guitarists prevent tinnitus and adjust the number of watts the amp produces. For more, contact Maven Peal at www.mavenpeal.com.

■ ELECTROSTATIC SPEAKER

MartinLogan announces a limited edition of their CLS loudspeaker system in honor of MartinLogan's 20th Anniversary. First manufactured in 1986, the CLS represents MartinLogan's second loudspeaker design to apply the patented curvilinear line source (CLS) electrostatic technology. A unique design enables faithful reproduction of the entire audio spectrum from a single curved electrostatic transducer (or stator). The current iteration of the CLS, the CLS IIz, remains nearly identical to the first unit, but major revisions for the IIz series CLS include refined crossover topologies, improved electrostatic panel construction, impedance stability, and advancements in power supply design. For more, contact MartinLogan, Ltd., 2101 Delaware, Lawrence, KS 66046, 785-749-0133, FAX 785-749-5320, www.martinlogan.com.

■ CONCRETE CABINETS

Faraday Sound introduces loudspeakers with a unique low-density concrete cabinet. The cabinets have rigidity and inertia that virtually eliminate unwanted panel vibrations. Because of the low density, the mass of each loudspeaker is less than 20kg, and they are of manageable dimensions for lifting. Faraday Sound will also offer an instruction booklet for do-it-yourself enthusiasts. Contact www.faradaysound.co.uk.

A 100W/Channel AB Differential Input Amplifier Part 1

This high-flying contributor presents a differential amp design that features some interesting, down-to-earth characteristics.

By Norman Thagard

The Pass/Thagard A75 amplifier[1] offered differential input capability for balanced sources. The only drawback to this option was the very low differential input impedance of just 950Ω. There are preamplifiers that would struggle with load impedance of 1kΩ or less.

I designed the 100W/channel amplifier presented here in 1992, and it was one of the circuits–the so-called A40M being the other–that I submitted to Nelson Pass when he was contemplating the A75 design. The DIFF 100 presents 20kΩ differential input impedance that should be acceptable for most preamps. Its schematic is drawn as *Fig. 1.*

I acknowledge at the outset that this amplifier contains some transistors that are no longer available. Some readers have complained about such situations, so more is said about that later in the article.

In past articles, I have taken a tutorial approach. I shall assume that you are familiar with the principles discussed in those articles and limit explanatory material in this article to those particulars that I left uncovered previously.

DIFFERENTIAL INPUT

The point of offering balanced input capability is to reduce hum and noise that are introduced into the system prior to the input. For effective reduction, the hum and noise voltages must be nearly equal at both the (+) and (–) inputs of the balanced input amplifier; i.e., they must be almost purely common-mode voltages. It does not matter whether the common-mode voltages are generated by electronics earlier in the chain or induced electrostatically or electromagnetically, but, of course, it is usually induced signals that are the target of balanced-mode operation.

This was my first and so far only attempt at a differential input amplifier. I have no differential sources, so this was an intellectual pursuit specifically initiated for the purpose of realizing a practical amplifier with generally acceptable parameters of input impedance, bandwidth, distortion, and power output.

Anyone who has read my previous articles knows that I like to stress the compromises that are inherently a part of engineering design. Raising input impedance in an amplifier such as the DIFF 100 invariably also raises high-frequency distortion because it increases impedance in the feedback network. Ultimately, it was this consideration that led to limiting input impedance at the 20kΩ value.

Refer to *Fig. 2.* In the A75 article Nelson argued that the input impedance at the inverting input was just 273Ω, which was, for that amplifier, less than the 475Ω value of R2. You should note that this value was derived based on an assumption of a purely differential input where the signals at the inverting and noninverting terminals were generated by two ideal voltage sources of equal magnitude but opposite sign.

Feedback is applied to the inverting or (–) input side of the input stage DIFF amp. This forces the voltage at the MOSFET gates on that side to be almost identical to the gate voltage at the noninverting or (+) input side, thus producing a "virtual short" between the gates. Because an observer "looking" into the negative input terminal cannot "see" the virtual short on the other side of the input resistor, the current that flows upon application of a given voltage at that terminal makes the input resistor appear to be much smaller than it really is. The virtual

PHOTO 1: Amp front.

ABOUT THE AUTHOR

Five-time astronaut Norman Thagard was the first American to enter space aboard a Russian rocket for a 90-day mission to the space station Mir. With a total of 140 days in space, he became the most experienced US astronaut ever. In addition to an MS degree in engineering science from Florida State University, he holds a doctorate in medicine from the University of Texas Southwestern Medical School. He is currently Professor and Director of College Relations at the FAMU-FSU College of Engineering. An avid audiophile, he designs and builds audio amplifiers as a hobby.

short thus produces an effect suggestive of the Miller effect that causes the apparent impedance on the inverting side to be reduced.

INPUT IMPEDANCE

The determination of an impedance looking into a terminal requires injection of a test current with measurement (or, during design, computation) of the resulting voltage or application of a given voltage with measurement or computation of the resulting current. To be valid, the resulting measured or calculated voltage or current must be due only to the test current or voltage that produced it. Any other independent sources that could alter this measurement must be appropriately zeroed. At least part of the current flowing in the (–) terminal will come through the feedback network due to the input at the noninverting terminal.

For this reason, some hold that the concept of input impedance at the inverting terminal of a difference amplifier does not apply unless the input at the (+) input is forced to zero.[2] In textbooks, the convention is, therefore, to invoke the principle of superposition, considering (+) and (–) inputs separately while holding the other input at ground. Considered in this manner, the A75 has an inverting input impedance of R2 = 475Ω because the virtual short becomes a "virtual ground." The differential input impedance by this convention is R2 + R4 = 950Ω.

A "floating" differential source must sink identically the same current into its (–) terminal that its (+) terminal sources. This case is illustrated by the simplified A75 input circuitry shown in *Fig. 2*, where the dashed line represents the virtual short. Although the two resistors labeled R ground-reference the voltage on the MOSFET gates, the source will nonetheless generate a current i = v/(R2 + R4), and it is clear, then, that differential input impedance is R2 + R4 = 950Ω for this case as well.

I suspect that most preamplifiers with balanced outputs would "see" the situation that Nelson describes. It is unlikely that the preamp outputs would be floating à la some kind of sensor or transformer coupling. Despite this, I shall take the liberty of convention to state that because R2 = R4 = 10kΩ in the DIFF 100, its differential input impedance is 20kΩ with an inverting terminal input impedance of 10kΩ when only that input is driven. Whatever the case, it is unarguably desirable to raise the input impedance to levels above that attained by the A75 if you can do this without unduly compromising other aspects of the design. Referring again to *Fig. 2*, which is also representative of the (+) input of the DIFF 100, noninverting input impedance is R4 + R ≅ 210kΩ.

A QUESTION OF BALANCE

Another factor that is different with a balanced input is the dependence of the closed-loop gain upon the source impedance seen by the inverting terminal. In textbooks[3], resistor R at the inverting gate is omitted. A series resistor similar to R2 sets the closed-loop gain in conjunction with the source impedance and a resistor R_{fb} from the output to the node labeled "feedback" as shown in *Fig. 2*.

Emphasis is placed on the special case where $R2/R_{fb} = R4/R$. This results in so-called "balanced bridge" operation, where differential gain becomes $R_{fb}/R2$ for zero source impedance but falls with increasing source impedance. If such an amplifier is stimulated only at the inverting terminal, the gain through mid-frequencies becomes $-R_{fb}/(R2 + R_S)$, where R_S is the real part of the generator (preamp) impedance.

Ideally, inverting input impedance would be high enough to swamp out any effect of source impedance and minimize the requirement levied on the preamp to deliver appreciable output current. The previous paragraph shows that as R2 is increased in order to raise input impedance at the inverting input, R_{fb} must also be increased by the same multiplication factor if gain is to remain the same. The output voltage will charge and discharge the gate capacitance of the MOSFETs at the inverting input by forcing a charging current that must pass through R_{fb}. Obviously, for a given output voltage, R_{fb} limits this charging current, and larger values of R_{fb} reduce the current proportionally more.

This is a potential slew-rate problem where feedback will be unable to correct output errors if output amplitude, frequency, or a combination thereof requires a feedback signal whose rate of change cannot be met. It is the feedback signal rather than the forward signal that is being primarily affected, but the effect at high frequency on amplifier performance will be measurable as a higher level of distortion. The high-frequency distortion can quickly reach unacceptable levels and almost always does if an impedance level typical of most stereo amplifiers—i.e., on the order of 50kΩ to 100kΩ—is attempted at the inverting input.

The 10kΩ inverting input impedance achieved by the DIFF 100 is quite good, but is still no better than the lowest impedance that I have ever encountered at the input of an ordinary stereo amplifier (Crown DC300). I wanted to set 10kΩ as the minimum input impedance, and it turned out to be the highest achievable while maintaining acceptable high-frequency distortion levels. Time after time, design decisions are set by compromises. It is unusual, however, to have no leeway at all.

To head off some potential letters, I acknowledge that a large part of the problem is the high gate capacitance of the MOSFETs used in the A75 and the DIFF 100. At the time the DIFF 100 and A75 amplifiers were designed, there was considerable general interest in all or mostly MOSFET designs. Thus, the MOSFET input stage was a design criterion. Today, I would probably use high-transconductance JFETs in the input stage DIFF amp and either keep (–) input resistance at 10kΩ to achieve lower high-frequency distortion or increase input resistance up to 50kΩ or so if high-frequency distortion remained at or less than its current level.

The A75 and DIFF 100 have switches that permit you to change between unbalanced and balanced input mode. *Figure 3* shows the placement of this switch for the DIFF 100. In the real world, people do make mistakes—especially if curiosity gets the better of them.

With a switch labeled "balanced/unbalanced" present on an amplifier back panel, the tendency is to flip it to the other position, even if you have the usual unbalanced signal source connected to the noninverting (+) input and nothing connected to the inverting (–) input. The figure makes it clear that the switch throw would then result in 100% feedback because there would be no current path through R2 or the MOSFET(s) gates and hence no voltage drop across R_{fb}. All of the output voltage would consequently be applied to the

TABLE 1
AMPLIFIER PARTS LIST

(ALL RESISTORS ARE 1%, ¼W METAL FILM UNLESS OTHERWISE SPECIFIED)

SCHEMATIC REFERENCE	PART #/ VALUE	DESCRIPTION	MANUFACTURER
Ctrim	1.8-6n	trimcap	
C1	3p	silver mica	
C2	100n	film	
C3	100n, 250V	film	
C4, C5	33p, 100V	silver mica	
P1	100k	trimpot	
P2	1k	trimpot	
P3	5k	trimpot	
R1, R2, R20, R21	10k		
R3	200k		
R4	150k		
R5-R8, R27-R30	221		
R9	274		
R10, R12	51.1		
R11	121		
R13	100		
R14, R16	237		
R15,R17	20k		
R18	7.5k		
R19	1k		
R22	68.1, 1W		
R23-R26	0.5, 1W (non- Inductive, may parallel 1Ω metal film resistors)		
R31	10, 2W metal film		
	jack-to-chassis resistor	10, 2W (see text; consider two 5.1Ω resistors in series)	
M1, M2	IRFD 110	n-ch MOSFET	IRF
M3, M4	IRFD 9120	p-ch MOSFET	IRF
M5	IRFD 9210	p-ch MOSFET	IRF
M6	IRFD 210	n-ch MOSFET	IRF
M7, M8	2SK134	n-ch Power MOSFET	Hitachi
M9, M10	2SJ49	p-ch Power MOSFET	Hitachi
Q1, Q2, Q8-Q10	MPSA56	pnp BJT	Motorola
Q3–Q7	MPSA06	npn BJT	Motorola
Q11	TIP29	npn power BJT	TI
Q12	SK 9041	npn power BJT	RCA
Q13	SK 9042	pnp power BJT	RCA
VR1, VR2	LM385Z-1.2	1.2V reference	National
D1, D2	1N4148	diode	
Z1, Z2	1N5231B	5.1V, 0.5W zener	
Z3-Z6	1N4739A	9.1V, 1W zener	
FB (2)		Ferrite bead	

Chassis-mount RCA phono jack, XLR jack, SPST or DPST toggle switch (see text), dual banana jack, TO126 and TO3 heatsinks, standoffs inverting gates producing a unity-gain amplifier, that is, a voltage follower.

The best that could happen is that the volume would necessarily need to be turned up significantly to maintain output level. At worst, the increased feedback would convert the amplifier to an oscillator, perhaps frying the high-frequency drivers in the speakers. The presence of resistor R at the inverting gate (*Fig. 2*) would provide a current path, forming the usual voltage divider feedback network. Gain for this off-nominal configuration would then be 1 + R_{fb}/R rather than unity. This gain is easier to compensate than unity gain and would provide reasonable behavior in the face of an unreasonable configuration of the switch.

In testing on my bench, the DIFF 100 is stable despite the unity gain in this situation. You could use a resistor of value 10kΩ for resistor R at both (+) and (–) inputs, just to be on the safe side, but I did not do so based on the test bench results.

If the source impedance driving the inverting input is taken as zero–not an unreasonable approximation in most cases–then differential gain for the A75 comes to about 35 and for the DIFF 100 to about 20. Gains for the incorrectly configured switch discussed previously–that is, for unbalanced operation in the balanced-mode switch position–drop to about 9 and 1, respectively. Both amplifiers are adequately compensated for these values of closed-loop gain so that the only result will be a drop in volume. It is nonetheless recommended to operate the amplifiers correctly, and notwithstanding test bench results, I make no claim of absolute stability of the DIFF 100 if operated incorrectly.

BLOCKING CAPACITOR

While on the subject of potential "bugaboos," I should mention the absence of DC-blocking capacitors at the input. This design implements a DC amplifier. If there is any possibility that an audio source driving the DIFF 100 has appreciable DC offset, then the addition of a capacitor in series with both inverting and noninverting inputs is required. The amplifier can tolerate some degree of DC offset, especially if it is in the common mode. If only unbalanced inputs are ever to be used, then only the noninverting input needs a blocking capacitor.

The capacitor needs to be rather large, say 10µF, if the low-frequency cutoff is to be at least a decade below 20Hz. Most high-end designers prescribe polypropylene or similar high-quality film capacitors, which are physically large. Rather than place them on the front-end PC board, it might be more practical to place them in series with the leads from the input connectors.

You saw earlier that balanced bridge operation was the condition for producing an amplifier whose output is the product of a resistor ratio and the difference between the voltages at the (+) and (–) inputs. This is also the configuration necessary for maximum rejection of common-mode signals when coupled with an ideal amplifier.[4] In the A75, Nelson was careful to maintain balanced bridge configuration insofar as it can be maintained given fixed 1% tolerance resistors. That is the reason that resistors attached to the gates of the A75's DIFF amp MOSFETs were sometimes paralleled even though it might have appeared unnecessary.

I chose to offer, instead, a means of adjusting for *unbalanced* bridge operation. This adjustment can yield higher rejection of common-mode signals because it allows partial compensation for imperfections in the amplifier circuitry that, of themselves, can reduce common-mode rejection.[5] The trade-off is the requirement for yet another initialization adjustment. You are free to replace the trimmer capacitor at the (+) input with a fixed 3pF capacitor. You can also replace the series 150kΩ fixed resistor and 100kΩ trimpot with a single 200kΩ fixed resistor.

At low frequencies, only the real part of the impedance must be adjust-

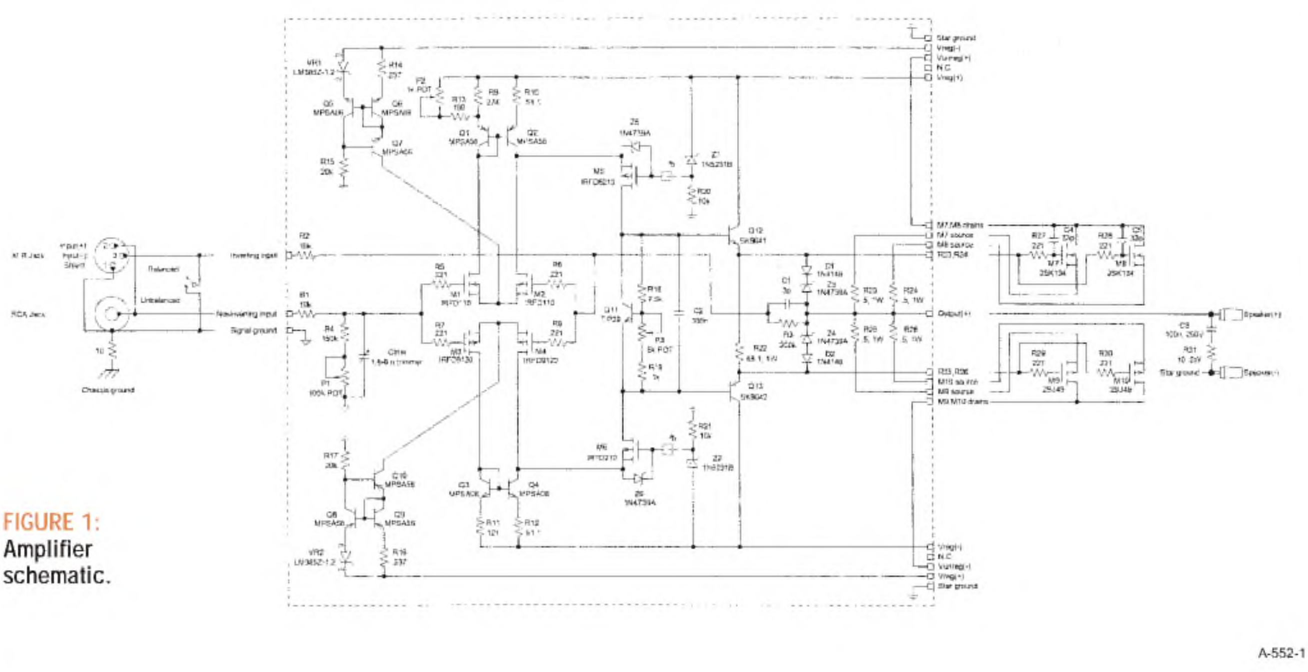
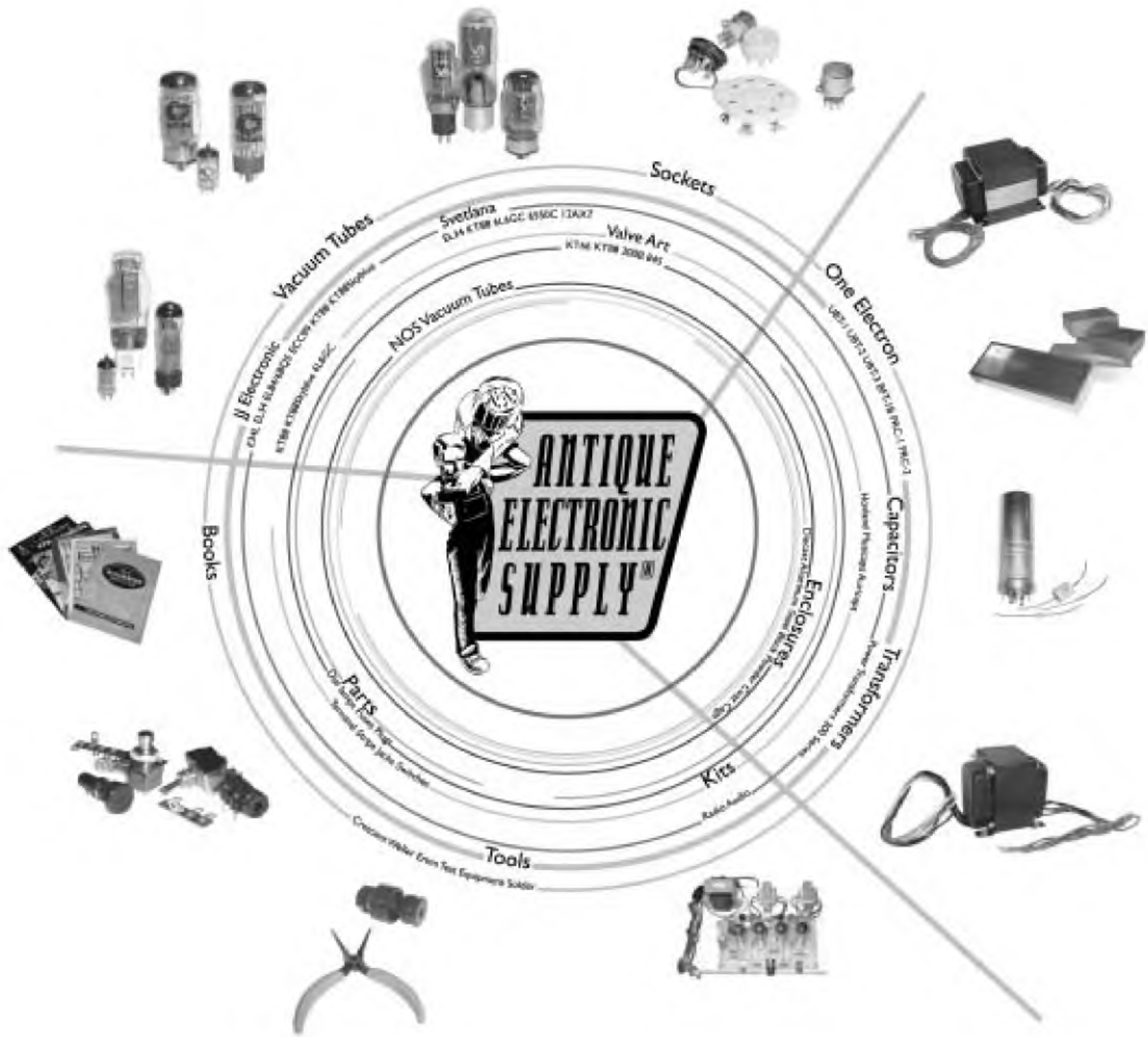


FIGURE 1: Amplifier schematic.



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ed in order to optimize common-mode rejection. For these frequencies, resistance matching alone is sufficient. At high frequencies, including the upper part of the audio band, the reactive component of the impedance at each gate becomes significant and also must be adjusted if good rejection is to be obtained. That is the purpose of the trimmer capacitor.

CIRCUIT DESCRIPTION

The MOSFET-based input stage is compound. The MOSFETs on the noninverting output side of the upper and lower DIFF amps form a folded cascode with complementary MOSFETs in common gate. Bias current for the upper and lower DIFF amps is set by separate current sources, as it is impossible to insert a common current source between the two halves.

These current sources look, at first glance, like modified Wilson current mirrors but are not. There is, however, a voltage that is mirrored from one leg to the other, specifically the 1.2V across a precision voltage reference. This mirrored voltage is applied across a 237Ω resistor in the other leg, which yields a current of about $1.2V \div 237\Omega \cong 5mA$. Assuming balance, each MOSFET in the DIFF amps will therefore have a drain current of 2.5mA.

A "reverse" Widlar current mirror sets bias current for the cascode (common gate) device. Again assuming a balanced diff amp and ignoring base currents in the mirror, drain currents in the common-gate devices are $0.0025A \times 120\Omega \div 51.1\Omega - 2.5mA \cong 3.4mA$. The 2.5mA is subtracted because that part of the current through the 51.1Ω resistor of the current mirror is supplied by the DIFF amp MOSFET. The idea was to make the current in both devices of the folded-cascode approximately the same. A trimpot associated with the upper mirror allows output offset voltage to be nulled.

Transistor Q11, along with R18, R19, P3, and C2, constitutes the usual V_{BE} multiplier. Adjustment of P3 varies, ultimately, the gate-to-source bias voltage for the output MOSFETs and, therefore, the output stage bias current. Adjustment range is from $[V_{BE\text{-}Q11}/(R19 + P3_{max})](R18 + R19 + P3_{max}) \cong 1.5V$ to $(V_{BE\text{-}Q11}/R19)(R18 + R19) \cong 6V$.

Due to the base-emitter voltage drops in Q12 and Q13, this adjustment range results in an approximate adjustment range of 0 to 4.5V for the gate-to-source bias voltage of the output stage MOSFETs. Since the data sheets for the 2SJ49/2SK134 power MOSFETs specify a gate-source threshold voltage magnitude in the 0–1.5V range, this adjustment range should be adequate for setting output stage quiescent current from zero to near-class A levels.

This is the only amplifier in which I interposed an emitter follower—here consisting of Q12 and Q13—between the voltage amplifier and the output stage. I did so to ensure adequate current drive capability for the gate capacitance of the output transistors; otherwise, slew rate might have been a problem. It is interesting that I subsequently learned that this topology has an additional advantage in lowering one form of distortion.[6]

The diode/zener network D1, D2, Z3, Z4 protects the output transistors against potentially damaging voltages. In the case of the Hitachi devices used, this voltage can be no more than ±14V.

SUBSTITUTIONS

The output stage is a push-pull source follower topology. Shown as C4 and C5 on the schematic are 33pF capacitors between gate and drain of the 2SK134 n-channel output transistors. These small capacitors were incorporated per a recommendation by Mr. Erno Borbely, who used these same Hitachi power transistors in some of his designs.[7]

I am a firm believer in publishing a design, warts and all, exactly as the prototype is constructed. Those capacitors are in the working amp, so they appear here. PSpice actually did show a slight improvement in the 20kHz distortion figure with these capacitors in place, so perhaps there is a valid argument for them.

There are also 1000µF capacitors placed from each drain of the four output MOSFETs to ground. While these, too, were placed due to a recommendation, I distinctly recall that this amp worked OK without them. Their inclusion or omission, then, is an option for the builder.

The 2SK134/2SJ49 complementary MOSFETs differ in significant ways from the (to me) more familiar IRF series of power MOSFETs.

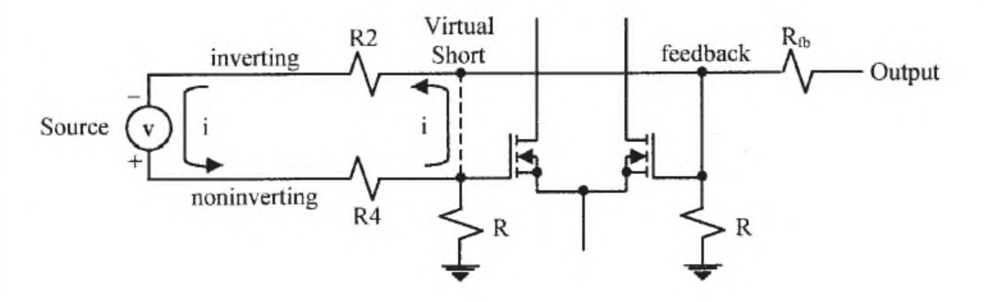

FIGURE 2: Simplified A75 input.

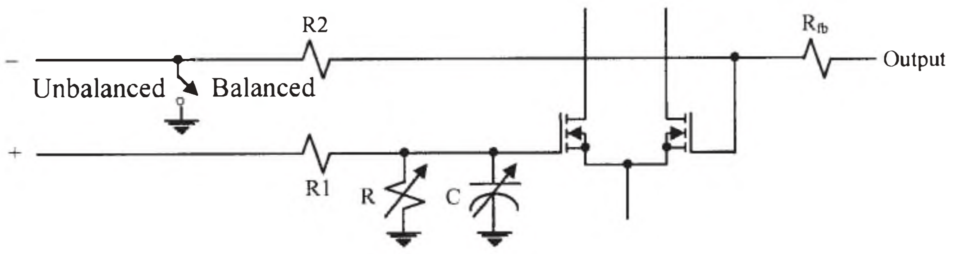

FIGURE 3: Simplified DIFF 100 input.

PHOTO 2: Amp rear.

The adjustment range of the V_{BE} multiplier evinces one difference. The Hitachi transistors have a much smaller gate-to-source threshold voltage than do the IRF devices.

Another difference is that the source of a Hitachi MOSFET is electrically and thermally connected to the case rather than the drain. In my prototype, the voltage across R22 was approximately 1.5V for each channel with the output quiescent current levels used.

I came to incorporate these devices because I had purchased a kit of parts for a stereo amplifier that included them. I was not pleased with the performance of that amplifier, especially its tendency to oscillate and to ring with square-wave inputs. Although these two phenomena might seem related, I cured the latter problem simply by paralleling the inductor in series with the output with a 1Ω resistor. The former problem was more intractable, and I finally turned to an entirely new design based around the Hitachi power MOSFETs. While I was at it, I somewhat arbitrarily decided to include the balanced input capability.

As I mentioned at the outset, the 2SK134/2SJ49 transistors are difficult to find, as they are no longer manufactured. In the PSpice version of the circuit I actually substituted IRF240 and IRF9240 devices because I did not have the Hitachi models. I modified the IRF240/IRF9240 Instance Models for threshold voltage and input capacitance to be a little closer to the Hitachi characteristics.

Use of stock IRF MOSFET models, however, worked fine as long as the V_{BE} multiplier was also modified for proper DC biasing of the output stage. For this, R18 was increased to 22.1kΩ. Even if you must substitute for the 2SK134/2SJ49 transistors, the circuit should still work fine, although a change in compensation may be required. If so, this may change the required range for Ctrim.

There is no series inductor between output and load in the DIFF 100. I hit the amplifier with fast rise-time square waves while I connected capacitors up to 1μF in parallel with an 8Ω dummy load directly at the output binding posts. The capacitor did cause an overshoot with a couple of cycles of rapidly damped ringing, but I think that this is OK, especially since the situation improved considerably if just a couple of feet of zip cord were interposed between binding posts and load. There is about 0.3μH/ft inductance associated with 18-gauge wire[8], so it takes very little inductance to mitigate the effect of capacitive loading at the output.

With the IRF output transistors, PSpice indicated that a 10μH series inductor was required to suppress oscillation. This requires only a few turns of wire for an air-core inductor. It may be necessary to wind these around and in parallel with a 1Ω, 5W power resistor in order to lower the Q of the inductor. Otherwise, the series inductor itself can lead to overshoot and ringing. I incorporated no inductor in the output of the prototype and have encountered no problems.

Full-power output THD decreased up to a point as output stage bias current was increased from zero. Since I used the original heatsinks supplied

PHOTO 3: Amp top.

with the kit for the Hitachi transistors, and these are not adequate for class A operation at the 100W/channel output level, this amplifier was intended to be operated class AB. Thus, based on the observed distortion behavior, the bias current for the output MOSFETs was increased until I observed little improvement in THD. This was achieved with a drain current of about 130mA, a value that squares with Erno Borbely's recommended value for these devices of 100mA.[9]

Be sure that your heatsinks are sufficient to dissipate 10W or so *per* mounted transistor. If in doubt, use the touch test, that is, after the output bias is set, be sure that the transistor cases do not become too hot to touch. Verify that the transistor cases are at near zero AC and DC potential with no input signal present before trying this. This should be the nominal situation since the Hitachi power devices have the source connected to case in a push-pull, source follower output stage topology.

Also, I learned the hard way as a ham radio operator that rf burns rather than shocks. (This was one thing taught in medical school that I had no trouble at all remembering.) Thus, for real caution, be sure your amplifier is not oscillating before touching the output transistor cases. The ferrite beads used in the amplifier are there to suppress a low-amplitude radio-frequency oscillation seen in the breadboard stage.

Part 2 concludes our look at this 100w/channel amp, with information about the power supply, construction, and more. ❖

REFERENCES

1. Norman Thagard and Nelson Pass, "Build the A75 Power Amplifier," *TAA*, 4/92, p. 20.
2. Allan R. Hambley, *Electronics*, Second Edition, Prentice Hall, 2000, p. 111.
3. See, for example, Sergio Franco, *Design with Operational Amplifiers & Analog Integrated Circuits*, Second Edition, McGraw-Hill, 1998, pp. 19, 20.
4. *Ibid.*, p. 73.
5. *Ibid.*, p. 75.
6. Douglas Self, "Audio Power-Amp Distortion, Part 2," *TAA*, 3/99, pp. 30–31.
7. Erno Borbely, "Letters," *TAA*, 2/85, p. 50.
8. *The 1997 ARRL Handbook for Radio Amateurs*, ARRL, Inc., 1996, p. 10.10.
9. Erno Borbely, "A 60W MOSFET Power Amplifier," *TAA*, 2/82, p. 12.

A Passive, Low Level Crossover

Here's an interesting topology for a passive crossover that produces pleasing results. By Cornelius Morton

I just completed a subwoofer project that brought out the need for a crossover that would:

a. Cross over at 40Hz
b. Sum the left and right bass channels for a true center subwoofer channel
c. Allow the high and low channels to be balanced for speaker efficiency
d. Work with my Carver C-1 preamp and Hafler 220 and Citation V amps
e. Simulate the Linkwitz-Riley alignment for 12dB per octave crossovers
f. Be easy to modify and inexpensive (in case the project went south)

The passive crossover presented here fulfills all these requirements. You should note that this is a passive device and exhibits insertion loss and a relatively low input impedance. A preamplifier or line amplifier with sufficient gain capable of driving a 3000Ω load is required.

CIRCUIT DESIGN

Referring to the schematics of *Figs. 1* and *2*, the basic low-pass and high-pass sections are inverse—the high-pass replaces the low-pass resistors with capacitors and the low-pass capacitors with resistors. Because the filters have two sets of RC units in series, R1, C1 and R2, C2, they are two pole filters and will exhibit a 12dB per octave attenuation in the stopband. Most of the documentation for this type of filter shows R2 = 10R1 and C2 = C1/10. While this is a good rule of thumb, it is not cast in concrete. What is necessary is that the two poles match in frequency, where f = 0.159/RC (0.159 = 1/2pi to three places).

In the low-pass filter R1 determines the minimum load presented to the preamp, and the parallel combination of R1 and R2 does the same for the high-pass filter. I considered a 3000Ω load the design minimum and 1.0μF caps were handy, thus R1 = $.159/(40 \times 1.0 \times 10^{-6})$ = 3975Ω. A value of 3900Ω is a readily obtainable part.

For the low-pass, I made the second section a repeat of the first. This results in an input impedance for the low-pass of 3900Ω, which is well above the 3000Ω minimum target. The two low-pass sections are tied together at the output through 2200Ω resistors to provide a summing function across the input impedance of the amplifier, that is 47kΩ on my Hafler. You could delete the 2200Ω resistors, but phase response becomes lumpy when one input is opened.

The high-pass exhibits a slightly different problem, as the input impedance is the parallel combination of R1 and R2. Setting C2 to 0.30μF (three each 0.10μF caps paralleled) results in a value for R2 of 13250Ω. I used 13200 for computational purposes. This results in an input impedance of 3010Ω. Unfortunately, C1 sees the 3010Ω as the R of that filter section with a resulting turnover frequency of 52.8Hz.

As determined previously, the calculated value for R1 was 3975Ω, which you can use to determine the value of R1 required so that the parallel combination of R1 and R2 equals 3975Ω. R1 = $(1/3975\text{-}1/13200)^{-1}$ = 5690Ω. I used a combination of 5100Ω resistor in series

PHOTO 1: Completed crossover unit.

ABOUT THE AUTHOR

Cornelius Morton retired from Motorola Inc. three years ago with 46 years' experience in the military electronics field. This included design, maintenance, test and troubleshooting of X and Ka band radar, C band transmitters, and various airborne communication systems. He has had an active interest in audio since 1958.

with a 620Ω resistor for a value of 5720Ω. The same problem also occurs when the conventional R2 = 10 × R1 values are used, and results in a 9% error in the effective value of R1, which is generally ignored.

The resulting input impedance of the high-pass is now 3975Ω and matches the low-pass. R2 is also in parallel with the level control and the input impedance of the power amp. If your amplifier has gain controls on the input, ignore the level pots. In any case, check the note for *Figs. 1* and *2* to determine adjustments to R2 and R4.

I put the level controls on the high-pass filters so that the subwoofer level could match the left and right speakers. This is required for two reasons: the low-pass section has about 2dB more insertion loss than the high-pass, and my bass unit is not as efficient as my other speakers. While the level controls increase the output impedance of the high-pass, cable lengths of 6–8′ do not affect the high-frequency response.

While computing values for R and C for other crossover frequencies is rather straightforward, finding values that are readily available may become tedious. *Table 1* provides values for half octave steps to 135Hz.

While the schematics and photos show separate inputs for the high- and low-pass sections, the inputs may be combined to provide single left and single right channel inputs. The combined

PHOTO 2: Component layout. The circuitry on the left is the low-pass. The high-pass circuitry is on the right. I used a bus wire ground for each section, even though the RCA jacks are mounted to the cover.

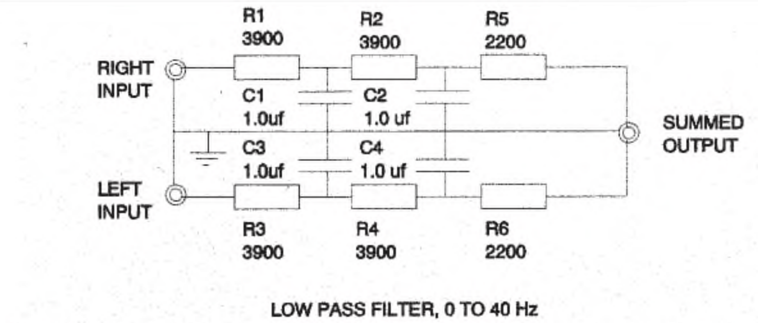

FIGURE 1: Low-pass filter section. See *Photo 2* for construction details.

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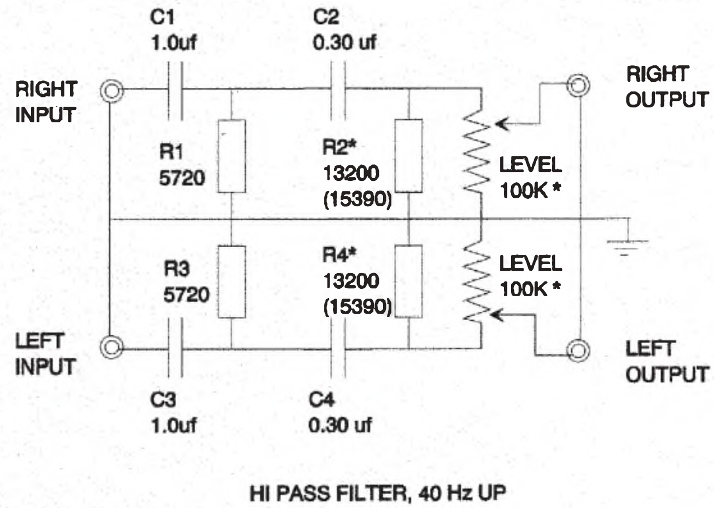

* IF THE LEVEL CONTROLS ARE USED R2 AND R4 SHOULD BE ADJUSTED AS FOLLOWS; RIN = THE INPUT RESISTANCE OF THE AMPLIFIER.

1/R2=1/113200 - (1/100000 + 1/RIN). FOR EXAMPLE;

RIN =1000000 OHMS, R2 AND R4 = 15440 OHMS (15k AND 390 ARE COMMON VALUES)
RIN = 47000 OHMS, R2 AND R4 = 22380 OHMS (22K AND 390 ARE COMMON VALUES)

FIGURE 2: High-pass filter section. See *Photo 2* for construction details.

B-2086-2

TABLE 1
HIGH-PASS SECTION

FREQUENCY	R1	C1	R2	C2	NOTES
60Hz	5720	0.68µF	13200*	0.2µF	2, 6
90Hz	5720	0.45µF	13200*	0.134µF	1, 4, 6
135Hz	5720	0.30µF	13200*	0.089µF	3, 5, 6

LOW-PASS SECTION

FREQUENCY	R1	C1	R2	C2	NOTES
60Hz	3900	0.68µF	3900	0.68µF	
90Hz	3900	0.45µF	3900	0.45µF	1
135Hz	3900	0.30µF	3900	0.30µF	3

Notes
1. The 0.45µF cap is two 0.22µF caps in parallel. Add a 0.01µF for real accuracy.
2. The 0.2µF cap is two 0.10µF caps in parallel.
3. The 0.30µF cap is three 0.1µF caps in parallel.
4. The 0.134µF cap is a 0.10µF cap in parallel with a 0.033µF cap.
5. The 0.089µF cap is a 0.068µF cap in parallel with a 0.022µF cap.
6. The 5720Ω resistor is a 5100Ω resistor in series with a 620Ω resistor. A 5600Ω resistor in series with a 120Ω resistor also works fine.
*See text and schematics for actual value.

input impedance will not go below 1900Ω. My C-1 has two sets of line outputs, so I chose to use both. Any preamp that can drive a 1000Ω load will have no problem driving the combined single channel inputs. If you are using one of Don Morrison's E.L.A.D.s, those will drive the world. *Figures 3* and *4* show simulations of low-pass and high-pass outputs.

CONSTRUCTION

I built the crossover in a Radio Shack 6″ × 4″ × 2″ box, part number 270-1806, and used the aluminum lid to attach connectors, level control, and four terminal strips. I pop-riveted the strips to the lid, but screws work well, too. Construction details and the completed unit are shown in *Photos 1* and *2*.

The level control is a Radio Shack 271-1732, 100k stereo control. Capacitors are metalized polyester or metalized polypropylene types, 100WV DC minimum. You can find these at Parts Express, Antique Electronic Supply, and Radio Shack. Resistors are ½W or ¼W, metal film, carbon film, or composite. Maximum dissipation with a 10V RMS input is 26mW for the 3900Ω resistors.

The speaker used as a stand in the photos is the subwoofer project that caused this effort. It uses four shorted quarter wave tunnels to control driver resonance and is based on original work by the late Stewart Hegeman and Don Morrison of Toronto Canada. It does produce enough energy at 16Hz to hurt my ears and rattle windows throughout the house.

SETUP

Hook it up between your preamp and power amps. A mono amp or one channel of a dual channel amp is required to drive the subwoofer. You may bridge that dual channel for an extra 3dB. Set the level control to minimum attenuation, load in a CD with a good spread of music and with a bit of low bass. Adjust the level for a balance between the left and right channels and the low bass. That should be 3–5dB of attenuation. If you have a sound level meter and an audio generator, you may achieve a precise setting. Sit back and enjoy. ❖

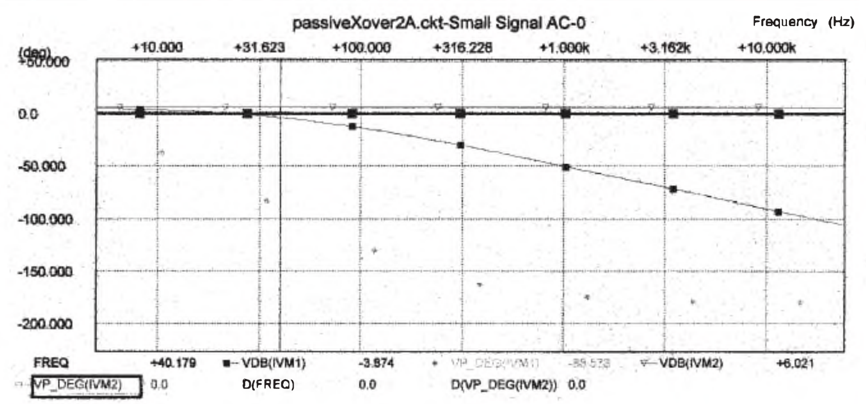

FIGURE 3: SPICE simulation of low-pass output. The input level, VM2, is +6.021dB. The output level, VM1, is –3.874dB at 40.179Hz and +2.15 at 16Hz. The actual circuit measured within ±0.2dB.

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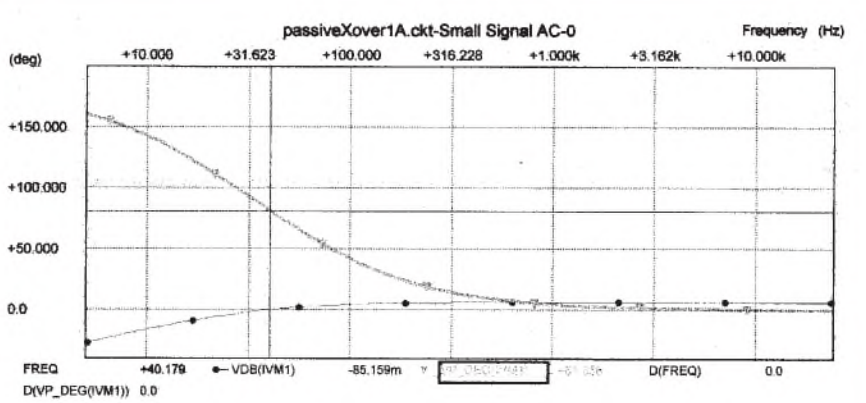


FIGURE 4: SPICE simulation of high-pass output. The input level, VM2, is +6.021dB. The output level, VM1, is –0.085dB at 40.179Hz and +6.002 at 1000Hz. The actual circuit measured within ±0.2dB.

B-2086-4

NOTE FOR DIYERS

0.159/frequency = RC, which makes it easy to find "quickie" values for resistors and caps in filters. For example, for 66Hz, 159/66 = 0.002409. If you have a $12k\Omega$ resistor, then 0.002409/12000 = 2.007 e-7 on my calculator, or 0.2µF. Works nice.

SOURCES

Parts Express
725 Pleasant Valley Dr.
Springboro, OH 45066-1158
937-743-3000
FAX 937-743-1677
sales@partsexpress.com
www.partsexpress.com

Antique Electronic Supply
6221 S Maple Ave.
Tempe, AZ 85283
480-820-5411
FAX 480-820-4643
info@tubesandmore.com
www.tubesandmore.com

Radio Shack
1-800-843-7422

Build a Low-Voltage Tube Hybrid Headphone/Line Amp

Here's a simple, safe, inexpensive project for beginners and those who wish to "try tubes." **By Pete Millett**

This project came about as a convergence of three things: First, since I've been in the headphone amp business, I'd been thinking about writing an article about a do-it-yourself headphone amp. Second, I had been hearing from quite a few people who wanted to try to build an audio project using tubes, but were put off by the high voltages involved. And third, I ran across some low-voltage tubes that were designed to run off the battery voltage in car radios.

My goal in designing this amp (*Photo 1*) was to come up with an easy-to-build, affordable project that's safe and fun for an inexperienced builder to experiment with. I wouldn't call it a high-end audio design, but it does sound pretty good. You can use it as a headphone amplifier or as a line amplifier to drive power amps.

This amplifier uses a single-ended tube voltage amplifier stage and a solid-state follower to get a low enough output impedance to drive headphones. It uses no negative feedback. I think this type of circuit is a good introduction to the sound of tube audio equipment.

LOW-VOLTAGE TUBES?

Most tube audio circuits–even low-level preamps–operate on power supply voltages of between 100 and 500V. With proper precautions while building and working on your equipment, these voltages really shouldn't be a safety hazard–but nevertheless, they can deter the inexperienced from attempting to build tube equipment.

Most audio experimenters probably don't know that there was an entire line of tubes designed to be operated from a low-voltage power supply. These tubes, sometimes called "space charge" tubes, were designed during the transition from tube to solid-state electronics, mostly for use in 12V DC automobile radios.

Automobile radios using high-voltage tubes were expensive to manufacture, because the low-voltage DC power (either 6V or 12V) available in the car had to be stepped up to a high voltage to operate the tubes. With AC power, this is just a matter of a transformer and rectifier; with a DC input, the battery voltage first had to be turned into a square-wave alternating current by the use of a "vibrator," an electromechanical device that operates a bit like a buzzer or relay.

When transistors were first commercially available, radio-frequency transistors were expensive and difficult to manufacture, so hybrid tube-transistor car radios were developed. Most often, these radios employed tubes in the RF and low-power audio stages, and a germanium power transistor to act as the final audio stage, to drive the low-impedance loudspeaker. This hybrid low-voltage tube plus transistor approach was used only for a short time before fully transistorized radios became cost-effective, making the hybrid low-voltage tube radios obsolete.

Because of their target application, many of the low-voltage tubes are RF tetrodes and pentodes. Fortunately for us tube audio fanatics, there is also a whole line of tubes that contain a small-signal audio triode plus two diodes. These tubes were used as the detector, AVC, and first audio stage in a typical AM radio. This is the type of tube I used in this design. There are several interchangeable types to choose from, and they are inexpensive and readily available. Two such tubes, a 12AE6A (left) and 12FM6, are shown in *Photo 2.*

The headphone/line amplifier presented here takes a similar approach to those old car radios: a low-voltage tube is used to amplify the audio signal, and a solid-state output stage is used to provide a low-impedance drive for headphones or a power amplifier.

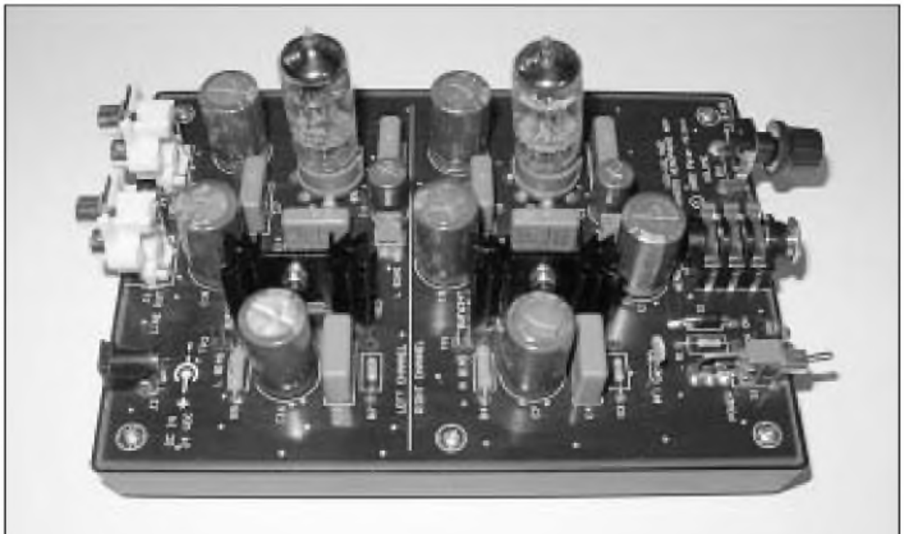


PHOTO 1: The low-voltage hybrid headphone amp.

THE CIRCUIT DESIGN

Refer to the schematic diagram (*Fig. 1*), as I walk you through the circuit and describe how it works.

Input Stage

The audio input from J4 is fed to a 50kΩ volume control potentiometer, RV1. The output of the volume control is connected directly to the tube's grid, so the DC voltage on the grid is 0V.

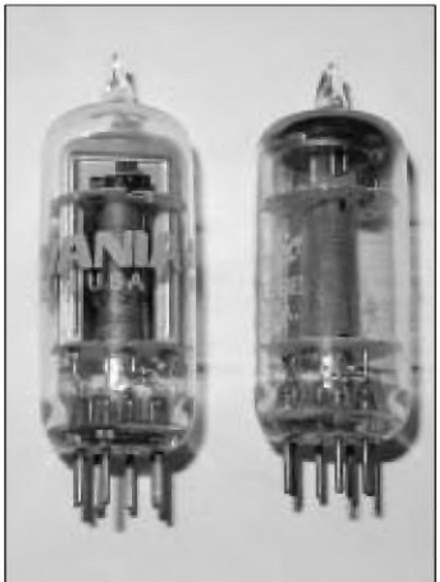


PHOTO 2: Low voltage triode/dual diode tubes 12AE6A (left) and 12FM6.

The triode of a space-charge triode/dual diode tube is used in a normal grounded-cathode voltage amplifier circuit. There are several such tubes that you can use; I tried the 12AE6 (or 12AE6A) and the 12FM6. Other tubes that may work, and all with the same pinout, include the 12AJ6, 12EL6, 12FK6, and 12FT6. Since the diode sections are unused, they are simply tied to ground.

Bias for the tube stage is developed across an adjustable resistor (R2, R6), which is paralleled by both an electrolytic capacitor and a film capacitor. DC current flowing through the tube raises the cathode voltage above the grid, which provides negative bias for the tube. The capacitors provide a low-impedance path for the audio signal. Note that the exact value of these capacitors is not at all critical.

The plate of the tube is loaded with a 0.56mA constant-current diode (D3, D4). You can think of this part as a resistor, which varies its resistance to try to keep a constant current flowing through it. The effect of this is to present a very high AC impedance load to the plate of the tube, which allows the tube to operate at high gain and low distortion. It also allows the plate to swing very close to the power-supply voltage.

There's nothing sacred about using 0.56mA as the plate current—looking at the plate curves, I thought it looked like a good point to operate the 12AE6 tube. I also tried a 1mA part, and got slightly higher distortion. You may want to try different currents, especially if you use tubes other than the ones I tried.

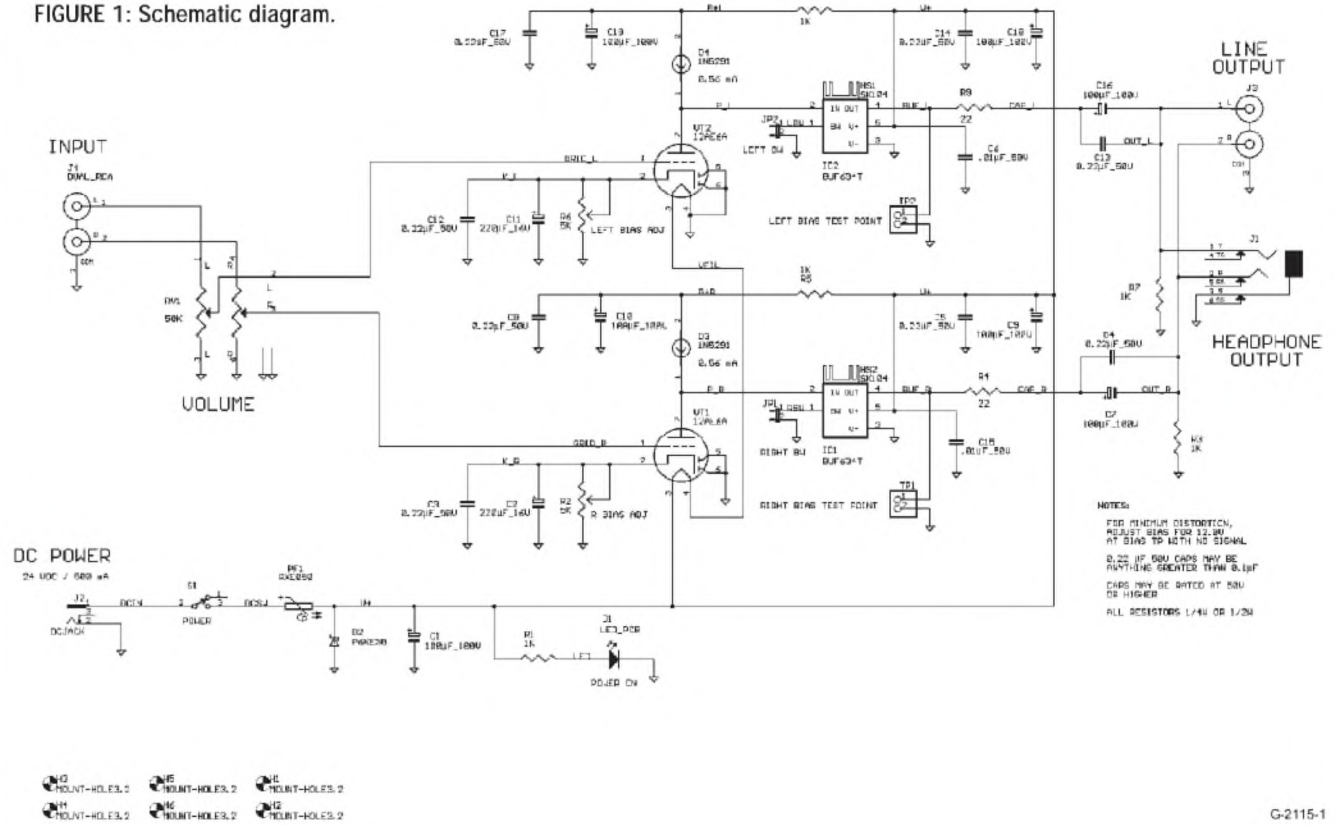
You can also experiment with using a resistor in place of the constant-current diode as a plate load. I tried resistors in the 47k to 100k range. I found that I achieved lower distortion and higher output levels with the constant-current diode. I didn't do extensive listening tests with the resistor load, though.

The DC voltage present (with no signal) on the plate of the tube varies, depending on the setting of the bias resistor. I'll discuss this setting in detail later, but normally this voltage is between 12V and 20V.

Output Stage

The plate of the tube is directly coupled to a unity-gain buffer amplifier IC, the

FIGURE 1: Schematic diagram.



BUF634, which is made by Burr Brown (now owned by Texas Instruments). The BUF634 is a follower, meaning that it has no voltage gain; the output voltage is the same as the input voltage. It has very low output impedance, and can provide up to 200mA of current from its output. Its input impedance is very high, so it doesn't load the tube stage significantly.

For those familiar with the BUF634, this application seems a bit strange. The typical use of the BUF634 is to be connected to the output of an op amp to boost its current capability. Normally, it is placed inside a feedback loop and is powered with bipolar (positive and negative) power supplies. Here, the BUF634 is used as an open-loop buffer, powered with a single positive supply. The DC coupling to the plate is required to provide the DC bias needed for the BUF634 to operate.

Normally, with no connection made to the BW pin, the BUF634 operates in a very low quiescent current mode. If you desire, you can operate the BUF634 in a wide-bandwidth, high-bias mode, by connecting the BW pin to ground (at JP1 and JP2). This lowers the open-loop distortion of the part ever so slightly.

The difference in THD is barely measurable, but I found that the character of the distortion did change. In the high-bias mode, I saw fewer odd harmonics. This is the mode that I used, but feel free to experiment with both settings.

The output of the BUF634 is connected through a 22Ω resistor, which is needed only to help protect the BUF634 in case of a short circuit of the output, but it also affects how different headphones sound. I usually recommend a series resistor of between 10% and 50% the impedance of your headphones—e.g., if you have 200Ω headphones, use a resistor between 20 and 100Ω. If you don't know what impedance your headphones are, or are going to use several different headphones, stick with a smaller resistor (such as 22Ω). Again, you can experiment with this resistor value to see what differences you hear without worrying about hurting anything. For line amp use, the value of the resistor makes very little difference.

Since the BUF634 is being operated with a single-ended power supply, its

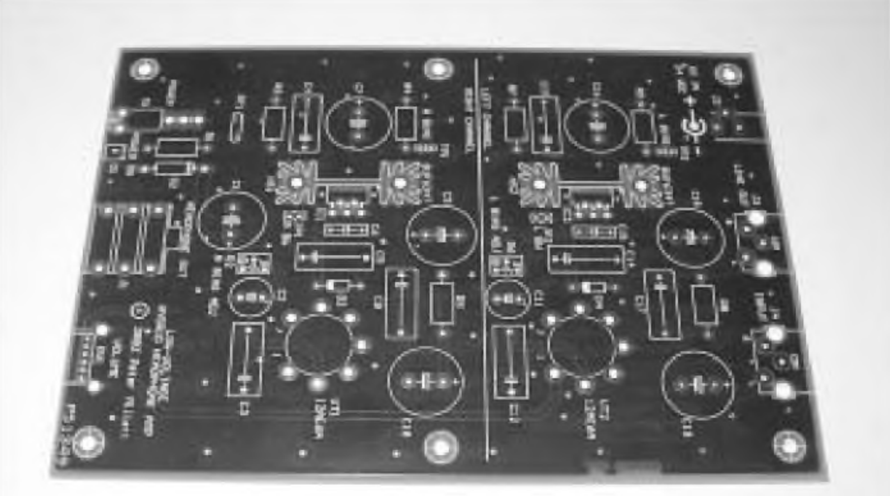


PHOTO 3: Bare PC board.

TABLE 1
PARTS LIST

REFERENCE	DESCRIPTION	MANUFACTURER/PN	DISTRIBUTOR/PN	COST EACH
C1, C7, C9, C10, C16, C18, C19	Capacitor, electrolytic, 100µF 100V	Elna ROA 100µF 100V	Welborne ROA102	$2
C2, C11	Capacitor, electrolytic, 220µF 16V	Elna ROA 220µF 16V	Welborne ROA221	$0.80
C3–C5, C8, C12–C14	Capacitor, film, 0.22µF 50V	Wima MKP10 0.22µF 160V	Welborne WM214	$1.80
C6, C15	Capacitor, axial ceramic, 0.01µF 50V	generic	Digi-Key 1103PHCT	$0.12
D1	LED, right-angle PCB mount	Dialight 550-0205	Digi-Key 350-1002	$0.69
D2	Transient suppressor, P6KE30	P6KE30A	Digi-Key P6KE30ADICT	$0.47
D3, D4	Current regulator diode, 1N5291	1N5291	Mouser 610-1N5291	$1.78
HS1, HS2	Heatsink, PCB mount	Aavid 531002B02500	Digi-Key HS190	$1.20
IC1, IC2	IC, buffer, BUF634T	TI BUF634T	Digi-Key BUF634T	$6.20
J1	Jack, ¼″ headphone	Rean	Mouser 550-22302	$1.36
J2	Jack, DC power, 2.5mm pin	CUI-Stack CP-102B	Digi-Key CP-102B	$0.38
J3, J4	Jack, dual RCA	DGS	Mouser 161-4219	$0.57
PF1	PTC fuse, RXE050	Raychem RXE050	Digi-Key RXE050	$0.59
R1, R3, R5, R7, R8	Resistor, 1kΩ ¼W	generic	Mouser 71-RN60D-F-1.0K	$0.21
R2, R6	Trimpot, 5kΩ	Bourns 3266W	Digi-Key 3266W-502	$3.58
R4, R9	Resistor, 22Ω ¼W	generic	Mouser 71-RN60D-F-22.1	$0.21
RV1	Potentiometer, stereo audio, 50kΩ	Panasonic	Digi-Key P2Y7503	$2.53
S1	Switch, toggle, PC mount	C&K	Digi-Key CKN1059	$4.50
VT1, VT2	Tube, 12FM6 or 12AE6A (see text)		AES 12FM6 or 12AE6A	$3.10
at VT1, VT2	Tube socket, 7-pin mini		AES P-ST7-195	$0.50
at RV1	Knob, press-on 6mm shaft	Rean	Mouser 550-67001	$0.46
	Case, plastic	Serpac 071I	Digi-Key SR071-IB	$8.88
	Wall supply, 24V DC 400mA	CUI-Stack DPD240040-P6P	Digi-Key T520-P6P	$8.75
	PCB			$20
				Total: $100.18

output sits at a DC voltage above ground (the same as the tube plate). To connect to headphones or other audio equipment, the DC must be removed with a coupling capacitor. I used an audio-grade 100µF electrolytic capacitor, paralleled by a small film capacitor.

The exact value of these caps is not critical, but it does set the low-frequency response limit of the amplifier. For most headphones, anything over 47µF is adequate. The output side of the capacitors then connects to both the headphone jack (J1) and an RCA line output jack (J3).

Power Supply

Input power is provided by a 24V DC wall-mount supply through the DC input connector, J2. By using an off-the-shelf DC wall adapter, there's no AC line voltage present anywhere in the headphone amplifier, so it's very safe. The supply does not need to be regulated; any voltage between 20V and 28V is fine.

The DC power is controlled by the power switch S1, and then flows through a PTC fuse device, PF1. This device is like a fuse, in that when too much current flows through it (in this case, over 500mA), it becomes an open circuit, stopping current flow. It is different from a fuse in that once it has a chance to cool off, it recovers and closes the circuit again.

D2, which is connected between the PTC fuse and ground, is a 30V transient protection diode. Forward biased (anode positive), it conducts current like a normal diode; reverse biased (anode negative), it does not conduct until 30V is exceeded, at which point it conducts. The purpose of this device, in conjunction with the PTC fuse, is to protect the circuit from the connection of a DC supply that either is wired with the wrong polarity or exceeds 30V. In either case, the transient protector will conduct, essentially shorting the supply, which will cause the PTC fuse to open.

A power-on LED, D1, and its current limiting resistor, R1, provide a visual power-on indicator. C1 acts as a filter, helping lower noise and hum coming in on the DC power.

The 24V DC power is applied to the filaments of the two tubes, which are connected in series. As long as the two tubes are the same, each filament will get one half of the 24V supply, or 12V. Note that the tubes designed for car radios are designed to work correctly with any voltage between 10 and 16V on their filaments.

The 24V DC power is also used to provide power to the two BUF634 amplifier ICs. This power is decoupled, or filtered, with several capacitors in parallel—an electrolytic capacitor, a film capacitor, and a ceramic capacitor. The reason for this is to provide a low impedance over a wide frequency range to the IC. Each type of capacitor has a low impedance over a different frequency range, and paralleling them accomplishes this.

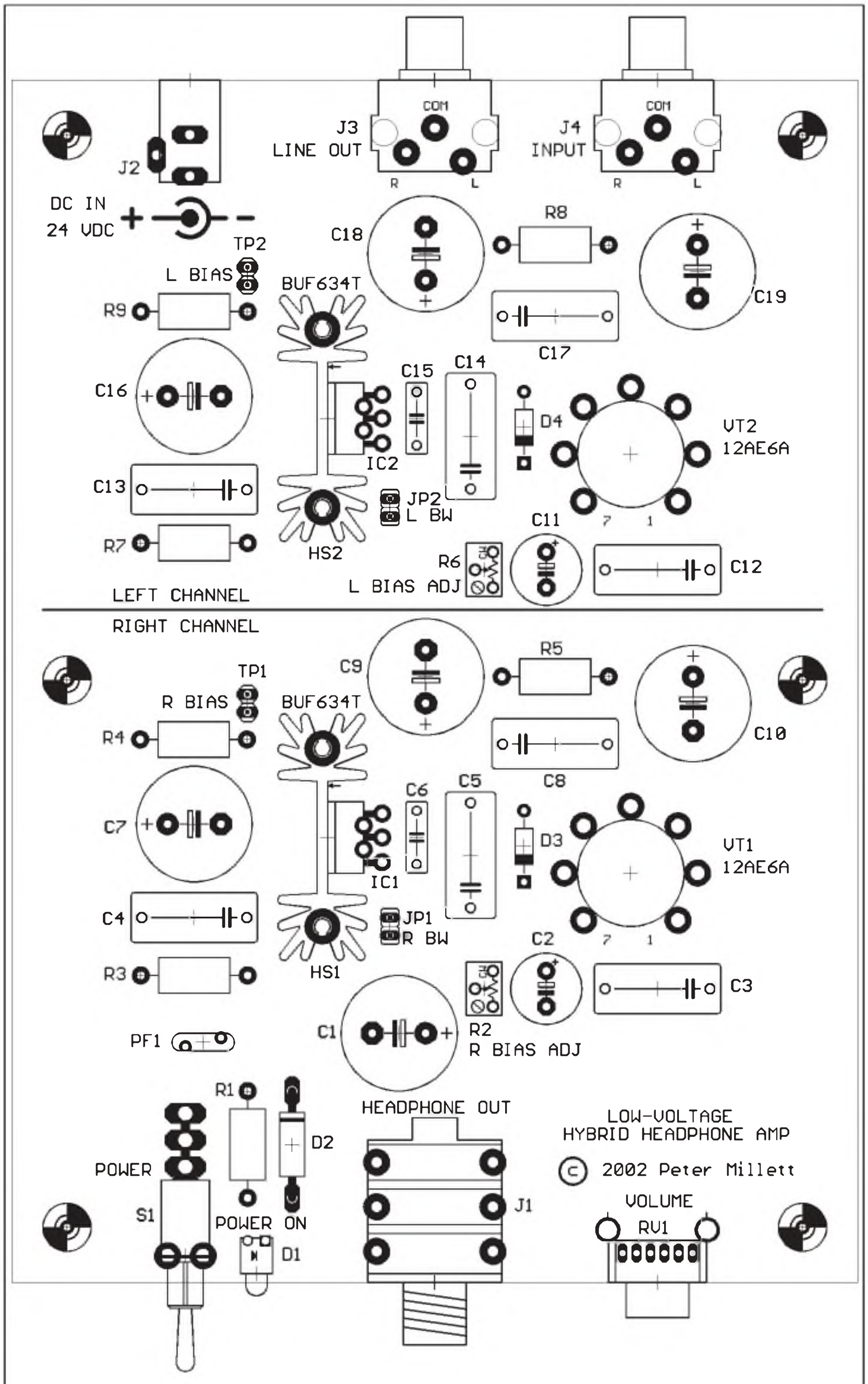


FIGURE 2: PCB component layout (PCB size: 4.75″ × 6.75″). G-2115-2

Power for the voltage amplifier tube stage is further filtered by a 1k resistor and another 100µF electrolytic capacitor, paralleled by a small film cap. The tube stage is sensitive to hum or noise on the power supply, so this filter prevents any ripple or noise present on the supply from being amplified and appearing at the output. The decoupling also prevents any feedback from the output stage to the input stage caused by perturbations of the power supply.

CONSTRUCTION

Since my goal with this project was to do something that would be easy to build, I designed a printed circuit board (PCB), which contains all of the components, including input and output connectors and the volume control (*Photo 3*).

Assembly

Assembly is a simple matter of inserting the components into the PCB, soldering the leads to the board, and trimming any excess wire from the back. Make sure that you install the electrolytic capacitors and diodes in the right orientation, matching the designation on the PCB.

The BUF634 buffers are bolted to small PCB-mount heatsinks. I found that even in high-bias mode, they run barely warm to the touch during normal operation, but the heatsinks will help protect the part in the event of a short-circuited output.

Since there are no dangerous high voltages present, I designed the PCB to mount into one half of an inexpensive plastic instrument case, with the top side of the PCB exposed. I just mounted the board into one half of the enclosure and discarded the other half (*Photo 1*). This made a simple, easy way to mount the PC board, and still allow access to the board to adjust bias, change tubes, and make measurements. You could also mount the PCB into a more conventional metal box if you choose.

Once the PCB is assembled and mounted to the plastic case, all you need to do is install the tubes into their sockets, plug in the power supply, and adjust the tube bias as detailed later. Of course, the more experienced builder could also build this project using conventional point-to-point wiring inside a chassis.

Parts

I used only parts that are readily available for a reasonable cost from mail-order distributors that cater to hobbyists. *Table 1* is a listing of all the parts used, where I purchased them, and about what they cost. Refer to the contacts listing at the end of the article for information on how to get in touch with the vendors listed.

The parts list is all-inclusive, including the plastic enclosure, volume control knob, and so on. You can see that the entire project can be assembled for about $100. The only tools you'll need are a screwdriver, wire cutters, and soldering iron. You'll also need a voltmeter (any analog or digital meter will do) to set the bias, which I'll describe later.

Exact part selection is not at all critical. Capacitors should be rated for at least 50V, except the cathode bypass capacitors, which can be rated as low as 16V. Capacitance values can vary between about 50% and 200% of the values

I used with little change. Resistors, too, can be anything close to what I used with not much effect.

The PCB

I hope to be able to provide bare PCBs for sale to individuals who want to build this project. You can also make your own, or have them made in small quantity for a reasonable amount of money at prototype PCB vendors who specialize in small runs.

Figure 2 shows the top side of the PCB, showing parts placement. *Figures 3* and *4* are the top foil and bottom copper foil layers. You can also download the artwork files from the author's website at: www.pmillett.addr.com.

SETTING THE BIAS

I decided to make the bias voltage for the tubes adjustable, partly so you could easily try out different tubes, and partly so you could try different operating points for the tubes. Since the bias voltage is just developed across the cathode resistor, making this resistor a trimmer potentiometer provides an easy way to vary the bias.

If you measure the voltage at the output of the BUF634 with no audio applied (which is the same DC voltage as on the plate of the tube), you can set the operating point of the tube by adjusting the bias trimpot. Measuring at the output of the BUF634 guarantees that the voltmeter won't load the voltage on the high-impedance plate.

Since the plate load is a 0.56mA constant-current diode, moving the bias point around does not affect the plate current. If you substitute a resistor for the constant-current diode, you can still adjust the bias in the same manner, but the plate current will vary with the bias setting.

Setting the bias is a great way to experiment with the "sound" of different tube distortion. For example, as you adjust the bias to get a progressively lower plate voltage, you get more and more "single-ended" second harmonic distortion. As you raise the voltage to one half the supply voltage, you can get a higher output voltage before clipping, at the expense of higher third-harmonic distortion at lower levels. Raising the voltage further lowers the distortion products at low signal levels at the expense of a lower maximum output level.

I looked at the distortion products of the output using an audio analyzer, but you really don't need sophisticated test equipment to set the bias on your amplifier. If you just put a voltmeter on the bias test points, you can adjust the bias based on the DC voltage you measure there. Then, use your ears to evaluate the result. I found that you really can hear differences in sound with changing the bias, especially in low-level detail.

With both the 12AE6 and 12FM6 tubes that I tried, adjusting the bias to one-half the input power-supply voltage provided mostly symmetric clipping, and the highest output voltage. My wall supply was putting out 27V, so I set the bias to 13.5V. *Photo 4* shows what the output looked like at this bias setting, using a 12FM6 tube, driven hard into clipping.

Note that clipping is nearly symmetrical, but the top of the waveform is clipped more abruptly than the bottom. This is the point where the output hits the positive power-supply rail.

Photo 5 shows the output signal and distortion residual (what's left of the

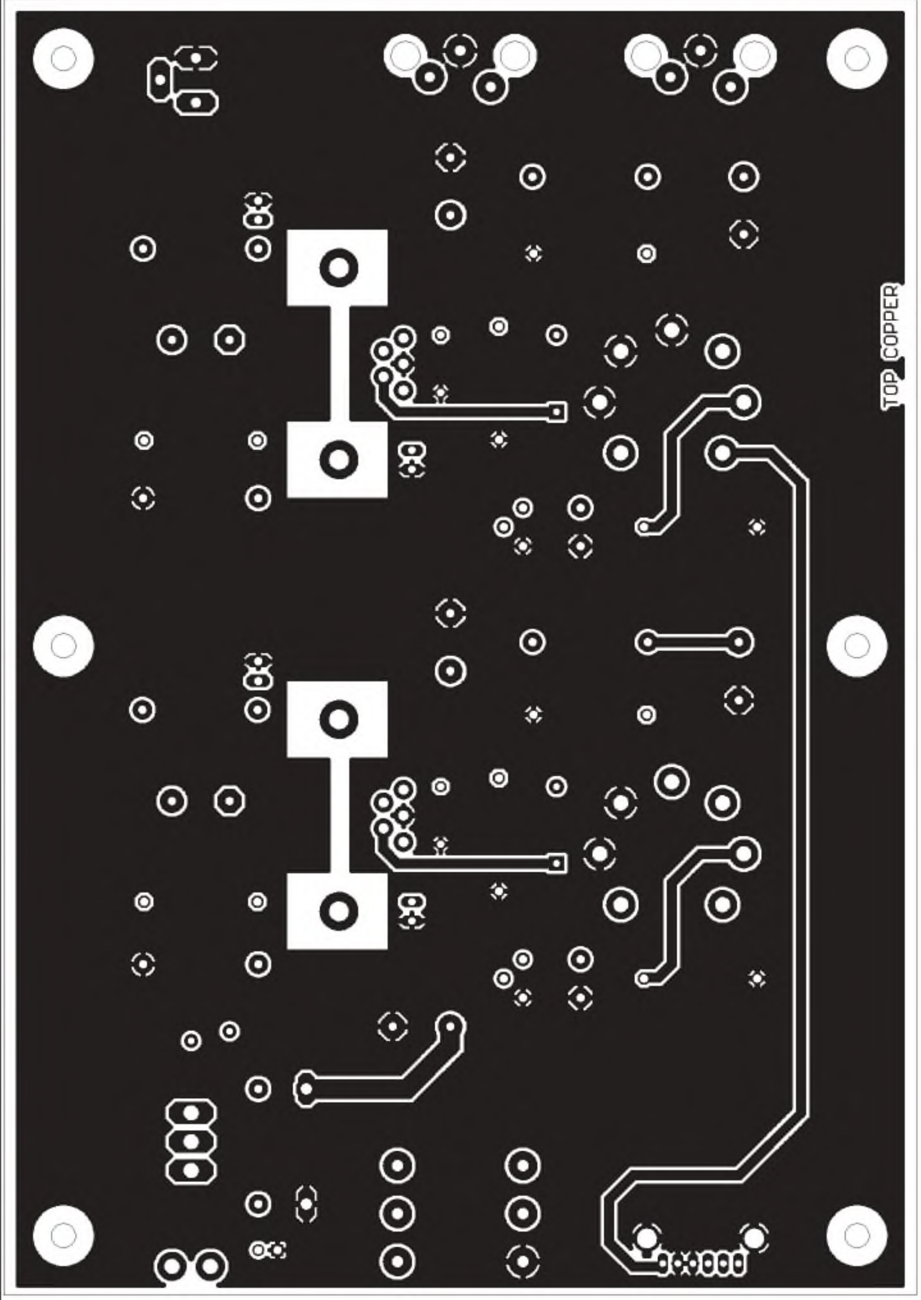


FIGURE 3: PCB top copper.

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signal after you cancel out the original input signal) for a 1kHz, 1V RMS output

PHOTO 4: 12FM6, 13.5V bias driven to clipping.

PHOTO 5: 12FM6, 13.5V bias, 1V RMS out (top) and distortion residual (bottom).

with the same bias. The residual, shown at a much higher scale than the

PHOTO 6: 12FM6, 19V bias driven to clipping.

PHOTO 7: 12FM6, 19V bias, 1V RMS out (top) and distortion residual (bottom).

output signal, is very nearly a sine wave at a frequency of 2kHz. This indicates that the distortion is primarily second harmonic.

If you adjust the bias voltage lower than one-half the supply voltage, you can avoid the sharp clipping at the top of the waveform—but distortion increases dramatically, since the tube becomes very nonlinear as the grid begins being driven positive with respect to the cathode. If you adjust the voltage above one-half the supply voltage, you can reduce the distortion at 1V RMS out, at the expense of slightly decreasing the maximum output that can be obtained before clipping.

Photo 6 shows the waveform in heavy clipping at a bias voltage of 19V. You can see that the top of the waveform is more clipped than in *Photo 4*. However, the distortion at 1V RMS out is actually slightly lower (*Photo 7*). I found this bias setting to be much more pleasurable to listen to than the lower plate voltage bias point.

CIRCUIT PERFORMANCE

For those not used to tube circuits, the distortion figures that follow will seem large. Indeed, this is not a low-distortion amplifier, but THD alone is not much of an indicator of perceived sonic performance. In fact, I believe that much of the appealing sound of a single-ended triode amplifier has to do with the introduction of second-order harmonics into the music. That's one of the purposes of this project—to allow you to listen to, and experiment with, this distortion.

I made distortion measurements of the amplifier, using both a 12AE6A tube and a 12FM6 tube, at two different bias settings, both using a 0.56mA constant-current diode as the plate load. Here are the results:

12AE6A, bias = 13.5V
Maximum output at clipping: 2.7V RMS
Maximum output, 5% THD: 1.8V RMS
THD, 1V RMS out: 0.6%, largely third harmonic

12AE6A, bias = 19V
Maximum output at clipping: 2V RMS
Maximum output, 5% THD: 2V RMS
THD, 1V RMS out: 0.5%, virtually all second harmonic

12FM6, bias = 13.5V
Maximum output at clipping: 3V RMS
Maximum output, 5% THD: 2V RMS
THD, 1V RMS out: 1.5%, mostly second harmonic

12FM6, bias = 19V
Maximum output at clipping: 2V RMS
Maximum output, 5% THD: 1.7V RMS
THD, 1V RMS out: 1%, virtually all second harmonic

The relatively low maximum output levels are limited by the low plate voltage used on the tubes. You can experiment with different tubes, different bias settings, and different constant-current diodes, and probably find operating points different than the ones I used that may provide higher output, and/or lower distortion.

For most headphones, the output from this amp is adequate to drive to quite loud listening levels. With Sennheiser HD600, Beyerdynamic DT831, and Grado SR60 headphones, there was ample output before distortion to well beyond the loudness that I can tolerate. This was verified both by ear and by looking at the waveforms to verify that the amp was not driving anywhere near clipping.

With my AKG K240 headphones, the amp couldn't drive as loud before reaching the onset of clipping. These are 600Ω impedance headphones, which require quite a lot of voltage. I would say this amp was marginally acceptable driving them.

Just for comparison, I measured the output level at the headphone jack of two portable CD players. With the first unit, a brand new midrange player, I could not drive the headphone output to clipping. The maximum output level, with a 0dBFS test CD, was 0.4V RMS. I could drive the second CD player, an older and more expensive unit, into hard clipping at 3V RMS.

As a line amplifier, the output of this amp should be more than adequate to drive all but the most insensitive power amps. Its low output impedance should be able to drive just about any interconnect cable.

The frequency response measured very flat, within ±0.1dB from 20Hz–20kHz, into a 200Ω load. At 100kHz, the

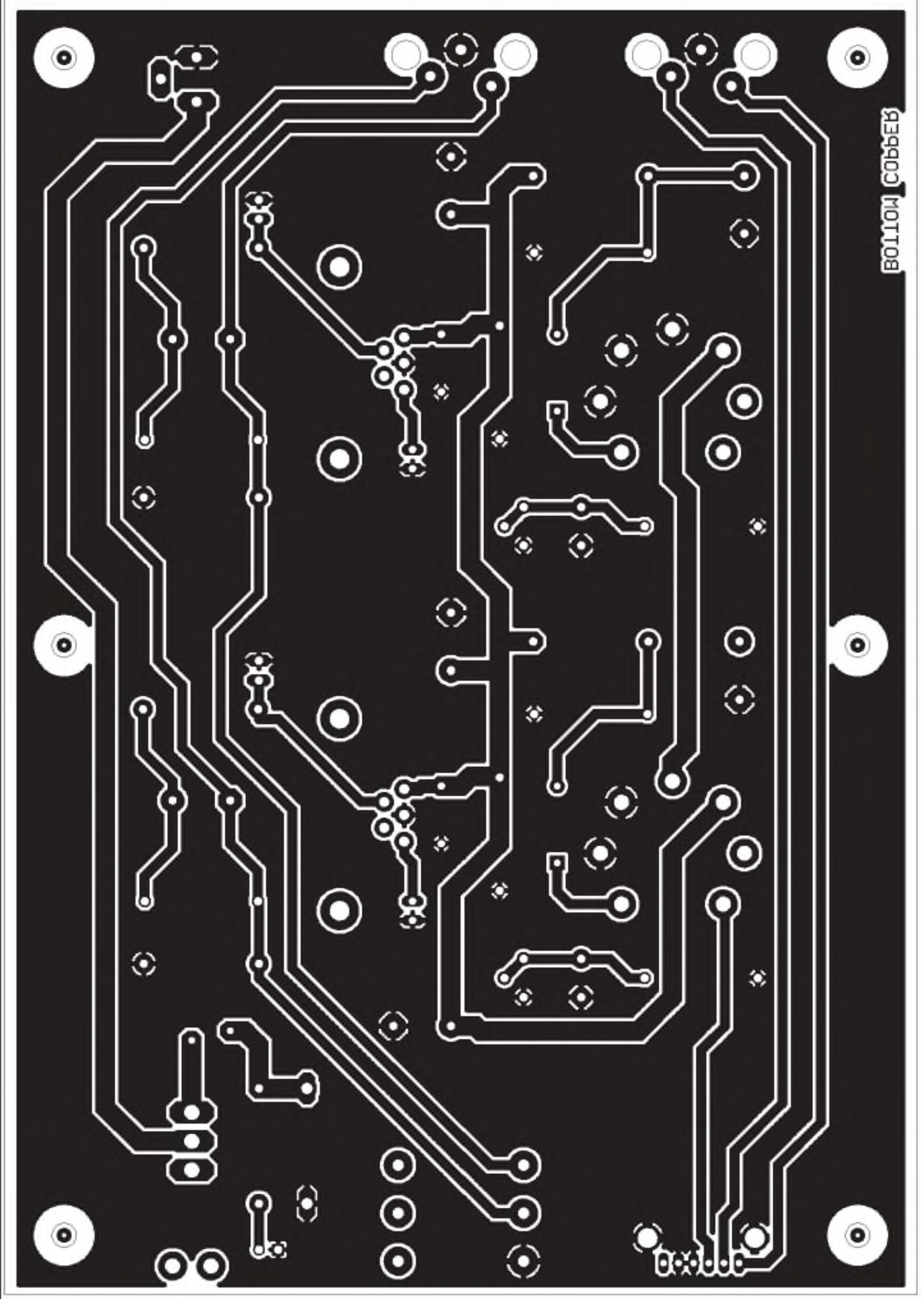

FIGURE 4: PCB bottom copper.

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limit of my audio analyzer, the output was down only 0.8dB. Into a lower impedance load, the low-frequency response will drop a little, since the output is coupled through a capacitor. With the 100µF capacitor I used, you can expect about –3dB into a 30Ω load at 20Hz.

I measured the noise at the output (terminated in a 200Ω load) with no input signal at 200µV. This was an un-weighted measurement, which corresponds to –74dB below 1V RMS, very near the measurement limit of my test setup. This is very quiet for a tube amplifier.

CONCLUSION

I'd be lying if I were to tell you that this is the best-sounding headphone amp I've ever listened to. But I've listened to dozens of headphone amps, some of which cost more than the average new car. This amp does well, considering its cost and the design compromises I made. It will certainly be an improvement over what you would hear with headphones plugged into a portable CD player. And being a fan of tube sound myself, I think it sounds a whole lot better than one of those $350 "op-amp in a wooden box" audiophile headphone amps.

The main goal with this project is not to build a headphone amp, but rather to build a project that can be a positive learning experience for someone who's just getting started with building and designing audio equipment. I think from that perspective, this design is a success: It's an easy, inexpensive project that will allow you to experiment with the sound of tubes. ❖

CONTACTS

Antique Electronics Supply (AES)—tubes, tube sockets, capacitors, and so forth.
www.tubesandmore.com
(480) 820-5411

Welborne Labs—resistors, capacitors, audiophile parts, and so on.
www.welbornelabs.com
(303) 470-6585

Digi-Key—full-line parts distributor
www.digikey.com
(800) 344-4539

Mouser Electronics—full-line parts distributor
www.mouser.com
(800) 346-6873

Pete Millett—author
www.pmillett.addr.com
pmillett@hotmail.com

Tuning in on the AM Band

The AM air waves, experiencing a revival, are abuzz with chatter. Here's how you can get in on the action.

By Carlos E. Bauzá

PHOTO 1: The Terk AM Advantage.

Listening to the radio on the AM (amplitude modulation) band was a strong social practice up to the 1950s. The family gathered around the radio for clear reception of a wide program menu. The way programs were designed, the imagination was piqued to visualize images that were grander than visual effects. But FM and TV captured the public's attention and seemingly relegated AM to oblivion.

AM producers were not willing to throw in the towel, though. They capitalized on AM's strong suit, which is carrying the human voice. And they had an eager audience listening from automobiles. Still, a new social custom is threatening AM yet again . . . talking on the cellular phone while attempting to drive, and chatting on the web.

Most of what humans talk about is carried on the AM band now. You receive a slice of real life whenever you tune in. Talk shows abound, with substantial audience participation through the telephone. You can see yourself in the voice of the callers!

Some shows are nationally syndicated. Dr. Dean Idell answers your medical questions from California. Dr. Joy Brown suggests strategies for solving relationship problems. Paul Harvey researches "The News" in his own unique way, and he later tells you "The Rest of the Story." Larry King is still on the radio.

Local shows inform you about the coal-burning power plant the government plans to build in your backyard. You can let the mayor know how you feel about his administration. You can sell items through flea markets "on the air." And experts often discuss subjects you never see on TV. You get a vague feeling of "the underground"; yet, AM is a human facilitator.

ABOUT THE AUTHOR

Mr. Bauzá recently retired from banking. He was a dedicated kit builder and speaker builder. Now he enjoys listening to analog as well as digital sources. His favorite is making music as a tenor in a concert choir.

Many stations link up in a transmission chain for coverage over longer distances. Even so, reception can be affected by low signal strength and noise from several sources. One solution is to have an outstanding antenna, short of getting a sophisticated tuner, such as Dynaco's AF-6 made 30 years ago.

THE TERK "AM ADVANTAGE"

This passive device consists of a 9″ loop with a base and tuning knob. It induces the tuned signal into your radio with, or without, a wire connection. Manufacturer: Terk Technologies, 63 Mall Drive, Commack, NY 11725. Price: $55. Their products are distributed nationally.

This product is elegantly designed in dark gray (*Photo 1*). It is durable and works entirely from the energy of the tuned signal. You place it inches from a radio (or tuner) that is already tuned to a desired station. Then you turn the Terk's knob until reception suddenly improves and becomes louder.

Station WOSO, 1030 KHz, from San Juan is one of my favorites. Its transmitter is 35 air miles from my tuner, with a range of mountains in between.

My tuner is the Proton 440 AM/FM model, a simple design with adequate performance under ideal conditions. It has a six-LED signal strength display. I could tune station WOSO with only a long-wire antenna on the roof, but it came in with annoying noise, some distortion, and some fadeout.

The Terk "AM Advantage" substantially lowered the noise and distortion and improved the fadeouts. Signal strength now registers on one of the LED indicators. Reception of WOSO is adequate, but I cannot tune it in at night.

The Terk antenna is operated by turning a knob, which is smooth. But you must develop a soft touch for weak stations.

There is no doubt that my tuner works substantially better than with the roof antenna or the tuner's own coil. Reception is now pleasurable instead of annoying, but a little residual noise spurred me to try a more sophisticated system.

THE JUSTICE AM ANTENNA

This active system is powered by one 9V battery or an included "wall wart"

PHOTO 2: The Justice AM antenna.

supply. It is contained in two "black boxes" about the size of an outstretched hand (*Photo 2*). One box is the "head," containing the tuning coil plus sophisticated circuitry on a PC board. The other box contains the two concentric tuning knobs (coarse and fine) plus the special power-supply circuit.

This system induces a chosen signal into your tuner with or without a wire connection. You can place the tuning "head" anywhere it will give the best reception. You can optionally increase the dedicated connecting wire between the two boxes to 50′ through an extension purchased from the manufacturer. You can also install the "head" outside the house. Manufacturer is C. Crane Company, 1001 Main Street, Fortuna, CA 95540-2008, 1-800-522-8863, www.ccrane.com. Price is $99.95.

The Justice AM Antenna brings specific refinements over the Terk AM Advantage (which already is a substantial improvement over your radio). Here's how.

First, you can place the "head" in a wider choice of locations for the best reception. It also has a grounding receptacle, which you can use for possible noise reduction on a case-by-case basis. The Justice's tuning knobs are so smooth and noncritical that tuning is easy without concentrated dexterity.

Sangean now builds this model for C. Crane Co. This makes a lower price than the original $150 possible. Also, they made a cosmetic change from two separate tuning knobs to concentric ones, and I did not get a chance to test them for smoothness of motion.

A well-tuned station comes through the Justice with higher signal strength, somewhat lower noise, and less fadeout. WOSO now registers on two display LEDs, with almost perfect reception quality. In short, the Justice antenna yields several small improvements over the Terk, which add up to discernibly better reception.

But that's not all! Both antennas together can be better than one alone! Tune the Justice system to a weak station. Then, place the Terk near your radio, and re-tune both antennas. Then, try rotating the antennas 90° for possible improvement. You will now receive the weak station at its optimum possibility. I would not part with either one of these fine products.

An added benefit of both antennas is that you can take them with your portable tuner almost anywhere. The Terk is more convenient for this. But these products are not designed for automobile use. They can be a lethal distraction while driving.

You can purchase both antennas from C. Crane Company, which also has a 94-page catalog of technological items, plus many books on radio and other subjects, including an AM-FM directory listing 15,000 stations in the US and Canada by call letters, frequency, city, and state. They also have a wonderful fired-enamel-on-steel replica of "Nipper" listening to "His Master's Voice." I have one hanging on the wall. Highly recommended! ❖

A Drum-Head Midrange

Here's an interesting restoration project that brings new life to a midrange. **By Thomas C. Coerver**

I had a full-range 8″ speaker with a deteriorated foam surround and a retired snare drum head. After cutting away most of the main cone down to the level of the whizzer cone, I cut out a diaphragm from the drum head (*Fig. 1*), folded it in half, and used epoxy to glue it into a slot cut in the remaining cone material (*Photo 1*).

PHOTO 1: Glue diaphragm to voice coil.

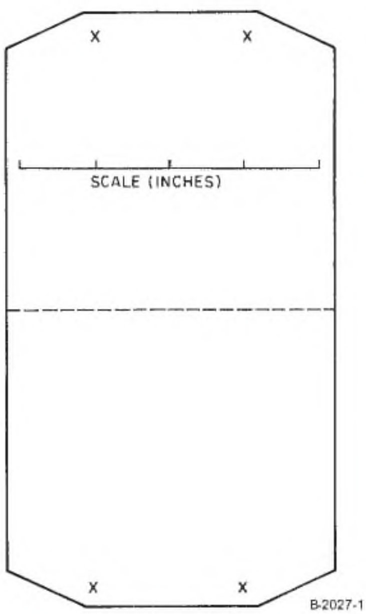
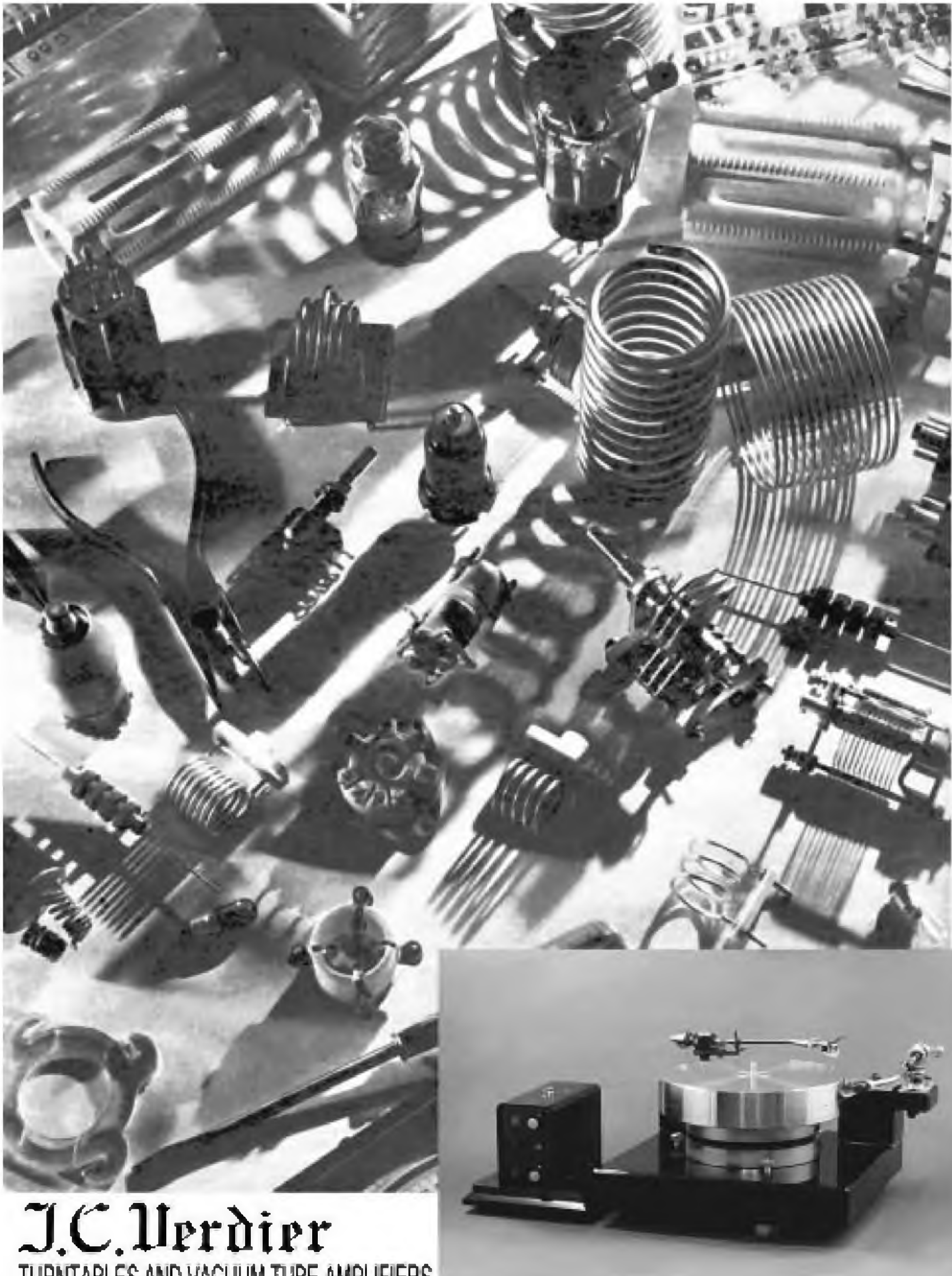


FIGURE 1: Pattern for diaphragm cut from drum head. Fold 180° at dotted line. Punch holes approximately ¼″ at Xs.

PHOTO 2: Markings on side of cabinet show internal baffles of transmission line bass portion with 8″ woofer. I stuffed the TL with polyester fiberfill from a discarded bed pillow. Midrange section on top is open to rear dipole style. To minimize diffraction, I covered the front with plastic foam and rounded all edges with a router.

ABOUT THE AUTHOR

Tom Coerver, a retired Professional Engineer (MSChE), has been a speaker builder and electronics hobbyist since serving as an electronics technician in WWII. His career (33 years in chemical manufacturing and ten years in environmental regulation) left little time for hobbies, but his eight years of retirement have allowed more time for them (when he is not busy with bowling or golf).



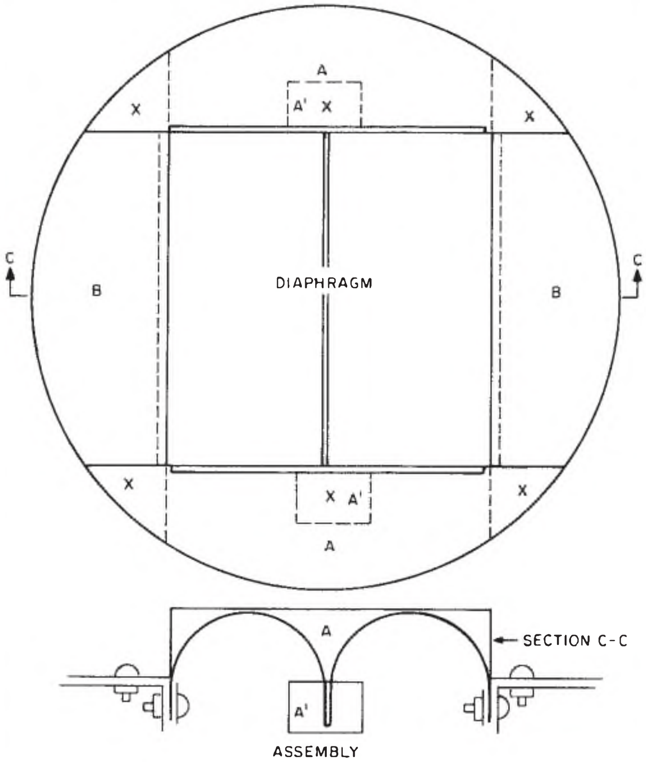


FIGURE 2: Assembly.

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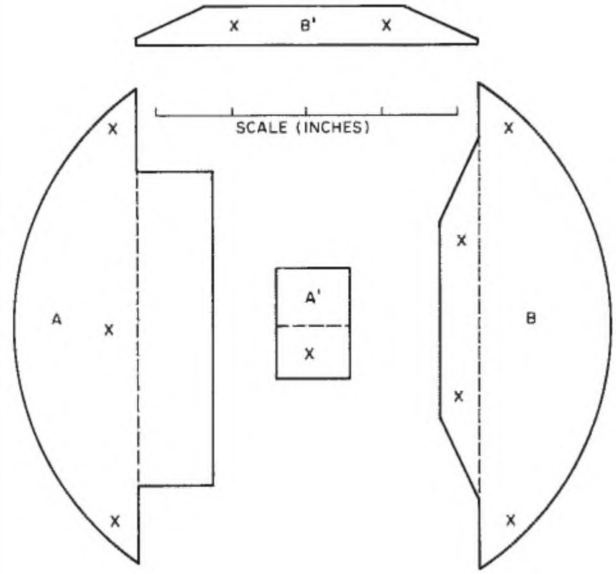

FIGURE 3: Pattern for front baffle (two needed). Use tin snips to cut from sheet metal. Bend 90° at dashed line. Drill ⅛" holes at Xs.

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PHOTO 3: Completed speaker with Linaeum tweeter ready for A-B test versus AR3a.

Using sheet metal, I fashioned the front baffle, to which I fastened the free ends of the diaphragm (*Photo 2*). *Photo 3* shows this midrange speaker in a TL enclosure. The back of the midrange is open to the rear.

ASSEMBLY

After cementing the diaphragm in place (*Fig. 2*), you should start assembly by bolting the "B" pieces (*Fig. 3*) to the free ends of the diaphragm. Then place the "B" pieces in position, curling each side of the diaphragm in the process.

Before putting the "A" pieces in place, feed a signal to the voice coil and adjust the diaphragm position as needed to eliminate any voice-coil dragging. This took me several tries to get it right. Finally, put the "A" pieces in place and bolt up.

I originally used plastic foam between the edges of the diaphragm and the flanges of the "A" pieces. But I later removed it, and the sound improved. I have no means to measure the response of the finished speaker, but to my ears it is as smooth as the AR3a, but is more transparent and handles transients better. ❖

Damping Loudspeakers in Series

Are there any deleterious effects in connecting two speakers in series? We turn to our speaker sleuth to provide the solution. **By Dick Pierce**

An entire mythology surrounds the notion of damping in loudspeakers. We have separately treated the oft-cited and nearly useless "damping factor" specification in a separate article ("Loudspeakers 101," *SB* 6/97, p. 52). Here, I discuss much of the myth surrounding the behavior of loudspeakers connected in series and the seemingly intuitive effect on damping that results. Of late, this topic is relevant for several reasons.

INTRODUCTION

First, we have seen a gain in use of multiple-woofer systems, exemplified by the popular, so-called "D'Appolito" configuration (often the woofers are connected in parallel, but there are applications in which a series connection could be appropriate as well). Second, there is an unfortunate trend in consumer electronics, especially in home theater receivers. More and more, these receivers have skimped on power supply and/or output stage design and are quite unable to drive the lower impedances of today's speakers. The question often arises whether speakers can be hooked in series, thus raising the impedance.

The answer often given to both of these scenarios is "Oh, no, you can't do that. The damping of each speaker will be severely reduced because of the series impedance of the other speaker!"

As you might have guessed, I will show you why this is wrong by taking an analytical approach, and testing analysis by actually measuring actual systems.

WHAT IS DAMPING?

The term "damping" has a very specific and unambiguous definition: technically, it is a measure of how quickly energy is removed from a resonant system. This definition stands despite attempts to co-opt the term for otherwise imprecise and often incorrect uses.[1] It is a measure of how quickly a resonant or oscillatory system is brought under control by removing energy that would otherwise keep the resonance going.

Energy is stored in reactive elements. These include masses and compliances (or springs) in the mechanical world, and inductances and capacitances in the electrical world. Mechanically, the energy of a mass is the kinetic energy due to the motion of a mass. The kinetic energy of a moving mass is equal to the mass times the velocity squared. In a spring or compliance, the energy is stored as potential energy in the compression or extension. The potential energy is equal to the spring constant times the compression or extension squared.

Electrically, kinetic energy is in the magnetic field around an inductor produced by the current flowing through the inductor. The energy is equal to the inductance times the current squared. In a capacitor, it's the potential energy in the electric field caused by the impressed voltage on the plates of a capacitor, equal to the capacitance times the voltage squared.

Energy is removed through loss mechanisms, such as frictional losses in the mechanical domain or ohmic losses in the electrical. These losses convert energy to heat, and once that happens, the energy is no longer available.

In loudspeakers, there is a direct measure of the ratio of energy stored to energy lost, and that is the so-called "Q" factor. In most loudspeakers, there are two predominant loss mechanisms, each with their own Q measurement. The Q due to mechanical losses is designated as Q_M, while that for electrical losses is designated as Q_E[2]. You can calculate these Q factors knowing the energy storage and losses mechanisms involved.

The mechanical Q_M results from the energy storage in the moving mass of the cone and the frictional losses in the suspension. It is calculated thus:

$$Q_M = 2\pi F \frac{M_M}{R_M}$$

where F is the resonant frequency of the system in hertz, M_M is the mechanical mass of the system, and R_M is the frictional loss in the system. Increase the mass, and more energy is stored in the system. Increase the friction, and more energy is dissipated from the system.

The electrical Q_E of the system results, again, from the energy stored in the moving mass, but now dissipated by the electrical resistance in the system. It is calculated as:

$$Q_E = 2\pi F \frac{M_M}{B^2 l^2} R_E$$

Here, B represents the magnetic flux

ABOUT THE AUTHOR
Dick Pierce has worked as a consultant to the industry for over a quarter of a century. That time has been divided between consulting with loudspeaker driver and systems manufacturers and professional digital editing systems manufacturers. In addition, he has written numerous articles on the fundamental physics of loudspeakers—digital audio fundamentals, and so on. In his spare time, Mr. Pierce has revived a 35-year old interest in fine-art photography, and regularly attends performances of baroque musical works.

density in the voice coil gap, l is the length of wire in the magnetic field, and R_E represents the DC resistance of the voice coil.

Of course, you can also consider the total Q, or Q_T of the system due to the combined mechanical and electrical damping, and it's calculated by the familiar formula:

$$Q_T = \frac{Q_M \times Q_E}{Q_M + Q_E}$$

The lower the Q, the more damped the system is. The higher the Q, the less damped.

THE INTUITIVE PREMISE

Here's the claim: Putting speakers in series is a bad idea because the series resistance of one speaker destroys the damping of the other. Why, even the equation for electrical Q_E says so: Having two voice coils in series doubles the voice-coil resistance (assuming for simplicity the voice coils are the same). So it must logically follow that adding two speakers in series must severely destroy the damping, and the previous equation shows that it should double.

It makes intuitive sense. It even seems to appeal to technical authority. But will it stand up to analytical and empirical scrutiny? Is this, perchance, another widely held belief that might not be so?

ANALYSIS

Consider the case in which you connect two of the same devices in series, be it two identical woofers in an enclosure or two identical speakers in series. Use identical woofers or systems to make the analysis simpler[3].

First look at the effects of two speakers connected together in the mechani-

TABLE 1
DRIVER MEASUREMENTS

PARAMETER	MEASUREMENT A	B	A+B	PREDICTION ANALYTICAL	INTUITIVE
Resonance F_S	71.20	69.03	70.1		
DC resistance R_E Damping	5.72	5.70	11.42		
Mechanical Q_M	1.29	1.33	1.32		
Electrical Q_E	0.88	0.92	0.91	~0.90	>1.80
Total Q_T	0.52	0.54	0.55	~0.53	>1.06

TABLE 2
SPEAKER SYSTEM MEASUREMENTS

PARAMETER	MEASUREMENT A	B	A+B	PREDICTION ANALYTICAL	INTUITIVE
Resonance F_S	110.5	113.7	112.6		
DC resistance R_E Damping	6.86	6.87	13.90		
Mechanical Q_M	2.77	3.06	3.15		
Electrical Q_E	1.02	1.09	1.14	~1.06	>2.28
Total Q_T	0.75	0.80	0.83	~0.80	>1.6

cal domain. It might seem obvious, but since no electrical effects are considered in the mechanical domain, it makes no difference on the mechanical damping or Q_M whether two speakers or two woofers are connected in series or parallel. Indeed, it doesn't even make any difference whether they aren't connected at all electrically.

In the mechanical case, you have doubled the moving mass M_M to 2 M_M (with twice as many cones, after all), but you have also doubled the amount of frictional loss from R_M to 2 R_M as well (twice as many surrounds and spiders, too). Plugging these changes into the equation for mechanical damping gives:

$$Q_M = 2\pi F \frac{2\,M_M}{2\,R_M}$$

You can simplify: 2/2 in the equation is equal to one, and thus you end up with:

$$Q_M = 2\pi F \frac{M_M}{R_M}$$

This equation, describing the effect on Q_M of connecting two speakers in series, is precisely the same equation for the case of a single speaker by itself.

Now look at the electrical damping or Q_E. Here, you have, indeed, doubled the resistance R_E to 2 R_E (the voice coils are hooked in series), but you have also doubled the moving mass from M_M to 2 M_M as well, and also doubled the length of the voice coil wire from *l* to 2 *l* sitting in the magnetic field. Now plug all those factors of two into the equation for electrical Q:

$$Q_E = 2\pi F \frac{2M_M}{B^2\,(2l)^2}\,2R_E$$

The next step expands and combines terms:

$$Q_E = 2\pi F \frac{2M_M}{B^2\,4l^2}\,2R_E$$

In another step, accumulate all these new factors (2 from the doubling of mass, 2 from the doubling of the voice-coil resistance, and 4 from the square of the doubling of the length of the wire) together for the numerator and the denominator:

$$Q_E = 2\pi F \frac{4}{4} \frac{M_M}{4\,(Bl)^2}\,R_E$$

And, since the fraction 4/4 is equal to one, you can reduce this equation to:

$$Q_E = 2\pi F \frac{M_M}{B^2\,l^2}\,R_E$$

Again, this result, showing the electrical Q_E of two speakers connected in series, is identical to the case of just a single speaker.

Thus the analysis clearly shows that the damping is not severely compromised by connecting two systems or drivers in parallel, because the measures of damping, Q_M and Q_E, remain the same for both the mechanical and electrical domains, and thus the total Q_T also remains the same. Q.E.D.

EMPIRICAL SUPPORT

The intuitive premise makes one clear prediction: The damping is seriously compromised by placing two speakers in series. This must be manifested by a substantial increase in the Q factors of the speaker. Specifically, the premise predicts that the electrical Q factor should be much greater.

How much greater is not clear, because the premise is woefully short of analytical precision. But, for instance, say that you should see at least a doubling of the electrical Q. And since the electrical damping predominates in most speakers, the total Q should be similarly affected.

On the other hand, this analysis predicts that the Q factors should remain essentially unchanged. Such an unambiguous difference makes this discussion an ideal candidate for falsification by experiment[4].

For the first experiment, I selected two woofers, a pair of Seas PR17RC 6½″ woofer-midrange drivers. I measured the relevant parameters, the DC resistance, resonance and the mechanical, electrical and total damping of each separately, and then with the two connected in-phase in series to see the effect on damping of such a series connection. The results are shown in *Table 1.*

The data seems to strongly support the analytical method's predictions, and refute those of the intuitive model. That's fine for single speakers, and this result has been validated in numerous home-built systems using multiple drivers in series. It might be a different question, though, for complete speaker systems, often the situation found in some installations.

So, I went to my storage room and grabbed a pair of rather ordinary bookshelf speakers, some ancient ones made by the old H. H. Scott company. I could well have used any two speakers, but these were handy and fully functional. I measured the resonant frequency, the DC resistance of the voice coil, and the relevant Q factors for each speaker alone, and the two in series. Along with these numbers, I also present the predictions made by the two competing theories, the "intuitive" premise and the "analytical" theory described previously (*Table 2*).

It seems that the empirical data

strongly supports the analytical model, and strongly refutes the intuitive premise. Q.E.D.

FREQUENCY-RESPONSE ERRORS

One problem with two speakers in series is that the frequency-dependent impedance variations of one will upset the frequency response of the other, and vice versa. As it turns out, this is not the case as well. (See "Damping Factor: Effects on System Response," *SB* 6/97, pp. 52-55).

Consider how the attenuation arises. Take the case of two resistances in series, R_1 and R_2. Given an impressed voltage of V_S, you can calculate the voltage across R_2. According to Ohm's law, the current through the entire circuit, I, will be:

$$I = \frac{V_S}{R_1 + R_2}$$

Given that current, again, by Ohm's law, the voltage across R_2 will be:

$$V_{R2} = I\,R_2$$

And, combining these two equations and simplifying, you find that:

$$V_{R2} = V_S \frac{R_2}{R_1 + R_2}$$

Now, in the case where $R_1 = R_2$, this reduces to simply:

$$V_{R2} = \frac{1}{2} V_S$$

Now, this can be generalized for impedances. If the impedances are the same, you can say that:

$$V_{Z2} = V_S \frac{Z_2}{Z_1 + Z_2}$$

Z_1 and Z_2 represent the complex, frequency-dependent impedances of the loudspeakers. If $Z_1 = Z_2$, which would be the case if the two speakers are the same (and this includes the frequency-dependent impedance variations as well), then the equation reduces to the fact that the voltage across each speaker would be:

$$V_Z = \frac{1}{2} V_S$$

Notice the complete absence of any frequency-dependent terms in this final equation: With two identical speakers in series, the voltage across each is simply half that of the voltage the amplifier is producing across the total, and is independent of frequency. There are, thus, no frequency-dependent variations in frequency as a result of putting two identical speakers in series. Q.E.D. ❖

REFERENCES

1. You often encounter hi-fi accessories, for example, that utilize "mass damping" to control resonances. That adding a mass will change a resonant system is hardly in dispute. That it "damps" a resonance is altogether a different and quite incorrect claim.

2. There are other loss mechanisms, most notably the acoustical losses. However, for direct radiator loudspeakers, these loss mechanisms are quite insignificant, most often representing less than 1% of the total losses. Not coincidentally, this number is not too dissimilar from the acoustical efficiency of such speakers as well, because in order to produce sound, real work must be done, and it is the work done into these acoustical "losses" that actually is the produced sound. Eliminate the acoustical loss, say, by taking away the radiation load by putting the speaker in a vacuum, and you've eliminated the sound. Not an entirely useful exercise for something like a loud "speaker."

3. While the case of non-identical drivers or systems is more complicated, the general principles apply, though there are confounding factors such as frequency-dependent attenuation resulting from different frequency-dependent impedances.

4. "Falsification" is a vital part of the scientific method. A theory must be falsifiable, that is, it must make a prediction that, by experiment or observation, can be clearly shown to be right or wrong. In the case here, either one theory, the other theory, or neither theory will be supported by the experimental data. No data can support both. If a theory makes a prediction that can't be tested, it's no good as a theory. You might have a theory: "I can levitate myself while no one is looking." It's impossible for anyone else to construct a test, because they can never look at you doing what you claim, thus the "theory" has no value scientifically.

TEST TRACKS

What are your favorite music selections you use to test audio gear or to show off some special capability of your system? Share your favorites with others. Simply describe your seven top pieces (not to exceed 1000 words); include the names of the music, composer, manufacturer, and manufacturer's number; and send to the address below. We will pay a modest stipend to readers whose submissions are chosen for publication.

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High g_m Smart Power Tubes, Part 1

This three-part article focuses on high transconductance vacuum tubes with low power handling capabilities (P_d less than 10W). The author presents hi-fi audio applications as well as cheap, easy-to-find Russian tube alternatives. **By Stefano Perugini**

High-transconductance vacuum tubes are strange animals, with electrical and mechanical challenges fully surpassed only at the end of the Western thermionic technology parabola. But what's their physical structure?

THEORY

In *Fig. 1* you can see the internal section of a triode with planar electrodes[1]. For this geometry, complex calculi show[2]:

$$\mu = \frac{2\pi nb}{\log\left(\frac{1}{2\pi nR}\right)} \tag{1}$$

$$J_p = 2.331\times10^{-6}\times\frac{1}{a^2}\times\left(E_g+\frac{E_p}{\mu}\right)^{\frac{3}{2}} \tag{2}$$

where:

μ = gain coefficient
n = grid wires/cm
a = cathode to grid distance (cm)
b = grid to plate distance (cm)
R = grid wire radius
E_g = grid voltage
E_p = plate voltage
J_p = plate current density

In a triode the transconductance, g_m, is given by:

$$g_m = \left(\frac{\delta I_p}{\delta E_g}\right)_{E_p = \text{const}}$$

Since $I_p = \Delta S J_p$ (where ΔS is the emitting surface area) from equation 2 you obtain:

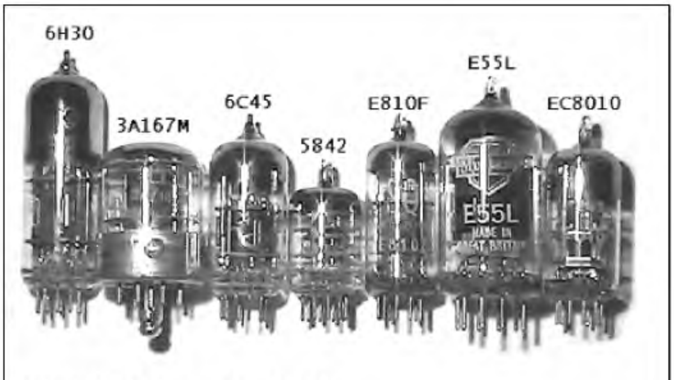

PHOTO 1: High σ tubes parade.

$$g_m = \frac{3}{2}\times\Delta S\times\frac{1}{a^2}\times\left(E_g+\frac{E_p}{\mu}\right)^{\frac{1}{2}} \tag{3}$$

The equations (1) and (3) are fundamental in the design of triodes (and particularly for the high g_m class).

Figure 2 displays the transconductance g_m and the μ factor versus the grid to cathode distance **a**. The graph refers to the Russian triode 6C45-ПE, but the trend is common for similar triodes (such as 6H30, 3A167M, and WE437A). You can observe that g_m increases when **a** decreases. Also, the μ factor slope is affected by **a** (since a+b is constant), but can be changed by the geometrical factors in equation (1) as n and/or R.

Figure 2 shows a strong dependence of g_m with respect to plate current (indirectly shown in the graph with E_p = cost and E_g = (−2...0)V), more marked in the low

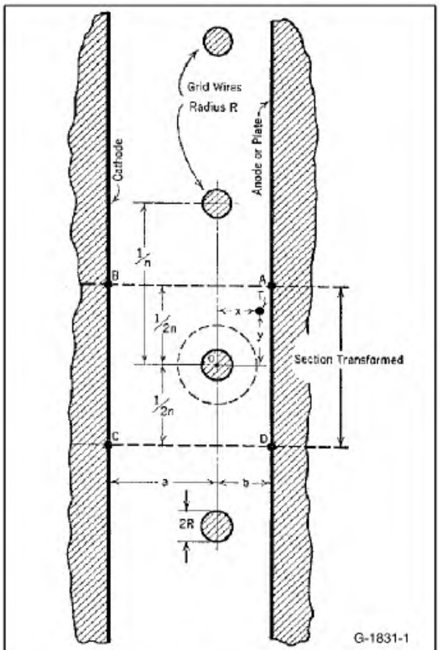

FIGURE 1: The section of a parallel plane triode.

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SUPREME 130 High Performance 28mm dome tweeter	8	680	220	91.5	$335.00pr
MDT10 28mm dome tweeter, Solin Range	8	1000	80	90	$23.15
MDT12 28mm dome tweeter w/ Neodymium magnet, 2.13" sq, 0.6" deep	8	1000	80	89	$23.15
MDT20 28mm dome tweeter	8	650	120	90	$25.05
MDT22 28mm dome tweeter w/ chamber and neodymium magnet	8	650	80	89	$25.05
MDT29 28mm dome tweeter	8	900	80	89	$35.50
MDT30S 28mm dome tweeter	8	700	200	90	$51.70
DMS30-S Shielded version of MDT30	8	700	200	90	$58.40
MDT32S 28mm dome tweeter with 110mm flange	8	700	200	90	$55.00
MDT33 28mm dome tweeter, double magnet, matched pairs available	8	700	200	92.5	$112.25
MDT37 28mm horn loaded dome tweeter	8	700	200	93	$49.00
MDT38 28mm dome tweeter, Top-mount	8	750	80	89	$39.65
MDT39 28mm dome tweeter, chambered	8	750	80	88	$39.65
MDT40 28mm dome tweeter, Neodymium magnet, Surface Mount tweeter.	8	750	120	89	$54.40
MDT41 28mm dome tweeter, Neodymium magnet, Top mount tweeter	8	750	120	90	$54.40
MDT43 28mm dome tweeter, Top mount, Double Neodymium magnet	8	750	120	92	$73.65
MDT44 28mm dome tweeter, Double Neodymium magnet	8	750	120	91	$73.65
R-29 Voice Coil for MDT29					$9.80
R-30 Voice Coil for MDT30S					$13.85
R-32/33 Voice Coil for MDT32S & MDT33					$14.85
R-37 Voice Coil for MDT37					$14.85
R-39 Voice Coil for MDT39					$9.80
R-40 Voice Coil for MDT40					$13.85
R-44 Voice Coil for MDT44					$14.85

Midranges

Model	Imp. Ω	Fs Hz	Qts	Vas liters	Nom. Power	Efficiency 2.83V/1m	Box Liters Sealed/Vented	F3 Hz	Price Each
MDM55 55mm dome midrange with 54mm VC and Neodymium magnet	8	380	-	-	200	90.5			$65.30
Classic Woofers									
MW113 4" Poly cone woofer	8	72	.75	4.3	150	87	10S	70	$63.90
MW114-S 4" Poly cone, Neo. magnet	4 or 8	69	.35	2.7	150	87	1.2V / 1S	87/135	$92.60
MW144 5" Poly cone woofer	4 or 8	45	.36	12.3	150	88	8V / 5S	54/88	$71.40
MW164 6" Doped Paper cone woofer	4 or 8	48	.55	14.3	150	86	20 S	62	$81.05
MW166 6" Poly cone woofer	4 or 8	48	.61	14.3	150	86	15 S	64	$81.05
MW168 6" Poly cone woofer, low Q	4 or 8	44	.41	16	150	88	14V / 8S	44 / 77	$81.05
MW265 8" Poly cone woofer, low Q	4 or 8	30	.44	88.6	150	90	75V / 56S	30 / 48	$87.80
MW266 8" Doped Paper cone woofer	4 or 8	29	.56	80	150	89	56- 00 S	38-3	$87.80
MW1075 10" Poly cone woofer	8	28	.44	155	200	90	90S	35	$108.90
Neolin Woofers - Neodymium Magnet System									
MW143 5" Poly cone woofer	8	47	.26	14	150	89	3V/2S	92/130	$124.50
MW167 6" Poly cone woofer	8	44	.35	19	150	88	10V/S	55/88	$133.45
MW267 8" Pole cone woofer	8	25	.33	113	180	89	52V/33S	34/53	$140.20
MW1077 10" Poly cone woofer	8	28	.44	155	200	90	190V/95S	25/45	$160.30
Hybrid Woofers - Neodymium & Ferrite Magnet System									
H5.1 5" Poly cone bass/midrange	8	43	.36	22	150	88	14V/9S	52/80	$69.40
H5.2 5" Paper cone bass/midrange	8	43	.36	22	150	88	14V/9S	52/80	$69.40
H6.1 6" Poly cone woofer	8	40	.32	30	150	91	14V/7S	55/95	$80.05
H8.1 8" Poly cone woofer	8	32	.29	65	180	90	29V/14S	45/75	$112.10
H10.1 10" Poly cone woofer	8	25	.35	191	200	90.5	95V/60S	33/51	$130.75

region of **a** values. Here you have a classic cost/benefit problem. The reduction of the distance **a** causes an increase of g_m (the benefit) and strongly submits this parameter to the caprices of plate current (the cost). The reduced dimensions (mechanically faced with a non-standard building such as a frame grid structure) are the main source of the electrical limits in this tube topology:

a) Microphonicity
b) Self-oscillating tendencies
c) High parasitics
d) Mismatching

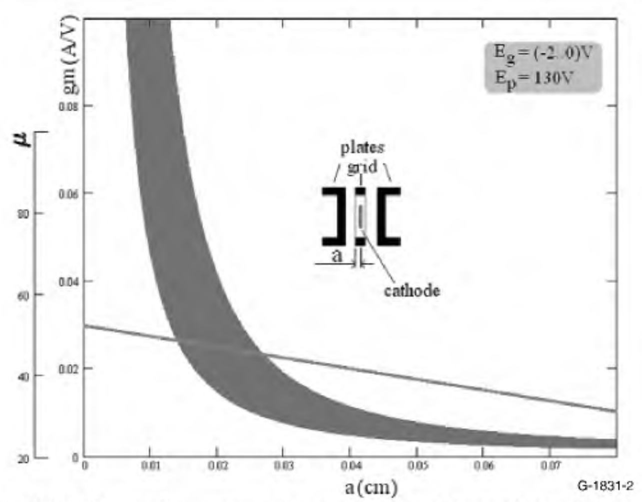

FIGURE 2: μ **and** g_m **versus distance a in a planar electrodes triode.**

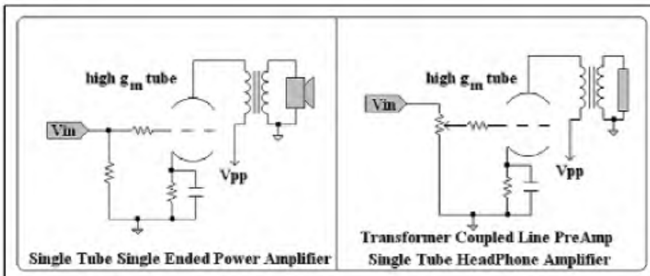

FIGURE 3: Transformer-coupled triode amplifiers.

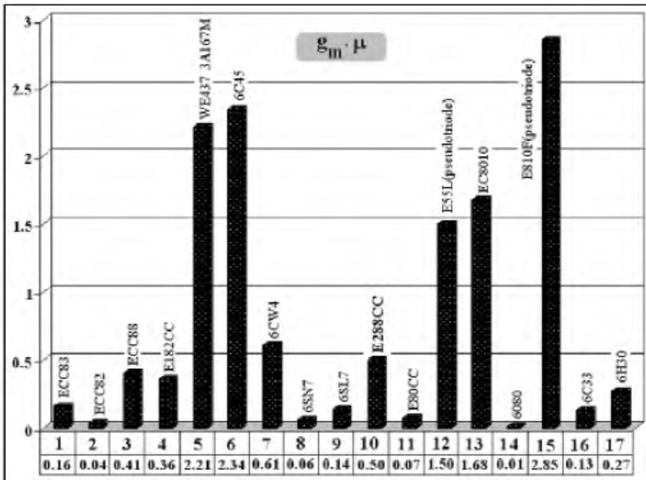

FIGURE 4: The σ **parameter in various triodes.**

The audio designer must manage these drawbacks properly for successful results.

APPLICATIONS

Figure 3 shows the basic schematics you can use as:

a) Transformer-coupled line amplifier
b) Single tube power amplifier
c) Headphone amplifier

For these applications the best results are obtained with triodes maximizing the parameter:

$$\sigma = g_m \times \mu. \tag{4}$$

A high value in the σ parameter (*Fig. 4*) reduces the input voltage swing and optimizes the transformer turns ratio for a

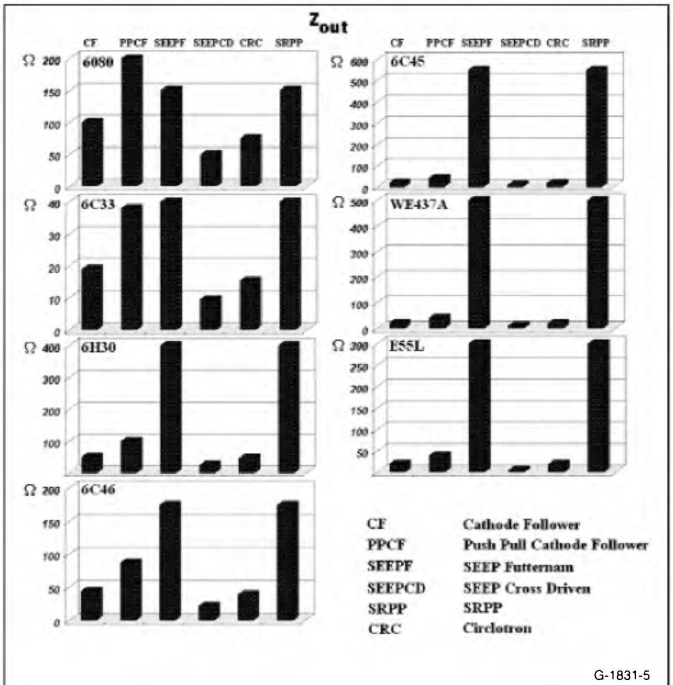

FIGURE 5: The output impedance Z_{out} **for standard OTL output stages in various triodes.**

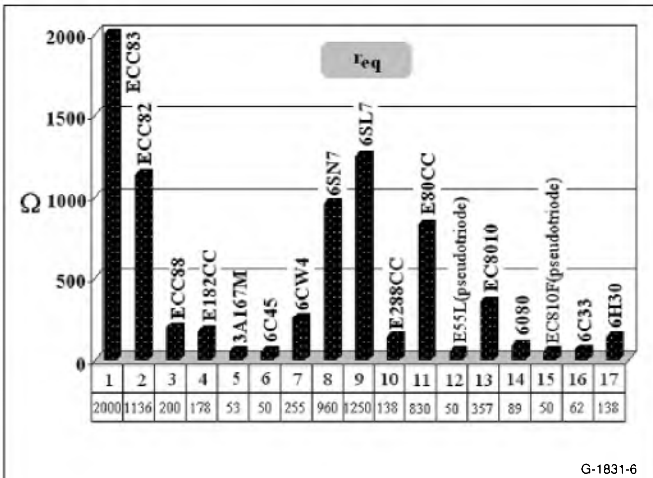

FIGURE 6: The equivalent noise resistance, r_{eq} **in various triodes.**

low output impedance with good current capabilities and high damping factors. High σ tubes (*Photo 1*) are also useful in the design of two-stage single-ended amplifiers—based on tubes such as 211, 845, 300B, and SV-572-XX—and give nice surprises in the design of low-power OTL amplifiers.

The essential requirement for an OTL amplifier is a low open-loop output impedance. The utilization of low r_p tubes appears as an irrefutable requirement, but you can also obtain excellent results with high σ tubes. In fact, the excellent performances in term of Z_{out} for the 6C33 triode (σ = 0.128) can be obtained and surpassed with other tubes such as 6C45-ΠE (σ = 2.34) and E55L (in pseudotriode mode, σ = 1.5). See *Fig. 5*.

Amplifiers designed around high g_m, high σ tubes present obvious advantages in term of noise, gain, and bandwidth. In fact, in a triode the equivalent noise resistance, r_{eq} (*Fig. 6*), is given by:

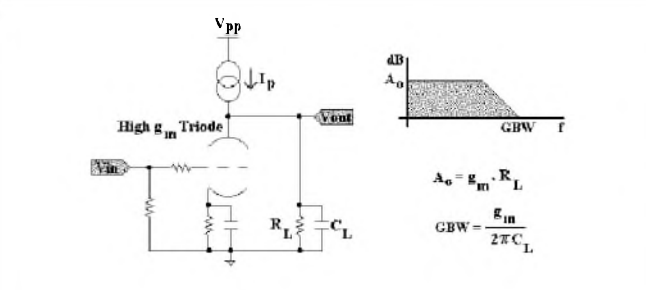

FIGURE 7: The high g_m common cathode amplifier. G-1831-7

$$r_{eq} = \frac{2.5}{g_m}$$

while for a common cathode amplifier (*Fig. 7*) you have:

$$A_o = g_m \times R_L$$

$$GBW = \frac{g_m}{2\pi C_L}$$

where:

A_o = gain
GBW = unity gain frequency

CONCLUSION

High g_m, high σ tubes present a wide spectrum of opportunity but also drawbacks derived by their intrinsic nature. The variability in the electrical parameters, as well as microphonicity and self-oscillating phenomena, must be evaluated with care by calculi, simulations, and prototyping.

Next month we'll take a closer look at some appropriate tube types. ❖

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Vacuum Tubes Born Again in Nanotube MEMS

Agere prototype integrates "triode" on standard silicon chip.

By R. Colin Johnson

Ever since the transistor was invented, engineers have lamented the slow, steady demise of the vacuum tube. But now researchers at Agere Systems Inc. (Murray Hill, N.J.) have demonstrated a prototype of a vacuum tube "triode" integrated onto a standard silicon chip. The device uses microelectromechanical systems (MEMS) to create anode, grid, and cathode components covered with carbon nanotubes that measures only 100 microns square.

"We believe that the first applications, which are many years away, will probably be to replace large, bulky solid-state basestations for microwave transmissions," said Wei Zhu, lead researcher on the project at Agere. "But eventually you could use these devices for any amplification need—even high-fidelity audio amps."

From an engineering viewpoint, vacuum tubes simplify circuit design because they are true amplifiers all by themselves. The basic concept is that a heated cathode "boils off" electrons into a vacuum; they pass through a screen-wire grid and finally are absorbed by the anode. By adjusting the voltage, the grid can modulate this current of free electrons passing through it, which transforms a small AC signal voltage to the grid into a larger AC voltage on the anode.

PROMISE OF EFFICIENCY

Transistors, though simpler, smaller, and cheaper to build, do not perform nearly as well in terms of precision, efficiency, or noise. "Even today, our transistor circuits are only about 5 percent efficient," Zhu said. "That's why microwave basestations have to be so big and bulky—they have to dissipate 95 percent of the energy you put into them as heat." In contrast, if Agere can convince the industry to start building MEMS-based triodes, tetrodes, and pentodes, engineers could downsize their solid-state equipment into smaller vacuum-tube-inspired designs that are 20 percent efficient.

According to Zhu's rough calculations, 100 MEMS-based triodes will eventually fit on a standard-size chip to handle 10W of amplification but will dissipate only one-fourth the heat of a 10W transistor amplifier.

"There is still a lot of engineering work that needs to be done to commercialize triodes again," Zhu said. "By demonstrating that . . . there are no big technological hurdles to overcome, [Agere] hopes to stimulate the industry to pitch in and help us do the necessary work."

Carbon nanotubes are the enabling technology. Without the large current densities nanotubes make possible, there would be no way to build an efficient field emitter to replace the vacuum tube's cathode. Other field-emitter-array technologies are being developed for display applications, but Agere claims its demonstration shows the technological superiority of switching to carbon-nanotube emitters.

To build a carbon-nanotube-based cathode, the team first laid down a silicon-nitride insulator atop a standard single-crystal silicon wafer. A three-layer polycrystalline silicon MEMS process was then used to etch out the cathode, grid, and anode electrodes from polysilicon panels with integrated "hinges" that let them stand vertically when the oxide underneath was etched away. After fabrication, the microtriode looks just like a tiny vacuum tube, right down to the screen-wire grid.

10 MICRONS A MINUTE

The carbon nanotubes were selectively grown atop the cathode by depositing a thin, nucleating catalyst layer of iron through a shadow mask. In the presence of a microwave plasma of ammonia/acetylene at 750°C, the 20- to 50-nm-diameter nanotubes grew in a tightly packed field at a rate of 10 microns per minute. The final 10-micron-high "patch" of carbon nanotubes on the cathode surface measured about 50 microns square.

While characterizing the device, the researchers verified that the microtriode performed in precisely the nice, mathematically simple manner of its bigger vacuum tube brothers. But the insulating layer was too thin, hampering performance, since operating voltages had to be kept below the level at which the insulator broke down. Thus, the characterized specifications (anode/grid current ratio of four, transconductance of 2.7 microsiemens, grid impedance 10MΩ) were not stellar.

"Our next step will be to improve the performance characteristics and to try out some other devices," Zhu said. Agere's next milestone will likely be to show a 1GHz version with higher grid impedance and better transconductance. At least five years of research will probably precede any development effort. ❖

consumeR PRODUCTS REVIEWED

S-5 Electronics K-12M

Reviewed by Charles Hansen and Duncan and Nancy MacArthur

S-5 Electronics, 1625 Twin Acres, Chandler, AZ 85249, 480-895-2521, FAX 480-895-4164. Kit price $140 or $170 assembled/tested; plus $10 shipping. Dimensions: 9.5″W × 4.5″H × 7″D. Warranty 90 days.

The S-5 Electronics K-12M is a push-pull stereo power amplifier rated for 8W per channel. The review unit came fully assembled, but I did receive the three-page kit assembly instructions along with the one-page operating instructions. The assembly instructions were brief but thorough, and presume that you have had prior kit-building experience and understand electronic components.

CONSTRUCTION

The amplifier doesn't come with an enclosure. A 7″ × 10.5″ pine board is furnished, along with a mounting/drilling template. The finished "breadboard" is shown in *Photo 1*. The PC board and transformers are mounted to the board with screws.

The front of the unit has two gold-plated phono jacks and a stereo volume control, both attached to the PC board. Two pairs of gold-plated speaker binding posts are located on the PC board just in front of the output transformers. These binding posts are spaced ¾″ apart, so you can use dual banana plugs.

The unit is furnished with a two-prong power cord, with a table-lamp-style in-line switch and an in-line fuse holder. One of the power-cord connections to the power transformer primary is made with a small wire nut. For safety's sake, I would pursue a metal enclosure for this amplifier. The tubes run very hot, and high voltages are present on the PC board.

The PC board is a very nice silkscreened double-sided glass epoxy board with a solder mask. It mounts to the pine board on four 6mm nylon spacers and the aforementioned screws. Ceramic tube sockets hold the four tubes.

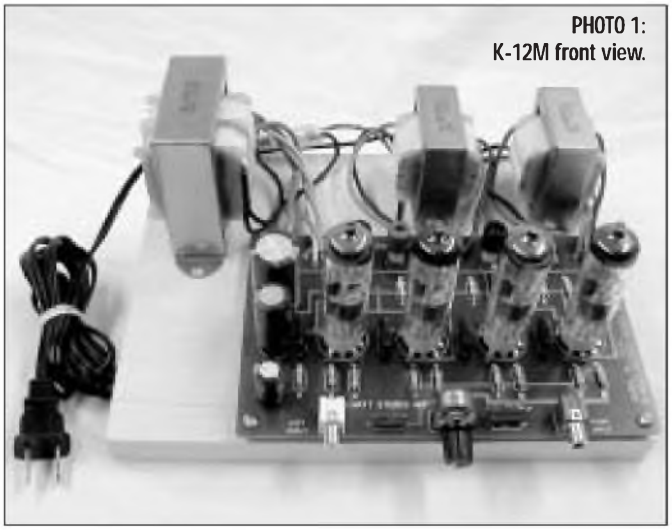
PHOTO 1: K-12M front view.

TUBE-POLOGY

A schematic was supplied with both the operating and the assembly instructions. Two 11MS8 triode-pentode vertical deflection amplifier tubes are used in each channel. The input signals are capacitively coupled to the 100k stereo pot.

The pot wiper connects to the grid of one of the triode sections used as a cascade amplifier. This triode drives the grid of the second triode, configured as a split-load phase inverter. This stage is capacitively coupled to the two push-pull pentode sections. Feedback is taken from the 8Ω secondary of the output transformer (the only tap available) to the cathode of the input triode.

All the coupling caps are metallized polyester, and Samsung aluminum caps are used to filter the full-wave solid-state power supply. Resistors are carbon film and metal oxide types. The volume control is a dual ⅛″ nondescript carbon pot. The three transformers are marked with a "UTK" logo.

S-5 Electronics supplied four 11MS8 "Made in Japan" tubes with this K-12M, packed as pairs in cardboard tubes. The 11.6V heaters are supplied from the "12V" AC winding of the power transformer. I measured an actual 11.5V AC at the heaters.

MEASUREMENTS

I operated the K-12M at 1W 1kHz into 8Ω for one hour to burn in the unit before measurements. The test data is summarized in *Table 1*. There was only a very low level of hum with my ear against the speaker, and no noise during power-up or shutdown.

Initially, the distortion (THD+N) in each channel measured 1.5%, but increased to 5% left and 2.9% right after the hour. I swapped tubes around to try to achieve a lower THD and settled on the "best" arrangement, which produced 1.6% left and 2.5% right after an additional hour of run-in.

The K-12M does not invert polarity. Input impedance at 1kHz was 81k left and 88k right, so this nominal 100k pot is on the low side. Volume control tracking was within 1dB from 10 o'clock to 3 o'clock, where it improved to less than 0.5dB up to full volume.

The full-volume gain at 2.83V RMS output into 4Ω and 8Ω loads was 25dB and 28dB, respectively. The output impedance was quite high: 6.6Ω at 1kHz and 5.5Ω at both 20Hz and 20kHz.

The frequency response for the K-12M was within −3dB from 14Hz to 58kHz, at an output of 1W at 1kHz into 8Ω. I measured −1dB at 20Hz and −2.8dB at 20kHz. This data, along with the response for 2W into 4Ω, and 8Ω paralleled with 2µF, is shown in *Fig. 1*. The dashed line in the figure represents the response to an IHF speaker load, which has an impedance peak at 50Hz. This amplifier will be extremely sensitive to any variations in speaker impedance with frequency, and will color the sound accordingly.

Hum and noise (maximum volume, input shorted) measured 0.6mV left and 0.2mV right. Viewing the noise content on an oscilloscope showed a 60Hz waveform with some noise, and obvious spikes near each power line zero-crossing, presumably due to rectification.

Crosstalk was noticeably higher L to R than R to L, as you can see in *Table 2*. The 100Hz levels are dominated by noise, while the fundamental test signal was predominant at 1kHz and higher due to capacitive coupling.

Figure 2 shows THD+N versus output power for various loads and frequencies. Notice the vertical THD scale, which is an order of magnitude higher than I usually require. I engaged the test set 80kHz low-pass filter to limit the out-of-band noise.

The right channel distortion (the two dashed curves) was significantly higher than the left channel. I was only able to obtain the rated 8W power at 1kHz with a 4Ω load, this at 28% distortion. The

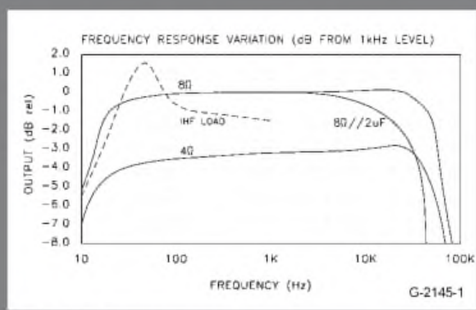

FIGURE 1: Frequency response.

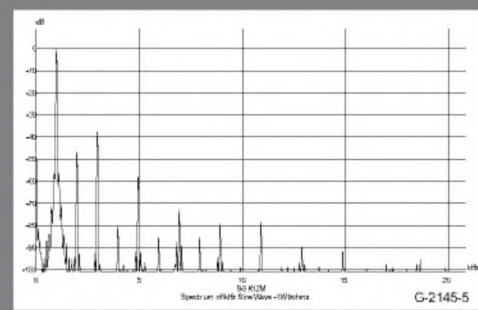

FIGURE 5: Spectrum of 1kHz sine wave.

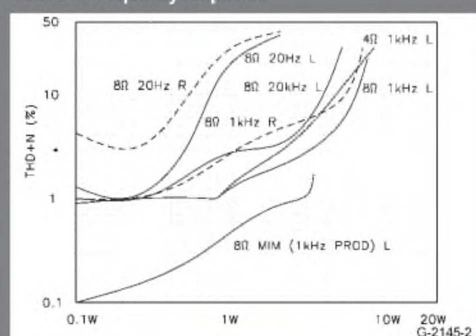

FIGURE 2: THD+N versus output power.

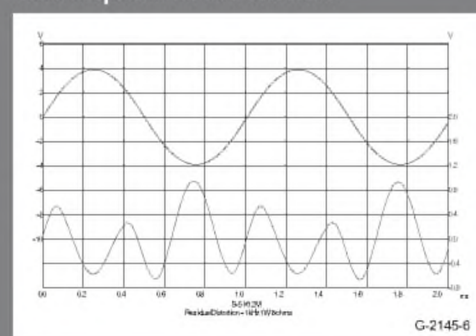

FIGURE 6: Residual distortion, 1kHz 1W 8Ω.

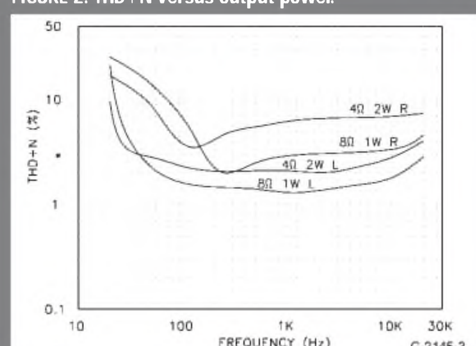

FIGURE 3: THD+N versus frequency.

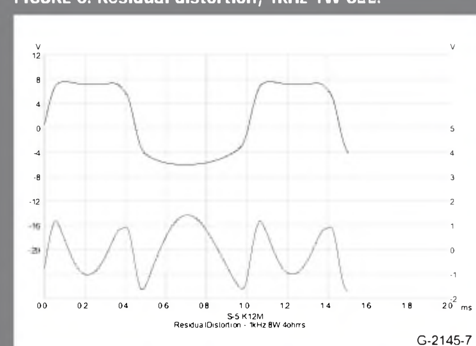

FIGURE 7: Residual distortion, 1kHz 8W 4Ω.

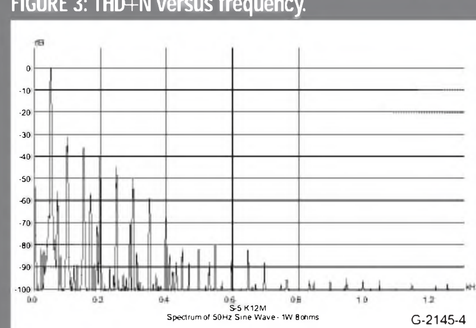

FIGURE 4: Spectrum of 50Hz sine wave.

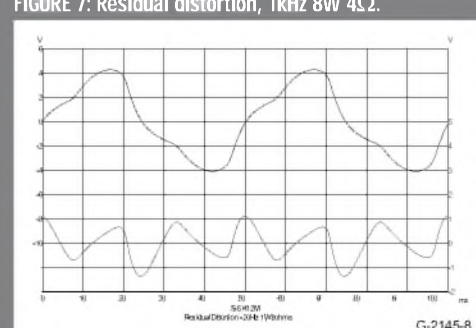

FIGURE 8: Residual distortion, 20Hz 1W 8Ω.

Reviewed by Duncan and Nancy MacArthur

What can you get for $140 nowadays? If you're willing to play by the rules as defined by this integrated amplifier, the answer is quite a bit. The K-12M produces a smooth, musical sound that rarely offends. When coupled with appropriate components, it is a joy to hear. We have not heard a better-sounding amplifier in this price range.

GUIDELINES

Unfortunately, during the listening process we discovered a significant problem with this amplifier's construction. After about 75 hours of operation at moderate levels, all four of the tube sockets were scorched and slightly melted. We discussed this problem with Larry Stafford at S-5. He mentioned that this socket problem is not limited to our sample and that S-5 is investigating ceramic sockets. We would not recommend the K-12M amplifier unless these sockets are replaced. Once the sockets are replaced, we'd suggest the following guidelines for enjoying the K-12M amplifier:

1. The K-12M needs efficient speakers. Eight watts are only 8W. When paired with 91dB/W speakers, the K-12M played loudly enough to make conversation impossible in a small room but not loudly enough to move objects or hurt eardrums. When we switched to Genesis 400 speakers with an efficiency (on a good day) of 89dB/W, the K-12M amplifier played beautifully at lower levels. At higher levels the highs became hard and the midrange congested. Playing the K-12M loudly through our less-efficient speakers made our ears suffer (and not just from the volume).
2. Don't expect ultimate bass extension and dynamics. Although the K-12M's midrange was excellent and its highs extended, its deep bass, while present, was notably weak. We didn't notice this lack on many types of music (vocals, strings, piano, and so on). But the slam was missing from the bass drums, and the fundamentals were gone from the low organ notes, on any music that included these instruments.
3. The K-12M is not a complete amplifier. A kit including electrical parts and PCB is available for $140, and an "assembled" version costs $170; however, neither version could be used in any but the most monastic environment without adding an enclosure, knobs, and connectors. This flaw isn't fatal in this price range, and the manufacturer prominently mentions the need for building an enclosure in the instructions. On the other hand, you should be aware that $140 (or $170) probably won't be the final price.

In fairness, the K-12M, as supplied, is a completely operational amplifier using only the parts included in the kit. As I write this, I am listening to a K-12M with the sole addition of an overturned milk crate to keep pets and children at bay. Although this combination sounds excellent playing Beethoven (Robert Silverman, pianist, *Ludwig Van Beethoven's Complete Piano Sonatas*, Stereophile KSP 830), its looks and long-term safety are not appropriate in a household with children, animals, or even other (non-fanatic) adults. The K-12M kit offers the opportunity to audition the sound before you spend more money on cosmetic and safety requirements.

Spousal note: Let me add another perspective on this issue: the K-12M as it stands is a circuit board and transformers on a raw pine board. Its appearance is unlikely to find favor with even an extremely tolerant spouse. To maintain domestic harmony, plan on building an enclosure. (S-5 has a good-looking example on their website at s5electronics.com.) Your loved ones will thank you.—NM

VOLUME MATTERS

Our sample of the K-12M arrived with a damaged volume control, which extends beyond both the PCB and the breadboard. According to S-5, this type of damage isn't uncommon. Replacing the pot isn't difficult given a little soldering skill, but in our case the pot had failed open—resulting in a full-power buzz through the offending channel. At this point we were glad that the K-12M only produced 8W per channel. Once we replaced the pot, we had no operational problems with the K-12M. (The melting tube sockets we mentioned earlier caused no operational problems, but left us wondering about the longevity of the amplifier.)

The volume pot on the K-12M appears to have a linear taper rather than the more common logarithmic taper. The linear taper makes the channel-to-channel tracking better but caused most of our listening to take place below "9 o'clock" on the pot. Using this type of pot probably represents a reasonable compromise in an inexpensive amplifier.

The lack of a "power on" indicator occasionally became confusing. Several times we resorted to resistance measurements to check the position of the (unmarked) AC switch before inserting the plug. Although the K-12M's speaker connectors are typical "5-way" terminals, they are mounted in such a way that banana plugs are the only easy way to connect your speaker cables.

DETAILED LISTENING RESULTS

We auditioned the K-12M using Genesis 400 and Audax A-652 speaker systems. Following our standard 50-hour break-in period (during which we observed little change), we did most of our serious listening with the Genesis speakers. We briefly compared the K-12M directly with our reference tube integrated amplifier, the Manley Stingray. This comparison is absurd, as the sales tax on the Manley probably would be more than the cost of the entire K-12M. Suffice it to say that the Stingray sounds better, as well it should.

We also compared the K-12M with our son's solid-state (of course) boom box as well as with an older solid-state Audio Source AMP-1. The comparison with the boom box was again extremely one-sided, but this time in favor of the K-12M. Although the boom box has a solid-state amplifier "rated" at 72Wpc as opposed to the K-12M's 8Wpc, the K-12M's bass response was tighter, deeper, and better defined.

The K-12M's midrange was clearer, better defined, and more dynamic. The high-frequency response of the K-12M was more extended. In short, the K-12M integrated amplifier represents a sonically valid step up from the amplifier included in the all-in-one systems

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maximum power available at 20Hz was a meager 2W, with 20kHz producing almost 5W, both again at distortion levels above 25%. At full power the supply transformer and tubes audibly resonated at the test signal frequency.

The clipping level is normally defined as that power where THD+N reaches 1%. However, the baseline distortion in tube amplifiers is fairly high at low power levels, so the generally accepted practice is to use 3% THD+N as the clipping point. Even with that allowance, this little amplifier was well into clipping at 2W or less, depending on the test condition. I never saw less than 1% THD+N for any test condition except at 0.1W into 8Ω 1kHz. Swapping the tubes around also affects the distortion, even for a left-right swap in the same channel.

The THD+N versus frequency for 2W into 4Ω and 1W into 8Ω is shown in *Fig. 3*. Here you can see the high levels of distortion, especially at low frequencies (LF). The right channel produces noticeably more distortion than the left, but neither channel quite gets down to the 1% maximum listed in the specifications.

The spectrum of a 50Hz sine wave at 1W into 8Ω is shown in *Fig. 4*, from zero to 1.3kHz. The THD+N measures 2.2%, and the harmonics are distributed throughout the spectrum. The second, third, fourth, and fifth measure –31dB, –36dB, –37dB, and –45dB, respectively. You can also see –57dB power-supply artifacts at 60Hz and 180Hz.

I also recorded a 1W 1kHz 8Ω spectrum in *Fig. 5* from zero to 20.8kHz, where fewer harmonics are present. This produces the lowest THD+N reading at 1.3%.

The left channel distortion waveform for 1W into 8Ω at 1kHz is shown in *Fig. 6*. The upper waveform is the amplifier output signal, and the lower waveform is the monitor output (after the THD test set notch filter), not to scale. The 1.3% distortion residual signal shows mainly the second and third harmonics, with no evidence of any noise or fuzz.

The clipping at maximum power (8W 1kHz 4Ω) shows a scooped-out positive half-cycle, and a noticeably asymmetrical negative half-cycle (*Fig. 7*). However, the residual distortion waveform is still predominantly second and third harmonic.

In *Fig. 8*, I show the left channel response to a 20Hz sine wave, amplified to 1W with an 8Ω load. You can readily see the distortion in the fundamental signal (top), with the residual distortion waveform shown below. THD+N measures a high 21%. Since the LF power is quite limited, I assume this LF distortion is due to output transformer saturation. None of the coupling capacitor –3dB points are higher than 10Hz, even allowing for the actual 81k input pot resistance and capacitor tolerances.

Figure 9 shows the K12M's output spectrum reproducing a combined 19kHz + 20kHz intermodulation distortion (IMD) signal at 12Vpp (about 2.2W) into 8Ω. The nonlinear tube transfer characteristics produce a wide range of intermodulation products. The 1kHz IMD product is a high 1.25%, with the 18kHz product about the same level.

REFERENCES

1. "Guitar Direct Box," Hansen, C., *Performer's Audio*, pp. 18–21, 44–45, 3/97.

The 12Vpp output test level for a multi-tone intermodulation signal (MIM, *Fig. 10*) of 9kHz + 10.05kHz + 20kHz produced a 1kHz response product of 0.89%. I also plotted the 1kHz product of this MIM signal versus output power in *Fig. 2*. This gives a better indication of the K12M's nonlinear response, since it is a closer approximation to music than a sine wave.

I applied square waves to the input of the K12M at three frequencies. The 2.5Vpp square wave at 40Hz into 8Ω showed signifi-

TABLE 1
MEASURED PERFORMANCE

PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Input impedance	100kΩ	81k left, 88k right
Output impedance	8Ω	6.6Ω at 1kHz; 5.5Ω at 20Hz, 20kHz
Power output, maximum	8W per channel	8W 1kHz 4Ω (28% THD)
Minimum input for full output	0.4V	0.427V RMS, 1kHz 8Ω 0.504V RMS, 1kHz 4Ω
Frequency response	20Hz–20kHz	14Hz–58kHz ±3dB 8Ω
Total harmonic distortion	<1% at 1W	1.3% 1W 1kHz 8Ω
IMD—CCIF (19 + 20kHz)	N/S	0.90% CCIF 6Vpp 8Ω
MIM (9 + 10.05 + 20kHz)		0.57% MIM 6Vpp 8Ω
Signal to noise ratio	N/S	–66dB
Noise	N/S	0.6mV maximum
Gain	N/S	21dB 4Ω, 25dB 8Ω

TABLE 2
CROSSTALK

FREQUENCY	CROSSTALK R–L	CROSSTALK L–R
100Hz	–60dB	–64dB
1kHz	–56dB	–51dB
10kHz	–44dB	–33dB
20kHz	–43dB	–27dB

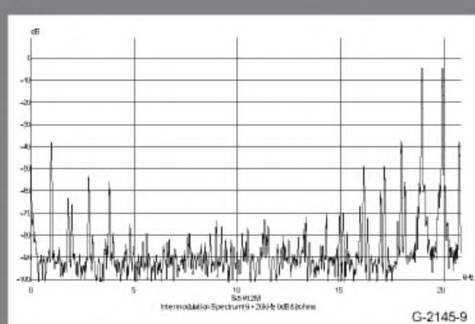

FIGURE 9: Spectrum of 19kHz + 20kHz intermodulation signal.

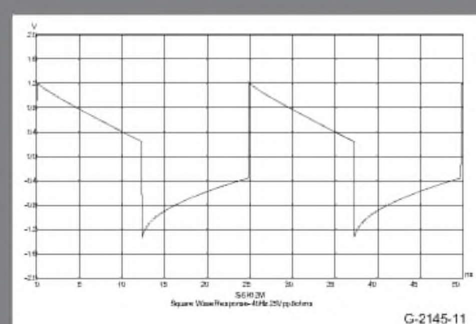

FIGURE 11: Square wave response, 40Hz 2.5Vpp 8Ω.

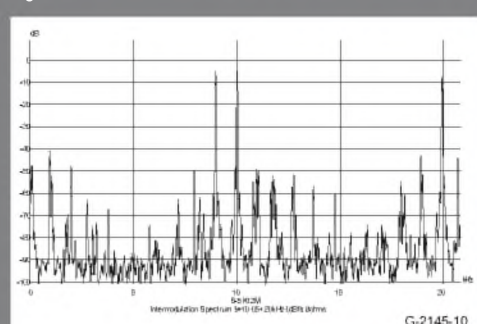

FIGURE 10: Spectrum of 9kHz + 10.05kHz + 20kHz intermodulation signal.

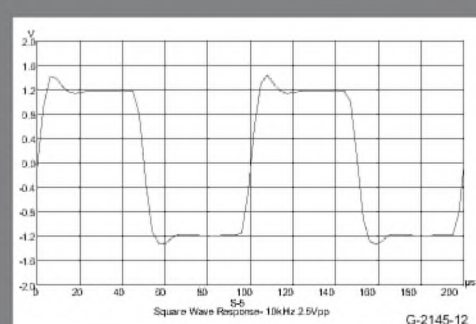

FIGURE 12: Square wave response, 10kHz 2.5Vpp 8Ω.

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typically owned by beginning listeners.

The comparison between the AMP-1 and the K-12M was much closer. The AMP-1 had a deeper and stronger bass response and an overall good sound, but when compared with the K-12M it sounded slightly harsh, unnatural, and "hi-fi." Although the AMP-1 was significantly more expensive, we both preferred the natural sound of the K-12M.

We auditioned the K-12M using a number of tracks from the *Hi-Fi News and Record Review Disk III* (track 2: Parry's "Jerusalem"; track 4: Vivaldi's trumpet concerto; tracks 5 and 6: excerpts from Prokofiev's "Peter and the Wolf"; track 7: Purcell's "Welcome, Welcome Glorious Morn"; track 10: a Corkhill percussion piece; and track 14: Rio Napo RSS demo). We also listened to a wide variety of other CDs, including a recent live album by BROTHER (This Way Up, Rhubarb Records RR07CD). See sidebar.

The K-12M produced an enjoyable sound with music ranging from Jethro Tull (*A Little Light Music*, Chrysalis, F2-21954) to Mozart (*Horn Concertos*, Dennis Brain, EMI CDH 7 61013 2). It was at its best reproducing instruments such as the horns and oboes in "Peter and the Wolf" and the trumpets in the Vivaldi trumpet concerto.

The high-frequency response of the winds in "Peter and the Wolf" was clean, sweet, and reasonably detailed, but the high frequencies of cymbals and miscellaneous noises on the "Rio Napo DSS Demonstration" were slightly recessed and lacking in sparkle. When we played the Purcell at reasonable volumes, the K-12M's sound was uncongested and very musical; however, this quality suffered as we increased the volume.

As you might expect, the K-12M amplifier lacked the extreme clarity and detail produced by some much more expensive amplifiers. Microdynamic contrasts were excellent; the Corkhill percussion pieces were crisp, and the individual drumbeats in the RSS demonstration were well defined, fast, and solid. Rhythmic recordings, including several songs from the BROTHER CD, generated an appropriate feeling of aliveness. This dynamic quality was lost at higher volumes.

At higher volumes some glare also became apparent on midrange instruments such as horns. The Vivaldi trumpet concerto made us cringe at high volume, although it sounded fine at our normal listening levels.

The K-12M was capable of reproducing a reasonable amount of ambience information. On the recording of "Jerusalem" the space around and above the choir was clearly audible. The K-12M produced a wide soundstage extending beyond the speakers on the "RSS Demonstration"; however, the instruments were not precisely placed within that soundstage but were slightly smeared. The drum kit on the BROTHER recording was appropriately centered, located above and to the rear of the other instruments.

TEST FOR TRANSIENTS, BASS, IMAGING, AND "SLAM"

BROTHER is a rock band with roots in Australia; they add bagpipes and digeridoo to the usual mix of guitars, vocals, and drums. Their live album, *This Way Up* (Rhubarb Records RR07CD), contains two tracks that test a music system's ability to reproduce bass transients realistically. BROTHER's recordings probably aren't available at your local CD store, but you can order them online at http://brothermusic.com/. (If you're wondering what a digeridoo is, this page has some good pictures.)

On a good system the drums on track 1, "Thetimeisnow," sounds as close to "live" as I've heard from a CD. The bass drum, although not especially deep, is fast, rhythmic, and insistent. If you are at all given to tapping your toes or stamping your feet in time with music, this track should get you going.

The drum sound on track 5, "The Unknown (Granny MacLeod & Rory MacLeod)," is deeper and less defined than on "Thetimeisnow" but just as insistent. This tune is another "foot-stomper"—with good woofers you should be able to feel each drumbeat. The digeridoo sounds as unique as it looks, and on this track it should be appropriately "twangy" with a deep bass fundamental. If it sounds like any normal instrument, the midrange resolution of your system and/or room needs work.

Both tracks have a very strong bagpipe lead. The bagpipe isn't a kind instrument to digital recording techniques. If you've turned these pieces up loudly enough to feel the drum and the bagpipe doesn't drive you out of the room, you've solved your digital sound problem.

On a good system the drum kit on both tracks is clearly located between the speakers (fairly easy), behind the speakers (harder), and significantly above the speakers (harder still). The drummer on this live album probably was sitting on a platform behind the other musicians. This spatial information is present on the CD and can be retrieved by a good reproduction system.—N + DM

FINAL THOUGHTS

NM: Tube amplifiers often receive good reviews for sound but poor reviews for measurements. For me a component's sound—particularly its sound when played with our system—takes precedence.

The K-12M's sound coupled with its modest price tag blew me away. I've heard amplifiers I liked better with our system but none for under $800. If you'd told me before I heard the K-12M that a $140 amplifier could sound this good, I wouldn't have believed you.

That being said, I have a few reservations. Don't buy this amplifier if you have inefficient speakers, if you must have deep bass, or if you're unwilling to build a box for it. The melted tube sockets (and to a lesser extent the easily-damaged volume control) still worry me. S-5 tells us they will try to replace the tube sockets. If they haven't done so before you buy this amplifier, consider either postponing the purchase or replacing the sockets yourself.

DM: The K-12M has many of the sonic characteristics attributed to classic tube designs. It possesses a wonderful midrange, a clean but slightly rolled off high end, and a somewhat muddy bass response. Although I have not compared the K-12M directly to any of the classic designs, to my ears its deficiencies are less pronounced than those of many earlier amplifiers. Its combination of midrange and high frequency response is ideally suited to taming the sound of an inexpensive CD player.

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cant tilt, as you can see in *Fig. 11*. This is indicative of the fairly high 14Hz –3dB point in this design.

The 10kHz square wave showed one half-cycle of peaking at the leading edge (*Fig. 12*). This peak was also present in the 1kHz square wave response (not shown). The leading edge of the 10kHz square wave showed a full cycle of ringing when I connected 2µF in parallel with the 8Ω load. However, there was no evidence of additional high-frequency peaking or instability, nor did I measure any in the frequency response test (*Fig. 1*).

After the measurements, I installed the K12M in my system for some brief listening. Then I had the idea to connect the left channel to my guitar amplifier speaker via my guitar direct box[1]. With its distortion performance, I found it to be a terrific little practice amplifier. The bass was reasonably full, and the high end had "bite" but without any treble harshness. The treble responded nicely to string dynamics, with easily controllable overload distortion at high volumes. Maybe the best enclosure for this little tube amp would be two reasonably efficient 8″ musical instrument drivers in a cabinet.

Manufacturer's response:

In keeping low cost and ease of construction as primary goals, the design was optimized for practical use with emphasis on the musical listening experience. NM's comment "The S-5's sound coupled with its modest price tag blew me away" attests to the design success.

We also comment that the tube sockets used have been completely satisfactory in the field with over 100 amplifiers in use by customers. We still plan to change to ceramic sockets, but this necessitates changes in the circuit board, which requires considerable lead time.

We also point out that the need for output power has been generally over-emphasized. Practical listening levels are of the order of 1W per channel. ❖

Larry Stafford
S-5 Electronics

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Along with the classic sonic signature, the K-12M integrated amplifier has managed to retain a price reminiscent of yesteryear. If its tube sockets were replaced, I would unhesitatingly recommend the inclusion of the K-12M in a starter system, assuming you've considered the three guidelines mentioned in the introduction. If you're even slightly competent with a soldering iron, I recommend buying the K-12M as a kit, not because the price is lower but because fitting an enclosure and replacing the sockets will be easier. Also, the volume control probably won't break if it's shipped as part of a kit. ❖

Product Review

Vidsonix VCB-100

Reviewed by Kent English and Nelson Pass

Vidsonix Design Works, 28415 Industry Drive, Unit 510, Valencia, CA 91355, 661-775-2760, Fax: 661-775-4967, www.vidsonix.com.

You see them in the catalogs:

"3 way crossover. 8Ω. 800Hz and 4500Hz @ 12dB/octave. 150W RMS. Made of only the highest quality components."

If you have a woofer, midrange, and tweeter already in mind whose manufacturer ratings resemble these specs, you could order this crossover for about $30, wire it up as instructed, and take a listen. Maybe you will be happy with the results, maybe not.

There is no doubt that these crossovers perform as advertised, and if loudspeakers behaved as neutrally as resistors, they would probably be close to perfect. Unfortunately, loudspeakers are not neutral. The input impedance, output amplitude, phase, and distortion will vary with frequency and power.

The result is that while a ready-made crossover will probably work, the chance that it will give you optimal performance is close to zero. There is no way to predict in advance which crossover components will give the best sound.

Most successful loudspeaker designers work with a mix of measurement and listening, and to them it is clear that both are absolutely essential. The product that represents the best mix of objective and subjective evaluation almost always results from much iteration and test. Usually the crossover components end up quite different from "textbook" values.

Designing a crossover involves a lot of clip leads, solder, time, and sweat. If you do this often, you start thinking about some kind of shortcut.

SOLUTIONS

At Pass Labs we designed an electronic crossover that allows rapid measuring and prototyping for active loudspeakers. Each of up to four poles for each filter is individually adjustable from 22Hz to 18kHz, resulting in millions of possibilities. Very nice if you can afford it and have the power amplifier required for each loudspeaker driver. For the average loudspeaker project this is too costly—both for the manufacturer and the do-it-yourselfer.

We've always thought the ideal solution would be some sort of switchbox that allowed standard components to be configured into first-, second-, and perhaps third-order systems for two- and three-way systems. A deluxe version would also allow substitution of nonstandard values and audio "candy" components in place of those already provided by the crossover switchbox.

You'd think such a device would be commonplace, yet it is not. Nelson relates how he used a big box of substitutable caps, coils, and resistors at ESS (remember the Heil transformer?) in the early '70s, but this was built in-house and was not very convenient to use.

Enter Charlie Miltenberger of Vidsonix. Charlie and his company not

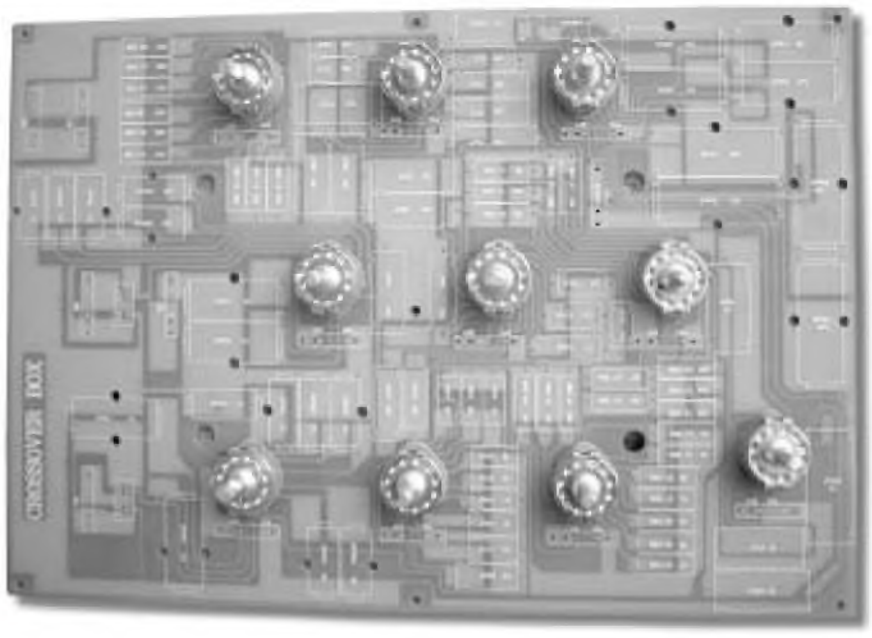

PHOTO 1: VCB-100 printed circuit board with rotary switches installed (courtesy of Vidsonix Corp.).

ABOUT THE AUTHORS

Kent English works at Pass Laboratories in charge of purchasing toys and creating trouble. Prior to this, he spent 12 years at the UC Crocker Nuclear Labs running the cyclotron, purchasing toys, and creating trouble.

Nelson Pass plays with the toys purchased by Mr. English and also creates trouble.

only thought about the practicality of such a tool, but they also produced a mature design and marketed it. I ran into Vidsonix's display at CES and thought, "What a cool product." When I saw the $299 price, I began to wonder how good it could be. Fortunately, Pass Labs has a big toy budget, so I called Charlie and ordered a crossover.

The service was excellent, and Charlie was available by phone or e-mail to answer any questions. True to audiophile tradition, on arrival I promptly unpacked the crossover, lost the instructions, and voided any warranty by disassembling the unit. Within an hour we ordered five more crossovers.

FEATURES

The crossover is very well made. The single circuit board is high-quality glass, and its conductive traces are minimum etch (*Photo 1*). Capacitors appear to be of high quality, inductors are of sufficiently low gauge, and the wirewound sand cast resistors are adequate—all typical of what you see in the majority of upscale commercial crossovers.

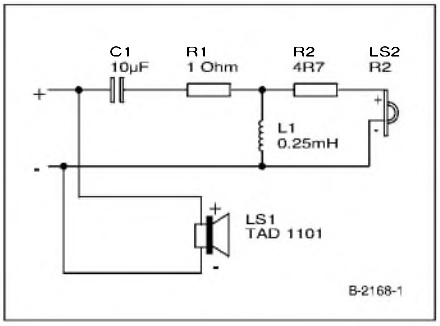

FIGURE 1: Filter configuration.

All the components are secured to the PC board with nylon zip-ties and what appears to be hot glue. The VCB-100 is not intended for permanent installation, nor is it intended to handle continuous power levels above 100W. Looking at the inside, we noted that "value engineering" did not materially affect the quality of parts or build quality (*Photo 2*).

A common concern with multiple inductors in a box is interaction between them. Vidsonix's solution is simple and effective: their inductors are contra wound on ferrite core bobbins to contain the magnetic fields within the inductor. As a result, there is negligible interaction between the coils, and the product is compact and portable.

The VCB-100 is easy to use. Vidsonix's website supplies a thorough and entertaining manual with 16 pages of text and diagrams describing the operation and theory of passive crossover network design in pdf format. They do not go into the mathematics of filter calculations, but provide reference to excellent resources. With the product they include a pocket slide rule that simplifies speaker-design-related calculations.

Input to the VCB-100 is through a pair of color-coded push-button spring terminals which will accept bare wire or pin connectors. Output occurs through up to three pairs of the same type of connectors. They will not accept spade lugs, alligator clips, or banana plugs, one of the very few flaws in the system.

The front panel is silkscreened with a block diagram showing the building blocks of first- and second-order filters for a two-way or three-way system (*Photo 3*). The woofer has an additional switched 47µF to 200µF capacitor in series making its filter a quasi bandpass. At each building block there is a silkscreened legend on the front panel around a 12-position rotary switch allowing selection of up to ten internal components, one external component, and none. The external position allows you to place components of your choice in the circuit. The out position opens for parallel components and shorts series components, bypassing the part.

Midrange and tweeters networks have series attenuation resistors of values up to 15Ω. The outputs feature both straight and reverse polarity by means of a switch, a very thoughtful addition.

We have used the VCB-100 a number of times in projects in the past few months, and it does not disappoint. Typically it pays for itself in one project. During this time we encountered no issues with reliability or sound quality of the parts.

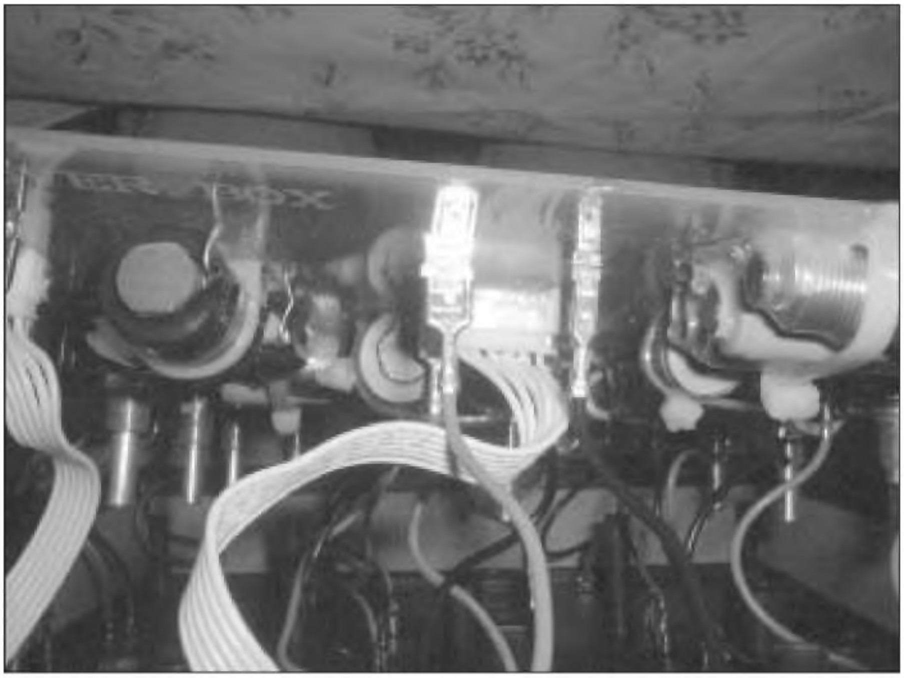

PHOTO 2: Close-up look inside the crossover box.

RECOMMENDATIONS

As with all crossover design work, ancillary equipment is highly recommended, though not strictly demanded. We use MLSSA and real-time analyzers, but anything with a microphone will be a big help.

Two or more VCB-100s may additionally be cascaded to produce networks of increased complexity, either with higher order networks or with four- or five-way systems. Using Vidsonix's crossover box reminds me of "free form" construction toys. Only your imagination and the number of crossovers at your disposal limit you.

The Vidsonix VCB-100 is not a speaker design panacea by itself. We recommend some sort of speaker measurement equipment to accompany and shorten the design process. Designing

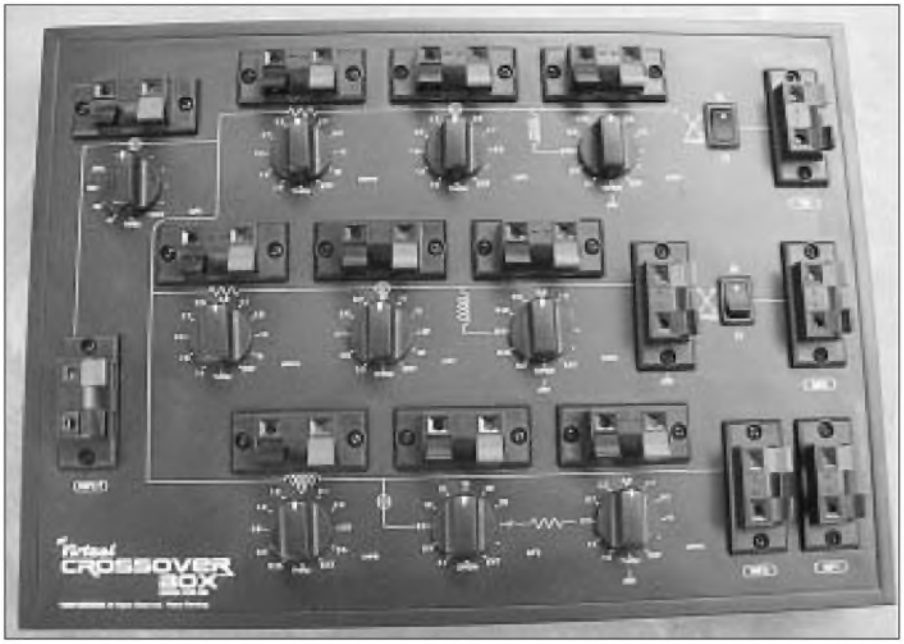

PHOTO 3: VCB-100 front panel (courtesy of Pass Labs).

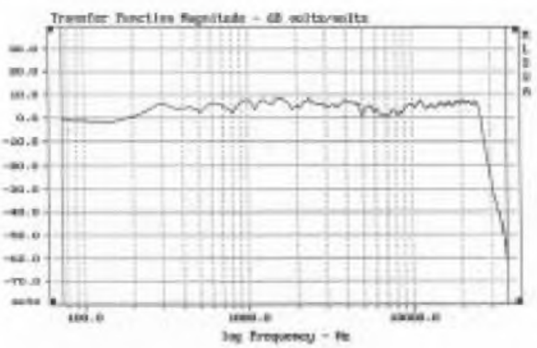

FIGURE 2: MLSSA response curve of TAD 1101H/Raven R2 taken at 1m, ACO Pacific microphone crossover configuration (courtesy Pass Labs).

B-2168-2

a good crossover filter still requires some knowledge, taste, and long hours of tuning and listening. This product can shorten the process dramatically.

We like the VCB-100 a lot, and recommend it highly. It is hard to believe that Vidsonix is making any money on it at this price, so we recommend that you buy one or two quickly before they come to their senses.

NOW FOR SOME FUN

Like most loudspeaker enthusiasts, you probably have a set of drivers put away for a rainy day. Our taste runs towards dynamic and efficient drivers, and in this case we had a pair of TAD (Technical Audio Devices) model 1101Hs, 11″ woofers rated 38–3000Hz with 97dB sensitivity and rated at 500W peaks. They have a particularly smooth character at the top end of their response, which is hard to come by in efficient wide range cones.

We mated them with the Raven R-2s from Orca Engineering. For those of you unfamiliar with the Raven, it is a 4″ ribbon tweeter; one of the best high-frequency drivers currently on the market.

We fit the TAD 1101H into a 3ft^3 reflex enclosure filled with Dacron, and auditioned it with a number of inductors/capacitors to limit its high-end extension. But in the final analysis we judged the natural rolloff of this particular driver the most natural, and so we used no crossover for it at all.

The Raven R-2 has been recommended for frequencies as low as 1800Hz, but with sixth-order slopes. We tried that and about 20 other possibilities, but in the course of a long afternoon decided that our preference was for a two-pole high-pass filter using a 10µF capacitor and .25mH inductor, with 5.7Ω resistance in series with the tweeter (*Fig. 1*).

We documented the system using Doug Rife's MLSSA system and an ACO condenser microphone. The result was a system that measured ±5dB from 70Hz through 20kHz (*Fig. 2*).

The sound was very impressive for such a marginal amount of effort. Part of the credit goes to the high quality of the drivers, but the ease with which the Vidsonix VCB-100 allows quick and reliable adjustment is not to be undervalued. Being able to make an instantaneous switch between crossover component values while you still remember what things sounded like before is extremely helpful. As indicated before, it paid for itself the first time we used it. Maybe you should order a few. ❖

SOURCES

DRA Laboratories
4587 Cherrybark Court
Sarasota, FL 34241-9213
941-927-2617
Fax: 941-925-0964
dra@gte.net
MLSSA

TAD—Technical Audio Devices
Pioneer Electronics Service Inc.
PO Box 1760
Long Beach, CA 90801-1760
800-872-4159
Model 1101H woofer

Orca Design and Manufacturing Corp.
1531 Lookout Drive
Agoura, CA 91301
818-707-1629
Fax: 818-991-3072
Raven R-2 tweeter

Product Review

Sounds Cylindrical

Reviewed by Gary Galo

Sounds Cylindrical–A Unique Collection of Music Recorded on Cylinder Between 1900 and 1929. *Old Colony Sound Laboratory, PO Box 876, Peterborough, NH 03458, 888-924-9465, FAX 603-924-9467, custserv@audioXpress.com, www.audioXpress.com; part # CDME1, $21.95.*

This collection is a slightly revised edition of a CD called *Pandora's Drums*, issued in 2000 by the British publication *Electronics World*. It is sure to interest sound-recording history buffs. The CD consists of cylinder recordings made by the Edison, Indestructible, Lambert, and Lioret companies between 1900 and 1929. The transfers were done by electronic means.

There is no technical information on the transfer process, but Joe Pengelly, producer of this CD and a well-known sound-recording historian, provided me with an article from the Oct. 2000 issue of *Electronics World* describing his all-electronic cylinder phonograph ("When Sound was Cylindrical"). I also visited the website of cylinder collector Christer Hamp, which has information on a surprisingly large assortment of modern electronic equipment for cylinder playback, including the Pengelly-Stringer Mk. 3 Phonograph used to make this CD (http://members01.chello.se/christer.hamp/phono/index.html; under "Axial Movement" click on "Joe Pengelly"). Most of the technical information in this review comes from these two sources.

The Pengelly-Stringer Mk. 3 Phonograph was designed and constructed by Mike Stringer, a faculty member in the Mechanical Engineering Department at Plymouth University in the UK, based on Pengelly's general concepts and specifications. The phonograph is a "moving mandrel" design (*Photo 1*, with thanks to Bill Klinger for supplying a high-resolution scan of Mr. Pengelly's photograph).

MACHINE OPERATION

This machine, taking a cue from Thomas Edison, includes a feedscrew mechanism that moves the entire motor and mandrel assembly under a stationary reproducer. Two motors are used in this design–one to turn the feedscrew and another to rotate the cylinder. Both motors are DC printed-armature designs with servo control.

Playback speed is continuously variable from 0- to 235-rpm. Most cylinders made after 1900 run at around 160-rpm, but many were recorded at around 120-rpm, and there were numerous other speeds in use. The mandrel speed can be varied between 2.5 and 91mm per minute, accommodating both two- and four-minute cylinders, plus any others likely to be encountered. The feedscrew and mandrel are both belt driven, since gears would have added pitch instability. The mandrel direction and cylinder rotation can be reversed, allowing the cylinder to be played backwards. This can sometimes give a cleaner playback on worn recordings.

The tonearm is a pivoted, free-stand-

PHOTO 1: The Pengelly-Stringer Mk. 3 Phonograph, used to transfer the cylinder recordings on *Sounds Cylindrical*. The straight, pivoted tonearm and moving-mandrel design ensure low tracking error and minimal record wear. (Courtesy of Joe Pengelly and Bill Klinger)

ing device fitted with a Shure M44C cartridge and an assortment of custom-ground playback styli of different sizes and shapes. The pivoted arm combined with the "moving-mandrel" design retains the near-zero tracking error advantage of the cylinder record.

But, the pivoted arm allows lateral movement of the reproducer, minimizing wear on the cylinder record. This is especially important if the rotational speed and the feedscrew speed are not properly matched. It is also helpful on cylinders that have become physically deformed, or have less-than-ideal physical properties due to manufacturing irregularities.

Pengelly's machine is equipped with a variety of mandrels to accommodate cylinders of various sizes, including large cylinders used as soundtrack records for the Edison Kinetophone film system (*Photo 2*). The wide range of playback and mandrel speeds, plus the variety of mandrel sizes, allows the Pengelly-Stringer Mk. 3 phonograph to play virtually any known cylinder record.

The phono preamplifier is solid-state, using Burr-Brown's excellent OPA604 FET op amps. The OPA604 can operate on supply rails up to ±22V, which helps prevent overloading by the clicks, pops, and other extraneous noises often found on cylinders. Sometimes, these noises are louder than the program material.

Acoustical recordings approximate a constant-velocity characteristic over their limited frequency range. A magnetic cartridge will yield a more-or-less flat response over this range, since magnetic pickups are velocity-sensitive. But, acoustical recordings played completely flat often sound thin and noisy. So, a mild equalization is built into the phono preamp—a bass turnover +3dB at 200Hz, and a treble rolloff −6dB at 6kHz. This gives a warmer, more musical result than completely flat playback.

Baxendall-type treble and bass controls and a Quad-style variable low-pass filter enhance the equalization capabilities. Finally, a 15-pole (!) Butterworth rumble filter, with a −3dB point of 160Hz, gets rid of nearly all rumble. Since cylinders rarely have any information below 200Hz, the radical rumble filtering should still leave the program material intact. The tone control and filter circuits are also based on OPA604 op amps.

EARLY RECORDINGS

Many of the recordings featured in this collection were made when the cylinder business was in decline. Emile Berliner developed the flat-disc record during the winter of 1887-88 (like cylinders, early disc-record speeds varied widely, but "78-rpm" became the standard descriptor for them). By the turn of the century, the longer playing time of the disc record, plus ease of duplication and storage, was rapidly making it the medium of choice.

When Columbia pulled the plug on the cylinder record in 1909, the format was already doomed. By the time the earliest Blue Amberol cylinders included on this CD were recorded—there are 14 examples made between 1912 and 1929—Edison was the sole American cylinder manufacturer among the major record companies. Edison closed his phonograph and recording business the day before the 1929 crash.

It is no secret among specialists in sound recording history that Edison had developed a recording process superior to most of his competition. Modern electrical playback can reveal just how remarkable some of his recordings are. In *Marriage Bells*, a 1913 Edison Blue Amberol recording, the clarity and transient response of the bells and xylophone is remarkable—this is, after all, an acoustical recording. Similarly, in a piccolo solo on another 1913 Blue Amberol—*On Her Pic-Pic-Piccolo*—the wide response of the Edison process captured the instrument's harmonic series with far greater accuracy than any other record label could achieve at that time.

Among the rare and unusual items on this disc is a 5½-minute "soundtrack" cylinder for the Edison Kinetophone film *The Old Minstrel*, recorded in 1913. The original cylinder is 7½" long, 4½" in diameter, and weighs 1 pound 3 ounces! At the beginning of the recording you can hear the crack of coconut shells, used to help establish an initial sync between the film and the recording.

Two Blue Amberol cylinders date from 1929, Edison's last year in the recording and phonograph business. As the booklet notes, these cylinders were dubbings of electrically recorded discs, and they don't sound quite like original acoustical recordings (the booklet says the dubbings were also electrical, but they sound as though the dubs were acoustic).

During his last years in the record business, Edison mastered cylinders on vertically cut Diamond Disc records. These master recordings were then copied—sometimes horn-to-horn—to a cylinder duplication master. Many Edison records from this period were is-

PHOTO 2: Some of the many types of mandrels used with the Pengelly-Stringer Mk. 3 are shown on the left. These will accommodate a wide variety of cylinder records, shown on the right. (Courtesy of Joe Pengelly)

sued three ways: as vertically cut, electrically recorded Diamond discs; laterally-cut, electrically-recorded "Needle-Cut" discs (as Edison called them); and dubbed cylinders.

I must take issue with Joe Pengelly's comments on the so-called countertenor Will Oakland. Pengelly asserts that "Oakland was a true countertenor—a tenor voice in overdrive—and not a falsetto male alto." Most accepted sources, including *The New Harvard Dictionary of Music* (or its predecessor, *The Harvard Dictionary of Music*), and *The New Grove Dictionary of Music and Musicians*, 2nd Edition, describe a countertenor as a male alto singing in head voice, or falsetto. Oakland mixes the head and chest registers, so he was not a true countertenor by the accepted definition.

VINTAGE CYLINDER

Nearly all of these recordings were celluloid cylinders, rather than wax. The transfers are well done, and the surfaces are quite low in noise for these types of recordings. The copies used appear to be in excellent condition. There is a small amount of pitch flutter on most of these recordings, which may be due to a slight elliptical warping that can take place if cylinders are stored horizontally for long periods of time.

There are no notes included with this CD, just a listing of the contents, identifying the recording dates and original catalog numbers. This is unfortunate, because Mr. Pengelly is an extraordinarily knowledgeable individual, and program notes authored by him would surely have enlightened the listener. Nonetheless, *Sounds Cylindrical* is an interesting and educational collection of vintage cylinder recordings, and is highly recommended to anyone interested in sound-recording history.

Readers interested in a more in-depth survey of cylinder recordings should check out the website www.tinfoil.com. Proprietor Glenn Sage offers 17 CDs that he has transferred using modern electronic playback equipment, including Brown Wax recordings from the late 1880s. Sage's playback system is also featured on the website mentioned earlier in this review. ❖

Xpress Mail

VIAGRA FOR WIMPY BASS

I thoroughly enjoyed Graham Maynard's article on "Sub-Bass Equalization" in the March '02 issue of *aX*. Integrating the sub-bass equalizer circuit with an active variable crossover would be great. Perhaps PC boards, kits, or assembled units could be marketed. Many hobbyists who have constructed Aria 5s or similar satellites would welcome a DIY project that could deliver relatively low distortion 20Hz bass from a $1ft^3$ enclosure. Plans for the sub cabinet, material sources, and so on, and hopefully testing by Mr. Joe D'Appolito, would make a most desirable project.

A project such as this may also benefit vendors who advertise products or services in *audioXpress*.

James M. Annal
Evergreen Park, Ill.

PASSIVE RIAA EQUALIZER

You talk about passive equalizers in your Dec. '01 issue ("Passive Phono Preamps," p. 8). On page 9 in Fig. 1 there are two RC networks that are configured for active equalization. Do you know how to configure these two networks (especially the one on the left) for passive equalization?

I have used a passive equalizer for years and now wish to remove the metalized polycarbonate caps and install some hermetically sealed Teflon and foil. They are very large and the present values of .1µF (100nF) and .3µF (300nF) will not accommodate Teflon. I have the values for the left side of Fig. 1 at 16k, 200k, .0047µF (4n7), and .015µF (15nF). These values will work fine with the larger Teflon caps.

Mark Huffman
Dayton, Ohio

Paul Stamler responds.

Neither of the networks of Fig. 1 will work in a passive circuit; they are suitable only for active (feedback) equalizers.

However, you can easily scale the network of Fig. 2 to your smaller capacitors. Here are scaled values, with easy-to-do parallel equivalents in parentheses.

Part	*Original value*	*Scaled value*
R1	*46.4k*	*145.4k (= 280k ‖ 301k)*
R2	*6.772k*	*21.22k (= 47.5k ‖ 38.3k)*
C1	*47nF*	*15nF*
C2	*16.1nF*	*5.14nF (= 4.7nF ‖ 430pF)*

The trade-off is that the higher impedances of the scaled network will be slightly noisier; however, I calculate that they will worsen the noise performance by only about half a dB—worst case, not enough to worry about.

Don't forget to reduce R1 by the output impedance of A1, if it's greater than about 1.4k.

HARMONY AND DISTORTION

Thank you very much for the two-part article entitled "Harmony and Distortion" (Feb. and March '02)—especially Part 2. Even if it has been more than 20 years, it is still very refreshing and kills some false ideas about distortion.

I have a little problem, however, when you explain the way we perceive pure sound when correctly distributed harmonics are provided. In your example, you take a 400Hz tone at 76dB SPL, and the second harmonic is at 61dB SPL. Because they are sound pressure levels, the relationship between the levels is the one with "3dB for doubling," or 10 log P1/P2.

So for harmonic 2, we are at 76 – 61 = 15dB less (or –15dB), which gives 3.16% of level content. For H3 we are at 1.58%, H4 is at 0.79%, and H5 is at 0.25%

In the same example, 30 years later, with some very slight changes, the same experience is conducted, but translated to voltage values, so we are in the "6dB for doubling" world, or 20 log V1/V2. So we obtain H2 = 7.9%, H3 = 5%, H4 = 3.2%, H5 = 1.4%, and H6 = 0.5%. These values are 100 times higher—and they are the right ones—than those you publish, simply because we count distortion in percent (%), so we must multiply the result by 100.

I'm sure that it is only a typo, but it is well worth noting that, in reality, a signal with a THD as high as around 10% (the vectorial sum of all harmonics) is perceived as a pure single tone, when harmonics are correctly distributed. So it explains very well why some amplifiers with "high" distortion can sound much better than others with 0.0000001% THD figures, depending on how they spread harmonics.

Thank you very much again for the article, and to *audioXpress* for publishing it 20 years later, showing that it is not the amount of distortion that counts the most.

Hervé Delétraz
Geneva, Switzerland

Jean Hiraga responds.

As stated in the article, the first experiments *by Wegel and Lane in 1930 used acoustic transducers, which did not have enough performance to provide an accurate result. Measurements made by H. Sakai of N.H.K. in 1980 are more accurate. However, loudspeakers used for these tests were also unable to produce a sound pure enough to give very precise results, even by reciprocity methods.*

Concerning sound level, I think that the use of decibel is not the proper unit for this kind of subjective appreciation. I suggest the use of sone, a unit of loudness for the individual listener (1 sone = subjective loudness of a 1kHz tone that is 40dB above the particular listener's threshold of hearing).

Besides that matter, music content is essentially made of transients. A sound and its harmonics levels are changing constantly with time, and we don't have enough fast analyzers to compare, in succession, very short samples of the real level of instantaneous harmonic distortion.

Several tests on phono cartridges and loudspeakers made in Japan around 1980 showed that harmonic distortion is also subject to frequency instability, a frequency jitter that is important enough to be mentioned. For example, an LP frequency test record showed that some phono cartridges had frequency instability as high as ±3Hz compared to a groove recorded at 1kHz. Note that loudspeakers are producing much higher frequency jitter effect, and that can more dramatically influence the feeling of sound purity.

Standard CDs at 44.1kHz/16 bits are also generating problems due to the fact that square and triangular waves take a shape close to sine waves for frequencies above 6.3kHz.

You need to consider these details if you think that the F on a piano is as precise as 349.23Hz.

TRAMPLED BY ELEPHANTS

Thank you for many years of the excellent publication *Speaker Builder.* I was pleasantly shocked one day in 1983 or so to pick up one of those magazine promotions and learn there actually was a magazine written for me and other speaker nuts. I have subscribed off and on since then. My objective from the beginning was better sound for less money, and a magazine with tried and tested systems could save much agony.

With a mechanical engineering degree, a number of electrical engineering courses on circuit design, 35 years of speaker building, and having built a scratch "kit" amplifier (parts and schematic: had to build the chassis, and so on), I didn't feel like a novice when upgrading my primary listening speakers a couple of years ago. I was armed with many issues of *SB*, catalogs from Goldsound, Zalytron, Madisound, and a box program from Bullock and White; this was going to be fun and easy, like shooting fish in a barrel. (Not!)

I thought the 2sat/2sub system I built (Swan 305, Seas P17RCY, Vifa D26tg) sounded pretty good to my "experienced ears," until I heard friends' tiny store-bought speakers, which sounded better. As best I could tell (no testing), my mid/highs were too forward and there wasn't enough bass or midbass.

"Must be the crossovers." (No testing for me.) So then I splurged and bought Bassbox6 (BB) with Xover3, for $200, which sounded as though it would not only design the perfect box and crossover, but would almost build them for you! You could select the perfect drivers right out of the database. Magic!

I used Xover3 to develop and try thousands of combos on-screen and then wired/tried all kinds of crossover topologies . . . but *the speakers still did not sound as good as cheaper, smaller commercial units.* Bassbox and Xover told me I had the perfect flat line response . . . but I didn't. By now I was becoming a little annoyed.

"Must need higher quality drivers." (Still too cheap to buy testing equipment.) The economy has been good, so I "deserve Focal or Audax upgrades!" After again much on-screen design and poring over the catalogs, I concluded with Elliot Zaliet's (Zalytron) input that the Audax HM 130C0 very light carbon cone woofer might even be better than the small Focal Kevlar, and industry standard Focal TD90tdx should be hard to beat. I replaced the two Swan 305s with a single Goldsound 12″ dual 8/8 sub to save my marriage. My new sweetheart wife had been complaining about "kitty coffins" in the den and living room.

The new Goldsound sub/Audax/Focal setup is very bright and detailed, but too much so. Again, Bassbox and Xover show that I have the perfect speaker . . . it just doesn't sound like it. A 12″ sub in 3ft ought to be belting it out, but it sounded thin.

I gave up in exasperation, purchasing

the $35 analog Radio Shack SPL meter and placing it at the listening position, on the wall opposite the bookshelf-mounted sats and sub built into a custom cabinet behind cloth-covered doors.

ARRGGGHHH. I can barely admit how nonlinear (±8–10dB) the response from my Xover and Bassbox "perfect" speaker was. The bass was relatively weak, the midbass really nonexistent. To get any midbass, I had to run the satellite without lower-end cutoff. I ordered the 4Ω/4Ω version of Goldsound Sub. Big improvment (+3dB) by pulling double power out at 4Ω. Minor lesson, but midbass thin. I am starting to get really perturbed.

By this time, I was waist-deep in building a home theater (HT) system, having moved the Swan sub, Seas+ now 1″ titanium dome (Parts Express) sats from the den to the theater room. I had already ordered "upgrade" drivers for those satellites: Audax HM170G0 (treated paper), whose frequency response was nearly identical to their well-respected very light and stiff carbon unit HM170C0 with flat response up to 6K (more on this later) and Focal TD90tdx. The initial test of the perfect Xover/Bassbox system was very, very bright, almost piercing. Many iterations did not improve it.

By this time, I was testing both systems with the Radio Shack meter at the listening point on a photo tripod. I still won't–will never–haul everything outside or rent an anechoic room. Besides, isn't it time to start recognizing that the sound we are looking for is at the listening position, not outdoors or in some guy's test room?

For 18 months, I flogged Bassbox6/Xover3, wired and tested "perfect straightline" designs in every imaginable combination at many cross points of 1st, 2nd, and 3rd order and mixed topologies. My wife wonders what I am doing in the basement all the time. No significant improvement: a hole in midbass from 150–400Hz and a huge mid-high spike rising +16dB from 3k to 6k!

Technical support at BB/XO couldn't have been nicer, checking out my designs and making numerous suggestions. But they pointed out that "room acoustics are important" and "drivers may vary from manufacturers specs. You probably need to test." The first hint of the elephants. HMMM.

I disconnected and tested only the midbass in the box with no crossover, at 1m with both speaker and meter on stands in the center of the room. Bass rolloff was well above 100Hz, not the f_3 point of 60 predicted by BB. From 200Hz, output climbs +3dB to 800Hz, then levels off to 3k. Then from 3k to 6k climbs +16dB! (Manufacturers specs show nearly flat from 800Hz to 6k; Bassbox modeled bass response as flat to below 80Hz.)

Elephant #1: Manufacturers specs way off. More than 16dB spike in response, nearly unsolvable without massive corrective electronic "surgery." And a clue that there may be another elephant.

After much conversation with BB6/XO3 tech support about how to compensate for my bad driver, they referred me to the LDSG website for a replacement driver. That site happened to feature a short article describing "Baffle Diffraction Step."

Elephant #2: Baffle Diffraction Step (BDS) not disclosed or compensated for in design software. Up to a 6dB efficiency mismatch from a "90dB" midbass and 90dB tweeter caused by the transition from 2π (half hemisphere) radiation for wavelengths smaller than the baffle dimension to 4π (full hemisphere) radiation of bass frequencies which wrap around the baffle and do not propagate forward unless reflected. It affects the region from 1kHz down to 300–500Hz for speakers not placed directly on a wall (2π) or the floor.

BDS is apparently the real reason for the widespread use of the (D'Appolito) MTM configuration among kit-makers: You need two midbass drivers to make up for the 6dB loss in the midbass region when placing a speaker in the now preferred location—on a stand at ear level, usually somewhat away from floors and walls. (There are a number of websites with an explanation of BDS. Just search "baffle diffraction step" on Yahoo and pick one.)

The pros and the kit-makers already knew this. Zalytron, Inc. (Elliot Zaliet, with input from Joe D'Appolito and other very high powered talent) offer many kits, many with MTM, whose crossovers all "mysteriously" had a 6dB shelf in bass/midbass when reproduced on Xover Pro/Bassbox6. I now know why. Same for an HT satellite design from Madisound.

Elephant #3: "Some User Testing Recommended." Actual speakers varying widely from manufacturer specs and up to an uncompensated 6dB Baffle Diffraction Step don't give the DIYer much chance for obtaining listenable results without investing in and using expensive, sophisticated test equipment. This, in addition to expensive box and crossover programs. However, a cheap $35 meter from Radio Shack and $10 CD (20–20k in about 100 critical steps of warble tones, Stryke.com Stryke Audio, Basszone) did dramatically improve my efforts.

Conclusions

It took me several very exasperating years to find the three elephants in my amateur speaker building: (1) supposed high quality manufacturer's speakers not even close to specs (2) "Baffle Diffraction Step" caused as much as 6dB error for a typical speaker location, which is not reflected in commonly available design software and (3)

"Some User Testing Recommended" could locate or fix the problem, and I should probably have used something more sophisticated than I did.

These have been some of the most aggravating and frustrating hours of my life. I had spent many hundred valuable hours before realizing that the elephants were preventing my success.

This prognosis isn't great news for speaker building. The explosion of kits is an obvious response, and a good choice for anyone short of a fanatic prepared to invest the time and effort I suggest.

However, you have always been forthcoming in the best interest of the hobby, and if we all recognize the elephants, maybe my wasted time can prevent others from having the same experience and will encourage the manufacturers and software people to do the right thing. The technology, which is not terribly expensive, is available. I believe that the manufacturer/OEM connection is preventing the solution: the OEMers are the big market, earning a nice markup for the voodoo of solving this problem.

Thanks for 18 good years.

Paul E. Davis
Atlanta, Ga.

TO BYPASS OR NOT

I read your article, "Big Mike and the Jimmy, Part 2," in the March '02 issue (p. 22). What caught my eye, and worried me a little, is your statement that audiophile designers' common practice of bypassing large electrolytic capacitors with polypropylene film caps produced sharp resonances due to the inductance of the electrolytic when combined with the capacitance of the bypass cap. You do have a way around this problem in your power supply; that is, the 100µF 450V Panasonic electrolytic has a low enough inductance and adequate damping to work for you.

Could you explain this a little further? What will this sharp resonance do to the quality of sound? Would it be better for some of us who are not sure about this–the level of inductance and damping in a particular electrolytic in a power supply, and so on–to just *snip out* our bypass? I may be okay because I use "black gate electrolytics and polystyrene bypasses."

Your thoughts expanding the implications of this beyond your article would be appreciated by many of your readers, I'm sure!

William R. Harren
Ridgefield, Wash.

Paul J. Stamler responds:

No, I don't suggest that you snip out the bypasses on your electrolytics! But neither do I suggest that they be used automatically, without critical analysis.

I can't tell you what the audible effect of sharp resonances in the power supply will be, because in my experience supplies and circuits differ hugely in their sensitivity to such problems. A sharp resonance could cause anything from no effect whatsoever to—in an extreme case—oscillations at the resonant frequency. (I've had it happen.)

If you don't have test equipment capable of measuring power-supply impedance, I suggest you try listening carefully to a single piece of gear, with and without bypasses. If you hear an improvement with the bypasses in place, leave them in. If not, don't.

However, if you have a good audio signal generator and a sensitive audio voltmeter with wide frequency response (a DMM will not do), you can measure the impedance of an electrolytic and determine the frequency of its impedance minimum (the point where the capacitance and inductance neutralize one another, leaving only the equivalent series resistance—ESR).

From this, you can determine the capacitor's inductance. You can then plug a modeled capacitor (resistance, capacitance, and inductance) into a program such as PC-ECAP, bypass it with a pure capacitance, and look at the impedance. Often you'll find a sharp, nasty resonance someplace not far above the audio range. (The little polypropylenes often used for bypass purposes have very low inductances and ESRs; they behave very much like pure capacitance at the frequencies we're discussing, or at least close enough for a first approximation.)

An appropriate answer can be a Zobel network, which can flatten the Electrolytic's impedance curve by neutralizing the inductance, allowing you to then roll it off with a small poly. However, note that the natural candidates for constructing Zobel networks—largish tubular polypropylenes—themselves have substantial inductance, sometimes requiring an even bigger capacitor for a second Zobel, which requires a third...

A better solution, or at least a more practi-

cal one, is to use capacitors with low enough inductance and high enough series resistance that a Zobel isn't necessary, such as the 100μF 450V Panasonics I used in this power supply. In general, smaller-diameter caps have lower inductances, and within a given line, lower-capacitance units are way less inductive than higher-capacitance. (Some capacitors, such as stacked-film devices, are optimized for low impedance; in practice, though, they're not usually available in higher voltages, limiting their usefulness for tube aficionados.)

Incidentally, there's no guarantee that the Black Gate electrolytics are appropriate, although they may be excellent in other respects. Black Gates, if I'm correct, boast unusually low ESRs, not necessarily best in this application. Likewise, tubular polystyrenes are more inductive than some of their flat polypropylene cousins (the ones that resemble Chiclets), and so may not be ideal for producing a power supply whose impedance is flat across a broad band.

My suggestion, therefore, is to experiment, sonically or in the lab (preferably, of course, you'll do both). I am conservative enough in design to want to minimize sharp electrical resonances in or near audio circuits whenever possible, and I suspect that some of the erratic reports I've heard from experimenters who have bypassed electrolytics reflect problems with such resonances.

This answer is, of course, only a cursory glance at a topic that's more complex than it seems. Perhaps an article is gestating, if Ed agrees . . .

THANKS

I appreciate the articles for building the THOR TL speaker and the moving-coil transformers. Keep up the good work!

Carlos E. Bauzá
Bauzace@netscape.net

CONVERSION CIRCUIT

Michael Kornacker describes his conversion of an Eico stereo power amplifier from one using a split-load push-pull driver to a "self-balancing phase inverter" driver (Audio Aid, March '02, p. 66). He is correct in thinking that the split-load circuit has differing output resistances from plate and cathode, although this difference is quite modest if a medium-mu tube is used. There is also, as he states, a difference in frequency response from these electrodes, although the difference is only significant at or above the upper limits of audibility.

In any case, these problems (if such they be) do not disappear through use of the self-balancing inverter. In fact, the output impedance differences between the non-inverting and inverting plate signals in the self-balancing configuration are more significant than in the split-load version. The reason is that the two triodes operate entirely differently. The triode receiving the input signal (V1B in Mr. Kornacker's circuit) directly drives the lower output tube and is an ordinary voltage amplifier. It also drives the other triode (V1A) via C3/R9, which in turn drives the upper output tube with an inverted signal.

However, the inverting tube is not an ordinary amplifier because negative feedback from plate to grid via C2/R8 reduces its gain to close to unity. This feedback makes the output resistance from V1A much less than from V1B. Also, it makes the bandwidth of V1A greater than that of V1B and its distortion less, which is desirable because it passes the signal from V1B without much further alteration except to invert it.

These considerations will be somewhat, but not essentially, modified by the author's unfortunate use of positive feedback in the circuit. If he had used a 12AX7 without this feedback, the gain would have been a little greater than in the original Eico circuit and there should have been no instability or unusual noise assuming a good tube.

Adjustment of the plate loads to balance the outputs are a better way to go.

Personally, I doubt whether there is much to choose from between these configurations. The split-load inverter will have better inherent output signal balance, but may be more prone to hum from higher heater-cathode potentials. In any case, audio beauty lies in the ear of the listener, who is the only one who counts.

Lawrence Wallcave
Santa Rosa, Calif.

Michael Kornacker responds:

Thank you for your comments to my article entitled "Self-Balancing Phase Inverter." I appreciate feedback—positive and negative—because I usually learn something new from what other more knowledgeable people have to say. I must admit that I am not an expert in tube circuits, but I like to experiment.

My reason for changing the split-load inverter to another type was based on criticisms of that circuit I had heard or read in various audio magazines, this one included. These criticisms were probably based more on people's opinions than on scientific evidence. Regardless, whether the criticisms were justified or not, I knew there were other types of inverter circuits out there and I wanted to try one. Not so much for the reason that I could really improve the sonics to a noticeable difference, but just simply to experiment with a simple circuit. In the end, I was amazed both that I could do that with a simple reconfiguration and with the results I obtained.

As for your comments, you sound very knowledgeable and your words are the first definitive words I think I've heard on this subject. Maybe they can dispel some bad rumors and shed some good light on the venerable split-load inverter circuit. I know I just learned something new. However, I think I shall leave the self-balancing circuit in place, but make some changes. In one change, I will eliminate the positive feedback resistor. I do agree it is probably not a good idea here.

When I first played some music at a high volume, every once in a while I heard a quick and sharp squeal. The first thing I suspected was the positive feedback generating an oscillation. I reduced the positive feedback in the circuit by increasing the resistor, and I think that cured it.

However, in the future, I will just eliminate the resistor completely and adjust the plate resistor for the correct amplitude level as you suggest, just to be on the safe side. I also want to experiment with all of the resistor values of the second tube. I'm really still not done playing with that circuit. I also will try the 12AX7, as you suggest, or one of its cousins to see who performs the best.

Another mod I want to try is from Rickard Berglund (GA 4/96, p. 64). He suggests replacing the 6BQ5 output tubes that the amplifier originally has with 6CW5s, which supposedly have less distortion. I'll have to construct a regulated screen voltage for them as recommended. It all should be fun.

Thanks again, Mr. Wallcave, for your input. I appreciate the education.

TUBE AMPS

Thank you for presenting the recent article by Mr. Modafferi regarding his Class-B design amplifier (April '02, p. 6).

Mr. Modafferi is one of those names that recurs over and over in the realm of audio design, and I have no doubt that these amplifiers function as he describes.

However, I don't think these designs quite reflect the motivation behind the continuing popular use of tube amplifiers in serious high-fidelity service. If low maintenance and high reliability were the primary objectives, we would not be building vacuum-tube amplifiers unless vacuum tubes were the only—or at least cheapest and easiest to acquire—devices (as was once true), or unless we were operating in an environment where extreme ESD, or EMP, induced failures were common. If I were forced to operate in Saudi Arabian sand dunes or if nuclear exchange was a significant likelihood, to be honest, I'd probably consider my stereo system a secondary concern.

The user of vacuum-tube amplifiers is almost by definition an "extremist," from a commercial point of view. His goal is the highest performance, and if he is building his own vacuum tube amp, he is certainly willing to do maintenance and the occasional repair; indeed, he probably enjoys it to a degree.

In my own case, my home-built amplifiers run a pair of 807 beam power tubes in regulated screen pentode mode at a Class-A power point of 40W.

Operated thusly (in the summer I turn the bias voltage up somewhat for thermal reasons as I listen less and less to dynamic music) a pair of output tubes lasts roughly 2500 hours. I set the bias on initial tube installation and at 30 minutes, 2 hrs, 5, 20, 50, 200, and thereafter roughly 500 hours. I consider this eminently reasonable, and while 6550s would probably last longer, they would need to last five times as long to make up for the cost differential. I can still buy 807s at local ham swap meets for $5 each.

Should I feel guilty about using up the power tubes? No. Indeed, the consumable nature of vacuum tubes is why there is still such an excellent selection of NOS vacuum tubes (once you get away from popular guitar amp and sweep (CB "linear") tube types). As long as people use them, they will make them. Excellent new tubes are now being made, and when the price of 807s, 1625s, and such gets to new tube levels, I will only use new tubes. If you make them last forever, they will quit making them.

This is not to advocate the wanton waste of tubes: considering a lot of old guitar amps are set up on purpose to get a couple hundred hours from a set of EL34s, I believe the 807s I use serve me well during their lives. But if you put the emphasis on sound where it belongs, the rest will fall into place.

There's no doubt that the classic McIntosh circuits, and the Bereskin, Nestorovic, de Paravicini, and Wiggins derivations of the basic concept (enhanced triode mode operation of beam power pentodes having been well-known even to ham radio operators building modulators: see the W6SAI Radio handbook as late as the 1970s) will provide superior tube life over Class A/AB operation in "classic" configurations. The majority of old McIntosh amplifiers in the hands of aficionados are not operated on a regular basis, though, and indeed generally McIntosh collectors are just that—collectors. Sonics have never been the strong suit of the McIntosh amplifier.

It's also fascinating that Mr. Modafferi would choose, of all the McIntosh output transformers in their catalog, the only one not actually wound by McIntosh and not using the unity-coupled n-filar design. The last McIntosh tube amplifier patent, issued in 1973 and not renewed, ran out in 1990. I suspect the fact that no one not connected to McIntosh has ever manufactured such a configuration is significant.

Again, it's not my intention to denigrate or discourage Mr. Modafferi's work in any way. I hope he will continue to publish interesting work. But I believe that trying to build a tube audio power amplifier which plays off sound for tube life is probably like putting a heavy diesel engine in a sports car: it might last longer, but would you buy one in the first place?

Keith Carlsen
Kc21130@hotmail.com

Richard Modafferi responds:

Mr. Carlsen's remarks reveal that he and I are on opposite sides regarding vacuum-tube audio amplifier design. I won't argue Mr. Carlsen's design philosophy, he can do as he wishes. However, I'll make one point concerning my design philosophy, and correct two misrepresentations of fact in his letter.

My idea of a vacuum-tube amplifier is that it should work like a refrigerator—plug it in and it goes. What a joy when you put on Mahler's Third, knowing full well that you will arrive safely to the end! You won't be resetting bias, or changing tubes, or even making major repairs, between record sides!

Now to two of Mr. Carlsen's allegations:

1. *McIntosh amplifiers are "collector-items" and not put to regular use. Wrong! Almost all of the thousands of vintage McIntosh amplifiers I have refurbished for Audio Classics since 1986 have come from owners who use them daily in a primary audio system.*
2. *Modafferi triode amplifiers "play off" sound (quality) for tube life. Wrong again. Opus one works correctly as intended, with no loss of sound quality, when driving "easy" speaker loads such as any Joseph Audio, or at one time, a pair of QUAD ESL-63s I had in for repair. No one in Audio Classics' listening panel was able to hear a difference in sound between the reference Krell and Opus One.*

IC SUBSTITUTION

I have a question regarding your article, "An Electronic Speed Control" (p. 89), reprinted in *The LP Is Back!* book (available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 603-924-6371, 603-924-9467, E-mail: custserv@audioXpress.com). Because the ILP HY60 module is no longer available, you suggest in your letter, published in the Jan. '01 issue of *aX* (p. 99), the use of the Amplimo A60 module. I found that this module is quite expensive.

What about building an amplifier using the National LM3875T power amplifier IC, as published by Charles Hansen in the Aug. '01 issue ("A High-Quality Two-Channel Chip Amp," p. 12)? This amplifier seems to have all the features to replace the HY60. Just today, I found out that *audioXpress* is selling ILP modules, but they have different model numbers. Could one of them be used in place of the HY60? (I suppose the 60W model should be OK).

Evangelos Kenourgios
Athens, Greece

Gary Galo responds:

The National LM3875T power amplifier should do the trick, as long as you keep the supply rails between ±35 and ±40V DC. You will also need to provide appropriate heatsinking for this device. Now that Old Colony is carrying the ILP line, I think that the HY2003 is actually the best bet. It comes complete with a power supply and integrated heatsink, and will therefore be easier to implement than the LM3875T. Just be certain that you select a power transformer that will give you supply rail voltages between ±35V and the ±40V maximum rating of the HY2003.

SUB-BASS RESPONSE

I read with interest the article "e-Bass: An Application of Phase Linear Sub-Bass Equalization" by Graham Maynard (March '02, p. 38). The more I read, the more I was struck by the similarity between the "speaker in a tiny box" and the 12dB per octave equalization and the work of Edward M. Long and Ronald J. Wickersham. They call their system the ELF system, which has several refinements not thought up by Mr. Maynard.

Their work is covered by patent 4,481,662. Bag End Loudspeakers has a well-respected line of subwoofers made under a license of this patent. Mr. Maynard states in the article that "The circuit is not patented . . ." (p. 47) and also "As far as I am aware, this is the first 'phase linear' sub-bass equalizer circuit to be constructed and published." I have material copyright 1994 (and that is version 1.2 of the "Guide to ELF systems").

You can find considerable information on ELF theory and Bag End's implementation in various commercial systems on their website at http://www.bagend.com. I do not know what Long/Wickersham Labs' attitude is about patent infringement, but at the least, readers should be aware that the patent has been granted. This is especially true in case anyone should try to develop the idea commercially.

Ted Miller
Elkhart, Ind.

THE DVD DRAMA CONTINUES

I have added the following very fruitful changes to my video system after publishing "A DVD Rescue" (July '02).

TV Receiver: I replaced the two filter capacitors for the 215V DC gun voltage with the largest caps which would physically fit their locations. An oscilloscope check of this voltage showed several volts of unregulated wiggle, much reduced by the change. I amputated the AC power cord near the set and replaced it with a CORCOM filter-in-a-box and a beefy 16-gauge 3-wire cord.

These changes improved color purity and balance, edge sharpness and detail, plus put more snap in the transient response. The set will now reproduce anything sent to it with the appearance of complete faithfulness to the original, and all on the Composite feed, often maligned, but a quality link if the attendant capacitors are seriously upgraded.

DVD Player: The Sony service literature finally arrived, and identified a great sensitivity in the system–jitter in the recovered RF signal. Sony went to the length of driving all servo motors used in reading the DVD disk with signals originating as measured jitter!

I epoxied a 4 lb slab of steel under the bottom of the metal chassis, where the transport screws on. I added more lead foil, plus a set of four new Audioquest hemispherical Sorbothane feet–visibly better than the old feet. Although I did not dismantle the transport, I removed it and added foil to it and its tray wherever possible, backing my way out with camouflage enamel. I wrapped a strip of lead foil around the spindle motor, exercising great care to preserve its balance.

The schematic showed that polarized electrolytic capacitors are used to send the video to the outside world. I bypassed these with tantalum caps, 10% of the value of the originals. Also, the power cord got the same surgery as the TV set.

In summary, the DVD player owes 95 percent of its improvements to vibration control and AC power, and only the last 5 percent to output capacitors, which brought in the tiny details.

This concludes the Rescue Drama. Excellent DVDs–and there are many–make sharp, natural edges and sumptuous color. Fine, filigreed detail is present, to the limits of the raster size on the Zenith. Armed with all the experiences documented in this text, I am looking forward with great anticipation to the day I acquire an HDTV monitor–and extract all it can give. ❖

Darcy E. Staggs
Orange, Calif.

Audio Aid

A Log Control Using a Linear Pot

With the potentiometer in the center position (*Fig. 1*), the gain of the op-amp Q1 is (1.1 + 5): 1.1 = 5.45×. However, the following voltage divider between Q1 and point C (the grid of the 6SN7) gives the same attenuation, so the total gain between points B and C = 1×.

In full clockwise position the gain of the Q1 is (1.1 + 10): 1.1 = 10×, without any following attenuation. This results in a gain of +20dB between points B and C. In full counterclockwise position the gain of Q1 is 1×, but is followed by an attenuation of (10 + 1.1): 1.1 = 0.1×, giving a gain of −20dB between points B and C.

In this way a linear potentiometer, being much more precise than a logarithmic type, will give you a linear in dB calibrated scale from −20dB to +20dB. If you like, you can also make the same scale in −30dB to +10dB. Changing the 1.1k resistors will give a similar scale, but calibrated in −10dB to +10dB. The voltage divider between points A and B prevents overloading when connected to a CD player, which at 0dB outputs a 2V AC signal. Just adapt this divider to suit your particular needs. ❖

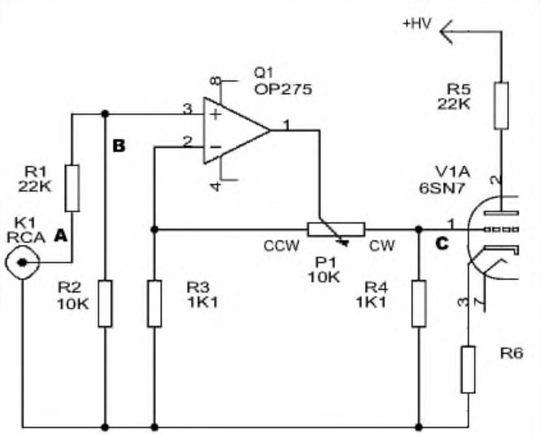

FIGURE 1: Log volume control with precision linear pot.

A. J. van Doorn
The Netherlands

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2 Velleman K4040 tube power amps, assembled, perfect, clean, like new, $1000 for both. Call 716-885-8250 or e-mail havenandhell3@cs.com.

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Hobbyist seeking pre-recorded, 2-track reel-to-reel tapes—7.5 or 15 ips. Can be 1950s/1960s commercial product or old studio dubs. Andy Pennella, 203-329-7498 or ajpennella@att.net.

Accuton C2-77 8Ω midrange driver, single piece or matched pair. Contact Kevin, kgallagh@risd.edu.

Two KEF B200 SP1075 woofers (triangular 8″ units used in the KEF 103.2 from the early 1980s). Call Perry Marshall, 708-788-4461 or e-mail perry@tannah.net.

Fellow audio enthusiasts for exchange of experience and ideas or just listening to music. Luxembourg city area in Grand Duchy of Luxembourg. Contact Jan Didden, jdidden@pt.lu.

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New Chips on the Block

AKM AK4560A By Charles Hansen

The AKM AK4560A has integrated six functions on one chip that, prior to this, were only done with multiple pieces of silicon. This will make the part useful to anyone designing an internet appliance, MP3 player, digital video camera, or other type of system where integration is the key to success.

Included in the AK4560A recording functions are an automatic level control (ALC) with wind-noise cancellation for the microphone amp, an internal input selection multiplexer (MUX), and digital HPF for DC-offset cancellation. Critical playback functions are a headphone/speaker amp, digital de-emphasis filter, equalizer, +2dBV lineout buffer, mute, 0 to –50dB analog volume control, beep and shutter input signals.

AK4560A FEATURES

Resolution: 16 bits

Recording function:

- 3-input selector (internal mike, external mike, line)
- Pre-amp/EQ-amp
- HPF-amp for wind-noise
- Digital ALC (Auto Level Control) circuit
- Fade in/Fade out
- Digital HPF for offset cancellation (fc = 3.7Hz @ fs = 48kHz)

Playback function

- Digital de-emphasis filter (tc = 50/15us, fs = 32kHz, 44.1kHz, and 48kHz)
- Lineout buffer
- Analog volume
- Headphone-amp
- Speaker-amp with a built-in digital ALC circuit
- Beep and shutter signal input

Analog-through mode

ADC characteristics

- DR = S/N: 86dB

DAC characteristics

- DR = S/N: 88dB

Master clock: 256fs/384fs

Sampling rate: 8kHz, 50kHz

Audio data interface format: MSB-First, twos complement (AK4518/AK4550 compatible)

- ADC: 16-bit MSB justified, DAC: 16-bit LSB justified

Multiple power supply: 2.6 5.5V

Low power dissipation, power management

Package: 64-pin LQFP, 0.5mm pitch

AK4560A pricing $7.04 US (10k pieces) AKM Semiconductor, contact Richard Kulavik toll-free at 888-256-7364, or icinfo@akm.com for info and engineering samples. ❖