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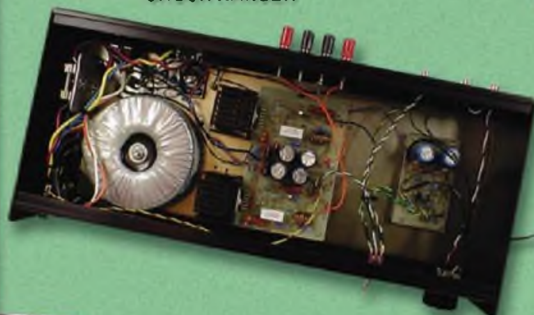

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Editorial

Out, Out, Damned Dot

Nearly everyone in the North American electronics establishment seems to me content to sit, seemingly happily and permanently, between two stools. The situation concerns the unhappy choice Michael Faraday made in choosing a quantity for that relationship between two layers of metal separated by nearly any dielectric that we refer to as capacitance. We refer to these units as microfarads, which isn't exactly the way it is with capacitance. Faraday's farad is very, very large in relationship to the components we normally use.

The microfarad is actually one millionth of a farad: 10^{-6} farads. Notice the minus sign before the exponent, which means that the number is smaller than one. We represent the micro with the 12th lowercase letter of the Greek alphabet, the μ, which is equivalent of our "m." We do not use the "m" for micro because it has already been taken by the next less small mathematical quantity, the thousandth, or milli (10^{-3}). Which is why when writing about microfarads we do not designate a capacitor as being 40mF. Unless we mean a 40,000μF device. We don't generally do that since we seem to have this passionate devotion to the little Greek μ and its decimal derivatives.

At least in North America we do. We also have a positive, abhorrent aversion for the perfectly respectable mathematical representation for numbers which are in the billionths, namely the nano. No, we would rather resort to the creaky, circumambulating, ditsy business of slicing up micros with decimals. What sense decimals make when talking about billionths escapes me. It is either laziness, ignorance, the herd instinct, or perverseness, take your pick.

Any quantity of farad slices that come in billionths can be designated as nanofarads. These are units which are 10^{-9}. The whole extent of decimated farads in this range, from 0.001μF to 0.999μF, is more properly and easily designated as 1nF to 999nF. The most often used 0.1μF as a bypass for op amps is far more easily written 100nF—same number of characters, and without the Greek letter (Alt 0181, on your computer keypad).

Fortunately North American electronics practitioners have made some small progress in the last three or four decades or so since we have, at least, abandoned "μμF" as a way of designating picofarads. Pretty generally we have accepted the fact that a trillionth of a farad (10^{-12}) is properly referred to as a pico. Thank goodness we have graduated from referring to these devices as accumulators or condensers, which we did for decades (shades of the Leyden jar).

Another major problem with this lazy decimation habit is the fragility of the little dot required for decimals. Some, who are obviously afraid the little decimal will get lost in the nineteenth copy of the schematic, put *another* zero in the lineup just to protect the puny little thing: as in 0.001μF. If you love redundancy, help yourself. However, isn't 1nF better? I submit it is not just better, it is *a lot* better.

It's easier to think of these generally used designators as three sets of threes.

Larger ⇐ μ n p ⇒ Smaller
000 | 000 | 000

In this simple way, you never need use a dratted decimal—that is, unless we can all agree on some of the larger units of electrolytic capacitance for smoothing low voltages such as a millifarad: 1mF meaning 1000μF, but that might confuse everyone. However, we should be able to manage a 1mF, realizing that the lowercase "m" always means milli or thousandths. Then, of course, we could move on to 1cF, the centifarad, or 100,000μF. This change would be very helpful on power supply schematics where the space around capacitors is usually limited. I'll be happy if we can all agree to just start using the nano regularly. Perhaps this reluctance is part and parcel of the U.S. fear of that "foreign conspiracy" known as the metric system, which is standard in most of the civilized world

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QUANTIFICATION FOR ELECTRONICS

LESS THAN ONE

	TRILLIONTHS PICO 10^{-12}	BILLIONTHS NANO 10^{-9}	MILLIONTHS MICRO 10^{-6}	THOUSANDTHS MILLI 10^{-3}	HUNDREDTHS CENTI 10^{-2}	TENTHS DECI 10^{-1}
ampere			μA	mA		
farad	pF	nF	μF	mF		
henry			μH	mH		
hertz						
ohm						
volt			μV	mV		
watt			μW	mW		

ONE AND LARGER

ONES	TENS 10^1	HUNDREDS 10^2	KILO 10^3	MEGA 10^6	GIGA 10^9
A					
F					
H					
Hz			kHz	MHz	GHz
Ω			kΩ	MΩ	
V			kV	MV	
W			kW	MW	

Note: The discoverers of these units are lowercased referenced quantities. The letter representing them is capitalized, except for Georg Ohm whose unit is designated by the Greek letter Ω, omega.

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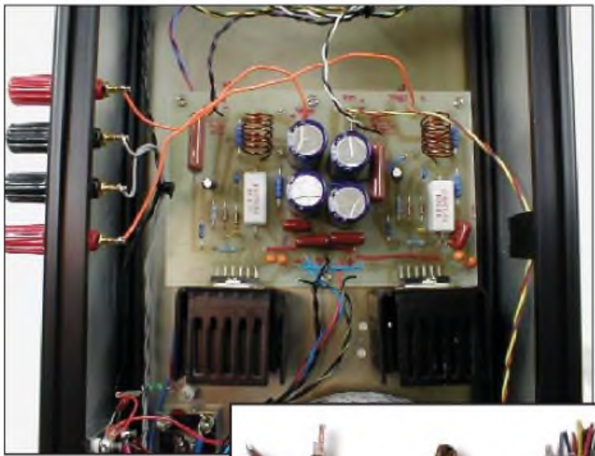

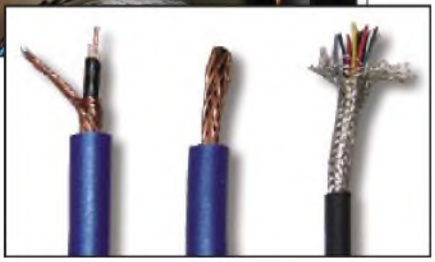

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.

JOHN STUART MILL

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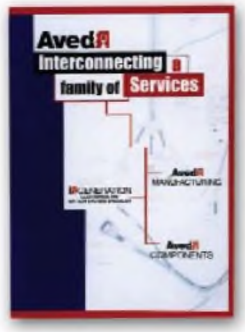
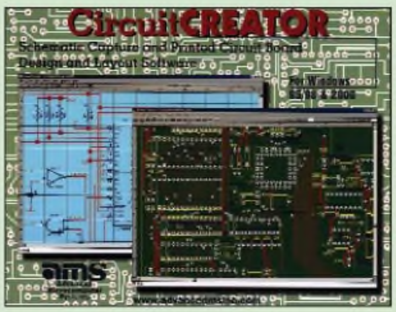


■ NEW PUBLICATIONS

Aved/Generation Electronics is offering a capabilities brochure describing design and manufacturing services for a wide range of custom cable assemblies, various types of battery packs and technologies, and interconnection components; for more information, visit www.aved.com. Focal Press (www.focalpress.com) announces the upcoming publication of *Modern Recording Techniques, Fifth Edition*, a leading teaching guide, comprehensive manual, and reference tool for those wishing to learn professional recording. Sams Technical Publishing (www.samswebsite.com) recently released their 2001 Annual Index. Sennheiser (860-434-9190) recently published *The Handy Guide to Evolution Wireless Systems*, a free, 32-page overview and application primer for Evolution wireless microphone systems.



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Slim Jims: Get the Skinny on an Easy-to-Build Project

Get in shape—sonically and physically—with these 4″ driver speakers.

By Ken Ketler

PHOTO 1: The completed speaker in place.

During my one-hour commute to work every weekday, I have plenty of time to just think about stuff. One particular morning, I had a revelation that hit me like a ton of Oreos: *Egad, I'm out of shape!* Although we have a stationary exercycle in our basement, why on God's green earth would I want to bother with that? If I could listen to music while pedaling, I just might be able to trick myself into a healthy dose of self-discipline.

The results certainly remain to be seen, but I decided to build a quick and easy pair of loudspeakers using some full-range drivers that were sitting around in storage. I decided this might be an interesting exercise to help new speaker builders get their hands dirty and experienced folks might like their time-coherence. You won't need much woodworking experience to complete this project, but you must love to listen to music and be able to have fun! You do not need to wear a pulse-rate monitor.

If you ask me, full-rangers often get a bad rap. I've heard for years about their poor high-frequency response, bad dispersion characteristics, and plenty of other reasons never to go near them. George Gershwin said it best when he wrote "It Ain't Necessarily So." Although I suppose it is possible that he wasn't referring to full-range speakers, it does apply here.

For now, please forget all preconceived notions. And to be fair, remember how many headaches multi-driver systems can cause with their crossover systems and driver time-adjustment issues. Every configuration has some trade-offs.

There is a small, but adamant, faction of people who swear by the time-coherence and imaging of a single-driver configuration. They are often the same folks who prefer low-powered single-ended triode amplifiers and ultra-efficient horn loudspeaker designs. If my little excursion into the full-range world happens to pique your interest, I encourage you to visit the Single Driver Website™ at http://melhuish.org/audio/index.htm. I have found this forum of people to be very helpful with their suggestions and willing to openly share their own results.

The enclosure for this project is simply a piece of PVC pipe. When I designed these loudspeakers, I planned to hang them from the ceiling with the back-ends of the pipes aiming into the wall/ceiling junction. I believe this makes Slim Jims good candidates for a garage or basement workshop. Of course, please feel free to modify and/or put this project wherever you'd like (*Photo 1*).

ABOUT THE AUTHOR
Ken is a Technical Writer who enjoys many flavors of music and audio equipment. He and his wife, Julie, often review consumer equipment for *audioXpress* magazine and happily spend large quantities of time with their son, Thomas.

PHOTO 2: Fitting speaker cables to the unit.

PREPARING THE INGREDIENTS

- Get two Radio Shack 40-1197 drivers (approximately $12 each).
- Cut two 26.5″ length pieces of 4″ diameter PVC pipe (enclosures).
- Cut two 2.75″ length pieces of 1.5″ diameter PVC pipe (bass reflex vents).
- Cut two 5″ × 5″ squares of any wood you have around. Using a hole saw, cut a hole in the center of each

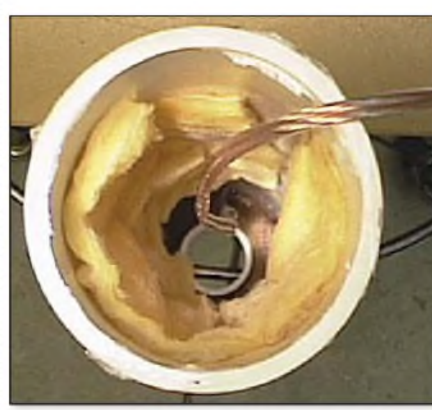

PHOTO 3: Inside the pipe.

square of wood that will tightly fit your 1.5″ diameter PVC pipes. Also, drill a small hole in each piece that will fit whatever speaker cable you prefer to use (*Photo 2*). Please feel free to use whatever type of connectors you like. I used cable simply because my budget was thin, unlike my stomach.

- Cut two 23″ × 13″ pieces of fiberglass speaker damping material (the yellow stuff from Radio Shack is fine).
- Cut two 30″ length pieces of speaker cable.
- Purchase a small can of polyurethane.
- Buy a roll of clay rope caulking.
- Obtain a tube of latex or silicone caulking. I prefer latex because it is stiffer. Since I didn't use screws to secure the driver to its enclosure, I thought that latex would have a better resistance to flexing during loud musical passages, which would reduce efficiency and final sound quality.

GET IT TOGETHER

For each loudspeaker:

1. Insert the 1.5″ pipe in the center of the wood square so that it is flush at the outside edge. Seal with the latex.
2. Run your zip cord through the drilled hole in the wood square so that the cable is long enough to stretch up the length of the whole enclosure to the driver. Again, seal with latex.
3. Glue (latex) the wood square to one end of the enclosure pipe. Make sure that there are no air gaps between the wood and the enclosure pipe.
4. Using latex caulking, glue the fiberglass sheet to the inside perimeter down the length of the 4″ pipe. Be sure it is flush with the inner-wall so it doesn't obstruct the path between the driver and the reflex vent (*Photo 3*).
5. Solder the speaker cable to the 40-1197 driver.
6. Glue (yep, latex) the driver to the open end of the enclosure pipe. Again, make sure there are no air leaks.
7. Splice the other end of the speaker cable to a longer piece, which you connect to your amplifier. This will make it easier to move the Slim Jims later (no huge pieces of cable hang from the enclosures). Don't forget to cover your splices with heat-shrink tubing or tape to avoid short circuits.

Of course, you must allow the latex to dry in between each step. If you'd like to decorate the loudspeaker, contact paper seems like a good cheap solution. I've used a funky granite-looking pattern that...well,—certainly looks better than just plain PVC!

TAME THAT MIDRANGE!

This is where the art in speaker building can really kick in! The 40-1197 driver has a rather bothersome mid-frequency peak. This tends to make it sound like an AM radio speaker. It is, after all, a "budget" driver. There are three practical ways that I know of to smooth out this peak to make the 40-1197 a very "musical" speaker. I suggest fine-tuning the system after it's built so that you can hear the changes as you make them. As a result, your changes will be based on your own hearing within your own acoustic environment.

The first and quickest solution to smooth out this midrange peak is to use a graphic equalizer. You'll also be able to tweak your room response if it is particularly troublesome. Another fairly simple solution is to build a notch filter with three simple components (*Fig. 1*):

R = 6.8Ω
C = 4.7μF
L = 0.8mH

This filter is often recommended on the Single Driver Website™, but I admit right here, right now, I haven't tried it myself.

The third solution is the most complicated, yet the most educational. Modify the driver! After reading a great deal of opinions on the subject, here are the modifications that I chose to use on my 40-1197 drivers (*Photo 4*):

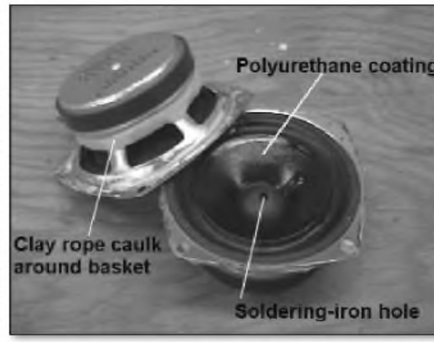

PHOTO 4: Modifying the drivers.

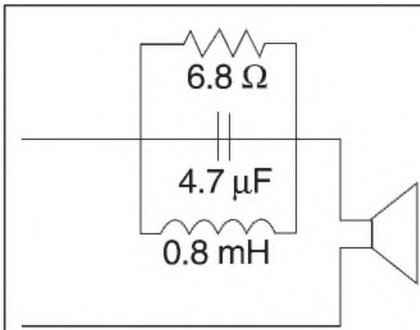

FIGURE 1: Filter circuit.

1. Apply two coats of full-strength polyurethane to the front side of the driver cone. This will help to damp various resonant modes and smooth out some of the "jagged edges." Be sure not to get any on the center dust cap or surround of the driver.
2. The basket of the 40-1197 is a stamped steel design. It can often have a bell-like resonance that you can easily damp. A few wraps of clay rope caulking around the magnet/basket junction does a great job of deresonating the frame of this driver.
3. I have heard many different opinions regarding this third step, but I have tried it and am pleased with the results. With a hot soldering iron, very carefully burn a 1/16″ diameter hole in the center dust cap of each driver. This will further smooth out the midrange response by stopping vibration breakup modes that can truly ruin the midrange quality of this driver. Again, do this with the system assembled, so you can fine-tune the size of the hole "by ear."

The clearest explanation I have heard was a simple analogy to "drilling

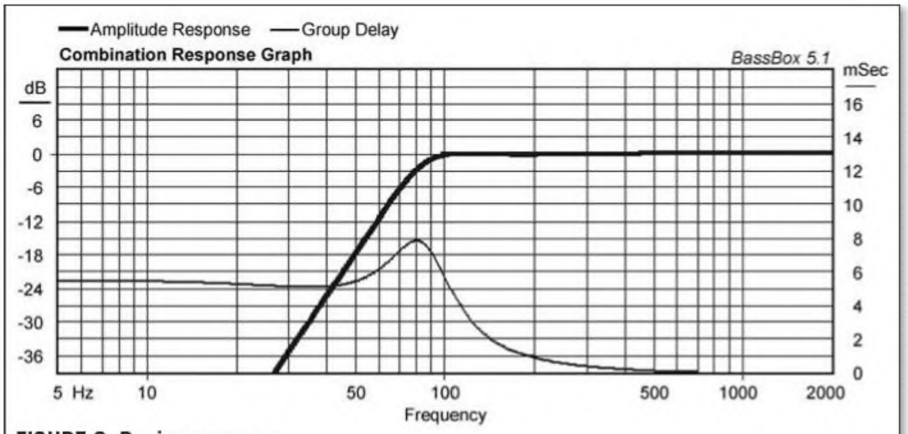

FIGURE 2: Design response.

a hole at the end of a crack to keep it from spreading." In fact, some models in the Fostex FE-AV Series of full-range drivers have a conspicuous-looking hole right in the center of their dustcaps! Hmm, maybe it's not such a crazy idea.

RESULTS

I hung the Slim Jims (using string) from the ceiling beams in my cellar so that the drivers are angled down toward me, the pedaler, while the port end is firing right into the corner where the ceiling meets the wall. If you are thinking of hanging them as I did, put dabs of superglue on the knots in the string so they don't come untied (*Photo 1*).

Overall, they sound amazingly good, especially for little 4″ drivers! The stereo imaging is excellent and the bass is clean although somewhat limited. The upper frequency response sounds great up to about 15kHz and drops off around 16.5kHz.

Now, I can hear you hecklers in the back row. Yes, a cone driver begins to beam its signal like a flashlight near the frequency at which wavelength equals speaker diameter (in this case, somewhere around 3.5kHz). I may be overly optimistic, but I think this is actually an asset. In my concrete basement (a very reflective acoustic environment), most of the high-frequency signal arrives at the listening position without being corrupted by early reflections. Great for musical clarity and stereo imaging!

The designed frequency response drops off at 81Hz and the enclosure/port is tuned around 84Hz. Firing the ports into the corner gives a smooth bass rise right around cut-off. When I swept the loudspeakers with a tone generator, their in-room output was strong to around 80Hz and continued to provide useful output down to about 62Hz, but then dropped off like a stone. Although I don't have a sophisticated audio test lab, I relied on the funny-looking microphones that are attached to my head and my Radio Shack sound level meter.

According to my computer design (using BassBox 5.1 software), the "group delay" at the tuning frequency is approximately 8ms (*Fig. 2*). This means that the bass response is quick and punchy. I think this is because I am not asking the driver to do more than it wants to—80Hz is right in the 1197s' ballpark. If you try to force this driver to work much below this, it will pump like a sweaty guy on an exercise bike and its group delay/transient response will go to heck in a hand basket.

If you are more of a bass fiend, consider adding a subwoofer or using larger-diameter full-range drivers in a similar arrangement. But for now, try building yourself a pair of Slim Jims. They're quick and cheap, kind of ugly, but surprisingly musical-sounding!

As far as the "other" results are concerned, I have actually been dragging myself out of bed every morning at 5:30 and cranking away to the best tunes I can find in the dark. After a few songs, I usually start jammin' like a rock star. Who'd have thought that playing air guitar could be so good for you? ❖

A High-Quality Two-Channel Chip Amp

This IC-based rear-channel amp has several uses to improve the quality of your sound system. **By Charles Hansen**

PHOTO 1: Front view of completed unit.

This project is a simple 25Wpc (watts per channel) power amplifier based on the National Semiconductor (NSC) LM3875T amplifier IC. I use it to power the rear ambience speakers of my audio system. The L+R and L–R ambient information is extracted from the stereo signal using the Audio Amateur QuadPod[1]. While I built the QuadPod into my amplifier chassis, the project as presented here is just as applicable to a two-channel amplifier project alone (*Photo 1*).

The 3875 has appeared in Audio Amateur publications before[2], but it was a single supply subwoofer amp from *Fig. 2* of the LM3875 data sheet. I prefer a dual-supply design, which dispenses with the large electrolytic output coupling capacitor and a number of bias components. I also made changes to the Zobel RC to match my speakers, changed the gain capacitor from 10μF to 39μF to extend the bass (my –3dB point is 4Hz versus 16Hz for data sheet *Fig. 2*), and changed the input network. Total parts cost is about $180 (without the QuadPod).

HOW IT WORKS

The schematic diagram for the audio amplifier is shown in *Fig. 1*, along with the LM3875 package outline and its five pin connections (the other six pins have no connection). The parts list is in *Table 1*. One of the nice advantages of toroidal power transformer T1 is the dual secondary winding.

To keep the power supply-induced noise to a minimum, I full-wave-rectified both windings with CR1 and CR2 to ±30V DC no load, or about ±27V DC at 25W. Capacitors C1–C4 snub the rectification noise at CR1, while C12–C15 do the same for CR2. Capacitors C5–C8 filter the rectified DC that is applied to power amplifier ICs U101 and U201. Film caps C9 and C10 filter RF noise from the DC supply rails.

In addition, the supply rails to the two ICs are locally bypassed with 10μF tantalum caps as recommended by the application notes. R1 limits current to power-on indicator LED 1, and fuses F1, F2, and F3 provide fault protection. Going from a single rectifier bridge to two bridges reduced the 180Hz power-supply component from –79dB to –91dB, taking a big bite out of the "N" portion of THD+N.

Each channel of audio amplification is provided by a National Semiconductor LM3875T audio power amplifier IC. This device is of much higher audio quality than the power amplifier ICs typically used for car audio and com-

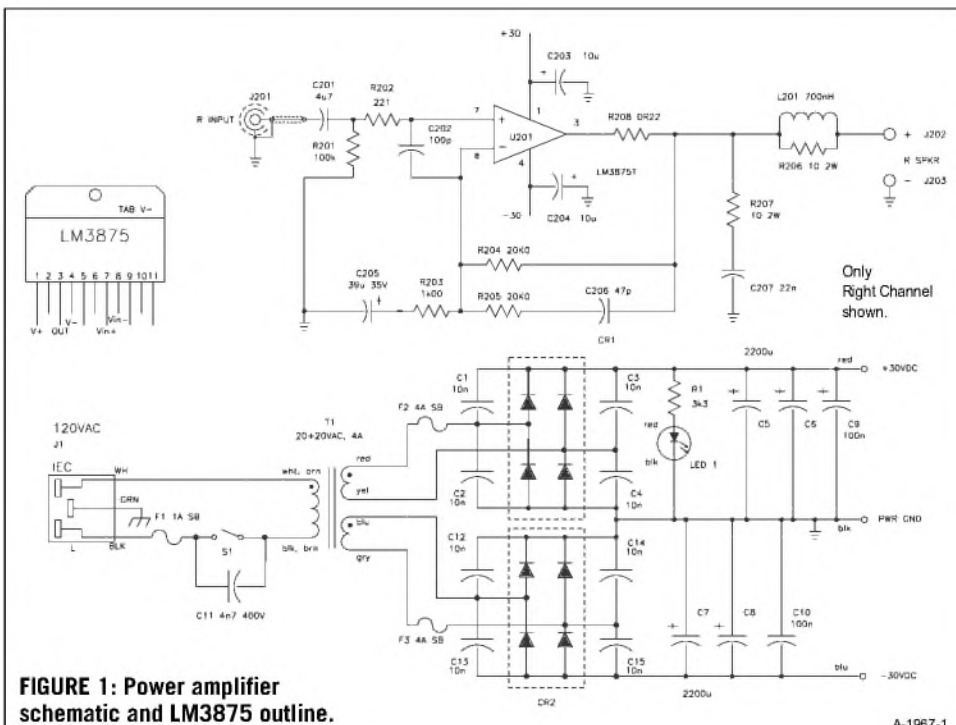

FIGURE 1: Power amplifier schematic and LM3875 outline.

puter speakers. It is housed in a plastic 11-pin metal-tab package. A data sheet for the LM3875T is available at the National Semiconductor website as a PDF file (www.national.com). I suggest you download it so you can avail yourself of the application notes, data, and test circuits.

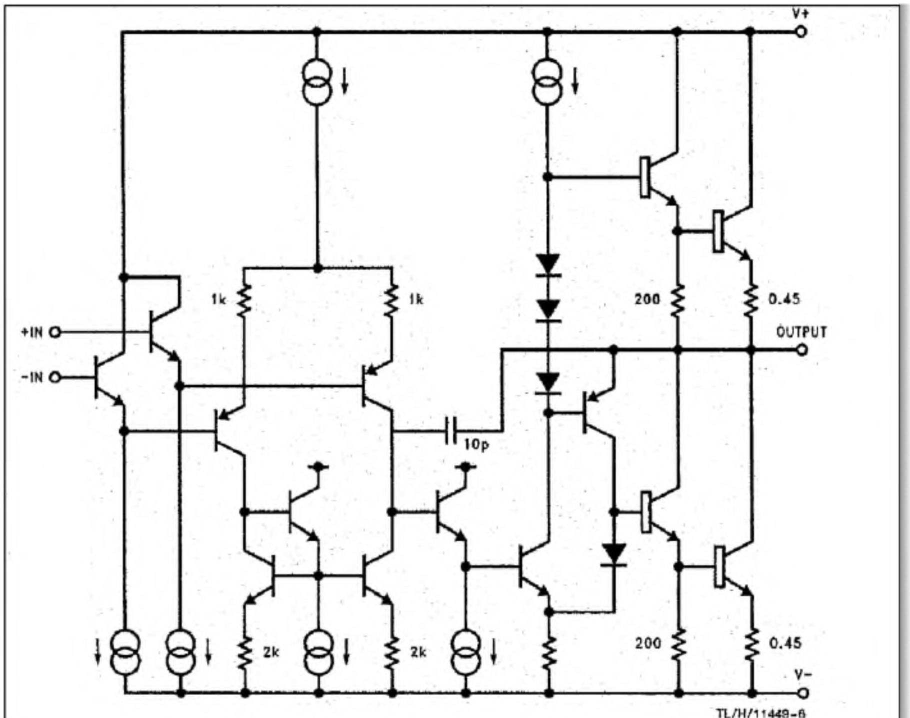

FIGURE 2: LM3875 equivalent schematic. © National Semiconductor Corp. and reprinted with permission of National Semiconductor Corporation. NSC reserves the right at any time to change said circuitry without notice. A-1967-2

I will describe the left channel since both are identical. The line-level audio signal is applied to DC blocking capacitor C101 and R101 through input jack J101. R102 and C102 reduce the gain at high frequencies to avoid quasi-saturation output transistor oscillation, and help suppress external RF noise. R102 also limits the input current that may be passed through to the output of U101 after power-down, once the supply rails fall below ±1.5V DC.

U101 is configured as a non-inverting amplifier. The mid-frequency gain is set to 21 (26dB) by R104 and R103 (Av = 1 + R104/R103). C105 produces a LF pole at 4Hz and ensures unity gain at DC. R105 and C106 produce a HF pole to roll off the gain above 170kHz. R105 is needed because the LM3875 is stable at or above closed-loop gains of ten or greater.

The amplifier output is coupled to the load through R108 and a parallel L-R network consisting of L101 and R106.

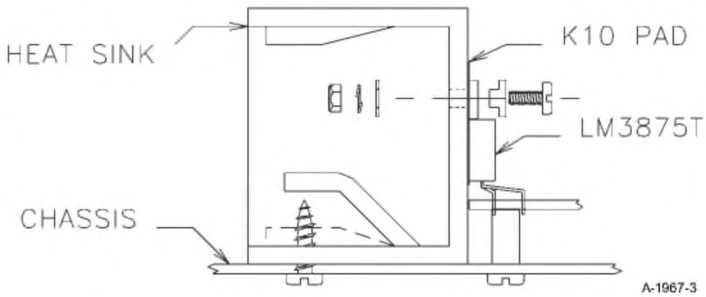

FIGURE 3: Heatsink detail cross-section.

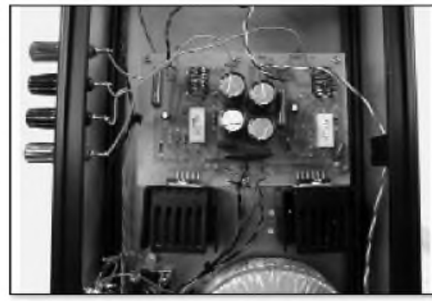

PHOTO 2: Amplifier circuit board close-up.

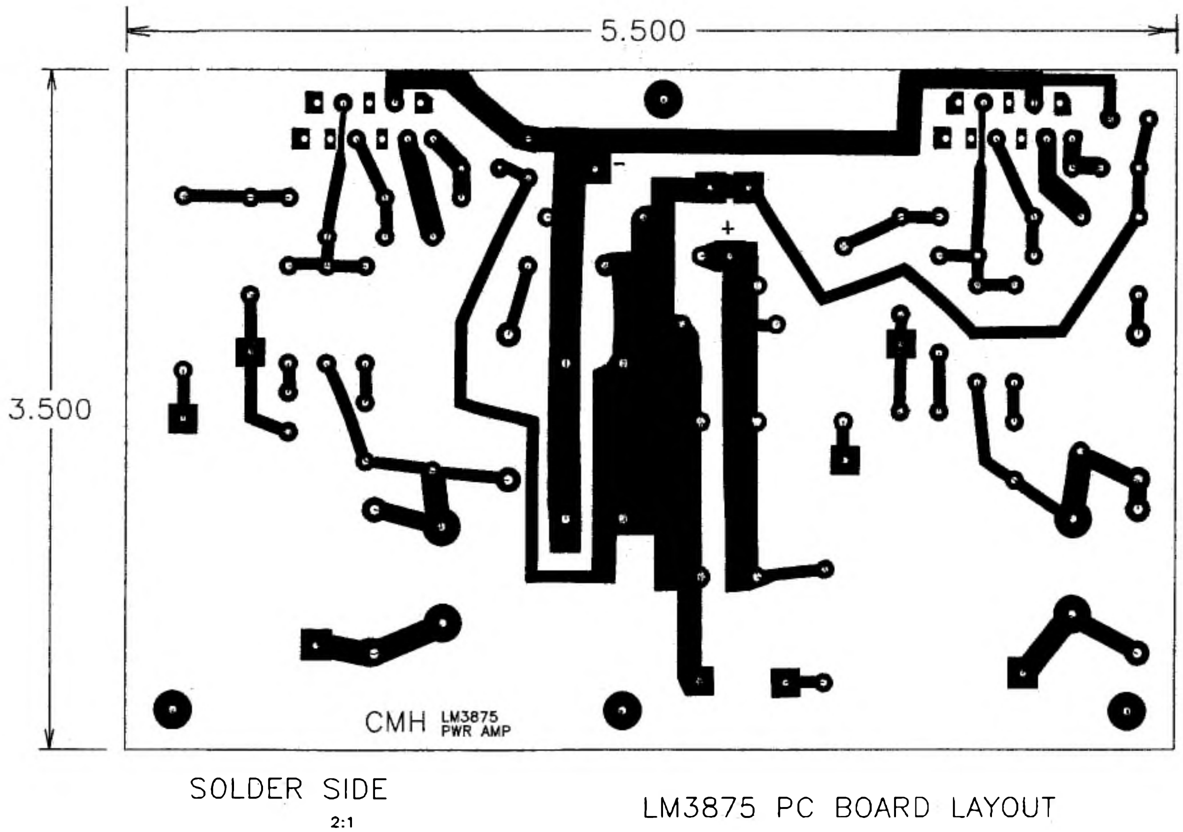

FIGURE 4: Copper side of circuit board. (100%)

The impedance of L101 increases at high frequency, so that R106 can decouple a capacitive load and reduce the Q of the resulting series resonant circuit. I wound L101 with 18-gauge wire so its resistance is negligible. However, after initial tests with a load of 8Ω in parallel with 2μF produced oscillation near 125kHz, and enough thermal distress that the chip protection circuits shut down, I added R108 to further decouple the IC from capacitive loads.

To achieve ideal pole-zero compensation with this inside-the-loop resistor, you could add another cap from output to inverting input. However, this capacitor would need to be fairly large, so I settled on some harmless frequency

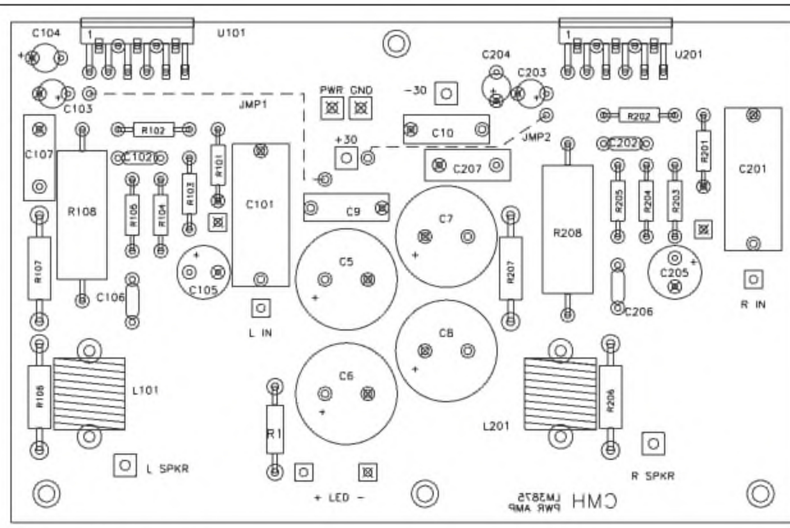

FIGURE 5: Stuffing guide.

peaking at 46kHz. This manifests itself as several cycles of damped ringing during square-wave testing. If you have no intention of driving capacitive loads such as electrostatic speakers or "difficult" crossovers, leave out the 0R22 resistor and enjoy the added efficiency.

The Zobel network, R107 and C107, produces an output stage pole that helps prevent high-frequency oscillation with real speaker loads. You may need to adjust these values to suit your own speakers, or use the default values in the LM3875 data sheet.

The LM3875 is capable of delivering 56W of continuous power to an 8Ω load

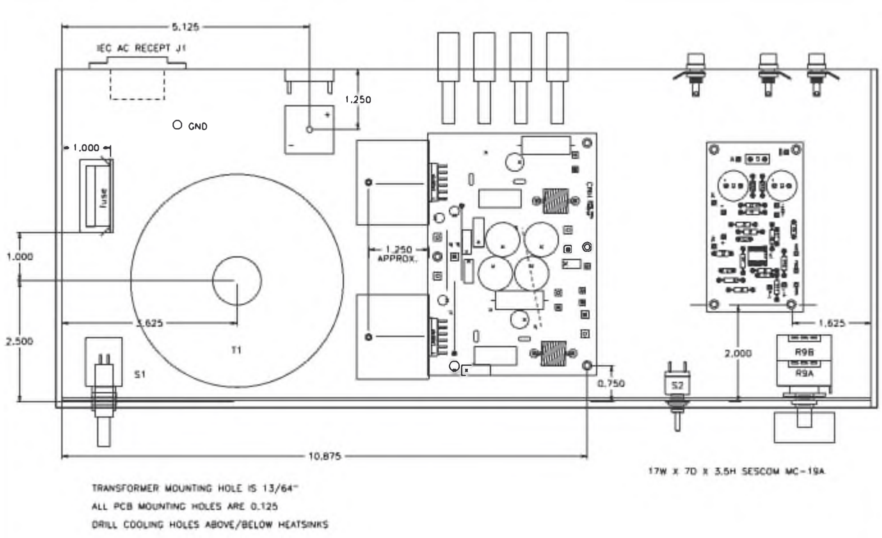

FIGURE 6: Chassis interior layout. A-1967-6

TABLE 1
PARTS LIST

DESIGNATION	VALUE, DESCRIPTION	VENDER P/N
C1–C4, C12–C15	10nF 100V V-stacked film	DK P4713-ND
C5–C8	2200μF 35V aluminum HFQ	DK P10335-ND
C9, C10	100nF 50V 2% PP film	DK P3104-ND
C11	4n7 400V film	DK P1066-ND
C101, C201	4μ7 100V Mylar	DK EF1475-ND
C102, C202	100pF 5% NPO ceramic	DK P4456A-ND
C103, C203, C104, C204	10μF 35V Tantalum	DK P2065-ND
C105, C205	39μF 35V aluminum HFQ	DK P5727-ND
C106, C206	47pF 5% NPO ceramic	DK P4452A-ND
C107, C207	22nF 50V PP film	DK P3223-ND
CR1, CR2 (see text)	25A 100V bridge	DK MB251WMS-ND
F1	1A SB fuse	
F2, F3	4A SB fuse	
J1	IEC filter receptacle	
J101, J201	Phono jack, gold (pair)	PE 091-1120
J102, J202, J103, J203	5-way binding post (pair)	PE 091-1140
L101, L201	700nF air-core	See text
LED1	Red LED indicator	
R1	3k3 ½W carbon resistor	DK 3.3KH-ND
R101, R201	100k ¼W 1% metal film	DK 100KXBK-ND
R102, R202	221Ω ¼W 1% metal film	DK 221XBK-ND
R103, R203	1k00 ¼W 1% metal film	DK 1.00KXBK-ND
R104, R105, R204, R205	20k0 ¼W 1% metal film	DK 20.0KXBK-ND
R106, R206, R107, R207	10Ω 1W 5% metal oxide	DK P10W-1BK-ND
R108, R208 (see text)	0R22 5W wire wound	PE 015-.22
S1	SPST toggle switch, 10A	
T1	20+20V AC, 4A toroidal	Plitron 057027201
U101, U102	LM3875T power amplifier IC	DK LM3875T-ND
	Fuse clip, single (3)	DK 3536-ND
	Heatsinks (2)	DK 294-1076-ND
	Heatsink insulator pad, K10 (2)	DK BER118-ND
	TO-220 mounting kit	Jameco 34121
	Chassis	Sescom MC-19A
	RCA jack insulators	DK SC1144-ND and SC1145-ND

with less than 0.1% THD+N over the audio band. This requires supply rails of ±35V DC. Since the device can function with supply rails ranging from ±10V DC to ±42V DC, I chose to design a more modest 25Wpc (8Ω) amplifier using ±27V DC rails.

If you prefer more power, you will need a larger transformer, with a higher secondary voltage. Higher current (do not exceed 4A) will also require a larger wire size for the speakers and power-supply wiring. Use the same wire gauge as that of your transformer secondary leads. You will also need to increase the voltage rating of the polarized caps. The tantalums should not see more than 78% of their rated voltage, and the aluminum caps should be limited to 80% of rated voltage at maximum audio output.

You can also add a stereo volume control between the input jacks and the amplifier board. I used a dual 20k audio pot, a small diameter Alps control I had on hand. Mouser sells 1″ diameter dual audio pots (the part number is 313-2420-value); you can use 10k or 50k (20k is not available).

LM3875 DESCRIPTION AND PERFORMANCE

The NSC LM3875 is one of three Overture™ amplifiers (the 56W LM3875, the 56W+Mute LM3876, and the 68W+Mute LM3886). DigiKey sells these devices as rated for 40W, 40W+Mute, and 60W+ Mute, respectively. Overture amplifiers are all based on Bob Widlar's classic LM12 design[3], and there are good application notes available on the NSC website.

Monolithic power amplifiers such as the LM3875 use a quasi-complementary output stage as shown in the equivalent schematic (*Fig. 2*). Even the complementary bipolar process is not yet able to produce matched-complementary NPN and PNP high-power transistors on the same chip. A monolithic PNP output device (whether lateral or vertical) has lower performance than the NPN device.

To compensate for this design compromise[4], the LM3875 uses a composite "power PNP" made from a lateral PNP and the usual NPN power devices, with some careful engineering for much lower Class-AB crossover

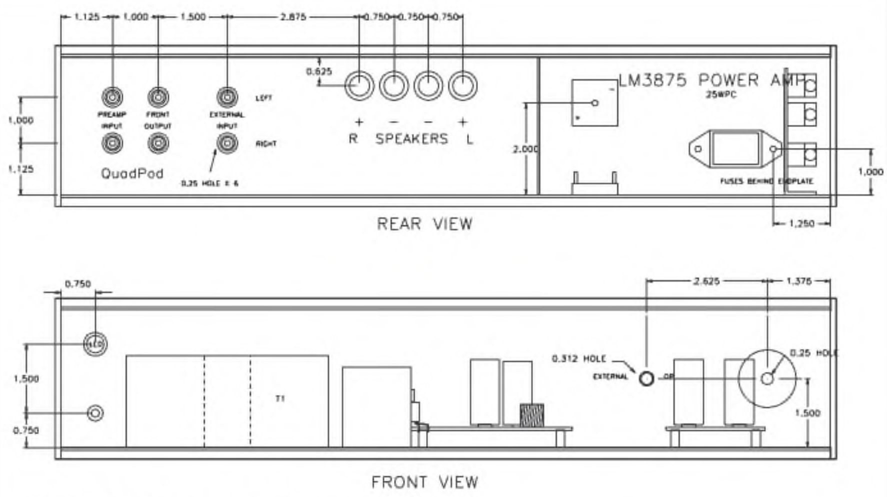

FIGURE 7: Chassis front and rear views.

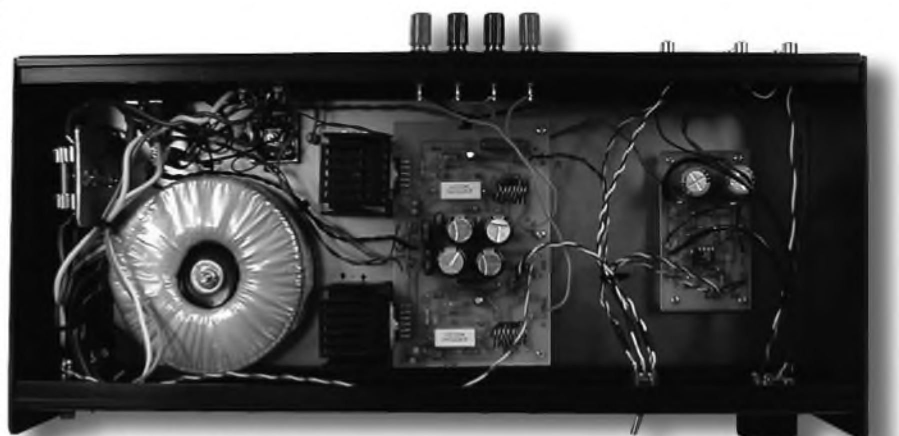

PHOTO 3: Interior view.

PHOTO 4: Rear view of unit.

distortion. The input stages are designed for low noise (the input is run "richer"). Since all the devices are on one piece of silicon, the thermal tracking is excellent, ensuring a stable output-stage bias point.

Undervoltage protection allows the power-supply voltages to approach their full levels before the amplifier is turned on. This prevents any DC output spikes. A similar function occurs on shutdown, where the output of the LM3875 is brought to ground before the power supplies decay. Overvoltage protection limits the output current to 4A peak, while providing output voltage clamping.

SPiKe (self peak instantaneous temperature, Kelvin°) protection limits the output array safe operating area (SOA) voltage-power-current profile. This involves small temperature-sensor diodes distributed all over the die and all around in the power transistors, so any hot spots cause it to shut down, rather than go into thermal runaway.

Thermal protection prevents long-term thermal stress. The LM3875 will

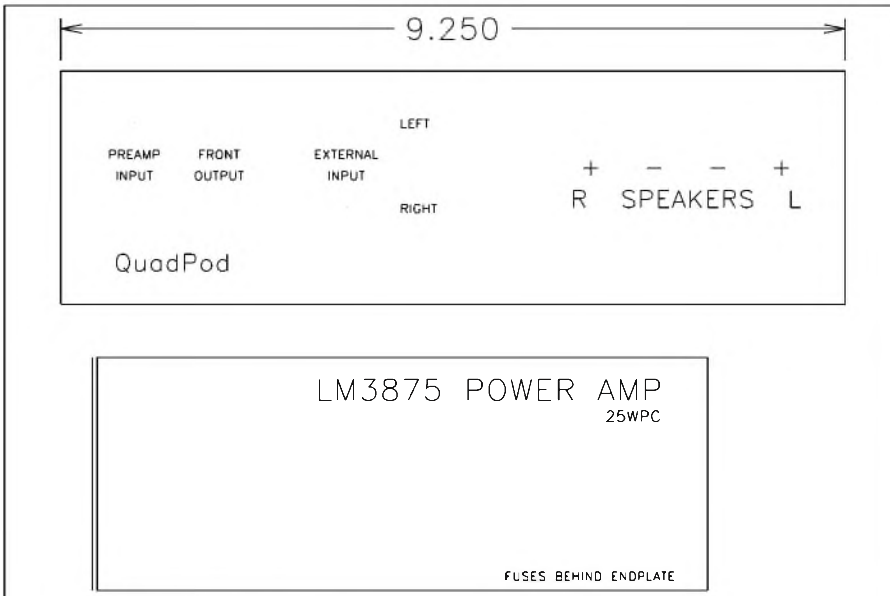

FIGURE 8: Rear panel lettering. A-1967-8

shut down if the die temperature reaches 165°C, and resume below 155°C. A sustained fault will cause the amplifier to cycle between the thermal shutdown limits, greatly reducing the thermal stress. Momentary thermal transients, such as a short-circuit at the output to ground or to either supply, or back emf from a reactive load, can cause the die temperature to rise 100°C in less than 100µs. This also results in a safe shutdown.

HEATSINK DESIGN

The methodology for the heatsink design is provided on the LM3875 data sheet, pages 10 and 11. Based on the formula provided, each IC dissipates 10.7W with 25W audio output into 8Ω. If you use 4Ω speakers, the IC dissipation will increase to 25.5W, and you may need larger heatsinks with a maximum θ_{SA} (thermal resistance sink to ambient) of 4.8°C/W. I decided to use a finned heatsink mounted on each IC (IERC p/n 7-404 with 3°C/W—see parts list). I made the heatsinks part of the PC board assembly.

I attached the bottom of each heatsink to the chassis with a 6 × ½″ sheet-metal screw to provide a good thermal path and to take the mechanical stress off the ICs (*Fig. 3*). *Photo 2* is a close-up of my prototype amplifier PC board. I made some changes based on testing, so it looks different than the final layout in this article.

Do not mount the heatsinks onto U101 and U201 until after all the components are stuffed and all the wiring is complete (more on this later). This pre-

TABLE 2
CROSSTALK

FREQUENCY	R TO L	L TO R
100Hz	–60dB	–60dB
1kHz	–57dB	–52dB
10kHz	–39dB	–35dB
20kHz	–32dB	–30dB

vents stress or damage to the amplifier ICs and insulators while you are handling the PC board.

CIRCUIT BOARD LAYOUT

I designed an etched, single-sided circuit board for this project. The prototype board was furnished by one of the *audioXpress* suppliers (Atlas Circuits Company, 1500 Old Lake Rd., PO Box 892, Lincolnton, NC 28092). The copper side of the PC board is shown in *Fig. 4*. The board uses two wire jumpers to complete some of the circuits.

If you use the prototype board supplier, the holes will be drilled with a single drill size. You must re-drill a number of these holes to accommodate wires and the larger component leads of some parts. Use the drill sizes in this table:

C5–C8, C101, C201, R108, R208	#56 drill
LM3875s, R1	#54 drill
L101, L201; wire and jumper pads	1/16″ drill
Mounting holes	1/8″ drill

The stuffing guide is shown in *Fig. 5*. Be sure to install both 20-gauge wire jumpers, using sleeving to prevent shorts to other components (or use insulated hookup wire). JMP1 and JMP2 indicate these jumpers.

I designed inductors L101 and L201 using a formula in the *Reference Data for Radio Engineers*[5]. Wrap nine turns of 18-gauge wire around a ½″ diameter wood dowel. Space the turns as evenly as possible, so the finished length is also ½″. Dress the free ends of the leads so they fit into the PC board holes.

In keeping with good assembly practice, install the least sensitive parts first, followed by the more sensitive parts. Begin by soldering the passive parts (inductors, resistors, then capacitors). Install the LM3875s last. Double-check the orientation of the polarized components.

CONSTRUCTION

I used a 17 × 7 × 3.5″ Sescom MC-19A chassis for my project, which has enough room for the QuadPod board. For the amplifier alone, you can use the MC-25A (12.75 × 7 × 3.5″). *Figure 6*

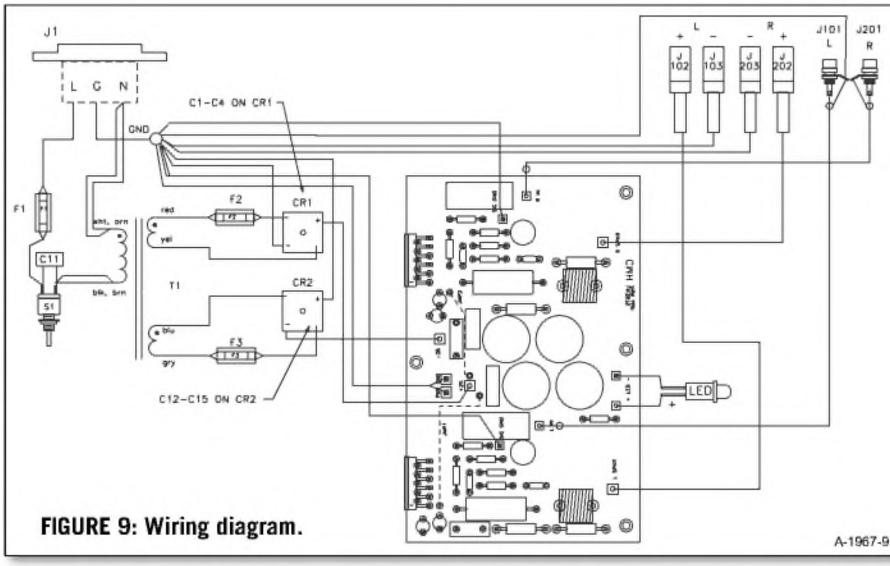

FIGURE 9: Wiring diagram.

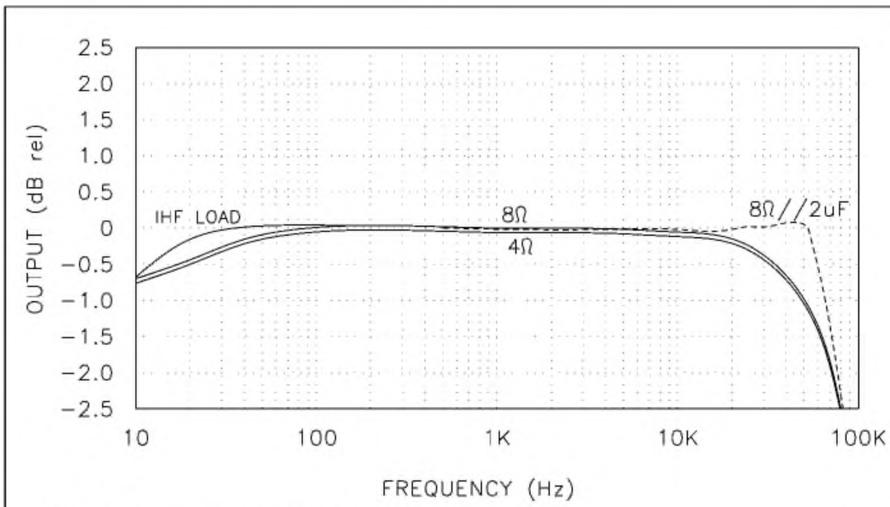

FIGURE 10: Frequency response.

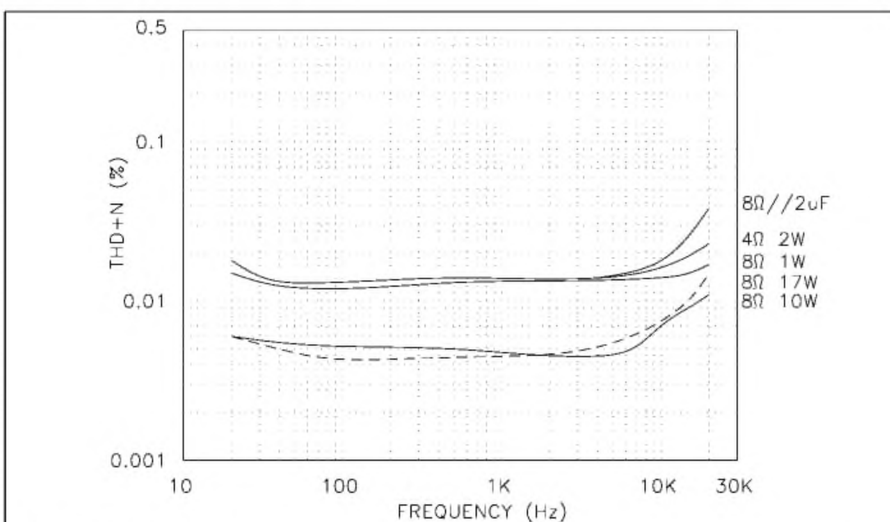

FIGURE 11: THD+N versus frequency.

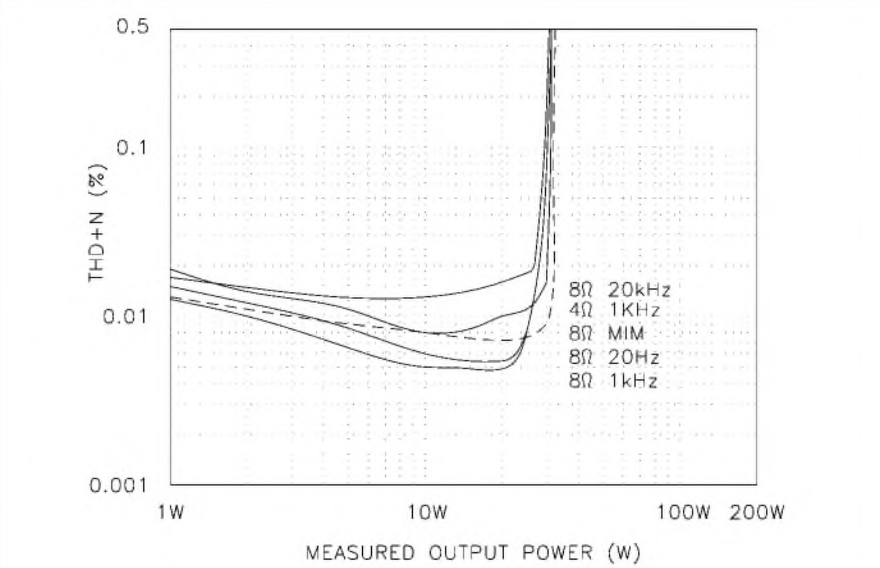

FIGURE 12: THD+N versus output power.

A-1967-12

shows the dimensioned interior of the amplifier chassis layout. All the wiring on the power-supply side is twisted to minimize magnetic fields, which could induce power-supply noise or hum.

I drilled a pattern of ⅛″ holes above and below the heatsinks to promote convective cooling. One hole must line up with the center slot in the bottom of each heatsink, on the center-line of the hole in the LM3875. The rest of the holes are evenly spaced ¼″ apart from each heatsink mounting hole. The slot in the heatsink can accommodate a fair amount of error toward or away from the PC board.

The interior view of my amplifier (*Photo 3*) includes the QuadPod circuit board, switches, jacks, and controls in my particular amplifier. I mounted the fuse holders on a 2″ × 3.25″ aluminum L-shaped bracket so they are easily accessible after removing the chassis end plate. I mounted CR2 on the inside of the rear panel.

I suggest this location because it gives more room for the wiring. However, feel free to rearrange the components as you see fit. You may even prefer to locate your DC supply filter caps at the rectifiers, which will reduce power-supply noise even further.

The front and rear views, with dimensions for chassis-mounted components, are shown in *Fig. 7*. I decided to paint the front panel flat black and use a minimum of lettering (*Photo 3*). If you build it without the QuadPod board, you don't need any lettering, since the front panel will have only a power switch and LED (self-explanatory).

All the detailed information for my

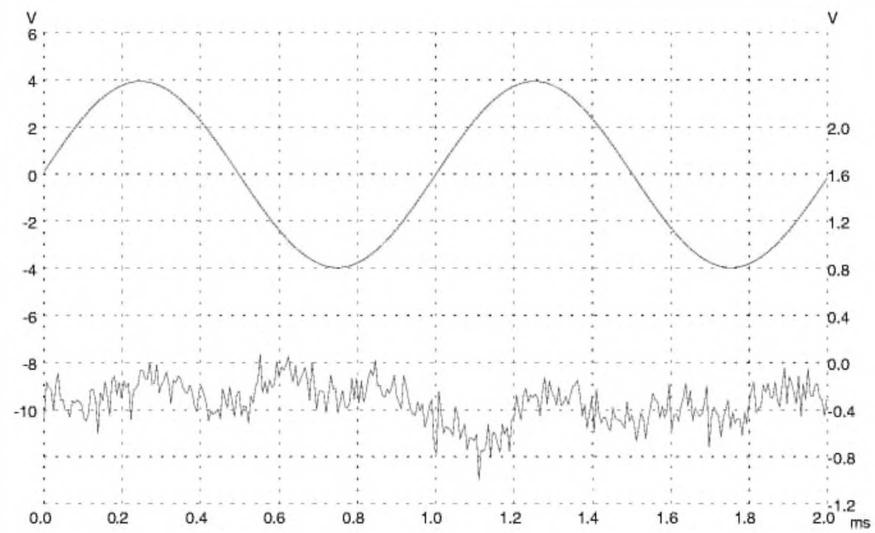

FIGURE 13: Residual distortion, 1kHz. A-1967-13

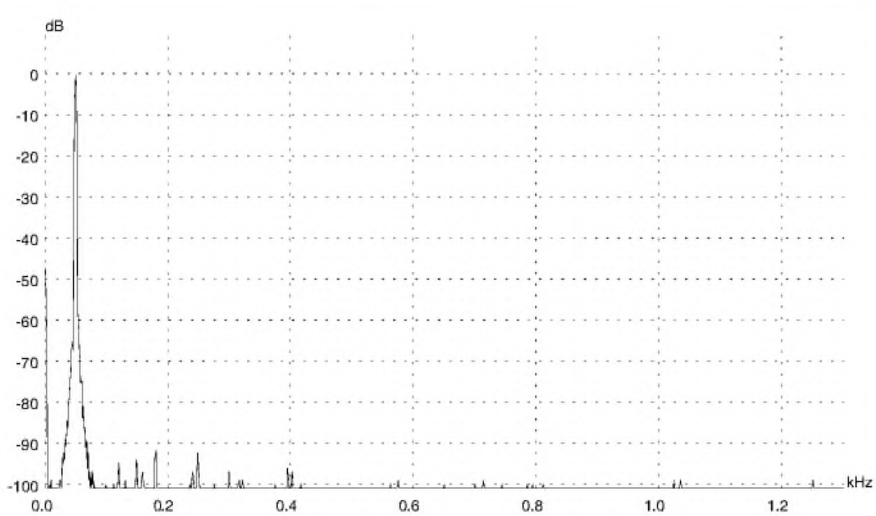

FIGURE 14: Spectrum of 50Hz sine wave. A-1967-14

amplifier is on the rear panel (*Photo 4*). *Figure 8* shows the lettering designations for my rear panel. I broke it into two sections since it is too long for a single 8½ × 11 sheet. I made a full-size copy of this lettering on drafting appliqué film, which is an adhesive-backed transparent plastic (see parts list).

The photocopy lettering is black, so I left the rear panel plain aluminum. Apply the lettered film to the panel in the proper locations. Cut through the film at the drilled holes and then spray it with two coats of clear polyurethane for protection. Then you can mount the components.

Mount the four snubber caps on each bridge rectifier. I took the easy way out, using just the four 10nF capacitors rather than calculating the optimum C-RC snubber[6]. You may also care to try low-noise, soft-recovery diodes, as recommended by Gary Galo[7]. I tapped power for the QuadPod board from the ±30V DC supply and stepped it down to about ±15V DC with series resistors (on the standoffs near CR1).

All circuit-board mounting holes have sufficient clearance for a 4-40 screw with flat washer. I used four ⅞″ long 4-40 tapped aluminum hex spacers to mount the board to the chassis. The speaker posts should be exactly ¾″ apart, with the terminals holes aligned vertically.

An easy way to do this is to mount the jacks in place finger tight. Plug in a dual banana plug to set the ¾″ distance. Bend a piece of 14-gauge solid bare wire (a stripped piece of Romex is fine) into a "U" shape so the legs are also ¾″ apart. Insert this wire through the holes in both adjacent connectors and tighten the mounting hardware.

Figure 9 shows the wiring to parts located off the circuit board. After installing the mechanical parts in the chassis, place the board near its final location and then wire to the switch, LED, jacks, and volume control if you decide to add one.

I used AWG-20 for the power supply and speaker wiring—two AWG-20 wires in parallel (equivalent to AWG-17) for the amplifier power ground and AWG-22 for the signal wiring. The easiest way to position the front and rear panels for wiring is to place them in the rails on the bottom plate and hold the three panels together with large rubber bands at each end.

The amplifier uses a single chassis star ground point, located near the fuse bracket. I ran separate ground wires from CR1, CR2, each PC board, the speaker (–) posts, and the RCA jacks to this point. Also connect the ground lug of the IEC connector J1 here as a safety ground.

I used insulators on the RCA jacks to isolate them from the chassis. Use shielded wire from the input jacks to the circuit board, with the shield grounded only at the jacks. The purpose of all these single-ended ground and shield connections is to prevent ground loops and noise currents on the shields, which could couple into the signal. Finally, after all wiring is complete, install the board in its final location. I used nylon wire ties to make the wire harnesses neater.

Since I also installed the QuadPod board in my chassis, there are additional twisted-pairs of wiring involved. Only one end of the ground lead in each twisted-pair is connected to a ground, at the jacks, or a circuit board. Again, this is to eliminate any ground-loop current.

The rubber feet that come with the chassis are kind of skimpy, so I purchased ¾″ square vinyl bumpers at a home-improvement center. I used five of them, with one under the heavy

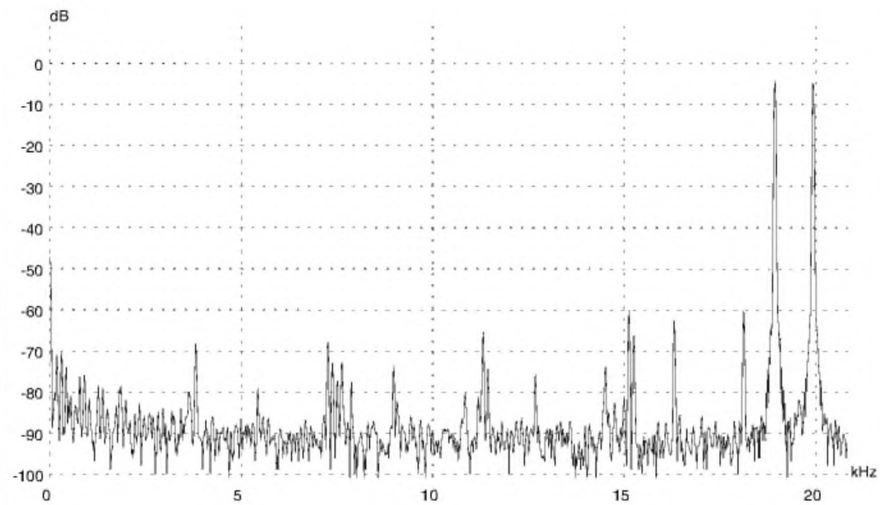

FIGURE 15: CCIF intermodulation response. A-1967-15

transformer so the bottom plate would not sag.

HEATSINK INSTALLATION

Once all the other construction is complete, you can install the heatsinks onto the LM3875 ICs. I used a TO-220 hardware mounting kit (see parts list) to insulate the heatsink from the IC. I did not use the mica insulator furnished with the TO-220 heatsink kit, replacing it with a K10 heatsink insulating pad (the tab of the 11-pin LM3875T package is connected to the V-supply).

Alternatively, you can use the LM3875TF package, whose mounting tab is electrically isolated from the IC. These devices are harder to find than the LM3875T, however. DigiKey does not carry them. You should be able to order them directly from National Semiconductor (www.national.com).

K10 is a thermally conductive Kapton polyimide coated with boron nitride resin. This material is thermally superior to mica, silicone elastomer and ceramic, and almost as good as beryllium oxide (which is brittle, and its dust is poisonous). In addition, the K10 does not need messy heatsink grease, since it is an elastomer that conforms well to the metal surfaces.

The heatsinks in the parts list do not have any mounting holes, so I drilled a ⅛″ hole in each heatsink about 1¼″ up from the bottom (see following) and centered side-to-side. You must carefully de-burr this hole on the side facing the IC to prevent damage and a possible short-circuit to the K10 insulator. I used a countersink for this purpose.

This hole must place the bottom of

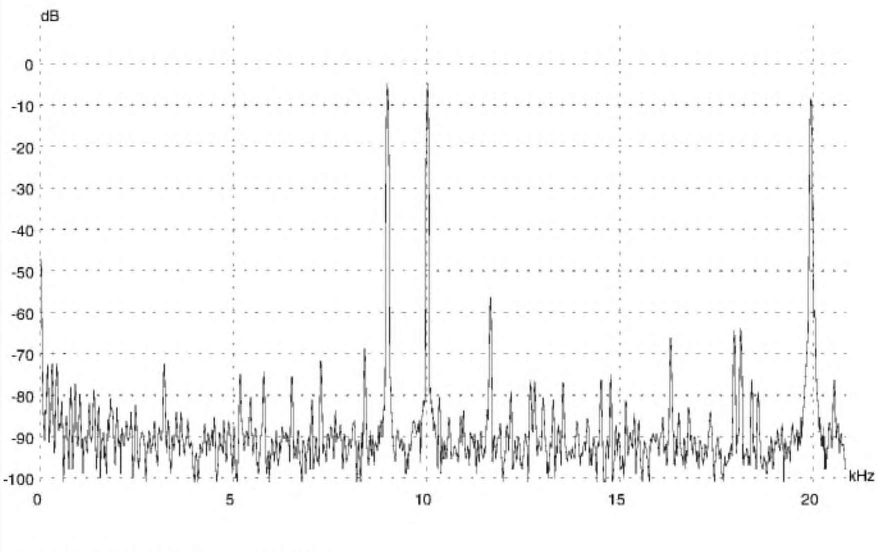

FIGURE 16: MIM intermodulation response.

the heatsink flush with the bottom of the chassis, so it is best to mount the PC board in position on its spacers and mark the center of the hole in the IC tab on the heatsink. Use this mark as the location for the hole in the heatsink. I also added a 4-40 flat washer under the lock washer and nut furnished with the heatsink kit (*Fig. 3*).

Lift the bottom center fin pin with needle-nose pliers, insert a ½″ long #6 sheet-metal screw through a hole in the chassis bottom plate, and screw it into the heatsink slot. Be careful not to put any force on the IC while mounting the heatsinks. The sheet-metal screw produces small metal chips as it cuts its threads, so be sure to turn the chassis upside down and dump out these chips, as well as any stray bits of solder and wire strands.

SOURCES

DK = DigiKey Corp.
701 Brooks Ave. South
Thief River Falls, MN 56701-0677
1-800-344-4539
www.digikey.com

Jameco Electronics
1355 Shoreway Rd.
Belmont, CA 94002-4100
1-800-831-4242
www.jameco.com

Old Colony Sound Lab
305 Union St.
PO Box 876
Peterborough, NH 03458-0876
603-924-6371
www.audioXpress.com

Mouser Electronics
958 N. Main
Mansfield, TX 76063
1-800-346-6873
www.mouser.com

PE = Parts Express
1-800-338-0531
www.parts-express.com

Plitron Manufacturing
8-601 Magnetic Dr.
Toronto, Ontario
Canada M3J 3J2
1-800-754-8766
www.plitron.com

Sescom, Inc.
517 Main St.
PO Box 839
Wellsville, KS 66092
1-800-634-3457
www.sescom.com

USING THE AMPLIFIER

As I mentioned in the beginning of this article, I use the LM3875 amplifier for my rear ambience speakers, and the QuadPod board drives it in my particular amplifier. The volume control sets the rear ambience level. The rear-channel outputs of the QuadPod are connected through switch S2 near the optional volume control to the line inputs of the LM3875 amplifier. The other side of this switch selects the external input, which bypasses the QuadPod and volume control.

LISTEN TO THE MUSIC

I did my listening audition using the external input jacks, placing the 25Wpc LM3875 amplifier below my 80Wpc NAD-214 front-channel stereo power amplifier. The NAD amplifier has a higher gain than the LM3875 (30.2dB versus 26.4dB into 8Ω), so I used a passive line stage equipped with input pass-through jacks to reduce the NAD's input signal so the levels were identical at 1kHz. This eliminated the "louder is better" psychological factor. Thus, I was then able to easily switch the speaker plugs from one amp to the other.

I used a number of CDs, including the HFN/RR Test Disc III (HFN020) that *audioXpress* has standardized for all their equipment reviewers. I listened with the full-range speaker system, as well as with just my satellite speakers alone. Rear-channel ambient information is almost devoid of bass, but is rich in the mid and upper audio frequencies. I also let it drive the subwoofer alone.

The LM3875 sounded a bit brittle or edgy at first in the treble, then became more transparent as it warmed up. In general, the NAD has a bit more soundstage and better imaging than the LM3875. It also has better bass extension. Of course, this compares a 4A integrated circuit with the paralleled 15A 130W discrete complementary bipolar pairs in the NAD (I use the NAD with its soft-clipping circuit turned off). The NAD's low-frequency rolloff point is also lower.

The bass with the LM3875 never got tubby or boomy, it was just a bit on the thin side, even with the satellites alone. (My speakers are not the most efficient at 86dB/2.83V/m, or maybe it's that extra 0R22Ω resistor I added?) Acoustic bass was sufficiently woody and resonant, and kick drums had a lively pop. It just didn't dig down to the lowest tones on a five-string electric bass as well.

The massed vocals in *Jerusalem* (Parry) were a bit less defined than with the NAD. This selection is a difficult test for any amplifier. The trumpets and strings in Vivaldi's *Trumpet Concerto in C* were very clear and bright. The only deficiency I could find vis-à-vis the NAD was a bit less definition in the harpsichord in the far right side of the soundstage. The male voice and individual instruments in *Peter and the Wolf* (Prokofiev) were very nicely reproduced by the LM3875.

Two tracks on Trudy Desmond's *Tailor Made* CD, "I'm Shadowing You" and "Make Someone Happy" (Jazz Alliance

TJA-10015), have a trombone and cello located closely together in the center of the mix. They interplay with each other in almost the same register, and sometimes with the upper notes of the acoustic bass. It's amazing how much alike these instruments can sound. The LM3875 had no problem separating them, providing adequate space around each instrument.

Next I listened to the big band in the Anne Hampton Calloway CD, *To Ella With Love* (Sin-Drome SD-8933). While the imaging and soundstage were again a bit less than with the NAD, the instruments were well defined. You could easily tell an alto sax from a tenor, from a baritone.

The initial transients of cymbals, triangles, and chimes were not as sharp as with the NAD, but they did remain clear and bright as their notes decayed into the low noise floor. The complex orchestration sounded full and natural, and her voice had a nice presentation.

The LM3875 is on a par with the better consumer audio equipment on the market. Not as detailed as budget high-end discrete amplification from Adcom, Creek, NAD, Parasound, or Rotel, but certainly respectable. At about $9 each, the LM3875 is one of the genuine bargains—in both cost and performance—in the audio IC marketplace.

MEASUREMENTS

I operated the LM3875 amplifier at 2W into 8Ω for one hour. The initial 0.03% THD reading dropped to 0.011% at the end of this run-in period. The distortion was essentially the same for each channel, so the right channel is presented here. There is a very faint thump during power-up and no noise at all at shutdown. The IC brings itself to life once the supply rails exceed ±7.3V DC.

The LM3875 amplifier does not invert polarity. Input impedance was 100k at 1kHz. The gain at 2.83V RMS output into 4Ω and 8Ω loads was 26.3dB and 26.4dB, respectively. The output impedance at 1kHz was 0.072Ω, increasing slightly to 0.078Ω at 20kHz.

The frequency response (*Fig. 10*) was within ±1dB from 8Hz to 41kHz, at an output of 2.83V RMS at 1kHz into 8Ω. When I connected a load of 8Ω paralleled with a 2μF cap (a test of compatibility with electrostatic speakers), the gain became extended at the high frequencies, peaking at 46kHz (dashed line). As I mentioned earlier, you can leave out R108 and R208 and enjoy the extra efficiency.

The IHF load, which simulates a loudspeaker impedance peak at 50Hz, produced a 0.2dB higher response at this frequency than the 8Ω resistive load alone. The LM3875 amplifier will be insensitive to variations in speaker impedance with frequency.

Output hum and noise (input shorted) measured 1.3mV, with −14mV DC offset. Crosstalk performance was limited by the close proximity of the two audio channels near S2 (*Table 2*).

THD+N versus frequency is shown in *Fig. 11* for the loads indicated at the right side of the graph. During distortion testing, I engaged the test-set 80kHz low-pass filter to limit the out-of-band noise.

Figure 12 shows THD+N versus output power for various loads and frequencies (in the order of the list on the right of the graph, when aligned with the 20W vertical line). There was absolutely no strain right up to the point of maximum power. The amplifier reached its 1% clipping point at 32W for all loads. With the 20Hz 4Ω load waveform on the scope, I could see the SPiKe protection activating intermittently.

I also plotted the 1kHz product of the multi-tone intermodulation (MIM) 9kHz + 10.05kHz + 20kHz test signal versus output power (dashed line). This gives a better indication of the LM3875's nonlinear response, since it is a closer approximation to music than a sine wave.

The distortion residual waveform for 10W into 8Ω at 1kHz is shown in *Fig. 13*. 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. This distortion residual signal just barely shows the third harmonic, overlaid with noise. THD+N at this test point is a very low 0.005%.

The spectrum of a 50Hz sine wave at 10W into 8Ω is shown in *Fig. 14*, from zero to 1.3kHz. The THD+N here measures 0.0055%. The second, third, fourth, and fifth harmonics measure −100dB, −94dB, −104dB, and −92dB, respectively. Low-level power-supply rectification artifacts are also present at 120Hz, 180Hz, and 240Hz, all below −90dB.

Figure 15 shows the amplifier output spectrum reproducing a combined 19kHz + 20kHz CCIF intermodulation distortion (IMD) signal at 12V pp into 8Ω. The 1kHz IMD product is 0.012%. Repeating the test with a multi-tone IMD signal (9kHz + 10.05kHz + 20kHz, shown in *Fig. 16*) resulted in a 1kHz product of 0.011%. (I plotted MIM distortion versus output in *Fig. 12*.) This fine performance did not change when I reduced the load to 4Ω.

A 2.5V p-p square wave into 8Ω at 40Hz showed the expected LF tilt. The 1kHz square wave was just about perfect, while the edge of the 10kHz square wave was slightly rounded. When I connected 2µF in parallel with the 8Ω load, the leading edge of the 10kHz square wave showed underdamped ringing at the 46kHz peak measured in the frequency-response test. ❖

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The Circlotron Amp

As a follow-up to a discussion of driving low-mu triodes, this author investigates the Circlotron amp configuration. **By John L. Stewart**

A response to my articles in *GA* 2/99 and 3/99 covering a different way of driving low-mu triodes got me thinking—about using a Circlotron amplifier. I have never had experience with this configuration and was curious as to why it would be used.

CIRCUIT FEATURES

The Circlotron has some important advantages. The power tubes are connected in such a way to drive the load in parallel rather than in series, as they would in an ordinary push-pull circuit (*Fig. 1*).

For a given power output in watts, one-half the voltage at twice the current will produce similar results. If you do the arithmetic with Ohm's Law (R = E/I), you get the following result:

$$R\ (load) = 0.5E / 2I = (1/4)\ \#R$$

This would let you use an output matching transformer having one-quarter the impedance normally used in a push-pull amplifier. That transformer will produce good results much easier than a high-impedance unit.

If you apply this to an amplifier using the 6AS7/6080 family, you would use a matching transformer of 1kΩ rather than 4kΩ. Since the load is in the cathode circuit, the resulting output impedance (not the same as the matching impedance) will also be very low. In a cathode follower, output impedance is reduced by the factor:

$$1 / \mu + 1$$

Again using the 6AS7/6080, the published plate impedance is 280Ω, giving you a source impedance of 93Ω as a cathode follower. Since you have a pair of these driving the load in parallel, the resulting source impedance becomes 46.5Ω.

The damping factor (DF) is the load impedance divided by the source impedance. In this example, it is 1000/46.5, or 21.5, which is a very high result if you take into account that you have not applied overall feedback. However, that is not the complete picture.

The output transformer will have some resistance in both its primary and secondary. That will somewhat increase the source impedance as seen by the load. In a practical circuit you could expect a damping factor of about 10 without external feedback. Other factors influencing the DF factor will be whether you use triodes or pentodes and what kind of driving scheme is ap-

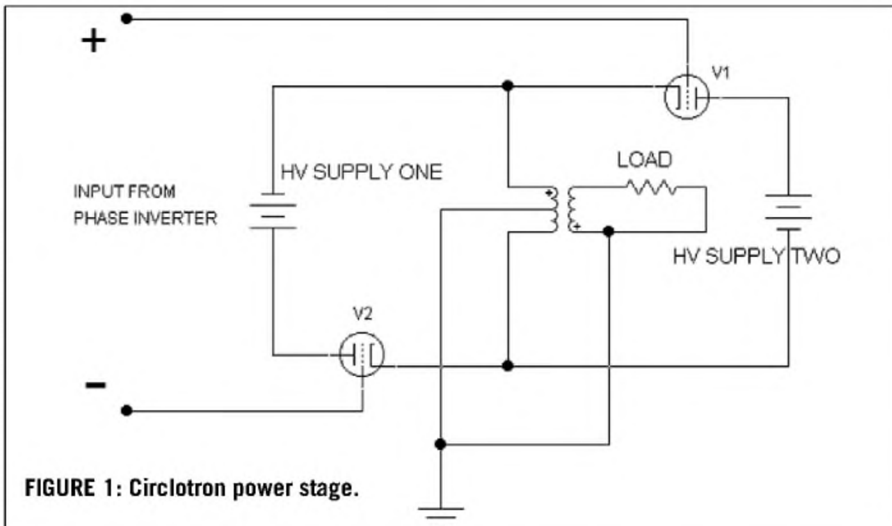

FIGURE 1: Circlotron power stage.

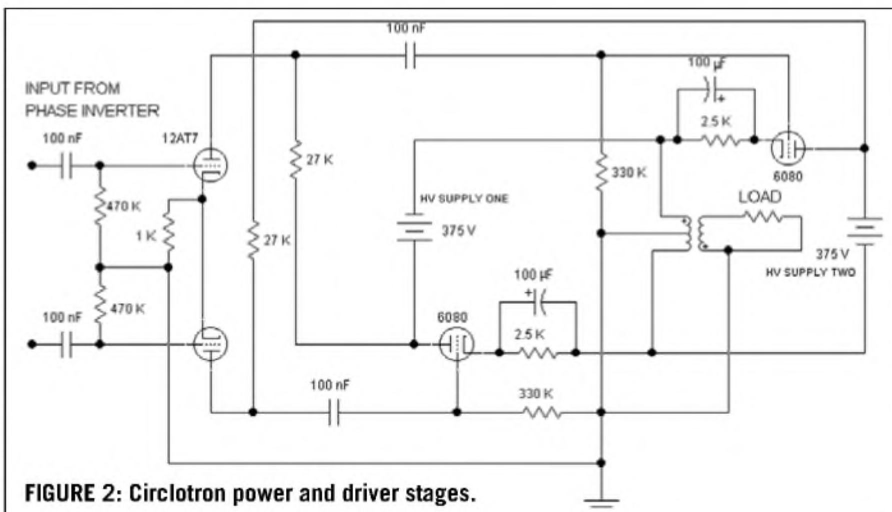

FIGURE 2: Circlotron power and driver stages.

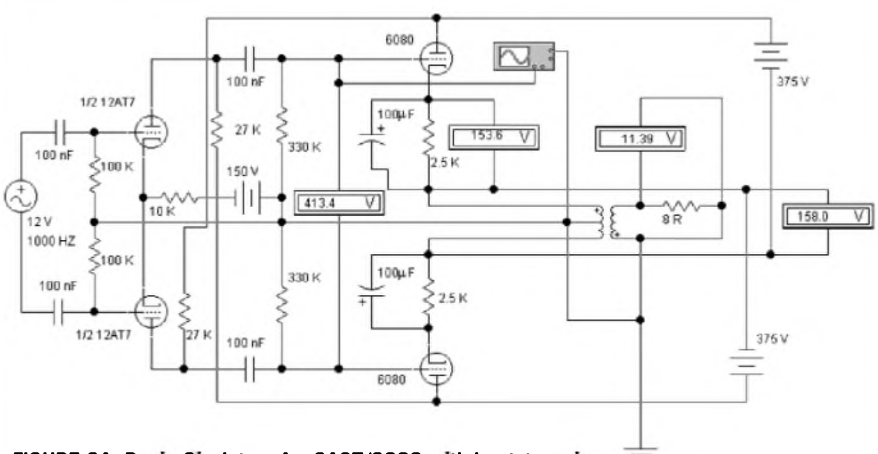

FIGURE 3A: Basic Circlotron for 6AS7/6080 with bootstrapping.

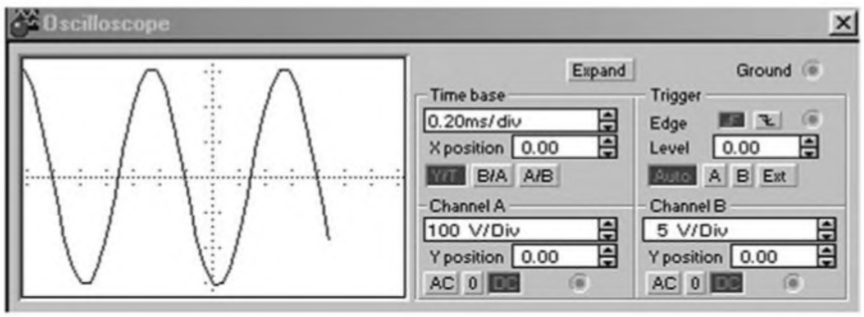

FIGURE 3B: Scope display of *Fig. 3A*.

plied. In recent work, I've found the final DF could be as low as two without external feedback.

Another advantage of this circuit is that it is completely symmetrical. You might counter that so is an ordinary push-pull circuit, and it is. However, you can match the Circlotron to a relatively low output impedance at close-to-ground potential.

You now have the possibility of using a number of power tubes in a push-pull parallel combination to drive even lower impedance loads. That makes it possible to drive a loudspeaker load directly without the help of an output transformer. You've avoided the asymmetry associated with the series push-pull (Futterman)[1] system.

WHY DON'T WE ALL USE IT?

With all these advantages, you would think that everyone would build vacuum tube amplifiers using this great topology. However, there are some problems.

First of all, you have probably noticed by now that I have shown two high-voltage power supplies in the simplified schematic. That's not a mistake. If you were to run a pair of the 6AS7/6080s in this circuit, you would need two of the 375V power supplies electrically isolated from each other (*Fig. 2*).

Another problem is how to provide enough driving voltage to the output stage. After all, it is connected as a cathode follower and has no voltage gain. The power gain is the same as it would be if the load were connected in the plate circuit. In practice you will need a drive of more than 2× the expected output voltage at the cathodes.

The driver stage is normally assisted by bootstrapping. *Figure 3A* is a simulation of a bootstrapped circuit using software by Electronic Workbench. I used a 12AT7 in the simulation to provide drive to the output tubes. It is connected as a differential amplifier with its cathode resistor returned to a 150V negative power supply, showing that a drive of 413.4V is required grid to grid of the output tubes to get 16W. That is 11.39V in 8Ω. The oscilloscope connected to one of the output tube grid leads has just started to show some clipping (*Fig. 3B*).

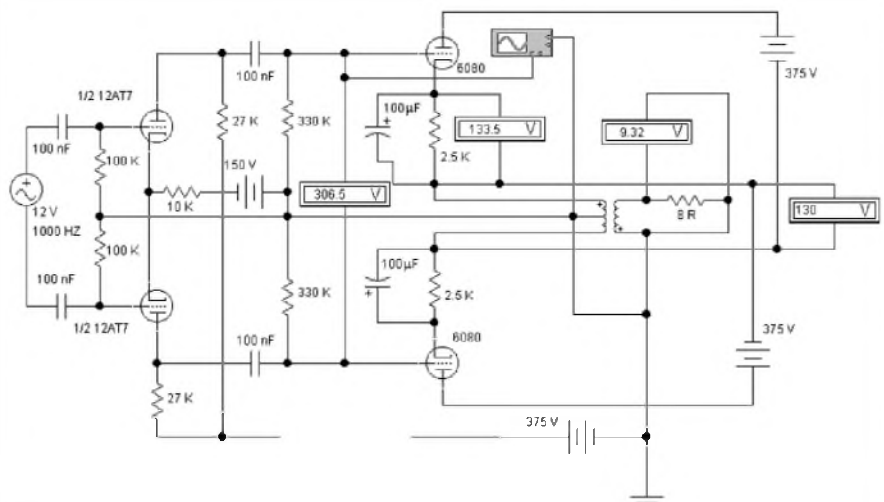

FIGURE 4: Basic Circlotron for 6AS7/6080 without bootstrapping.

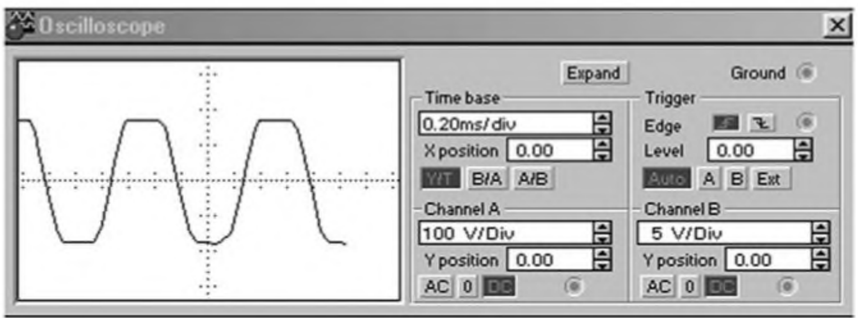

FIGURE 4B: Scope display of *Fig. 4A*.

Figure 4A shows the same circuit simulation, but this time the bootstrapping is unhooked. The 158.0V of signal developed between the output tube cathodes in the bootstrapped circuit is not available to assist in driving the output grids. There is much clipping (*Fig. 4B*).

To get results similar to those in the bootstrapped circuit, you would probably need a 500V or more power supply for the driver. This is a good example of transferring the problems in an amplifier from the output stage to the driver.

BOOTSTRAPPING

Some audiophiles are concerned that bootstrapping, because it is positive feedback, will cause problems for their circuit. They may be surprised to discover that one of the benefits of the circuit is the same as that found in the mu-follower. The bootstrap connection increases the load impedance seen by the driver stage in proportion to gain in the output stage. That has the same effect in lowering second harmonic distortion in the driver as it does in the mu-follower.

In this example of the Circlotron, the driver stage uses 27k loads. Because of the bootstrapping the load they will see is 1.62 times (44k) that. In the McIntosh the driver sees about 4× the load resistors used. I measured the effect of the bootstrapping in my version of its use and found one-half of the required drive was provided by the bootstrap. In that case the load resistors for the driver were 27k and would appear to be 54k to the driver.

The most important advantage of the bootstrapped driver is that it has a much larger output voltage capability. Without the bootstrap you would need to build a special high-voltage driver. That is why it is used by McIntosh and

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Box Stuffing–A Fistful of Fuzz

The function of stuffing in your speaker cabinet depends on application and box design. **By Ron Horowitz and John Schaffer**


PHOTO 1: AR's famous "Bee Butt" midrange.

With the exception of speaker cables, perhaps no other audio component carries more baggage than cabinet stuffing. Part of the confusion stems from a misunderstanding of various requirements for different and diverse applications. Some materials work well in most cases, while others provide more psychological comfort than acoustical benefit.

Likewise, costs, manufacturability, and stability must be considered. Sorting wisdom from witchcraft starts with understanding the various benefits of box stuffing, which can vary with enclosure type. When the correct type of stuffing is selected and properly applied, it can slow down the speed of sound, increase the acoustic compliance, and attenuate the midband while enhancing the low frequencies (*Figs. 1* and *2*).

WHAT'S GOING ON BEHIND THAT SPEAKER CONE?

About two years ago, while working on a project at the Dai-Ichi speaker plant in Manila, our associate, Mike Klasco, was asked to comment on the tweeter impulse-response printouts from a newly constructed tweeter test box. It was apparent that the ragged, extended decay time could not *possibly* be attributed to the tweeter, a balanced-drive unit of advanced construction. Mike knew the test mike had superior impulse response, as does the Bell Labs SYSid test and measurement system. So, new test box's acrylic-fluff "anechoic stuffing" was suspect.

Noticing the pink building insulation covering the ceiling of the plant, Mike had the acrylic box stuffing replaced with that. After installing the fiberglass in the test box and re-measuring the tweeter, Mike saw that the long decay time from the previous plots was gone. It had merely been an artifact of the acrylic fluff's poor midrange absorption!

Imagining that the simulated response of his design is adequate, the rookie speaker engineer, armed only with a low-frequency modeling program, might be tempted to omit box stuffing altogether. After all, no one sees this stuff anyway! Veteran engineers know better. Midrange roughness and impulse response signatures result when you omit enclosure stuffing.

If half the energy coming off a speaker's cone is emanating from its backside, the "back wave" energy must go somewhere. Though enclosures such as the bass reflex utilize the low-frequency portion of the "back wave" component constructively, enclosures such as the infinite baffle just wish it would all go away. In any case, box modes must be suppressed, a necessity that is increasingly overlooked.

At Menlo Scientific we are often contracted to do audio design for consumer electronics. A recent undertaking was a stand-alone Internet radio (no computer required), from which the best possible sound quality per dollar was desired. Although we'd specified box stuffing in the design, we caught the Chinese subcontract factory leaving it out. While the cost savings was negligible, the performance degradation was dramatic!

The omission of box stuffing in less-expensive, compact audio products is a disturbing trend, indicating a clear misunderstanding of the importance of this component. While enclosure stuffing is generally understood to influence the low-end response, its effects on midrange clarity aren't as widely appreciated. Especially in the case of small satellite enclosures (with woofers crossed over high into the midrange), a lot of midrange energy is dumped from the back of the cone.

THE FUNCTION OF BOX STUFFING FOR VARIOUS TYPES OF ENCLOSURES

So, which stuffing is best for which application? Let's look at enclosure design requirements, survey the designer's options, and discuss price and performance.

SEALED ENCLOSURES

Absorptive stuffing in a sealed box serves a number of functions. It increases the acoustical compliance, low-

ABOUT THE AUTHORS
Ron Horowitz and John Schaffer are employed at Menlo Scientific, Ltd., which serves as a consultant to a number of loudspeaker firms.

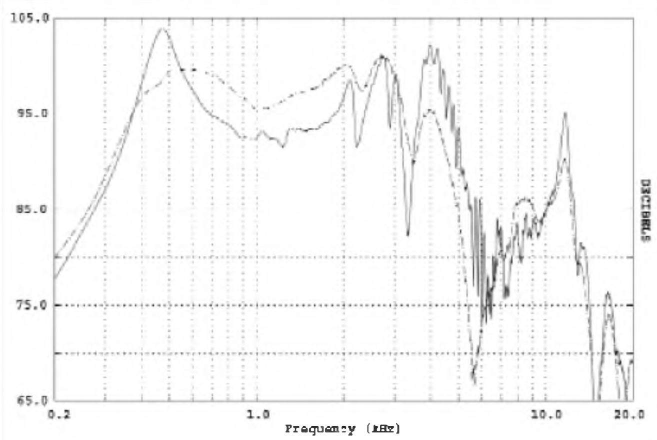

FIGURE 1: Frequency response without stuffing (solid) and with stuffing (dash).

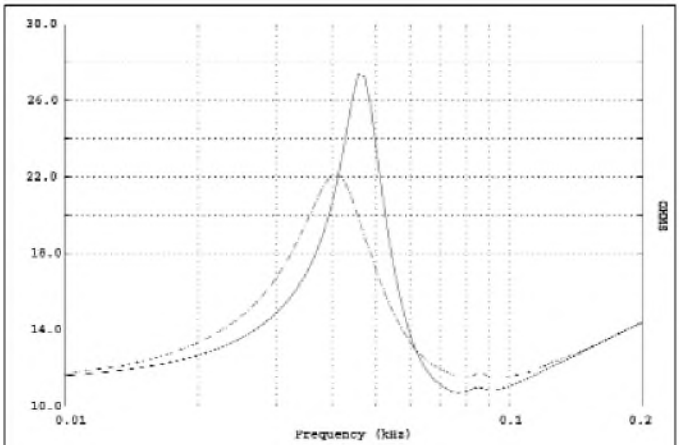

FIGURE 2: Impedance without stuffing (solid) and with stuffing (dash).

ers the woofer's in-box resonance, adds resistive damping, and lowers the system's Q. Absorption at higher frequencies suppresses box modes, smoothing upper frequency and transient responses. To a point, increasing stuffing density enhances all three effects.

Since stuffing costs can become significant with large acoustic suspension designs, it's left to the engineer to justify the cost. All too often, out of sight is out of mind as far as the corporate "bean counters" are concerned. On the other hand, accountants don't know anything about capacitors and inductors, either, and I understand a few speakers have been made with those! It must be made known that adequate damping is essential to optimal system performance, so stuffing is not a component to be omitted at the whim of a corporate know-nothing.

PORTED AND FASSIVE RADIATOR ENCLOSURES

Generally, ported boxes are merely lined with an absorptive boundary layer to help dissipate midrange energy and suppress box modes. Following this approach, low-frequency absorption is negligible. Occasionally, one portion of a ported enclosure is lined, while another portion is stuffed.

With this approach, the acoustical

compliance of the enclosure can be increased, though you may observe low-frequency damping effects. Regardless, unlike the obsolete aperiodic enclosure type, stuffing should not befoul the port and upset the port tuning. Since the largest possible port area is desired to keep port velocities low, it appears that modern, linear-compliance passive-radiator systems might have a slight advantage here, especially for two-way systems using a high crossover point.

BANDFASS

Designers of bandpass enclosures often omit absorptive materials altogether, not to mention crossovers, presuming the acoustical filter nature of the design will suffice. While this expedient has advantages in low-cost designs, nothing precludes the judicious application of box stuffing in a bandpass design. Ported sections can be lined, whereas compliance and damping effects should be considered in sealed bandpass sections.

ACOUSTIC LABYRINTHS AND TRANSMISSION LINES

The function of stuffing in a transmission line is two-fold: it lowers the speed of sound, and it absorbs midrange energy. By slowing the transmission speed, a line can be mechanically shortened, while retaining the same tuning frequency. Ideally, the stuffing would pass low-frequency energy, which can be used to enhance low-frequency output while suppressing box modes at upper frequencies. A boundary layer of stuffing lining the walls provides this function, though relatively little slowing of the transmission speed occurs, as compared to a fully stuffed line.

BASS HORNS

The back chamber of most bass horns can be treated like a sealed box, providing loading below the horn cut-off. (In the horn's bandpass range, the rear chamber has other interactions.) In his Pataxial (coaxial horns) loudspeaker, Dr. Eugene Patronis of Georgia Tech lined the walls of the low-frequency horn with Sonex, and, while developing hyperbolic (high Q) bass horns in the 1970s, Richard Long found that an effective means of damping resonances was to place a little bit of stuffing in the horn's throat. Of course, if the air column in a long, slowly expanding hyperbolic horn is stiffer than the diaphragm of its driver, the best place to put the acoustical stuffing may be in the listener's ears!

TWEETERS AND MIDRANGE DRIVERS

In order to smooth the frequency and time responses of dome-type midrange and tweeter units, the space behind the dome is often filled with stuffing. With this application, in addition to acoustical performance, you should consider mechanical and thermal issues. The stuffing will do more harm than good if it shifts out of position, or worse yet, if it melts or burns under high-power operation. Fortunately, because the volume of stuffing used is so small, material cost is of minor importance.

Though currently out of vogue, resistive damping of dome-type midrange drivers (achieved by positioning stuffing in front of the dome using an expanded metal cover) has been used in a number of commercially successful products. The Acoustic Research "bee-butt" midrange (*Photo 1*) drivers (so-called because they resembled a bee's butt), as used on the AR3 and the more recent AR 303, are examples of this approach. A remarkable smoothing of the response is the primary benefit, and cosmetic and mechanical concerns are the primary drawbacks. This method works well with cone midrange units with B&W's Nautilus being a shining example.

TYPES OF STUFFING CHOICES

There are two main categories of stuffing, foams and fibers. Each type has advantages and disadvantages.

FOAMS

Open-cell polyether and polyester foams have similar properties, in which foam density and cell structure are related to acoustical performance. The primary advantage of foam is ease of handling. Foams can be cut to order (hot wire or blade) in fixed, non-friable geometries, making for ease of warehousing, inventory, handling, assembly, and so forth.

Foams tend to lack low-frequency absorption compared to fibers, though, and are of limited benefit in sealed enclosures at low frequencies. They can deteriorate when exposed to UV, fungus, ozone, and the like, but these don't tend to be issues in sealed or passive radiator systems, where foam surrounds may go first! Small, die-cut foam buttons are often used under the diaphragms of dome tweeters and midrange drivers, where a non-friable, stable geometry is desired.

FIBERS

The major fiber groups used in acoustics include acrylics, wools, and glass fibers. Fiber length, texture, diameter, flexibility, and packing density are the major acoustical variables for a given fiber type. Optimal fiber geometries give best results, though such attention to detail increases costs over generic types.

Wool is an expensive fiber. On the other hand, when properly applied, it does have excellent acoustical characteristics. This is the "cost no object" choice for the ultimate audiophile, Jon Risch of Peavey Electronics.

While the long-fiber wool of the female sheep of the Outer Hebrides is the hands-down cult favorite (sheared in late autumn after a hard rain), it has many practical limitations. The fibers bunch up, it's difficult to get consistent packing density and volume, it's dimensionally unstable, and so on. As a practical stuffing material, it's a non-starter. Besides, who wants to make cedar-lined speaker enclosures?

Fiberglass is a low- to medium-priced fiber with good to excellent performance. Fiberglass, specified in thickness and density, is generally used precut in sheet form. While semi-rigid fiberglass is available, it generally lacks the non-friability of foam. Glass fibers are fairly brittle, especially when large cross-section fibers are present, and, unlike wool or acrylic fibers, are prone to fracturing.

As anyone who's ever worked with fiberglass can attest, its primary disadvantage is adverse worker reactions such as itching and allergic reaction. One of the better low-cost, "reduced itch" fiberglass types available in Asia is Owens Corning's TIW-20. This brownish fiber serves double duty as an oven insulation! In the US, Owens Corning's "Think Pink" thermal insulation is sometimes used, as is its yellow counterpart, which can be ordered in various densities and semi-rigid sheets.

Acrylics are medium-priced fibers with good acoustical performance, depending largely on fiber geometry and packing density. Unlike fiberglass, acrylic fibers are stable and non-allergenic, with no special handling issues. Acrylics come in sheets for lining, or in bulk for filling, and occasionally in small buttons of acrylic fiber for back-of-dome applications.

Select Sound Acoustic Core is a relatively new fiber technology from Owens Corning. Acoustic Core is an engineered glass fiber product that specifically addresses the issues of high acoustical performance. Select Sound fibers were designed using two different fiber types wrapped around each other with a random twist and a curved fiber shape.

After Mr. Risch's explanation of the long-fiber wool, you could almost describe Acoustic Core as a synthetic version of the wool. The fiber diameters were selected to afford a "non-stinging," reduced-itch fiber. Likewise, the resin was selected to minimize allergic reaction.

Select Sound has superior acoustical characteristics at a fraction of the cost of long-fiber wool. It is an engineered product, so it's consistent in its critical parameters and availability, unlike wool, which was designed originally to keep sheep warm. Select Sound is fairly resilient and springy, too, so it doesn't tend to bunch up as much as long-fiber wool, and it's available as a white, unfaced, loose fiber in banded, 275 lb bales.

CONCLUSION

Often regarded with confusion or indifference, box stuffing is an essential, if misunderstood, component. Understanding various applications, available materials, and necessary functions is the first step toward superior product design. ❖

All About Wire

This article discusses some of the different types of wire commonly used in audio equipment, and provides some practical advice about when to use each type. **By Pete Millett**

PHOTO 1: Solid (left) and stranded wire with the insulation removed.

PHOTO 2: Different wire sizes. From left: 12AWG, 16AWG, 24AWG, 30AWG.

On the surface, the function of wire seems pretty simple: get electrons from point A to point B. But there are many choices to be made when wiring audio equipment. A multitude of different wire types, sizes, and materials is available to the builder.

THE SINGLE CONDUCTOR

Single-conductor wire is what most people think of as "normal" wire. The single metal conductor can be made of one piece of metal called "solid" wire, or from many smaller wires wrapped together, which is called "stranded" wire. Single-conductor wire may be bare metal (un-insulated) or covered in a nonconductive material. (I discuss the materials used for the conductor and insulation in more detail later.)

Electrically, there is not much difference between solid and stranded wire. The main difference is mechanical. Solid wire is, well, solid, while stranded wire is more flexible. *Photo 1* shows solid and stranded wires with the insulation removed.

In old-fashioned point-to-point wiring, like that used in tube equipment, solid wire is generally used. It is easier to terminate to solder lugs (like those on tube sockets) since it doesn't fray out into a million little wires, at least one of which will never go into the hole you choose. Solid wire also holds its shape, so you can dress it into position and it will stay there.

On the other hand, stranded wire is more flexible, especially in larger sizes. It also survives being bent back and forth without breaking, so any wiring that involves motion should be done with stranded wire, which is also used when the wire is to be terminated into crimp or insulation displacement (IDC) connections. Solid wire doesn't generally make a reliable connection this way except in very small wire sizes and with connectors specially designed for it (such as telephone connectors).

A typical piece of audio equipment may use both solid and stranded wire. For example, AC power wiring is usually done with stranded wire, since it is more flexible in the larger size required for AC wiring, and can be terminated with crimp terminals. Point-to-point wiring, such as between tube

ABOUT THE AUTHOR

Pete Millett started playing with tubes as a kid, and has been an electronic design engineer for about 20 years. His experience ranges from consumer electronics to computer hardware and system design. He also runs Wheatfield Audio, a small manufacturer of tube headphone amplifiers. Pete can be contacted by e-mail at pete@headphoneamp.com.

TABLE 1
COMMON WIRE SIZES AND THEIR CHARACTERISTICS

AWG	DIA. (INCHES)	AREA (SQ. INCHES)	RESISTANCE (MILLIOHMS/ FOOT, COPPER 20°C)	RESISTANCE (MILLIOHMS/ FOOT, SILVER 20°C)	MAX CURRENT, AMPS (BARE COPPER WIRE)
4	0.2043	0.03278	0.2485	0.229	199
6	0.162	0.02061	0.3952	0.364	125
8	0.1285	0.01297	0.6281	0.579	79
10	0.1019	0.00816	0.9988	0.921	49.6
12	0.0808	0.00513	1.59	1.46	31.2
14	0.0641	0.00323	2.52	2.33	19.6
16	0.0508	0.00203	4.02	3.7	12.3
18	0.0403	0.00128	6.39	5.89	7.75
20	0.032	0.000804	10.1	9.34	4.89
22	0.0253	0.000503	16.2	14.9	3.04
24	0.0201	0.000317	25.7	23.7	1.93
26	0.0159	0.000199	41	37.8	1.2
28	0.0126	0.000125	65.3	60.2	0.761
30	0.01	0.0000785	103	95.6	0.477
32	0.008	0.0000503	162	149	0.304

TABLE 2
COMMON AUDIO EQUIPMENT APPLICATIONS OF DIFFERENT WIRE SIZES

WIRE SIZE (AWG)	APPLICATION
24–26	Low-level audio signals, power wiring <500mA
20–22	Power wiring <1.5A
16–18	AC line wiring (usually stranded) <5A, speaker wiring <40W, power wiring <5A
12–14	AC line wiring (usually stranded) <10A, speaker wiring >40W, power wiring <10A

sockets or terminal strips, may use solid insulated wire.

For under-chassis wiring inside a typical audio project, the choice between solid and stranded wire is mostly a matter of builder preference. Either one works fine.

WIRE SIZE

Wire is manufactured in sizes as thin as a hair to several inches in diameter. The size affects its electrical characteristics, such as resistance and ability to pass large currents, as well as its mechanical characteristics. Smaller wire has a higher electrical resistance, and very small wire is difficult to work with. At the other extreme, large wires take up space, are difficult to bend and fit into chassis, and are expensive.

Wire size is commonly specified with a number called "gauge." "American Wire Gauge," or "AWG," is the standard normally used for electronic wiring. The larger the number, the smaller the wire. *Table 1* shows the diameter and characteristics of a range of wire sizes that covers most of what is used in wiring normal electronic equipment. Note that commonly available wire sizes are only in even numbers; the odd-numbered sizes do exist, but are not commonly found. *Photo 2* shows differently sized wires for comparison.

Two factors influence what size of wire you use inside a piece of electronic equipment. Electrically, a wire needs to be large enough so that its resistance is low enough not to influence the circuit or be heated significantly by the power dissipated in the resistance. Mechanically, you need a wire that is neither so small that it's fragile and hard to terminate, nor so big that it is difficult to bend into shape.

For low-level signals, where the current is measured in milliamps, the wire size is chosen more for ease of use than for electrical characteristics. Usually, a size between 22AWG and 28AWG is chosen somewhat arbitrarily.

For power wiring (both power-supply and high-level signals like speaker wiring), you must choose the wire size based on the amount of current that must be passed. There are many rules-of-thumb for determining the current capacity of a given wire size—some based on temperature rise, others on voltage drop. Generally, it is best to be conservative and use a wire size that is a couple of sizes bigger than what is absolutely necessary.

The maximum currents listed in *Table 1* are based on temperature rise of bare wire. Consider these absolute maximums for wiring in audio equipment, and it would be best to use wire two sizes larger than this minimum.

BLACK MAGIC

Some audiophiles seem to have a nearly neurotic obsession about wire. The science and engineering says, within reason, wire is wire. Yet, there is a proliferation of exotic materials and technologies: from silver hook-up wire and silver-wound transformers to long-grain ultra-pure oxygen-free copper wire with some exotic polymer insulation. So, what's up with this?

Though I am an engineer by profession, I've learned not to dismiss all of this "black magic" as hallucination and marketing hype. As much as engineers and scientists hate to admit it, there are things that can be heard but not measured.

Can I hear the difference between identical tube amps—one wired with silver hook-up wire at $20 per foot, and one with Radio Shack copper wire? Nope. But I'm not quite willing to say that nobody can. If there's one thing I've learned in this business, it's to keep an open mind.

Things become even more complicated when you start looking at speaker wire and interconnect cables, which can cost you anywhere from a couple of bucks up to thousands of dollars. There are certainly measurable and audible differences between cables, but, to my ear, they don't necessarily relate to the price tag.

I've heard so much discussion about how solid wire sounds so much better than stranded wire (something about "strand interaction?"), and how PVC insulated wire sounds better than Teflon® (and vice-versa), and how you just have to use silver solder, that it makes my head spin.

So how do you resolve all this? Well, you don't. Build and buy what sounds good and makes sense to you. Experiment and make your own decisions—don't be swayed by the magazines and the marketeers.—**PM**

Table 2 lists some common wire sizes used for hook-up wire in audio equipment.

CONDUCTOR MATERIALS

Almost any metal can be used to construct wire, as can combinations of more than one metal. For example, wire designed to be strung between telephone poles sometimes uses a steel core for strength, with copper layered around it for good electrical conductivity. In the wiring of electronic equipment, though, the electrical characteristics of the material are more important than mechanical characteristics, so they determine what metal is used.

Of the commonly available metals, the best electrical conductor is silver, followed by copper and gold. Gold is far too expensive to consider for electronic wiring—you'd need to be a very wealthy person (not to mention one in need of psychiatric help) to wire a project using gold wire!

Silver wire is sometimes used in high-end audio projects. Although it is a better conductor than copper, its resistance is only about 8% less than copper, which is much less expensive. Even though from an engineering perspective there is no reason to use silver as a wiring material, many audiophiles swear by it (see sidebar).

So, copper is the material of choice for almost all electronic wiring. However, there are different types of copper as well!

Pure copper is a very soft metal, so it is often alloyed with other metals to alter its mechanical properties. There is also always some level of impurities left in the metal, so even "pure" copper has some amount of other metals—and nonmetallic materials—in it. As far as the electrical properties are concerned, purer copper is better.

There has been much discussion about "ultra-pure" and "oxygen-free" copper being needed to wire audio equipment. Certainly, impurities such as copper oxides are detrimental to the electrical properties of the wire, but they are very difficult to quantify.

Copper's problem is that it reacts with air and forms an oxide layer on its surface, which turns the shiny copper color to a dull brown. Copper oxide is not a good conductor, and it can make connections and solder joints work poorly. Because of this, most wire used in electronics has another metal—usually either tin or silver—plated on the outer surface of the wire. This plating dramatically improves the solderability of the wire and reduces the chance of a connection becoming open over time. Unless you have a specific reason to use unplated wire, you should always use wire that has tin or silver plating.

INSULATION

Most wire used in electronic equipment has a coating over the metal to insulate it, or to prevent it from making electrical contact with other wires, a chassis, or you!

There are applications that use uninsulated wire. Short connections, such as between adjacent lugs on a tube socket, are often done with uninsulated wire, which is fine, as long as the connection is short enough that there is no danger of the wire bending and making contact where it's not supposed to. Likewise, grounded connections, such as bussing together the grounded side of a bunch of phono jacks, are often done with uninsulated wire, which is commonly called "bus" wire.

Many different materials are used to insulate electronic wire. The two basic parameters to consider in an insulation material are its breakdown voltage (how high a voltage the insulation can withstand before failing), and its temperature rating (how hot the insulation can become before it melts or otherwise breaks down).

Various plastic and rubber materials have been used for hook-up wire in electronic equipment over the years. Today, for electronic wiring, the most commonly available insulation materials are PVC (polyvinyl chloride) and Teflon®. For special applications requir-

TABLE 3
COMMON UL AND MIL WIRE TYPES

WIRE TYPE	INSULATION	APPLICATION
UL1007	PVC, 300V, 80°C, 0.016″	General use hook-up wire
UL1015	PVC, 600V, 105°C, 0.031″	Higher voltages and temperatures—thicker insulation
UL1429 MIL-W-16878/1	Irradiated PVC, 600V, 105°C, 0.010″	Thinner, tougher insulation, resistant to soldering
UL1213 MIL-W-16874/4	Teflon®, 600V, 200°C, 0.010″	High reliability, high temperatures
MIL-W-16878/5	Teflon®, 1000V, 200°C, 0.014″	High voltage, high temperature
MIL-W-76B	PVC, 1000V, 80°C, 0.016″	High voltage

ing very high voltages, temperatures, or flexibility, there are many other materials available as well, but they are seldom used by the hobbyist.

The most common type of electronic wire, usually called "hook-up" wire, is normally insulated with PVC, which is a good—not great—electrical insulator, and is inexpensive. It's easy to work with because it strips easily. Its biggest downside is that it doesn't withstand high temperatures well. It melts easily when soldering, so you must be careful (and quick) when soldering connections. It comes in different thicknesses, which are rated for different voltages. Common PVC hook-up wire is rated for 300 or 600V, which is adequate for everything except some tube circuits.

The other type of wire used often in audio equipment is Teflon® insulated wire. Teflon® is a better electrical insulator than PVC and can withstand much higher temperatures. It is also very resistant to flame, and for these reasons Teflon® insulation has always been the preferred material in high-reliability and military applications. Teflon® has a distinctive slippery feel to it, and is usually shiny, as opposed to PVC, which is duller.

Teflon® insulated wire has a couple of disadvantages. It tends to be expensive (though in the quantity you would need for a typical audio project, the cost is probably not too great a deterrent), and it is also harder to strip the insulation from the wire. Unless you use a good, sharp wire stripper, Teflon® tends not to cut cleanly and to draw out in long stringy pieces.

So, which one should you use? It really comes down to personal preference. I like the fact that Teflon® doesn't melt easily, so you can solder a connection without worrying about the insulation, and you can run the insulation right up to the solder joint. But it's hard to find Teflon® wire in any type other than the silver-plated, stranded wire commonly used in military applications—if you prefer solid wire, it may be a bit more difficult to find.

Table 3 lists several commonly available types of insulated wire and their military (MIL) and UL designations.

MULTI-CONDUCTOR CABLES AND SHIELDED CABLE

More than one insulated conductor can be enclosed inside a common jacket to form a multiconductor cable. There are many different types of cables, many used in constructing audio equipment.

Shielded Wire and Coaxial Cables

A single conductor can be surrounded by braided wires or metal foil to form a coaxial, shielded cable, which is used in many audio interconnect cables. The signal travels through the center conductor, with the outer shield acting as the return or ground conductor. The grounded outer shield helps prevent noise from being coupled into the audio signal traveling within.

You should always use shielded cable for very low-level audio signals, such as those from a microphone or phono cartridge, to prevent noise or hum from being coupled into the signal from nearby power cords, transformers, or other components. You should also use it inside equipment, such as in an amplifier chassis, if the signal cable must be rout-

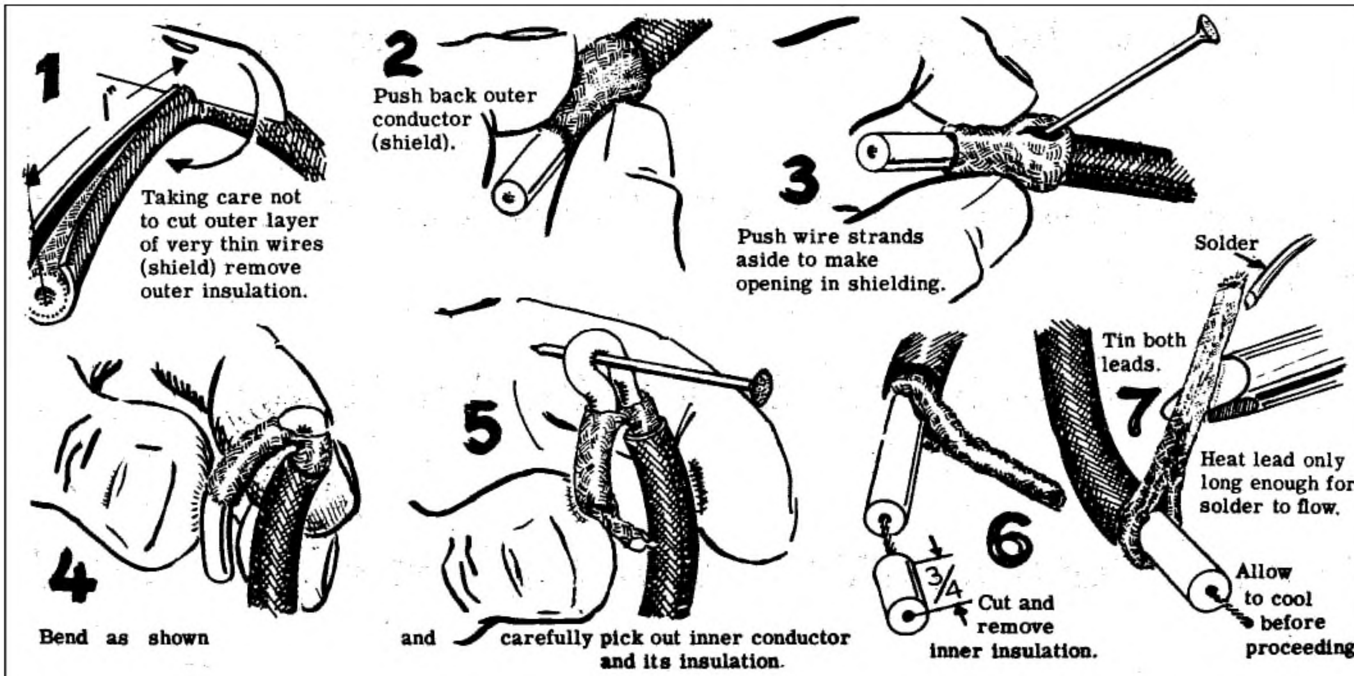

FIGURE 1: Cable termination (from Heathkit booklet #595-294).

ed near a power transformer or other potential source of noise.

When you use shielded cable, you need to consider that there is significant capacitance between the center conductor and the (normally grounded) shield. This can cause problems in sensitive circuits, such as phono cartridge inputs. This capacitance can cause degradation of the high-frequency response of your system. For this reason, it is best to use only cable that is specifically designed for audio use, and to avoid cables designed for digital or radio-frequency used in audio circuits.

Braided shielded cables can be difficult to terminate, since the shields are usually made up of many small wires. The best approach is to try to unbraid all the tiny wires, and then twist them together and tin them with solder before trying to connect them to anything (*Fig. 1*). You can use heat-shrink tubing over the twisted shield wires and over the end of the cable jacket for a neat appearance and to prevent stray shield wires from causing a short circuit (*Fig. 2*).

Foil-shielded cables normally have a wire running along the inside of the foil, called a "drain" wire. When terminating this type of cable, you simply cut off the foil shield and terminate the drain wire as you would any other wire.

Another common technique is to use a shielded cable with two conductors inside. At one end (usually the output of the cable) the shield is connected together with one of the conductors to ground, and at the other end the shield is left unconnected. This prevents the flow of current through the shield, which can sometimes introduce noise into the signal inside.

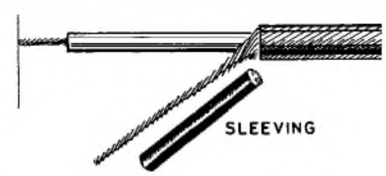

FIGURE 2: Cable modification (from Heathkit booklet #595-186).

Photo 3 shows some different shielded wires designed for audio use.

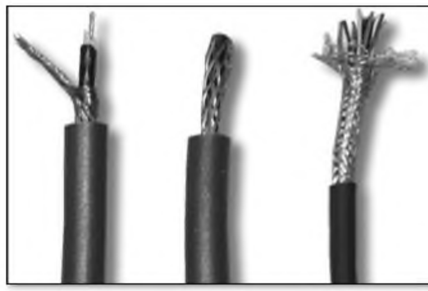

PHOTO 3: Shielded cables for audio.

Multi-Conductor Cable and Twisted-Pair Cable

A multiconductor cable is made up of several single conductors randomly oriented inside a common jacket. Twisted-pair cable is similar, except the wires are arranged in pairs that are twisted around each other. Multi-conductor cables may also have an overall shield around all the wires, as described above.

Multiconductor cables are not often used in audio equipment, except in pro-sound reinforcement and studio applications, where many signals must be transported. They are sometimes used between components, such as between an amplifier and a separate power supply.

Twisted pairs are sometimes used—with or without a jacket surrounding the pair of wires—inside equipment. Twisting a signal wire together with a return (or grounded) wire helps reject some noise, though it is not as effective as a shielded wire. Conversely, wires carrying AC power, like filament connections, are often twisted together to provide cancellation of the magnetic field emanating from the wires. This reduces the coupling of AC hum into surrounding signal circuits. [Tight twists can be achieved by clamping one end of a pair in a bench vise, twisting and soldering the other ends of the pair together, and twisting them with a bent nail in the chuck of an electric drill.—Ed.] ❖

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Electrostatic Odyssey

This article describes the construction of a hybrid electrostatic/transmission line speaker with highlights of one audiophile's lifelong journey for quality sound. **By Gary Gendel**

When my wife Dani decided we needed new furniture for our living room, she insisted my speakers had to go. I had designed and built these many years ago. They had a marvelous sound but weren't much to look at. Since Dani required that any speaker must blend in with the room's new décor, I needed to find a replacement (*Photo 1*).

This was a golden opportunity for me to acquire a pair of electrostatics. At the time, the commercial offerings were lacking, so I merged some ideas of my own with those found in *The Electrostatic Loudspeaker Design Cookbook* by Roger Sanders, and a beauty of sight and sound was born.

Many years of listening and tweaking had made me comfortable with the sound of my favorite home-brew design. These three-way, time-adjusted speakers were supported about 10″ off the floor by metal legs. The result looked like potbellied robots (*Photo 2*), and the bust of Beethoven on top of one bolstered the illusion.

I had built several other speakers over the years, trying side-firing woofers, phase alignment, and other concepts, but none sounded as good as these robots. They provided me with listening enjoyment for over 18 years without fail, except when I had to replace the woofers after they succumbed to speaker rot.

ABOUT THE AUTHOR
Gary Gendel has BSEE, MSEE, MSCS, and MSBE degrees from NJIT. He holds three patents and several awards, including the David Sarnoff Outstanding Technical Achievement Award and two Emmys. He is VP at Genashor Corp. and is a researcher for Sarnoff Corporation.

ESL SEARCH

My first encounter with electrostatics came in the late '70s when I heard the Dayton Wright Mk III at the Audio Guild in Hohokus, N.J., a place that I sorely miss. The Audio Guild had an impressive showroom spread across various rooms of a big stone mansion. There were systems set up in the living room, the sunroom, the poolroom, and the den. The electrostatics were displayed prominently in the living room.

Because the Audio Guild was such a unique place, it was frequently visited by some of the big names in audio. I had the privilege of talking to Mark Wright, Ed Nakamichi, Nelson Pass, and others. The Audio Guild also had some very interesting clients.

One day I was sitting on the couch, relaxing, listening, sipping a cup of coffee, and tasting the most fabulous Black Forest cake I've ever had (brought in by an Audio Guild client, a gourmet pastry chef), when my breath was taken away. I could no longer tell I was listening to speakers and all I heard was clear, open, effortless music. I had to have them but, as a college student, I settled for a pair of Stax electrostatic headphones. I hoped to own a pair of these speakers some day.

That day had finally arrived. Initially, I did not intend to build speakers, and set out on a quest to find a commercial electrostatic product. A local SoundLab dealer had only the full-range Auras, which I knew would be too large and expensive, but I went with my wife to look at them anyway.

Listening brought back those old feelings I had in Hohokus. Dani just kept staring silently. She finally broke the silence to tell me that they seemed too big and to ask how much they were. The response seemed to put her into a state of shock. I forgot to mention that I wasn't buying these $27,000 speakers. She wasn't amused!

PHOTO 1: The finished speakers.

After that we traveled around and listened to various hybrid designs, but they didn't have the sound I expected, and some even sounded awful. I just couldn't shelve my trusty robots for any of them.

I also scoured the Internet trying to find more electrostatic speakers, when I came across Barry Waldron of the Electrostatic Loudspeaker Design eXchange on an audio newsgroup. We had an animated e-mail discussion for the next few months, and I was per-

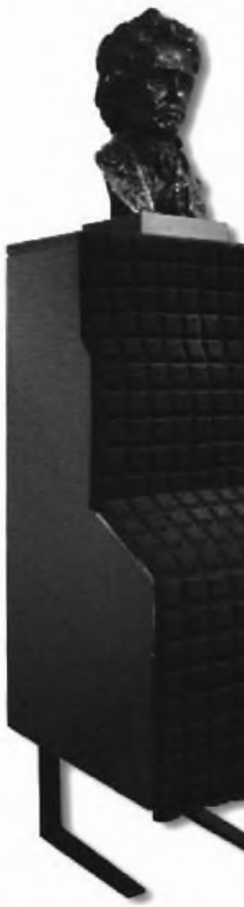

PHOTO 2: The time-adjusted design that served me for many years. Functional, but ugly.

suaded to build speakers instead. I felt like a novice since it had been so long since I had designed any speakers or audio equipment. I had been hooked on audio since I built my first transistor radio at the age of eight.

When I went to engineering college, I wished to design audio equipment. In my junior year, I produced a high-quality amplifier using the cascode transistor configuration. In my senior year, I built a digital-switching amplifier based on the brand new VFET technology.

As a student, I worked for a local acoustic testing lab and consulting firm. After graduation, I didn't receive any response from the resumes I sent to audio companies, so I took a job designing integrated circuits. Having the background, but no recent experience, I was both excited and nervous about trying to pick up where I left off. I purchased several books on loudspeaker design and dug in.

DIY DECISION

Given the limited wall space, I started with the "Compact" design from Roger Sanders' book, in which the woofer is loaded into a transmission line bent into an "L" to hold the electrostatic panels. I calculated that the design could use only about 25″ of wall space. Originally, I planned to produce the electrostatic grids with closely spaced welding rods. My biggest concern was how they would look.

Just about that time, Barry started to offer for sale assembled panels that seemed to fit the bill. These used perforated metal panels for the grids and were about 14 × 42″ overall. I was hoping for something more on the order of 70″ high, so I could provide an overall speaker height around 7′ that would perform well with the listener in a seated or standing position. I wrestled with this problem and finally decided to use two panels stacked vertically, since I had room to spare with 9′ ceilings.

I chose the 12″ Dynaudio 30W54 woofers in the transmission lines. In order to provide solid support for the panels, I made a complete frame around the electrostatic section. I located an active crossover unit designed specifically for hybrid electrostatic designs from Acoustic Instruments, Inc. This unit came with full documentation and enabled me to tailor the crossover points, equalization, and gain of the circuits for my design. The company is no longer in business, so an alternative source is required for future designs.

I produced a 3D model of the cabinet (*Figs. 1a* and *b*) in AutoCAD. I chose to angle the cabinet 60° to produce a parallelogram for three reasons. The first was to reduce by several inches the linear wall space required by the speakers.

The second was to provide a beam-splitter for the acoustic back wave. I calculated that the cabinet would deflect approximately two-thirds of the reflected wave when placed against the wall. I added a slight backward slope to the front to move back the center of gravity for stability. As an added benefit, with walls that are not parallel the cabinet's resonance characteristics should improve.

Third, the sonically dead region where the two vertical panels meet would be focused away from the head of a listener in either a standing or sitting position. The software model was needed to show Dani what I had in mind.

ASSEMBLY

I ordered panels, step-up transformers, and high-voltage power supplies from Barry. The first pair of panels came generously surrounded by bubble-wrap in a custom crate. I shipped back the empty container and was promptly reimbursed for the shipping charges.

The other pair came in a sandwich consisting of the panels individually covered in bubble-wrap placed between plywood plates and wrapped with cardboard. These panels had a gentle but complex warp when they arrived. They seemed to be fine after mounting, but I warned Barry about the problem. He assured me that future panels would be shipped using crates.

Each panel had one perforated sheet wider than the other and centered, so about ¾″ extended on the left and right sides. This made the panels very easy to mount using screws. I routed a channel around the mounting frame so that the smaller perforated grid would be recessed, hiding the panel edges from view. I made the routing deep enough so that there was ample room to keep the panels from becoming compressed.

The wider grids were also slightly shorter in length where the wires are attached, which gave easy access to the connection leads. The perforations of the front and back grids were aligned, giving them a semitransparent look. I noted only a couple of minor problems with the design.

PHOTO 3: The small speaker cabinet mock-up.

First, the larger grids were not cut the same. The panels were assembled with a significant difference in the position of one grid to the other. This was probably due to alignment of the perforations, but had the effect of the flange varying by up to ¼″ from panel to panel.

The second was a large (more than 1″) nonperforated area at the top and bottom of the panel. Because I was planning to stack the panels vertically, it meant it would be "sonically dead" at the 2″ region where the panels abutted. It should be possible to perforate more of the panel without sacrificing panel strength. The slope of the front panel in my design helped direct this dead region away from the ears of a listener. Overall, the look of each panel is excellent and the construction is good.

I was forced to increase some of my original dimensions to accommodate

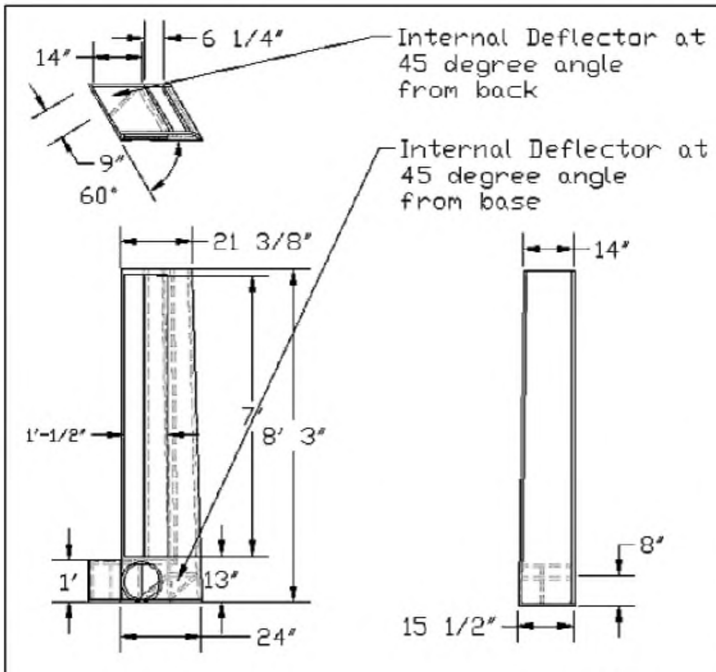

FIGURE 1a: Blueprints for the new design (using AutoCAD 2000).

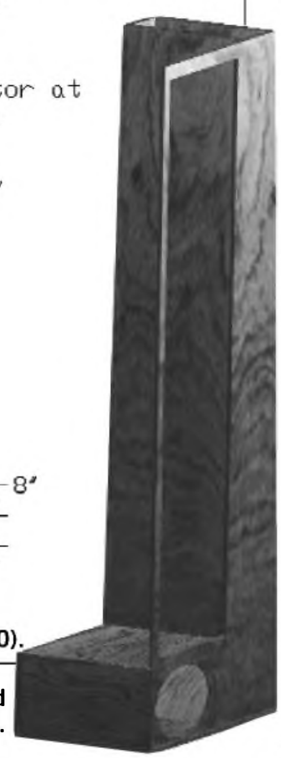

FIGURE 1b: 3D computer-rendered model (generated from Fig. 1a).

PHOTO 4: The woofer section of the TL with the top plate removed prior to final assembly.

the variances in the delivered panels. I needed at least another inch vertically, but ended up adding 3″ and an inch to the width to accommodate any possible problems due to woodworking tolerances. I built a roughly one-third scale model out of scrap lumber (*Photo 3*). This helped uncover several construction issues and ensured that the unit would be stable.

Using this model, I determined that the horizontal transmission line board above the woofer should be attached only after mounting the electronics, wiring the woofer and panels, and stuffing the transmission line. Otherwise I would need an access opening (*Photo 4*). This model was instrumental in reducing Dani's skepticism about my design while I was constructing the actual cabinets.

FINISH LINE

I used MDF for the cabinet, and routed the woofer opening to flush-mount the woofers. I assembled the cabinets with a glue-and-clamp approach, determining that additional bracing was not required. The final panel for the top of the woofer was made to fit snug. Sealing this later with silicon caulking made an air-tight, vibration-free, joint for easy removal to make repairs or modifications.

Once the glue dried, I applied a maple veneer and finished with a red oak stain, distressed by black flecks of paint, and added several coats of polyurethane. A worried Dani cried, "They're huge!" I reassured her the best I could and kept working.

Drill small holes in the transmission line adjacent to the wiring area on the panels and feed the connecting wires through these holes and down the transmission line. Apply beads of sili-

PHOTO 5: Wiring of the adjacent panels.

con caulking to hold the wires and keep them spaced apart to prevent arcing. Mount the panels so the connection wire leads are near each other at the center (*Photo 5*) and affix them with #4 sheet-metal screws every 4″ in the side. I soldered and insulated the connections with heat-shrinkable tubing and stuffed each transmission line with 3½ lbs of polyester fiberfill, reducing the density linearly starting from the woofers and ending about 6″ from the end of the transmission line (*Photo 6*).

Once the lines were stuffed, I mounted the high-voltage supply and step-up transformers in the space behind the first baffle directly behind the woofer (*Photo 7*). I installed five-way binding posts for external connections and tapped the "removable" panel into place and secured it using a silicon caulk along all inside seams and along the rear baffle. Finally, I caulked and mounted the woofers. A top screw also helped secure the "removable" panel.

Using a signal generator and calibrated microphone, I adjusted the active crossover. I was finally ready to listen and enjoy.

LISTENING CRITIQUE

I started by playing a few pieces. Just like my headphones, they had a warm sound with incredible depth and clarity. I listened to a portion of my collection covering rock, jazz, and classical on both vinyl and CD. Since I had never listened to a transmission-line woofer

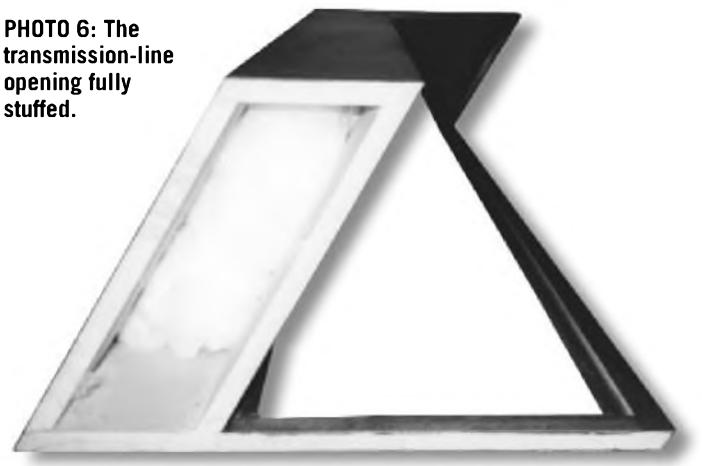

PHOTO 6: The transmission-line opening fully stuffed.

TABLE 1
PARTS LIST

QUANTITY	DESCRIPTION	SUPPLIER
4	ESL panels	ESL Information eXchange
2	Audio step-up transformers	ESL Information eXchange
2	High-voltage power supplies	ESL Information eXchange
1	Active crossover/equalizer	Acoustic Instruments, Inc.
2	Dynaudio 30W54	Parts Express
2 pair	5-way binding posts	Parts Express
4	4 × 8′ sheets MDF	Local lumber supply
4	4 × 8′ sheets maple veneer	Local lumber supply
1 quart	Red oak stain	Local hardware supply
1 pint	Black enamel (for distressing)	Local hardware supply
1 quart	Polyurethane clear coat	Local hardware supply
1 cartridge	Silicon sealer	Local hardware supply
1 box	#4 × ½″ sheet-metal screws	Local hardware supply

PHOTO 7: An overhead view showing the woofer area (stuffed), the rear woofer deflector, and the electronics area.

before, I was thrilled to discover their clean, tight bass.

The speakers looked much smaller in place against the wall and within a few weeks they were hardly noticeable. Dani even shows them off to guests as a great conversation piece.

It's been more than a year since that first listening session. Dani and I have enjoyed many hours listening together and also had the pleasure of showing them off to numerous people.

I suggest that my visitors bring some music that they are very familiar with. One brought a CD that he had been listening to continuously for several weeks. He commented, "I thought I knew this CD inside and out, but I'm hearing instruments that I never knew were there."

Another time, listening to Buddy Spicher's acoustic jazz rendition of "Georgia on My Mind," a friend exclaimed, "I could swear the musicians are right in front of me!" I saw that same look in his eyes that I had years before when I was sipping coffee, eating cake, and listening to the beauty of music from a pair of electrostatic speakers. ❖

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"An Electrostatic Speaker System," David P. Hermeyer, *TAA* 3 and 4/73.

"The Sanders Electrostatic Speaker," Roger R. Sanders, *TAA* 4/75 and 1/76.

"A High Efficiency Electrostatic Loudspeaker System," David P. Hermeyer, *TAA* 2 and 3/77.

Electrostatic Loudspeaker Design and Construction, Ronald Wagner, Audio Amateur Publications, Inc., 1993.

The Electrostatic Loudspeaker Design Cookbook, Roger R. Sanders, Audio Amateur Publications, Inc., 1995. Note: This last one contains the No-compromise Hybrid and the Compact/Integrated ESL/TL designs that I merged together for the basis for the new design.

SOURCES

ESL Information eXchange
2820 Miller Way
Placerville, CA 95667
530-622-1539
esl@information4u.com
www.jps.net/eslinfo
Note: Barry included replacement resistors for the high-voltage supplies to allow them to charge two panels each.

Parts Express
725 Pleasant Valley Dr.
Springsboro, OH 45066-1158
513-734-3000
sales@partsexpress.com
www.partsexpress.com
Note: Parts Express included a mounting kit including screws and sealant with the Dynaudio woofers.

A Start-Up Delay for Vintage Amplifiers

This circuit offers a different solution to delay the start-up of classic amps. **By Alexander Rubli Kaiser**

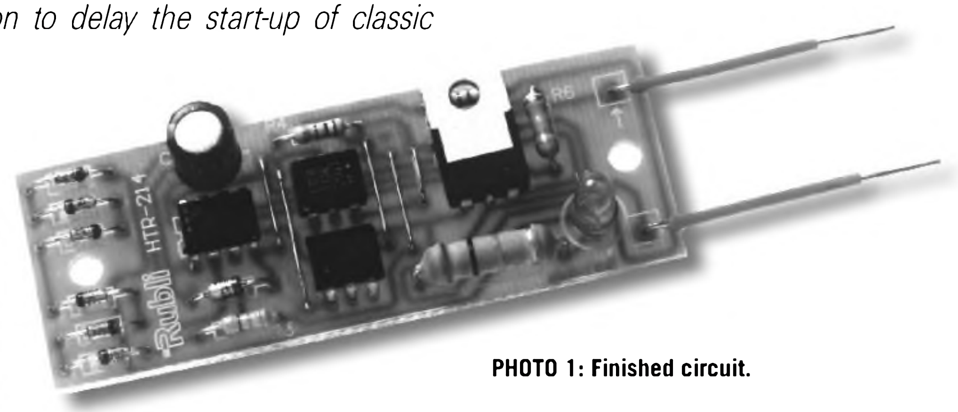
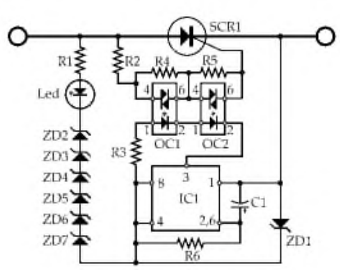


PHOTO 1: Finished circuit.

I have read dozens of articles and reviewed many recommendations about adding a retard circuit via a relay-controlled timer, a rectifier tube, or even a "standby" switch to avoid voltage jolts to cold components. All these solutions can work. But what if you own a vintage amp? There simply is no space to add a transformer, a tube socket, or relay. Drilling a hole into the chassis so as to add a standby switch is not an option. Most collectors would not purchase an original amplifier if major modifications had taken place.

My proposed circuit (*Fig. 1*) is a small PCB with only two terminals that go in series to the HV. Attaching the PCB to the amplifier can return it to its original state by reconnecting the cable you unsoldered. There are no connections to ground, no relay, no switch, and no cable runs (*Photo 1*).

The circuit contains an SCR that acts as the switch, a 555 IC timer, and other associated components. Once the SCR switches on, all components on the circuit vanish from the electrical path. The drop voltage of the SCR is less than 1V. The voltage of the associated components is taken from a shunting cascade of zener diodes.



Components List:

R1	10KΩ 1 W
R2,3	330Ω 1/2 W
R4,5	1M Ω 1/4 W
R6	330K Ω 1/4 W
C1	47uF 16 V
OC1,2	MOC3011
IC1	NE555
SCR1	600V 8 amp SCR S6008L
ZD1	Zener Diode 12V 1/2W 1N5242
ZD2	Zener Diode 17V 1/2W 1N5247
ZD3-7	Zener Diode 56V 1/2W 1N5263
Led	Gen Purpose LED

FIGURE 1: Circuit schematic. G-1731-1

INSTALLATION

Be careful, for this project involves high voltage; electric shock can be fatal. Before installing, always unplug the unit from the AC socket and fully discharge the electrolytic capacitors with a 10kΩ resistor to the ground. Then wait a few seconds before trying to handle the amplifier. Always check the voltage before soldering or unsoldering any cable.

Locate the positive lead of the main rectifier and the first electrolytic capacitor. You will notice several other connections that lead to the amp: one goes to each OPT, and a resistor goes to the next HV stage. The connections to the PCB go from one side to the junction of the positive side of the rectifier and the first electrolytic capacitor; the other connection goes from the remaining cables that were connected to that point.

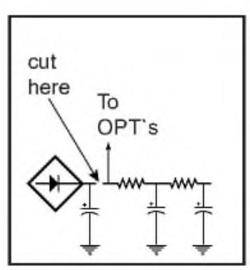


FIGURE 2: Connecting the PCB. G-1731-2

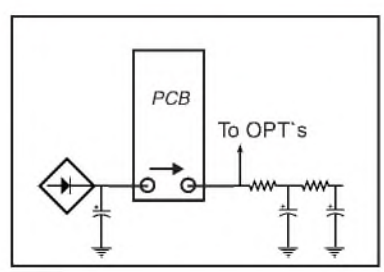


FIGURE 3: PCB placement. G-1731-3

ABOUT THE AUTHOR

Alex Rubli is a Macintosh computer consultant based in Cholula, Mexico. He worked as chief engineer/designer at Pierdant Electronica, the German DUAL audio company representative in Mexico back in the '80s, when some of his designs, such as turntables, solid state amplifiers, and speakers, hit the Mexican market. For personal reasons, he moved out of this business and went into computing science. His interest in tube audio re-emerged some years ago. In his laboratory, it is usual to see high-end Macintosh computers testing laboratory cards and software, next to a pair of vacuum tube amplifiers waiting to be restored.

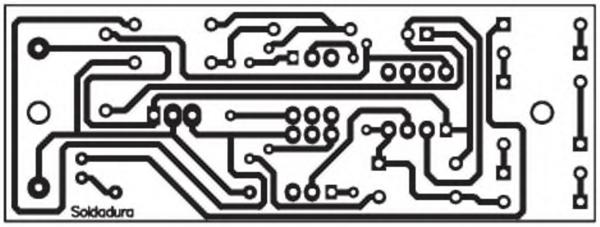


FIGURE 4A: Copper side. (100%) G-1731-4A

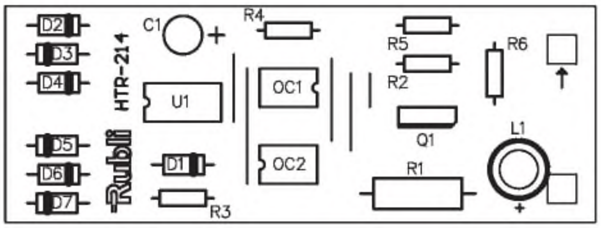


FIGURE 4B: Component side. (100%) G-1731-4B

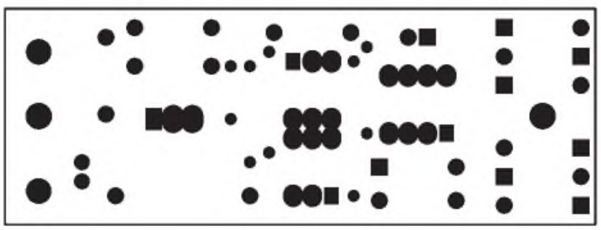


FIGURE 4C: Soldering mask. (100%) G-1731-4C

See *Fig. 2*; notice the direction of the arrow on the PCB.

The circuit is intended for use with an HV of 350-550V. If you have an amplifier with a lower HV, try bypassing ZD2. The voltage across R1 should be 45V or more.

THE LED

When the amplifier is powered on, the LED will turn on until the timer fires up and takes the SCR to the "on" state. After this occurs, the LED will turn off.

The timing circuit will fire up the SCR in about 45 seconds if the tubes are cold. If the tubes are hot, the time is reduced about ten seconds. The necessary current for this circuit originates primarily from the idle current in the output tubes. If you are troubleshooting the amp and remove the tubes, the SCR will not activate, and you need to bypass the circuit.

The PCB (*Fig. 3*) measures 3″ × 1⅞″. The completely assembled PCB is available from the author at rubli@pobox.com. You can find further information at http://www.pobox.com/~rubli. ❖

Recreating a 1948 Console Amp

Before super-low-distortion amplifiers, there was the classic sound of the Westinghouse console, which you can re-capture in your living room today. **By Larry Lisle**

In 1948 my parents purchased one of the last great radios, a Westinghouse AM/FM-Phonograph console, Model H-169, for $400. This was the top of the line, a no compromise "build it the best we can" radio that was at the tail end of a long, long period of development. The engineers knew the questions and had the answers. I don't play the Westinghouse much anymore, but when I do I'm amazed at the wonderful sound that comes out of this 50-year-old classic. I recently set out to see whether I could find out why, and build one that sounds the same.

A MARVEL OF QUALITY

The first thing you notice about the Westinghouse is the quality. The cabinet work is superb. There's a saying, "they don't build them like that anymore." In this case, I'm not sure they can. The quality is internal also. In 50 years the output 6L6s have been replaced twice, as well as, I believe, two other tubes. That's it! No capacitors, no resistors, no transformers, nothing. And there's not a trace of hum.

Electronically, the Westinghouse uses push-pull beam tetrodes in the output without feedback. D.T.N. Williamson would be spinning in his grave but the fact remains that big tubes—run at normal listening levels under a watt of average output—don't produce much distortion, and the Westinghouse has some unique circuit features to minimize even that.

You'll notice in the circuit diagram that there is a 5000pF capacitor from the plate of each output tube to ground. When I tried it in my version, there was a severe rolloff of the high frequencies—over 4dB at 10kHz and 8dB at 20kHz. But the Westinghouse has great treble! What I believe happens is that the output stage, which produces the most harmonic distortion, is purposely designed to roll off the highs, while the circuitry between the triode voltage amplifier and driver is used to boost them.

For example, say there's a 2kHz note at the input of the amplifier. The voltage amplifier is flat at this frequency and sends it on to the output stage, where a small harmonic at 6kHz is produced, as well as a smaller one at 10kHz. The harmonics are rolled off and are not as noticeable.

Suppose, instead, you put a 6kHz and a 10kHz note or overtone that's part of the music at the input. These are emphasized in the voltage amplifier, but rolled off in the output stage, so the output is reasonably flat with lower distortion. At least that's my theory, and it's the way the Westinghouse sounds. I prefer my amps to be flat, but this may be a technique worth exploring.

TWO-TAP VOLUME CONTROL

Another feature of the Westinghouse is the volume control with two taps that are connected to circuitry that boosts the bass at low volume levels. This, combined with the open-back-speaker sub-cabinet, helps give the Westinghouse a velvety bass at low listening levels. Poorly designed, an open-back speaker can sound boomy, but the Westinghouse is an integrated system,

PHOTO 1: The amplifier enclosure.

PHOTO 2: This is a basic push-pull amplifier, but it sounds very nice at quiet listening levels.

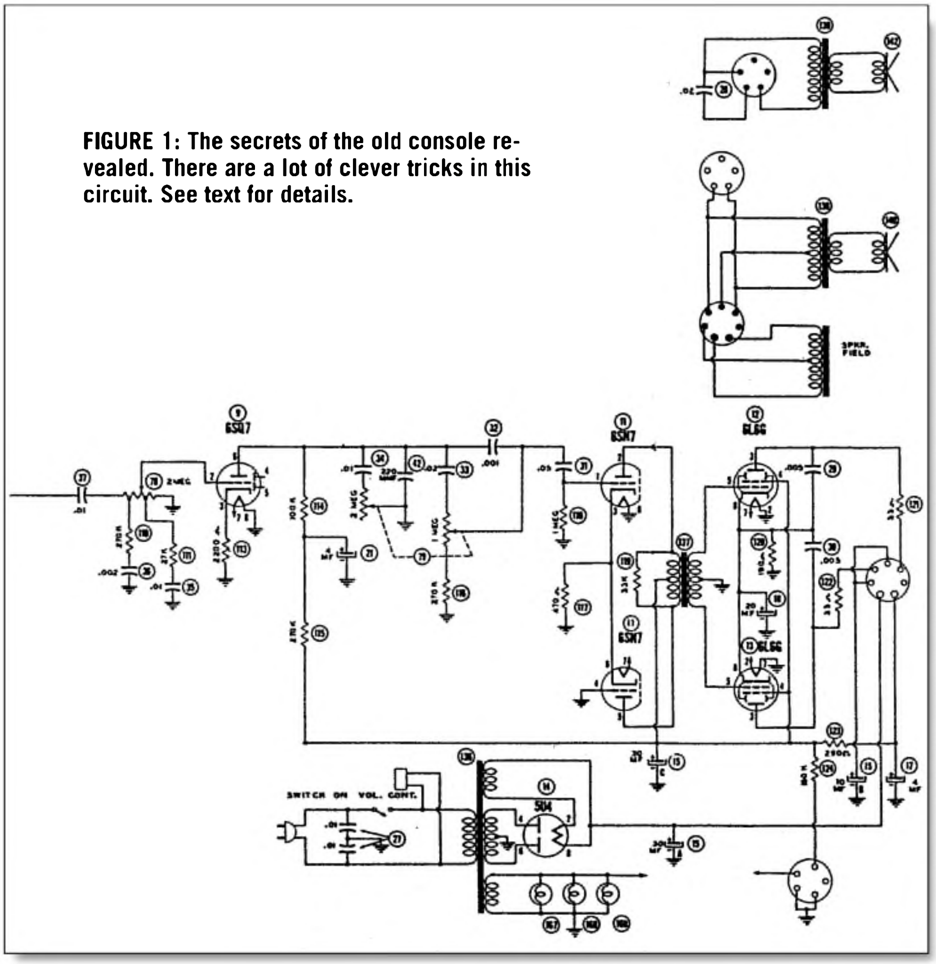


FIGURE 1: The secrets of the old console revealed. There are a lot of clever tricks in this circuit. See text for details.

and all of the sections work together. I'd love to talk to the designers!

The driver circuit is unusual, featuring a 6SN7 cathode-coupled phase splitter and a center-tapped interstage transformer. I've never seen both in any other amplifier. The cathode coupled phase splitter has the advantage over some other designs of canceling out hum from both the high voltage and filament supplies. It also prevents current from flowing in the primary of the interstage transformer, which would degrade its bandwidth.

The transformer, on the other hand, offers excellent balance at all frequencies, as well as low resistance in the input circuit if the grids of the power amplifier tubes are driven positive. It's another indication of the "spare no expense" mentality that went into the Westinghouse.

There are other examples of this attention to quality, such as the separate output transformers for each speaker, but now I'll describe my attempt to duplicate the sound using modern components.

USING AVAILABLE PARTS

I didn't attempt to build an exact duplicate of the Westinghouse since I already had one in the living room. Besides, some of the parts, such as the two-tap volume control, the paper capacitors, the field-coil speaker, and so on, aren't available. Instead I concentrated on the sound and on using parts that you could find without too much trouble.

I chose 807s as the output tubes because they're very similar to 6L6s and fine-quality NOS tubes are easy to find at low prices. I used an output transformer purchased at a flea market that was almost certainly rated for 6L6s. There are better transformers available, but I thought if I could make this one sound good, you could be confident in trying other transformers.

The screen and plate connection of each output tube is brought out to a separate terminal, so if you decide you really don't like the beam-tetrode sound, you can easily reconnect the 807s as triodes. Otherwise, the output stage is routine, with screwdriver-adjustable controls for the total bias and balance between the tubes. The capacitors from cathodes to ground are to bypass odd order harmonics.

In the driver stage I used a transformer from Antique Electronic Supply. There are much better interstage transformers, but again, if I could get this one to perform, the others will sound even better.

DRIVER STAGE

After considerable experimenting, I decided on a pair of triode-connected 3Q4s in parallel as the driver. The reasoning here was to get the lowest possible plate resistance in the driver stage, which is around 1700Ω with parallel 3Q4s, together with low distortion, hum, and other noise. It was interesting to see that the formulas for transformer bandwidth really work in trying tubes with different plate resistance. The 3S4s are even better, but their gain is a bit lower.

I used parallel feed to the primary winding of the interstage transformer to keep unbalanced DC from limiting its performance. Dropping from the 300V source down to about 60V for the drivers allows you to use a big resistor and consequently lowering the distortion, so don't put a bypass capacitor at the junction of the 10,000Ω resistor and the voltage divider.

As a result of all this, the amplifier droops about 1.6dB at 20Hz and falls off

PHOTO 3: The old Console. This Westinghouse H-169 from 1948 was one of the last of the great radios.

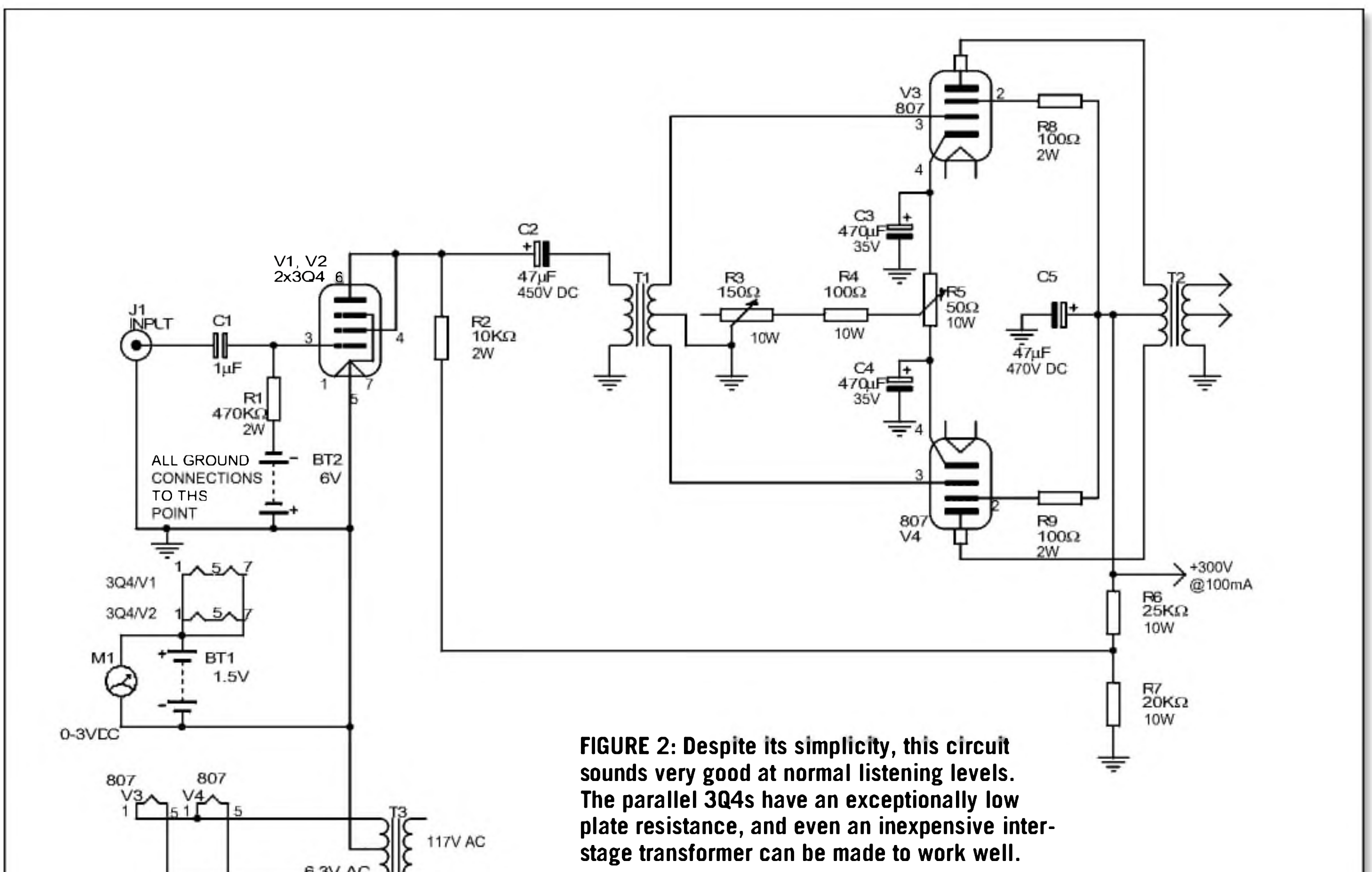


FIGURE 2: Despite its simplicity, this circuit sounds very good at normal listening levels. The parallel 3Q4s have an exceptionally low plate resistance, and even an inexpensive interstage transformer can be made to work well.

a barely perceptible amount at 20kHz, with a slight bump of about 0.6dB at 13kHz. THD runs about 2.5% at 10W and is under 1% at 1W.

The 807s are capable of much higher output, of course, if you raise the source voltage. (Voltage ratings of the capacitors may need to be increased.) But the present setup gives enough headroom for transients and peaks. 300V is also a comfortable limit for everyday use.

THE ENCLOSURE

I built the amplifier on a 24″ by 8″ (nominal) piece of red oak. I like this style of construction for its sound, the way it keeps heat localized, and its simplicity. The cover is made of a combination of poplar and birch. The meter on the front measures filament voltage for the 3Q4s, and the battery clip for a single alkaline "D" cell is on the back. Battery life is very good. Battery tubes were designed to operate over a wide range, and 3Q4s will sound fine down to 1.1V and lower.

The front panel drops down on hinges to access the controls for biasing and balancing the output tubes. The front and back panels are a little distance away from the baseboard to allow air to enter at the bottom of the box and exit through vents located at the top of the cover on the back side. The driver stage is built in a small aluminum box. I used battery bias because I like the sound.

You should be able to rotate the interstage transformer for minimum hum pickup, so leave the wires long and don't box it in with other parts. The hum actually comes from fields in the room, not from circuitry in the amplifier or power supply, which you should locate a little distance away. If you move the amplifier to a new location, take a minute to rotate the transformer for minimum output with no signal. You can make this amplifier very quiet.

TABLE 1
PARTS LIST

B1	1.5V alkaline "D" cell
B2	6V battery (4 "AA" cells)
C1	1μF
C2, C5	47μF, 450V DC
C3, C4	470μF, 35V DC
M1	0–3V meter
R1	470kΩ, 2W
R2	10kΩ, 2W
R3	150Ω pot, 5W
R4	100Ω, 5W
R5	50Ω pot, 5W
R6	25k, 10W
R7	20k, 10W
R8, R9	100Ω, 2W
T1	Interstage transformer, single plate to push-pull grids, 1:1/1 or 1:1.5/1.5 turns ratio primary to secondary. Antique Electronic Supply P-T20A14 or equivalent (6221 S.Maple Ave. Tempe, AZ 85283, 480-820-5411)
T2	Output transformer, 5000Ω plate to plate to voice coil Hammond P-T1645 or equivalent
T3	117V to 6.3V at 2 amps or more transformer
V1, V2	3Q4 vacuum tubes
V3, V4	807 vacuum tubes

POWER SUPPLY

You can use any power supply capable of delivering 300V at 100mA. Don't ever work on this or any other amplifier without unplugging every power cord and discharging the positive terminal of every capacitor to ground. You can locate the filament transformer with the power supply if you wish, but wires carrying the filament current should be twisted together and shielded from the wire carrying the high voltage. Switching arrangements and connectors can be chosen to suit the builder.

THE SOUND

So how does it sound? It's hard to describe the sound of an amplifier in words, but I'll try. It's not completely transparent, but almost. It doesn't sound liquid, as can a good single-ended triode amp. There's just a bit of an edge right at the threshold of perception. It doesn't sound good on electronic drum machines, but with big-band or other music using acoustic instruments it sounds bouncy. If I had to choose one word I'd say it sounds "happy." It certainly is not dull.

This amplifier is a good starting point if you wish to go further in the direction of push-pull amps. You can change the 807s to triode operation, add some negative feedback, try different transformers and input tubes, and so on.

As for me, I'm going down to my listening room and put on some Tommy Dorsey. ❖

Roksan Caspian CD Player

Reviewed by Charles Hansen
Listening Tests by Betty Jane & Richard Honeycutt

Roksan, 30 St. Peter's Rd., Huntingdon, UK PE18 7DB, www.roksan.co.uk, info@roksan.co.uk. Imported by May Audio Marketing, 2150 Liberty Drive Unit 7, Niagra Falls, NY 14304, 716-283-4434. $1750. Dimensions: 43.5cm W × 33cm D × 8cm H; net weight: 9.5 kg, 21 lbs. Limited two-year warranty.

PHOTO 1:
CD player front view.

The Roksan Caspian is one of the Caspian family of products, which includes a power amplifier, integrated amplifier, and tuner. *Photo 1* shows the front panel, with its combination front-loading door and fluorescent display. This panel swings down, and the CD loading drawer slides out with a push of the open/close button. Three push-button controls flank each side of the display, with play/pause buttons on each side.

CONSTRUCTION

The rugged chassis is black-painted steel with a thick aluminum front panel, and the door/display is also aluminum. The display itself is covered with a thin plastic film that gives it a ghostly green out-of-focus look. The user's manual did not say to remove it. Track information and total time are displayed for about 10 seconds after loading a CD, but the unit does not display index marks.

The optional remote control supplied with the review unit (with the all-too-common tiny buttons) can also operate the Caspian-series Tuner and Integrated Amplifier. Making quick track changes is not easy, due to the nonstandard number pad with the buttons arranged in two vertical columns. The CD-only remote control (not supplied) has the more familiar touch-tone telephone layout.

The rear panel (*Photo 2*) has the switched IEC power receptacle, two pairs of high-quality gold-plated Teflon™-insulated RCA analog output jacks, and a 75Ω BNC connector for the S/PDIF coax digital output. An optical output is not provided.

The unit is furnished with a power cord, but not with analog or digital interconnects. The power can be connected for 120V or 240V mains, and a line fuse is located in a drawer in the IEC receptacle. The third pin of the AC receptacle is connected to the chassis.

TOPOLOGY

A schematic was not supplied with the unit, and I did not attempt to open the unit for inspection. The Caspian uses a Philips TDA 1305T BCC-DAC2 with continuous calibration. The unit does not perform HDCD decoding.

The Caspian information sheet describes its patented Laser Environment Enhancer Light, which is a green laser that bathes the CD in green light. The Enhancer is supposed to produce an effect similar to painting the edges of your CDs with green paint, such as Audioprism's CD StopLight.

PHOTO 2:
CD player rear view.

PRELIMINARY TEST

Unfortunately, the Caspian arrived inoperative. The front door/display panel was flopped open and the loading tray was not fully retracted. When I powered it up and pressed the open/close button, the tray retracted, made some noises and opened up again. The unit appeared to be a (probably well-used) review sample. I returned the unit to May Audio Marketing, and they promptly furnished a new replacement.

MEASUREMENTS

I operated the Caspian for one hour with a loud music CD before making any tests. I performed all tests using the CBS Labs CD-1 and Pierre Verany test CDs (both available from Old Colony Sound Lab). The output impedance at 1kHz was 127Ω, and channel balance was better than 0.1dB.

The frequency response for the Caspian was within ±0.68dB from 17Hz-20kHz. The 0dBFS output was 2.12V RMS at 1kHz, or 0.51dB higher than the CD standard of 2V

TABLE 1
MEASURED PERFORMANCE

PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Output	2V RMS	2.12V RMS
Frequency response	20Hz–20kHz ±1dB	8Hz–20kHz ±0.68dB
Total harmonic distortion	0.003% (1kHz)	0.012% (see text)
IMD-CCIF (19 + 20kHz)	N/S	−70.5dB, 1kHz
Output impedance	N/S	127Ω
Signal to noise ratio	105dB (IHF-A)	
Channel separation	100dB (1kHz)	unmeasurable
Power requirements	<15W	

RMS. Crosstalk between channels to 16kHz was excellent and was below the noise floor of my test equipment.

THD+N versus frequency results are shown in *Fig. 1*. Normal practice when testing devices with digital-analog converter outputs is to engage a 22kHz six-pole, low-pass filter to remove out-of-band noise. My HP-339A distortion test set has two three-pole LP filters, at 80kHz and 30kHz.

With both filters engaged, the combined effect is a slightly better than three-pole LP filter at 28kHz. As such, my THD+N readings are somewhat higher than they would be if they were measured with the specified LP six-pole filter at half the 44.1kHz digital sampling rate. (When I repeated the THD measurements with just the 80kHz LP filter engaged, the THD+N was 0.02% higher across the graph.) The spikes in THD above 2kHz are the result of noise and distortion cross-products with the 44.1kHz sampling frequency and are not unusual in digital equipment.

The spectrum of a 50Hz sine wave at 0dBFS is shown in *Fig. 2*, from DC to 1.3kHz. The calculated THD+N was 0.0025%. There were no significant harmonics and only extremely low-level noise artifacts.

Figure 3 shows the spectrum of response to equal level 19kHz and 20kHz signal, each at −6dBFS, from DC to 20.8kHz. The 1kHz intermodulation difference product measured −70.5dB, while the 18kHz and 21kHz products measured −38dB. The spectrum analyzer FFT was still processing the data, and the IMD spectrum was decreasing when the 30-second CD test track ran out. The actual IMD is probably lower than it appears.

Figure 4 shows the reproduction of an undithered 1kHz sine wave at −90.31dBFS. At this level, the signal consisted of ±1 bit of data, producing two different voltage levels that are symmetrical about the horizontal axis (time). These discrete voltage steps were not obvious, probably due to out-of-band high frequency noise. The ringing after each vertical transition can be explained by the Gibbs Phenomenon[1].

The Caspian ignored defects on the Pierre Verany Test CD#2 out to track 32, which has a 1.25mm long section of blank data. At track 33 (1.5mm defect), the unit emitted a series of audible clicks at each pass through the defect. The unit easily met the Red Book requirement of 0.2mm max. ❖

REFERENCES

1. Arfken, G. *Gibbs Phenomenon*, "Mathematical Methods for Physicists," 3rd Ed., pp. 783–787, Academic Press, 1985.

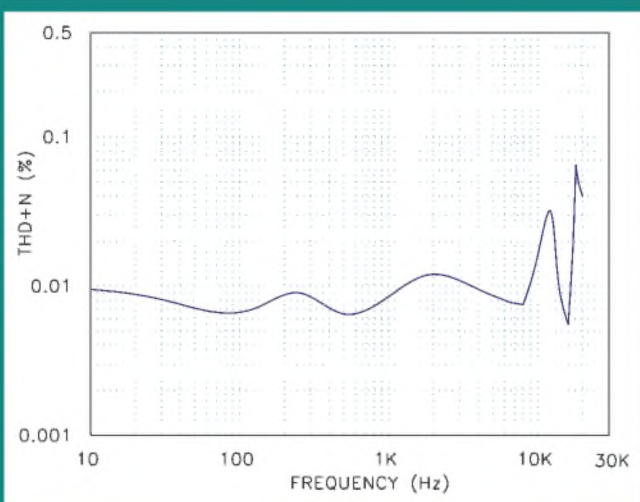


FIGURE 1: THD+N versus output power.

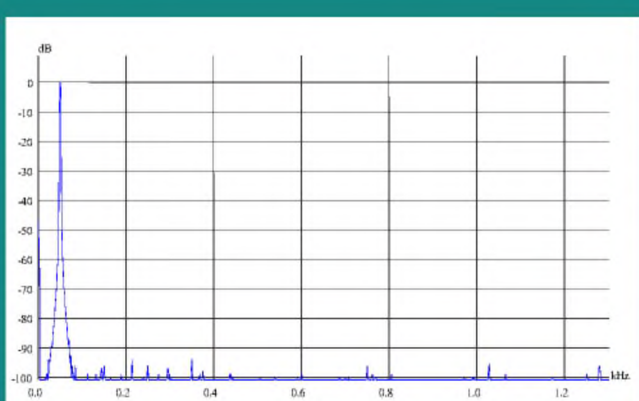


FIGURE 2: Spectrum of 50Hz sine wave.

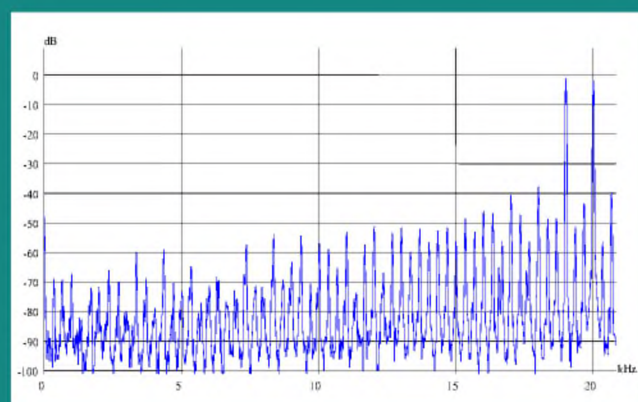


FIGURE 3: Spectrum of 19kHz + 20kHz intermodulation.

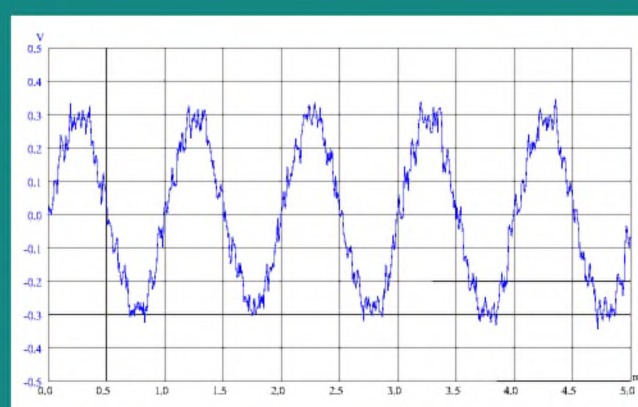


FIGURE 4: Undithered 1kHz sine wave at −90.31dBFS.

Reviewed by Betty Jane and Richard Honeycutt

It's usually the little things...The elegant front panel. The solid (and heavy) feel. The smooth, silent glide of the door as it opens and closes. The fact that it is a single-CD player, not a changer. The gold plating on the RCA connectors. The heavy rocker switch on the back panel. The heavy, removable power cord. And, of course, the name. (We're Narnia fans from way back!) By the time you actually start to listen to the Roksan Caspian, you're already impressed by the quality.

On the other hand, it is this very sort of first impression that makes it doubly important to be on your guard, especially when not testing double-blind against some standard. So between the Caspian's arrival in our home and our actual listening session, I did some thinking about just what sort of sonic differences you could reasonably expect among professionally designed CD players. My mind went back to the horrid first impression I had of compact discs.

A friend of mine had purchased an early CD player and several "Best of the Beach" or some such CDs. Those of us who were in radio back in the '60s used to comment that 45 RPM records were pressed on a mixture of chicken manure and sawdust. This first CD sounded even worse. Obviously the engineer thought you could overdrive a CD master by 10 or 15dB, just as he always overdrove 45 RPM record masters. The distortion was hideous.

Fast-forward to the '90s. By now, the requirements of proper digital recording were well understood by the professionals. Overdriving simply did not occur. But listening to a slow fadeout, you could sometimes hear graininess, especially near the "silent" end of the fade. This seemed to vary with the CD player.

Such graininess could arise from improper resistor tolerances in the D/A converters, causing amplitude errors. And since all CD players have analog sections, the usual bugaboos of hum, noise, high-order harmonic distortion, IM distortion, and slight frequency-response aberrations could rear their heads. As a result, the sound could be a bit thin or edgy, or inner voices could be somewhat muddy.

All that said, I must admit that I own a fairly low-priced five-CD changer, and have been generally pleased with its performance. In fact, my cassette recorder cost over twice what my CD player did. But now you know what we were looking for in our audition of the Caspian.

We listened to the following selections from *Hi-Fi News and Record Review*'s CD Test Disk III, with comments as appropriate.

TEST 1
Track 2—Jerusalem/Parry
BJH: The performers sounded very close to the listener. Because of the recording venue being outdoors, I could not tell much about the original acoustic environment.
RH: The outdoor recording, reproduced in our fairly acoustically dead living room, sounded cramped to me. Not the best environment for listening to this recording!

TEST 2
Track 4—Trumpet Concerto in C/Vivaldi
BJH: Excellent presentation! Very natural.
RH: The trumpets (which are very difficult to properly record) are superb. See note below on a further test.

TEST 3
Tracks 5/6—Peter and the Wolf (narr)/Prokofiev
BJH: Not quite the presence of the Vivaldi, but the sonority and locations of the various instruments are very well-defined.
RH: The narrator, especially, could have had a bit more presence, but the soundstage was excellent.

TEST 4
Track 7—Welcome, Welcome/Purcell
BJH: Generally good ensemble. However, the harpsichord's position seemed to roam according to the register in which it was played: lower register to the left, upper register to the right. The apparent change in position was twice the length of the typical instrument, so the instrument's orientation could not explain this effect.
RH: The overall sound would be wonderful for a Romantic composition, but the room sounds too large for a Purcell chamber piece. But then the equipment accurately conveys the size of the room. Singers sounded somewhat blanketed and lacking the intimacy that should characterize this type of music. But I believe this is all in the recording.

TEST 5
Track 10—Corkhill (Piece 2)
BJH: Crisp, lots of presence.
RH: The silent sections are truly silent. The full dynamic range of the piece (*pp* to *f*, I would guess) is reproduced effortlessly.

TEST 6
Track 14—Rio Napo RSS Demo
BJH: Presence and stereo effect are exceptional.
RH: Fine studio recording.

Notice that by far most of our comments on these tracks did not relate to the ways any professionally designed CD player might conceivably affect the music; rather, they pertained to the recordings. Consequently, the accompanying chart of sonic characteristics ratings, based on these tests, should be taken with a grain of salt. So, to focus on any possible graininess, edginess, hum, noise, or distortion, and thereby give an honest and useful rating of the sound of the Caspian, we made...

FURTHER TESTS
First, we played the Vivaldi back on our own CD player, A/B-ing the two. The Caspian definitely produces a rounder, more perfect trumpet sound. My CD player has a slight edginess on that sound. As a trumpet player myself, I (Richard) definitely considered the Caspian to deliver superior reproduction in this test.

Second, we listened to Chadwick's "Noel" from *Symphonic Sketches*, on the CD *HDCD Sample, Vol. 2* (Reference Recordings RR-905CD). This silken choral performance would easily reveal any artifacts in frequency response as a deviation from the absolutely perfect balance and sonority of the choir, and the wide dynamic range provides a fine test of low-level imperfections in the D/A converter. No flaws in the Caspian were found.

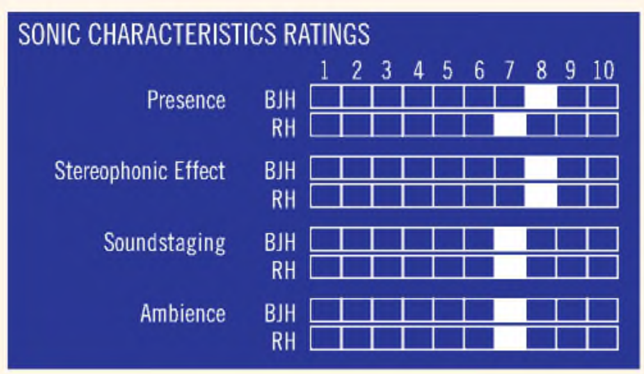
Third, we listened to Chesnokov's "Spasénïye Sodélal" ("Salvation is Created"), also from the *HDCD Sample* CD. On this one you could hear the colors of the singers' dental fillings. No distortion could hide from this test.

Finally, we auditioned several tracks from Michael Card and John Michael Talbot's *Brother to Brother*, which contains some of the best-recorded blends of nylon-string guitar, electric bass, percussion, and male voice we have heard. Again, the Caspian performed flawlessly.

CONCLUSION
We had only one minor concern: the operation of the remote control was not exactly as we expected. In our experience, when you press a button on the remote corresponding to a track number, the CD player immediately begins playing that track. On this one, you must first stop the current selection, then select the new track, then press play. Of course, we had no manual with the unit, so there may be a simpler mode, but we did not find it.

We were not told the price of the Caspian; perhaps that's a good thing. But if you, like myself (Richard) prefer a single-disk CD player, I don't see how you could do better than the Caspian. Even the remote-control issue is minor, because I seldom use a remote with a CD player anyway. And the appearance, construction, and sonic performance are superb.

ABOUT THE AUTHORS
Richard Honeycutt is an accomplished musician and author. He is Chief Engineer at Electroacoustic Development Company and EDC Sound Services, where he is engaged in acoustical and electroacoustical consulting and design. Betty Jane Honeycutt is a professional composer and musician with a BA in Music from Greensboro College. She and Richard have three grown children. In her spare time, Betty Jane helps operate EDC Sound Services and also teaches literacy and English as a second language.



Product Review

Music Hall MMF-2.1 and Goldring Cartridge

Reviewed by Charles Hansen

Music Hall Audio, 108 Station Rd., Great Neck, NY 11023, 516-487-3663, www.musichallaudio.com. $299 US. Supplied by Music Direct, 800-449-8333, www.amusicdirect.com. Dimensions: 16″ W × 4″ H × 13″ D (including dust cover); weight: 17 lbs.

The Music Hall MMF-2.1 is a single-play two-speed turntable equipped with a factory-installed Goldring Elan moving-magnet (MM) cartridge and dust cover. *Photo 1* shows a front view of the MMF-2.1. The 4.5 lb platter is belt-driven by a shielded synchronous-reactance motor that has two step-crowned pulleys, providing speeds of 33⅓ and 45 rpm (a 45 rpm spindle adapter is supplied). The motor is mounted to the plinth by an elastomer isolator.

PHOTO 1: MMF-2.1 front view.

You can change speeds by removing the platter and shifting the flat rubber drive belt from one pulley step to another. A plastic speed change key is supplied with the unit so you do not have to touch the belt with your hands, thus avoiding oily contamination (MHA also supplies a spare drive belt). The aluminum platter, equipped with a felt record mat, sits on a die molded plastic sub-platter. The tonearm lift/cueing lever has a nice damped action that sets the stylus gently on the record.

FEATURES

The straight tonearm measures 9″ from pivot to stylus and has a dual bearing gimbal mount. The cartridge is mounted at a 25° angle in the headshell. Two opposing hex-key screws on the counterweight shaft allow for vertical tracking alignment (VTA) adjustment, and a hex-key wrench is supplied.

The counterweight has an elastomer isolation insert and threads onto the counterweight shaft for tracking force (TF) adjustments. The TF range is 1.25 to 5.25 grams, with a cartridge weight of 3gm. Music Hall supplies a slotted ½ × ¾″ sheet-brass weight that can be installed under very lightweight cartridges.

Anti-skating is implemented by means of a small cylindrical weight. You place its thin monofilament line in one of three notches in the anti-skating rod attached to the rear of the tonearm. I found that this is best done with tweezers. The weight then loops through an anti-skating lever that directs the monofilament line away from the tonearm and allows the weight to hang vertically above the plinth. Notch 2 is used for the Elan.

The tonearm wiring is hard-wired to a pair of shielded interconnects with gold-plated RCA phono connectors. An integral turntable ground wire is included with this cable pair to allow you to ground the MMF-2.1 to your phono preamp chassis.

The standard Goldring Elan cartridge is premounted to the tonearm, with VTA preadjusted at the factory. The Elan has a 15-micron spherical diamond stylus. No performance curve was supplied, as it is with the more expensive cartridges. My Shure V15 Type V came with a 20-page booklet, with curves for frequency response and trackability versus recording velocity.

If you prefer, you can install a different cartridge. Music Hall supplies a plastic mounting pattern that you set onto the platter spindle. The pattern has three reference points that allow you to adjust the cartridge mounting screws so that the stylus-to-pivot and cartridge/headshell alignment are in accordance with specified dimensions.

I found the tracking force very easy to set up, even without a stylus force gauge. You simply remove the stylus guard and adjust the counterweight so the cartridge just floats at the record-playing level. Then you hold the counterweight in place and turn the force dial ring until the top of the scale reads zero. Finally, turn the counterweight to the recommended TF reading (1.7gm for the Elan). When I checked the TF with my stylus force gauge, it read

1.8gm, so the counterweight scale appears to be sufficiently accurate.

LISTENING TESTS

I used a number of LPs for my listening audition. I enjoy jazz and classical music, but I also played some popular music (well, popular from the '60s–'80s). I used the phono preamp in my Control Preamplifier with HeadRoom™ Module (*AE* 6/97, p. 8), a Parasound HCA-1000A power amp, and NHT SuperOne's and SW2P subwoofer.

The MMF and Goldring combination has a nice, warm midrange, with a slightly rolled-off treble that complements jazz very well. I don't believe it has a really extended deep-bass response when compared to the Shure V15 type V, but it is no slouch either. It acquits itself quite nicely for a $60 cartridge.

The table and cartridge had no trouble with the Shure Audio Obstacle Course test LP, or the high velocity tracks on the Stereo Review SR12 Stereo Test Record. The massed strings and choral voices were clear and well-defined—much more so than CDs of the same label and title. The Nutcracker Suite is one of my favorite classical recordings, and I have yet to find a CD that comes close to the Fritz Reiner/Chicago Symphony Orchestra LP (RCA Red Seal VCS-7100).

The MMF-2.1 brought out all the fine details in instruments such as the brushed cymbals, triangles, acoustic guitar, strings, and the woody tone of the acoustic bass. I think it is a real price/performance bargain.

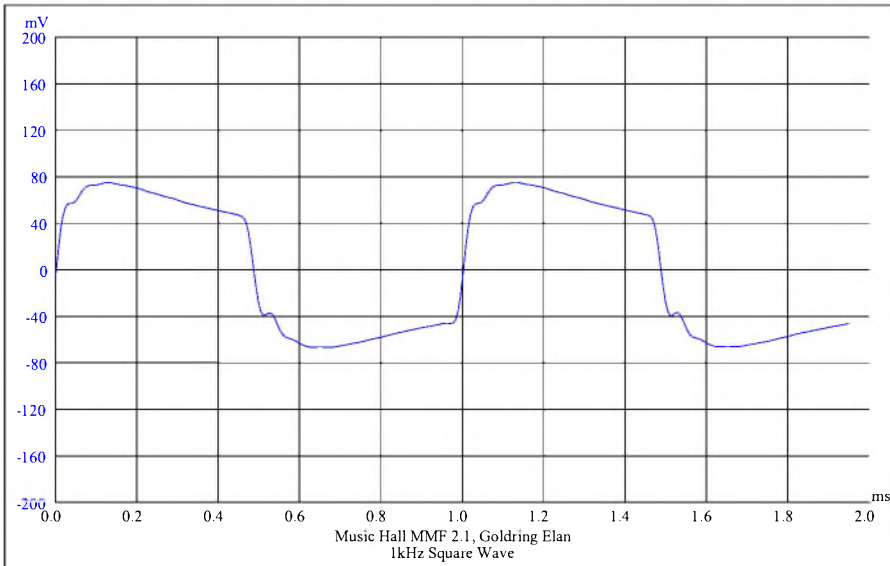


FIGURE 1: Frequency response.

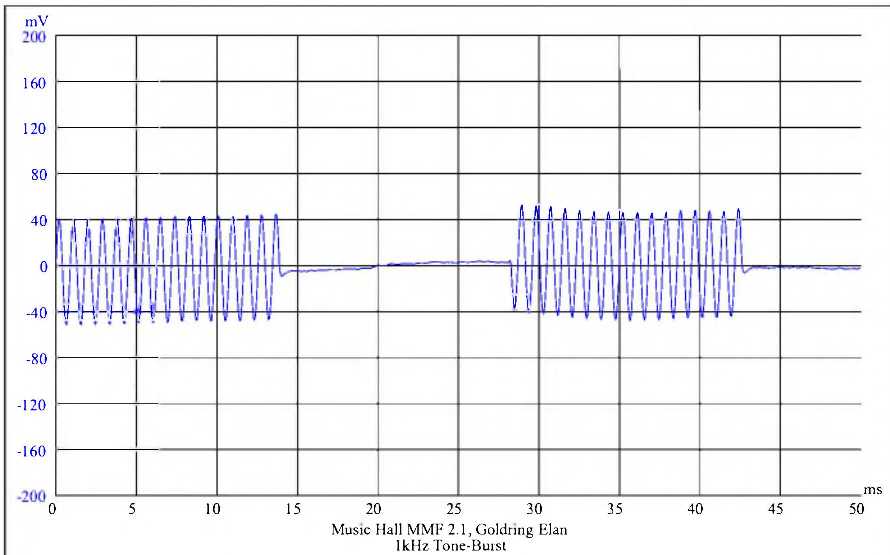


FIGURE 2: 1kHz square-wave response.

MEASUREMENTS

Ed Dell asked me to take as many objective measurements as I could. I have two records with test-tone tracks: Stereo Review's SR12 Stereo Test Record (1969) and *Hi-Fi News & Record Review*'s Test Record (HFN-001, 1996). Many of these test tracks are designed for subjective listening evaluation, but I selected those I believed were best suited for measurement.

I measured the frequency response for the MMF-2.1 (*Fig. 1*) at the phono preamp output jacks. I established 0dBr as 400mV using the 1kHz reference track on the SR12 test record. The overall response was determined by measuring the output voltage for the 19 warble tracks covering 20kHz down to 20Hz.

HF response rolled off gradually at each end of the audio spectrum, but remained within the ±3dB specification for the Elan. Music Hall recommends a cartridge loading of 150–400pF, 47kΩ. My preamp load is 100pF and 47k5.

Channel separation for the six test tracks is shown in *Table 1*. The results were essentially the same for the R-L and L-R tests. Goldring specifies –20dB for the Elan at 1kHz, and I think that is probably within my measurement error range.

THD+N measurements proved difficult, since the test tones all have over 5% THD+N on the test records I used. One low-distortion 3kHz test tone for use with a wow-flutter meter measured 0.8% THD. Since there is no indication of the recorded THD+N of this test signal, it may or may not represent the actual performance of the phono cartridge.

My LED strobe showed the platter speed to be very slightly on the fast side, with one of the test bars on the strobe disk moving one bar width

TABLE 1
CHANNEL SEPARATION

FREQUENCY	SEPARATION dB
6.4kHz	–13.6
3.2kHz	–16.7
1.3kHz	–18.4
800Hz	–19.2
400Hz	–21.0

clockwise in 30 seconds. This is of no real-world consequence. There was no flutter or wow that I could observe with the strobe, which is synchronized to the 60Hz power line.

Figure 2 shows the cartridge response to a 1kHz square-wave test track. The illustration in the test booklet shows a slight tilt and cupping in the "ideal" as-recorded waveform, so it does not represent a perfect square wave shape (I wonder whether any cutter stylus can make an instantaneous 90° turn). The Elan shows a more rolled-off wave-shape, with a slight oscillation at the leading edges.

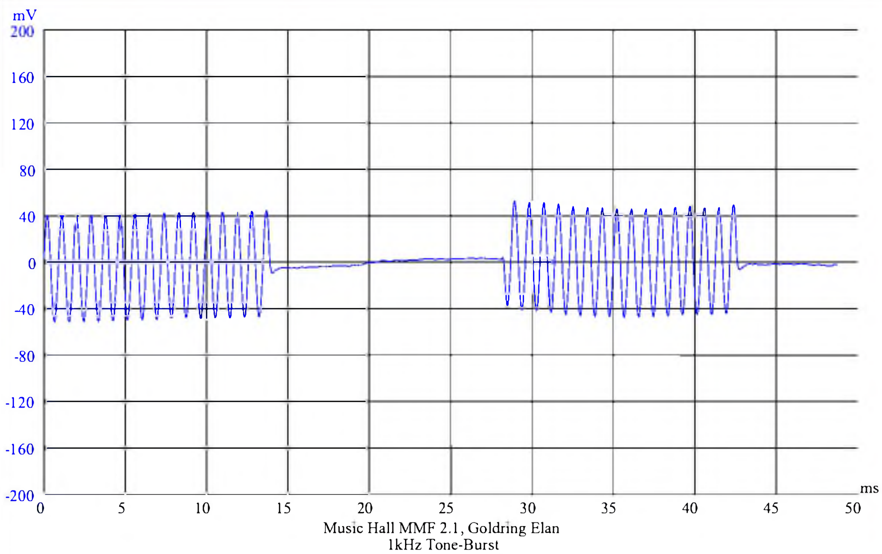


FIGURE 3: 1kHz tone burst.

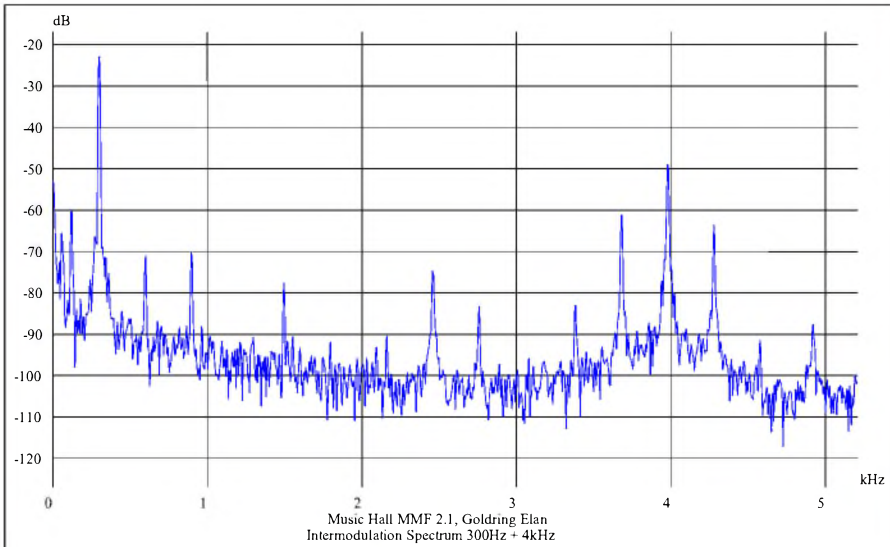


FIGURE 4: Spectrum of 300Hz + 4kHz intermodulation.

Figure 3 shows the response to a 1kHz tone burst. The Elan achieved very accurate results here, with no evidence of response dips, resonance, or spurious response for any tone burst in the sweep from 500Hz to 10kHz.

Figure 4 shows the MMF-2.1/Goldring Elan output spectrum reproducing a combined 300Hz + 4kHz intermodulation distortion (IMD) signal. The 4kHz signal is recorded at 7.5cm/s (–50dB rel), and the 300Hz signal is recorded at 9cm/s (–25dB rel). The 4.3kHz IMD product is –67dBr, and the 3.7kHz product is –61dBr. There are additional response peaks at or below –70dBr.

Interestingly, the SR12 test booklet states that the IMD tones are 400Hz and 4kHz. The spectrum analyzer and my comparison with a 300Hz oscillator tone show the lower frequency to be 300Hz.

The manufacturer's specifications and measured results are shown in *Table 2*. ❖

TABLE 2
MEASURED PERFORMANCE

PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Frequency response	20Hz–20kHz ±3dB	20Hz–20kHz ±2.91dB
Separation	20dB, 1kHz	18.4dB, 1.3kHz
Tracking force range	1.5–3.0gm, 1.7gm nom.	
Recommended loading	150–400pF, 47kΩ	
Compliance	16µm/mN	
MM output	5mV, 1kHz, 5cm/s	
Wow and flutter	0.15%, 33-⅓, DIN 45-507	
Rumble	–70dB, DIN 45-539-B	
Speed error	±0.9%	
Power requirements	2VA	

Product Review

Sovtek 6550 and 2A3 Platinum Matched Tubes

Reviewed by Charles Hansen

Sovtek 2A3 Power Triode, Sovtek 6550 Beam Power Tube, New Sensor Corp., 20 Cooper Square, 4th Floor, New York, NY 10003, (212) 529-0466, www.newsensor.com.

INTRODUCTION

I previously reviewed the Sovtek 2A3 in the 5/00 issue of *Glass Audio* ("2A3 Sovteks," pp. 46–48). New Sensor furnished two more "Platinum Matched" 2A3s along with four 6550s for this evaluation. As before, these tests were strictly objective measurements. I did not install them in an amplifier for any subjective testing.

2A3 TEST DATA

At 5.375″, the tubes exactly met the maximum height of the RCA ST16 Outline 51. The maximum diameter of the Sovtek tube was 2.0″, just under the ST16 maximum of 2.063″. The filament currents for both tubes were within the specified 2.5A maximum at a filament voltage of 2.5V AC 60Hz.

I measured the dynamic characteristics using the Audiomatica Sofia vacuum tube curve tracer. *Figures 1* and *2* are the plate curves for samples 1 and 2, respectively.

The variations in plate current at a fixed plate voltage of 250V DC and a grid voltage of –48V DC is shown in *Table 1*. I have also listed the G_m, µ, R_P, and P at the selected test points for information.

I also ran the Sofia matching program, and initially the two 2A3s matched with an ERR of 116. This was somewhat high for matched tubes, so I ran the dynamic test data and the match program two more times.

ERR is a dimensionless figure of merit representing the smallest root-sum-square differences between the measured values of plate current and plate voltage over the specified range of grid-bias values[1]. The lower the number, the better the match. An ERR of less than 100 is a very good match.

The second trial resulted in an ERR of 103, and the third trial produced an ERR of 98. However, you should note that I

PHOTO 1: The Sovtek 6550.

TABLE 1
COMPARISON OF 2A3 PLATE CURRENTS AT Vak = 250V DC, Vg = –48V DC

2A3 SAMPLE	IP (mA RMS)	G_M (mA/V)	µ	R_P (kΩ)	P (WATTS)
1	27.7	6.34	6.49	1.02	6.94
2	29.2	6.03	6.34	1.05	7.33

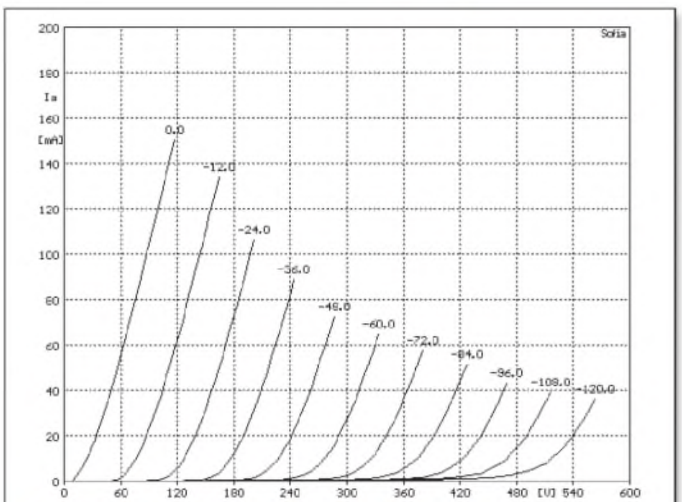


FIGURE 1: Plate curves for Sovtek 2A3 sample 1.

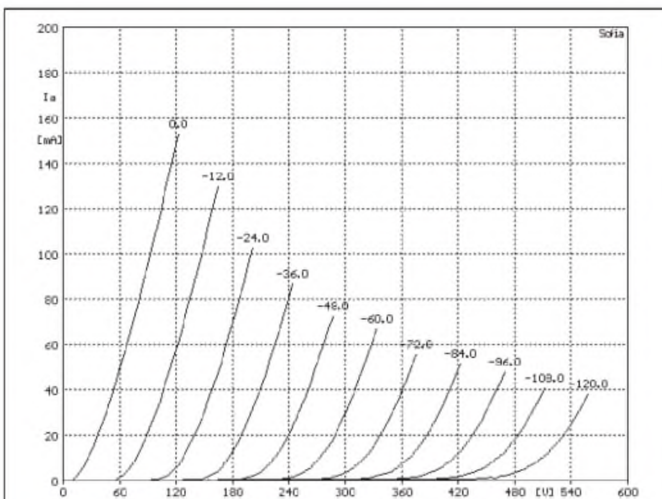


FIGURE 2: Plate curves for Sovtek 2A3 sample 2.

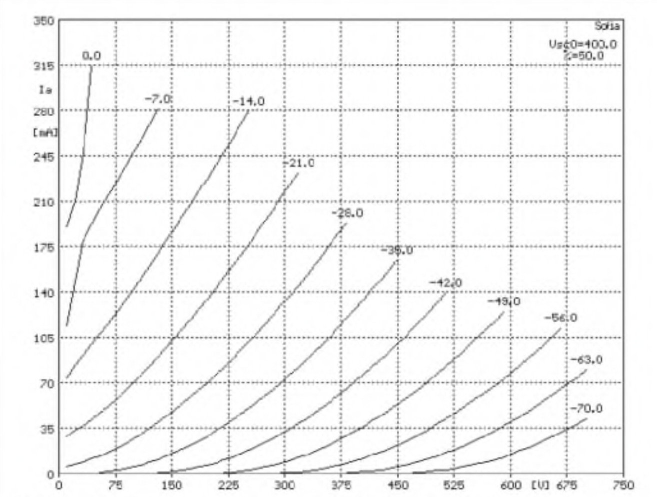


FIGURE 3: Plate curves for Sovtek 6550 sample 1.

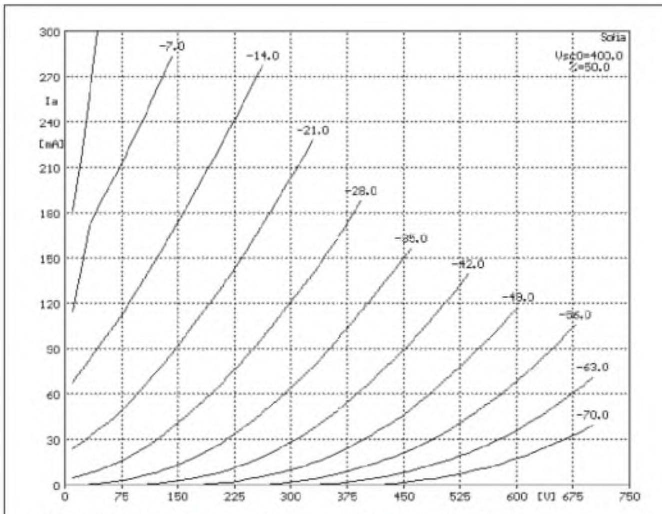


FIGURE 4: Plate curves for Sovtek 6550 sample 2.

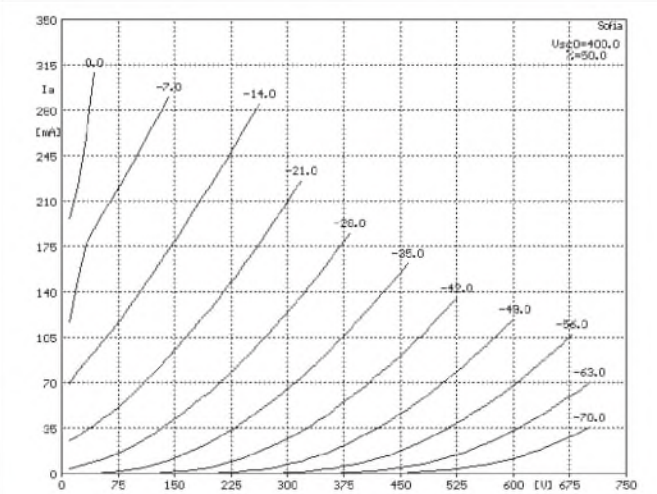


FIGURE 5: Plate curves for Sovtek 6550 sample 3.

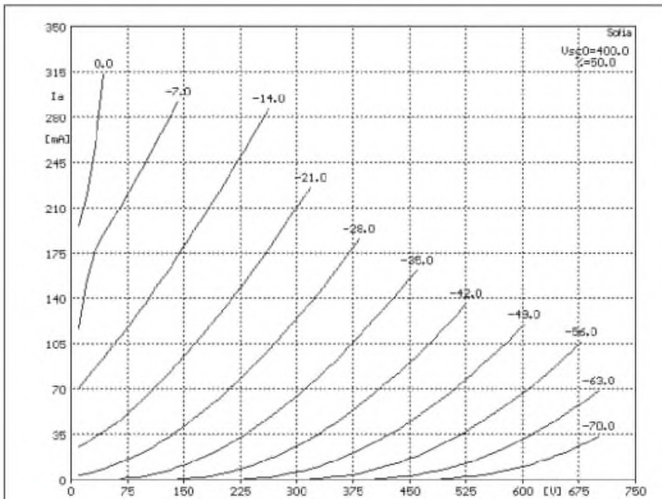


FIGURE 6: Plate curves for Sovtek 6550 sample 4.

TABLE 2
COMPARISON OF PLATE CURRENTS AT
Vak = 450V DC, Vg = –49V DC

6550 SAMPLE	IP (mA RMS)	G_M (mA/V)	μ	R_P (kΩ)	P (WATTS)
1	53.9	6.68	16.6	2.48	24.3
2	46.2	6.09	17.0	2.78	20.8
3	46.7	6.33	16.7	2.64	21.0
4	44.9	6.48	17.1	2.64	20.2

measured an ERR of 88 with two unmatched 2A3s in the earlier 5/00 *GA* test.

6550 TEST DATA

At 4.47″, the Sovtek 6550 tube was within the 4.75″ maximum height of the ST16 Octal outline. The maximum diameter of the tube, at 1.72″, was under the maximum of 2.063″. The filament currents for the tubes were within the specified 1.8A maximum at a filament voltage of 6.3V AC 60Hz.

I measured the dynamic characteristics of the four "Platinum Matched" tubes using the Audiomatica Sofia in 6550 ultralinear mode. *Figures 3-6* are the plate curves for 6550 samples 1 through 4, respectively.

The variations in plate current at a fixed plate voltage of 450V DC and a grid voltage of –49V DC is shown in *Table 2*. I have listed the G_m, μ, R_P, and P at the selected test points for information.

I ran the Sofia matching program for the four 6550s, and sample numbers 3 and 4 matched with a very low ERR of 28. The remaining two tubes matched with an ERR of 209. The ERR spread for all four tubes (Quartet mode) was 99. ❖

REFERENCES

1. Don Jenkins, "The Sofia," *V&T News* 1/99, pp. 11–19.

Product Review

Sony CDP-XE400 and CDP-XE500 CD Players

Reviewed by Gary Galo

Sony CDP-XE400 and CDP-XE500 CD players. Sony Electronics Inc., 1 Sony Drive, Park Ridge, NJ 07656, 800-222-7669 or 941-768-7669, FAX 941-768-7790, www.sel.sony.com. CDP-XE400: $160; CDP-XE500: $180. Warranty: one year parts and labor.

PHOTO 1: The Sony CDP-XE500 CD player with its supplied remote. The lower-cost CDP-XE400 does not come with the remote control, but the internal infrared remote receiver allows you to use it with the remotes supplied with most Sony receivers, as well as many universal remotes.

Sony's CDP-XE400 and CDP-XE500 are the least expensive single-play CD players from the Japanese electronics giant. Inexpensive single-play CD players are becoming increasingly difficult to find, since most manufacturers are pushing CD changers.

The CDP-XE400 and CDP-XE500 players are identical except for a few basic features. The CDP-XE400 is a bare-bones player—no remote, no digital output. For $20 more, the CDP-XE500 adds a Sony Remote Commander infrared remote control, a Toslink digital output, and digital volume control operated from the remote. The digital volume control affects both the line outputs and the headphone jack.

The CDP-XE400 chassis contains the receiver for the infrared remote control, so full remote capability (including the digital volume control) is possible with the remotes supplied with many Sony receivers. You may also control the CDP-XE400 with a "universal remote," with one caveat: Sony's track-selection system does not allow direct entry of tracks higher than ten—all Sony remotes contain a "greater than ten" button (>10) for this purpose. If your universal remote does not have this button, it may not be compatible with these players. Among the incompatible remotes is the Radio Shack 15-1994 remote I reviewed in *aX* 3/01.

FEATURES

The Sony CDP-XE400 and CDP-XE500 players include a full array of standard features. Play modes include program play, in which you can program up to 24 tracks on a disc to play in any order you choose, and shuffle play, in which the player randomly selects the program order. The repeat feature can repeat an entire disc, track, or selected program, or begin another shuffle of the entire disc. The edit/time fade button allows you to produce a program to fit any standard cassette tape length. You can also program the player to fade out at a specific time on the CD.

The peak search feature quickly locates the highest recorded level on a CD, allowing you to set the level on a cassette deck at that spot. When you insert a disc in the player, the total number of tracks and total playing time are displayed. If the disc is not played after insertion, the displayed information disappears after a few seconds (the display reads "0" tracks).

Pressing the time button restores the total time and track information for about three seconds. The time and track display is always active when the player is in the play mode. The time button also allows you to toggle between elapsed time, remaining track time, and total remaining time on the disc.

The music scan feature plays the first ten seconds of each track—when you hear the track you are looking for, simply press the play button to play this track from the beginning. You can change the playing time of each track sample to 20 or 30 seconds by toggling the music scan button.

These players do not contain a keypad for track selection—you select individual tracks using a rotary control located in the upper right-hand corner of the front panel. Sony calls this control the AMS ("Automatic Music Sensor"). Direct keypad entry is supported on the remote supplied with the CDP-XE500, or the remotes supplied with Sony receivers. Both players come with standard double-arrow search buttons. You can conduct the search silently, monitoring the time and track display, by pressing the pause button.

INTERNALS

Sony uses what it call a "Hybrid Pulse D/A Converter" in these players. This chip bears Sony number CXD8567AM and is a combination 8× oversampling digital filter and bitstream-type D/A

TABLE 1
MANUFACTURER'S SPECIFICATIONS (IDENTICAL FOR BOTH MODELS)

Frequency response: 2Hz–20kHz, ±0.5dB
Total harmonic distortion: 0.0045%
Signal-to-noise ratio: 100dB
Dynamic range: 98dB
Channel separation: 95dB
Line output voltage: 2V with 50kΩ load
Recommended load impedance: >10kΩ
Headphone output: 10mW/32Ω
Power consumption: 10W
Dimensions: 17 × 3¾ × 11 ⅝″ (430 × 95 × 295mm)
Weight: 6 lbs, 10 oz (3.0 kg)
Supplied accessories:
Stereo audio patch cord (both models)
Remote control and (2) AA batteries (CDP-XE500 only)

converter with analog voltage outputs. The analog circuitry is about what I've come to expect in low-cost players. Each channel uses a JRC NJM4558 op amp for analog filtering and output buffering, and a Mitsubishi M5218AL for the headphone amplifier. The M5218AL will sink ±50mA of load current, making it suitable for headphone amplifiers.

The CDP-XE400 and CDP-XE500 players are surprisingly well-built, particularly considering the price. The chassis and cover are all-metal, and the front-panel controls have a solid operational feel. The mechanism is quite sturdy, and the CD tray operates smoothly. The isolation feet are mounted with the screw holes off-center to help diffuse vibration modes.

Both players bear the designation "Made in China" and, in case you haven't noticed, the quality of Chinese manufacturing has improved considerably in recent years. Overall, these players exhibit none of the chintzy look and feel so prevalent in inexpensive consumer electronics.

PERFORMANCE

At these prices, the sonic performance of the CDP-XE400 and CDP-XE500 can't be beat—these players are as good as any I've heard for under $200, and better than most in this price range. Although Toslink is not the best digital interface, the CDP-XE500 makes a respectable entry-level CD transport when used with an outboard digital processor. I recommend a high-quality Toslink interconnect, such as the Kimber OPT1 I reviewed in *AE* 4/99.

Looking for an under-$1000 system for your son or daughter? Try mating the CDP-XE500 with Sony's STR-DE445 receiver (street price $250) and a pair of Boston Acoustics CR-9 loudspeakers ($420 per pair; www.bostonacoustics.com). You can operate the Sony receiver in the two-channel stereo mode or as a five-channel Dolby surround unit (with additional speakers, of course).

The receiver has both Toslink optical and S/PDIF coaxial digital inputs, so you can digitally link the CDP-XE500 CD player to the receiver, which also comes with a Sony universal remote to operate the CD player. I assure readers that I haven't abandoned perfectionist, high-end audio, but I do find it remarkable what $820 will buy these days, if you pick your components carefully.

Over the past two years we at The Crane School of Music, SUNY Potsdam, have purchased around 20 of these players for use in the faculty studios and music library. About six months ago I installed a dozen CDP-XE400 players in library listening carrels, where students tend to give any piece of electronic equipment a real workout. So far, they are holding up extremely well. My dealer told me that several local radio stations are using these players, with similarly positive reports on reliability.

These players typically sell for about $30 below the manufacturer's suggested retail price, whether through mail order or a local retail dealer. If there is no dealer near you, try J&R Music World (www.jandr.com). At these prices the Sony CDP-XE400 and CDP-XE500 are a great bargain, and are my first choices among under-$200 CD players. ❖

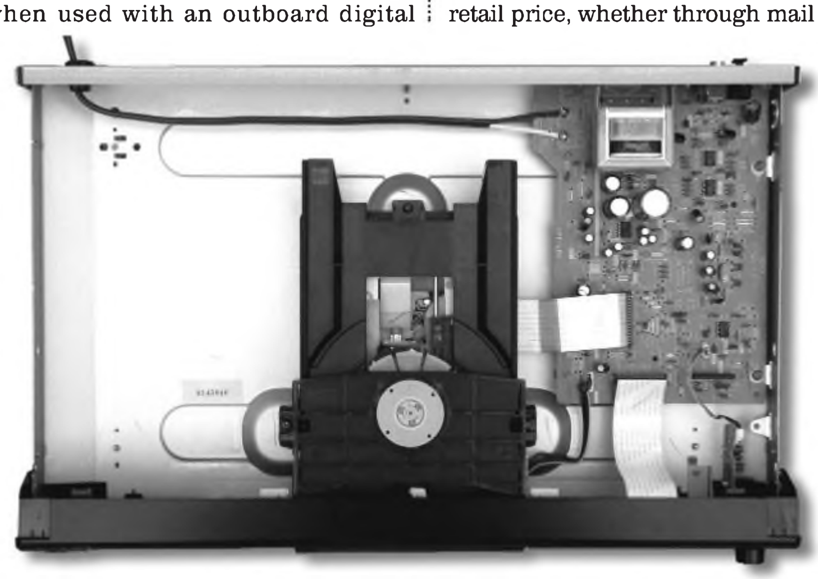


PHOTO 2: Inside view of the CDP-XE500 (the CDP-XE400 is virtually identical). The build quality belies the low cost of these players.

Book Review

Cunningham Radiotron Manual

Reviewed by Scott Frankland

PHOTO 1: Cunningham Radiotron Manual.

To say that the triode is alive and well seems almost a truism nowadays. Witness this vintage 1934 tube manual, newly minted by Audio Amateur Press[1]. This edition of the *Cunningham Radiotron Manual* was originally published at the peak of the triode era. As such, it lent technical support to RCA's growing lineup of vacuum tubes, including the 2A3, first released in 1933. Once seen as a relic by "progressive" audiophiles seeking better sound via high-power beam tubes, the 2A3 is now seen as a legitimate contender for best power tube of the 20th century[2,3].

What first impressed me about this book is the striking way in which the cover announces its contents (*Photo 1*). The invitation is almost palpable; you need only turn the page. So, what then, do you find inside?

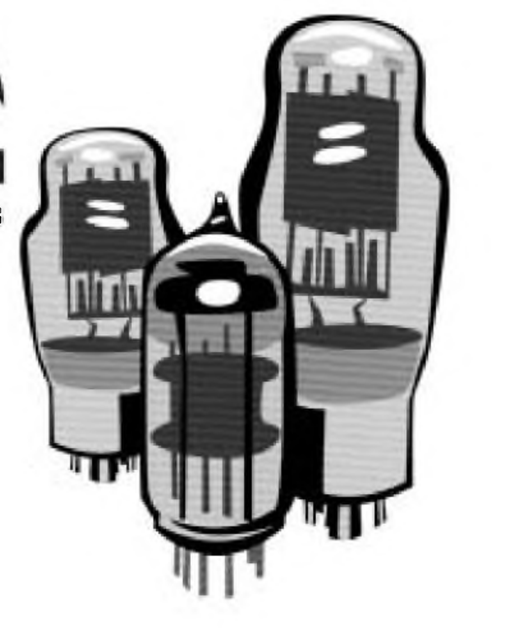
THE CONTENTS

The first part of this 154-page manual contains the familiar RCA primer on tubes, wherein tube lore of the most essential sort is laid out, including the usual caveats and operational hints. This section is often overlooked, but it retains its value as a boiled-down statement of what is most essential in putting tubes to work.

Following this is the central core of the book: the tube data. All of the tubes listed were specifically designed for either power amplification or radio reception. There are no television tubes, no hearing-aid tubes, and no computer tubes. Anyone seeking definitive data for power triodes such as the 2A3, 10, 26, 31, 37, 45, 50, and 71-A need look no further (*Photo 2*).

Nearly every tube is accompanied by a graph of its plate characteristics, along with operating points and application notes. In many cases the recommended loadline is drawn across the plate characteristics, and the corresponding power output is given. Small-signal audio tubes are also represented, but these usually combine their function with other tube elements designed for rf amplification, detection, or conversion.

For each tube type there is what appears to be a very fine charcoal drawing (or duotone) of the tube itself. There are, in addition, numerous photos not usually found in manuals—a dozen of which depict tubes on the assembly line (*Photo 3*). These photos are welcome, as they help put the book into context.*



Those who consider pentodes interesting will find them well documented here. A number of vintage tubes are represented, such as the 2A5, 6A4, 33, 38, 41, 42, 43, 47, and 59. In some cases, plate curves for triode-connected operation are shown.**

There is also vintage-tube rectifier data covering the 1-v, 5Z3, 12Z3, 25Z5, 80, 81, 82, 83, and 84. The type 80 and 81 include full families of graphical data, including a schematic of a typical application, in this case a complete choke-input power supply.

Three pages at the back of the book are devoted to tube testing—the how-tos and the wherefores. Following this is a collection of circuits, most of which are radio-related, but some of which are amplifier-related. There are, for example, a push-pull triode circuit with interstage transformer, and a single-ended triode circuit with input transformer.

Lastly, there are pages devoted to tube dimensions, an index, and a reading list of vintage textbooks. In short, this is a thoroughly executed model of what a good manual should be.

CUNNINGHAM AND DE FOREST

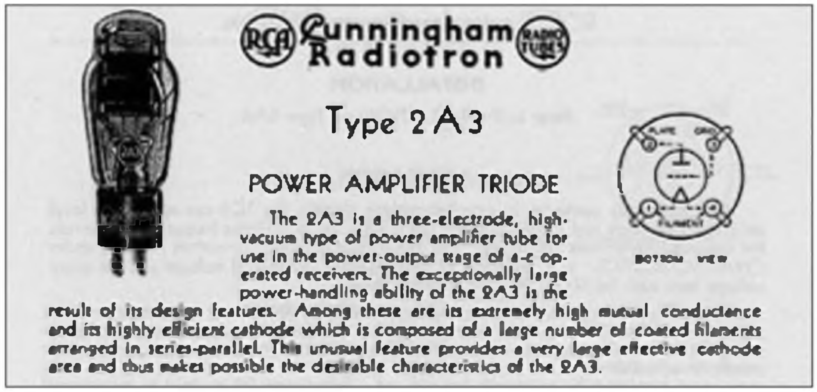
I would be remiss if I didn't remark on the mysterious figure of Elmer T. Cunningham, one of the most prolific bootleggers of the early tube era. From all reports, Cunningham was hugely successful on the West Coast, with a devoted following in the amateur market.

Then again, he was a thorn in the side of de Forest, the inventor of the Audion (triode). De Forest wished to sell complete receiving kits along with his tubes, but Cunningham cheerfully undercut him by offering raw tubes. Rather than Audions, Cunningham labeled his tubes "AudioTrons" (*Photo 4*). The Audiotron was made in the San Francisco Bay Area and was the most widely sold of all the independent brands[4].

ABOUT THE AUTHOR

Brought up as a musician, Scott Frankland studied philosophy of science at college, and in 1976 he turned his attention to electronics. Throughout the '80s he was a co-designer at MFA Systems, manufacturer of the Luminescence preamp. He was subsequently principal designer for Wavestream Kinetics, where his continuing mathematical studies helped him secure a patent in 1996, covering multi-tube triode amplifiers operating in class AB_2. He can be reached at audioeng@pacbell.net.

Cunningham's ads boldly stated: "Every AudioTron is guaranteed to have a life of 1,000 hours and to arrive in perfect condition." One-thousand


Cunningham Radiotron

Type 2A3

POWER AMPLIFIER TRIODE

The 2A3 is a three-electrode, high-vacuum type of power amplifier tube for use in the power-output stage of a-c operated receivers. The exceptionally large power-handling ability of the 2A3 is the result of its design features. Among these are its extremely high mutual conductance and its highly efficient cathode which is composed of a large number of coated filaments arranged in series-parallel. This unusual feature provides a very large effective cathode area and thus makes possible the desirable characteristics of the 2A3.

BOTTOM VIEW

PHOTO 2: Sample page.

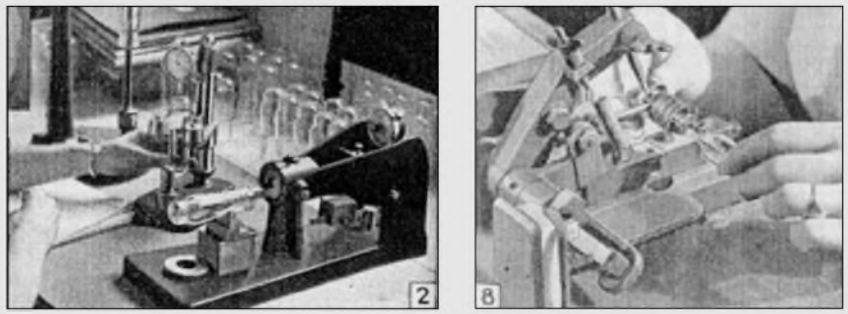


PHOTO 3: Testing tubes on the assembly line.

hours was a long time in 1915, but Cunningham's guarantee was evidently dependable. It is said that he (or his staff) discovered a new chemical reaction that allowed Cunningham to manufacture long-life tubes ahead of his time[5].

Nonetheless, in 1916 de Forest filed suit against him for patent infringement. Cunningham promptly posted bail and continued selling Audiotrons. Eventually, he and de Forest settled their differences, and apparently a joint-venture of sorts ensued.

CUNNINGHAM AND RCA

The Radiotron label had been extant since 1920, when RCA issued its first tubes: the UV-200 and UV-201. That same year, RCA brought suit against Cunningham. The upshot was that Cunningham agreed to become a distributor for RCA. It is clear that Cunningham's long experience in the tube market afforded him serious leverage at the bargaining table.

As a distributor for RCA, Cunningham wrote his own ticket. He bought only those tube types he chose to sell, and he persuaded RCA to label them with his own name. In return, Cunningham sold an enormous number of tubes for RCA.

Throughout the radio boom, RCA functioned as a joint distributing arm for GE and Westinghouse—those two old rivals in the electrical business. RCA was a sales company, not a manufacturer. In 1929, RCA purchased the Victor Talking Machine Company of Camden, N.J.[6]. The resulting RCA Victor Company was the first step in David Sarnoff's grand unification scheme to transform RCA into an integrated manufacturing, engineering, and sales entity. Victor's factories would provide the necessary facilities for manufacturing.

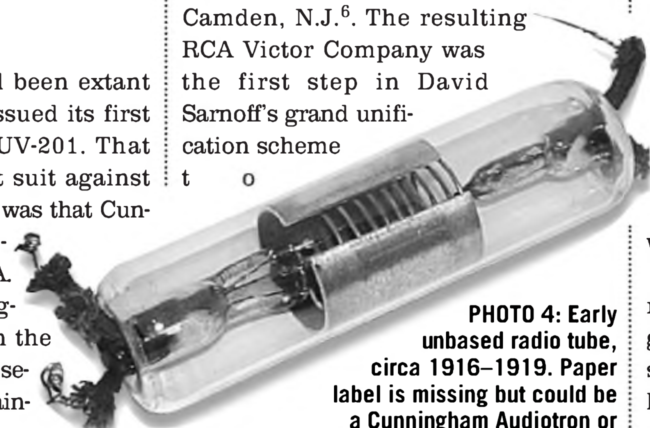


PHOTO 4: Early unbased radio tube, circa 1916–1919. Paper label is missing but could be a Cunningham Audiotron or similar. There are no stampings on the cylindrical plate (from Kilokat's website).

As part of a complex deal with GE and Westinghouse, RCA spent millions to adapt the Victor factories, and thereafter began to manufacture radio sets using Victor cabinets and phonographs. In April 1930, RCA obtained control of GE's tube manufacturing plant in Harrison, N.J.[7,8].

In May 1930, the US government filed suit against the "Radio Trust" (GE, Westinghouse, AT&T, and RCA), and by 1932 forced the combine to break up. For Sarnoff, however, this proved a blessing in disguise. RCA's independence was by this means assured, and Sarnoff's unification plan was fulfilled.

The following year, 1933, saw the arrival of the 2A3. It turned out to be a very good year for audio, as this was the same year in which Western Electric released the 300A (a 300B except for the location of the bayonet pin[9]). In the meantime, the radio boom having fizzled following the market crash of 1929, Cunningham ads began to carry the tagline "A Subsidiary of Radio Corporation of America." By 1931, Cunningham Inc. had been absorbed by RCA[10], and Cunningham himself entered their employ.

But Cunningham soon got his revenge. By 1933, he (*Photo 5*) had taken charge of the manufacturing division! For Cunningham, 1933 was a very good year indeed. From then until 1935, all tubes made by RCA were marked, "RCA Cunningham Radiotron." After 1935, the Cunningham name was gradually phased out, and RCA again took center stage.

THE LAST WORD

Audio Amateur Press has done a thoroughly professional job in the construction of this reprint. The paper, the reproductions, and the typography are all extra fine, as befits the subject matter. The *Cunningham Radiotron Manual* was originally priced at 25 cents, but you will have to pay a bit more for it today: $14.95. That, however, is less than half of what they've been trading for on Ebay lately—35 to 50 bucks a copy (circa February 2001). So this is a

PHOTO 5: Elmer T. Cunningham, President, RCA Manufacturing Co., Inc (RCA Victor).

no-brainer. The *Cunningham Radiotron Manual* is one of the classic databooks of the vintage triode era.

* You can see more such photos in Deketh's classic text, *Fundamentals of Radio-Valve Technique*.[1,11]

** Pentode fans may also wish to obtain *Valves for Audio Frequency Amplifiers*,[1,12] which contains extensive data for the EL-34 and EL-84. ❖

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Xpress Mail

PARALLEL FEED

With regard to Barney Vincelette's comments considering direct feed versus parallel feed ("Letters," *GA* 6/00), I agree with his opinion that normal SE transformers are highly linear inductors. There has been increasing hype towards choke (also a linear component) loaded tubes having a tiny OPT coupled with a capacitor. First, there was a tendency to have huge linear irons and to eliminate all capacitors from the signal chain; now it's okay to add a cap to a critical place of the chain using minimal sized iron stacks. The idea is some 60 years old.

Without further criticism, there may be a difference in sound. My purpose here is to offer one explanation. In the beginning of my career I was involved in designing transformers, mainly ferrite cores for telephony carrier-wave equipment.

A common trick was to add a series capacitor to feed a transformer from a resistive source to a resistive load. If you have a suitably sized capacitor, a series resonance is present with the transformer primary inductance forming a filter that extends the bandwidth about half an octave downwards for a final high-pass slope of 12dB/octave.

In the amplifiers' case, you may have a capacitor of several microfarads and an inductance of tens of henries, fed by the resistive triode impedance, loaded also approximately resistively. Although probably not on purpose, the resonance mentioned may hit the area of 5–30Hz and, with good luck, even give some flattening to the response.

There is, however, another aspect that may be more important. Downwards, towards the resonance frequency, the load impedance the tube sees also falls markedly due to the resonance, and has sonic influence. The bass to a lower transformer load is more rigid than to a higher one using the same tube, at the expense of distortion. More study is needed to look into this practice.

Mauri Pännäri
Finland

MTM FEEDBACK

As a satisfied user of an MTM 18 speaker system from Speaker City (and after reading the dismal review in March 2001), I feel compelled to pass along some suggestions for optimizing the speakers. I did not purchase a kit from Speaker City, as I had been using most of the drivers in a previous project. Since I was purchasing only the cabinets and crossovers, I was on my own in terms of optimizing the internal bracing, damping, and wiring.

My first recommendation is to purchase the 9500 tweeters rather than the 9900s. They offer 95% of the performance of the Revelators at about half the price. Note that when purchasing the crossovers that are designed to be used with the 9500s, both high- and low-pass sections are on a single board that fits perfectly on the back wall of the cabinet. No need to reposition the binding posts. About half of the money you save here will be spent tweaking the insides of the cabinets and wiring up the components.

The cabinets benefit from "bracing" as described by Mark Wheeler in *Speaker Builder* 8/99 ("Navigating Speaker Design"). I used narrow strips of solid oak, epoxied vertically along the center of the side and rear walls. I also glued Deflex panels (four per cabinet, available from Madisound) along the side and back walls to minimize standing wave problems and further reduce panel resonances. Lastly, a modest (less than 1 pound per cabinet) amount of long fiber wool spread evenly throughout the box seemed to eliminate the last little bit of congestion (slap back through the woofer cones?). BTW, the drivers take a long time to break in.

I used Analysis Plus Oval Theater (CL-3) speaker cable for all internal wiring. This cable is slender, very flexible, and easy to work with. It is an odd shape, but crimps or solders easily. It is very reasonably priced (less than $3 per foot, and you need only 10 to 12 feet; I got mine from Greg Lucas at MUSIC for your EARS, 216-296-2263), and sonically it is wonderfully musical.

Neil Walker
walkern@aii.edu

I read with great interest Joe D'Appolito's review of the Speaker City MTM-18 kit. I noted your belief that the internal volume of the speaker was too small. I am interested in using these same drivers in a design of yours, the 717 for A&S. I computed the volume of the 717 to be approximately 33 ltrs without bracing and other losses. This seems close to the MTM volume.

In addition, I have built an ARIA 5 with the Raven, which seems to exhibit the same problem at the bottom end as the MTM-18 and has less volume than you seem to recommend for the MTM. I know that you can't give individuals speaker-building advice, but since you seem to give contradictory volume recommendations, I wondered if you can clarify this.

Harold Goldman
LINHARSETH@aol.com

Joseph D'Appolito responds:

The A&S 717 was designed in the early '90s. Parameters for the Scan-Speak 180mm woofer have changed substantially since that time. In particular, V_{AS} *has increased greatly, requiring a larger enclosure.*

I am not aware of any low-end problem with the ARIA 5 as I designed it, but, again, driver parameters have changed over the in-

tervening years. I have not made any update to the enclosure design. It is quite possible that the volume originally specified is no longer valid.

GOOD CHANGE

Well, the change from *Audio Electronics* to *audioXpress* is perhaps giving me more tube electronics than I would desire, but, on the whole, the combination of three publications into one makes sense and I like it.

Regarding the Jan. '01 issue, I had previously read Anthony New's article in *Electronics World*, and I agree with some of the response letters, which are worth mentioning. First, IM distortion measurements in audio have been around many, many years and were frequently, if not always, included as part of an amplifier specification.

Second, IMD and (T)HD are both measures of linearity and go hand-in-hand. A reader might well conclude from Mr. New that if an amplifier sounds "bad," yet has low HD throughout the audio range, IMD measurements will reveal the problem. Wrong! There is no mechanism that I know of whereby an amplifier can have low HD and high IMD, or vice versa. I would agree, however, that an inadequate HD measurement, say at only 1000Hz, would not reveal problems at lower and higher frequencies than a standard IMD test at 60 and 7000Hz might, and in this sense an IMD test is superior and faster.

Some years ago, perhaps in the '60s or '70s, the Consumer Reports people tested power amps by an IMD method that I thought was great. As I recall, they fed the amp with white noise filtered to remove frequencies below 500Hz or so and then measured the total amplifier output at all frequencies below 500Hz. This low-frequency output could only be as a result of IM distortion. Pretty neat!

Finally, Dr. Thagard's article on a phono preamp design reminded me that I once worked out for my own amusement a QuickBasic program for a generic feedback phono preamp into which you could plug in various resistor/capacitor values and compare the results with the standard RIAA values. I went back and polished up the program and made it more user-friendly, I hope.

Any reader interested can E-mail me (Lwallc@aol.com) for my address and for $1 (or a blank floppy and a couple of stamps), I'll mail back a disk containing the BASIC program and also a compiled stand-alone program for those who don't know about or don't care to bother with the Microsoft Qbasic program that is included with Windows 95 and 98 (and, probably, Win 3.1) or in DOS versions 5.0 and higher.

Lawrence Wallcave
Santa Rosa, Calif.

GOLDEN CALCULATION

While working on a new project, I read "Determining Optimum Box Dimensions" by Louis C. McClure (*SB* 2/00). He presents a way of calculating the dimensions according to the golden ratio.

Being inspired by this article (thanks, Louis!), I found an easier and more accurate method of doing so, by using the little-known "solver" function in Microsoft Excel. In this case, it solves the box volume (defined by you) by changing the box dimensions, where the ratios between the dimensions are defined by the golden ratio (0.618).

To do this you first determine cell A1 as 1, A2 as "=0.618*A1" and cell A3 as "=0.618*A2", where A1 to A3 define each dimension of the box. The box volume is represented in cell A4 and defined as "=A1*A2*A3."

Next, go to "tools" and then "solver." The target cell is A4 (box volume), equal to the value of (fill in the desired box volume), by changing cell A1 (the only independent dimension). Click "solve" and your optimum dimensions are there! For example, if the box volume is 100 ltr (100 cubic decimeters), the dimensions are 7.51, 4.64, and 2.87 decimeters (75.1*46.4*28.7cm). Of course, you can use this for any unit of length, and any other ratio.

If the solver function does not appear under the tools menu, you'll need to install it first by going to "tools" and then "add-ins." Click on "solver add-in" and then OK. Now the solver should appear.

The values are extremely accurate, since the solver's accuracy is about a millionth of a percent by default. McClure's method was found to be about 2% accurate for each dimension (it all depends on rounding errors). Once you've set it up it's very easy and fast to get the optimal dimensions for a new box volume.

Justus Verhagen
Oxford, England

TUBE WARM-UP

In Neil A. Haight's letter entitled "Buyer Beware" (Jan. '01 issue), he refers to certain brands and types of tubes (Telefunken and Mullard 12AX7/ 12AU7/ 12AT7s, among others) having a "defect," which causes their heaters to flare brightly when initially powered up. This is not a defect in the tubes—rather, they are designed to warm up and reach operating temperature quickly.

Almost all metal filament resistors (including incandescent lamp filaments and tube heaters) have a lower resistance when cold than when hot—which causes them to draw a "warm-up surge" when connected to a constant voltage source. In these tubes that effect has been maximized to provide a quick startup.

Offhand, I can't recall where I learned about these, but I certainly remember running across them—and they do warm up more quickly than their "normal" counterparts. I don't know whether they normally have a shorter life expectancy than standard tubes, but I would not be surprised to find that heater supplies that are capable of supplying unusually large amounts of current might provide an unexpectedly large warm-up surge and blow them out prematurely as well.

Keith Levkoff
Massapequa Park, N.Y.

Neal A. Haight responds:

I would like to begin by thanking Mr. Levkoff for taking time to respond to my recent letter, regarding the behavior of the heaters in Euro 12AT7s/U7s/X7s.

After reading his letter, I decided to experiment to see whether I could verify what he stated about Euro tubes being designed for quicker warm-up. To do this, I loaded a Telefunken 12AX7 and a U.S.-made 12AX7 into my stereo phono preamp. As is usually the case, each 12AX7 contains both preamp stages for each channel.

Next, I played a monophonic record album, i.e., placed the needle on a revolving disc, and then switched on the preamp with audio amp already on, and set to normal listening level. The result was that each channel took the usual 11 seconds to operate. I verified this visually with the aid of vu meters in a hooked-up tape recorder.

In conclusion, unless there is a lab test that may prove otherwise, it appears that the Euros do not reach operating temperature any faster than U.S. makes. It is also possible that quicker warm-up may, however, have been the intention of the Euro manufacturers, even if it didn't materialize.

It won't be long before Mullard and Telefunken 12AX7s will go for no less than $100, due to the ever-dwindling supply. While I would never attempt to discourage anyone from buying them, I would, however, encourage anyone who does to heat them with DC voltage. Not only will they last longer, they will also sound better, too. Those who don't have big bucks to spend on top-notch brands of 12AX7s can rest assured that affordable versions continue to improve in performance and reliability.

Issues #14 and 15 of Vacuum Tube Valley *(PO Box 1499, Lakeport CA 95453, triode@vacuumtube.com) contain a two-part evaluation of 12AX7s, which may prove helpful to those trying to decide which brand of 12AX7s to purchase. It is important to note that 12AX67s that sound good in guitar amps may not necessarily sound good in hi-fi amps, and vice versa. Until you decide which 12AX7 sounds good to your ears, avoid spending big bucks on bulk quantities. Also, make sure you know a tube retailer's return policy—many retailers do not allow returns under any circumstances. Good luck and happy hunting.*

POWER-SUPPLY DESIGN

If a little bit of filter capacitance is good, then is a lot of filter capacitance better? For example, Thagard's excellent article, "A Phono Pre-Preamplifier for the CD Era" (*audioXpress*, Jan/Feb 2001), points out what would happen if you try to build a power supply with very small ripple (10mV) by using only a capacitor input filter. He estimates that even on his small 25mA power supply, you would get narrow 5A current pulses flowing through the diodes, capacitors, and transformer. His solution was to use a two-stage filter to minimize these current pulses and reduce the ripple.

What he doesn't mention is that these same narrow current pulses also flow through and radiate from the power lines. While their fundamental frequency is only 60Hz, because of their narrow width, they will contain harmonics throughout the entire audio band.

If a 25mA power supply can cause ampere size current pulses, consider what the much larger power supplies in your PC, TV, power amplifiers, and other electronic equipment are doing. They are all conducting at the same time—the peak of the line voltage wave. If you look at the voltage waveform of the power lines, you will probably notice that the tops of the sine wave have been flattened as these current pulses load down the line.

In the standard 120/240V system used in US homes, these current pulses tend to flow in one hot wire and out the other. Only the unbalanced current flows in the neutral wire. But in an industrial, or power distribution wye-connected three-phase system, the current pulses on each phase occur at different times and all return through the neutral wire. The three sine-wave line currents in a balanced three-phase system normally add up to zero and no current flows through the neutral wire, but in this case they all add in the neutral line.

When these current pulses get to a delta-connected utility transformer, they cause a circulating current that heats up the transformer. In office buildings where there is a PC on every desk, there have been cases where the insulation on the neutral wire has burned, due to these excessive current pulses. This is because the heating depends on the square of the current—a large narrow pulse will cause more heating than the same charge in a smaller wider pulse.

Even if the contribution of your small home audio system is not a problem, it would be wise to design more environmentally friendly equipment. I think a better solution would be to design power supplies with 10 to 20% ripple and use a three-terminal regulator to reduce this ripple to a reasonable level. This would significantly reduce the stresses on the transformer, diodes, filter capacitors, and the environment.

Walter Schillinger
Santa Cruz, Calif.

Norman Thagard responds:

I had heard of the problems that Mr. Schillinger mentioned, but did not know that capacitor-input filters in line-powered equipment were the culprits. There is a tendency to take a casual approach to power-supply design, but perhaps we should not, for the reasons that he gives.

In my summer semester special topics course, "Feedback Amplifier Principles," I teach this concept and assign a homework problem similar to the computation of the preamp capacitor-charging pulses that was given in the article. At least my students understand the ramifications of large capacitors in capacitor-input filters.

It is too bad that transistor amps are relatively low-voltage, high-current components. Tube amplifier power supplies could either have a choke-input or capacitor-input LC filter that did not generate such huge current spikes. This is possible when a power supply provides only a few hundred milliamps, but it is not an easy matter to find a choke of 1H that won't saturate with several DC amps through it.

Such phenomena as Mr. Schillinger describes are in the purview of power engineering, which is a specialty within electrical engineering. There is a recent renewal of interest in the field and in its practitioners and students for obvious reasons.

SERIES CROSSOVER NETWORK

Here are some comments on Xpress Mail published in the May '01 issue. Angel Luis Rivera has written a long and cogent response concerning the application of series crossovers to loudspeaker design, and I wish to throw in my two cents' worth.

The one property of the first-order (6dB/octave) series network that bears repeating is that the sum of the voltages across the two drivers is always equal to the amplifier voltage. The practical consequence of this is that aberrations induced by one of the drivers or by an inductor will be compensated for by the other driver, so in a sense, this topology is "self-healing." In practical use, this constitutes an advantage.

Another advantage, and one not mentioned by Angel, is that the source impedance seen by a driver, looking back into the amplifier, is low outside of the driver's working range in addition to being low within the driver's working range. For instance, with 8Ω drivers in a two-way system, both the woofer and tweeter will see an impedance approaching 0Ω away from the crossover frequency, and an impedance of 8Ω right at the crossover frequency. This helps their sound by damping out diaphragm resonances both within and outside of their working range.

In the 1970s I designed a four-way system using the Electro-Voice T35B horn tweeter and accompanying Electro-Voice X36 12dB/octave crossover. The lower three drivers were run from the X36's low-pass through a compound series first-order network. I cascaded 6dB/octave sections, each one a series constant-voltage network, to achieve 12dB/octave rolloffs for the 12″ woofer's low-pass and a 2″ dome midrange's high-pass, and 6dB/octave for high-pass and low-pass on a 5¼″ filler driver working between 500 and 800Hz.

This four-way system is linear phase from about 10kHz all the way down to the woofer's fundamental resonance (I used a Bessel fourth-order reflex alignment). This setup has extremely comfortable and dynamic sound. Since a 6dB/octave slope by itself is not steep enough for most purposes, you must elaborate on it and adapt it to a collection of drivers that make sense in a particular application.

All four drivers had Zobel networks added across them as part of the crossover network, and several resistors were added strategically so that L-pads would work properly with the dome midrange and horn tweeter, without altering the frequency response during adjustment. The horn tweeter was time-adjusted with the dome midrange.

The trick to making it all work, as I mentioned, was to choose the crossover slopes and frequency ranges to match the driver complement, and to build it up with cascaded series first-order sections. The crossover concept and the driver complement needed to be specified as an integrated system.

Victor Staggs
victors@hq.cc-inc.com

CURRENT PULLDOWN

I have some questions concerning op-amp applications and a little bit of input on various topics, as well. It's been a few years since current pulldown on op amps has been discussed, and I wonder whether anything has changed, or if indeed it is useful on newer op amps such as the AD817 or maybe the AD825. The latest I recall involved 2mA; I use the LM334Z with 33Ω bias. I have always used a heatsink on the chip when applying pulldown. Is this really necessary?

I've been having trouble finding AD827 dual op amps. Are there other dual op amps of similar or better performance out there?

Apparently a number of people preferred the AD825 to the AD817 in Mr. Jung's super regulators. Would the AD825 be a good bet for a line stage also?

I've been looking for an upgrade for my LT1115-based phono preamps (*TAA* 2/90). I'm also looking for a use for all of the AD797s I pulled out of my super regulators. Would the AD797s work well in the phono preamps? If not, what would? Are there any other ideas for the AD797s?

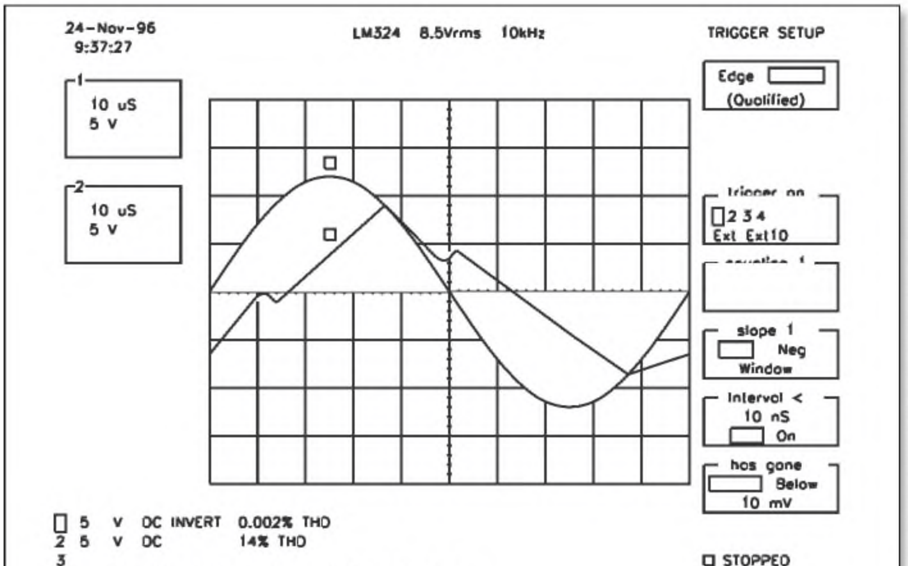


FIGURE 1: Sample pulldown current.

A recent issue of *Stereophile* mentioned that Holco resistors are no longer non-magnetic. It's easy to tell the newer ones—the steel leads are much stiffer (you could also use a magnet). They sound pretty much like "Resistas" to me. Welborne Labs (www.welbornelabs.com) is shipping Vishay-Dale RN60Ds in place of Holco H4s. I like them; Mouser (www.mouser.com) also sells them.

Here are some tips for anybody who might be trying to quickly put together a good, inexpensive amp. MCM sells power-amp modules—discrete transistor Class AB boards complete with (rather small) heatsinks—by a company called Cebek. They also sell power supplies for them (a bit wimpy).

I bought a pair of 25-watters (various power ratings available), did a resistor/cap upgrade including some added on-the-board rail capacitance, and used an oversized choke-input power supply. It sounds better than my "baseline" Adcom GFA 5200—even to my wife! MCM (www.mcmelectronics. com) "AXON" and Parts Express "Dayton" (www.partsexpress.com) poly caps work well for projects like this; take a look at them and you might have a suspicion about who actually makes them. I also like Nichion "Muse" electrolytics—try Michael Percy for them.

Finally, before you spend major money on line conditioners, look at the Sola CVS 2000W constant voltage transformers that Fair Radio (www.fairradio.com) has for $250 each, 2 for $450. The Navy uses these extensively in small-signal applications where minimal line noise is a must and they really work.

M.B. Wilcox
Altamont, Tenn.

Charles Hansen responds:

*I have used pulldown biasing to cure "scoop-out" and crossover distortion problems in the LM324 quad single-supply op amp (*Fig. 1*). In terms of audio use, Gary Galo gave us three excellent examples of current pulldown using a resistor, a constant-current FET, and the LM334 in the "New Buffers" designs by Gary, Rich Markell, and Walt Jung (*TAA *2/90 p. 48). The pulldown current is indeed 2mA and biases the op amp into Class A operation. Additional discussions on the topic can be found in* TAA *1/91, pp. 48-49; 1/94, p. 37; and 3/97, p. 39. The authors of the 2/90 article reported cleaner, smoother, and more dynamic sound using pulldown biasing.*

The 2mA pulldown current adds additional power dissipation to the IC. Whether this is a problem or not can be determined by a worst-case power dissipation analysis. I will use the AD825 data sheet for the analysis information.

Assume a 500Ω load and ±15V DC supply rails. You find the worst-case supply current (over the full temperature range, since the chip will run hot) is 7.5mA or 225mW. The 2mA pulldown bias adds 30mW to the output stage. The AD825 can swing a maximum output of ±12.9V DC with the ±15V DC supply. The worst-case chip dissipation due to the load current occurs at an output voltage of about 6.75V RMS, or 83mW. For this example, the worst-case chip dissipation is the sum of these three components, or 338mW.

According to Figure 2 of the data sheet, the AD825 8-lead plastic SOIC package can dissipate 730mW at 25°C ambient and a junction temperature Tjmax of 150°C. To be conservative, derate Tjmax to 125°C. This puts the useful dissipation at room ambient at about 600mW.

Using the package θ_{JA} of 160°C/W, the temperature rise for 338mW will be 54°C, for a case temperature of 79°C at room ambient (25°C). While 79°C is not an excessive amount of power dissipation, my rule of thumb is that any semiconductor with a case temperature that I cannot comfortably touch (50°C) at room ambient is running too hot and requires a heatsink. Removing the 2mA pulldown bias won't help very much—the case temp will drop only 5°C.

So, whether or not you need a heatsink in your application will depend more on the load current than the 30mW added due to pulldown bias. And remember, the majority of that temperature rise will be due to the supply current alone.

The AD827 dual op amp is available from Newark Electronics (www.newark.com or 1-800-4NEWARK) in four different package types. Other duals worthy of consideration for high-quality audio applications include the Analog Devices AD826 and the Burr-Brown OPA2134 and OPA2604.

I have no personal experience with the AD797. It has a bipolar transistor input stage, so at first glance it looks like it may work in a phono preamp. Readers who have tried the AD797 in this application are encouraged to share their experiences

LINE ARRAYS

As a long-time subscriber to *Speaker Builder*, I am interested in building speakers that are line source instead of point source. I would like to know where I might find information on design. I don't recall seeing anything in your magazine on this configuration. Is there a reason for this? I would appreciate any help you can give.

Keith Hamilton
khamilton@accutek.com

Bill Waslo responds:

I am assuming that your question is about line source (or quasi line source) bass arrays rather than midrange or tweeter drivers. But just in case, I'll first mention that there have been a number of articles about using line source ribbon drivers for the higher frequencies, including one in the Jan. '01 issue of audioXpress. *You can find references to more articles on this driver type by going to* audioXpress*'s website, www.audioXpress.com, and doing a search on the word "ribbon." One notable article from* Speaker

Builder *2/92 described a 20′ ribbon tweeter made by SpeakerLab.*

I've not seen as much published on the subject of line array woofer systems. There is the July '63 AES Journal *article, "Constant Directional Characteristics from a Line Source Array," by Klepper and Steele (republished in the AES's "Loudspeaker Anthology" Vol. 1). Phillip Witham's Linear Array articles (starting in* Speaker Builder *5/94) describe some of the directional effects of driver arrays as related to his sound-field replication system.*

I did some work with woofer line arrays during my focused array loudspeaker project (starting Speaker Builder *5/98), though only as a complement to the high-frequency section, which was the subject of the articles. I can say that, sonically, these stacks of ported boxes gave the best bass I've ever heard in any of my systems: very powerful and very smooth, with the major disadvantage of totally dominating the room with their massiveness. I've had a more reasonably sized woofer array cabinet design on paper for some years now, waiting for me to find the construction time.*

Line woofer arrays are a good approach toward generating clean, high-level bass, while providing some relief from room boundary effects. If you take on such a project and are interested in documenting the process and its results, I would certainly look forward to reading about it.

JOB SEARCH

I read your article in the March 2001 *audioXpress* ("Confessions of a Former Driver Designer," p. 70) and found it quite intriguing. I am a sophomore at the University of Maine studying electrical engineering. I, too, have been building stereo speakers from the age of twelve.

My father got into it and I kind of followed in his footsteps, reading *Speaker Builder* magazine and others. I love music (I am a guitarist myself) and have always thought the perfect job would be to build and design speakers. I never really considered actual driver design as an option. Where would an EE find his place in stereo speaker design?

So did you really enjoy the job, find it challenging, and so on? I am working for Massachusetts Electric this summer, just for a co-op. I would really like to get a toehold into the audio industry for next summer's co-op. If you could give me any tips or pointers it would be greatly appreciated.

Nathan Sherwood
University of Maine

Perry Sink responds:

Well, try places like Harman, Onkyo, larger consumer electronics or professional companies, and so on. Yes, these are real jobs and real careers. They tend to pay slightly less than other engineering disciplines, but you may find the work more enjoyable.

I liked the challenge of making something so inexpensive sound respectably good. And I worked on a few "skunkworks" projects that were a lot of fun, too. Start anywhere you can get a toehold and just begin talking to people that are inside the industry. Like any specialty, it's a fairly small industry and everyone knows each other.

Once you know a few key people, it won't be hard to find something. First, look at the job advertisements in the AES Journal *or* Voice Coil. *Be a little ballsy and get in touch with the engineering people (bypass the HR departments) and ask them whether they're looking for an engineering co-op. In many cases they could really use some extra help. OK, you might not get a prestigious project at first—you may sit around running response curves half the week—but you'll have a good start.*

I worked for Oxford Speaker Company in Chicago around the same time that you were at Jensen (yes, the competition). I was on the Ford account, and reading your article just gave me a huge "Matrix"-type flashback of memories.

Working at Oxford was one of the highlights of my life. I couldn't agree with you more that you learn so much about engineering when you have no budget to get high performance from a "mere paper" cone. For example, it must withstand a drive from Alaska to Death Valley in 15 minutes, 50 times a day, and withstand side impacts, explosions, high-pressure hoses, and—oh, by the way, if it is free it is too expensive, so make it cheaper!

I left the high-pressure, fast pace of Oxford after three years and ventured to the luxury and Club Med feeling of northern California and Lucasfilm. According to my Oxford co-workers, I had "made it out," instead of the usual incestuous "let's get a job at Onkyo, Panasonic, Foster, Harman."

People here in the high-end market have absolutely no concept of OEM, and I think that they actually disbelieve me when I tell them the stories that you have actually written about.

Warren Mansfield
Product Development Engineer
Lucasfilm, Ltd.

UNDERSTANDING V_{AS}

Was the article by Ray Alden on the Bella Voce (*SB* 4/00) correct in stating that the V_{AS} (liters) is doubled? Are all my books wrong when they say that the V_{AS} is halved?

Chris Turner
charger90@hotmail.com

Ray Alden responds:

In response to your question regarding whether V_{AS} should be doubled or halved with regard to the Bella Voce speaker system, I am sure that doubling V_{AS} is correct for my situation. Naturally, this seems to raise a red flag if you are correct that all your books say it should be halved. However, based on where in the book you are getting this assertion, there might be a simple solution.

On page 30, section 1.91, of the Loudspeaker Design Cookbook *(Vance Dickason, version 4), it states that for a standard configuration (two identical woofers in the same enclosure), "V_{AS} (and the associated box volume V_B) will be twice that of a single driver." On page 32 in section 1.93, he states that for a compound configuration, "V_{AS} (and the associated box volume V_B) will be half that of a single driver." Since the Bella Voce uses two woofers in standard configuration, V_{AS} is doubled for use in all calculations. Possibly you are looking in your books at the compound configuration?*

To get a better intuitive idea as to why V_{AS} behaves differently, I suggest you read page 16 of Testing Loudspeakers, *by Joseph D'Appolito. His section 2.5.2 on "air springs, mechanical compliance and V_{AS}" should help you visualize these two situations. Also, pages 17, 27, and 61 of my book,* Advanced Speaker Systems, *will help you imagine V_{AS} in a more intuitive manner and describe why there is a different behavior when using the standard as opposed to compound configuration.*

NO FOOLIN'

I've enjoyed reading some of your publications over the past few years and have found little to quibble about in the articles. They have been sober-minded even when they were obvious commercial messages. However, I must take exception to the article entitled "A 50W/Channel Composite Amplifier" (April '01 *aX*, p. 6) and another one in this same issue.

If this was an attempt at April 1st foolishness, it must be rated as quite successful! However, it is usually considered good practice to inform the readers of this at the end of the article.

An article addressing "Bass-Boosting Network" (p. 24) appears to be a good discussion of crossover networks, complete with test data. The author is credited with a patent application for circuitry to improve incandescent lighting efficiency. Don Lancaster has a review of one such approach on his website, www.tinaja.com. The technique involves operating 30V bulbs from standard 120V line voltage, using a dimmer function to reduce the effective voltage. He indicates that the measurement techniques could not produce accurate results and generally debunks the idea.

If this is the same person or technique, then this article is also suspect. Not a nice thing to provide to your less experienced readers!

The referenced article has all the makings of a commercial announcement—verbiage that applies new words to standard techniques, enough technical discussion to lend an air of validity to the presentation, and so on. The author's use of terms related to feedback and servo techniques certainly indicates his familiarity and expertise in that field. This, however, is not an indication of the efficacy of this design.

The article text seems to be a mix of assembly manual, advertising material, and theoretical tutorial. Well mixed up! Use of the term "composite" seems to be overplayed, since the term is so broad. Almost all amplifiers consist of more than one gain stage, including the op amps used here. To consider a power amplifier as an op amp seems to stretch terminology, too. The first paragraph under "Other Features" seems to equate "compound" with composite, without any introduction. This could well be confusing to a novice.

The schematics on page 6, Fig. 1-3, have severe errors—there is no Vin terminal provided for two of them, and the third has an unlabeled open resistor connection that might be the intended input. Again, very confusing to beginners.

In the main circuit diagram, a curious mix of European and IEEE part value labeling is used, sometimes for the same component type! And the additional definition of mF* as millifarads in very tiny print is difficult to understand.* Consistency would be beneficial, particularly to newbies.

There are two jumpers whose functions don't seem to be explained in the text. It makes reference to "PC100" without any description thereof. Some terms in the text appear to be of European origin; e.g., "stuffing guide," which is more commonly referred to as the component placement diagram.

And, no doubt, Mr. Zobel and the IEEE would certainly appreciate capitalization of the network that is named after him. By the way, it would be nice if such complex schematics were allocated more space.

In paragraph two of this section, the statement is made to the effect that the slew rates of the amplifiers multiply each other. If I understand this comment, it is saying that frequencies that the first stage couldn't amplify will be amplified in a faster stage following the first. Or, to put it differently, if the first stage has a slew rate of 10V/microseconds, a signal of 20V/m-second would not be passed to the second stage, and its 100V/microseconds slew rate would allow the signal to be amplified! NOT! Slew rate and frequency response are effectively inversely proportional to each other, if I remember correctly.

In paragraph 3 of that section, this statement is made: "Two amplifiers in a loop virtually guarantee oscillation..." If memory serves me, that is only true if there is positive feedback. Negative feedback negates this.

Despite the avowed intention of zero distortion, the design seems to violate at least some of the more commonly held beliefs in the field. It said to be operating in Class B mode, which is a no-no to most purists. As is negative feedback. The statement is made that such feedback corrects distortion. If memory serves me, there is little if any connection between negative feedback and low distortion, other than perhaps keeping the active devices in their nominal operating regions. It won't eliminate crossover distortion in a push-pull stage, for instance.

There is a reference to mounting fuse holders to a metal surface with silicone sealant. The insulating properties of the sealant are important here, because many fuse holders have exposed connections on the underside. Note that RTV sealants contain acetic acid, which is why they smell like vinegar, and are not, therefore, a viable choice here. The acid will eat away at the metal parts and cause eventual failure.

All in all, these discrepancies could well lead to many problems in implementing or understanding this design. I am disappointed that it appeared in your otherwise laudatory publication.

Phillip Milks
Indianapolis, Ind.

*Reader Milks' comment is typical of the very provincial attitudes of US engineers as to nomenclature for capacitors. While millifarads are not common for capacitor values they should be. We have no problems with such references in chokes. And nano is a standard in *all* mathematical values. Americans are just too backward (or worse) to use the term. The nanofarad would effectively banish all those 0.001µFs with 1nF and 0.1µF with 100nF—E.T.D.

Kenneth P. Miller responds:

First, given the overall tone of your letter, I probably will not find the right words to change your opinion of my amplifier. Thus I will focus on other objectives.

For one thing, your letter does offer some legitimate criticism. This needs to be addressed and I will do that. Some of your comments are based on a faulty understanding of elementary electronic terms, definitions, and concepts. Here, too, I can help. Finally, some of your judgments are based on misconceptions picked up during your travels through the audio field. I'm not at all sure I can fix those problems, but I'll give it my best shot.

You have one particular misconception that will be very, very difficult, indeed, to dismiss. Most of this difficulty is due to the fact that you are not traveling alone. I will save the tough task for last. I would prefer to address each issue in the same order in which it was raised in your letter, but I can find no way to do this coherently, so I will, in my own way, eventually answer most of your comments.

Despite heroic efforts from the publisher, the author, and the printer, there are mistakes in my article. We both know that. Your assertion that Zobel is a proper noun is correct. I did not catch this mistake when I proofread the copy, nor did the editor. Even worse, I misspelled the word elsewhere in my presentation.

The unlabeled open resistor (Fig. 2, p. 6) should go to ground. I think this a printing error as it did not appear on my proofread copy. In Figs. 1, 2, and 3, the input signal is applied to the + terminal of A1. In Fig. 5, Vin is applied at the + terminal of the BUFF op amp.

I agree that the mix of labels is confusing. I submit all capacitor values in either picofarads or microfarads. The editor prefers other units.

There are mistakes in the amp schematic, Fig. 6, page 7. The collector of Q2 should be connected to the base of Q4. And the input side of C3 should come from the junction of R2 and C2 rather than from the output terminal of A1. The jumpers are also missing on this schematic. These are nothing more than pieces of wire used to connect eyelet 2 to eyelet 3, and 4 to 5. The link between 2 and 3 connects the output of A2 to the input of A3, and the other link connects the base of Q2 to ground. This is the normal condition of the links during composite operation. During the construction process or for troubleshooting purposes, the links can be removed and repositioned such that A2 and A3 can be operated as conventional, independent amplifiers. This detail is of most interest to someone building the amp and covered thoroughly in my separate stuffing guide.

I have no plans to commercially exploit my amplifier. I sell my stuffing guide and boards at cost to those who would like to try my amp. Although it is not illegal to turn my project into a cash cow, it is probably a violation of both the spirit and intent of the magazine. The real motivation and satisfaction is in sharing a better idea with other audiophiles.

The fuse holder I specify has no exposed connections on the underside, and thus the insulating properties of the sealant are irrelevant. Since I'm not a chemist, I can't refute your claim that the sealant will eventually eat away at the metal. But I did check my own unit, which has been in continuous service for the past 16 months and found not a trace of deterioration. I will contact the manufacturer and if he confirms your claim, I will advise builders to use a different mounting system.

The term "composite" is a definition. It is only broad enough to include all topologies in the family.

Two op amps inside a purely resistive feedback will definitely oscillate. The high frequencies will be shifted 180° relative to the lower ones, exactly the right condition for positive feedback. The output will oscillate vigorously, usually at the lowest frequency capable of producing the full 180° phase shift.

I can understand your concern over Class B operation. Optimum Class B is a term referring to recent discoveries and is probably not common knowledge just yet. It is well known that push-pull devices deliver minimum distortion when biased for full Class A operation. If, for the sake of efficiency, the bias is reduced, pushing the output into Class B or AB operation, then distortion will increase. As bias is increased toward Class A operation, you can expect the distortion to reduce.

This is all true with one exception: the bipolar transistor. I use a bipolar output and recommend a bias current of 11.5mA. At this current, my amp delivers minimum distortion out, except for full Class A operation. If the bias setting is changed from 11.5mA to 5mA, for example, distortion will increase due to crossover artifacts in the output. If the bias is increased to, say, 20mA, distortion will increase once again, but not due to crossover artifacts, but rather to a phenomenon known as transconductance doubling. The distortion will stay at this higher level until it is adjusted to near Class A operation where it drops to the lowest level of all.

If you get the drift of what I'm saying, it *should be obvious that, with bipolar transistors, there are only two viable bias options—optimum Class B or full Class A. Why not use a different output device, such as a power MOSFET, since it is immune to transconductance doubling? Because any bipolar pair operating optimum Class B will produce lower distortion than any pair of MOSFETs biased at Class AB. The same is true at Class A. There can be legitimate reasons for using MOSFETs in the output stage, but there is always a distortion penalty.*

Your statement that there is no connection between negative feedback and low levels of distortion is simply untrue. I suspect that your claim is based on some "zero-feedback" amplifier having respectable distortion figures.

The catch is this. The manufacturer here means zero global feedback. His amp uses local negative feedback and gobs of it. There is no way to have respectable distortion figures without the heavy application of negative feedback. Negative feedback is negative feedback, whether local or global.

There is a misconception about feedback, which goes something like this: Negative feedback in an amplifier is generally a bad thing. The less feedback used, the better—zero feedback is best. This all started some years ago and, of course, has been heard by designers and others. Their reaction to it? Most audio designers do not buy it, and academia totally rejects it. That's why universities are still teaching conventional negative feedback theory.

In a nutshell, it goes something like this. Negative feedback is a powerful tool, so use it. The one and only penalty for too much negative feedback is instability. Instability is hardly a subtle defect and easy to cure—back off the feedback level a bit until stability is sufficient.

Let me say at once that I do not question the sincerity of those who believe that negative feedback is a bad thing—they are quite sincere about their belief. But the truth or reality of any given proposition cannot be determined from the sincerity of convictions. Most mistakes are honest. The preponderance of evidence is what matters and it supports the use of negative feedback. As far as proof goes, it can't be had—but only because the purists cling to a disprovable hypothesis.

TONE CONTROL

Thanks for publishing the project on "An Unusual Tone Control" (April 2001, p. 50). For years I've been looking for a loudness control schematic that did not involve exotic multi-pole switches or tapped potentiometers. There is, however, a detail missing from the parts list and text. What type of potentiometers are P1, P2, P3, and P4? Logarithmic or linear?

Simon Balderson
S. Balderson@btinternet.com

Tech staff here thinks they are all log taper. Some have suggested that the unit would benefit from an added cathode follower output stage.—E.T.D.

BEST EDITION

I enjoyed Larry Lisle's "Books for the Tube Audio Beginner" (April '01, p. 76). I have most of the books he lists, including my introduction to tube amplifiers (the two-stage 6BF6 job!), circa 1965, at the tender age of 8 years: Alfred Morgan's *The Boys' First Book of Radio and Electronics*. I would like to know if, possibly, he can recommend *one* particular year in the 1950s that the *ARRL Handbook* may have stood out as the "best" edition for tube audiophiles. With ten editions (1950-1959) at about $30 a pop (from one used book source), it might get expensive for me, as well as redundant, to build this segment of my library!

John Agugliaro
JAGUGL4546@aol.com

Larry Lisle responds:

Thank you for your interest. The ARRL Handbook *changed very little from year to year. Editions from the late '40s and early '50s had a greater variety of building techniques, using wooden racks as well as all-metal construction. Later editions will have more recent tube data. These are about the only differences for our purposes, and I'm sure you will find any* Handbook *useful.*

VINTAGE AMPS

Going through the pages of a recent edition of *audioXpress* (March 2001), I became very interested in the article about remaking the Dynaco Mark III by Kara Chaffee (p. 18). The reason I am so interested is that I also own a vintage amp, which is a Harman Kardon Citation V in original working condition. I would like to remake it like Kara Chaffee did with the Dynaco Mark III.

Would it be possible to see this project in the pages of *audioXpress* soon? I would like to know whether this amplifier has ever been rebuilt to today's standards, or if anybody out there is interested in this project.

Fernand Binette
Victoriaville, Quebec

SHED SOME LIGHT

As a reader of your magazine and tube audio enthusiast, I was pleased to see one of JKL's products mentioned in "Switching Power Supplies for Tube Preamps" by Eric Barbour (April 2001, p. 16). I'd like to add a few notes on the JKL 12200 lamp inverter. The 12200 is the hobbyist version of the JKL BXA-12529, which is designed with pins to mount onto a circuit board. The BXA-12529 and other JKL inverters, including EL inverters, are available through Digi-Key (WWW.DIGIKEY.COM). As seen from the accompanying schematic (*Fig. 2*), the DC-AC inverter is a Royer-type oscillator circuit; the frequency of the oscillator is about 35kHz.

A word of caution: The open-circuit output voltage of the inverter can exceed 1000V RMS, as it is primarily intended for ignition and operation of discharge lamps. Capacitor C1 is used to give the output waveform a more sinusoidal shape; capacitors C2 and C3 split the output from the transformer for two lamps and provide a means of ballast for the lamps by limiting lamp current to about 5mA RMS each. When using the BXA-12529 (12200) as a high-voltage DC-DC converter, you should use diodes and capacitors that can handle the aggressively high voltage and frequency.

I look forward to more insightful articles in your publication.

John Kahl
JKL Components Corp.

SOUND RECOMMENDATIONS

In his review of Cool Edit Software in the April issue of *audioXpress* Perry Sink mentions that you must be careful what sound card you use for recording, and that some cards introduce significant noise pickup or have marginal signal-to-noise ratios. From his experience, can he tell us any sound cards that should be avoided, and also which sound cards he recommends?

Richard Mains
Ypsilanti, Mich.

Perry Sink responds:

I have done no personal evaluation of various cards myself, other than to note that the typical card that comes in a PC is most likely built to minimal standards and is noise-prone. If you get excellent sound from such a card, you're a bit on the lucky side.

However, I know that many high-performance cards are available at price points up to $1000+, often designed with serious 24-bit home recording in mind. In the middle are a number of excellent products for under $100.

The following are some resources on the web that may take you a step further:

http://www.epanorama.net/pc/sound.html
http://www.music-software-reviews.com/sound_cards_PC.html

Also, check sources such as ZDnet, PC Magazine, *and various musician magazines. Note: Turtle Beach cards have been very*

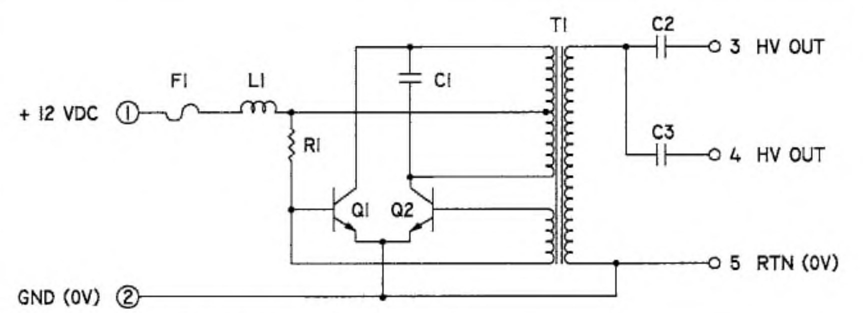


FIGURE 2: JKL's BXA-12529 inverter.

positively reviewed in PC Magazine *by Craig L. Stark.—E.T.D.*

I SEE ICS

Regarding Charles Hansen's column "Chips Gone from the Block" (April 2001, p. 89), I'm sure you'll get a ton of e-mail pointing out that datasheets for both the BA1404 and the TC9147 are freely available for download from http://www.freetradezone.com/ (Actually, they had three different versions of the Rohm datasheet, and just one version of the Toshiba datasheet.)

Scott Newell
Fort Smith, Ark.

Charles Hansen responds:

Thanks for the information. This site is certainly worth a Favorites/Bookmark!

HELP WANTED

Does anyone have the instructions and drawings for the Le Petit Onken? Also, do you have enclosure instructions and drawings for any Norelco or Philips full-range 12″ drivers? These enclosures were made back in the late 50s. I have some vintage 12″ drivers and wish to build the Le Petit Onken for them, and I was hoping that some reader would have the solution to my problem.

Evangelos Fondas
efondas@mercymiami.org

I'm trying to find out the value of a Jensen model 610, 15″ triaxial with bridge tweeter circa 1953. I've been told that it is the holy grail of vintage speakers. Can anyone enlighten me?

Paul Baron
f246@compaq.net

Many years ago (approximately 20), I subscribed to the *Audio Amateur* magazine (and *Speaker Builder*) when I was a Professor of Chemistry at Syracuse University. A few years ago I retired from teaching and settled in Keene, N.H.

Searching the web, I noticed that the successor to these magazines is headquartered in Peterborough, N.H. I was disappointed that there were no audio amateur clubs listed on your website (not even in Peterborough!) except for one in Connecticut (no address given). Are there no groups of people in the Keene-Peterborough area who are interested in getting together to listen to good, well-reproduced music and talk about audio (the old Hi-Fi)?

Clarence Pfluger
clep2@earthlink.net
Keene, N.H.

I have a pair of Clements 206di speakers, which I enjoy, but was wondering if some modifications or upgrades would be possible. To begin with, the binding posts could definitely be upgraded, and I thought maybe a similar crossover with higher-performance parts could replace the existing one. I am not an expert in speaker building, but I have had my DAC modified by an expert named Dan Wright in Portland, with very good results. Could anyone provide information and/or service on modifying or upgrading speakers?

Jason Smith
jssdg3@yahoo.com

I'm looking for information and notices about the triphonic JR loudspeakers (model lpa). I think it's 15/20 years old.

J.J. Guicheney
JJ.Guicheney@noos.fr

I am building a box for a 1998 Dodge Ram 1500 quad cab for 2 rockford punch 10″. Is there box designer for this?

Robert Dicks
kwbeav1972@yahoo.com

I have an Altec 5″ tweeter that is not working, and I wish to find out how to rebuild it. ❖

Greg Muellerleile
blues@dailypost.com

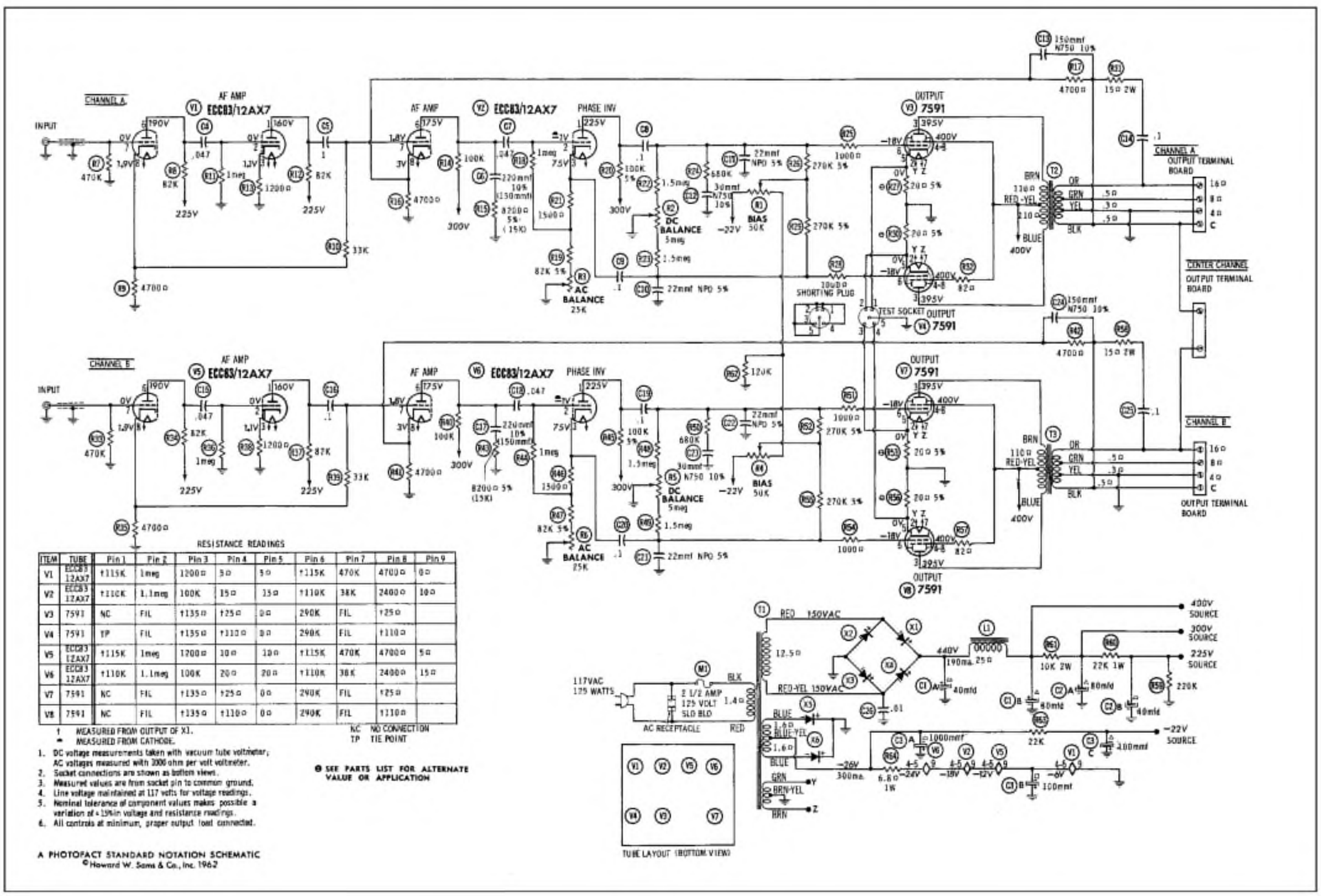
Readers with information on these topics are encouraged to respond directly to the letter writers at the addresses provided. –Eds.

Classic Circuitry

Pilot 264

This is a seemingly generic-looking basic stereo power amp from 1962, but look at the output section. The 4Ω tap is grounded! Audio Research did not originate that concept. Further, feedback comes both from the ungrounded "common" tap and from the 16Ω tap. Most unusual. ❖

Lance Cochrane
San Francisco, Calif.



Pilot 264.

Circlotron Amp

from page 29

the Circlotron. It is also the key feature of the 6AS7/6080 I described in the earlier *GA* article.

Now, if someone would contribute the transformers, we would have a project. ❖

*Obviously bitten by the Circlotron bug, John Stewart, in a subsequent issue, provides further details about this exotic amp...and enough information for you to build your own.–**Eds.***

ENCLOSURE VOLUME

In figuring box volume, a higher V_{AS} speaker would require a larger box than a speaker with a lower V_{AS}. By definition, V_{AS} is the volume of air having the same acoustic compliance as the driver's suspension. To me that sounds as though a low V_{AS} driver would have a stiff suspension and a high V_{AS} driver would have a loose suspension. But doesn't a loose speaker require a small box and a tight speaker need a large box?

Wilbur Cothrun
lcothrun@prodigy.net

Dick Pierce responds:

First, you are correct about the meaning of V_{AS}: it is the volume of air having the same effective acoustic compliance as the driver, and there is a direct relation between the mechanical compliance C_{MS} of the driver and its V_{AS}. The following equation shows that relationship:

$$V_{AS} = \rho_0 c^2 C_{MS} S_D^2$$

where ρ_0 is the density of air (about 1.18kg/m^3), c is the speed of sound (342m/s), C_{MS} is the mechanical compliance of the driver suspension (in m/N), and S_D is the effective emissive area of the diaphragm. This equation confirms your notion that a high V_{AS} implies a high C_{MS} and vice versa. And the correlation is direct, assuming the cone diameters are the same.

However, your guess—as intuitive as it might seem—that a driver with a large V_{AS} needs a small enclosure and vice versa is not correct. One of the problems with making such a sweeping generalization is that it ignores a large number of important factors.

In a sealed box system, for example, the combination of the driver's V_{AS} and the enclosure volume V_B together determine the total mechanical stiffness of the system. That stiffness, combined with the effective moving mass of the cone, determines the system resonance. So, you could correctly conclude from this that the proper enclosure volume V_B for a given driver is that needed to raise the system stiffness to achieve the target system resonant frequency. This conclusion makes no suggestion that a large V_{AS} requires a small V_B, or vice versa.

There is another relationship that's important here. Raising the resonant frequency also raises the system Q at resonance. The Q increases directly as the system resonant frequency increases.

These two parameters—system resonance and system Q—change together in a predictable fashion. In a closed box system, for example, there is a parameter, α, which is the so-called "tuning ratio." It is the ratio between the driver equivalent volume V_{AS} and the enclosure volume V_B:

$$\alpha = \frac{V_{AS}}{V_B}$$

The change in resonance and the change in system Q can be calculated once you know the tuning ratio. For example, you can calculate the resonant frequency of a driver in the enclosure F_C from the tuning ratio and free-air resonance of the driver F_S:

$$F_C = F_S\sqrt{\alpha + 1}$$

Similarly, you can calculate the system's electrical Q, Q_{EC}, from the driver's free-air electrical Q, Q_{ES}, as follows:

$$Q_{EC} = Q_{ES}\sqrt{\alpha + 1}$$

From this, I hope it's clear that it's the ratio of the driver compliance to box compliance that's important, regardless of whether you have a high-compliance or low-compliance driver. For example, if you had two drivers, both with the same free-air resonant frequency and the same free-air Q, but with different equivalent compliance volumes, to achieve the same target resonant frequency and system Q, the ratio of the V_{AS} to V_B for both speakers would need to be the same. The result is that the speaker with the larger V_{AS} would require a large enclosure volume V_B.

Now, seldom do you encounter such a situation in actual practice (though there is no inherent physical reason why it couldn't happen), but the example serves to illustrate the point that the necessary enclosure volume is not inversely related to V_{AS}. Rather, the ideal cabinet volume is determined by the desired system response and the total of the driver's electromechanical parameters. There is no hard-and-fast rule that suggests that high-compliance speakers require small boxes and vice versa.

Part of your confusion may derive from the original claims made by Acoustic Research back in the early 60s. They used a very high-compliance driver in a very small enclosure, at a time when people generally used low-compliance drivers in large enclosures. The latter is easy to explain; people simply did not understand how loudspeakers and enclosures worked as a system. The breakthrough that Villchur and Allison achieved was to consider the driver and enclosure as a system, though it would take Thiele and Small to generalize the principle.

The AR idea was to marry a high-compliance, high-mass driver with a low-compliance enclosure. The primary reason for doing this was that it was the enclosure that largely determined the total system stiffness. In AR and similar "acoustic suspension" systems, the tuning ratio, α, was often on the order of five and higher. The consequence of this is twofold.

First, it meant that the system could tolerate large variations in the driver compliance, and those variations would have a minimal effect on the system overall. For example, if the enclosure compliance was ⅕ that of the driver compliance (a tuning ratio of 5), any variations in driver compliance on the total system compliance will be reduced by a factor of 6. A ±25% variation in driver compliance will result in only a ±4% or so change in system compliance, and then only about a ±1% change in system resonant frequency.

Second, it is the enclosed air that provides most of the restoring force for the cone, not the driver's suspension. In that way, the more linear operation of the entrapped air overwhelmed the nonlinear mechanical suspension. However, as a tradeoff, the acoustic suspension system requires *a driver with a high moving mass to achieve a low system resonance, and that combined with the long-overhung voice coil resulted in a system with very low efficiency.*

This serves further to illustrate the point I am trying to make: the enclosure size is determined by the parameters of the driver (compliance, Q, resonant frequency, and so on) and the target system performance (system resonance, system Q). In some designs, that means a high-compliance driver in a small box, in other systems, it may mean something else.

ZOBEL POWER DISSIPATION

As an avid subscriber to your journal, I was wondering whether I could bother you for some technical advice concerning Zobel networks in two-way crossovers. It seems the problem is in the power-handling characteristics of the board we've designed. Because the Zobel network equalizes the impedance of the woofer in the upper part of the woofer's range, it necessarily must dissipate some power. My calculations show that, over a narrowband of frequencies, the power required is 40-50W when an amplifier is putting 100W into the speaker at 8Ω.

In order to claim the real power-handling characteristics of a speaker, is it necessary to "beef up" the Zobel network to account for the power dissipation through the network?

Bob McChesney
BobM@Port of Everett.com

G. R. Koonce responds:

The power requirement for resistors used in driver Zobels has always been a bit of a mystery. The "standard" Zobel consists of a series-connected resistor/capacitor network connected across the driver terminals where the resistor value in ohms nearly matches the driver resistance. It is clear that when such a network is driven by a voltage source, the current through the network increases as frequency rises. The maximum Zobel resistor dissipation will thus tend to be at the top end of the crossover range for each driver/Zobel combination.

If the woofer crossover frequency is high enough—likely with a two-way system—the Zobel resistor may dissipate more power than the woofer at the top of the band because of the woofer's rising input impedance. That is exactly what the Zobel is supposed to do! The problem is how to size the Zobel resistor for your intended application.

Just like semiconductors, resistors need derating. For high ambient temperature applications such as in military gear, a resistor is used only to half its nominal rated power. Typical commercial application would require less derating. In a high-power speaker system where the Zobel is mounted inside the box with all the driver dissipation, you should consider the possible rise in ambient temperature within the box. The remainder of this letter refers to the derated value of the Zobel resistor.

Narrowband Applications

In a case where the input signal may be a sinusoid or other narrowband signal, you can compute the actual worst-case Zobel resistor dissipation, as when you plan to test with sinusoids at high power or pro-sound applications such as bass guitar or organ music. Here you can take the maximum input voltage the system will see, and at the maximum frequency the Zobel will compute the current to the Zobel resistor and thus the dissipation. The computation should include any "peaking" in the crossover network that can subject the driver/Zobel combination to a voltage somewhat higher than the maximum system input voltage. In these cases you may find the Zobel resistor must be rated to the same power rating as that of the driver.

Music Applications

When you play music, the Zobel resistor has reduced power dissipation as it receives less power at the low end of the driver frequency range and music tends to spread the energy out over frequency. Many years ago, my computations of the power a Zobel resistor could dissipate matched those results reported by Mr. McChesney; i.e., the Zobel resistor could see about one half of the power delivered to the driver/Zobel combination. Thus for a 100W woofer pushed to its limit, you might need a Zobel resistor rated at 50W with music.

Practical Results with Music

I normally develop speaker systems with 100W amplifiers. With the high peak to average power requirements of CDs and LPs, this means about 10W-20W of average power available. The limiting used on FM would allow a somewhat higher average power from a 100W amplifier, but 10-20W produces loud music and I normally would not exceed this level.

I started developing systems with 10W resistors in the Zobels, and after playing quite a while at 10-20W average, the resistors were never even warm. Thus they clearly were not dissipating anything like the 5W to 10W the worst-case calculation would indicate. Why was this?

First, while the music may reach 20W average for some period of time, it generally does not stay at this level. Thus some time averaging of the dissipation occurs. Power resistors have the ability to take very high power dissipations for short periods of time; it is the average dissipation over time that sets the resistor heating.

Next, testing showed that with a three-way system with the lower crossover frequency in the 800Hz range, about one half of the average system input power goes to the woofer and the other half to the midrange and tweeter. The tweeter normally receives about 10% to 15% of the power. For a two-way system with its relatively high crossover frequency, it's best to assume the woofer sees all the system input power.

My current rule of thumb for home systems is that I use at least a 10W resistor on the woofer Zobel and at least a 5W resistor on the midrange Zobel. The lower power rating on the midrange is based on the fact the midrange is normally more sensitive than the woofer and thus has some attenuation ahead of the Zobel. I normally don't worry about the tweeter Zobel resistor power at all; I use 5W units for convenience. In over 30 years I have never had a system returned with a failed Zobel resistor.

Why not just use high-power resistors, which are relatively cheap? The problem is finding room for these physically large resistors, which will run cooler than smaller ones but probably actually put out more heat. This is because the resistor types normally used (wire wound) have a resistance that rises as they heat up, making them somewhat self protecting. This resistance change will, however, slightly modify the Zobel from its design values.

As noted, there are some cases that require special attention to the dissipation in the Zobel resistors as follows:

1. *Very high average system input power. This would be in music or pro-sound application where the driver is pushed*

(continued to page 95)

New Chips on the Block

National Semiconductor LM 4651/4652 Class-D Amp

By Charles Hansen

The IC combination of the National Semiconductor Overture® LM4651 driver and the LM4652 MOSFET power amplifier provides a high-efficiency, two-channel, Class-D subwoofer amplifier. Applications include powered subwoofers for home theater and PCs, car booster amplifiers, and self-powered speakers.

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Maximum power at 8Ω, THD ≤1%: 75W
Maximum power at 4Ω, THD ≤10%: 170W
Maximum power at 8Ω, THD ≤10%: 90W
Signal-to-noise ratio: 80dB (typ)
THD+N 0.3% (typ), 10W, 4Ω, 10–500Hz
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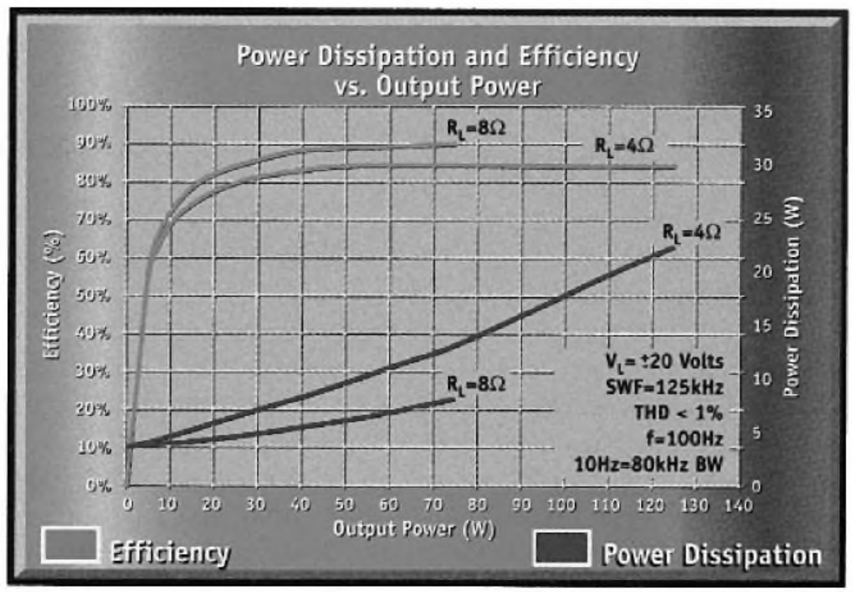
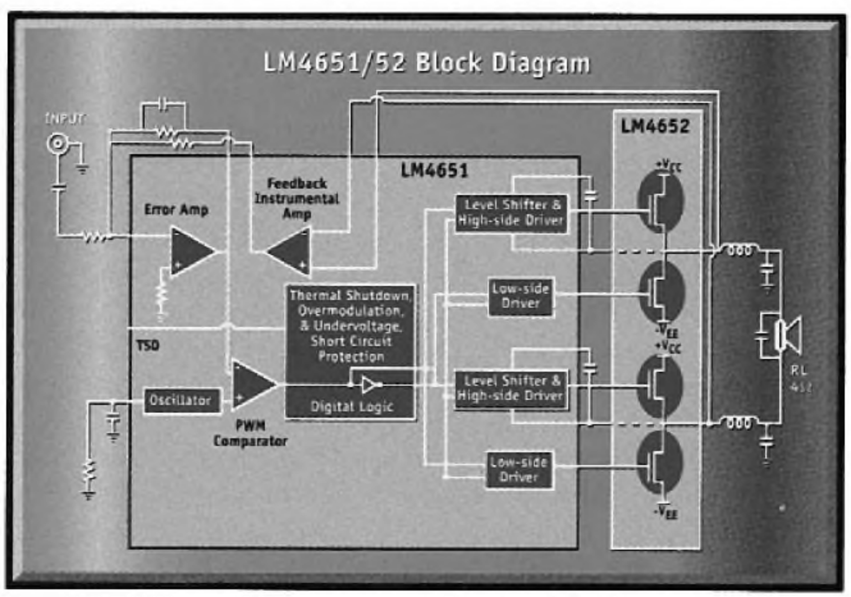
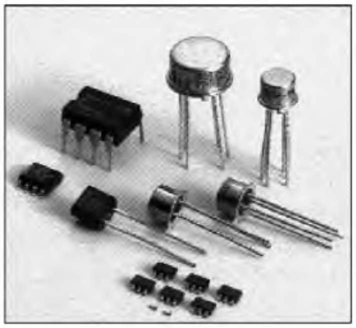




New Chips on the Block

Linear Systems

By Charles Hansen



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Glass Shard

Choosing Coupling Capacitor Values

Along with negative feedback, the seemingly innocuous and trivial coupling capacitor is arguably one of the greatest single factors you can easily control in determining frequency response of an amplifier. Other factors, such as the Miller effect, the upper frequency response limit, and the output transformer response, are somewhat set and not easily altered.

In correspondence with Mr. Dell, recently, he questioned the size of the coupling caps I was going to use in my Eico HF-60 amplifier. This raised a question in me, how do you go about selecting the value of coupling capacitors in an amplifier? So I hit the books to see what the experts said, and then started to experiment with my amplifier.

FORMULAIC SOLUTION

Let me first relate my understanding of how to derive the values for the coupling caps. The Eico HF-60 manual states that the lower frequency limit is 20Hz. The coupling cap of 25nF and the output tube grid resistor of 330kΩ form a simple RC high-pass filter. By plugging these two values into the formula for an RC filter circuit, you can find the –3dB rolloff point of the high-pass filter:

$$freq = 1/(2\pi RC)$$

which equals 19.3Hz, very close to the 20Hz lower limit.

Electronic Designer's Handbook (Robert W. Landee, et al., McGraw-Hill, 1957, p. 3-19) confirms this as long as R is much greater than the parallel combination of the driving tube's plate load resistor and its dynamic plate resistance. For a more accurate value for C, add this parallel combination resistance to the grid resistance in the previous formula. But that only slightly increases C by a few nanofarads, which is not much difference.

F. Langford-Smith's *Radiotron Designer's Handbook* (p. 483) also says the preceding formula is valid, but adds, "In high fidelity amplifiers a fairly large value of C is generally adopted, thus not only improving the low frequency response but also reducing phase shift and possibly also improving the response to transients. However, excessively large values of C are undesirable." This last sentence is particularly true because with negative feedback excessively large coupling caps could lead to distortion and oscillation. But I wonder how large he meant by "a fairly large value of C"?

Radio Handbook (William I. Orr, Editors and Engineers, 1970, p. 147) gives an empirical formula for coupling caps. "For high-fidelity work the product of the grid resistor in ohms times the coupling capacitor in microfarads should equal 25,000 (i.e.; 500,000 ohms × 0.05μfd = 25,000)." In my case CμF would equal 25,000/330k, which is .076μF or 76nF. Plugging this value for C back into the formula for an RC filter circuit (just to see where the new low limit would be) comes to 6Hz, which is quite a bit lower than the original.

REACTANCE

Another way to look at the capacitance of the coupling caps is from the standpoint of capacitive reactance. The reactance of a 25nF cap at 20Hz is $X_c = 1/(2\pi fC)$, or 318kΩ. The reactance of a 76nF at the same frequency is 105kΩ. Quite a large difference! From this view, using the larger cap would make more sense, but again, this is predicated on the condition that it does not cause instability, oscillation, or distortion. ***Table 1*** shows the reactances of the two caps versus frequency from 1Hz to 20kHz. You can see how capacitive reactance affects the whole audio frequency band, not only the lower limit, but the upper end as well. The 78nF cap is about three times larger than the 25nF, and about three times lower in reactance and in the low-frequency limit, too.

I realize that these equations and formulas—empirical or otherwise—are just intended as a starting point. You must experiment by increasing the capacitance (lower the low-frequency limit) until the amplifier shows excessive distortion or oscillates (if it does), then back it off for a safety margin. This is the value to use. You do this, of course, using an oscilloscope connected to the properly loaded output, with some kind of generator injecting an audio test signal to the input.

In my particular case, after trying various values, I ended up using 47nF coupling caps. Using larger values caused high-frequency oscillations to ride piggyback on my output, test square waves, but the 47nF behaved perfectly through the whole audio range. The new low-frequency limit is now 10Hz, and my amp sounds sweeter, with a more robust bass. So with a little experimentation I was able to successfully double the size of the original coupling capacitors and improve the response, while introducing no degrading side effects. ❖

Michael Kornacker
Palatine, IL

TABLE 1
REACTANCES OF A 25NF COMPARED TO A 76NF CAP VERSUS FREQUENCY ALONG WITH THE 47NF.

FREQUENCY IN Hz	Xc FOR 25nF (Ω)	Xc FOR 47nF (Ω)	Xc FOR 76nF (Ω)
1	6.4M	3.3M	2.1M
10	636k	338k	210k
50	127k	68k	42k
100	64k	34k	21k
200	32k	17k	10k
400	16k	8.4k	5.2k
600	11k	5.6k	3.5k
800	8.0k	4.2k	2.6k
1k	6.4k	3.4k	2.1k
2k	3.2k	1.7k	1.0k
4k	1.6k	850	520
6k	1.1k	565	350
8k	800	423	260
10k	640	340	210
12k	530	282	175
14k	450	242	150
16k	400	212	130
18k	350	188	116
20k	318	169	105

Classifieds

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I have 12 pairs of Hitachi J56 and K176 MOSCODE and want to build a MOSCODE 600, but I don't have any of its design info. Can anyone help me? I can spend some money for its PCB, schematic, and operation guides. C.K. Wong, chi_keung_wong@hotmail.com.

I have an old 1986 vintage Akai CD player, model CD-A30, but no manual or schematic for it. It has a multipin outlet on the back labeled "Digital Subcode." I am trying to figure out what outputs are available through this outlet. Any information GREATLY appreciated. Please contact Don, dleitner@sju.edu.

HP339A audio distortion analyzer. Must be in good working order. Andy Rawling, FAX +1 630-214-3687, e-mail andy_rawling@ yahoo.com.

Two Focal 7N402DBE drivers, preferably NOS. Ruud, Phone/FAX 415-383-4958.

Editorial

from page 2

today. But not here. It cost NASA a Mars satellite the other day.

If you argue that decimalled farads are like Gary Galo's (one of our regular contributors) favorite, the current flow issue, in that the millions of textbooks have all adopted the "mistake," and it would be just too expensive and difficult to start changing all the literature at this stage of our existence, then I do not believe the issues are comparable. We have made the change from the old micro-micro farad of the thirties and forties. We could easily begin changes in the books, catalogs, and spec sheets starting in the new century. Most of the European electronics press already has this style in place.

One historical note might be in order here. We honor Faraday and other discoverers of electronic properties by designating such quantities with a number and the capital letter of their surnames. Faraday's µF, Henry's µH and mH, Nicolo Volta's 1V or µV. Poor Georg Ohm has the misfortune to have a name whose capital is too similar to the zero, so the elders resorted to the good old Greeks again, picking the omega, Ω, to designate the ohm. When we spell these quantities, however, the proper use is lowercase, which indicates that we're talking about a quantity, not the distinguished person.

It is time to stop decimating the farad. The standard math notation for these little quantities is much more economical, rational, logical, and also neater. These magazines will continue to convert those ugly decimals to their proper designators. I hope you will spend some time thinking through the change and adopt it for yourself.—E.T.D.

This column originally appeared in GA *6/99,* SB *7/99, and* AE *6/99.*

Test Tracks

from page 96

5. Fleetwood Mac, *The Dance*. Warner Records 38486-2. This group's personal conflicts and music defined a generation. Twenty-two pure gold songs. First-rate cinematography. Double-sided disc with 5.1 Dolby Digital on one side, stereo LPCM on the other. Solid live quintet sound, with a substantial amount of backup musicians. Plenty of instrument details, but the soundstage is not quite there. The bass is a little overwhelming on the stereo track, but it is nothing that cannot be compensated with the bass control. Lyrics may be subtitled.

6. Peter Frampton, *Live in Detroit*. NuagesMusic D88161GDVD. Nobody rocks the house like Peter Frampton. This is a solid performance of classic Frampton songs. Extra-crisp widescreen video thanks to HDTV mastering. Plenty of bonus features. Available surround tracks in both 5.1 Dolby Digital and DTS. Choose the latter. The four-piece rock band does not have much soundstage, but the recording has engaging audience ambience. The stereo track is heavy on bass, so turn down that control. ❖

Fernando Garcia
Brownsville, Tex.

Let's hear from you. Simply describe your seven favorite pieces (not to exceed 1,000 words); include the names of the music, composer, manufacturer, and manufacturer's number; and send to "Test Tracks," Audio Amateur, Inc., Box 876, Peterborough, NH 03458. We will pay a modest stipend to readers whose submissions are chosen for publication.

Ask AX

from page 89

to its rated electrical power: the case Mr. McChesney raises. There is the implicit assumption here that your box design is not displacement-limited at any frequency so that the electrical power limit applies.

Another case would be the home user who uses 1kW amplifiers and plays at very high power levels. If I was sure the application was playing full bandwidth "typical" music, I would size the Zobel resistor at about one quarter the driver rating. If the music to be played maintained a constant high level, then I would raise the Zobel resistor rating to one half the driver power rating.

2. *Narrowband input frequency range. This is where the system may see a high power input at a single frequency or narrowband of frequencies. Here, for safety, I would use Zobel resistors with a power rating matching the average power you plan to put into the system. To consider physically smaller resistors, I would measure or calculate the maximum Zobel resistor dissipation that could occur.*

3. *Multiple driver systems. Remember, if you have a group of woofers using a single Zobel, the resistor (and capacitor) must be sized for the power to all these woofers. Multiple Zobels might make better sense in terms of physically smaller resistors and less current through the Zobel capacitors.*

Summary

So, in general, for home speakers used at rational power levels, I get by with 10W Zobel resistors for the woofer and 5W for the midrange driver. When running a woofer to its full power rating, the Zobel resistor dissipation might approach one quarter of the full power rating of the woofer, depending on the application. Not knowing the application, I'm afraid I can't be more specific than this.

As always, experience is the best guide. If you are not having problems with the Zobel resistors as you are currently sizing them, then why change? I address the subject of power dissipation in the resistors used in crossovers more generally in SB *8/96 ("SB Mailbox," p. 51).* ❖

Test Tracks

Reader-submitted favorite selections to test audio systems.

Those who have experienced music videos on DVD know that it will revolutionize the way music is enjoyed, especially for live performances. Not only does it have the benefits of the plain old CDs, but its added visual dimension provides a stunning extension to the listening pleasure.

Additionally, a well-produced DVD has a host of features that conventional CDs can only dream of. True, some newer CDs feature video clips—with puny resolution—and computer interactive software, but the medium is constrained by its limited capacity. DVDs offer that plus high-quality video clips, still pictures, bios, lyrics, interactive features, and much more. Some come with selectable camera angles.

And the sound...well, the sound will literally blow you away. Many people will hotly argue this statement. I believe that it may be related to a bias—or, shall I say prejudice—against multichannel sound. I must confess I was in that same group. After all, surround sound employs lossy compression, and we audiophiles are always complaining that even linear PCM steals away precious music information.

Well, don't despair. DVDs always carry an LPCM stereo track with at least 48k/16-bit sampling; you can always listen in that mode if you wish. But the good news is that audio engineers are finally catching up with surround sound's capabilities and limitations. Early surround recordings, especially those found in movies, were artificial, aggressive, and in-your-face—more a technology showcase than a musical experience. But an increasing number of music videos, some of which I have picked here, have a three-dimensional quality and open soundstage in the surround mode that cannot be reproduced by only two speakers, no matter how high their quality.

About the only complaint I have is that it can quickly escalate into an expensive proposition. Think of it as an audio hobby on steroids. You have the added expense of not only the audio components (multiple speakers and amplifiers and surround decoder), but that of the television monitor itself. The high video resolution will leave you starving for a large-screen monitor with component or S-video inputs, interlaced to progressive scanning conversion, or maybe a widescreen format. Hey, an upgrade to a HDTV set would be nice!

Second mortgage, anyone?

1. Eagles, *Hell Freezes Over.* Geffen Records ID5529EADVD. The standard by which music DVDs will be measured for years to come. The production is impeccable. Video quality is first rate. The sound quality is outstanding and detailed.

This DVD features both LPCM and DTS-encoded tracks, and I like listening to it in one mode or the other depending on my mood. The stereo LPCM track is incredibly tight and detailed, with powerful percussion and precise imaging. But it is the 5.1 channel, DTS-encoded track that really shines. My personal judgement, which I have seen expressed many times over, is that DTS is superior to Dolby Digital for music. The DTS version of the songs, although not as detailed and powerful as the stereo track, nevertheless offer the best soundstage I have discovered in music DVDs and an impressive three-dimensional sound. This DVD features an audio-only DTS track that is a showcase of superbly executed surround sound. And the music, well, you know the Eagles songs.

2. The Moody Blues, Hall of Fame: *Live from the Royal Albert Hall.* Threshold Records ID9757TEDVD. British band The Moody Blues has always attracted fans of soft rock for its lush, orchestral sound, of which the hit "Nights in White Satin" is a premier example. The quartet plays alongside the World Festival Orchestra at the fabled Royal Albert Hall. The video takes full visual advantage of the opulent venue. The digital video production is striking; the images are so detailed.

The audio portion features three different tracks: stereo LPCM, Dolby Digital 5.0 and DTS 5.0, and I would recommend the latter. This is not a typo—5.0 is the correct format description, and it means that the low-frequency effects channel is not enabled. The recording does not have killer bass, and the LFE channel is not really required.

The recording does have a stupendous amount of detail and space, although image wanders a little. Hall ambience is fantastic.

3. John Fogerty, *Premonition.* Warner Records 38496-2. At first glance, the opportunity to listen to some old Credence Clearwater Revival songs with just a member of the original CCR lineup is not very appealing. But John Fogerty was CCR, and the addition of new musicians with renewed spirit provides for an incredibly energized rendition of the old songs, in some instances livelier than the original versions. Watch for the extremely energetic drummer. Encoded in 5.1 Dolby Digital. Lyrics may be subtitled.

4. Rick Wakeman, *The Classical Connection.* Beckmann Communications ID8814CLDVD. Keyboard maestro comes alive in this DVD. Watching Wakeman's fingers fly over the keyboard is a treat in itself. The music includes classical arrangements of popular music. The maestro is supported by veteran guitar player David Paton. This album is best heard in stereo, as the 5.1 Dolby Digital track does not add any significant depth to the soundstage.

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