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Build a 50W HYBRID AMP



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Finding a TL Home
For 3" Drivers

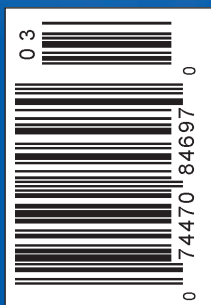
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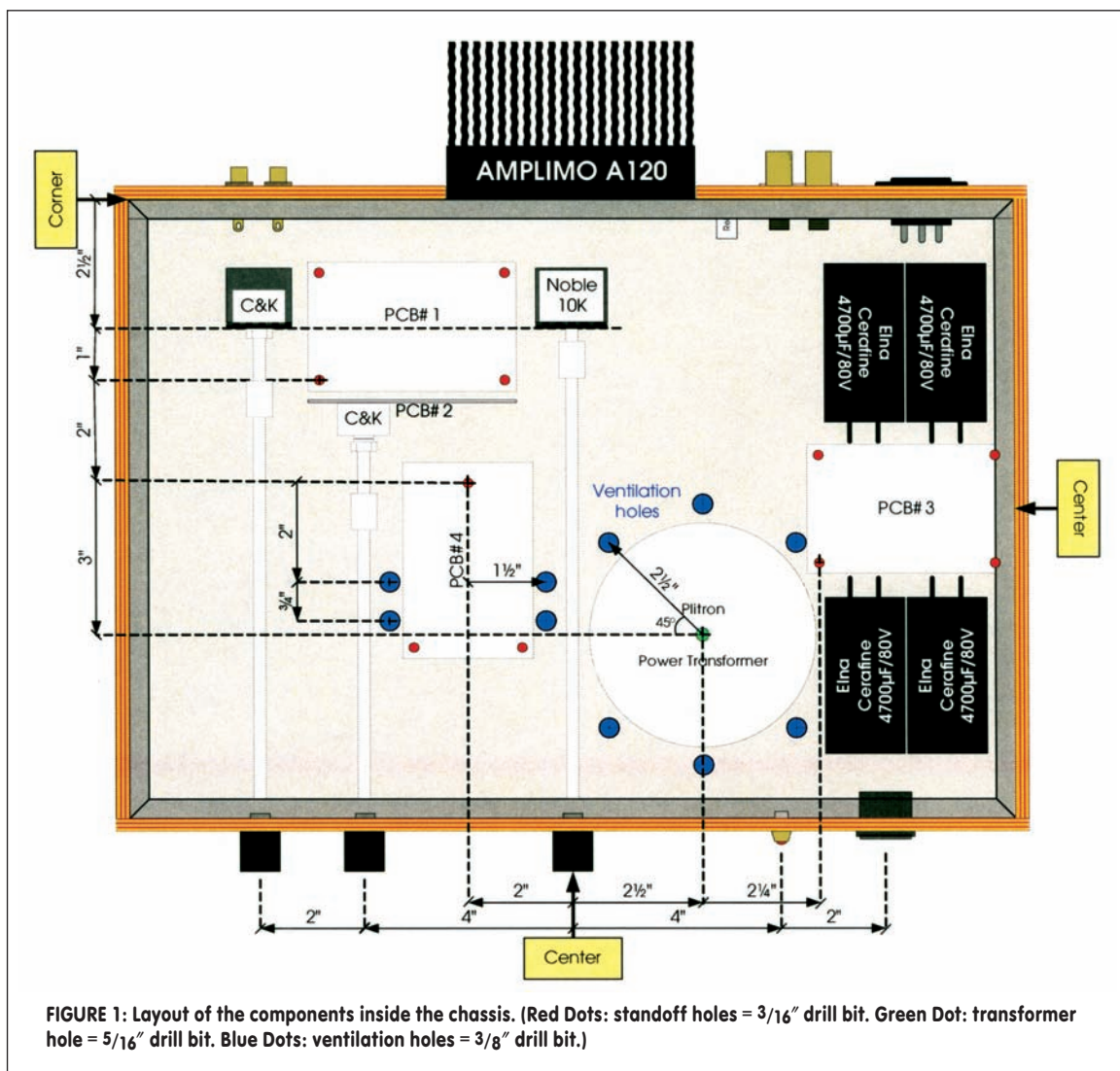


After discussing the theoretical and practical aspects of the circuitry in the first two parts of this series, it's time to put it all together and build the subwoofer amp.

In my opinion, having the leisure to produce something of your own is one of the most rewarding aspects of this hobby. Even with a limited set of tools, you can build a piece of equipment to be proud of.

When I started to design this cabinet, I decided to build a low-profile amp because my space was quite limited. This particularity turned out to be an advantage from an aesthetic perspective because low-profile amplifiers are usually rather elegant. I also wanted to integrate various materials to give it a unique personality. **Photo 1** shows the completed subwoofer amp.

The first thing you must do is to prepare the steel chassis by perforating all holes (round and square); finishing them with the appropriate mill files; sanding the inside and bottom of the chassis with automotive-type sandpaper; and, finally, cleaning all surfaces with a damp cloth to remove all particles.



For this project, I used a Hammond steel chassis measuring 12" x 17" x 2". **Figures 1 and 2** give you all the information you need to prepare the chassis, while **Fig. 3**

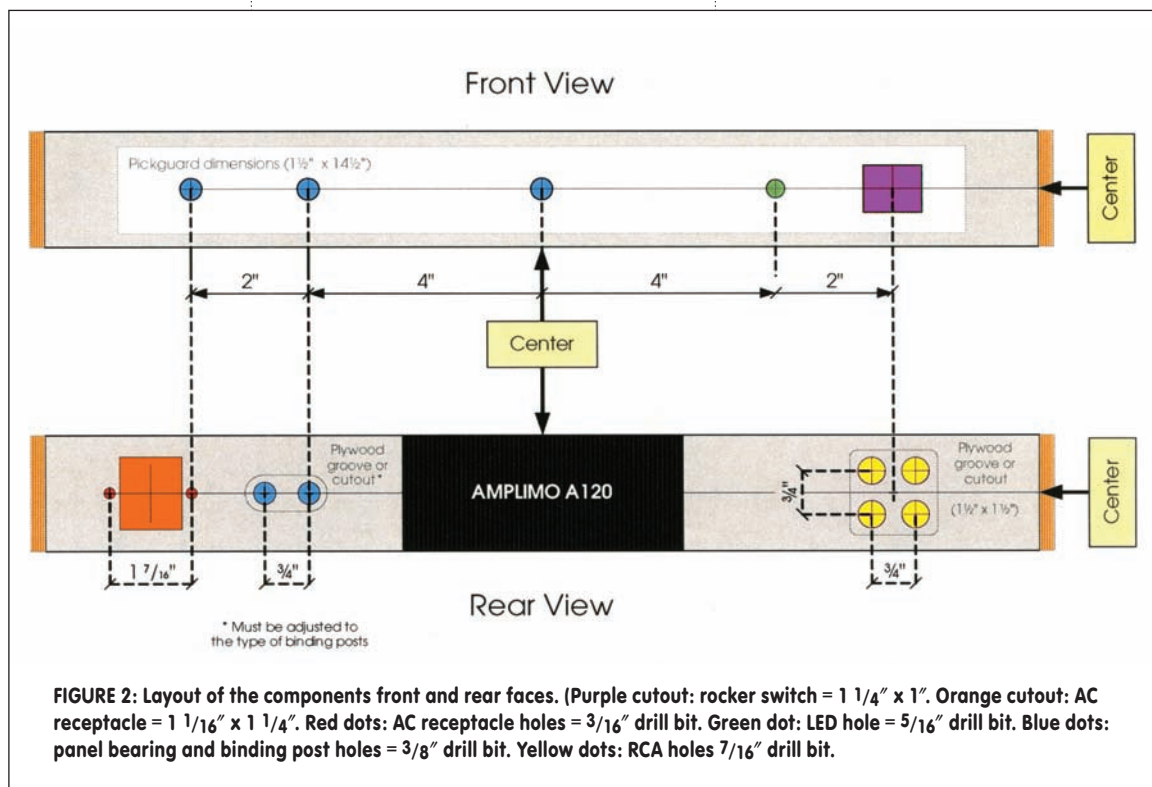
shows the cover plate with its ventilation holes. Among my personal set of best practices, I always draw and print templates and temporarily stick them to the chassis. This helps me achieve accurate results when drilling or punching holes.

The cabinet is made of various materials that combine harmoniously. I covered all sides of the chassis with ¼" thick plywood that I

glued to the chassis with "no nail" glue. Though I used plywood, HDF would be an excellent alternative here, if not better. Note that the rear face is not completely

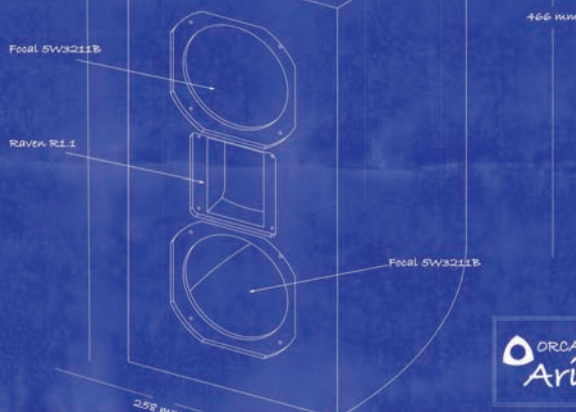
covered. You must leave a section of 4¾" free in the middle to install the power amp module directly to the chassis.

The dimensions of the panels are as



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- Front panel: 2" × 17½" × ¼"
- Side panels: 2" × 12" × ¼"
- Rear panels: 2" × 6 ⅜" × ¼"

These are approximate dimensions that might require some adjustment

- Front panel: $2'' \times 17\frac{1}{2}'' \times \frac{1}{4}''$
- Side panels: $2'' \times 12'' \times \frac{1}{4}''$
- Rear panels: $2'' \times 6\frac{3}{8}'' \times \frac{1}{4}''$

Depending on the RCA connectors and binding posts you use, you will probably need to groove the plywood, or even cut out a section, to ensure proper assembly. **Figure 2** and **Photo 2** indicate

how to proceed. If you look closely at the photo, you'll notice that the IEC connector is next to the amplifier module. This is a design flaw because the mains AC wires should not be so close to the sensitive circuitry. I needed to shield the AC wires from the IEC connector to the primary of the transformer in order to avoid hum induction in the circuitry. I corrected all this in **Fig. 2** by placing the IEC connector in the corner of the amp as it should be.

After cutting your panels to length, apply the veneer, beginning with the outside and then the bottom and top edges. I chose an exotic veneer called

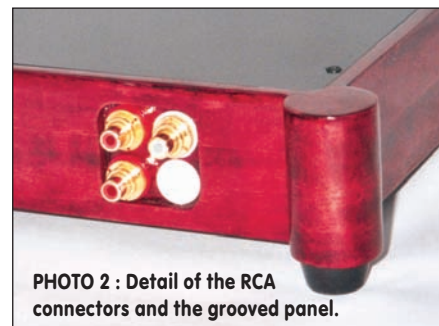


PHOTO 2 : Detail of the RCA connectors and the grooved panel.

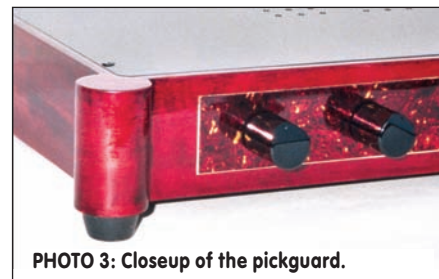


PHOTO 3: Closeup of the pickguard.

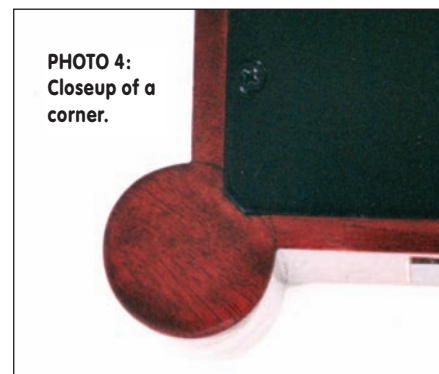


PHOTO 4:
Closeup of a
corner.



PHOTO 5: EAR
isolation foot
installed under
a corner.

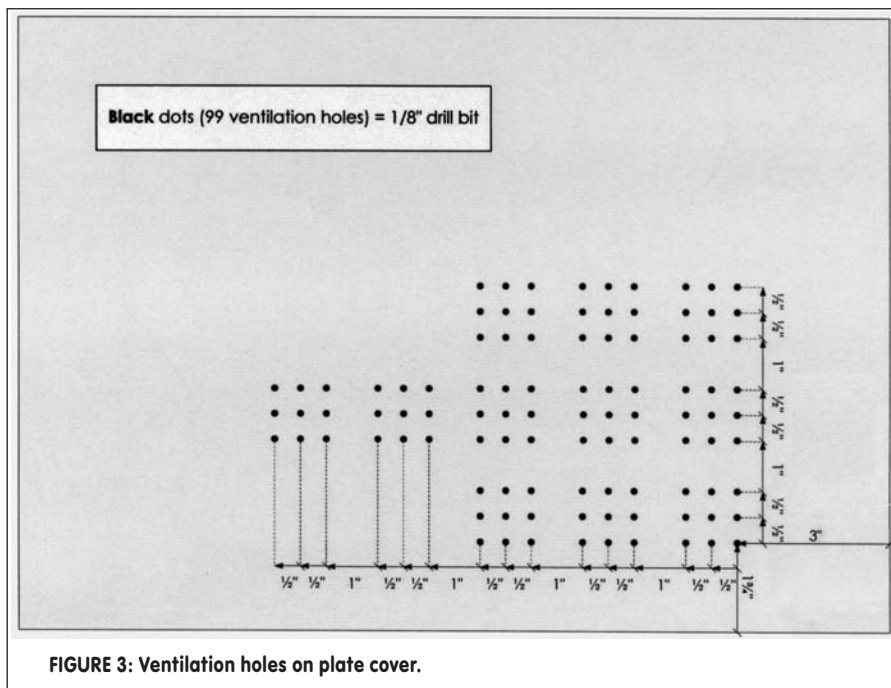


FIGURE 3: Ventilation holes on plate cover.

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■ **Day 1:** Apply two coats of Waterbase Brushing Varnish with a brush. If you intend to varnish your cabinet with a spray gun, use the Waterbase Spraying Lacquer. Follow the manufacturer's instructions, especially for the time required before recoating.

■ **Day 2:** Gently sand the cabinet with 600-grit sandpaper (waterproof type) and some water. Make sure you do not reach the veneer. The objective here is to provide good adherence for the next coat. Clean the cabinet with a damp lint-free cloth. Apply two more coats of water-base varnish.

■ **Day 3:** Same procedure as in Day 2.

■ **Day 4:** Gently sand the cabinet successively with 320-, 400-, and 600-grit sandpaper and some water. Your objective is to progressively level the surfaces of the cabinet (to help me with this task, I made myself a small sanding block with a piece of wood 1" x 2" x 1/2", under which I glued a piece of 1/4" thick liege). You should not attempt to completely level the surfaces now, because you might reach the veneer in some places. Clean the cabinet with a damp lint-free cloth. Apply two more coats of water-base varnish.

■ **Day 5:** Repeat the same procedure as in Day 4.

■ **Day 6:** Repeat the same procedure as in Day 5. When sanding the cabinet, your objective is to get perfectly level surfaces.

■ **Day 7:** Gently sand the cabinet successively with 600-, 800-, and 1200-grit sandpaper and some water. Again, the end result should be perfectly level. After the 1200-grit sanding, the water-base varnish should be slightly shiny. Apply the last coat with the utmost care.

■ **Days 8 to 14:** Put aside the cabinet to let the varnish harden.

■ **Day 15:** This last step will provide the ultimate finish. Using a medium compound, gently buff the entire surface following the recommendations of the manufacturer. For best results, use a foam polishing pad with a drill. You may also repeat the same procedure with a fine compound and another foam polishing pad. This will virtually eliminate any tiny scratches. Using the swirl remover, polish the varnish with a soft cloth. Voilà! Your cabinet now has an outstanding finish! ■

Anigre Figured, which is a beautiful, pale wood offering gorgeous grain patterns that resemble waves. The paleness of the veneer was important to me because I intended to stain it to match the colors of a vintage-style pickguard I wanted to put on the front panel. **Photo 3** shows a closer look at the harmony between the tinted veneer and the pickguard.

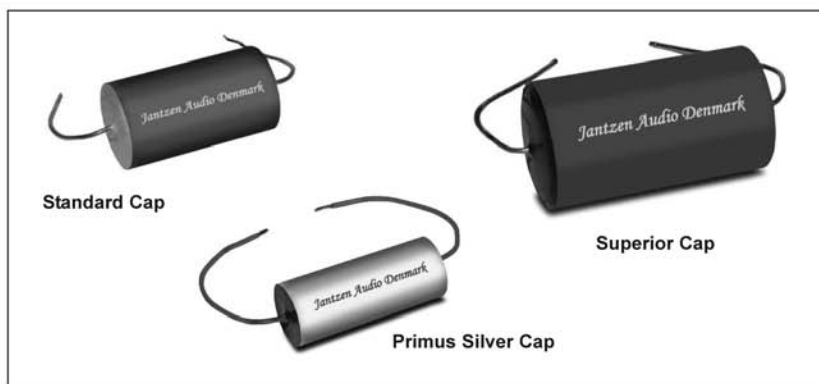
Gluing the panels to the chassis is an easy task but requires some thought beforehand. You must align the cover plate and top edges of the panels to make sure everything will end up flush when you finish the cabinet. I recommend that

you first screw your cover plate in place, then apply some glue to the inside of your panels and place them in their final position. Carefully align the top of the panels with the cover plate, secure them in place with masking tape and make sure everything is square, vertically and horizontally. Having the surrounding panels and the cover plate flush with each other will give a professional touch to your work.

Once the glue has dried, perforate the wood everywhere there is a hole in the chassis. Be careful not to damage the delicate veneer. It is also a good idea to

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TABLE 1: FINISHING MATERIAL.

Component/ Quantity	Description	Supplier	Part no.
1 sheet (1' x 8')	Anigre figured veneer	lhh	
1 sheet	vintage style pickguard	SM	#4417
3 bottles	liquid stains (red mahogany, black, vintage amber)	SM	#5032, 5040, 5030
1 pint	Waterbase Spraying Lacquer	SM	#5566 or #5570
1 can	polishing compound - medium*	SM	#1202
1 bottle	swirl remover	SM	#1204
1	foam polishing pad	SM	#3414
1	spray paint can, black, high gloss	lhh	
1	spray paint can, black, textured	lhh	

lhh: Local home hardware

SM: Stewart-MacDonald (800-848-2273)

protect the veneer with masking tape before you glue the panels. This will shield it from the excess glue that will be squeezed out.

You must now assemble the corners. Mine were made out of a wood pole of 1½" diameter. The first thing to do is to apply veneer around the pole. Then, using a bench saw, cut out one-fourth of the pole (**Fig. 4**). After that, cut four identical sections of 2", the same height as the bare plywood panels.

Veneer the top and bottom of the corners only after you glue them in place. This methodology allows you to hide the joints of the panels. Again, it is a good idea to try to protect the veneer where the glue might be squeezed out. When you glue the corners in place, do not forget to allow for the thickness of the bottom and top veneers that will be positioned later on. Use masking tape to secure the corners and let the glue dry.

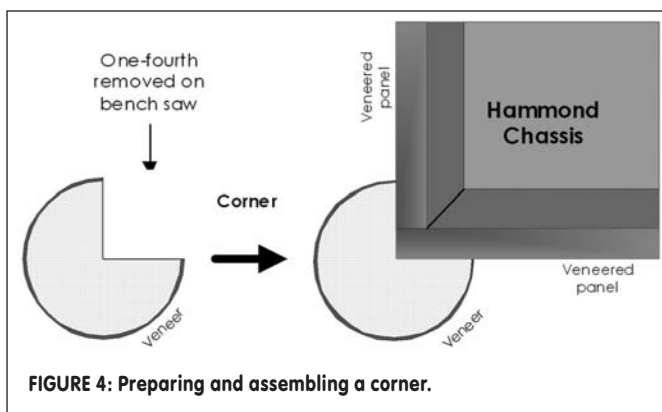
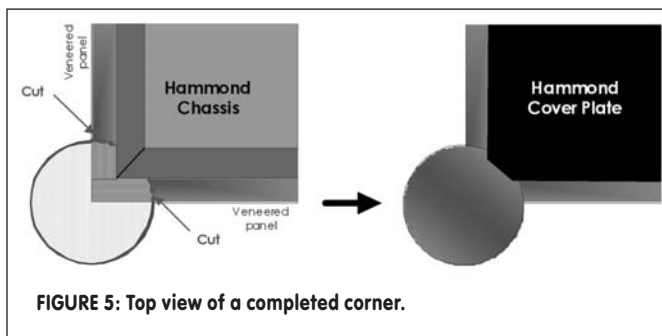
When the cabinet is ready, apply veneer to the bottom and top sections of the corners. Using a remaining section of the veneered pole, draw eight circles on a piece of veneer and cut them out. Before you glue them in place, you will need to remove a small section of the veneer that covers the top and bottom edges of the panels to allow the round piece to fit perfectly. **Figure 5** and **Photo 4** show the final result.

The last step is to sand the veneered cabinet very gently starting with 320-grit sandpaper and finishing with 600 grit. When completed, clean the veneer with a lint-free cloth and some turpentine. You are ready for finishing.

FINISHING THE CABINET

I finished my cabinet by applying over

ten coats of shellac. My father, who makes stringed instruments as a hobby, thought my technique similar to that used by luthiers when they


FIGURE 4: Preparing and assembling a corner.

FIGURE 5: Top view of a completed corner.
TABLE 2: SUBWOOFER AMP PARTS LIST.

Component/ Quantity	Description	Manufacturer	Supplier	Part Number
4	completed PCBs (see part 2)			
C14-15, C18-19	4700µF/80V capacitors	Elna Cerafine	PC	NOS*
C16, 20	100µF/100V capacitors	Elna Cerafine	WL	ROA104
C17, 21	0.1µF/100V capacitors	Wima MKP-10	WL	WM209
S1	4P3T rotary switch	C&K	DK	CKC7007-ND
S3	DPDT toggle switch	C&K	DK	CKN1035-ND
S4	10A/125V rocker switch	C&K	DK	CKC2014-ND
P3	10kΩ stereo potentiometer	Noble	MP**	
D5	LED with hardware	Lumex	DK	67-1163-ND
Z1	1.6A time-delay fuse (GDC)	Busemann	DK	283-2275-ND
1	power transformer	Amplimo or Plitron		58019/057019201
1	120W amp module (A120)	Amplimo	A or P	A120
1	loudspeaker relay	Amplimo	A or P	LRZ
3	RCA connectors	WBT	PC	WBT-53445
1	pair of binding posts	WBT	PC	WBT-55244
1	AC receptacle with fuseholder	Qualtek	DK	Q201-ND
1	power cord	Qualtek	DK	Q115-ND
10'	hookup wire - 19 AWG (red/black)	Kimber	PC	KIMBER-51905/51906
10'	hookup wire - 16AWG (red/black)	Welborne Labs		TR16/TBLK16
2'	hookup wire - 23.5 AWG	Cardas	PC	CARDAS-51900
1	steel chassis 17" x 12" x 2"	Hammond	ES	1441-34 or 1441-34BL3
1	steel cover plate 17" x 12"	Hammond	ES	1431-34 or 1431-34BL3
3	knobs ¾" gloss black, beveled face	AES	KNO	SL94 SML
3	shaft extenders (¼" diameter)		local supplier	
4	E-A-R isolation feet (1")	E-A-R	MP**	
3	panel bearings for ¼" shaft	Keystone	ES	119
11	½" 6-32 standoffs	Keystone	DK	1952K-ND
2	¾" 6-32 hinged standoffs	Keystone	DK	353K-ND
13	¼" 6-32 slotted screws	building fasteners	DK	H154-ND
9	6-32 hex nuts	building fasteners	DK	H220-ND

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www.welbornelabs.com

www.e-sonic.com

www.audioelectronicsupply.com

* May be replaced with 4700µF/63V capacitor. I recommend the excellent Nichicon KG Gold Tune. See Percy Audio.

** Michael Percy does not use part numbers. Consult his catalog for more details.

make violins. Though the end result is superb, I believe you should use water-based varnish, which I use in all my projects. The finish is more resistant and equally beautiful, but will not make the amp sound any different.

Following this project, I developed my own technique that I now use when I finish a cabinet. This procedure, summarized in the sidebar, provides exceptional and consistent results. I am confident you will reach the same results if you follow this protocol. **Table 1** is a description of the recommended finishing materials.

Prior to the application of varnish, you must decide whether or not to tint the veneer. This is a matter of preference. As mentioned earlier, I stained my cabinet to match the nuances of a vintage-style pickguard I put on the front face of the amplifier. Experimentation led to the development of my personal recipe composed of three different colors of liquid stains: red mahogany, black, and vintage amber. I diluted these in methyl hydrate and applied them to the veneer with a lint-free cloth. Following this step, I applied the shellac using a technique similar to that described in my protocol.

To complete the aesthetic part, I spray-painted the cover plate, the inside, and the bottom of the chassis. After I perforated the 99 holes of the cover plate as suggested in **Fig. 3**, I removed the original paint from the Hammond cover plate using a paint remover solution. I then sanded it with 600-grit sandpaper (automotive type), cleaned

it with a solvent and a lint-free cloth, and spray-painted it with five thin coats of high-gloss black paint. I spray-painted the inside and the bottom of the chassis with a textured black paint. Three coats were sufficient to produce a good finish. Of course, you must completely cover the varnished veneer with masking tape prior to this task.

As soon as the paint is dry, install the four rubber feet. I prefer the small EAR isolation feet, which are efficient, inexpensive, and fit nicely under the corners. **Photo 5** shows one corner completed.

The very last finishing touch concerns the pickguard. As shown in **Fig. 2** and **Photo 1**, I applied a piece of $1\frac{1}{2} \times 14\frac{1}{2}$ " of vintage-style pickguard to the front face of the amplifier. To enhance its patterns, I attached the pickguard to gold paper using a thin adhesive sheet made by 3M. I let the gold paper exceed the pickguard by about $\frac{1}{16}$ " all around it to further enhance the effect. To avoid the production of bubbles, stick the gold paper to the front panel by "rolling it" from one end to the other. Careful alignment will guarantee satisfying results.

Note that you can install the three panel bearings under or over the pickguard. In my case, I chose to install them under the pickguard.

ASSEMBLING THE SUBWOOFER AMP

The last part of the project includes no unique procedure, so there should not be any problem for the DIYer having a minimum



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BK-16 Kit

Madisound is pleased to offer the **BK-16** folded horn kit.



**F
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We have chosen the **Fostex FF165K** 6.5" full range for use in the BK-16 cabinet. The **FF165K** has a **Kenaf fiber cone**, inverted foam surround and aluminum dust cap.

The **FF165K** is run full range with a frequency response out to 15kHz. The **T90A** super tweeter has been added to cover the upper frequencies. The **T90A** is a top-mount horn tweeter with an **Alnico magnet**. The tweeter is rolled off on the low end with a **Fostex Tin & Copper foil capacitor**. The system



frequency response is 55Hz to 35kHz at 95dB.

The **BK-16** cabinet is made from **Baltic Birch** plywood and is sold flat, unassembled, unsanded and unfinished. Cabinet dimensions are 9.75" W x 14.75" D x 29" T.

The kit includes:

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- Pair of FF165K - full range
- Pair of T90A - horn tweeter
- Pair of DB-Cup - Input cup
- Pair of Crossovers
- Nordost 2-Flat wire for tweeter
- Instructions

Kit Price: \$635.00 /pair

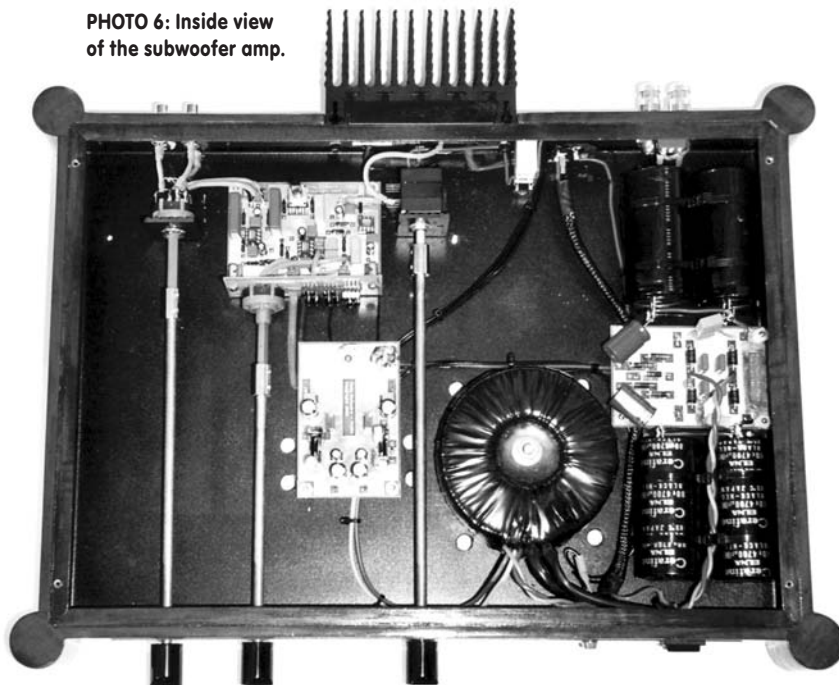
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PHOTO 6: Inside view of the subwoofer amp.



of experience. All you must do is assemble all the components as shown in **Figs. 1** and **2** and **Table 2**. The final result should look like **Photo 6**. Also refer to the figures of Part 1 to realize the proper connections between the components.

In addition, you need to pay particular attention to the installation of the following components:

- the source selector S_1 ;
- the volume control P_3 ;
- the phase inverter switch S_3 ;
- the loudspeaker relay;
- and, finally, the chassis wires.

The source selector S_1 and the volume control P_3 each require a separate mounting bracket. This kind of hardware is sometimes difficult to find and quite expensive. You may wish to make your own as I did. Using a small Hammond plastic box, I simply cut out the part I needed. As shown in **Fig. 6**, you can extract many

mounting brackets from a single box, all customized to meet my specific requirements. I stuck my mounting brackets to the chassis with double-sided adhesive, which is strong enough for this task, but you could also screw the mounting brackets to the chassis.

Photo 7 shows the phase inverter switch S_3 of my subwoofer amp. I placed this switch close to the binding posts for maximum convenience and short wiring. I used a three-pole double-throw switch, which I had on hand and found it easier to cross-connect the pins together. A double-pole double-throw switch is all you need, however. In this photo,

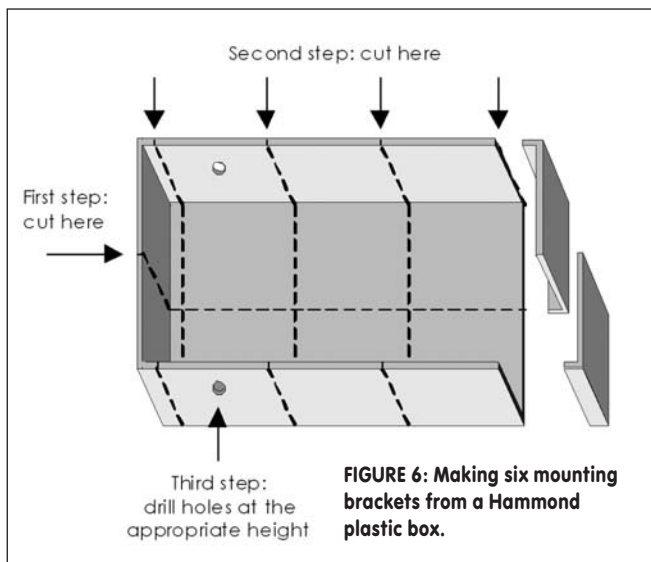


FIGURE 6: Making six mounting brackets from a Hammond plastic box.

PHOTO 7: Closeup of the phase inverter switch S3.



TABLE 3: WIRE GAUGES AND THEIR USES.

AWG Application

16	From the LSP pin of the Amplimo A120 to S3 From S3 to the ground point of PCB #4 From S3 to the binding posts From C14-17 to V1+/V1- of PCB #3 From V1+/V1- of PCB #3 to the Amplimo A120 From C14-17 to the ground point of PCB #4 From the ground planes of PCB #1 and #2 to the ground point of PCB #4 To extend the mains AC leads of the power transformer <i>Note: The mains AC leads have been twisted and shielded to prevent humming.</i>
5.5	From the RCA connectors to S1 and from S1 to PCB #1 Between PCB #1 and PCB #2 From PCB #1 to P3 and from P3 to the IN pin of the Amplimo A120 <i>Note: Shielded interconnect cable, see text.</i>
19	All other wires

you can also see the loudspeaker relay positioned next to the Amplimo A120. The relay is delivered with double-sided adhesive that facilitates its installation within the chassis.

Finally, **Table 3** describes the wire gauges I used to interconnect the components. As I mentioned in Part 2, the use of shielded wires in the signal path had a noticeable contribution by further reducing the level of hum and noise from 1.6mV down to 1.0mV at full level. To perform this task,

I used a Cardas interconnect shielded wire, which offers a pair of conductors that were very convenient to interconnect the RCA connectors to S_1 and S_1 to PCB #1. For the other signal wire connections, I needed only one conductor. It is important to mention that the drain wires must be soldered to the ground plane of PCB #1 to make the shield effective.

I believe there is no need to give you more instructions than that. Take your time and enjoy!

CONCLUSION

This third part has provided detailed information on the making of the cabinet and its assembly. In the fourth and final part, I will discuss the performance of the subwoofer amp. As a "bonus feature," I will give you some hints on how to make two preamplifiers using the same PCBs. *aX*

NOTE: At the end of Part 1, my website address was incorrect. The correct address is www3.sympatico.ca/r.godmaire.



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Amplifier War and Peace

If you have noticed that “passive” components aren’t passive and find the idea of a simple power amp in which the signal goes through a very small number of components attractive, this “hybrid” amp may be for you.

By Simon Brown

Tchaikovsky rises to a crescendo. The cannons boom through the loudspeakers. But there’s something wrong here; it’s a piano concerto, not the *1812 Overture*! The “cannons” continue, out of time with the music, and now more like the continuous crackle of small-arms fire. A well-placed thump on the front left corner of the amplifier and Peace breaks out.

A ghost in the machine? My Fonz of *Happy Days* impression? More like a dubious solder joint or a fatally wounded capacitor. This valve/MOSFET hybrid amplifier is now a fickle beast. Twenty years of hardwired mods on modified PCBs has left it an irascible veteran. This makes me reluctant to do even minor exploratory surgery for fear of causing more complications. It’s time to rebuild the amp.

Twenty years ago I designed my original Valvet amplifier around Hitachi MOSFET devices. Modern power FETs have significantly better properties enabling the use of one complementary pair per channel where previously two had been needed. This has the dual advantage of providing a reduced capacitive load for the valve driver stage and avoiding the need to match parallel pairs of devices.

SHORT AND SWEET

A hybrid amp lets each device do what it’s good at. Valves can provide good voltage gain but are limited when it comes to current. Power MOSFETs are great at driving real-world loads, such as speakers.

So here is a 50W power amp with enough gain to be fully driven by a CD player without extra amplification. Yet you can view the circuit as one valve triode, one capacitor, and a MOSFET fol-

lower. It also manages acceptable distortion performance by maximizing the linearity of the circuits rather than relying on a bandage application of negative feedback. But I should point out that if you want an amplifier whose distortion has more noughts than a bagel factory, I suggest you shake your head sagely and move on to look for more of a “textbook” amp.

COMBINED FORCES

This amp (**Photo 1**) is a “hybrid” in many ways. There is the obvious way in which it uses both valves and MOSFETs. But it’s also a hybrid in its use of enriched Class A/B, which means it is running Class A up to a few watts. This is deliberate. The bottom bit of the transfer curve is the wart on the face of MOSFET specs that Bipolar Bullies like to point out (“MOSFETs have higher distortion than bipolar transistors”) before poking out their tongues. Running higher bias current moves you into a happier playground.

If you choose suitable heatsinks and voltage rails, you can run this design in “pure” Class A (or cooler if your budget points you that way). I have chosen the ~50W/8W Class A operating characteristics to suit my ~90dB/W speakers, and the heatsinks are simply the ones I have in stock that match my “mid”-size ezChassis chassis. **Table 1** shows alternative power and bias possibilities for a number of common transformer secondary windings and three heatsink sizes. Choose the bottom right option for a “pure” Class A amp.

THE CHARGE OF THE LED BRIGADE

The valve “driver” stage of this amp has evolved over the years. Originally it was a triode plus cathode follower before



PHOTO 1: The completed unit.

TABLE 1: VALVET POWER / BIAS OPTIONS.

Heatsink Rating °C/W (for 30°C rise)	Heatsink Power /per heatsink (for 30°C rise)	Possible Bias Levels for DC Voltage rails /Set Voltage(TP2)		
		±40V (~30VAC)	±35V (~25VAC)	±23V (~18VAC)
0.65	46W	0.6A /0.26V	0.7A / 0.31V	1.2A /0.53V
0.42	70W	0.9A /0.4V	1.0A* /0.44V	1.5A /0.66V
0.33	90W	1.1A /0.48V	1.3A /0.57V	1.9A /0.84V
*As built	Aprox 8Ω power>	75W	50W	25W (~ClassA)

stepping up to a SRPP (Shunt Regulator Push-Pull¹) circuit. Then there was a significant improvement to a μ -follower^{2,3} despite my misgivings about the additional components. I had been considering adding a current source when I came across the β -follower⁴ (nice to see it reinvented in *aX* recently⁵). This proved even more effective. What you see here is effectively a hybrid of both of these two designs using LEDs to define bias voltages.

The other major change I've made since my older amp is

to the tube type. Power MOSFETs are relatively easy to drive, relative to big bipolar transistors, that is. They still make

serious demands on the driving device, especially with their high levels of capacitance. 6DJ8/E88CCs, which were

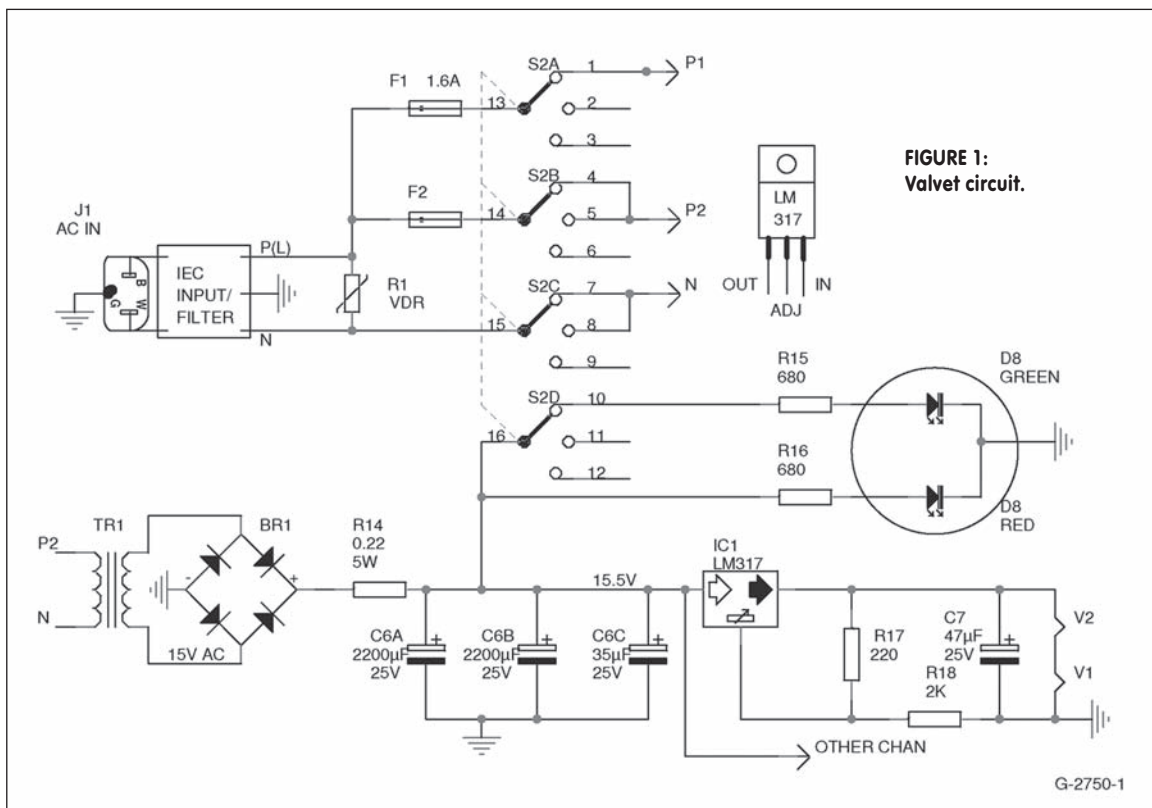


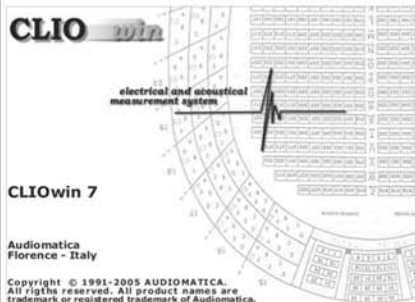
FIGURE 1:
Valve circuit.

G-2750-1

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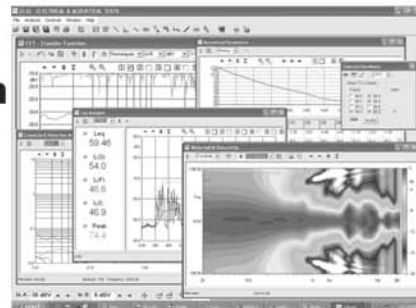


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my favorite preamp tubes, are powerful (transconductance-wise) with reasonable gain, and can be low noise devices. So they seem good choices for a driver in a hybrid amplifier, as witnessed by their

use in a number of commercial amps. These are what I've used until now.

Recently I've started to question how these tubes behave when called upon to swing larger voltages. In looking at com-

VALVET PARTS LIST

	Value	Notes	Supplier	Part no
RESISTORS				
R1	2M2			
R2	330			
R3, R8	100k			
R4	100			
R5, 7, 24, 25	12k			
R6	82k			
R9, 10, 28-31	150			
R11-12, 14, 22	0.22 (5W)			
R23, 26, 27				
R13	10			
R15, 16	680			
R17	220			
R18	2k			
R19 (aka R22)	1M			
R20-21	560			
P1	100k log	Alps blue	RS	219-9146
P2	10k	10 turn		
P3	20k	10 turn		
CAPACITORS				
C1	4.7µF	16V min		
C2, 3	4.7µF	200V min		
C4	4.7µF	16V min		
C5	0.1µF	35V min		
C6	4400µF min	25V min		
C7	0.47µF	15V min		
C8, 9	100µF	350V min		
C10, 11	16µF min	350V min		
C12-15	4700µF	50V min		
	(1000µF Panasonic NHG)	RS		
C16-19	3300µF min	50V min	RS	339-6859
DIODES				
D1-3	red LED			
D4, 6	1N4148			
D5, 7	12V zener			
BR1	P04 6A			
BR2	UF4007 (x4) 1000V			
BR3, 4	35A/400V bridge			
BR5, 6	MBR1100TR Schottky		RS	469-0691
D8	red/green LED		RS	578-294
TRANSISTORS AND TUBES				
V1, 2	6SN7GT		(DBL)	433610
T1	BC558			
T2	IRFP140	T0-247	RS	254-0948
T3	IRFP9140	T0-247	RS	189-0949
Reg1	LM317	T0-220		
INRUSH SUPPRESS.				
	25R-2A (for230V)		RS	210-689
MISCELLANEOUS				
Sw1 (DBL)	rotary 433818	2P6W		
IEC1	6A/230V	mains filter/inlet	(DBL)	433817
VDR	275V (for 230V supply)			
F1	1.6A	(for 230V)		
F2	0.25A	(for230V)		
Sw2	rotary	4P3W	(DBL)	433819
L1	15H (380Ω)		Hammond	
F3, F4	3A			
HARDWARE				
heatsink	0.42C/W (30C rise)		DBL	433804
chassis	ezChassis 120mm		DBL	432006
valve mounting panel			DBL	432301
RCA sockets/speaker terminals/valve bases			(DBL)	
shaft extension kit 1/4"			DBL	433215
brass knobs	50mm (2") and 30mm		DBL	433008/9
handles	96mm black		DBL	433023

Supplier Notes:

I've included some RS part numbers for convenience (but it isn't the cheapest place to buy) www.rs-components.com or www.alliedelec.com in North America.

DBL is www.designbuildlisten.com.

(DBL) indicates this is available via DBL but is a generic component than can be bought anywhere (but listed here for convenience).

mon alternatives (I avoided using the sort of tube in which the entire world stock is in a locked cupboard of a locked room in a former KGB building), I decided to use the 6SN7GT because it is a linear valve with middling transconductance. Here I chose to parallel up both halves to improve its driving ability (and noise).

This proved to be not strictly necessary: the completed amp is 3dB down at around 500kHz (and 3Hz) and is very quiet. Gain is $\times 20$, so can usefully be driven to clipping (56V p-p, equivalent to 50W into 8 Ω) with the output of most CD players. Distortion is near the limits of my admittedly crude test gear (USB instruments DS1M12 USB oscilloscope/signal generator), but around 0.1% THD.

CAMPAIGN PLAN

I configured this as an integrated amp, but you can omit the inputs/switch and replace the potentiometer with a resistor. The pot is quite a high value (100k) to help my tube phono stage; reduce this to 47k if you have a more convention-

al preamp. R1 and so on ensures quiet when switching inputs. I like to use a six-way switch (for my <4 inputs), giving me "free" mutes at either end of the switch travel (or in the middle to reduce crosstalk).

I included an input/output connection after the potentiometer to run a subwoofer and to make it easy to bypass the pot in other systems. R2 is a grid-blocking resistor, which helps avoid any RF oscillation. This is probably not necessary, but is a pragmatic choice to leave it there. Solder it as close as physically possible to the grid. R9-10 play a similar role for the MOSFETs but are essential and again must be attached directly to the gate.

The actual operation of the SRPP/ μ -follower/ β -follower is open to some debate, so what follows is my interpretation. You can view V1 as "simply" a triode in which V2 and the current source T1 and so on combine to act as a very high impedance load, maximizing gain and linearity. The LEDs act as a low-noise biasing arrangement. V2, which also acts as a "cathode follower" to give

some grunt to drive the MOSFETs, does this job superbly as this operation is enhanced by the high impedance action of the current source.

The signal is then capacitor-coupled by C2-3 to the individual MOSFETs. This is because the MOSFETs must be biased with a DC voltage (around 3.7V) to get them to operate correctly. R5-8/P2-3 form this bias network. P3 adjusts the current flowing through the MOSFETs and thus the level of "Class A" operation. P2 adjusts the balance between the devices, ensuring you can adjust for negligible DC current through the speakers.

The FETs are operated in "source follower" mode, so theoretically "track" the input with no distortion (and no voltage gain). I chose the IRFP 140/9140 pair because they are well matched for capacitance. IRP540/9540 have similar specifications but in a smaller/cheaper package (TO-220). To me the larger (TO-247) package size is potentially important because it reduces the potential for transient thermal effects. However, I did use the smaller devices to get



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PA060S Improved replacement power transformer for ST70

PA135 replacement power transformer for Dynaco (tm) Mark 4.

P782S replacement power transformer for Mark 3 and Mark 2

C354 Exact replacement for chokes in ST70, Mk2, Mk3 and Mk4

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the amp running OK.

My minimalist knife is evident at the output, where there is no inductor. Such is the advantage of a simple amplifier with no global NFB. I have left the “conventional” output filter C5/R13, which rolls off the output gently at high frequencies. I have no real evidence to say that it is necessary, similar to the grid resistor.

SUPPORT PERSONNEL

The diode/zener tree D4-7 protects against the MOSFETs being accidentally turned on hard. C4 performs a dual function, ensuring that both FET gates see the same voltage. It also acts in combination with the bias resistors to gently ramp up the voltage to the gates to gradually turn on the FETs (taking about 5s after the power is applied).

The rail fuses, which are open to debate, are essential when you first build the amp and may provide some long-term protection. But fuses have a nasty habit of blowing just because it's Tuesday. If only one blows, it can lead to a

nasty DC voltage at the output, so it may be wise to remove them once the amp is bedded in.

The dual LED is entirely optional and displays red when only the heaters are on and amber (both red and green halves on) when fully on.

POWER BLOCS

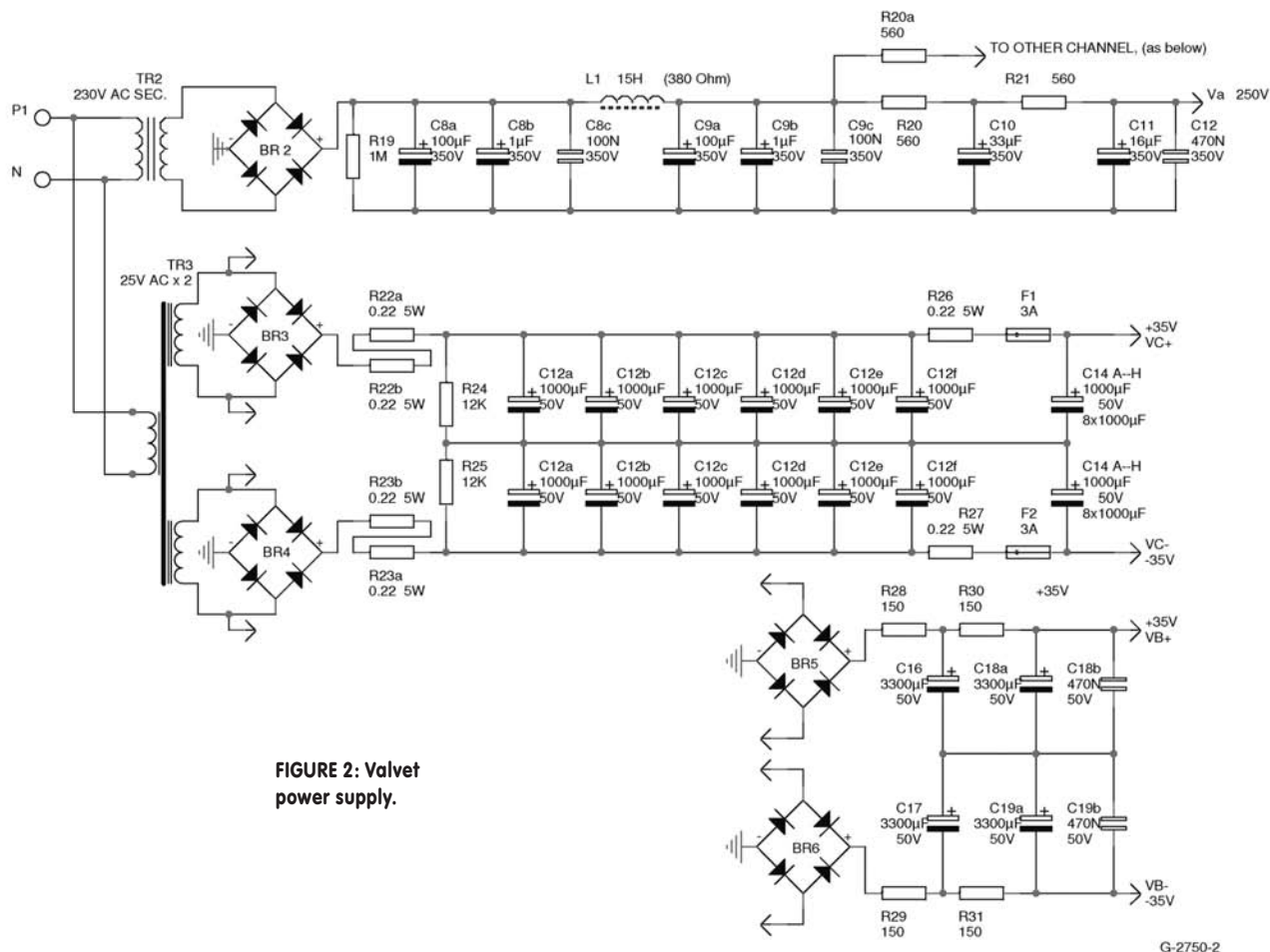
While the active part of this amp is simple, the need for three or four separate power supplies adds significantly to the complexity and cost (\$800 ballpark). This isn't entirely a bad thing as I'm a strong believer in having separate supplies for power amp driver and output stages—here they occur by default.

Input: IEC 1 is a combined IEC power inlet socket and filter unit rated at 6A (230V); it is snubbed by a voltage dependent resistor to further take out voltage transients. While both of these are optional, I recommend them or at least some form of input filtering. SW2 has a “standby” position that enables the heaters to be warmed prior to turning on the HT. Here I've used a separate heater

transformer, but if you have a combined HT/heater transformer, you may choose to switch the HT by relay, ideally with a delay mechanism.

Heater Supply: LM317 regulators provide a cheap, clean DC supply to the heaters, but because they are dissipating a couple of watts, they must be heatsinked. One of the nice aspects of 6SN7s (over 6DJ8/E88CCs) is that they can stand 200V from the heater to cathode, so there is no need to jack up the supply (or worse, limit the HT rail). However, if you are using “dual” 6SN7s as I have, with heaters in series, I suggest using the grounded heater for the lower valve. You may need to adjust the exact value of R14, depending on the “sag” of your heater transformer secondary.

HT Supply: I've shown a pretty conventional supply here based on a 230V transformer secondary to give ~300V. But if you are starting from scratch, you may consider running a rail closer to 400V (possible with the 200V heater/cathode voltage limit). The actual capacitor values are pretty arbitrary, but I don't



recommend using much less than I have here. If you want to avoid the inductor, you should consider more capacitance or a regulated HT supply.

MOSFET Bias Supply: Because I wanted to keep the biasing as stable as possible, I incorporated a separate power supply for the MOSFET biasing. This avoids the potential of being disturbed by the output. I haven't actually used a separate transformer (if you do, choose one to give slightly higher voltage rails than the main MOSFET supply), but the separate bridge rectifier alone gives good isolation. If all this seems unnecessary or expensive to you, you can attach the biasing resistors to the "main" voltage rails.

MOSFET Power Supply: Having been brought up in the 1980s low impedance school of amplifier design, I've now undergone somewhat of a sea change and can be caught sneaking resistors into power supplies. A small amount of resistance dramatically improves the effectiveness of the

filtering. The resistor immediately after a rectifier further helps by increasing the conduction angle of the diodes even if you "lose" voltage. But you're free to experiment without the resistors.

I chose to use banks of paralleled capacitors rather than single large value caps to give low impedance. I'd already (less scientifically) chosen these Panasonic NHG 1000 μ F (50V) capacitors before Paul Stamler's helpful article⁶ in *aX* which also uses them in parallel. I considered 10,000 μ F a minimum value,

with money and space being the upper limit. I ended up using two 160VA transformers, which is about the minimum size I'd suggest for the bias level I used. These gave ~80W into 4 Ω before clipping. To avoid blowing fuses on power-up, I used a NTC thermistor in the feed to the MOSFET power-supply transformers.

Rectifier Choice: To avoid high-frequency noise from the rectifiers, I prefer to use Schottky diodes, but failing that, fast/soft recovery ones. If this is difficult or you remain skeptical,

use snubber capacitors across regular diodes. I originally planned to use a tube rectifier for the HT supply, but this limits the possible capacitance at its output (although it does provide a desirable slow HT start-up).

AN EXPLODING CHASSIS

I must declare my involvement in the ezChassis used here to house the Velvet (**Photo 2**). This is not surprising, since I designed it

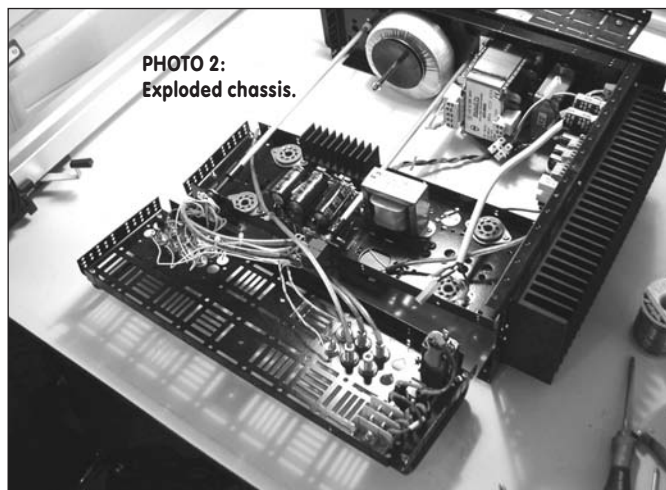


PHOTO 2:
Exploded chassis.

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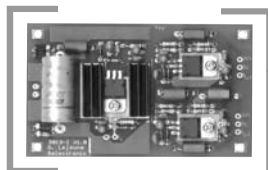
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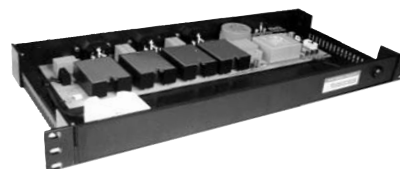
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after frustration with previous amplifier projects. Of course, if you've got the time/skills, you can build it into any old box that suits you and skip this section.

Here I have used the 120mm (~5") high ezChassis and matching 100mm × 300mm (4" × 12") heatsinks. Part of my thinking behind ezChassis was to make it possible to remove individual panels for servicing. Many audio enthusiasts like me forever want to try out some improvement or other. This feature helped in construction because I was able to build and test it in modules before final assembly.

I haven't provided a "step by step" construction guide here; otherwise, this article would be considerably longer. In this form this amplifier is not suitable for a novice builder (but I may yet offer a PCB-based kit in the future). I will offer a few general comments, which won't be news to old campaigners:

- Build and test the tube driver separately. I built mine as a hard-wired unit including the coupling capacitors on one "Tube Mounting panel" (**Photo**

3). I can't pretend that I didn't need to drill any holes, but there were very few because this panel is provided with mounting holes for octal and B9A tube bases and a raft of slots and holes which proved useful. In hindsight, I should have mounted this panel vertically (not horizontally) to support the back end of the shaft extension.

- To keep the signal path short I used shaft extensions to mount the input switching and volume pot at the back of the chassis, enabling the back panel to be built up as another module. The front end is supported by the structural panel behind the cosmetic front panel (**Photo 4**). I found it easiest to install all the back panel components, including handles

(because the amp weighs ~35 lb) before attaching it to the other panels. Knock-outs are provided for the IEC mains entry and labels (supplied with ezChassis) are used to blank off unused holes (**Photo 5**).

- Because the panels have baked on paint, I locally scraped paint off (under screw heads) to ensure they were earthed.

- I built the power modules on PCB "spot" board (I don't know the "correct"

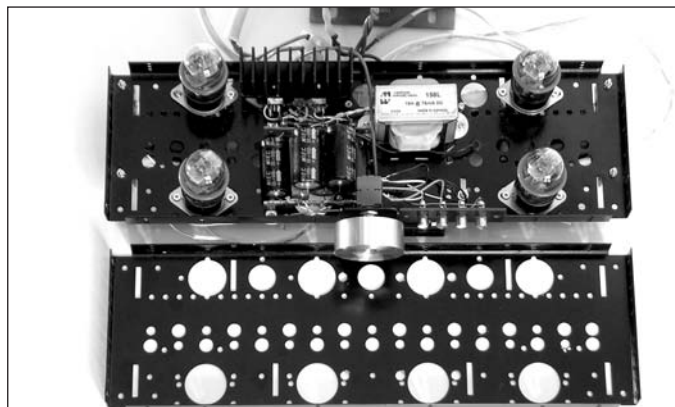


PHOTO 3: Tube mounting panel from above (and blank panel).

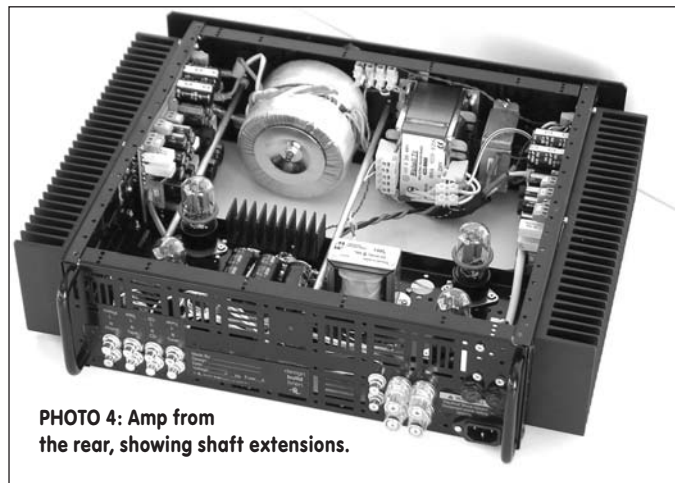


PHOTO 4: Amp from the rear, showing shaft extensions.

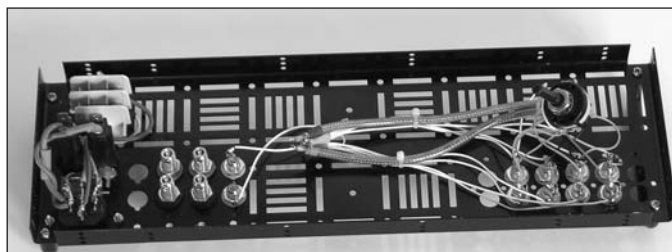


PHOTO 5: Back panel before installation (and labels).

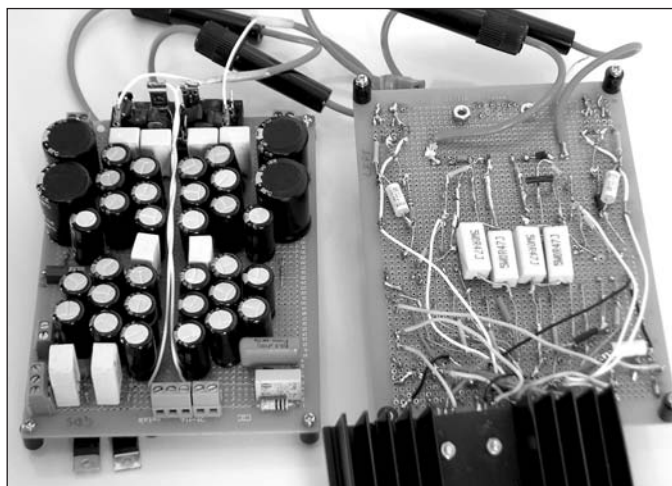


PHOTO 6: Power module on "spot board."

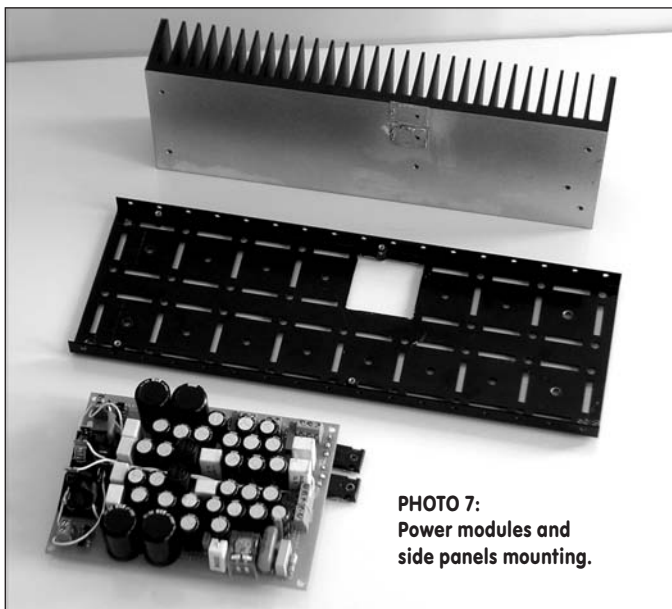


PHOTO 7: Power modules and side panels mounting.

E 3:
t power
y (2).

The circuit diagram shows a Class B push-pull amplifier. The input stage consists of a switch S1 (1P4T) connected to a 2M2 resistor (R1) and a 100K resistor (P1). The output of P1 is connected to the base of a 6SN7 vacuum tube (V1A). The other 6SN7 tube (V1B) is connected in a push-pull configuration with V1A. The load is connected to the outputs of V1A and V1B. A current source is connected to the bases of V1A and V1B, consisting of a 100K resistor (R3), a 4μF capacitor (C1), and a 100 resistor (R4) in parallel, with a PNP transistor (T1) in series. The current source is powered by a +4V supply. The output stage consists of an IRF9140 MOSFET (T2) and an IRF9140 MOSFET (T3) in a push-pull configuration. The load is connected to the outputs of T2 and T3. The circuit is powered by a +35V supply (Vb+) and a -35V supply (Vb-). The feedback network consists of a 10K resistor (P2) and a 100 resistor (R9) in parallel, with a 100K resistor (R8) in series. The feedback signal is connected to the bases of V1A and V1B. The circuit also includes a 1N4148 diode (D4) and a 12V Zener diode (D5) for protection. The output is connected to a 10 resistor (R13) and a 100N capacitor (C5) in parallel, which is connected to the output terminal (OUT).

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FIGURE 4: Distortion components of a 1kHz signal at 50W into 8 Ω (54V pp).



PHOTO 8:
Complete amp
without cover.

name), which makes it easier to deal with wiring up all those capacitors. This board is a recent discovery for me and I'll enthuse about it to anyone.

I assume I'm not alone in spending hours hunting unintentional joins in "stripboard" and even more time replacing the components I blew up when I missed one. This board enables you to use the small solder pads around each hole to locate the components while enabling hard wiring. Its holes are at a standard pitch

that enables you to place many electronic components without drilling extra holes (**Photo 6**). I then mounted these boards on standoffs on the side panel with a section "knocked out" to access the heat-sinks (**Photo 7**).

• I (re)learned a lesson when building this amp. Don't recycle components. It's not worth it. I spent hours trying to figure out a problem with the HT supply. It was due to some dodgy capacitors.

TEST FIRING

As I mentioned, I built and tested the tube stage separately (heater first then HT). Similarly I strongly suggest a step-wise approach to each channel of the power section. I tend to use a protective RCD until I've confirmed that all is well (keeping my finger poised on the off button while looking for trouble).

• Before placing the FETs, try the other parts of the circuit. The independence of the bias supply enables that to be tested first. Check for the 6-9V adjustment across the gates (TP1) before turning down to the minimum value (around 6V). Next check the main MOSFET power supply before installing the FETs. With any power supply, I usually flick my multimeter over to AC volts after confirming the DC value; this is a simple noise/hum check. You should get a near zero reading (though my Fluke multimeter gives more sensible results here than my "El Cheapo" meter).

• After connecting the MOSFETs (and attaching them to the heatsink), check once again that P3 is adjusted to zero before powering up. You need two multimeters (even cheap ones) at a minimum, because it's pretty handy to be able to watch the gate voltage (TP1) at the same time as the bias voltage drop across R11-12 (TP2). There should be no voltage across these resistors when you start (the FETs are turned off).

Adjusting P3 upwards, when the gate voltage gets to around 7.2V current should start to flow through the FETs. But note that IR's datasheets give this turn-on point a very wide 4-8V tolerance, so do this very gradually. Stop at about 150mV and watch until you are sure it's stabilized. (You can do this without the valve driver stage or speakers connected.) This is about the minimum level of bias. I chose 0.44V (~1A)

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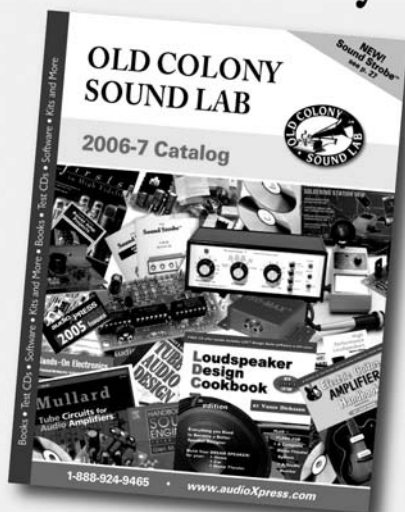
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for my heatsinks. You should watch very carefully over a few hours that it's stable (I checked on each start-up for the next few days).

- Check the DC offset across the output terminals and adjust P2 to get it near zero (I try to keep it below 50mV) before connecting speakers.

TCHAIKOVSKY ENCORE

Describing the sound of your own gear is full of obvious dangers. But suffice to say its sound is somewhat "hybrid" too, having a "valve" midrange but a "transistor" bass end (if a little "softer" due to the lack of global feedback). The Velvet amp (**Photo 8**) proved to be a nice match for a friend's Thor transmission line speakers, driving them comfortably. With my newly built Fostex full range driver speakers (93dB+), I've been caught grinning like a James Bond villain who has just perfected his Death Ray.

Conversely at home I hear musical detail I haven't noticed before. This isn't detail for its own sake but more the big musical picture, seeing how the parts/harmonies fit together in a coherent sense. One of the things that surprised me was how quiet the amp is. Other amps I've used are apparently silent, but here delicate music is audible against a dark background.

The new amp is great; it no longer makes cannon fodder of Tchaikovsky.

Thanks to Alan, Richard, Julian, John, Phil, and Jean for helping me with suggestions and support for the amp and this article.

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CUSTOM WORK



Speech Intelligibility, Part 1

By Ed Simon

No one has ever sung into my ear from 3" away. I have never even put my ear in the bell of a horn being played, or listened to a stringed instrument from a foot away. I have met some good-size singers, but none with a soundstage of 20'. As a result, I really do not know how some of the studio recordings should actually sound.

EVERYBODY'S TALKING

However, every day you and I listen to people talk. We all know what they should sound like. Even more important, there is little room for argument as to whether we can understand what they are saying.

More sound systems are used for conveying speech information than any other use. The wired telephone does not have very good fidelity, but you can understand almost everything that is said. The cellular telephone can, of course, allow you to take notes and pictures, play games, mpeg audio, and video files, and even talk to another phone user. If your iPod had as low fidelity as a cell phone, you would consider it broken.

The primary use of the telephone is to communicate speech. You also use larger sound systems for movies, sporting events, paging, music, public speaking, and uncountable other uses. There are more than 2 billion sound systems in the US!

You are familiar with what happens when a speech sound system does not work. You can usually hear the person talking, but cannot understand what he/she is saying. In much rarer cases the person's voice is not loud enough to overcome the noise, and you just can't hear him/her.

Much effort has been put into understanding the requirements for good speech intelligibility and how to achieve it. The telephone is a great example of providing good speech intel-

ligibility at minimum cost.

RULES OF THE GAME

In developing the telephone system, engineers devised a method called an articulation test to measure the performance. This test, which is done infrequently with public sound systems, is easy-to-do, can give very accurate results, and allows the knowledgeable user to make adjustments. The test is almost never done for one simple reason: It more often than not embarrasses the designer and installer of the sound system.

It is a great way to see whether an auditorium space is good or bad, and can be used to measure performance of a sound system, determining particular problem areas. This test does not require any special equipment—one or two people to speak, and a group of listeners with paper and pencils to record the results. Live speakers are preferred, as the scores increase slightly when done live, a bit more when people can see the talker. Often the test speech is recorded and played back. The scores are slightly lower, but the results can be more consistent.

There is actually a standard test method described in ANSI standard S3.2-1989 R1999. You can buy this document for \$100. Of course, you could use the list of words I have provided with this article for free (**Table 1**). The expensive version has complete instructions on each aspect of the test and proven test word lists that are properly phonetically balanced to represent normal speech. I used harder words to point out problems.

The test is not complicated: One person speaks to a group of test listeners at a rate of three syllables per second, using a carrier sentence such as "Write the word ____ now." A good reader will go through a list of 50 test words in about two minutes. Too fast is just as bad as too slow. The carrier sentence allows

The first part of this series explores the variables affecting a good sound system design.

reverb to build up and surround the test word. If you just read the words, scores will be higher.

SCORING

When you read the word list to people with normal hearing in a small nonreverberant room, scores should be 95% or higher; 100% for any size group almost never happens. The best part is that spelling does not count!

If the test word is "FIVE" and you write down "PHEYEPHJ," you are right! After completing the test, each listener is always convinced he/she answered them all correctly. That is why you never let test-takers score their own tests.

My list contains 300 words. You should test your group in a small room using about 25 words to get them used to the test and weed out the people who hear or write poorly. After all, you must be able to read what they wrote.

You can reuse the list with the same group of listeners as often as you like, but mix up the order. Scores will improve with repeated experiments, but you still can isolate problems. A score of 75% is actually enough to allow understanding of normal speech. A score of 85% or more is great. Our brains put normal words into complete sentences, so we have a built-in form of error correction. If you are up for a bit of fun, try the test with cellphones!

For a thorough test, one reader should be male, one female. Spread the listeners throughout the room. The reader presents 25 or 50 words. The listeners write down where they are and what they hear, then they move to a new position.

After you have combined at least 500 to 1500 sample words from all the listeners, you can score the tests. You are interested in two kinds of mistakes. The first involves the wrong consonant; for example, "HIVE" or "FIRE" instead of "FIVE." When scoring, you must be

careful to allow for bad spelling while making sure the sound is correct.

The second mistake occurs when the vowels are wrong. So when someone writes "FAVE" or "FEVE," this is not a spelling error and must be counted wrong. Some folks assume that only consonant sounds are important in these tests. The theory is that vowels are closer to pure tones, and most masking problems do not affect them. I have found that in large spaces the atmospheric absorption of high frequencies can cause a significant loss of vowels. Yes, air absorbs sound. Available charts show you the loss for a standard distance at different temperature and humidity levels.

NO LAUGHING MATTER

If you test in a mostly empty room in the worst spots, and 15% or less of the words have consonant errors, then the communication method is just acceptable. (Some believe this could be 20% loss, but they are wrong.) An error rate of 10% is desirable when counting only consonant errors. When you include vowel errors, the error rate total should be less than 20% to be just acceptable and about 15% to be good.

Note that most words have two consonants and one vowel. If either or both consonants or the vowel is wrong, that is an error. A single test word can have more than one error in my method of scoring. If you notice frequent multiple errors in single words, you know you have a problem even before you total the test. This is how I score my tests; others may do it differently. Some just count consonant errors; others count only one error per word. Do not assume that similar sounding test names and scores will produce the same results.

The acceptable level of errors will vary according to your intended audience. Churches tend to have older listeners with less hearing ability. Sports venues have few problems if the team wins; more if they lose. Even with some articulation loss most errors are corrected by being placed in the context of a sentence. Even with a word loss of 25%, sentence recognition will still be acceptable, except, of course, in a comedy club. Missing the last word of the punchline 25% of the time might just keep you from laughing. As people's expectations

rise, so should your design goal.

It is also handy to total each individual's results separately and compare them to the others. It is useful to have people change seats with each other if there seems to be a particular problem area. That way you can tell whether the problem is with the person or the room. In a symmetric room you need test in only one half of it. Very few rooms are symmetric when you consider internal and external noise sources.

If you are planning on designing a sound system, it is useful to measure the distance from the unaided reader at which the speech intelligibility is still good (15% consonant and vowel loss combined or less). Use this distance for loudspeaker selection and spacing. Keep in mind that adding loudspeakers decreases this distance, while using directional loudspeakers increases this distance.

PEOPLE ABSORB SOUND

Be aware that testing in a large empty room will give different results than when it is full. People absorb sound. They also make noise, but you assume

the audience is attentive. So the lower numbers are okay under this condition.

If the room has plush padded seats and carpeting, then the lower results probably are not a good choice. The other assumption is that the listeners and readers are native speakers of the same dialect of American English. The use of my table of words is based on American English; other languages will have some differences. Not all languages use the same methods to transmit intelligence.

Do not confuse acceptable with good. If the worst seats are acceptable when the room is empty, that's acceptable. If the best seats are acceptable when the room is full, then you need to change something.

Of course, if you want to do the test by yourself, you can record the list, forget about it for a while, then play it back. Your scores will be higher this way but will still show problem areas. I have provided wave files suitable for making your own CD with this article (posted on the *aX* website at www.audioXpress.com).

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OTHER TESTS

There is another, newer (1958) spoken test called the diagnostic rhyme test, which requires less training for the listeners and is simpler to score because it usually does not test vowel sounds. My lack of experience correlating it with sound system results is why I do not use it, although others certainly do.

If you prefer, there are machine-based tests to determine speech intelligibility. Two Dutch researchers, Houtgast and Steeneken, devised a mechanized method of measuring intelligibility (Steeneken, H.J.M., and Houtgast, T., "The Modulation Transfer Function in Room Acoustics as a Predictor of Speech Intelligibility," *Acustica* 28, 66-73, 1973). In their speech transmission index, they amplitude-modulate a carrier tone and measure the recovered modulation. This is done for many carrier frequencies and modulation frequencies. A weighted sum of the results gives a score ranging from zero to one.

Some technical debate exists, of

course, about the frequency bands of interest, the modulation rates, and the weights that should be applied to give the final score. As a result, you will find labels such as RASTI or STI-PA used for variants of the test. These tests are an extremely valuable method. However, the basis for criticism is that you can fool the test under some situations. (Of course, there is no better machine-based method, and human-based tests have far more variables.)

Instruments that can make these measurements are available today from manufacturers such as NTI, Sencore, and Goldline. These are typically DSP-based units that include STI-PA as one of their functions. Goldline is the exception, making an STI-PA-specific gizmo. You can buy a basic system for a few thousand dollars and quite a nice setup for under \$5000.

Using one of these analyzers is stupidly simple. You play a CD containing a special test signal, setting the sound system to the volume normally used for

speech with the SPL meter built into the STI-PA analyzer. You push the test button, wait less than a minute, and the screen displays "ERROR" or the magic number. My experience is that the number you get is a good one. When you get an "ERROR," the cause is obvious—the system was not loud enough, somebody talked during the measurement period, or a truck went by.

With very little practice you can conduct a useful survey of a room in under an hour. The only purpose-built unit to measure RASTI-PA comes with a label that sums it all up: 0.00-.30 Bad, .35-.45 Poor, .50-.60 Fair, .65-.75 Good, .80-1.00 Excellent!

I once designed and installed a sound system for a church that reputedly was acoustically designed by a noted acoustician named Sabine. The RASTI score was .9 for the seating in the main part of the nave and dropped to .70 in the worst audience seats.

After you have designed and built what you think is a great sound system, it is disheartening to have a little black box come by and tell you it won't work. Bass may be good, treble great, all the openness you could want, but no one can tell the difference between "Tire" and "Fire." Under those conditions having a tire sale could be very hazardous! At some time in the future you can expect that the safety codes for buildings will require such a test of the emergency paging systems. Right now it is an option.

There are also some circumstances under which it is desirable to minimize speech intelligibility. The best known example is the open office plan. The common US scale is the CIS rating. The very best privacy requires a 95% articulation loss to prevent overheard conversations. A bit less loss will work for most cases. The common US technique to reduce speech intelligibility is to use pink noise to raise the noise level. If done subtly, it will reduce speech recognition without being annoying.

Now that you know the speech intelligibility rating for your room, you can go to the dean, principal, pastor, or person in charge with the knowledge that the room does or does not work for speech. (Music is a different story, so don't confuse the two!) But you may wish to know why.

Table 1: Sample word list to test room performance.

1. noose	39. stride	77. mole	115. oak	153. hath	191. might	229. lens	267. tide
2. hoe	40. rife	78. gate	116. such	154. brown	192. cage	230. close	268. peeve
3. part	41. spare	79. floss	117. tricks	155. fort	193. seat	231. we	269. rut
4. grog	42. tip	80. gab	118. oar	156. coil	194. nut	232. heat	270. base
5. dive	43. rend	81. nil	119. debt	157. rim	195. rice	233. bee	271. hour
6. weighs	44. our	82. buck	120. bake	158. shack	196. slip	234. wean	272. ale
7. cuff	45. hub	83. sate	121. auld	159. buck	197. hitch	235. cops	273. seen
8. rang	46. rid	84. rid	122. fib	160. pass	198. stroke	236. boat	274. cokes
9. urn	47. feet	85. rice	123. crypt	161. perk	199. flap	237. fish	275. frog
10. peep	48. desk	86. van	124. frame	162. heat	200. miss	238. bloat	276. drop
11. hike	49. punt	87. clout	125. drip	163. seek	201. rove	239. frond	277. swap
12. sad	50. hand	88. wife	126. lid	164. spread	202. awe	240. feast	278. crook
13. ding	51. yes	89. frump	127. this	165. kneel	203. spurn	241. big	279. gain
14. rove	52. fault	90. vault	128. tar	166. source	204. fudge	242. rove	280. quit
15. then	53. hook	91. writhe	129. raft	167. ram	205. bin	243. steak	281. grass
16. shrew	54. nine	92. sit	130. catch	168. poke	206. for	244. wave	282. hitch
17. bog	55. bar	93. dock	131. ream	169. whole	207. scar	245. pith	283. hand
18. goose	56. knit	94. hog	132. cat	170. wall	208. bass	246. when	284. strife
19. lick	57. ramp	95. pit	133. dab	171. wag	209. wow	247. leaf	285. vane
20. fudge	58. board	96. wire	134. oh	172. terse	210. have	248. roust	286. hay
21. pill	59. fizz	97. pop	135. till	173. woof	211. roar	249. crane	287. joke
22. flush	60. tan	98. fit	136. clasp	174. came	212. bean	250. pest	288. cream
23. dove	61. sauce	99. change	137. tar	175. spice	213. ship	251. fig	289. blind
24. cleft	62. mush	100. stash	138. plush	176. taste	214. out	252. oat	290. tread
25. zest	63. tab	101. right	139. false	177. cinch	215. curl	253. thrush	291. rut
26. pain	64. blob	102. flop	140. mart	178. wheel	216. vine	254. tuck	292. code
27. elf	65. dog	103. prize	141. pluck	179. soup	217. lapse	255. pets	293. deed
28. false	66. reed	104. slush	142. seas	180. bland	218. tea	256. tap	294. draw
29. chore	67. slight	105. fleck	143. nape	181. wood	219. jab	257. sap	295. lead
30. heart	68. pounce	106. nigh	144. row	182. blush	220. ah	258. floor	296. odd
31. murk	69. spice	107. vote	145. ebb	183. slip	221. fish	259. grain	297. he
32. puke	70. bag	108. wow	146. sub	184. cry	222. wit	260. bin	298. oh
33. bang	71. rash	109. boss	147. tear	185. tent	223. herb	261. sweet	299. ex
34. mar	72. crib	110. wall	148. scotch	186. pave	224. cents	262. geld	300. press
35. range	73. rid	111. clash	149. kit	187. lodge	225. rash	263. fur	
36. breed	74. ban	112. eight	150. lend	188. dote	226. cloud	264. mart	
37. cross	75. knot	113. mew	151. pad	189. fleet	227. posh	265. bash	
38. fuse	76. bull	114. clank	152. top	190. purge	228. rail	266. sack	

NOISE

You frequently treat noise as three different problems: the source, the path, and the receiver. You have some control over the first two, which I will address. If you are doing community noise work, then the nonlinear response of the third is your problem.

In the common speech model, spoken words are made up of distinct sound groups called phonemes. If you can hear the phoneme being presented at a level that is not masked by noise, then communication should be good. If most of the phonemes are received but a few are missed, you can usually figure out what was meant. Noise can be from external sources such as air conditioning, traffic, or other conversations. Or it can also be self-generated from the reflection and reverberation of phonemes that have preceded the one you are now trying to receive. In extreme cases it can be distortion induced by the amplifying apparatus.

The first design issue is how loud should speech be for optimum understanding. In 1929, Vern O. Knudsen (UCLA) determined that speech intelligibility improved in a quiet room as level was increased to 70dB. It began to degrade slowly starting at 80dB. This was astoundingly good research.

Modern acoustics as a science started in 1900. The vacuum tube was invented in 1906. Here were experiments that had accurate decibel readings, used a "distortionless" amp in a sound-reinforcement system and measurements by articulation tests. This is amazingly better work than some of today's efforts.

More recent research has shown that the level of speech above the background noise is important. This ratio of speech plus noise level divided by noise level is called the signal-to-noise ratio. Most people forget that when you are measuring the speech level it is not separated from the background noise. In the old days it used to be called S+N/N. Today we usually include the reverberant tail of the target speech as part of the noise number.

The intelligibility scores with various signal-to-noise ratios have been the subject of many experiments. Obviously, if you have lots of noise at a low frequency (say 20Hz), it will not cause as much

masking as noise at midrange (3kHz) frequencies. In the RASTI method when you have the equivalent of a 15dB signal-to-noise ratio for a given band and modulation, it is given the highest score of 1.00. A test in which the noise equals the signal gets a .50.

BANDWIDTH RESEARCH

Much research has been done on which bands of frequencies contain the information you need for good understanding. It should come as no surprise that the human ear is most sensitive to speech in the midrange frequencies of 1200-4000Hz.

One question of importance to system designers is how narrow the bandwidth should be to communicate intelligible speech. In one experiment to determine how narrow a bandwidth the ear can discriminate, a sine wave is compared to a narrow band of filtered noise. When the subjects cannot tell the difference, the width of the filter is noted. The other version of the experiment uses two oscillators set to the same frequency and then one is slowly changed until

two tones are perceived. This bandwidth, called a "Critical Band," should not be confused with the ability to perceive pitch. Select individuals can determine pitch differences to an amazing degree.

A common misperception is that the critical bandwidth is about $\frac{1}{3}$ octave wide. When Fletcher did the original research the bandwidth was much narrower. Modern methods (and definitions) give different results that are closer to a third of an octave over much of the auditory range. If you could just use $\frac{1}{3}$ octave equalizers to fix frequency response errors, life would be simpler. Resonances and other problems are often narrower or wider or, even worse, not on $\frac{1}{3}$ octave centers.

In addition, most of you would like to think that in critical listening, you can tell differences in level between two notes even when they are close together. As nice as it is to just slide a few knobs, pros will usually use fewer but more adjustable filters. A rule of thumb on large sound systems is if you are really using five or more parametric filters, you have a problem.

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In other experiments, just a narrow band of frequencies carrying speech is passed and the percentage of correct words received is measured. Some experiments use the critical bandwidth of frequencies, some use $\frac{1}{3}$ octave spacing, and a few use other bandwidths. This allows the prediction of the intelligibility results when you have less than full bandwidth communication or when noise masks some bands of frequencies but not others. It is handy to know how little bandwidth you can use to convey speech and which bands to use.

Of course, understanding which frequency bands contribute to intelligibility has been the subject of research for almost 100 years. That is why the wired telephone uses 300–3000Hz. It is cheap to implement and the second best choice of frequency bands to allow effective communication.

Many experiments have resulted in establishing a number of tables of speech recognition percentage versus a given frequency band. These have been incorporated into different standards. There is an ANSI standard S3.5-1997 R2002 that defines a method for predicting the Speech Intelligibility Index (SII, a more accurate name!) which correlates well with the results of the articulation tests for predicting speech intelligibility. The SII is scored from zero to one.

It is possible to estimate the speech intelligibility by measuring the noise (including reverberant tail) in $\frac{1}{3}$ octave bands, calculating how much effect each band of noise will have on the speech spectrum, and coming up with a single number score. There are programs that allow you to do this if you have all of the measurements. Some software allows you to estimate this based on the building information. You could refine this to perfection if you had identical humans talking and listening, the noise source was completely constant, you had full and accurate data on the room materials and construction, and you had no other room effects. That is why you use estimates or well-educated guesses in many cases.

A good rule of thumb is that when the speech is 6dB louder than the noise, speech is good; 15dB is great. When you measure the noise use the “A” weighting and slow response on your sound level meter. That combination will give you

one method of approximation as to how noise is perceived as interference. There are other scales used for better frequency weighting of psychoacoustic measurements, but the dB is the one I use the most so I will stick with it.

A method by Kryter (1962) is often simplified to look at the signal-to-noise ratio in octave bands, each band scoring its highest value (one) when the sound level *peak* was 18dB above the noise in that band. The lowest score (zero) occurred when the noise was 12dB louder than the speech. Each octave band had a maximum score depending on frequency (250-.06, 500-.15, 1K-.23, 2K-.32, 4k-.25). The final score gives an articulation index, in which .5 is just acceptable and .8 is pretty good. There is some argument as to which variant of the scale is most accurate and also as to exactly where the quality is acceptable.

SLUMBERLAND

There is a second factor that comes into play and Knudsen nailed it: a minimum level at which people pay attention! When the volume of the speech is below 70dBA, people fall asleep. In high schools and churches, I have set a noise level of less than 35dBA, and still people’s heads will nod when I drop the level below 70dBA. It works almost as well as a light switch!

In a large room people cannot talk at 70dBA. If you want that level for maximum speech intelligibility, you will need amplification. The simplest amp is the megaphone, a beautiful piece of acoustic design. You can build yours as a straight conic section, an exponential, or, for a real tweak, use a tractrix! I will stick with the straight cone, because rolling a piece of cardboard to this shape is easiest. Bad news: you need to do a bit more to meet modern expectations.

The conclusion is simple. The noise level in an auditorium should be as low as possible, but do not spend money getting it ridiculously so. A person talking normally without amplification can be 50dBish without really trying too hard. A great signal-to-noise ratio is 15dB. So a noise level of 35dBA can be acceptable for medium-size rooms. A larger room may require less noise if no amplification is used. It isn’t easy, but it is achievable. Besides, once the room is right, amplification can be added to overcome the noise.

NOISE SOURCES

The most common source of noise is the ventilation system. Most large modern rooms use a moving air (ventilation) system. Older rooms may use just steam or hot water heat. In steam-heated rooms, just replace the noisy valves regularly. In ventilated rooms the more the air moves the more efficient the transfer of energy. To save money, smaller ducts resulting in higher air velocity are sometimes used, thus more noise.

There are “low noise” outlets for air handling. These have a larger open area than normal to allow for slower airflow for the same volume of air. There are also “lined ducts” which have sound-absorbing material on the inside. The idea is to reduce the noise traveling in the duct. The best attenuation in an air duct is, of course, a right angle bend with absorbent material. The high frequencies travel into the absorbent wall and are diminished. Obviously the bigger the duct, the less flow is required to get the same volume of air motion and lower noise.

The second most common built-in noise sources I have encountered are noisy lights, which have bad—or, in some cases, wrong—ballasts. There are ballasts made for outdoor lighting, and can make much more noise than the ones designed for rooms. These outdoor ballasts are cheaper, but you risk rendering a room useless. Just replace the ballast.

The third in-room noise source is electronic devices. Both stationary devices such as computers, light dimmers, and so on, and portable ones such as video games and cellphones make noise. How annoying are cellphones ringing during public performances! I once went to the theater and the entire time someone was playing a portable video game. Signage and announcements may help prevent these noises and help keep the peace.

Most of these fixes are difficult to apply after a room is built but should be addressed during the design stage. The after-build fixes are usually limited to changing noticeably noisy diffusers, sometimes adding lined bends, and fixing broken parts such as squealing belts or bearings.

Another source of noise can be traffic, which is why carpet in the room(s) above or the hallway outside is a good

idea. If you reduce the sound generated, less is transmitted.

GEE-FORCES

Using a bit of attention, you can identify and sometimes even control the sources of noise. The next step is to fix the path by which noise gets into the room. The easiest way for noise to enter is through holes in the walls—often referred to as doors, windows, or “Gee I never noticed that.”

There are many versions of doors; obviously an open door transmits the most noise, thin veneer wood doors transmit a bit less. A heavy metal door much less, but if the door has a large gap around the frame a tight wood door would be better. This is where a good threshold is important.

Windows can be thick glass, or even double pane. Closing them greatly increases their ability to block noise.

The most common “Gee” is the invisible gap. Many rooms have ceilings of “acoustical” tile, which is designed to absorb sound, not block it. Plaster would block more sound, but then you would have other problems.

The problem with the ceiling tile is simple: Some sound goes through it. It is usually not the actual top of the room. There is a plenum space above the ceiling and below the floor above which may be used as part of the air handling system or just a space to run wires and pipes. You actually must look at the wall above the ceiling. I hope you will find the wall actually goes up to the floor above. Often it does not! Also, ensure there are no gaps between the wall and the upper floor. With a corrugated floor system, there could be many gaps.

Even if the walls are properly extended to the floor above, check the “penetrations” where the pipes, wires, and steel go through the wall. They should be filled with a rated fire-stopping compound. If not, this should be fixed. Old newspaper is not good here. In a recent room I filled the gap around the fire sprinkler head with foam to reduce the noise coming in by 3dB (**Photo 1**). The small gap reduced the effectiveness of a sound-rated door with seals, double-glazed windows, alternate stud walls, isolated floor, and all the other measures by half!

You may also need to look under items such as stages to be sure all your “Gee” gaps are filled. Do not forget that some things that look solid pass sound as if they were not there! Use your fingers to tap on what look like solid surfaces—a hollow ring will be a hint things are not as they seem.

I recently ran across a room in which the studio floor floated above the con-



PHOTO 1: Gee gap sprinkler head.

crete floor on a resilient system. Unfortunately, the isolated floor extended outside the studio to the hallway. In the studio, when somebody walked in the hall you could hear and feel it!

In a purpose-built room you often have two sets of wall studs, one set for each side. That way the fiberglass inside is not bypassed by solid contact. Of course, someone will drill holes to pass wires, or mount fixtures bridging the gap. Look for these things, too.

There are other issues such as wall thickness and construction that come into play during design and building stages. These are usually not inexpensive to fix later. My experience is that even with good design and construction, unless a room was purpose-built and frequently inspected during construction, you will have problems.

If you can find or calculate the ratings of all the building materials you can calculate what the losses will be before a room is constructed. To do this you must convert from decibels to the actual value of attenuation, then multiply this by the percentage of the total surface area and sum all the surfaces. When you turn this loss back into decibels, you will have the isolation for the room, except for what you forgot and the holes that will remain unfound. (You forgot that AC outlet wall box, I bet!)

A common misconception is to com-

pletely cover the walls, floor, and ceiling with sound-absorbing material. This will keep the energy from bouncing around and reduce it a bit, but it is better to keep it out. A completely “dead” room is not as good as one with the right kind and amount of “liveness.” But in rare cases that may be your only resort to reduce a problem. It almost never will fix it.

I once visited a television production facility that was having too much noise go from its machine room into the edit suite. An editing suite is a special case of a room in which the occupants wishing to make sound judgments are more sensitive to noise than usual. There was a hole in the wall for an equipment rack between the two rooms and all the windows had at least 2” of gaps above and below the glass. It was a design style! I probably could have charged them a healthy fee if I did not break up laughing. Oh yes, they had paid someone else for the acoustic design.

To check whether a room is quiet enough, you should have a sound level meter. There are a number of inexpensive and surprisingly accurate ones available. When you use the meter you will notice the meter reading bounces around a bit. The better meters do this a bit less. Do not use the highest or lowest number. With some experience you will see that this is a number that it all seems to swing around. This is often about 10-12dB lower than the *high* number when measuring speech.

Do not be tempted to use the “C” scale because it is “flatter.” The “A” weighted scale (slow) is much more useful for these purposes. You can use the “C” scale to see whether you are fixing low-frequency problems, particularly if you have an inexpensive meter that does not have a very sensitive lower range, but be careful you are not chasing something that is barely heard instead of what counts!

If you ever plan on building an auditorium or buying a sound reinforcement system, be sure the designer or seller actually owns a meter. If they are in the business and don’t . . . WOW!

If you stop the noise from coming into a listening room, you also keep it from leaking out. Noise is not the only problem to worry about. . . as you’ll see in next month’s part 2 article. **ax**



An Audio Isolation Box

This simple transformer-coupled unit allows you to send line-level audio 100' away, while isolating AC line ground-loop pickup by over 120dB. Audiophile quality is maintained.

By Dennis Colin

My wife wanted to make CD copies on her computer of LPs and tapes, whose players were about 70' from the computer. A direct cable feed caused audible 60Hz hum. A 2.4GHz wireless A/V link (Radio Shack, about \$100) was noisy at that distance. I quickly dismissed the idea of floating the AC line safety ground at one end as dangerous.

The unit I describe here uses a stereo pair of Hammond 850H audio line transformers, rightly described as broadcast quality: electrostatically shielded, hum-bucking windings, rugged metal case, 4Hz –90kHz BW at –1dB, < 2% THD at 20Hz at 2V RMS (0dB full-scale Red Book CD standard), probably less than 0.01% THD at normal line levels.

CONSTRUCTION

Photos 1–5 show the unit, built in an aluminum Hammond 1590S box, 4.35" × 3.21" × 1.57", Digi-Key P/N HM158.

The output RCA jacks must either be of ground-insulating type or floated from the metal box. I did this by punching four 3/4" holes in the box with a Greenlee chassis punch ("knock out punch unit," model 730BB-3/4), then epoxying a strip of 1/8" hardwood inside, covering the holes. Then I drilled 1/4" holes in the wood, centered in the 3/4" box holes, and mounted the RCA jacks. Before mounting the wood strip I painted it with enamel to prevent moisture absorption electrical leakage. As a testament to the box's overall insulation including the transformers, the input/output leakage resistance is 4000MΩ.

The input/output ground (cable shield) capacitance coupling is 436pF, which at 60Hz is a 6MΩ impedance. Therefore, even in the unlikely (and dangerous!) event that the audio equip-

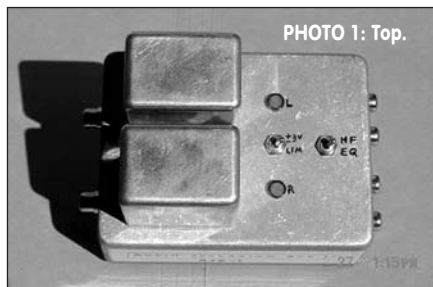


PHOTO 1: Top.



PHOTO 2: Front.

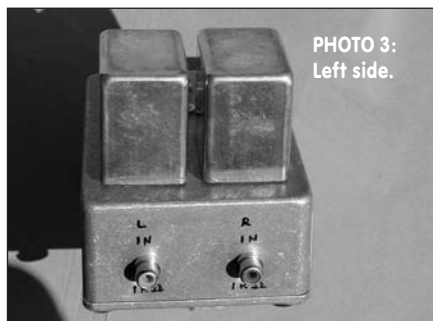


PHOTO 3: Left side.



PHOTO 4: Right side.

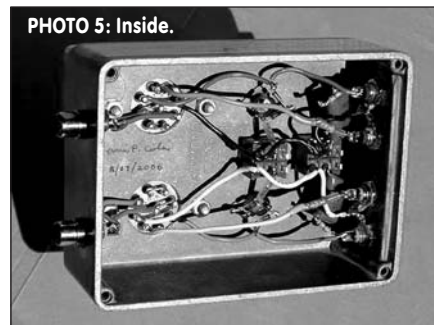


PHOTO 5: Inside.

ment being driven has its "ground" shorted to the hot 120V AC line, the 60Hz leakage current through the box would be only 20μA, not even enough to feel a tingle.

SAFETY WARNING: VERY IMPORTANT—Before you contact both source and receiving setup cables simultaneously (yes, people have been electrocuted by "only" 120V!), please measure the AC voltage between those two cable shields! If all is well, it shouldn't be more than 5V, the maximum drop expected between line grounds at opposite ends of a house.

My house originally had the safety "ground" of an outside outlet connected

to the hot 120V by some "electrician"—I discovered this when some contractors were using the outlet for some equipment, when someone started yelling about shocks! Thankfully, he lived—of course, I should have checked the wiring when moving in. I promptly had it fixed and tested it (and all the other outlets, inside and out). Whew!

I recommend reading Charles Hansen's excellent articles about the AC power grid and safety in this magazine. The reason I'm emphasizing this is that the box described here isolates two audio systems' (supposed) AC line grounds, and that it's all too easy to take "only 120V" for granted. OK, enough preaching.

SCHEMATIC

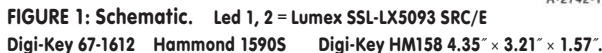
The transformers (**Fig. 1**) have a nominal stated impedance of 1200 to 600Ω, but will provide this 2:1 impedance transformation (1 to 0.707 voltage step-down) over a wide range of source and

Two stereo pairs of box ground-isolated RCA jack outputs are provided. J3 and J4 are the cable-driving outputs, while J5 and J6 are 300 Ω headphone-

With no load, or up to 100' of cable driving hi-Z loads, the transformers have a "ringing" resonance at between 200kHz (no load) and about 50kHz (driving 100' of Alpha 2466C twin-insulated pair cable). Source impedances below 1k Ω (such as headphone outputs) exacerbate this resonance. The resonance is between transformer leakage inductance and load capacitance

While the resonance frequency is ultrasonic, SADCs can have content up to 100kHz, and vinyl records can also have ultrasonic content. R3/C1 and R4/C2, when switched on by S2 (“HF EQ”), form a resonance damping network.

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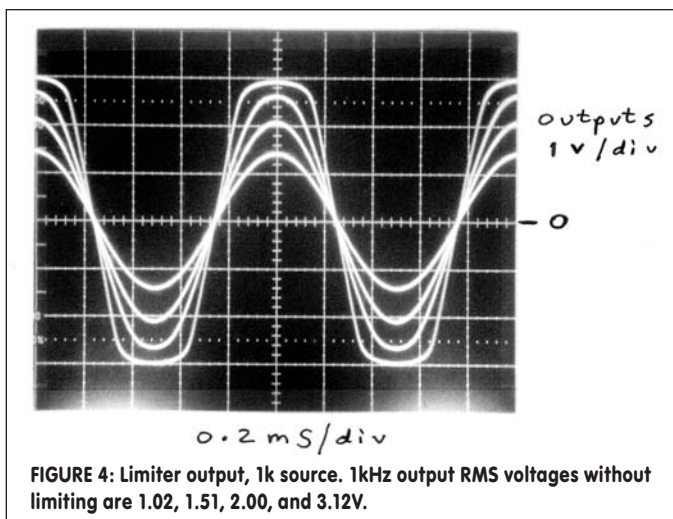
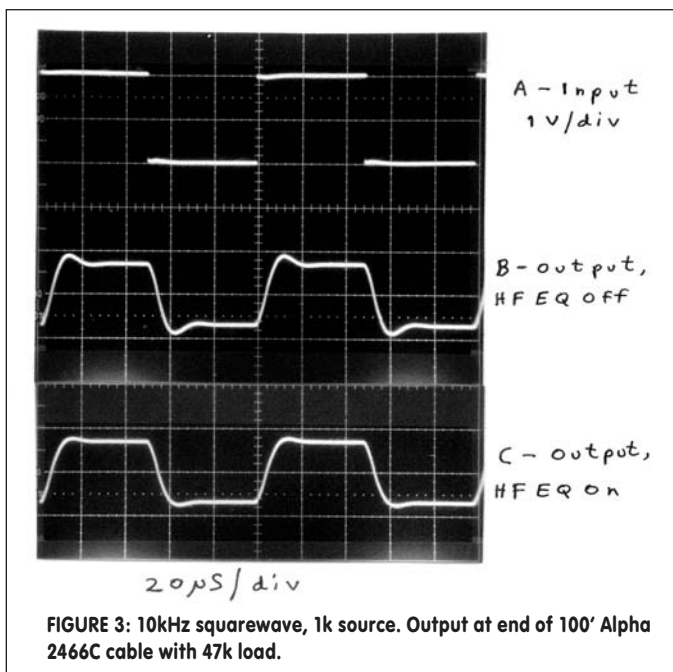
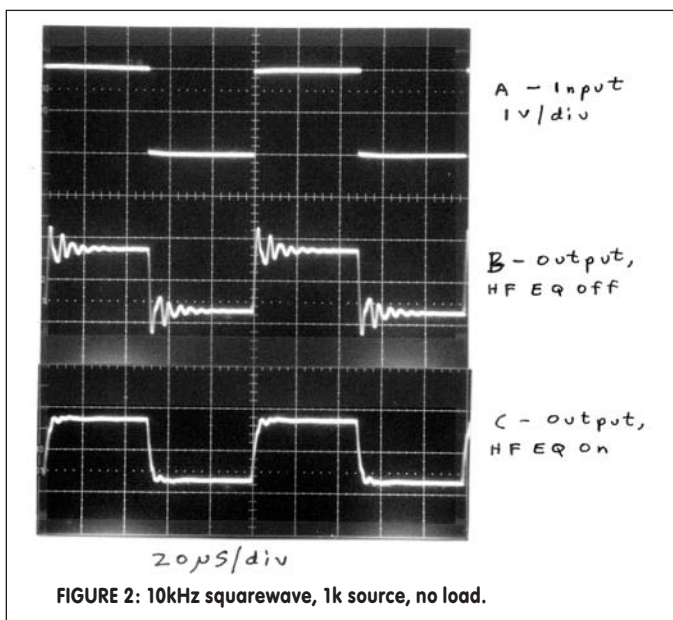


Figure 2 shows 10kHz squarewave responses with 1k source and no load, with (2C) and without (2B) the HF EQ.

The reason for including switch S2 is that depending on cable and load, there might be enough HF damping to not need the EQ.

Figure 3 shows the outputs at the end of 100' of Alpha 2466C cable (5.35nF capacitance) terminated in a 47kΩ resistive load. With HF EQ on (**Fig. 3C**) the overall frequency response reaches a "peak" (insignificant) of 0.07dB at 10kHz, and is -0.03dB at 20kHz. Above this the response monotonically falls (no peaks or dips) to -3.2dB at 50kHz, and -31dB at 250kHz. The latter provides good rejection of AM radio interference and the digital noise surrounding computers.

PEAK LIMITER

You do not want to ever overload the input to an A/D converter preceding a digital recording setup. Even momentary peak "digital clipping" produces a broken-glass-sounding "crunchy splash" of musically irritating wideband IM transients.

The Red Book CD standard defines 0dBFS (full scale) as 2V RMS for a sinusoidal peak, meaning a maximum peak voltage excursion of ±2.83V. Several CD/SACD players I've measured can output peaks (with some recordings) reaching about ±3.5V.

The -3dB voltage stepdown of this isolation box provides additional margin. Just in case (e.g., driving the box from a headphone output, or a phono preamp or tape recorder), switch S1 enables a diode bridge/LED peak limiter. The specified LEDs have a low drop (1.70V at 5mA) and high brightness, visible even with 0.1mA.

Figure 4 shows the limiting characteristic, which limits very smoothly, producing low-order odd harmonics similar to a push-pull tube amp without feedback. Much, much better than "digital clipping!"

MEASUREMENT SUMMARY

Table 1 shows data on the isolation box alone, **Table 2** is with the box driving 100' at Alpha 2466C cable, and **Table 3** shows characteristics of the cable. Parts cost for the isolation box is about \$220.

CABLE DATA

I purchased a 100' spool of Alpha 2466C SL 005 (SL for slate color) cable from Digi-Key, P/N A131-100, for \$43.52. It has two individually shielded pairs of 22-gauge conductors, plus an overall shield. I think the three shields are in contact, but I connected them and one conductor of each pair together. At each end I connected a red (right) and green (left) RCA stereo pair of plugs. The green plugs (Digi-Key CP-1452) connect to the cable's green wire (hot) and white wire plus all three shields (plug shell); the red plugs (Digi-Key CP-1448) connect to the cable's red wire (hot) and black wire plus three shields (plug shell).

This provides a nice convenient thin and flexible 100' stereo RCA cable, easy to use and roll up when not in use (wives don't like leaving long cables all over the floor, but

neither do I after having tripped on it).

Figure 5 shows 200' of this cable (left plus right looped back) responding to a 100kHz squarewave. At the cable input, you can see the delayed reflection (in phase with open load, **Fig. 5A**; out of phase cancellation with shorted load, **Fig. 5C**). **Figure 6** shows a time-expanded view of the cable's delayed and HF-attenuated response.

You may take such waveforms for granted, but **Fig. 6** shows a signal traveling at 449 million mph, which is 67% of the speed of light. Needless to say, with 100' of this cable having a delay of 0.3% of a 20kHz cycle period (1° of linear phase shift), there are no "wavelength effects" to influence sound quality! (Of course, you could spend \$30,000 for 100' of "audiophile interconnect.")

LOW FREQUENCY PERFORMANCE

With a 1kΩ source, the 850H transformer has an LF (-3dB) point of 1.7Hz, and is down only 0.32dB at 20Hz. But I wanted to determine its 20Hz signal-handling ability. Ham-

mond states +15dBm as maximum power, which is 4.36V RMS across 600Ω; I'm using the "600Ω nominal" winding as the output secondary. Hammond states 1.5% THD at 20Hz, +15dBm.

But that's with the windings each seeing their nominal impedances for source and load, equivalent to driving the 1200Ω winding (used here as primary) from a 600Ω source with no load, as far as magnetizing current distortion is concerned (more on this follows).

With my 1kΩ source/47kΩ load testing, the nonlinear magnetizing current will produce more distortion of the signal voltage. **Figure 7** shows 20Hz sine response. In **Fig. 8**, a 20Hz squarewave is used. **Figure 8A**, with an input of ±1V peak (1V RMS for a squarewave), shows only a slight tilt due to phase shift from the 1.7Hz 3dB rolloff.

But in **Fig. 8B** I increased the driving amplitude until core saturation became evident. The output, at each squarewave jump, starts with a large, fast overshoot, as the core "snaps" out of saturation from the previous half-cycle.

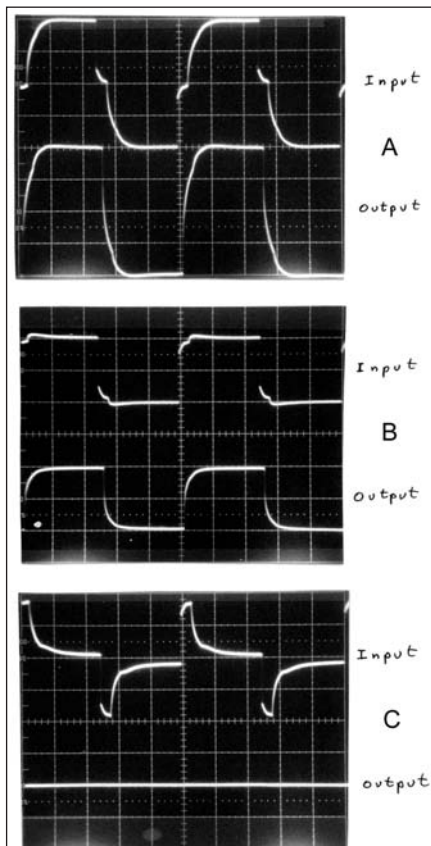


FIGURE 5: 200' Alpha 2466 cable, 100kHz squarewave from 50Ω source. 1V/div, 2μS/div.

TABLE 1: AIB-1 MEASUREMENTS.

In/out ground isolation: 126dB at 60Hz, 76dB at 20kHz
in/out ground coupling: 436pF, 4000MΩ
AC line pickup, 12" from an 80W power supply:
1.8μV RMS = -113dBu = -121dBFS (0dBFS = 2V RMS)
impedance stepdown ratio = 2:1
voltage transformation ratio = 0.688 = -3.25dB
insertion loss re stepdown ratio = 0.24dB
frequency response, 1kΩ source, 47kΩ load, HF EQ on:
-0.32dB at 20Hz, +0.08dB at 20kHz
-1dB at 4Hz, 90kHz
-3dB at 1.7Hz, 150kHz
maximum input (2% THD): 2.0V RMS at 20Hz, 10V RMS at 100Hz
THD at 0dBu (0.775V RMS) input:
0.3% at 20Hz; approximately 0.01%, 100Hz - 20kHz

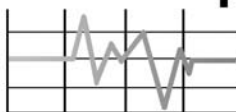
TABLE 2: AIB-1 DRIVING 100' OF ALPHA 2466C SL005 CABLE.

HF response w/47k//100pF load, 1k source to AIB-1:
+0.6dB at 20kHz (HF EQ off)
-0.03dB at 20kHz, -3dB at 48kHz (HF EQ on)

TABLE 3: CHARACTERISTICS OF 100' ALPHA 2466C SL005 CABLE.

(2 pair, shielded individually and overall; between one #22 conductor and other one connected to shield)
DCR: 1.61Ω (conductor), 1.57Ω (shield)
C = 5.35nF
L = 12.6μH round trip, measured at 50kHz
Zo (characteristic impedance) = 48.5Ω at 50kHz
fo (quarter-wave resonance) = 1.348MHz
propagation velocity = 67% speed of light

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Then it maintains the desired 10V level for about 10mS, followed by saturation, gradual at first, then complete just before the next jump. At this point, the magnetic flux can't continue increasing (as needed to maintain an induced voltage) so the voltage collapses to zero.

The product of voltage and time duration is a constant for a given number of turns on an iron core; in this case that saturation constant is about 100 millivolt-seconds.

HYSTERESIS LOOP (B-H CURVE)

I saved the most interesting waveshapes for dessert (**Fig. 9**). "H" symbolizes magnetic driving force (magnetization), analogous to voltage or water pressure. It's proportional to current times the number of winding turns (ampere turns).

Iron atoms have an organized elec-

tron-spin effect, making the atoms microscopic magnets. In unmagnetized iron, the atoms are randomly oriented, so the individual magnetic fields cancel, averaging out to zero. But an applied external magnetic field makes these tiny magnets tend to align. In "hard" steel, they stay aligned after the external force is removed. This is called "remanence" (for "remaining") and is used in permanent magnets.

But in "soft" iron used for transformer cores, remanence is undesirable, causing energy to be wasted overcoming this magnetic "stickiness."

"B" symbolizes magnetic flux, analogous to electric current or water flow rate. In iron, and other magnetic materials, the permeability (ratio of flux to magnetic force, or B/H) is greater than that of a vacuum or air (defined as permeability = 1). This is because an

external magnetic field starts aligning the tiny iron atom magnets; this alignment increases the overall flux much more (by thousands) than in air. But as the externally applied field is increased, eventually the iron atoms are as aligned as the molecular structure allows; this is saturation. Beyond this point, only the vacuum permeability of 1 can add flux; beyond hard saturation you have an air-core transformer, useless except perhaps at radio frequencies.

Figure 9A shows the test setup for displaying B-H curves. Primary current, which is sensed by R1, is proportional to H, the magnetic driving force. The primary voltage, proportional to the rate of change of flux, could have been sensed, but I used the secondary voltage for two reasons: 1) the primary voltage has an additional component, the drop across its DC resistance, whereas the secondary current

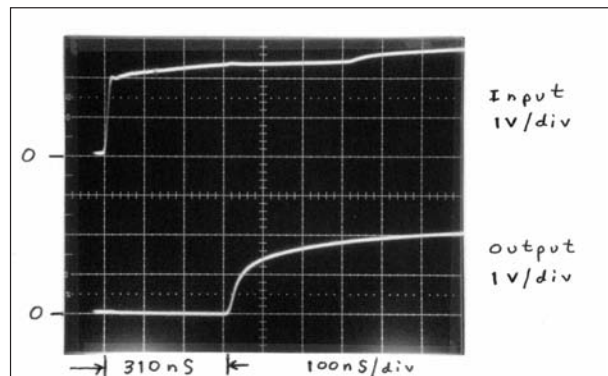


FIGURE 6: 200' Alpha 2466 cable + 5' RG58 C/V cable, 500kHz squarewave from 50Ω source. Cable load = 50Ω, 2466C cable $Z_0 = 48.5\Omega$. BW (-3dB) = 1.2MHz.

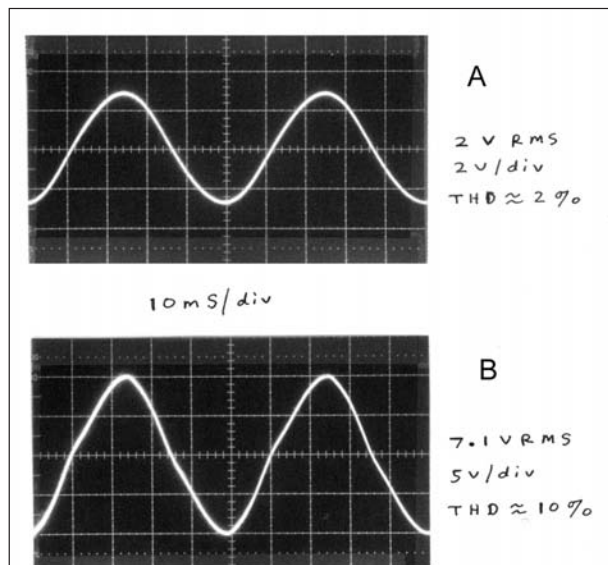


FIGURE 7: 20Hz output, 1kΩ sine wave source, 47kΩ load.

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Because the induced voltage is proportional to the rate of change (time derivative) of flux B, the latter can be obtained by integrating (the inverse of the derivative) the winding voltage. R2 and C1 serve as an integrator. An op amp with feedback cap would form a virtually ideal integrator, but because the

2. The opening, or separation, of the

Fig. 9C shows the B-H curve with only 1V RMS applied, again at 18.9Hz. Here, the curve appears smooth and el-

FIGURE 8: 20Hz output, 1k Ω squarewave source, 47k Ω load.

FIGURE 9: Hammond 850H Primary B-H curves (hysteresis loops).

liptical. At this level, much below saturation, the elliptical shape is probably not from hysteresis, but rather from resistive losses, especially the 10k secondary load. A volt-amp X-Y plot of an ideal inductor would be an ellipse or circle (with equal X-Y excursions).

But here the R2/C1 integrator's 90° shift would cancel that of a pure inductor. However, any resistive losses (core loss, winding resistance, or most likely the secondary R2 load) will cause a net overall (including integrator) phase shift. That the shape of **Fig. 9C** is not due to nonlinearity is supported by the observed low distortion of this trans-

former with 1V RMS applied at 20Hz and 18.9Hz.

Figure 10 shows the locations of parts that I used on the box.

CONCLUSION

Table 4 is the parts list. While describing a simple isolation box, I've taken the liberty of graphically displaying some fundamental physical principles of cables and audio transformers, and also electrical safety regarding AC line grounding imperfections. One or more of these boxes will allow you to use your house as a low-noise recording studio. *ax*

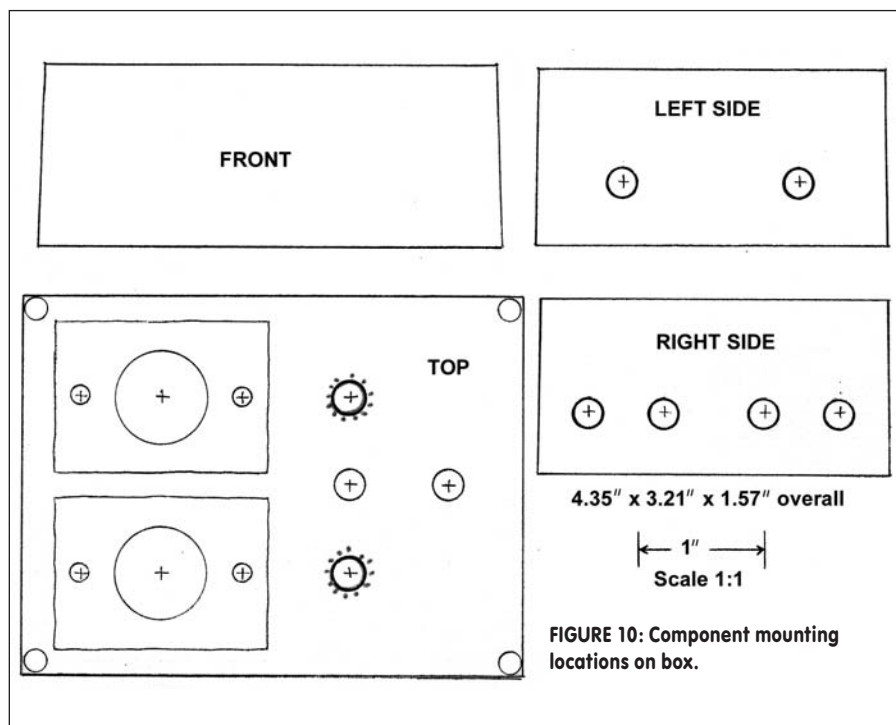


TABLE 4: PARTS LIST.

Ref	Description	Manufacturer	Part#	Distributor	Part#	Price
Box 1	box, 4.35" x 3.21" x 1.57"	Hammond	1590S	Digi-Key	HM158	\$10.88
C1, 2	cap, 1.8nF, 100V			Digi-Key	PS1182J	
D1-8	diode		1N4148			
J1, 3, 5	jack, RCA, green			Digi-Key	CP-1417	
J2, 4, 6	jack, RCA, red			Digi-Key	CP-1413	
LED1, 2	LED, red	Lumex		Digi-Key	67-1612	
R1, 2	Res, 180Ω ¼W					
R3, 4	Res, 1.8k, ¼W					
S1, 2	switch, toggle, DPDT			Digi-Key	360-1156	
T1, 2	transformer, 1.2k-600Ω	Hammond	850H	Spartan*		\$98.77ea
	cable, twin, 100'	Alpha	2466C SL005	Digi-Key	A131-100	\$43.52
	plug, RCA, green (2)			Digi-Key	CP-1452	
	plug, RCA, red (2)			Digi-Key	CP-1448	

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TABAQ Tang Band Quarter Wave

A transmission line designed and optimized for the Tang Band 3" full-range drivers.

By Bjorn Johannesen

Inspired by debates on the Internet about quarterwave designs with Tang Band 3" full-range drivers, I decided to design my own quarterwave loudspeakers. They sure deserve a better construction than those found on the Internet.

In the last couple of years, I have worked with the very useful simulation models from Martin J. King and designed and built a successful TL sub ("The HideAway TL Sub," 7/06 *aX*). Working with Martin's models is quite educational, and I feel confident about

quarterwave (transmission line) design. Please refer to my article "Martin J. King for Dummies" on www.t-linespeakers.org/.

The TABAQ Sound Pressure Level (SPL) is impressive for such a small driver, with a very smooth and broad frequency response. The dashed line in **Fig. 1** is the Infinite Baffle (IB) response, and the solid line is the summed output from the driver and the opening.

The sound is open with no unwanted resonances, and the bass is relatively deep and clean. The first time you listen to this speaker, you will be impressed by

the quality of the sound and performance of the bass, bearing in mind the size of the driver and the lack of financial problems building it. The prob-

lems with baffle step are corrected with a simple filter, which balances the SPL in the listening room.

TANG BAND DRIVERS

You can use all the 3" drivers in this construction. My first test was with W3-926S, which I replaced with W3-315SC. The Tang Band drivers have a relatively high F_s and very high Q_{ts} , which means the rolloff is more gentle than a low Q_{ts} driver, and it is therefore possible to design a quarterwave with useful output well below F_s . However, high Q_{ts} drivers can be difficult to control.

The whole idea with a quarterwave—or transmission line—is to extend the bass by the contribution from the opening. At the cabinet resonance, the output adds to the driver output. However, there are some unwanted higher harmonics, which are out of phase with the driver, causing uneven frequency response. These problems can be solved

by different design parameters, which I have successfully used with the TABAQ design.

DESIGN

The tuning frequency of the cabinet alone should be set lower than the driver F_s . Because the drivers all have a very high Q_{ts} , I started to see how low I could go, getting maximum bass performance out of the drivers without losing the overall frequency balance.

After hours of simulation with Martin's MathCad models, I found the best compromise was 55Hz, which is low for drivers with an F_s of 100 or 110Hz. But it works!

The tuning frequency is set by the length of the pipe and the geometry. A tapered pipe is shorter than a straight

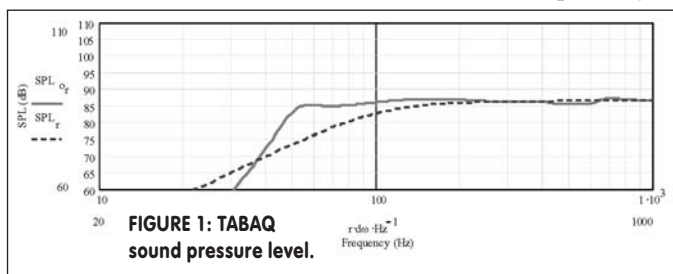


FIGURE 1: TABAQ sound pressure level.

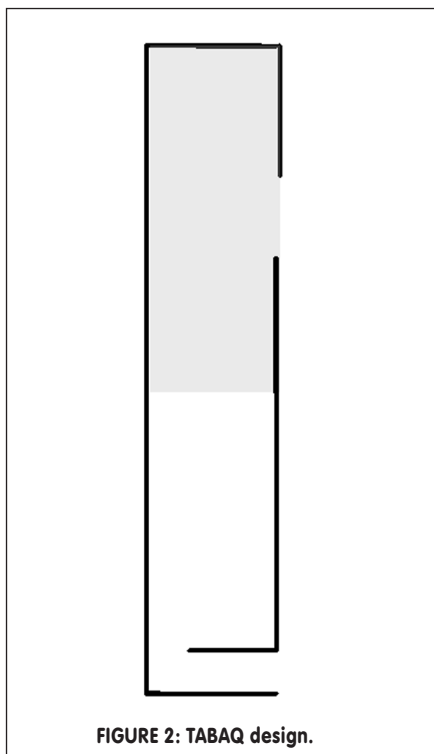


FIGURE 2: TABAQ design.



PHOTO 1: TABAQ.

pipe for the same tuning frequency. TABAQ uses the mass loading principle, in which the last part of the pipe has a smaller area than the rest of the pipe. The air in this part of the pipe adds resistance to the driver (more mass-mass loading). Compared to a straight pipe, the length is shorter.

The problems with unwanted higher harmonics are solved by driver placement, stuffing, and applied the geometry as in the mass loading. The driver is one-third down the pipe, a common and well-known trick to attenuate the upper harmonics.

However, this is not enough, and damping of the cabinet is needed (as it is in every quarter wave). The damping is high (1 lb per cubic foot) and is placed in the upper part of the pipe. The last third of the pipe is not stuffed. The stuffing will attenuate the output from the opening, so I solved this by increasing the volume of the cabinet.

Increasing the volume also increases the output from the opening, so this compensates for the high density stuffing. The area of the pipe is four times the area of the driver (S_d).

The TABAQ design is shown in **Fig. 2**. Don't be confused. This is not a bass reflex enclosure; it is a quarterwave. Rob Brines has a good explanation on his site: www.geocities.com/rbrines1/Pages/QWR_not_BR.html.

LISTENING ROOM

Until now, I have focused on design theory. Real life is different, and I must deal with

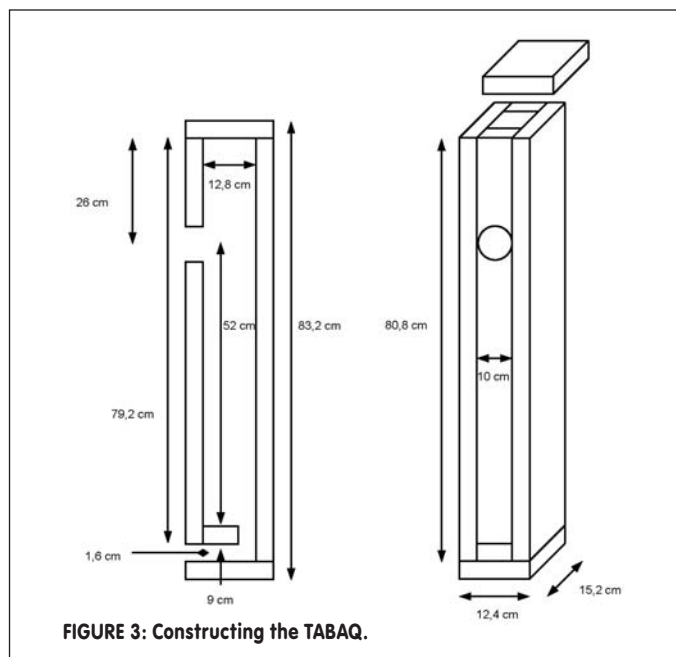


FIGURE 3: Constructing the TABAQ.

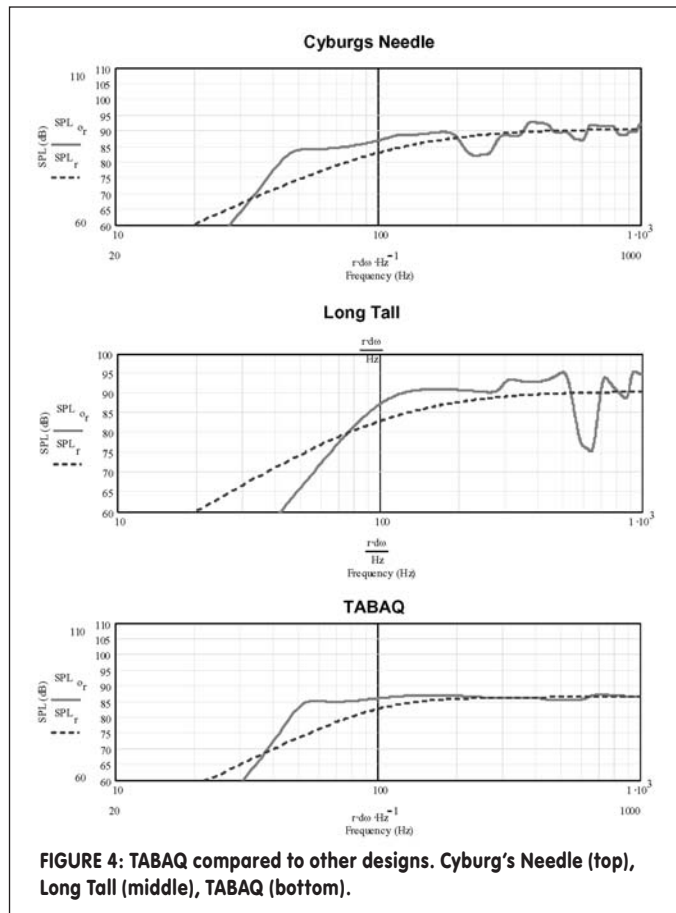


FIGURE 4: TABAQ compared to other designs. Cyburg's Needle (top), Long Tall (middle), TABAQ (bottom).

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the baffle step problem, which generally applies to all full-range designs.

You already know that a speaker placed in the middle of the room has a weaker bass than one placed against the wall. The same thing happens for the middle and high tones, which get extra help from the front of the cabinet. The result is a speaker that sounds bright with a focused middle tone. This “analytical” sound will be irritating after some time.

Therefore, you need a baffle step compensation circuit to attenuate the higher

frequencies. Does it sound complicated? It is not. Simply include a resistor and an inductor in parallel and place them in the + cable to the driver.

Martin King's latest MathCad models now include calculations needed for baffle step compensation. For the TABAQ, you need a resistor value of 3.9Ω and an inductor value of 1.0mH . This is 4dB attenuation at 700Hz. If your cabinet is more than 1m from the wall, or you would prefer more attenuation of the higher frequencies, 6dB attenuation is accomplished by using 6.8Ω and 1.5mH instead.



PHOTO 2: Parts for building TABAQ.

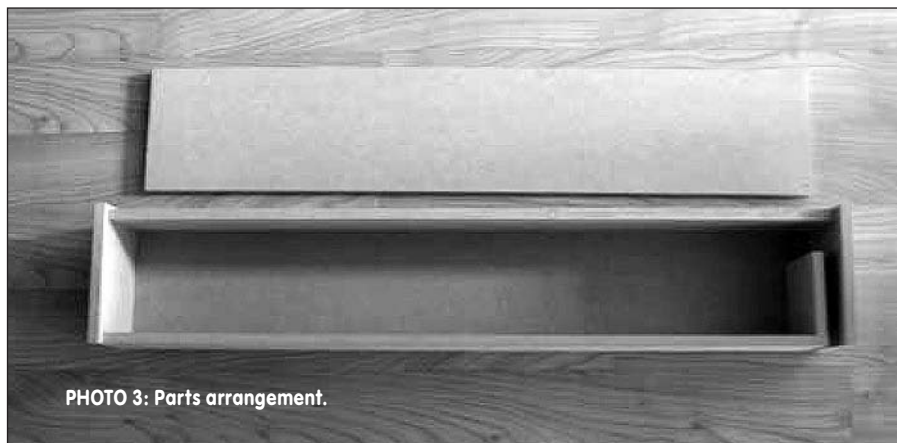


PHOTO 3: Parts arrangement.

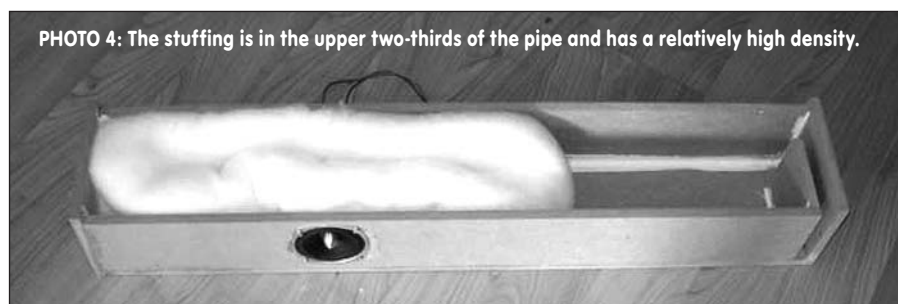


PHOTO 4: The stuffing is in the upper two-thirds of the pipe and has a relatively high density.

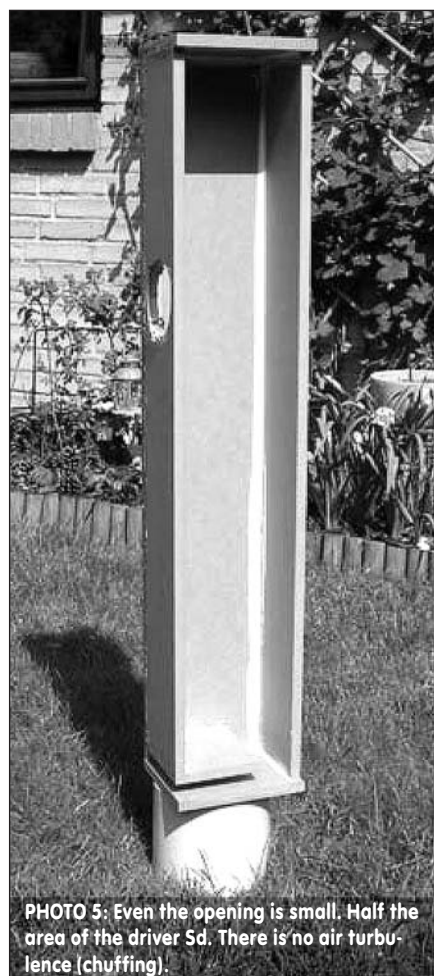


PHOTO 5: Even the opening is small. Half the area of the driver S_d . There is no air turbulence (chuffing).

I have tested TABAQ with different kinds of music and placed the cabinet in the middle of the room, at the wall, and in the corner. The speaker is easy to place in the listening room, and the bass is well controlled.

I have even listened to organ music. Of course, it is not like being in church, but I was impressed by the lack of distortion.

CONSTRUCTION

You need 12mm MDF pieces in the following sizes:

- 12.4cm × 15.2cm, 2 pcs, top and bottom
- 79.2cm × 10cm, 1 pc, front
- 80.8cm × 10cm, 1 pc, back
- 9cm × 10cm, 1 pc, opening
- 80.8cm × 15.2cm, 2 pcs, side
- Stuffing: 100 gram (0.23 lb) in the upper two-thirds of the cabinet
- Filter: 1 resistor 3.9Ω and 1 inductor 1.0mH

Information about the Tang Band drivers can be found at www.tb-speaker.com/



PHOTO 6:
Please note,
the beer is
non-alcoholic.

I would like to thank Martin J. King for his outstanding work and for educating us in the noble art of quarter wave design. www.quarter-wave.com

I also want to thank the Danish Tang Band reseller Michael Sjorslev for his support www.jumon-audio.dk/

As a final note: TABAQ is also doing very well with the 4" driver W4-1320SA. *ax*

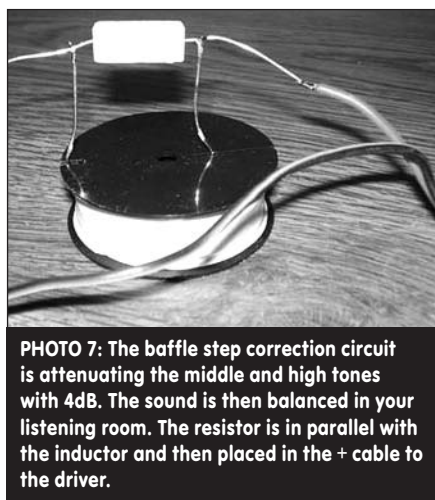
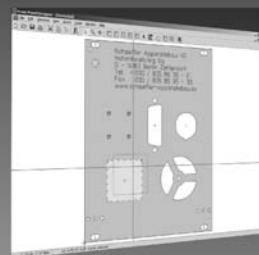


PHOTO 7: The baffle step correction circuit is attenuating the middle and high tones with 4dB. The sound is then balanced in your listening room. The resistor is in parallel with the inductor and then placed in the + cable to the driver.

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By Bill Waslo

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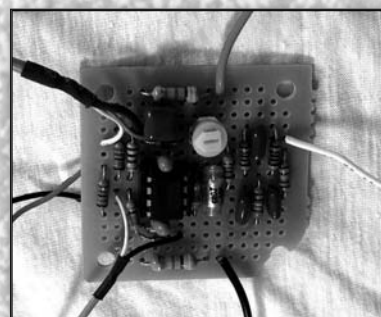
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A Prototyping System for Passive Crossovers

By Ramkumar Ramaswamy

Today's computer simulation software allows you to design passive crossovers extremely accurately. Nevertheless, the proof of the pudding invariably lies in building a "brick-and-mortar" prototype that you can plug some real drivers into, and play around with some real component values to achieve the final result.

Assuming you have completed a computer-based crossover design, you will have a set of finalized-in-software component values—inductors, capacitors, and resistors—that are ready to be realized in a prototype. When you build a prototype at this stage, you will typically find it more appropriate to validate and fine-tune the results by playing around with the capacitors rather than the inductors. With inductors being bulkier and more expensive, it is usually more practical to design a crossover around standard inductor values and use capacitors in parallel to fine-tune the crossover characteristics, rather than the other way round. This observation was an important consideration for me in arriving at an effective prototyping model.

Alligator clips and breadboards are two popular ways to produce a prototype. However, my experience was that the plug-and-play process using these methods can be quite cumbersome for the experimenter and quite taxing for the components involved, too. It is not a pleasant experience to render an audio-ophile-grade capacitor with bent, weakened, or broken leads after a few plug-and-play operations using traditional

breadboards or alligator clips.

Is there a way to effortlessly and reliably plug and play with a set of (fixed) capacitors across crossover prototypes without the fear of ending up with broken leads? This question drove me to design this passive crossover prototyping system.

It would be appropriate to point out that this prototyping system involves a nontrivial investment of money, but if you are someone who builds more than a couple of prototypes a year, you will find that this system makes your job much easier and quickly pays for itself.

THE SOUL OF THE IDEA

As **Photo 1** shows, the central idea is



PHOTO 1: A capacitor mounted on a dual banana plug.

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to have each capacitor mounted on a dual 4mm stacking banana plug. I used a Pomona MDP, which is a solderless type with set screws. Another possibility is Pomona 1330-ST, which comes with solder turrets. The use of stacking banana plugs helps in multiple ways:

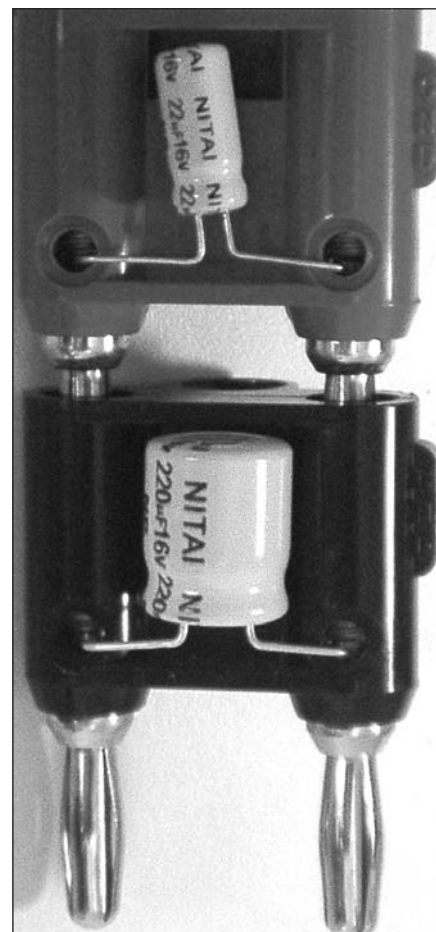


PHOTO 2: A pair of stacked up COBs.

• First, the contact resistances are of the order of a few milliohms each—very important when you are dealing with signal paths that are of the order of just several tens or hundreds of milliohms.

• Second, once you fix the capacitor or resistor to the banana plug, the act of handling the capacitor leads is done with once and for all: in all future plug-and-play operations you handle only the banana plug. This means that you can plug and play as much as you wish without fear of component damage or contact deterioration.

• Third—and this is key to the rapid prototyping ability—you can easily build up the desired capacitor values by simply stacking up the capacitor-on-banana plug (COB) combination. **Photo 2** shows a pair of stacked-up COBs.

If you are using banana plugs with set screws, experiment first with a couple of throwaway capacitors to assess the “fit” of the screw. All the Pomonas I used held the component leads very snugly indeed, but I had somewhat less ecstatic experiences with less expensive brands of plug.

Quality banana plugs such as Pomonas when purchased in retail are pretty expensive—about \$3 a piece. It might

initially seem strange to mate a \$3 plug with a capacitor of similar or perhaps even lesser value, but those three advantages make for a good return on investment as you go. It is certainly worthwhile to do some research and smart shopping for surplus plugs for this purpose. I managed to get a few dozen new Pomona plugs from eBay for as little as \$10 for a dozen, which is less than a buck a piece. Some savings!

In my experience the 4mm sockets are probably less critical than the plugs. You should be able to use more or less any brand that is easily available, especially as surplus stock. I used a standard nickel-plated 4mm socket-cum-binding post with solder tags.

CONSTRUCTION

The choice of material for the board is not critical: I used a 3mm phenolic board, but you may choose to use 3mm hardboard, for example. The board I used measured about 15 × 12” and holds enough sockets for prototyping three-way 18dB/octave parallel crossovers.

Photo 3 shows the component side layout of the completed system, and **Photo 4** shows the reverse side with interconnections. The overall layout of the board can resemble the topology of

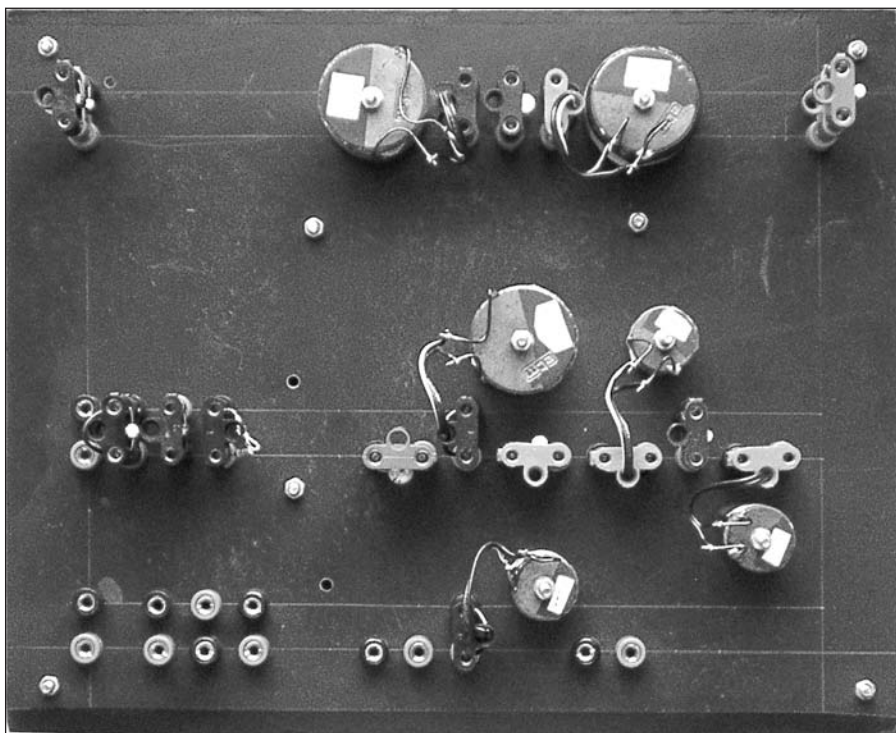


PHOTO 3: The component side of the completed system.

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the way you would schematically draw the crossover. I can also use this system for two-way and lower order crossovers simply by plugging appropriate shorting cables on the component side attached to 4mm banana plugs, or by affixing them to the binding posts.

During construction it is important to make sure you drill carefully to account for the spacing between the two pins of the dual plug (19mm) so you don't need to exert unusual force to insert or extract the COB. In drilling the holes for the sockets, begin by drilling a pilot hole and then drill the main hole. A thin round file is useful if you find, after drilling, that the hole is not perfectly centered and the dual banana plug does not quite fit perfectly. The file allows you to "extend" the hole in the appropriate direction. (Needless to say this is a last resort; as always, "Measure Twice, Cut Once" should be your guiding principle.) In my design the distance between two banana plugs was about $\frac{3}{4}$ " to allow for space for a large capacitor; to this you may add any additional space you think you may need to accommodate your finger.

The wire used for all interconnections must be thick enough to keep resistances insignificant compared to the resistances of the inductors. You should use at least 22 AWG wire, which has a DC resistance of approximately $16\text{m}\Omega$ per foot; 18 AWG wire has less than half that resistance. You may find it helpful to mark the interconnections on the component side too for convenience.

You may also be worried about the existence of stray inductances due to the long lengths of wire hanging around, but this is not a problem at audio frequencies: a foot of straight wire has a self-inductance of less than 0.0005mH , which is insignificant.

The board needs supports to keep it clear of the table. Because the board is fairly large, it may be susceptible to some bending as you insert the banana plugs. Therefore, in addition to the four corner supports, I added a few more internal ones to make sure there was adequate support. The supports I used were inexpensive plastic furniture legs, 1" in height, affixed with a nut and bolt.

The inductor leads, too, are soldered or screwed to dual banana plugs. When affixing 4mm sockets for the inductors,

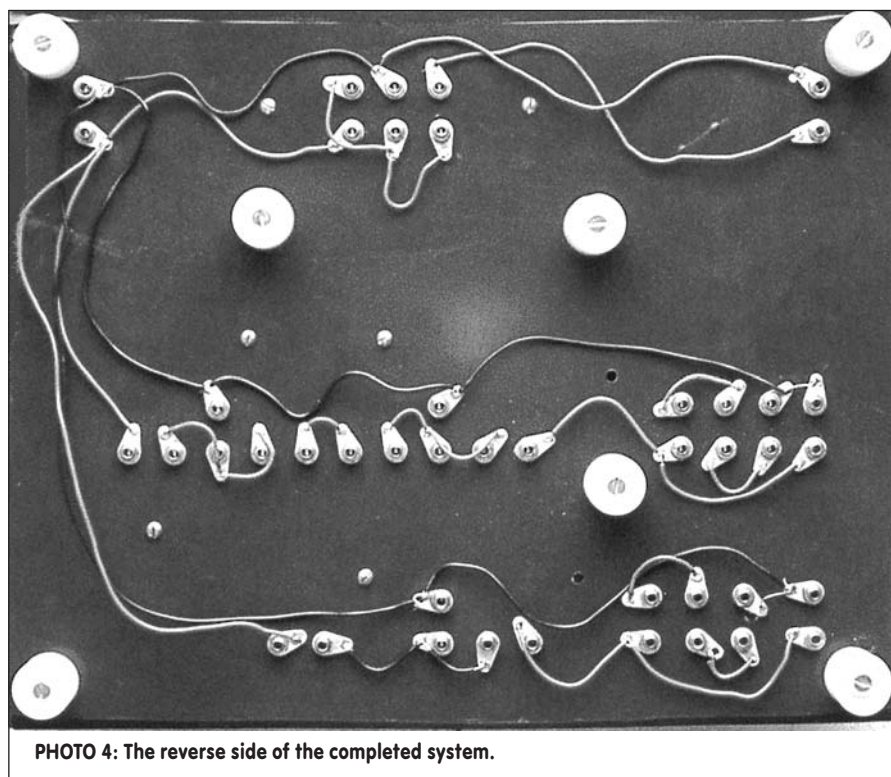


PHOTO 4: The reverse side of the completed system.

note that the inductor will sit on one side of the socket. Make sure you leave enough space around the socket to be able to accommodate the type of inductors you plan to use.

In my system (Photo 3) the inductors are wound on a ferrite drum (bobbin) core, fixed to the board using brass screws. (Always use brass or nylon screws for inductors.) I left a radius of about 1.25" around the inductor center. (Note that you cannot build up inductor values quite as easily by stacking because parallel inductors do not add up—they combine like parallel resistors.)

BUILDING A COMPONENT LIBRARY

Recall that the idea is to produce a permanent and durable COB by screwing/soldering a capacitor to a banana plug. When you build the first crossover using this prototyping system, you will find yourself investing time in building the COBs required for this specific crossover. If you are building a parallel crossover, you can even fine-tune your investment by prototyping the low/mid/high paths separately, reusing the COBs for each frequency range as you go. Once everything seems perfect, you can then build all three paths by building only as many additional COBs as you need

since you have already determined the exact component values.

When you are done with this crossover, you should put these COBs aside for reuse. (In addition to these COBs you can also consider building a few mini-decade boxes that basically consist of a few capacitors connected to a rotary switch. You can use these in parallel with a COB in those rare cases where you need to get an odd capacitor value within a very small tolerance and are not able to do so easily with your COB library.) The next crossover you build will require you to only add incrementally to the set you already have. In this manner, over the space of a few prototyping exercises, you will have built up a set of COBs most suitable for your needs, and which will last you indefinitely. *ax*

PARTS LIST

- 4mm nickel-plated dual banana plug with solder turrets (Pomona 1330-ST or equivalent)
- 4mm nickel-plated dual banana plug with set screws (Pomona XXXX or equivalent)
- 4mm nickel-plated sockets
- Brass/nylon screws
- Phenolic board or hardboard, 12×15 ", 3mm thick
- Suitable supports (generic furniture legs, 1" high)
- Soldering equipment, wire, and so on.

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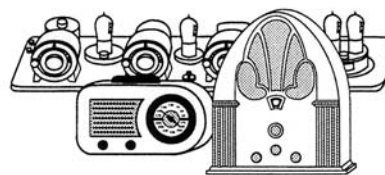
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Terk VR-1 TV Volume Regulator

By Charles Hansen

I grew tired of the jumps in volume when commercials came on our TV. It is especially bad with our local Public Broadcasting affiliate, whose news is aired by a much larger PBS network. Whenever a PBS promo comes on the volume jumps way up. The loud commercials during sports events are another example of dB madness. I saw a reference to the Terk VR-1 TV Volume Regulator a few months ago, and looked into it.

Terk is well known for its vast antenna product line. They are now part of the Audiovox line of equipment. The Terk VR-1 is a small plastic device with an AC power adapter and a cheap set of tin-plated RCA interconnects. It was easy to attach between my cable TV box and the television. You just connect the VR-1 between the cable box or satellite receiver composite audio outputs, and the audio line inputs on the TV. If you have an older TV, you can add a Terk Mini RF modulator to handle the line output from the VR-1 and then connect the composite video signal from the cable or satellite box to the modulator video input.

The spec sheet accompanying the VR-1 also has examples for interconnecting various other system components, such as VCR tape monitor jacks, TV audio outputs to a receiver, and so on.

The VR-1 can accept line level audio signals up to 5Vp-p (about 1.77V RMS or +7.18dBu). While this is not a problem for TV stereo audio signals from a cable or satellite box (the NTSC audio limit is +4dBu or 1.23V RMS), you should not feed audio line sources into the VR-1, which could be as high as 2V RMS (+8.24dBu). Some CD and DVD players have been known to be fractions of a dB higher than the Red Book standard.

There is a bypass switch that sends

the line input signals directly to the outputs without processing.

INSIDE THE VOLUME REGULATOR

Photo 1 shows the rear panel, with the line input and output jacks and the 12V DC power jack. The front of the unit has a green power LED and the bypass switch. The chassis is all plastic with four small round rubber feet, and the ID sticker says it was made in China.

Photo 2 shows the VR-1 with the cover removed. This side of the single-sided PC board contains all of the surface mount components. A 9V linear regulator chip provides all the internal DC power. Each channel uses an 8-pin SOIC IC, two SOT transistors, and a SOT diode as the active devices. The vendor logos and part numbers for all these parts have been obliterated. The unit is entirely electronic and does not use opto-isolator devices for the audio compression/limiting functions. The opposite side of the PC board contains all the radial electrolytic caps and the wire jumpers needed to complete circuits on the surface mount side of the board.

TOPOLOGY

A schematic was not furnished with the unit. I have read opinions on the web stating that the VR-1 uses Plantronics Octiv's Volume Logic/Optimax technology. One of the video chat rooms suggested a visit to octiv.com, but that site only lists software downloads for MP3 and iPod players. Their software

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631-231-7750
www.audiovox.com
Street price: \$36
Warranty: 1 year limited



PHOTO 1: Front View.

is used to provide automatic gain reduction, compression, multiple audio band EQ, and noise reduction.

According to the buzz on the web, the next logical step was to move the Octimax technology to a stand-alone single chip device for consumer/pro audio use. In cooperation with Terk/Audiovox, they reportedly produced an IC for the VR-1. If you want to read all this for yourself, I suggest you Google Terk VR-1. The claims seem to change frequently. *Caveat legens*: let the reader beware.

In order to resolve all these conflicting claims, I took some time to trace the pc layout and make a schematic. Even with-

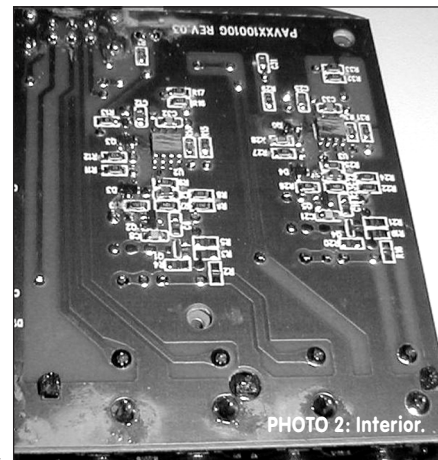


PHOTO 2: Interior.

out a device part number it was obvious that the 8-pin IC was an op amp due to the relationship of the IC pins to the VR-1 input/output jacks. The following description applies to each channel.

The audio signal is input to the non-inverting input of the op amp, whose resistive feedback network is configured with very high gain. In parallel with this is a variable-gain feedback network that uses an NPN emitter-follower circuit and a diode to store the positive peaks of the audio waveform. This peak signal is capacitively coupled to a common emitter amplifier that sinks some portion of the input signal to ground, depending on the amplitude of the output signal. This variable-gain feedback signal has a fast attack and slow release, with a compander (compressor-expander) gain characteristic.

There are no specifications with the VR-1 owner's manual except for the warning not to exceed an input signal level of 5Vp-p. The few specifications that are listed in **Table 3** were gleaned from an Internet site that sells the VR-1¹, but they did prove to be fairly close to my measurements.

MEASUREMENTS

Before removing the VR-1 from my TV for measurements, I held a portable AM radio directly over the plastic case. There was absolutely no buzzing sound

that would indicate the presence of a digital processor clock. The case also has no metallic shielding, which would be required to keep any digital clock noise at bay.

The VR-1 maintains normal polarity. The input impedance measured about 18k, and the output impedance was 823Ω over the audio band. Without any input signal there was no transient noise when turning the unit on or off, or selecting Bypass.

Figure 1 shows the steady-state compander (compressor-expander) performance of the VR-1. The dashed straight line represents unity gain. When the input is less than approximately -25dBu, the VR-1 provides a reasonably linear gain of about 11dBu. The unity gain point occurs with an input signal of 139mV, or -14.9dBu.

When the input is greater than -15dBu, the compressor characteristic takes over and is frequency dependent, as you can see from the way the four curves deviate from each other above that level. The 20Hz curve (bottom) has the greatest amount of compression and becomes flat at an output level of -13.5dBu. At 32Hz the compression ratio is reduced and only flattens at the highest limit of the input range. The 63Hz compression is within 0.5dBu of the 1kHz curve.

The maximum gain at low input levels is 11.6dBu, while the maximum compression at 20Hz is -20.8dBu. This is equal to 32.4dB of what the ref-

erenced website calls "normalization." Normalization is used in digital audio recording to apply a constant amount of gain to an entire recording to bring the highest peak to a level close to 0dB, so I'm not sure whether normalization is an accurate description of the VR-1's companding.

I recorded the frequency response (**Fig. 2**) into a load of 100k with input signal levels from 10mV to 1.77V RMS (-37.8 to +7.18 dBu). The frequency response for the VR-1 with a 500mV (-3.8dBu) input signal was within -1dB from 20Hz to 20kHz. At lower input levels, the HF response fell off more rapidly. Reducing the load to 10k did not appreciably affect the LF response. At higher input levels, the LF response fell off more rapidly. Because the maximum input limit is specified by Terk at 5Vp-p (1.77V RMS), there was nothing to be gained in exploring input levels beyond this, since the VR-1 would not experience even that level in its intended application.

Because the bandwidth is quite wide in the VR-1, wider even than SACD, this is further evidence that there is no analog-digital-analog conversion in play here.

Output hum and noise (input shorted) measured 1.15mV or -57dBu, with no DC offset. Viewing this output signal on my analog oscilloscope showed just broadband noise.

Crosstalk performance with a fixed 500mV input signal and the opposite input shorted showed the response to be the inverse from the usual audio equipment, where parasitic capacitive coupling increases crosstalk with frequency

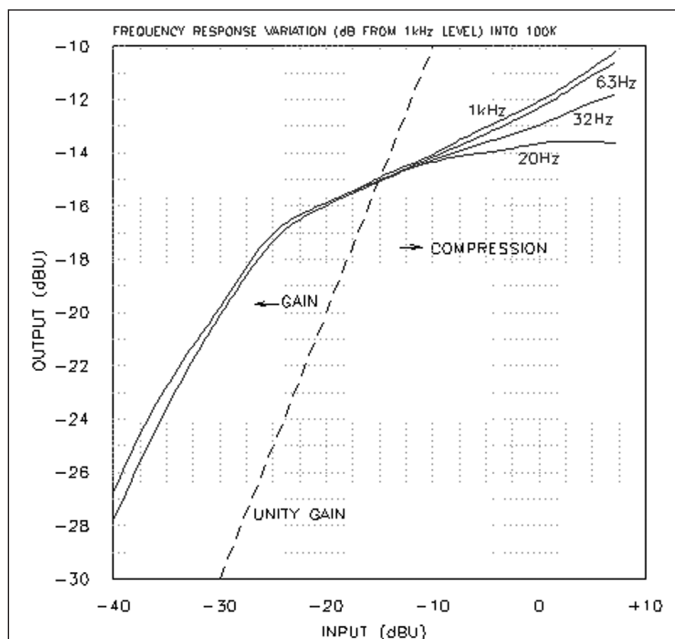


FIGURE 1: Output Compression Ratio vs. Input Level.

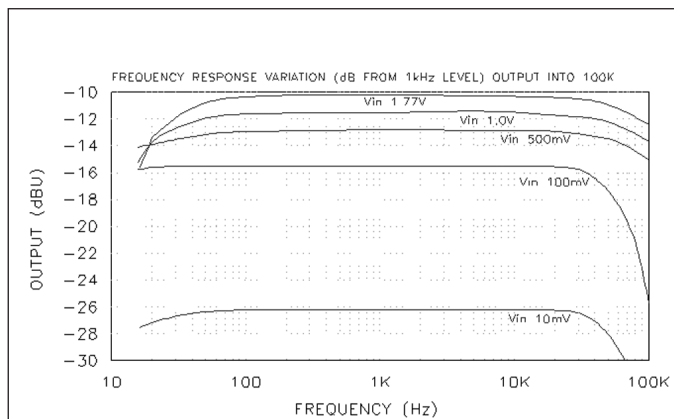


FIGURE 2: Frequency Response.

(Table 1).

Figure 3 shows the output THD+N versus frequency for the VR-1 at various input signal levels. Above 100mV a peak occurs at 60Hz which may be due to insufficient filter capacitance in the 12V DC 100mA plug-in power adapter. Decreasing the load resistance to 10k Ω accentuated the 60Hz peak (not shown).

The graph of output THD+N versus input voltage in **Fig. 4** shows that the area of lowest distortion is quite narrow, and occurs at an input voltage of 325mV. There is a noticeable discontinuity in the drop in THD+N on the left side of the graph that roughly corresponds to the transition from gain to compression at the 139mV unity gain line in **Fig. 1**.

The distortion residual for a 325mV RMS input signal at 1kHz is shown in **Fig. 5**. The upper waveform is the distortion test set sine wave input signal,

TABLE 1: TERK VR-1 CROSSTALK PERFORMANCE.

Frequency	Crosstalk
100Hz	-41 dBu
1kHz	-45 dBu
10kHz	-47 dBu
20kHz	-48 dBu

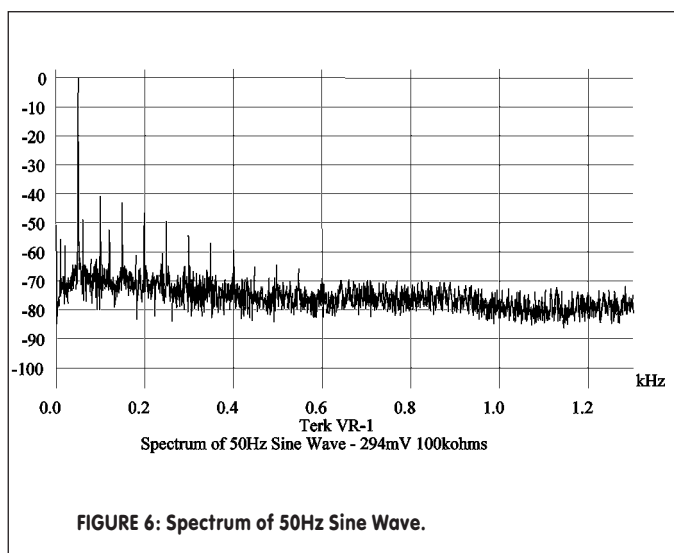
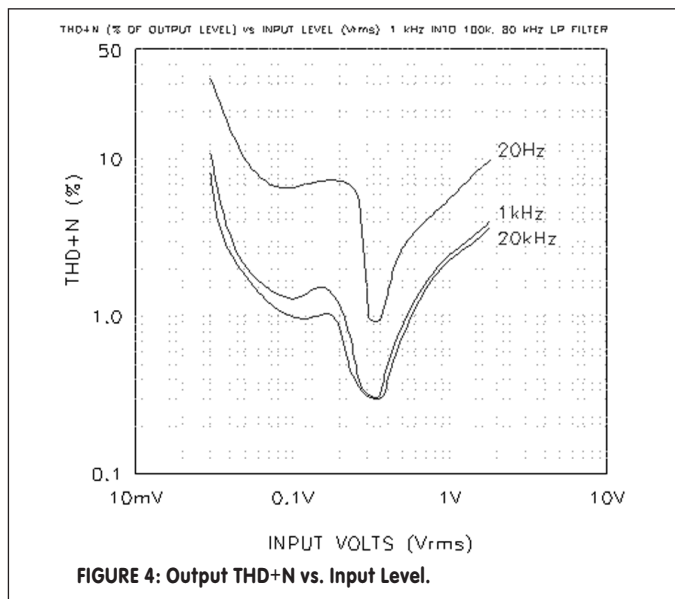
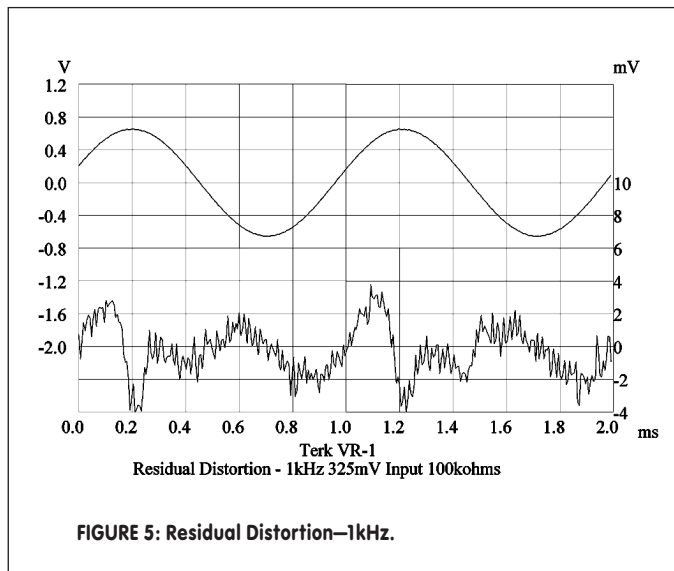
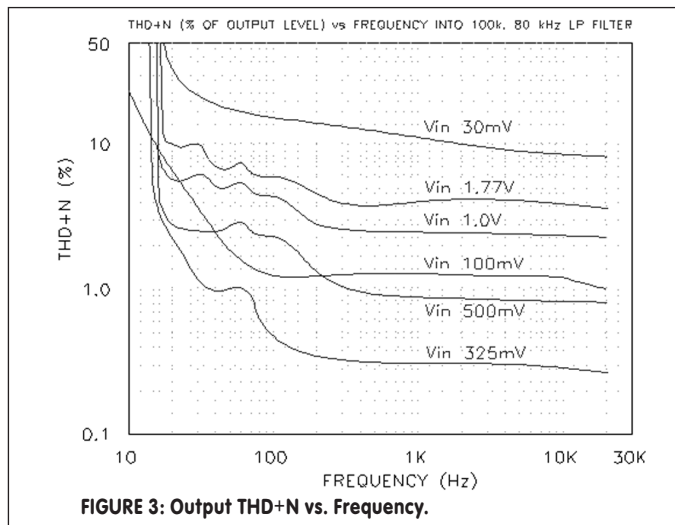
and the lower waveform is the monitor output (after the THD test set notch filter), not to scale. This distortion residual signal shows discernible harmonics along with the noise. The THD+N at this point is 0.31%, corresponding to the minimum distortion point in the curve of **Fig. 4**.

The spectrum of a 50Hz sine wave is shown in **Fig. 6**, from zero to 1.3kHz. I again adjusted the input signal level for the lowest output THD+N. A decreasing series of harmonics is present along with a series of 60Hz power line

harmonics. The 2nd harmonic is -41dB (0.89%) while the 60Hz line is -49dB (0.36%). The measured THD+N is 1.3%, and the THD calculated from the 50Hz harmonics is 1.01%.

I repeated the FFT spectrum analysis with a 1kHz sine wave (**Fig. 7**), with the input signal adjusted to 325mV RMS. The 2nd harmonic is -56dB (0.16%). The measured THD+N is 0.31%, while the THD calculated from the 1kHz harmonics is 0.16%.

Figure 8 shows the VR-1 output spectrum reproducing a combined 11kHz + 12kHz intermodulation distortion (IMD) signal at 2.5Vp-p into 100k Ω . The 1kHz IMD product is -35dB (1.78%) with a veritable picket fence of IMD products throughout the spectrum, due to the intentional non-linearity of the compander circuitry. The signal peaks from 18kHz-20kHz are



particularly puzzling.

SQUARE WAVE TESTS

I viewed the response of the VR-1 to three square wave test frequencies at 2.5Vpp input. The response at 40Hz (Fig. 9) shows a large amount of tilt, compression, and phase shift. The 1kHz square wave in Fig. 10 has just a slight amount of tilt. The best looking of the bunch is the 10kHz square wave in Fig. 11, which

had just a slight bit of rounding on the leading edge, with no hint of peaking or ringing.

Dynamic Response to Tone Burst and Step Transients

I performed all the foregoing tests with steady-state test signals, and used dynamic test signals for the next series of tests.

Figure 12 shows the response to a 1.77V RMS 20Hz sine wave tone burst.

The initial attack is quite sharp, with a slow release. There is also noticeable phase shift, which I would not expect to see if the VR-1 used a digital filter in the signal chain. At 100Hz (Fig. 13) the attack is not as sharp, but there is a noticeable difference between the positive and negative half-cycles, with slewing on the negative side of the output waveform. Figures 14–16 are tone bursts of 400Hz, 1kHz, and 10kHz, all at 1.77V RMS.

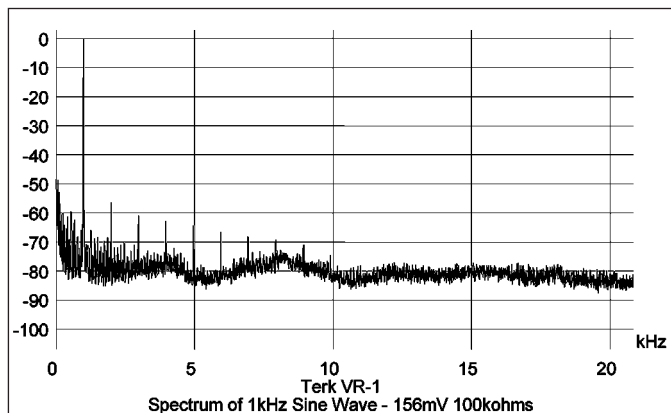


FIGURE 7: Spectrum of 1kHz Sine Wave.

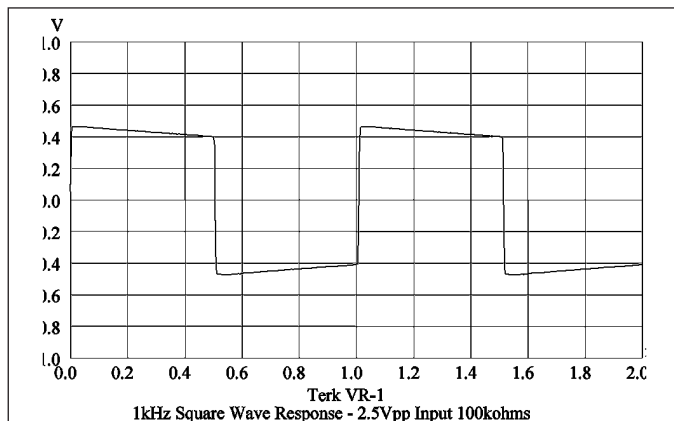


FIGURE 10: 1kHz Square Wave Response.

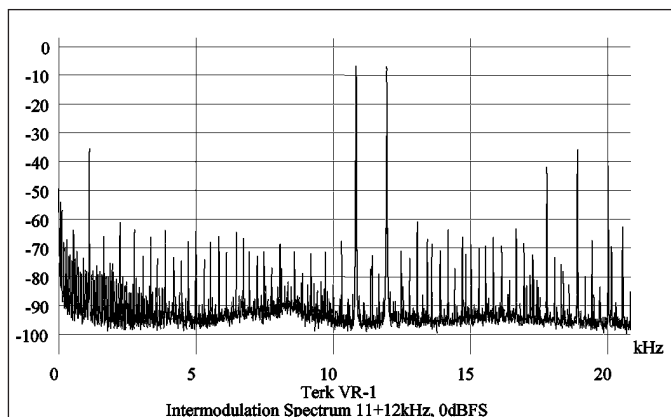


FIGURE 8: Spectrum of 11kHz + 12kHz Intermodulation Response.

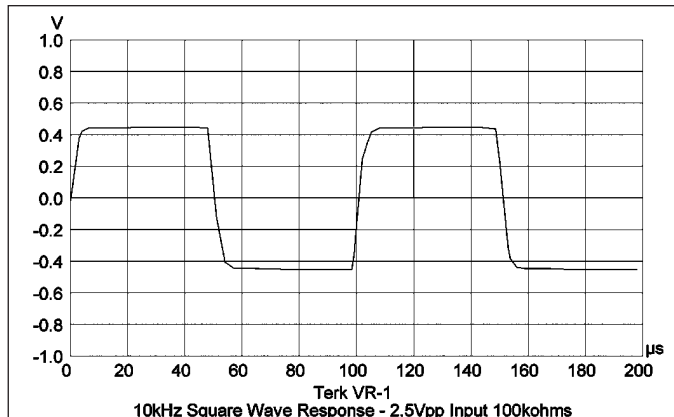


FIGURE 11: 10kHz Square Wave Response.

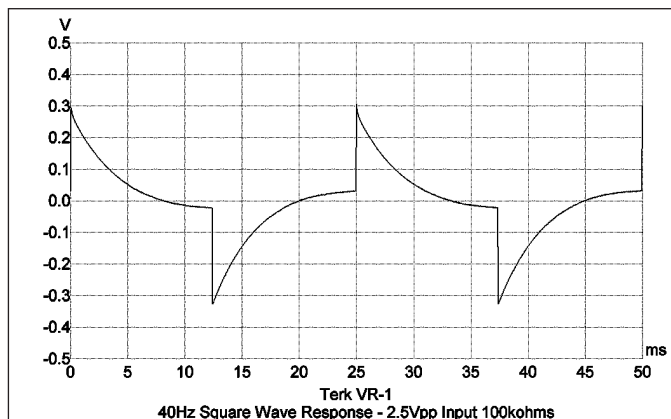


FIGURE 9: 40Hz Square Wave Response.

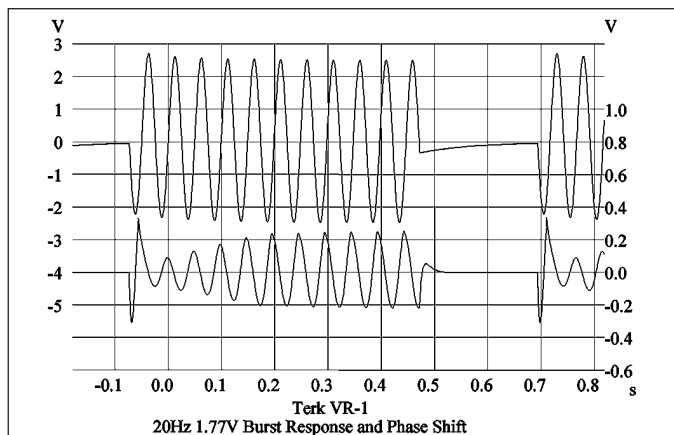


FIGURE 12: 20Hz 1.77V RMS Tone Burst Response.

Table 2 shows the percentage to which each of the tone burst peak-peak voltages was reduced from the peak of the initial attack to the following peak. You would expect the percentage of reduction to decrease with frequency in accordance with **Fig. 1**, but the distortion level is very high at the beginning of output waveform compression. Given this high level of distortion, the crest factor would introduce considerable error when converting the peak-peak voltages in the scope traces to RMS.

What the tone bursts do show is that it takes about 20msec for the VR-1 to make its maximum reduction from the start of the tone burst. **Figure 16** shows only 1.5msec of the total burst, but when I increased the time/division (not shown), the delay to the minimum peak-peak level was also 20msec. The release time is much slower. **Figure 12** shows that it takes about 200msec from the initial attack to the final steady-state level. This initial attack-release is quite audible when the cable box and TV are first turned on.

Once the circuit adapts to the average output signal amplitude, it is only

TABLE 2: PERCENT COMPRESSION IN INITIAL RESPONSE LEVEL.

20Hz	26% of initial in less than 50msec
100Hz	85% of initial in 20msec
400Hz	83% of initial in 20msec
1kHz	70% of initial in 20msec
10kHz	68% of initial in 20msec

TABLE 3: COMPARISON OF "SPECIFICATION" VS. MEASURED RESULTS.

Parameter	Rating (see Ref. 1)	Measured Results
Input Impedance	20k Ω	18k Ω
Input Signal Range	0 to 5Vp-p	Did not test
Frequency Response	20Hz-20kHz	20Hz-20kHz +0/-1dB
Output Impedance	600 Ω	832 Ω
Output Signal Range	0 to 2.8Vp-p	239mV RMS (676mVp-p)
Normalization	30dB	32.4dB

audible when there are sharp impulses, such as words that start with "p" or "t." The resulting pop produces a transient in the bass frequencies that causes the VR-1 to produce its maximum fast attack and slow release envelope. This causes a snapping sound followed by an overall reduction in volume until the completion of the release delay.

Figure 17 shows the 1kHz tone bursts at a reduced level of 500mV. Here the peak-peak level is attenuated to 81% of

the initial value as compared to 70% for the 1.77V RMS signal, again in 20msec.

My final test of the VR-1 transient performance was to make step inputs of +2.5V DC (**Fig. 18**), then -2.5V DC (**Fig. 19**). The input step is at the top, and the output response is at the bottom.

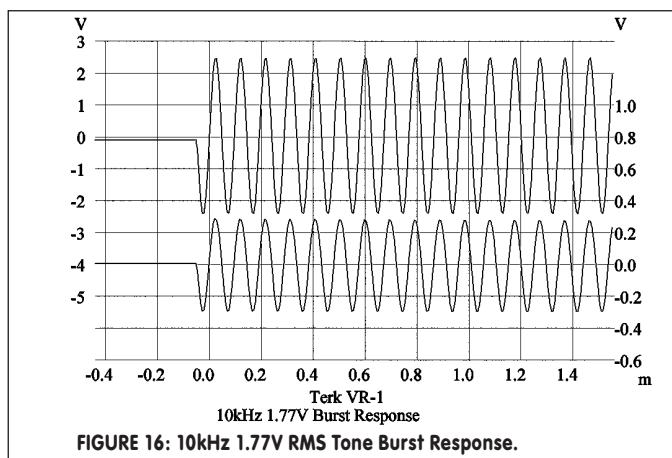
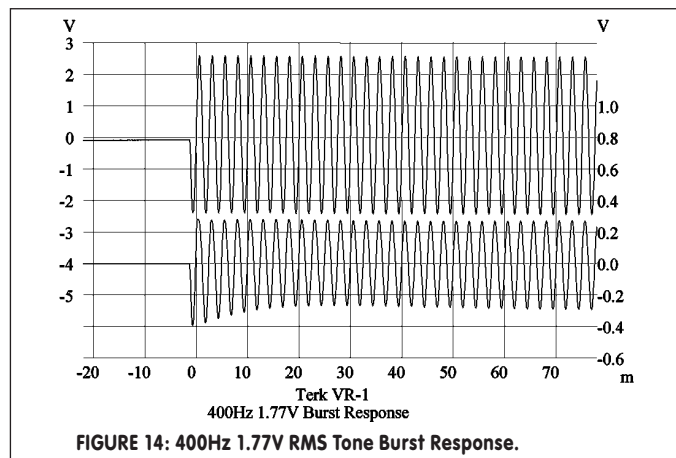
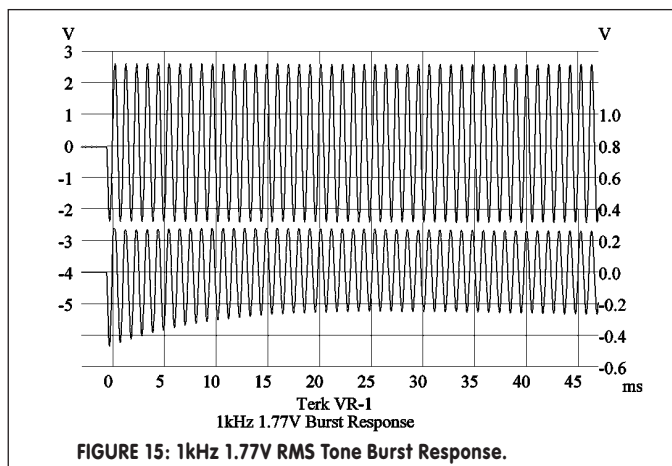
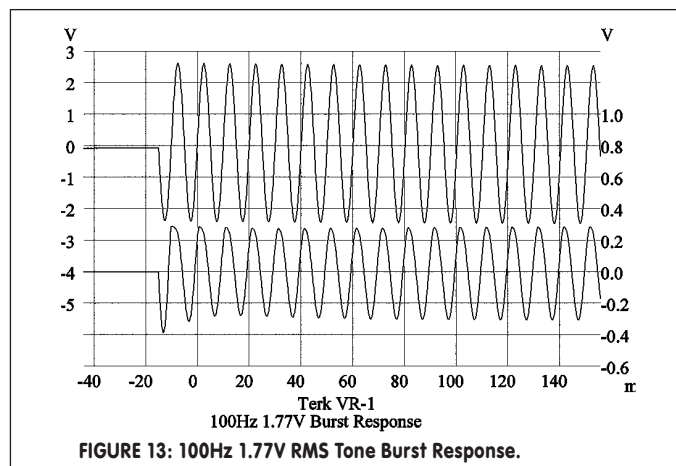
In a capacitively-coupled circuit, the time constant $\tau = RC$, where τ represents the time it takes the step response to get to 63.2% of its final value. The time constant for the output response envelope for both the positive and negative step inputs is 2.3msec.

Note the short burst after the initial output peak. **Figure 20** is an expanded plot of the initial +2.5V step response. This burst has a frequency of about 70kHz and lasts for about 100 μ sec.

A comparison chart of "specified" data versus measured results is shown in **Table 3**.

LISTENING RESULTS

So how does the VR-1 Sound? I connected the VR-1 between my preamp and power amp, making sure to set the preamp volume control low enough so as



not to exceed the 5Vp-p input limit. The VR-1 closely approximates the heavily compressed audio you would hear on an FM rock station, but with a bit more distortion. However, the VR-1 is not intended to be used for this purpose. If I want to watch concerts or the symphony on TV, I use the Bypass switch. Otherwise, the added distortion will be evident on most music.

I now have a few months' experience with it as a TV volume regulator, and it does indeed fulfill its role in leveling the differences between loud and soft audio passages.

The only annoyance with the VR-1 is the snapping noise that occurs with words that begin with "p" or "t," as I described. The initial attack of the VR-1 is quite aggressive, and many TV stations use compressors and/or limiters to stay below the maximum allowable audio signal modulation. When you place a compander (VR-1) in series with a compressor/limiter, the results can be audible. **aX**

REFERENCE

1. www.chumbo.com/Info.aspx?id=252665

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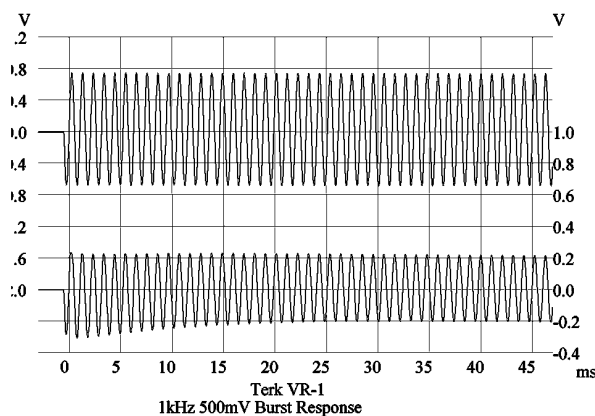


FIGURE 17: 1kHz 500mV RMS Tone Burst Response.

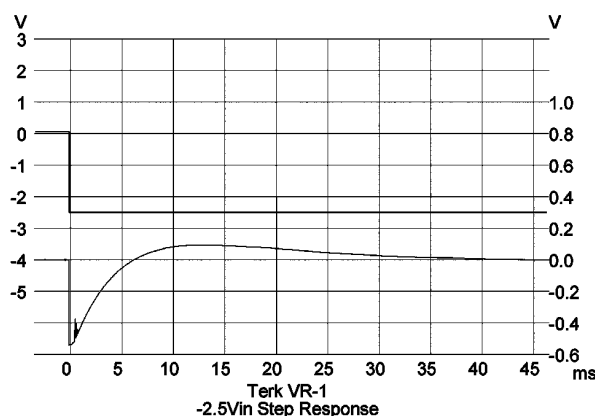


FIGURE 19: -2.5V Step Response.

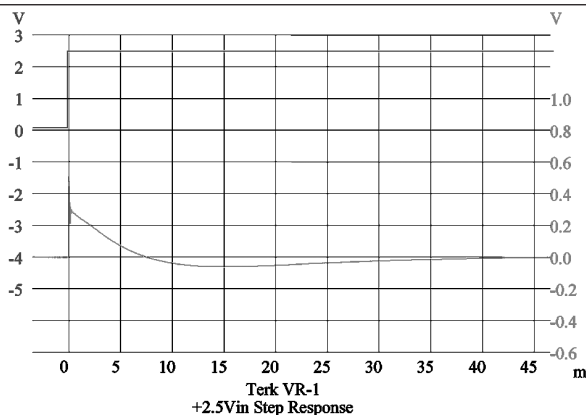


FIGURE 18: +2.5V Step Response.

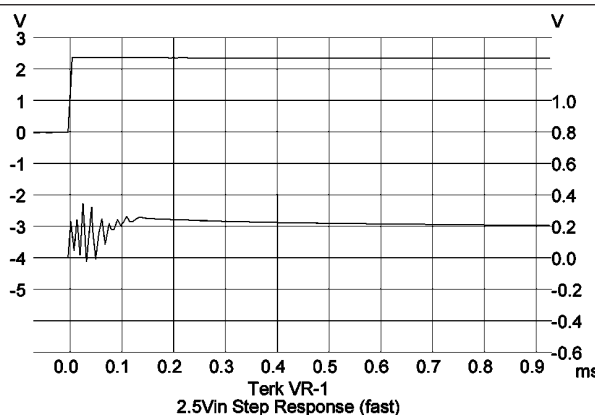


FIGURE 20: +2.5V Step Response Expanded.

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PNF Audio Cables

By David Milford

Do your tastes run to turquoise, orange, purple, or gaudy yellow-green? PNF Audio can customize the braiding on their cables to match your desires. I approve of non-drab audio gear! Normally, cable companies will either assign a color to a particular model or performance level or have a common color throughout the line (black or gray usually). Interesting. The question is do they glow in ultraviolet light?

FEATURES

Most of PNF's cables utilize CC (continuous cast) OFHC (oxygen-free high conductivity) copper. PNF touts the fact that the CC process eliminates most of the crystal barriers found in standard OFHC copper. Apparently, as many as tens of thousands of barriers per meter exist in normal cold-cast samples. Their hot-cast CC copper reportedly has less than one barrier per meter. I have heard of this method of producing copper before, but have not listened to any in my own system until now.

They further assert that capacitance, inductance, and resistance are noticeably affected by the multitude of crystal barriers in standard copper. It makes sense from a scientific standpoint that crystal barriers might be bad for signal

propagation. It is certainly an inhomogeneity that could have unexpected and asymmetrical effects on the purity of the music. Does the listening give evidence of the theoretical benefits? Because I did not have an identical but non-continuously cast set of cables for comparison, I can't say for sure. But I did find them to have some very nice qualities.

The Icon interconnects reviewed here are their top-shelf offerings. At \$199.95 per meter pair, you could buy a few and outfit your entire system for what you would spend on a single pair of some other company's flagship offerings. They use CC process, 23awg OFHC, silver-plated copper for the conductors. The company further states that the skin effect penetrates the entire diameter of the conductor at audio frequencies.

PNF claims that the dielectric has the ability to absorb the electrostatic energy stored in the conductor (electrostatic energy in music? I thought nothing was static in music?). To absorb as little of the energy as possible, PNF uses what they call Airlon dielectric material, which is Teflon (PTFE) foamed into a honeycomb matrix with air pockets. They state the Airlon achieves a 1.4 dielectric constant, the lowest currently possible with a solid material. For comparison, unfoamed Teflon is 2.1 (praised

for its sonic performance), and a perfect vacuum (the ideal) is 1.0.

PNF uses a solid foil 100% coverage shield in the Icon, to reject external EMF and RF noise. A PVC jacket lends the outside of the package toughness and flexibility. The connectors are a high copper content alloy, locking RCA with a conductivity rating of 93.3%.

The PNF Audio Symphony speaker cable uses CC copper OFHC conductors and a custom polyethylene dielectric. The spiral geometry and proprietary burn-in process all work together to produce a very neutral cable, according to the company. They come with either gold-plated spades or gold-plated banana plugs, depending on your needs. They are both crimped and "silver soldered." This termination method should be robust enough for plenty of amp or speaker swapping. There are four 12-gauge conductors per bundle, yielding approximately 10-gauge overall.

PNF offers a 30-day no-questions refund. Quite a convenient way to try things out in your own home when buying from their website. All the cables also come with a one-year parts and labor warranty.

LISTENING TESTS

When I commenced listening—I must

PNF Audio

2598 Tuckahoe Rd.

Franklinville, NJ 08322

1-877-57AUDIO www.pnfaudio.com

Icon interconnects: \$199.95 1m pair with RCA.

Symphony speaker cable: \$314.95 8' pair with spades.

Associated equipment: Rotel RDD-980 transport, Assemblage DAC-3 D/A, home-built STAMINA Zen/Bride of Zen integrated amp monoblocks, Muse 150 power amps, B&W Matrix 804 speakers.



be honest—I was immediately taken by the detail and natural quality of the sound. My favorite interconnects right now are the Kimber Hero and Kimber Silver Streak, both affordable over-achievers. The Hero is a better balance for my system frequency-wise, and the PNF Icon came very close to it in that department. The Icon was actually a touch livelier on the top, but somehow it all came forward as extra musical information and not anything unwanted. I tried not to pay close attention until all the cables had been broken in for 50 hours or so, but it was hard. The music was so alive it drew me in.

With Incubus's *Morning View* (Epic EK 85227) all the information was there, without slapping you in the face with it. The exciting distorted guitars made my temperature rise. The melodic singing on "Wish You Were Here" brought back to me why this song is one of my favorites. I was a bit surprised the Icon lived up to my expectations. They disappointed in no department, no matter how hard I searched for sonic flaws. My ultra-simple amps are more revealing than most, a picky person's dream (or sometimes nightmare). The Icon left me wanting nothing. And, more important, there were little or no additive distortions that called attention to themselves.

I also tried a pair of balanced Icon interconnects with XLR connectors, running fully balanced from D/A to Muse 150 power amps. With no volume control, the only things I could listen to were very quiet recordings, such as *A Meeting By The River* (Water Lily WLA-CS-29-CD). My nature trip continued at a volume as high as I would want to listen.

The Icon spoke with ample liquidity to do credit to this phenomenal recording. I could feel the tension on the heads of the tablas, hear the subtlest resonances of the guitars, and even get a feel for the space the musicians were playing in. Really very fine balance to everything.

I had balanced Kimber Hero on hand and Straight Wire Maestro for comparisons. The Icon won out over the Maestro in many categories. Between the Icon and the Hero, however, the choice was ambiguous—a good thing. Sometimes I thought I preferred one or the other in

a particular area, then my choice would flip-flop. The Icon had maybe a touch more detail, but then the Hero was the champion when it came to warmth.

Then I switched back to the STAMINA amps, using both Icon interconnects and also an 8' pair of Symphony speaker cable. With an all-PNF cabled system (except digital), the dynamics were impressive. It was just as I like it—all the detail with no harshness. To be fair, the top octave was just a touch more noticeable than with the high capacitance Goertz MI-2 T-series that are my favorites. It was a slight difference in frequency balance, and in many systems I think the Symphony would be preferable.

I put on Simple Minds' *Live In The City Of Light* (A&M CD 6850) for some very familiar material. The natural reverb bouncing off the Paris buildings in this concert bled back through the mikes to produce a wonderful sense of spaciousness. It is not a flawless recording, yet there are great depths of detail to explore. I was nothing short of enthralled while hearing this served up via PNF cabling. The quantity of detail was almost staggering. Even the crowd noise showed off the realism these cables are capable of.

As with the interconnects, the top octave was something special. Though perhaps a tiny bit too much for my system, which is already over-gifted in this department. In many systems, the PNF's top octave will be a welcome blessing.

What of bass and midrange, you might ask? Well, these were strongly delivered as well. Bass had a pleasing roundness when provided by the recording. Voices had plenty of in-room intimacy, evidence of midrange quality. Wooden piano soundboards seemed as large as they are in real life. Melodic lines drew in the listener with as much importance as they deserved.

I very much enjoyed my time with the PNF CC cables. I recommend both the Icon interconnects and the Symphony speaker cable. They manage to deliver impressive detail as well as smoothness at this price point. In most areas, they compete well with my other favorite cables. They brought life and excitement to any and all types of music that I threw at them. **AX**

CONTRIBUTORS

Rudy Godmaire ("A Flexible Subwoofer Amp Pt. 3," p. 6), a sales consultant for Bell Canada, has been interested in DIY audio since 1998.

Simon Brown ("Amplifier War and Peace," p. 14) is a Professional Engineer and has worked in Product Design in New Zealand and England. Simon has long been involved in DIY HiFi, having built speakers, amplifiers, and turntables. This led him to set up Design Build Listen Ltd. to supply DIY HiFi components. Simon has had over a dozen articles published in HiFi magazines. Contact him via www.designbuildlisten.com.

Ed Simon ("Speech Intelligibility, Part 1," p. 24) received his B.S.E.E. at Carnegie-Mellon University, and has installed over 500 sound systems at venues including Jacob's Field, Cleveland, Ohio; Museum of Modern Art Restaurants, New York; The Forum, Los Angeles; and Fisher Cats Stadium, Manchester, N.H.

Dennis Colin ("An Audio Isolation Box" p. 30) has demonstrated the audibility of phase distortion at Boston Audio Society, and has designed the "Omni-Focus" speaker (bipolar coincidental with phase-linear first-order crossover), ARP 2600 analog music synthesizer, 1kW biamp and PWM supply at A/D/S, and Class D amps.

Bjorn Johannesen ("TABAQ: Tang Band Quarter Wave," p. 38) is an avid DIY hobbyist involved in building amplifiers and loudspeakers. His speaker designs have included horns, full-range, and transmission lines. The interest in quarterwave designs has accelerated in the last few years, thanks to audioXpress, www.t-linespeakers.org, and www.quarter-wave.com. Bjorn lives in Denmark, just outside Copenhagen. His e-mail address is bjohannesen@post.cybercity.dk.

Ramkumar Ramaswamy ("A Prototyping System for Passive Crossovers," p. 42) is an information technology professional. He owns Audesine (www.audesine.com), a firm that provides loud-speaker design services. He is a doctoral Fellow of the Indian Institute of Management Calcutta and is a member of the AES.

Charles Hansen (Product Review: Terk VR-1 TV Volume Regulator, p. 46) holds five patents in his field of engineering and enjoys modifying guitar amplifiers and effects to reduce noise and distortion. With hundreds of articles to his name, he is a frequent collaborator to the magazines of Audio Amateur Publications.

David Milford (Product Review: PNF Audio Cables, p. 54) received a bachelor's in physics from Oberlin College. He was a contributing editor with *Stereophile* magazine for three years. He has enjoyed modifying all sorts of audio equipment.

Vance Dickason ("Special Report: Audio in the Airport," p. 62) is the Editor of *Voice Coil*.

CORRECTION

Equation 4 in my article "A Flexible Subwoofer Amplifier" (Jan. '07, p. 37) contains a couple of typos. The correct equation should read:

$$\text{Phase (rad)} = \tan^{-1} \left[\frac{\frac{2\omega}{R_1 C_1}}{\omega^2 - \left[\frac{1}{R_1 C_1} \right]^2} \right]$$

Rudy Godmaire
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FLEXIBLE SUBWOOFER

I enjoyed Rudy Godmaire's article in the January issue, but have a few comments. There are several instances in which polarized capacitor symbols are used, but I'm sure Mr. Godmaire intended to show film capacitors:

Figure 2: C29, C31

Figure 6: C3, C4, C5

Figure 10: C17, C21

There is no parts list in the article to verify the capacitor types he used and, unfortunately, tantalum polarized capacitors are available in some of these values. They would be most unsuitable for the filter frequency determining capacitors.

I am concerned with his method of powering LED D6 through the two series 24V zener diodes ZD5 and ZD6 without any series resistance. The dynamic impedance of a 24V 5W zener (such as the 1N5359) is only 3.5Ω. With 5% tolerance zeners, the breakdown voltage could be as low as 45.6V. Any surge in the +48 supply could deliver high damaging current to the LED.

Another concern is the use of pots P4 and P5 to set the DC rail voltages for the op amps (Fig. 11). The most likely failure mode for any potentiometer is an open wiper. When (not if) this occurs, one of the V3 voltages would increase from 8V to 13.8V. It is much better in

terms of noise and reliability to use 1% fixed metal film resistors to set the rail voltages. Op amps are relatively immune to small differences in the rail voltages, but they may not appreciate a 73% difference.

I'm not certain why he used a supply voltage of only ±8V for the OPA-2604 op amps, and he didn't elaborate on his choice in the article. The OPA-2604 has a maximum supply limit of ±24V, and is safely usable to ±21V. I've seen articles that state some op amps sound their best when used at higher rail voltages, but this may be op amp specific. I don't think there is any downside to using the traditional ±15V rail voltages. He could raise the input voltage to the linear regulators to ±18V, there would be less power dissipated in IC4 and IC5, and he could reduce the dissipation in the series zener voltage string ZD1 to ZD4. The remaining dissipation resulting from stepping the original ±48V down to ±15V would be spread across all the op amps, which are well below their maximum rated dissipation.

While Mr. Godmaire had a space limitation in his amplifier, if the reader has more room, he/she could use a small separate 25.2 VCT "filament" transformer to provide the DC input to IC4 and IC5. This would eliminate the power dissipation in the zener diodes and provide a lower voltage for the LED indicator series resistor. The entire amplifier would benefit from the reduced power dissipation.

Finally, I am firmly in the camp of Gary Galo when it comes to adding protection diodes to linear regulator ICs (see "Preamp Power Supply," *TAA* 4/90, p. 47). Mr. Godmaire did not violate any by-the-book rules in his linear regulator supply application, but five more 5-cent 1N4002 protection diodes are cheap insurance, and as Gary says, their use is not negotiable.

This is a well thought-out article, and I certainly look forward to Part 2 of Mr.

Godmaire's article.

Chuck Hansen
Orange, N.J.

Rudy Godmaire responds:

I would like to thank Mr. Hansen for his interest in my article and for taking the time to share his concerns. I feel privileged to receive comments from such an authority.

Regarding Mr. Hansen's first concern about the potential misuse of polarized capacitor symbols, in Fig. 2 C29/C31 are film capacitors and the symbols appearing in the schematic seem to be appropriate. As I mentioned at the end of the article, all op amps have been bypassed with 10μF/16V electrolytic capacitors and 0.01μF/63V polypropylene capacitors (pins 4 and 8 in the schematics). The parts lists will appear in Parts 2 and 3 as support information for the construction of the amplifier.

C3, C4, and C5 are electrolytic capacitors. I assumed the circuit could allow for the use of these types of shunt capacitors. Measurements in Part 4 tend to confirm that. Having said that, there is no doubt Mr. Hansen's suggestion will improve the performance of the filter, and I will take the opportunity to replace these electrolytic capacitors (Black Gate) with metallized film capacitors.

For now, readers may already consider replacing the parts shown in Table 1 of Part 2 with the following metallized film capacitors:

C3: 0.22μF/50V	Digi-Key Part No.: P4667-ND
C4: 0.33μF/50V	Digi-Key Part No.: P4669-ND
C5: 0.47μF/50V	Digi-Key Part No.: P4671-ND

As a better alternative, you could replace C3 and C4 with a single 0.56μF/50V capacitor (Digi-Key Part No.: P4672-ND). Note that I chose this model of metallized film capacitors mainly for its small size. Other brands or models will do.

While Fig. 10 shows C17 and C21 as electrolytic capacitors, they are, in fact, polypropylene capacitors. This detail es-

caped my attention while proofreading the article, and I thank Mr. Hansen for pointing out this mistake.

Though Mr. Hansen is concerned with my method of powering LED D6, I can report that I have never had any problems so far (two years). If I may build on his suggestion, LED D6 might simply be connected through a series resistor to V3+, without the need for any zener diode.

I am surprised to read that Mr. Hansen is concerned about pots P4 and P5. The voltage regulator datasheets of Linear Technology suggest, in many schematics, the use of a potentiometer as a means to adjust the output voltage. I have selected quality cermet potentiometers for this task and I am confident they will provide reliable operation for years to come. Nevertheless, Mr. Hansen's intervention is certainly guided by his longstanding experience and expertise. I will retain his advice for future projects.

Like Mr. Hansen, I also heard about the benefits of raising the supply voltage of op amps. Unfortunately, this was not the case in my application. The supply voltage of $\pm 8\text{V}$ provided me with the lowest level of hum and noise (H&N). As mentioned in Part 2, I measured 1.0mV of H&N at the binding posts with the volume control at full level.

As an example, increasing the supply voltage of the op amps to $\pm 10\text{V}$ increased the H&N to 1.7V. Further increase of the supply voltage would worsen the situation. Not being an expert, I can only speculate on the reasons for this situation. My best guess is that it might be related to the design of my PCBs as discussed in Part 2.

Finally, I am glad to know I did not violate any by-the-book rules in my linear regulator supply application. The circuit was designed with full respect to Linear Technology's recommendations described in their datasheets. Though I did not read the article Mr. Galo published some time ago, I have already prepared a revised version of PCB# 4 that takes his sound advice into consideration. Readers may communicate with me through my website (www3.sympatico.ca/r.godmaire) should they wish to get a copy of the revised templates.

I hope Mr. Hansen will enjoy the rest of the article. I invite him to keep on expressing his concerns, or appreciation, for the benefit of all readers, including myself.

MUST-SEE SEMINARS

I just finished reading the article

"Grounding and System Interfacing" by Gary Galo in the January 2007 issue of *audioXpress*. It was a great distillation of many of the most important factors that affect noise in sound systems. I was especially interested to see the new balanced ICs available from THAT Corporation. Nice job, Gary!

In 1999, the Chicago section of the AES hosted a seminar by Neil Muncy titled "Grounding, Shielding, and Things that Go ZAP! in Audio Signal Processing Systems." I was fortunate to attend, and to say it was an eye—and ear—opener is an understatement. Through a series of visuals and live demonstrations, Muncy led the attendees through the various aspects of the power distribution system and also the correct and incorrect ways to handle noise suppression. At one point in the demo, he showed how effective proper techniques could be by running a line level signal about 200' through the room including over fluorescent lights, video monitors, and other noise-generating devices with virtually no noise, even though the various pieces of equipment in the test were powered

from different lines within the building. Truly amazing!

The one thing I did not see mentioned in Gary's article was the pin 1 problem. Even though devices may be connected using balanced lines, if the termination of the shielding connected to pin 1 is not done properly, noise can creep in. Connection of pin 1 to the chassis as close to the entry point as physically possible is the proper choice. A bad choice is to connect pin 1 to signal common even if it ultimately is connected to the chassis. I hope by now most commercial equipment will be connected properly, but if you are doing your own projects, this is important to remember.

If any reader ever gets the chance to attend seminars by either Mr. Whitlock or Mr. Muncy, they should run—not walk—as fast as they can for an experience not to be forgotten and immensely valuable.

Tom Perazella

tom.perazella@calumetphoto.com



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Gary Galo responds:

Mr. Perazella is correct in pointing out the "Pin 1 Problem" (a term coined by Neil Muncy). Bill Whitlock did discuss this problem in his presentation, and readers should refer to the .PDF documents that I mentioned on pages 29 and 30 of *aX* 1/07 for detailed information on this issue. Essentially, pin 1 problems occur when star grounding has not been observed in equipment design and layout. On page 120 of Whitlock's Power Point presentation, he illustrates both incorrect and correct methods for connecting pin 1 on XLR input and output connectors inside the equipment chassis. Incorrect methods include various daisy-chain ground schemes and others that result in multiple paths back to chassis and/or power supply ground. Correct methods bring all XLR input and output grounds to a single "star" chassis ground near the power supply.

Mr. Perazella is correct: No one who works in audio should miss the opportunity to attend a presentation by Bill Whitlock or Neil Muncy.

HUM PROBLEM

In constructing amplifier circuits—namely, preamps—the most frustrating thing is hum. A poor design, or perhaps a simple construction error, can result in a circuit that hums like a buzz saw.

The first thing you tend to blame is a faulty ground. If that's OK, then you suspect inadequate filtering in the power supply.

You eventually get to the point where you've checked every possible culprit—except for one, which is most often overlooked—the power transformer.

Because of the principles in which it functions, the power transformer behaves like an electromagnet whose field reverses 60 times per second, as does the power being applied to its primary. In turn, the chassis on which it is mounted becomes magnetized as well. The magnetized chassis won't attract a tack or nail—it's not that strong. It is, however, a strong enough field to be easily picked up by the preamp tube, usually a 12AX7.

This was proven when I disconnected the power supply of one of my "buzz saw" preamps, and obtained the heater and plate voltage from a second preamp, which also had a horrible hum. The result was a loud hiss—not a lick of hum whatsoever.

I've tried newer hi-tech power transformers and still had hum. I've also tried plastic spacers between the transformer and the chassis, and that didn't help. The preamp tube and the power transformer are like two guys, each from a different street gang that cause trouble when they come near each other. Separate the two, and the result is peace and quiet.

Through experimenting I've found that when using an outboard power supply (no flowing AC or magnetic fields near the preamp tube), even the simplest power supply will do just fine. No need to fool with exotic capacitors and regulators. With the power transformer away from the preamp, construction is no longer critical; it still sounds great.

I thought Joseph Norwood Still's preamp (*aX* 3/01 p. 36) would put a dent in my theory here. With the power transformers (three in all!) mounted on the preamp chassis, he reported no noise, hiss, hum, or whatever with the volume at $\frac{3}{4}$ rotation. Then I remembered that when a turntable is off, the cartridge is shorted out (as is the preamp input as well). Any preamp will be quiet when its input is shorted to ground. An externally powered preamp is quiet with the turntable connected and switched on.

I hope this helps solve at least one DIYer's hum problem.

Neal A. Haight
Castro Valley, Calif.

DOPPLER DISTORTION AGAIN

While perusing my *aX* library, I noticed a letter response by G.R. Koonce presenting a good case for the audibility of speaker Doppler (FM) distortion (*aX* Aug. '02, pp. 60-62). My enclosed *Fig. 1* shows the derivation of his equation. But I propose an additional factor, the increased audibility and objectionability with increasing LF modulating frequency (f_1), whereas the actual distortion sidebands are independent of f_1 . Three examples show the rationale:

1. A singer's or instrument's vibrato (FM) rate sounds pleasant around 5-7Hz; faster rates sound "choppy" (as many inexperienced singers have yet to learn).
2. If a voltage-controlled oscillator (vco) is frequency modulated at rates above 20Hz, the sound is very discordant.
3. An extreme example is shown in the

data row in the *Fig. 1* table with $f_1 = 0.1\text{Hz}$, $A_1 = 0.5''$ peak, and $f_2 = 300\text{Hz}$. Unweighted Doppler distortion is 4.9%. But consider that the speaker cone's f_1 takes five seconds to move forward, and five seconds to move back. Intuitively this would be no more audible (disregarding motor/suspension nonlinearity) than moving your head forward 1" in ten seconds!

The reason is that, while there is 4.9% FM "distortion," the (predominant) 1st order sidebands (at -29dB) are only 0.1Hz away from the 300Hz input. And the peak frequency deviation Δf_{zp} is only 0.007Hz, a change of only 0.0004 semitones, which is certainly inaudible!

Thus I propose the factor "S" in *Fig. 1*, standing for "subjective Doppler distortion." "S" is a weighting factor equal to the ratio of f_1 to 20Hz. 20Hz is arbitrary; it's the proportionality to f_1 that's significant.

Of practical significance, note the example that Mr. Koonce presented (the first row in *Fig. 1*). As he said, the (unweighted) 33% Doppler distortion should certainly be audible. Paul Klipsch demonstrated the Doppler effect with an "unnamed" (the Bose 901) system using nine 4" drivers. It should be easy to see on a spectrum analyzer, and you can rule out ordinary harmonic/IM distortion by choosing a very linear driver and viewing the 50Hz signal alone.

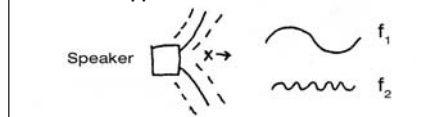
I agree with Mr. Koonce's observation of the superior midrange clarity (at high power) of a three-way versus two-way: I reviewed the excellent Swans M1 by itself (*SB* 3/99) and crossed over at 220Hz to an 8" Scan-Speak 21W/8555-00 sealed sub (*aX* Sept. '05). With sub, it sounds clean even with my 125W per channel "Mad Katy" tube amp (*aX* June/July '06) at full power; the midrange was "muddy" before installing the sub. Likewise for my three-way "Venue" speaker (*aX* Nov. '06); very clean midrange. A notable two-way exception is Joe D'Appolito's "Thor" (May '02), which uses two 18cm (7") mid/woofers with enough cone area plus TL loading to keep the bass excursion low enough to avoid audible Doppler (or any other) midrange distortion.

I'm sorry for being four years late in appreciating Mr. Koonce's letter,

but I look forward to his thoughts on my thoughts regarding Mr. Doppler's thoughts!

Dennis Colin
dcolin@worldpath.net

FIGURE 1: Doppler distortion



$$x = A_1 \sin 2\pi f_1 t + A_2 \sin 2\pi f_2 t$$

Assume $A_2 < A_1$

c = speed of sound = 1128ft/second, M = modulation index = $\frac{\Delta f_2}{f_1}$

velocity $v_1 = 2\pi f_1 A_1 \cos 2\pi f_1 t$, peak velocity (f_1) = $v_{1p} = 2\pi f_1 A_1$, peak frequency deviation =

$$\Delta f_{2p} = f_2 \frac{v_{1p}}{c}, m = \frac{\Delta f_{2p}}{f_1} = \frac{f_2}{f_1} \frac{v_{1p}}{c}$$

$$M = 2\pi A_1 \frac{f_2}{c}; \text{ for } M < 1, d =$$

$$\frac{\text{rms sideband amplitude}}{\text{rms unmod amplitude}} = \frac{m}{\sqrt{2}}$$

$d = \sqrt{2} \pi A_1 \frac{f_2}{c} = \text{RMS Doppler distortion (upper and lower 1st order sidebands)}$

$D = \% \text{ Doppler distortion} =$

$0.03282 A_1 f_2$ (A_1 is in peak inches).

Define S = Subjective % Doppler distortion, proportional to $\frac{f_1}{20\text{Hz}}$;

$$S = 0.001641 A_1 f_1 f_2$$

A1 excursion, peak	f_1	f_2	d RMS	S RMS	relative amplitude 1st order sidebands, each of 2
0.5"	50Hz	2000Hz*	32.8%*	82.1%	-12.7dB
0.125"	40Hz	3000Hz	12.3%	24.6%	-21.2dB
0.5"	20Hz	300Hz	4.9%	4.9%	-29.2dB
0.5"	0.1Hz	300Hz	4.9%	0.025%	-29.2dB
0.125"	100Hz	10kHz	41%	205%	-10.7dB

* Example shown by Mr. Koonce

HORN GUIDELINES

Ed Simon's article, "A Combination Horn You Can Build" (*audioXpress*, January 2007) was a real source of inspiration to me. I've always liked horn enclosures, and Ed's seems to me to be one that is thoughtfully designed, promises great performance, and is easy to build! What more could a speaker-putterer with marginal woodworking skills ask for? I really want to put together a pair of these, but feel a bit at-sea due to the lack of a dimensioned drawing. The Fig. 1 Cutting Pattern is a big help, and Photos 3 and 4 show the box in the process of assembly, but a picture's worth a thousand words and front/side drawing with all the hidden lines seems a good idea to make sure that it comes out right.

Jim Wood
Brea, Calif.

Ed Simon responds:

I am sorry I did not do a dimensioned assembly drawing when I built them. Because they are fully assembled and I do not have access to the interior, I cannot do an actual accurate as-built diagram. Perhaps someone else building them would be kind enough to do one.

Of course, if you want to try it out on your own without risking any serious investment in time or materials, you could build a prototype out of drywall. It is cheap and you need only a knife and straight edge to score and snap it.

Sorry about that. *aX*



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Modifying Mighty Mouse

By Bob McIntyre

The Little Amp That Could
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I'm an inveterate tinkerer. Give me an amplifier—even a brand-new one—and I'll tear into it in an attempt to extract that last little bit of performance. This cannibalism extends even to those projects I've built myself.

My little tube amp Mighty Mouse appeared in the April 2006 issue of

audioXpress. I thought the design was reasonably finalized, but in the year or so I've lived (and tinkered) with it, the relatively minor changes I've made have resulted in a huge improvement in its performance. ■

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New Products

The TA2 Tube amplifier, from **DLS**, contains silver wire capacitors, DLS Audio capacitors, and WIMA polypropylene capacitors, and has a dual mono power system design as well as a separate power supply for tube and preamp stage



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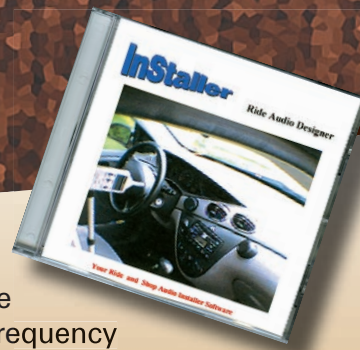
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by F. Langford-Smith

Long out-of-print and hard to find, the complete fourth edition (originally a 1,498 page text) of this classic is reproduced on CD-ROM. This self-explanatory handbook covers design and application of radio receivers and audio amplifiers. The 50-page table of contents has been linked to the more than 7,000 items in the text. Adobe Acrobat reader included on the disc. Sh. wt: 1 lb.

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AUDIO IN THE AIRPORT

By Vance Dickason

The San Francisco Airport Museum, in cooperation with the Audio Engineering Society, put together an exhibit entitled “The History of Audio: The Engineering of Sound.”

Installed to coincide with the 121st AES Convention that was held at San Francisco’s Moscone Center October 5–8, 2006, the exhibit runs through May 2007 in the North Connect Gallery of the airport’s Terminal 3. Exhibit display case topics range from “Then and Now” and “Sound Gets

Its Groove: Edison’s Phonograph” to “Digital Sound Comes to the Cinema,” “Breaking the Noise Barrier,” and “Home Entertainment in the 21st Century.” Between the cases are displays with huge photos honoring audio pioneers. Among the photographs are those of Sidney Harman, PhD (b. 1919); Frank McIntosh (1906–1990), who invented an amplifier “with an unprecedented combination of high power, low distortion, and wide bandwidth”; Edgar Villchur (b. 1917), inventor of

the first acoustic-suspension and compact loudspeakers; Avery Fisher (1906–1994), who founded Fisher Radio after WWII; and Saul Marantz (1911–1997), whose wife persuaded him to build and market the high-quality preamp he had developed. Among the exhibit are vintage AR speakers, KLH components, a 60s Heathkit preamplifier, a belt-driven turntable, a horn from a theater speaker system, recording equipment, and other articles from the audio industry’s past.

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PHOTO 1: Audio pioneer Henry Kloss.



PHOTO 3: Vintage hi-fi gear.

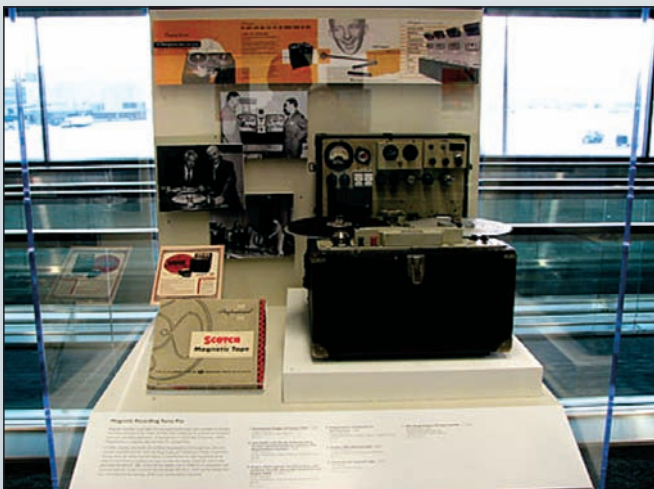


PHOTO 2: Vintage recording equipment.



PHOTO 4: Vintage cinema horn.