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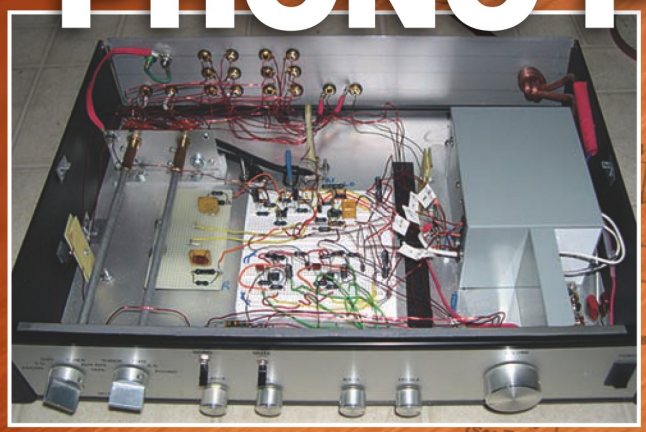

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Moving On. . .

Gutenberg started it in 1455 by printing 180 copies of the Latin Vulgate Bible. Thus began the habit of reading books. His Bibles were over a thousand pages and they began a revolution that profoundly changed society and reshuffled the power of kingdoms and religious beliefs.

Since then his basic printing process has evolved steadily. Workers go into the woods and cut millions of trees, chipped into bits, mushed into pulp, rolled into huge rolls of paper on machines which cost over a billion dollars and are the size of two football fields which must run 24/7, and smell bad. The rolls are trucked to presses costing a million or more each, which pour out little collections of stitched or glued-together pages on which computers write addresses, followed by bundling in packs for the correct postal facility and finally delivered to your mailbox.

The owner of the *Wall Street Journal* predicts this will all be gone in a decade.

Nobody, probably including you, will like the change from a printed magazine to one that arrives at your computer monitor after your e-mail tells you it is ready. We did not realize that the computer would make physical addresses redundant. Web addresses are now becoming the method for finding stuff and each other.

At Audio Amateur we have always moved with the technology and have hauled a distressing lot of expensive equipment to the recycle center. The time has come when we must change again. Our small size as a niche specialist publisher makes us more vulnerable than most in this field. We already prepare our pages for the big lithographic press in a digital format sent electronically overnight to the printer. We OK our printer's proofs over the same route two days later.

Now, with some inexpensive software, we can process those same pages, enhance them in some impressive ways, place them on a big, very secure server which we can rent inexpensively, and send you an e-mail telling you the address of the server where your magazine's content for the month is located. You will have supplied to us a username and passwords which will unlock your copy.

We will continue to produce *audioXpress* in both printed and digital formats. For how long will depend on your choices. If many of you make the change, we can probably lower the subscription price. As this happens the price of a subscription to the printed version must rise.

Our costs have steadily risen for everything we use to publish and deliver your copy of *audioXpress*. We are also in the middle of the worst economic decline in 80 years. Advertising budgets of our most faithful clients are down. We have taken every step we can to economize and to become more efficient. So these are powerful incentives to move in a new direction to put the publication on a firmer, economically sound foundation.

It is particularly critical that overseas subscribers change to the digital version, since postal rates are rising more rapidly than ever, and delivery is slow and often unreliable. The digital *audioXpress* will be published ahead of the printed version. You will find a survey on our website (http://www.audioXpress.com) which asks about your preferences. Our website will also contain a form for changing your subscription to digital delivery which will require only a minute to fill out. You will need your magazine's mailing label to complete the form.

I believe you will find a digital version of *audioXpress* more fun and more useful than its printed form. Digital magazines are interactive and ours will have all sorts of links to enhancements which are available with a click of your mouse. We hope to add audio and video clips to articles, and links to the sources of parts for a building project, and the bibliography can be "live" rather than difficult to find. Advertising will also be more active with direct links to those who offer you the products which make this avocation possible. When we can again do reviews of recordings, it may be possible to include short samples.

Please give some serious thought to this proposed format change. Gutenberg probably could not have imagined how much his work would revolutionize the world. The reading habits he instituted are woven deeply into our brains. But we must ride this tide for our efforts to survive. Your help is vital. **—ETD**

An Inductively Equalized LP Preamp

The design for this phono preamp is simple, includes inexpensive parts, and features the use of inductor over capacitor.

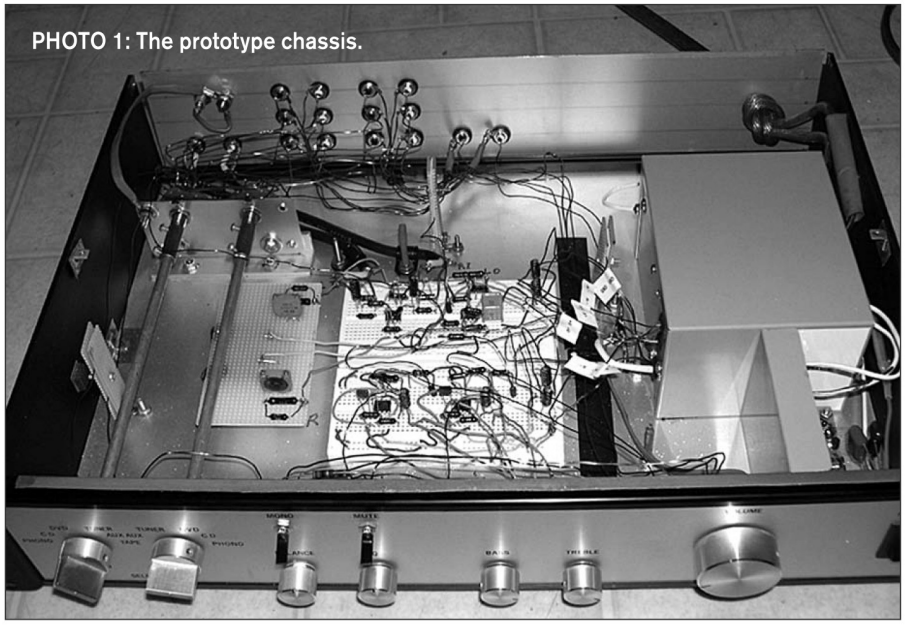
PHOTO 1: The prototype chassis.

Despite the mass attraction of cheap (and typically highly compressed) downloads in the digital age, the LP still lingers in some circles as a desirable musical alternative (or antique collectible). Oftentimes the implementation of a phono preamp for playback is either looked at as an afterthought, or given the status of a black art of the past, with subtle intricacies not readily appreciated nowadays.

BACKGROUND

In order to understand the job of a phono preamp, a brief overview of the various components in the system is in order. Stating the obvious is the LP itself. Typically made of vinyl and usually mastered to the RIAA standard, the amplitude information stored in its groove is not flat with frequency, but rather follows the curve of **Fig. 1**. Extracting the information from the groove has typically been the job of a cartridge/stylus device (though lasers have been used in some instances). There have been different types of playback cartridges developed, but the most popular by far has been the magnetic type.

Consisting of small magnets and coils, the magnetic cartridge is essentially a small generator whose output exhibits a rising amplitude with increasing frequency for a fixed record groove amplitude (**Fig. 2**). Superimposing the two curves of **Figs. 1** and **2** yields the curve of **Fig. 3**, the inverse of what the phono preamp must do to produce a flat system frequency response.

Traditionally the RIAA playback curve (**Fig. 4**) has been implemented with an RC network, either in the feedback loop of an amplifier, or using a "passive" approach; i.e., an RC attenuator to ground between two amplifier stages. The feedback configuration has the advantage of needing only one amplifier per channel (helpful with complex discrete designs), but places a high open-loop gain requirement on the amplifier itself. The passive design splits the gain requirement between two amplifier stages, making it easier to achieve high performance with modest amplifiers.

CIRCUITS

A third topology also exists, a variation of which is presented in **Fig. 5**. Essentially a hybrid of the two approaches, one part of the curve is corrected for in the first stage, while the other part is corrected for in the second stage. Referring to **Fig. 5**, the first stage of the preamp corrects for the slope of the magnetic phono cartridge (**Fig. 2**) via the use of an inductor, L1, and also produces the 50Hz breakpoint in **Fig. 4**.

The use of the inductor to compensate for the cartridge slope seems to be pivotal for sound improvement, because no capacitor implementation sounded comparable (the reason for this remains speculative). The type of inductor doesn't seem too critical, as long as it has sufficient ambient noise rejection, and doesn't saturate in the audio band. I used a small 25mH toroidal inductor (originally manufactured by Dale) in

the prototype. Others have wound their own on shielded pot cores and actually achieved better electrical performance than with the toroids (see source recommendations).

The second stage of the circuit corrects for the record groove characteristic in the midband areas at 500 and 2122Hz (**Fig. 1**). It's interesting to note that you could plug a ceramic phono cartridge into this second stage with nothing more than the addition of a 1.5M bias resistor to ground and the removal of C13.

The ±9V power supply is provided by pi filters. No regulator is used, as sufficient headroom to the op amp's maximum operating voltages exists with the 9V rails, and the pi filter is "stiff" enough not to exhibit much load sag by the drain of the small op amps. The circuit itself is built on prototyping boards. With most unity gain stable op amps, the proto boards have proven reliable. As long as reasonable layout is implemented, there's no reason to expect any trouble using other construction techniques. Output offset in the prototype ran around 2.7mV with the NJM4559 op amps. If this is unacceptable, you can implement output coupling capacitors.

There are many different types of op amps that will work in the design. I tried over a dozen different types before settling on the somewhat obscure 4559s. For whatever reason, in the application they compared well tonally to discrete designs (they're also very inexpensive). Obviously you could upgrade to any op amp of choice as long as the operating voltages and unity gain stability criteria are met. The input stage is fairly conventional, with C2 and R2 forming a high-pass filter to help reduce record rumble.

If you desire, you can replace C2 with a jumper for a slightly more transparent sound. C1 is basically a place holder for the desired value of C loading for the cartridge of choice, and should be selected on an individual basis, because turntable wiring, as well as cartridge selection, will influence its value. The resistances of R5, R6, and the DCR of L1 add up to 8Ω and, combined with the inductance of L1, produce the 50Hz breakpoint shown in **Fig. 4**. When using various inductors or winding your own for the application, DCR will inevitably vary, so you may need to adjust the values of R5 and R6 to maintain an overall DC resistance of 8Ω in the circuit leg.

COMPONENT SELECTION

In my experience, you don't necessarily need expensive parts for most audio applications (and I've tried many of them over 35 years). The fact that some people hear a difference doesn't automatically equate to the more expensive part being the better one. The Vishay/Dale RN60 metal film resistors offer good performance at a reasonable price. Pot

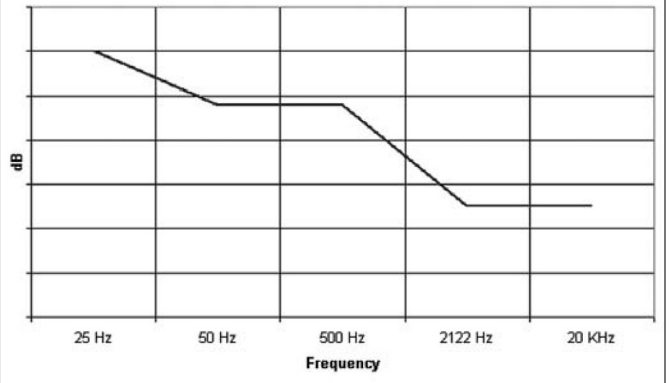

FIGURE 1: Groove amplitude versus frequency for an RIAA mastered LP.

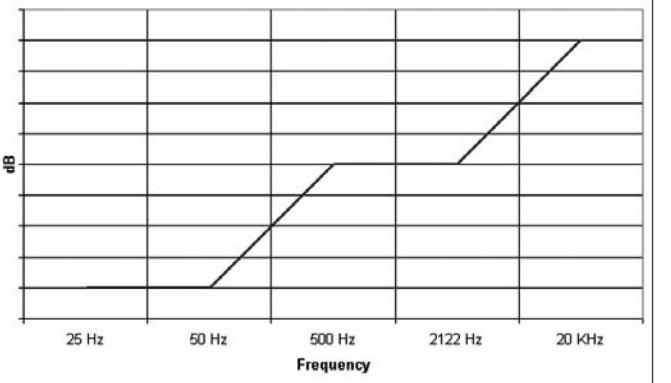

FIGURE 3: Magnetic cartridge output signal for an RIAA mastered LP.

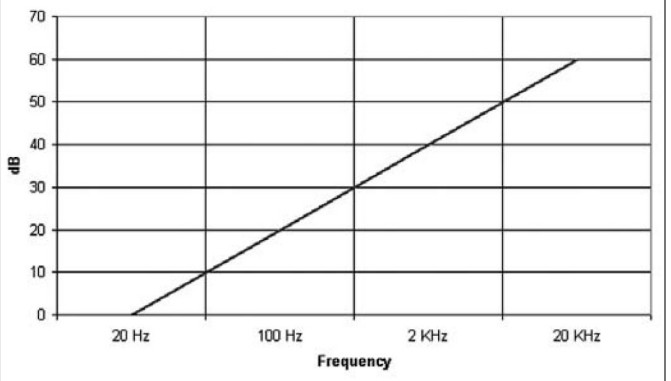

FIGURE 2: Magnetic cartridge response to a constant groove amplitude.

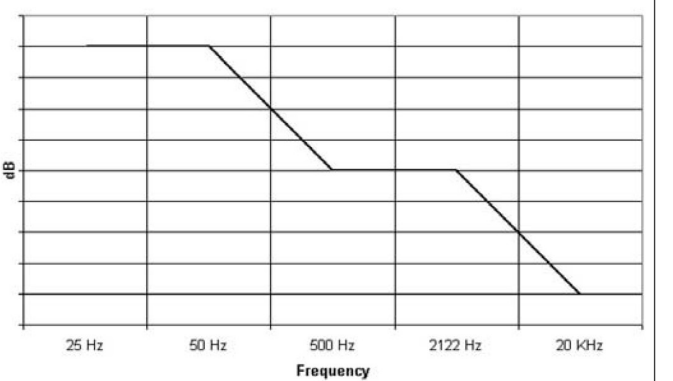

FIGURE 4: Preamp response for a magnetic cartridge.

cores for constructing L1 are available for a few dollars on the Internet. The 4559 op amps are available for around 50 cents each. The SMG electrolytics made by Nippon Chemicon are quite inexpensive and have some of the best sound I've heard for an electrolytic—no graininess, fairly neutral tonality, and natural dynamics.

This isn't to say that you shouldn't experiment—quite to the contrary, it's a great way to learn—but building the preamp with the recommended parts as a baseline is an inexpensive and prudent idea before experimenting. The power supply is adequate to power two channels, but the parts list of the preamp itself will need to be duplicated for the second channel (except for the dual op amps).

ONLINE RESOURCES

Amidon for the L1 inductor cores:
www.amidoncorp.com/items/73

Digi-key—diodes, op amps, and capacitors, including the Nippon Chemicon (United Chemicon) SMG electrolytics:
www.digikey.com/

Mouser—diodes, op amps, capacitors, and resistors, including the Vishay/Dale RN60s:
www.mouser.com/home.aspx

Avel Lindberg—toroidal transformers:
www.avellindberg.com/

NJM4559 Datasheet (listed under "low noise" op amp):
www.njr.com/index4.html

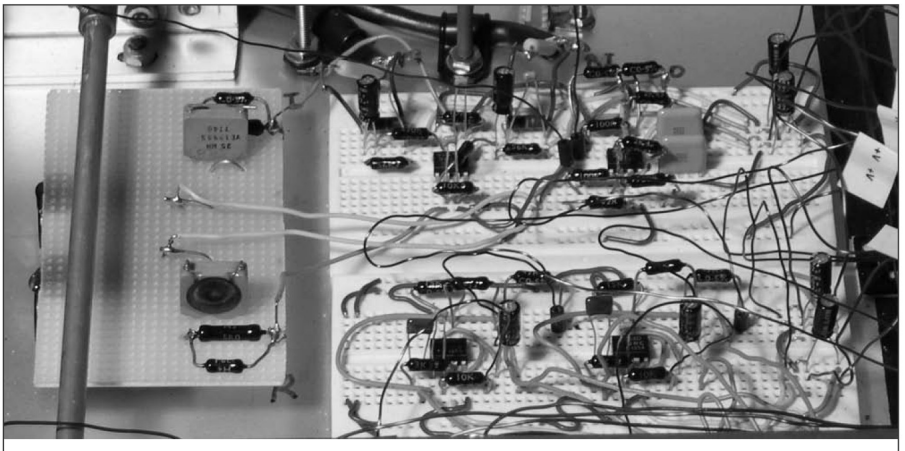

PHOTO 2: The prototyping boards.

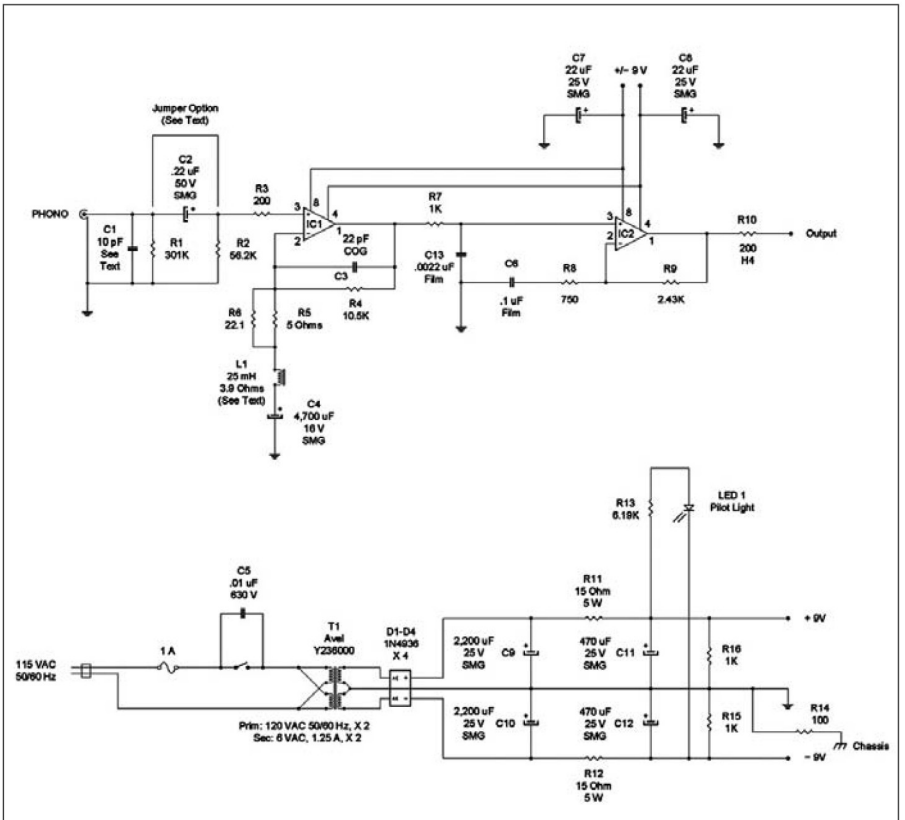

FIGURE 5: Schematic, basic preamp.

PARTS LIST

Capacitors

C1	0pF, COG or Film
C2	0.22µF, 50V, SMG, Electrolytic
C3	22pF, COG
C4	4,700µF, 16V, SMG, Electrolytic
C5	0.01µF, 630V
C6	0.1µF, Film
C7, C8	22µF, 25V, SMG, Electrolytic
C9, C10	2,200µF, 25V, SMG, Electrolytic
C11, C12	470µF, 25V, SMG, Electrolytic
C13	0.0022µF, Film

Resistors

R1	301K, Vishay/Dale RN60
R2	56.2K, Vishay/Dale RN60
R3	200, Vishay/Dale RN60
R4	10.5K, Vishay/Dale RN60
R5	Vishay/Dale RN60 (see text)
R6	Vishay/Dale RN60 (see text)
R7, R15, R16	1K, Vishay/Dale RN60
R8	750, Vishay/Dale RN60
R9	2.43K, Vishay/Dale RN60 (use 2.49K if 2.43K is unavailable)
R10	200, Vishay/Dale RN60
R11, R12	15Ω, 5W, Wire Wound
R13	6.19K, Vishay/Dale RN60
R14	100, any type

Inductor

L1	25mH shielded pot core or toroid, see text

Semiconductors

D1-D4	1N4936 or similar
IC1, IC2	NJM4559, or user's choice
LED1	User's choice

Transformer

T1	Avel Lindberg Y236000

Miscellaneous

Power Cord
1A Fuse
Fuse Holder
Chassis
RCA Jacks
Power Switch
Construction Board
Wire

Chassis 101, 201, 301

A guide to constructing your own enclosures and cabinets.

Many of the projects published in *audioXpress* require a chassis or enclosure for a professional appearance. Unfortunately, in some cases the cost of such a chassis or enclosure can substantially add to the cost of a project. If you have seen some of my past articles ("Scrounging 101," July '08), you know that I tend to make use of what's available. Even with this frugal side, I have had projects appear on the front cover of *audioXpress* ("Junk Box Tube Amp Jubilee," Dec. '05).

I will share with you a variety of techniques developed over the years to allow you to produce a professional, functional, and inexpensive custom chassis or enclosure for your next project. I will consider three chassis groups: first, a chassis that you can salvage from other projects and repair; second, a simple, sturdy and attractive chassis that virtually anyone can build with just a few hand tools; and, third, a more complex version that requires at least a few power tools and some precision to complete. I am an amateur woodworker and have—through the years—become an acceptable metalworker.

TOOLS NEEDED

- Drill/drill driver and assorted small sharp drill bits
- Countersink bit
- Assorted hole punches (for mild steel) or bimetal hole saws (for acrylic and aluminum (and mild steel, if you are very careful) (½″, ¾″, 1″, 1 1/8″)
- Acrylic hook scribe for cutting acrylic
- Hand saw or hand miter box, or ideally a table saw with 60+ tool sharp carbide blade
- Air brad gun or small nails, clamps (one web clamp will do for most chassis), and pilot bit to pre-drill holes in hardwood (if you don't have an air brad gun)

MAKING DO

The first group of items may include old tube equipment chassis that were stripped for parts or transformers (see my "Scrounging 101" article). I never (well, almost never) throw anything away, but I do have a well-organized junk room, which is the first place I go when I want to start a project. I evaluate what I need for a chassis or an enclosure, hoping to find one that will work with few modifications, before I decide to build or buy (Heaven forbid) a new one. I spend time in surplus outlets and places such as Goodwill searching for gear that may be useful later on.

The old-style Pilot amplifier ("Upgrading a Classic Pilot Amp," April '08) started life as a commercial rack-mount telephone amplifier using four pin tubes. After stripping it of all parts, I spent some time figuring out how to make the best use of the original chassis holes while allowing additional new ones. Place all major top-mounted parts on the candidate (include tube sockets and can caps, as well as transformers) and move things around to find a pleasing look. In some cases you will be able to mount transformers, caps, and chokes under the chassis, so consider this also.

Don't be afraid to arrange things in an unconventional manner; just try to keep parts such as preamp tubes and low-level circuits away from transformers. I have one amplifier that has a V arrangement for tubes, and everyone who sees it thinks it is very unusual (not sure whether that is good or bad).

During the course of building Bill Christie's LM 12 IC AMP (*audioXpress* Aug. '07), which included an excellent discussion on building enclosures for these amps, I decided to use a really cool chassis that had been the power supply for a home-built pulse laser (I bought it at a surplus store, which had bought the estate). I coveted the two very heavy 60V center-tapped transformers and capacitor banks. It also included a multitude of relays, big bridge rectifiers, switches, and heatsinks and was made out of heavy aluminum angle and plate.

I used one of the transformers for another project, sold the other one (paying for my whole purchase), salvaged the other parts, and left the capacitors on the chassis. Some careful measurements demonstrated that the parts, heatsinks, and boards would fit, and would provide a very unusual-looking amplifier (**Photo 1**). I cut Plexiglass for the tops and bottoms. I really like acrylics, because they are easy to work with and can be very visual (more about this later).

HALF-BAKED

The second category includes items, such as cake pans, that were designed for something else, but can be made to serve a new purpose. I once used a cake pan for a chassis, when I was 13 years old, but got so much grief for it, I never did it again. (Mom was a little upset that I used "her" cake pan and my friends laughed at me.) Some builders commend them as cheap, sturdy, and attractive, but I just think they look like cake pans. If you use cake pans, don't take offense; instead apply some of the suggestions in this article

to build a quality, good-looking, and budget friendly chassis. Believe me, this will save you a ton of grief when your "cake pan" overshadows your wonderful project.

If you haunt surplus stores, you will find a wide variety of chassis and enclosures—many of them industrial or military surplus, containing a selection of cool parts and usually a very rugged chassis. It is quite easy to fashion a new aluminum front and back panel to cover existing holes (more on working with aluminum later). I see tons of really nice rack-mount equipment originally designed for phone/CAT 5 junction boxes and servers. These are great for tube amps because most of the top and bottom panels are solid and relatively hole free. Again, you make a new front and rear panel and you are good to go.

This also makes a pretty good environmental statement. I recently wanted to write an article on building an inexpensive PP tube amp (less than $150 for everything) that would be easy and inexpensive to maintain and retube. My first stop was eBay, where I picked up a nice set of power and output transformers from a Harman-Kardon 300 integrated amp ($62 including shipping). I used almost half of my budget, so it was time to find a very inexpensive chassis (**Photo 2**). I was in Gateway Electronics in St. Louis and found three enclosures used for serial and parallel port printer switching. They were 10″ × 6″ × 1½″, beige powder coated (one of my favorite finishes), and the front switch and back sockets were already removed. These were marked $4, so I bought all three.

I laid the major components out on the cover of one of these and decided I could make it work. I found an appropriate size pilot light for the existing front hole, drilled the rest I needed, and then cut a couple pieces of aluminum to cover two of the three back slots and then used a barrier strip to cover the last one. I also used a piece of scrap angle aluminum underneath, to stiffen the top of the chassis.

Before I go any further, a word about aluminum. I really like working with aluminum—it is easy to work, drill, and machine, looks good, and can be finished in a variety of ways. The best place to find sheet aluminum is at metal recyclers. They often have racks of it for sale.

The price is usually by the pound, and you can buy many scraps for very little money. Even if they don't have a display of this type of stuff, be very nice to the counter person and explain what you are looking for (flat sheets, angle, and so on) and ask whether there is any place you can look for it. My experience is 90% of the time they will send you to the scrap area for aluminum and you can freely pick what you want. If it has some scratches or corrosion, don't worry—I will also talk about that later.

I recently walked into one of my usual places, where the counter guy had saved some "nice stuff" for me. He had four sheets of 1/8″ plate about 2′ by 2′ with one side covered in white plastic. This is really high-rent material! The plastic protects the satin or brushed finish and makes a great canvas for your layout. I paid about 8 bucks a sheet, and later you will see the use of some of this. Obviously, I thanked him profusely. I try to have a variety of thin and thicker material on hand, as well as some "angle" aluminum (I figure that if I die suddenly, my wife can always sell it as scrap).

If you can't find any recyclers close by, most large home stores or hardware stores have a "metals" rack containing steel aluminum, brass in sheets, angles, tubes, and squares. This is obviously more expensive, but it is usually new and in good condition, which may require less work in the long run. Keep in mind that your finished product will still cost only 10-30% of a pre-manufactured enclosure.

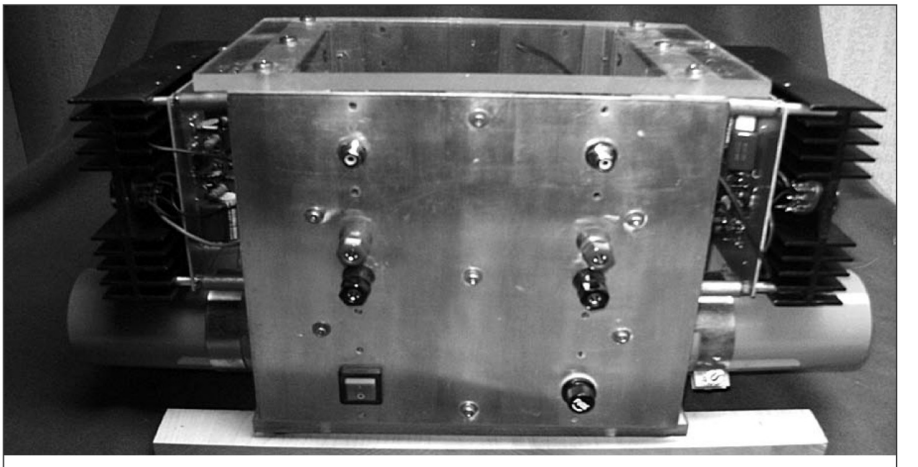

PHOTO 1: Aluminum castle.

PHOTO 2: Complete amp.

FROM SCRATCH

I have mostly taken existing parts and modified them to work, but you can also build some custom gear using some very helpful tools. While you can build some gear with hand tools, you should invest in a table saw—even a tabletop model—which you can purchase for $150-300. I had a Makita 10″ tabletop model for many years (since upgraded). Or, you might get some help from one of your friends with a saw. You will be cutting wood, acrylic, and aluminum, so you will need 60+ tooth quality carbide-tipped blade. Don't scrimp; a good blade can be resharpened 6-8 times (cost is usually about $15-20 per sharpening).

Acrylic and wood make a great first chassis, because they are easy to work, even with just hand tools. You can buy 1″ × 1″ or 1″ × 2″ hardwood at any home improvement store and cut it with a hand miter box or hand saw to build the base of the chassis. The same process for the base is used for either an aluminum or acrylic top. The acrylic should be 1/8 to ¼″ thick, depending on the load that will be on it. It is available in a variety of colors, finishes, and thicknesses (check Plastics in the Yellow pages).

Most stores will usually cut pieces for you, and charge by the square foot or by the sheet. They use a tablesaw with a special (read: expensive) plastic cutting blade that leaves a very nice edge. You can cut acrylic on a regular table saw, but if you don't have one, you can buy an acrylic scribe (a sharpened hook used to scribe the acrylic, **Photo 3**).

Lay down a good metal rule and pull the scribe from one end to the other on the line you want to cut. Do this four to six times, applying more pressure on the subsequent passes. You should see a little continuous plastic curl coming up from the acrylic. The thicker the material, the more passes you should make.

Once you have cut a very strong groove, place the larger part of the piece with the groove up on a table or piece of wood with a sharp edge, align the groove over the edge of the table, and sharply hit the part hanging over with the palm of your hand. You should be rewarded with a sharp snap and a perfectly "cut" piece. Repeat this on the other edge for the size you want. It is very helpful to make your breaks with at least 2-3″ hanging over the edge.

If you are using fairly thick material, you will want to sand the edges. Start with 100 grit and when you get a nice smooth edge, move to 200, repeat, then go to 400. If you want a really nice clear edge, use 800 and 1000 grits, and then use plastic polish. I very seldom go past 200 because I don't care whether I have a clear edge.

This particular chassis (**Photo 4**) took me less than 15 minutes from start to finish. One of the side benefits of using acrylic for the top or front of a chassis or enclosure is that it forces you to be very careful in your wiring (after all, you don't want everyone to see a rat's nest).

MORE BOXES

You can make a very simple enclosure out of three pieces of hardwood and two pieces of acrylic (or two pieces of wood and three pieces of acrylic). I built many gain clone chip amps (**Photo 5**) for friends using a piece of hardwood (or acrylic) for the bottom and two pieces of hardwood for the ends. I then cut acrylic for the front and back, drilled screw holes into the wood, and in about 20 minutes built a very attractive enclosure.

A couple of notes about using acrylic: Don't place it near heat (such as class A amp heatsink mounts) because it will bend and deform. Different kinds will take different amounts of heat with polycarb probably being the most heat resistant. Acrylic obviously doesn't conduct electricity, provide a ground plane, or shielding, so you must be creative with those needs. For input connectors, you can drill large holes in the acrylic and mount an aluminum

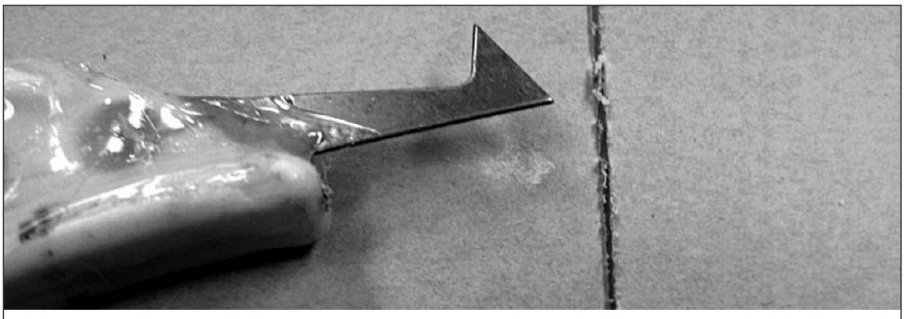

PHOTO 3: Acrylic hook.

PHOTO 4: Complete acrylic and wood chassis.

PHOTO 5: Gain clone amp case.

plate to the inside of the panel.

If you mount LEDs in acrylic, try just press fitting them rather than using commercial mounts. This will give you a lighted edge on the acrylic, providing a very nice visual effect. I typically use blue LEDs and recently found some aqua ones. I have seen other colors including violet, so you have several choices.

I built one amp for a friend with a thick acrylic base and clear front and back. He liked the blue edge on the front panel so much that he asked me whether I could incorporate LEDs into the other panels. I put two in the rear of the bottom panel and one on each side panel. He now has a glowing blue box in his living room and receives many favorable comments on it.

Traditional wood and metal chassis are also quite easy to build, and you can make them any size you need. When building the base for the chassis, you can use either miter joints or butt joints. Pick through your wood supply to find the best grain, and use this for the front of the base. Your front and back pieces need to be the exact same length, as do the sides.

Once everything is cut, you can glue-clamp the pieces together with a web clamp (**Photo 6**). It is a good idea to also use a couple of clamps to secure the box to a flat surface while the glue dries. You can also use a small square to check that the frame is square. I sometimes do my metal work and finish before I glue chassis together, and sometimes I don't glue the back panel; instead I use two or four countersunk screws from the back into the side panels. For bigger chassis this allows better access to all corners.

Sand the base pieces with 100, then 180, and finally 400 grit paper. There are hundreds of "finish" products out there. I like polyurethanes because of their durability and fast drying time. I also use Watco Danish™ wipe-on finish or Tung oil. Lay on a heavy coat with a foam brush, allow it to soak in for about 15-20 minutes, and then wipe off the excess with a paper towel. Allow to dry for several hours, then lightly sand with 400 grit paper between coats. Repeat the process until you get the gloss or finish you are looking for.

For boxes that mount the panel on the top, you will need to attach the top to the base. The process from here is pretty much the same whether you are using acrylic, mild steel, or aluminum for the top. Start by drilling a series around the perimeter of the top, about ¼″ in from the outside edge. (I generally put a screw in about 1″ from each corner and then space a few others between them.) If you are using acrylic or aluminum, you may want to countersink the screw heads. For acrylic I use brass #4 square drive production screws with a 5/64 pilot hole and a 7/64 hole in the top plate.

For aluminum or steel I use #6 round-head machine screws with a 3/32 pilot hole and a 9/64 shank hole.

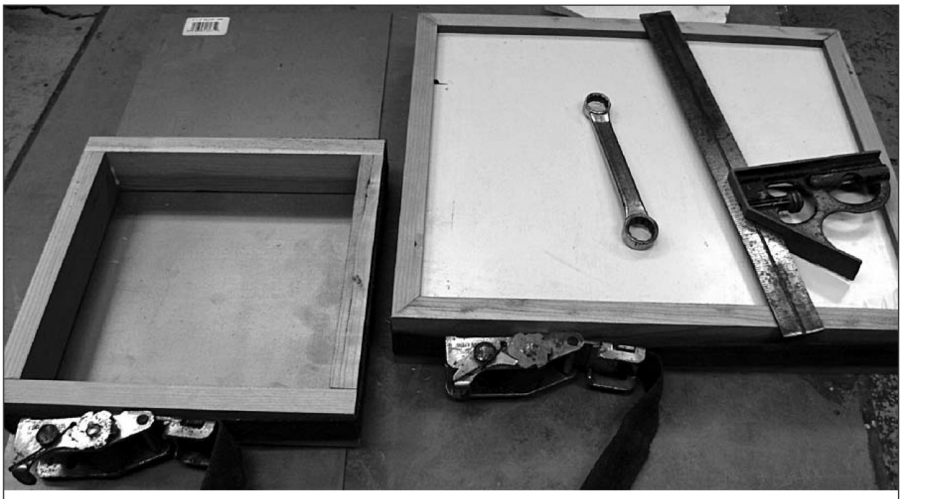

PHOTO 6: Boxes being clamped.

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If you are using heavy aluminum for the top, you can also countersink them. The best way to buy screws is by the box of 100 (save money), or order them from McFeely's (www.McFeeleys.com). Once this is all done, you can take the top off and cut and drill the necessary holes for your component.

You can make a little fancier version of this type of chassis by using miter corners and a recessed panel (**Photo 7**), which requires more work. Your front, back, and side rails are "grooved" for the metal or acrylic. You will need a tablesaw, setting the fence for ¼″ below the height of the base. Set the blade height about ¼″ and run the back side of the front, back, and sides through the saw. If your panel is thicker than the cut, move your fence on the saw a little, and run them through again.

Once you have the top panel machined and punched, you can use a variety of finishes (**Photo 8**). Steel is easy to paint with a spray can. Rustoleum™ makes some very nice textured and antique veined finishes. Clean the surface well with lacquer thinner and spray several light coats (or as directed on the can) and then let dry for 24 hours. Then heat the oven to 250° and place the panel on the rack, close the door, and turn off the oven. This baking will make the finish tougher.

My favorite finish on steel is powder coating, which you can do at home (www.eastwood.com), but I have a local shop that does it for me. They are very reasonable, especially if I am in no hurry and can wait until they are using the same color on many other pieces. Powder coat is so tough that it is virtually impervious to chipping, wear, and corrosion. You can also do other parts including transformer bells and choose almost any color for your amp.

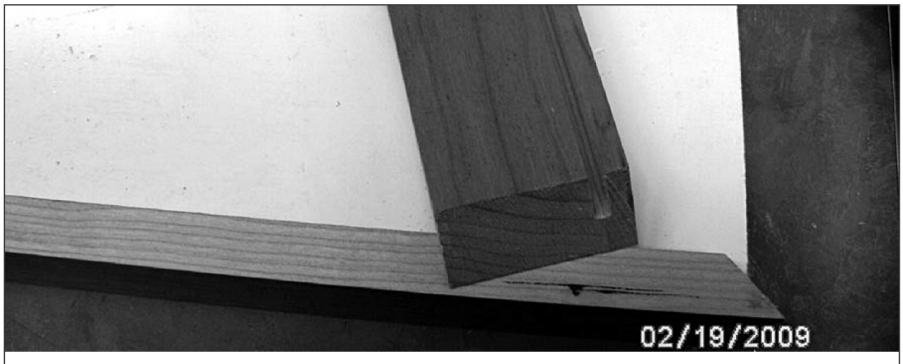

PHOTO 7: Recessed miter joint.

PHOTO 8: Three completed chassis.

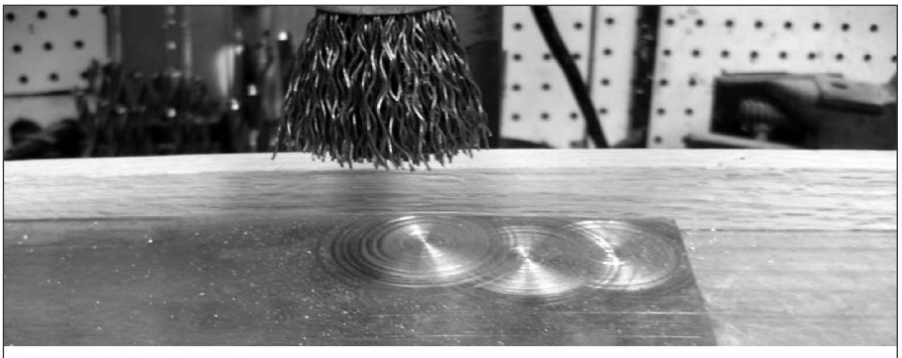

PHOTO 9: Turning setup.

WORKING WITH ALUMINUM

Aluminum panels present another challenge. If you have used material, it will no doubt have a variety of scratches and abrasions. The simplest way to deal with this is to sand starting with 80 or 100 grit paper and proceed to finer grades to achieve the finish you desire. If you use an orbital sander, you will get small circles, which diminish in size with each grade of finer paper. If you don't want these circles, hand-sand with a block of wood backing the paper. Another interesting finish is called an "engine turned" finish, which you see in very high-end, vintage car dashes and trim. This requires a drill press and small steel bristle brushes, as well as a great deal of time to build a jig to hold and move the panel for overlapping circles. I recently did a very large panel with a 1″ steel bristle brush designed for wire brushing metal. This wasn't as time consuming because it used a 1″ brush; each overlap was ½″.

Photo 9 shows the setup to do this. It requires that you have some way to guide your metal panel through a straight line. I set the depth range stop on my drill press so that each depression cut is the same depth. You can do the overlaps on the length visually, and move the table to the next row each time you finish a row.

Aluminum requires no cutting abrasive be used on the steel brush, but if you are going to do this to mild steel,

you will need to apply a paste of silicon carbide and oil on the brush periodically. You can purchase a brush at virtually any hardware store, and small brushes (with compounds) from Brownells (www.Brownells.com), which is a gunsmithing supply store in Montezuma, Iowa, carrying tons of really neat tools including metal coloring solutions and chemicals.

Graham Dicker (*audioXpress* June '04) previously discussed another type of chassis using C channel. I haven't found a good source for C channel, so I make my own steel C channels, with a small sheet metal brake (www.harborfreight.com). This is an inexpensive tabletop version of a brake you will find in a large sheet metal shop. If you don't want to buy a brake, you can sometimes get a heating and air conditioning shop to bend the metal for you. The rest of the construction is just making your end panels from the wood strips you used above. The only difference is you need to have two that are dimensionally smaller glued to the external pieces, for attachment to the channel.

The last type of chassis/enclosure I discuss is all aluminum. For this type of construction you really must have access to a tablesaw with a 60+ tooth *sharp* carbide blade. You can use any fine tooth carbide wood blade to cut aluminum. It is extremely important that you be very careful! I know this sounds crazy, but I cut heatsinks up to 5″ thick with my tablesaw, in addition to the angle and sheet used in this construction.

CUTTING METAL ON A TABLESAW

1. Always wear eye, ear, and hand protection when using any power equipment. I know this sounds obsessive, but a small piece of aluminum can blind you, or at least involve a trip to the ophthalmologist. I wear prescription shop glasses made with polycarb lenses with eye wings on the sides. They are expensive, but you have only one set of eyes! For hand protection I wear a set of Mechanix™ gloves, which are available in most hardware automotive parts stores and home improvement

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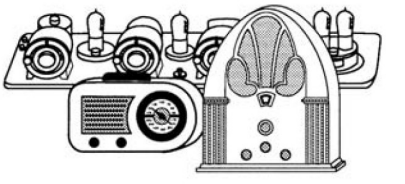

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stores. Buy a good pair that fit tight and have reinforced palms. A good pair of these will not only protect your hands, but still offer good tactile feel.

2. If you are cutting channel or angle, have the openings resting on the bed of the saw; do not attempt to cut angle material with either one of the edges facing up! Most of the time this will not cause a problem, but if you ever have an edge catch a blade at the wrong angle, you may have a piece of aluminum traveling at 60 plus miles an hour flying in your shop.
3. Make sure you have no distractions. My dogs go most everywhere with me, but they are not allowed in my shop (three 75 lb. labs can get into a lot of trouble).
4. When you have everything set up, turn on the saw and let the motor come to speed before feeding any material, and as soon as you are done with your cuts (before you reach for scraps of material), turn off the saw and make sure the blade has completely stopped.
5. NEVER, NEVER attempt to cut steel with a tablesaw!

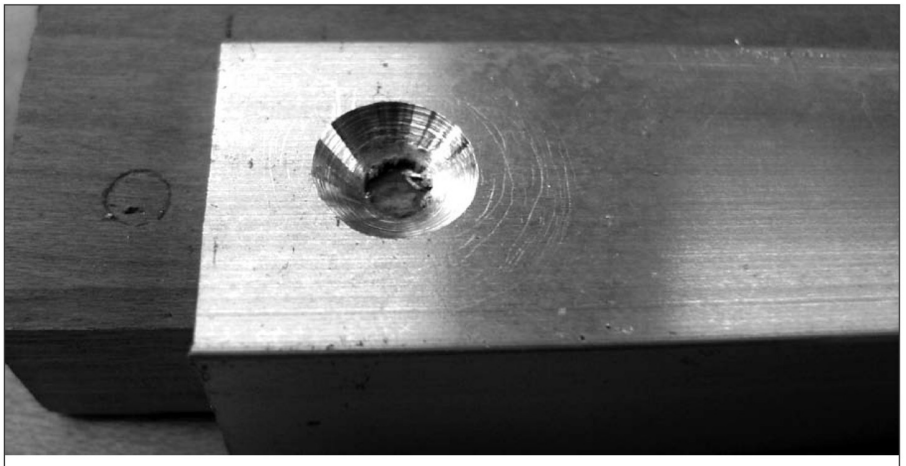

PHOTO 10: Countersunk hole.

You can easily build a complete enclosure for amplifiers that use heatsinks. Start by laying the heatsinks on each side of the components you are going to use. You need to calculate how much space you want between components and move things around until you get it right.

Now you can measure from the proposed attachment points on the heatsinks to establish the length of the angle aluminum you need. I use heavy ½″ × ½″ angle aluminum, but for very heavy stuff you can use the 1″ × 1″ material. Ideally you will cut four pieces to this exact length.

In this case, I was very lucky in that

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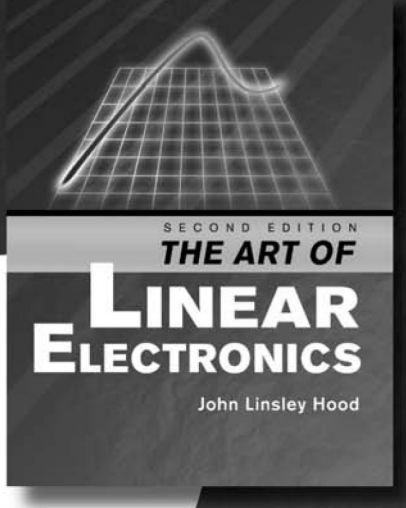

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the manufacture of these heatsinks included "screw slots." If you don't have these, you will need to drill the angle and heatsink and then tap the heatsink for a bolt. I have a drill, die, and tap set that provides 4-40, 6-32, 8-32, and 10-28 sets (available from hobby and model stores and online at www.micromark.com). Precision is the key here; clamp the angle firmly to the heatsink, drill the hole, tap the heatsink, and then drill a slightly larger hole in the panel and countersink the head (**Photo 10**).

Once you have all four in place, it's time to make the front and back panels. First check the squareness of your panel material (no matter how well your saw is set up, if the material isn't square the resulting panel won't be perfectly true). Cut your panel based on the squarest edge or by using your miter gauge to cut a true edge (**Photo 11**).

After you make the panels, it is a good idea to clean up the edges. I use an orbital sander with 80 grit paper to remove the cutting marks from the edge and follow with 100 grit for a matte finish. You can obviously move to finer and finer grades of paper and continue up to a polished edge.

Now clamp the panels to the chassis cage and drill holes for mounting through the front and rear panels. You can use panel head screws, or countersink them, but I prefer to use more decorative bolts for the front panel. I use black or chrome Allen head bolts, which add a special touch to the panels and are pretty inexpensive. You will notice that I have left the plastic covering on the panels for this chassis; they make great layout canvas, are easy to drill through, and protect the finish until I am ready for the final mounting of jacks, switches, pilot lights, and so on. The finished amp is shown in **Photo 12**.

Well, you now know all my tricks. Feel free to experiment and try different types of materials and finishes. If you have any additional questions or comments, you can contact me directly at Tuninfork@aol.com. I look forward to seeing your home-built chassis in the pages of *audioXpress*. *aX*

PHOTO 11: Cage and cut panel.

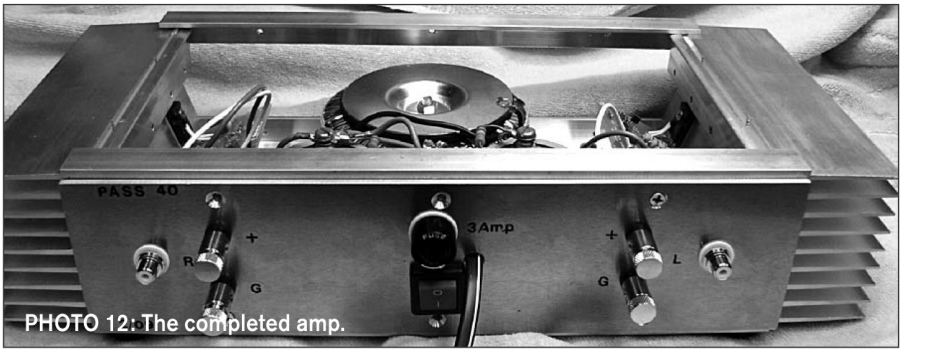

PHOTO 12: The completed amp.

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A SET OTL Amplifier

The author presents a new, improved output-transformerless amp design.

Using parts I had on hand, I presented a good-looking OTL amplifier in the May '08 issue ("Build a Hybrid SE OTL Amp," p. 34). Although it performs very well, I decided to change some basic things. The result is this new OTL amp design.

In the previous amplifier a PL509 pentode delivers the current to the speaker via an I/V converter and is assisted by a power booster in the form of a TDA7294 power op amp that also modulates the plate voltage when powering the speaker. First, this plate voltage variation of the pentode has no influence on the plate current itself, so no feedback exists. Second, the power supply voltages were a little too high for the booster and too low for the pentode.

In the meantime, I received a bunch of 6080WA double triodes, which I decided to use to make a new SET OTL amplifier. I needed a new power supply to deliver

a) ±30V DC for the booster
b) -120V DC for the triode
c) 6.3V AC for the filaments
d) ±15V DC for the op amps.

* The layout of the new amplifier is basically the same as the first amplifier. The first stage is the input circuit, followed by the SE triode circuit, then the conversion circuit, and finally the booster circuit. For each of these circuits I made a complete circuit diagram plus parts lists (**Figs. 1-4**).

THE INPUT STAGE

The 6080WA is a sturdy double triode with a rather low plate resistance and an even lower gain. The PL509 could be modulated straight from the line transformer, but the triode needed some more amplification (**Fig. 1**). I changed the original line transformer for a new Lowter model 1975, which has a balanced input. The RCA input is connected to the non-inverting input of the OPA2134 op amp that gives the 3× amplified input signal to half the primary of the line transformer Tr1, so the secondary signal to the grid of the triode is six times the RCA input signal.

THE TRIODE STAGE

As you can see on the complete SET-OTL schematic (**Fig. 5**), the next stage is the SE triode circuit. The 6080WA

PHOTO 1: Sowter line transformer model 1975.

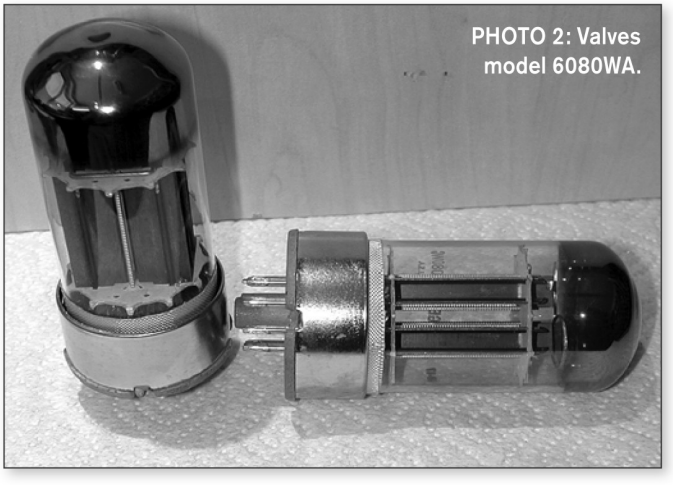
PHOTO 2: Valves model 6080WA.

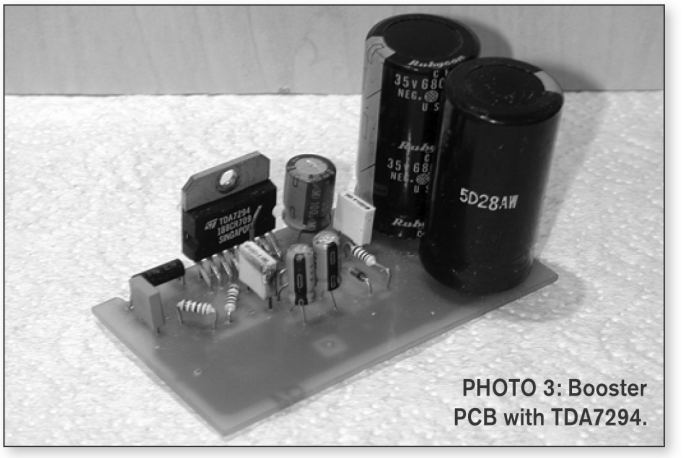

PHOTO 3: Booster PCB with TDA7294.

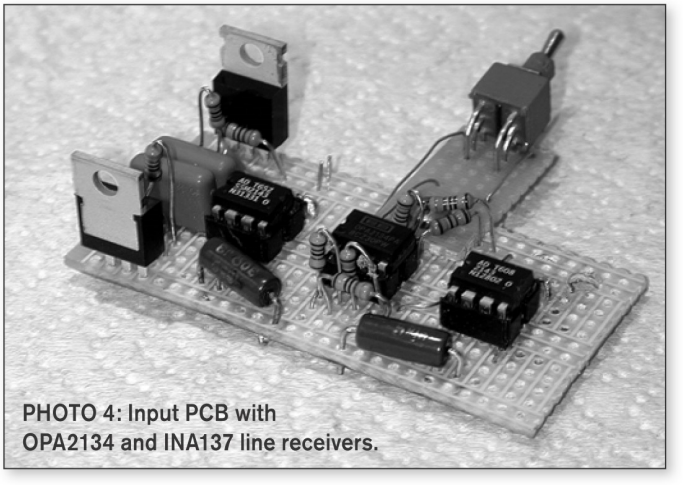
PHOTO 4: Input PCB with OPA2134 and INA137 line receivers.

consists of two triodes, each with a plate current of 100mA DC. With both triodes connected in parallel, the plate current is very sensitive to plate voltage variations. Stabilization of the plate current is a must and could easily be done by a simple constant current source (CCS) in the cathode line. The 15Ω resistor in each cathode line equalizes the plate currents within 5%. This CCS not only stabilizes the total plate current to 200mA, but nicely filters the residual ripple of the power supply as well. The output of the CCS is grounded for AC by 2 × 470µF Elcos (C2-3).

Through resistor R5 the grids of the triode get a negative bias voltage, which is filtered to ground by C1 (470nF). Resistor R3 prevents resonating of the line transformer at higher frequencies.

Zener ZD12 gives a stable 12V DC to the ten-turn trimmer P1, so you can preset the plate current to 200mA DC. The CCS circuit works very well; when varying the AC mains voltage with a Variac, the plate current didn't move.

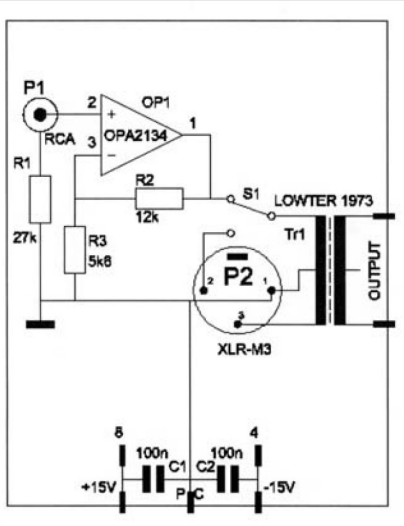

FIGURE 1: Input circuit.

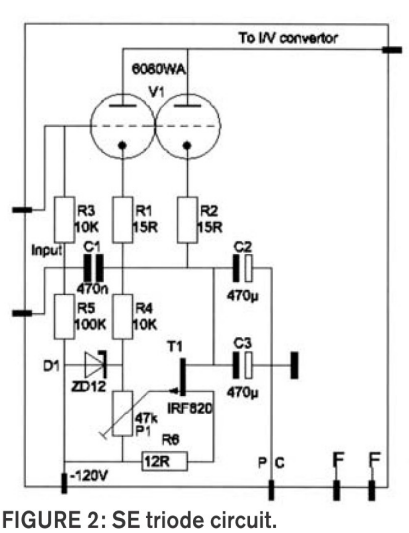

FIGURE 2: SE triode circuit.

THE I/V CONVERSION STAGE

The plate current of the triode flows through resistor R1 to the positive speaker terminal. With a DC current of 200mA, the input of the INA 137 line receiver is offset by 5.4V DC, and, as in this configuration, the gain of the line receiver is ½× the output to the following booster stage with a DC component of 2.7V, which is blocked by C1 (470µF). A rather high value for the booster has a low input resistance. Say that the plate current of the triode varies ±100mA, then the INA137 line receiver outputs a signal of 2.7Vp-p to the booster input. Being a line receiver,

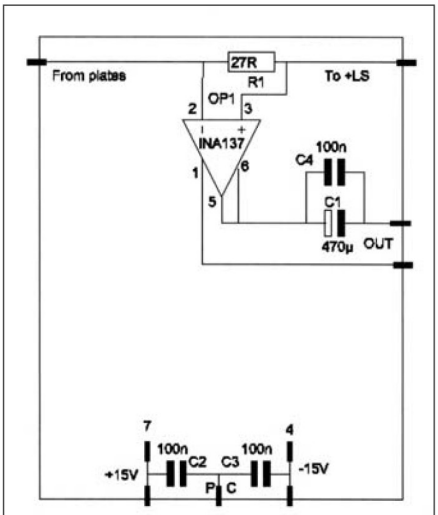

FIGURE 3: I/V converter circuit.

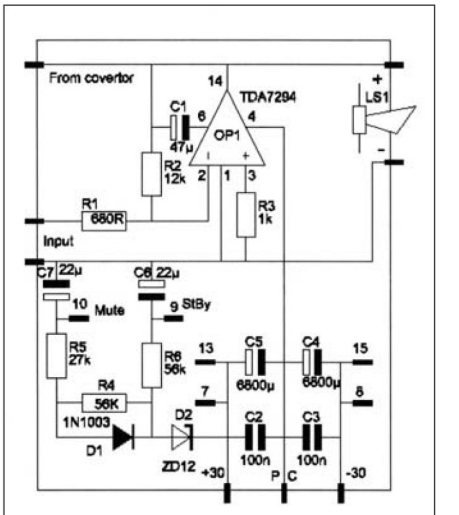

FIGURE 4: Booster circuit.

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the INA137 has a high common-mode resistance, and both inputs may vary almost ±40V before clipping occurs.

In this respect, there is no danger at all as the output of the following booster cannot change more than its supply voltage of ±30V DC. As with the first stage, only ±15V is needed for proper operation.

THE BOOSTER STAGE

The booster stage features a TDA7294 power chip, which is widely known as one of the best sounding chips of the lot. **Figures 3** and **4** show that the output signal from the INA137 should go straight to the input of the booster, to prevent hum and noise. This input signal is a perfect copy of the triode AC plate current; even at high frequency no phase shift occurs, which is a must for proper operation of the system.

The functioning of the booster is quite simple, but there are certain peculiar things you should know. As mentioned in my previous article, the plate current goes via the current converter to the positive terminal of the 8Ω speaker, which, in principle, should offset the DC speaker voltage, theoretically –0.2 × 8 = -1600mV DC. But the TDA7294 has a very high DC gain, so an offset of maximum -10mV is on the speaker terminal, and all the plate current goes through the output transistor of the TDA7294 to the +30V supply, but the loss in the chip is still only 6W.

But now the output stage of the TDA is strongly biased toward class "A," which is very beneficial for the sound. If the plate current of the triode varies ±100mA, then the AC input to the TDA7294 is 950mV, and with a voltage gain of 19×, the voltage on the speaker is 18V AC, which means a power output of 40W. The TDA chip has a standby and mute pin that gets a positive voltage from the +30V via zener D2; you can replace this with a switch. The mute function has a longer delay than the Standby when switching on, and is faster when switching off.

TRIODE FEEDBACK

Looking at the characteristics of the tube 6080WA, you'll find a very low plate resistance and, consequently, a plate current that is very sensitive to plate voltage vari-

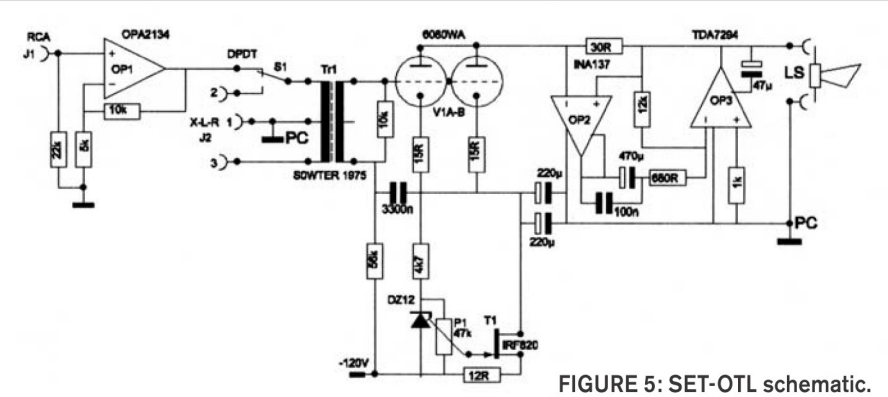

FIGURE 5: SET-OTL schematic.

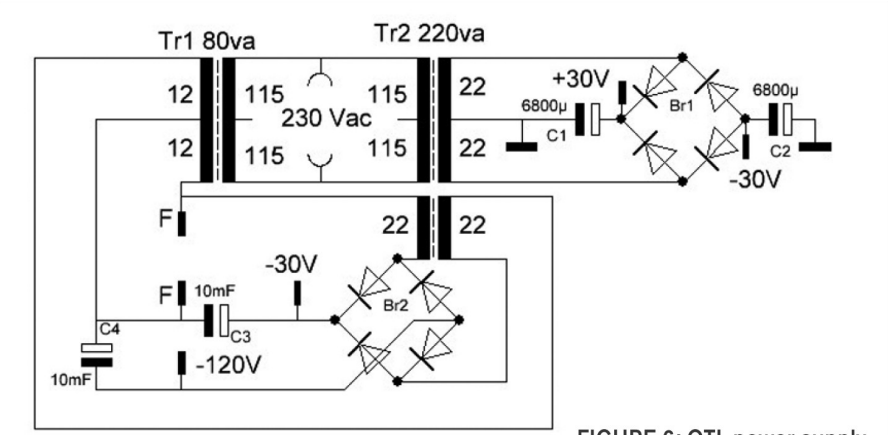

FIGURE 6: OTL power supply.

PHOTO 5: 19″ housing.

PHOTO 6: Inside the amp.

ations. So there is a strong feedback from the signal to the speaker to the triode current and hence the booster input signal.

Since the output of the booster TDA7294 is an exact copy of the triode plate current variations, the amplifier acts like one big triode. Changing the gain of the TDA7294 will change the sensitivity and sound of the amp. Increasing the gain gives more sensitivity and less triode sound. With less gain the triode needs to deliver more signal to the booster so the amp has lower gain, more triode sound, and a little more distraction (mostly second harmonics). The sound is absolutely fantastic, and it seems that there is a solid grip on the speaker as well. I was really very surprised with the improvement over the first OTL amplifier.

POWER SUPPLY

There is not much to say about the power supply, which you can make from standard parts. But it is important to know that the triode plate current flows to the +30V, and to avoid too much ripple the -30V supply should be part of the -120V supply as well. This is easy to do by just adding -90V to the -30V booster power supply.

I used a 220VA transformer to deliver 2 × 22V AC for the ±30V booster voltage plus 2 × 22V AC that, with the addition of 12V AC from the filament transformer, delivers the -90V—the positive side is connected to the -30V to make the -120V supply. If you fancy a delayed switch of the -120V for the triode, John Broskie of Glassware Audio Design can supply you with a nice switch.

SUMMARY

We took the following measurement at the output pins with an 8Ω speaker load:

Bandwidth at –0.5dB 25Hz to 45kHz

Bandwidth at –0.1dB 50Hz to 40kHz

Distortion at an output of 1W 0.6% second harmonics only.

Noise on speaker with no input connected about 2mV AC and 8mV DC.

aX

POWER SUPPLY PARTS LIST	
Br1, Br2	bridge, 400V 2A
C1	6800µ, Elco 40V
C2	6800µ, 40V
C3, C4	10mF, Elco 50V
Tr1	80Va, 2 × 12 sec
Tr2	220Va. 4 × 22V AC sec
INPUT CIRCUIT PARTS LIST	
C1, C2	100n, poly 100V
Op1	OPA2134, op amp
P1	RCA, socket
R1	27k
R2	12k
R3	5k6
S1	switch, DpDt
Tr1	Lowter 1973, line transformer
TRIODE CIRCUIT PARTS LIST	
C1	470n, poly 100V
C2	470µ, Elco 250V
C3	470µ, 250V
D1	ZD12, zener 12V
P1	47k, trimmer 10 turn
R1, R2	15R
R3, R4	10k
R5	100k
R6	12R
T1	IRF820, MOSFET 400V
V1	6080WA, double triode
BOOSTER CIRCUIT PARTS LIST	
C1	47µ, Elco 40V
C2, C3	100n, poly 100V
C4, C5	6800µ, Elco 40V
C6, C7	22µ, Elco 40V
D1	1N1003
D2	ZD123, zener 12V
LS1	connector, to LS
OP1	TDA7294, power chip
R1	680R
R2	12k
R3	1k
R4, R6	56k
R5	27k
CONVERSION CIRCUIT PARTS LIST	
C1	470µ, Elco 40V
C2-4	100n, poly 100V
OP1	INA137, line receiver
R1	27R, 2W

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The Essex Echo: Audio According to Hawksford, Pt. 2

Jan Didden continues his discussion about audio technology with Professor Malcolm Hawksford.

JD: Let's move to amplifier electronics, because one thing that comes across clearly from your publications is that you enjoy electronic circuit design.
MH: Is it that clear? But it is true. My first amplifiers were tube-based, of course, and I still have a certain fondness for them. Most were simple, first-order circuits, with some pleasant coloration usually added by self-induced microphonics and vibrations. Different manufacturers using different tubes even with similar circuits show up different issues, but they err benignly, so to speak. It is very seldom that a tube amplifier's sound can't be enjoyed despite its technical limitations; the errors tend to be quite musical.

JD: What triggered your interest in error correction (EC)?
MH: Peter Walker's *Current Dumping* concept did that. I thought it an extremely clever and elegant solution (still do), and a "thinking out of the box" amplifier design that was *en vogue* at the time. There are various ways of looking at Current Dumping, but I explained it as a combination of feedback and feedforward techniques. The clever bit, as I saw it, was that it allowed you to design a structure that didn't require infinite gain to obtain theoretically zero distortion over a fairly broad bandwidth. In a feedback amplifier, as you move up in frequency, the feedback decreases leading to increasing distortion.

In this (then) new concept, the feedforward path compensates for the loss of feedback with frequency, and in theory you can keep up the "zero distortion" over the audio band. Of course, it depends on what stage of the amplifier produces distortion. It started me thinking about some way to generalize the concept of combining feedforward (ff) and feedback (fb)—which, of course, is at the core of Current Dumping—and explore other trade-offs in ff and fb. As the most objectionable distortion in a power amplifier is generated in the output stage, would it be possible to locally correct that output stage so that the remaining distortion signals that are fed back from the output to the input stage would be much cleaner (i.e., devoid of output stage distortion) thus also contributing to lower input-stage distortion? As N (the uncorrected output stage gain) approximates to 1, the error tends to zero and this makes the difference (correction) amplifier much more linear as it only amplifies small signals, and this holds even when the output voltage swing is large.

The conceptual view (**Fig. 8**) made it clear that, in theory, combining ff and fb can completely eliminate the forward loop nonlinearity, without the need for infinite loop gain, simply by choosing suitable combinations of transfer functions *a* and *b* in **Fig. 8** providing (a + b) = 1. Practical ff or fb networks will most probably need to have some active components and will thus be at least first-order low-pass circuits. But, if the "*a*" network has a first order 1/(1 + sT) characteristic, you could make "*b*" a conjugate sT/(1 + sT), and the elimination of distortion independent of frequency still holds.

Now, for the feedforward component "*b*," there is the practical problem of combining the forward and feedforward signal in the output (power) stage, so that is less attractive. Therefore, one solution would be to use only the "*a*" fb path, as it is much easier to combine low-level signals at the amplifier input. Because you now can no longer compensate for the first-order rolloff, the full curative properties of the system break down at higher frequencies so zero distortion is out of reach.

Yet, employing this type of error correction locally in, for instance, output stages still has significant advantages. Such fast-acting local correction does a good job to linearize the output stage by one or two orders of magnitude and, as a bonus, give very low output impedance before global feedback is applied. I also showed that you can implement a correction circuit virtually without needing more components than those used for biasing, so it's essentially free.

The local loop does not impact stability much, so you can have your cake and eat it, too. You end up with a more linear power amplifier for the same parts investment

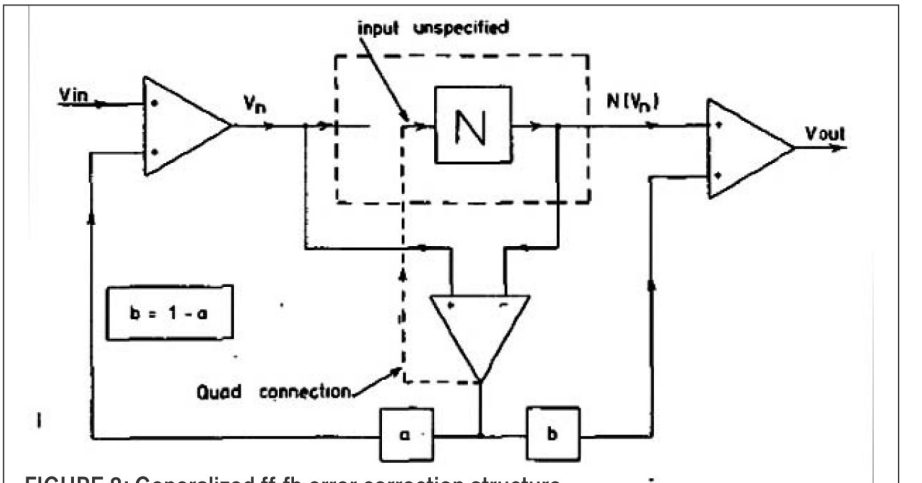

FIGURE 8: Generalized ff-fb error correction structure.

and that's always worthwhile. Bob Cordell had a very elegant implementation of this concept which I like very much[8].

JD: At one point there was a great discussion on diyaudio.com between Bob Cordell, yours truly, and other very smart circuit designers. The question was whether error correction is really a different circuit concept or whether it is another way of using negative feedback (nfb). That it was, to paraphrase evolutionary biologists, a matter of exploring the "space of all possible nfb implementations."

MH: Well, I guess that conceptually it is indeed a different way to apply nfb, but with some interesting different issues which also lead to more insight into this type of circuit. For instance, in **Fig. 8**, assuming that b = 0, then Vout/Vin = G = N/ (aN - (a - 1)). The target for Vout/Vin = 1, so now you can calculate the error function ε representing the overall input-to-output transfer function error, that is the deviation from "1," thus ε is defined as ε = 1 – G.

Substitution gives you ε = (a - 1)(N - 1)/(aN - (a - 1)). Now you immediately see that the error function has two zeros, i.e., (N - 1) and the balance condition represented by (a - 1). This succinctly explains the operation and power of EC, especially with near unity-gain output stages as you get two multiplicative terms in the error function which should both be close to zero. Half of the art of understanding and developing circuits lays in finding the right viewpoint!

JD: I know of at least one commercial implementation of what appears to be your EC concept, based directly on Bob Cordell's circuits, by Halcro. Presumably based on a patent by Candy, which came later in time than your publication.

MH: Yes, I am aware of that. At the time I sent Halcro my papers and wrote to them asking for some clarification, but never received a reply. So it goes. Anyway, life's too short to worry about such things. It's not my problem. Bob Stuart of Meridian Audio also used the circuit in his amplifier range for a period of time, which was most gratifying as he is a very gifted audio circuit and system designer.

There's analogy to error correction in the digital domain, and that is noise shaping. I wrote a paper with John Vanderkooy comparing digital noise shaping with nested differential feedback in analog circuits[9] and concluding that they can be seen as different views of similar issues! If you look at a first-order noise shaping configuration (**Fig. 9**), you see that, similar to EC, you take the difference between the forward block (the quantizer) input and output, which is the noise it generates, and feed it back to the input, properly shaped like $H = e^{(-sT)}$. Now, if you look at the noise shaping transfer function (1 - H), it looks very similar to the error reduction function of EC you showed before. So as you go lower in frequency, where the loop gain gets higher, the noise also gets lower.

Now this is a simple first-order case, but as you go to higher order noise shapers, your in-band noise gets lower at the expense of forcing more and more noise above the audio band. Now, if you put in a coefficient in (1 - H) of less than 1, then the reduction curve bottoms out at lower frequencies, so it is analogous to the bottoming out of your EC curve due to a less than 1 error-feedback coefficient. So, you could say that quantization noise shaping in sampled data systems is analogous to distortion-shaping in feedback or error correction in continuous signal systems. You often see that when the distortion is driven down by feedback or EC, it works for the first few harmonics at the expense of increasing higher harmonic components. Again, just like what we observe with noise shaping in digital systems!

You should look into the literature about Super-Bit Mapping (SBM). Michael Gerzon and Peter Craven in the UK worked on that as did Stanley Lipshitz and John Vanderkooy and also SONY. I well remember a rather heated argument between Michael and a Sony engineer during an AES convention some years ago! The idea with SBM is to apply noise shaping to a digital signal in the context of CD. Normally, with uniformly quantized and dithered 16-bit/44.1kHz LPCM, the

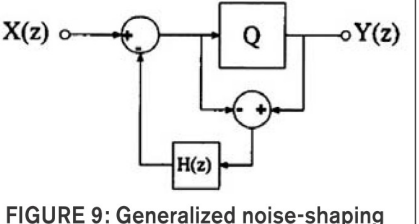

FIGURE 9: Generalized noise-shaping structure.

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Specifications:
HF Power Output: 30Wrms
LF Power Output: 80Wrms
THD: 0.03%
S/N ratio @ rated W: 90db
Input sensitivity: 1V
Input impedance: 22Kohms
4th order X-over: 2KHz AP-016A
2.7KHz AP-016B
3.5KHz AP-016C
Weight: 2.6Kgs (5.7lbs)
Dimensions W x H x D:
W: 137mm (5.4")
H: 218mm (8.6")
D: 81mm (3.2")
Cut-Out W x H:
W: 108mm (4.25")
H: 190mm (7.5")
AC Voltage: 115V / 230V

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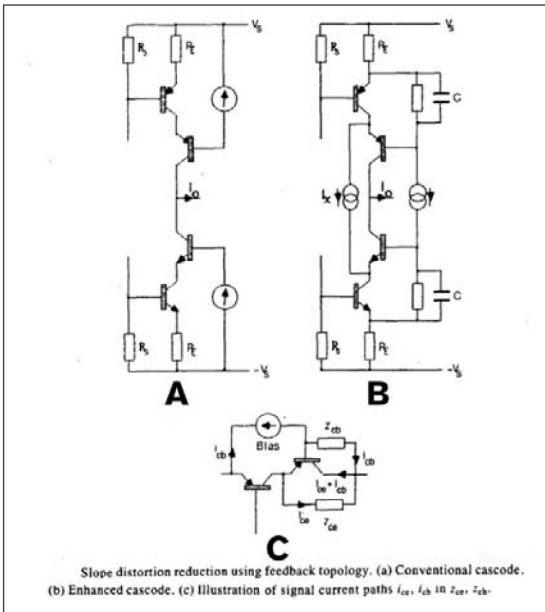
noise floor is essentially flat from DC to 22.05kHz.

Now, they asked, suppose we start with a 20 or 24-bit source, and we re-quantize and noise-shape the signal, can we somehow retain some of those additional bits of resolution below those 16 bits? Of course, the noise that you reduce in one part of the spectrum needs to go somewhere, and what SBM does is to decrease the noise in the mid band so you get perhaps 18-bit resolution in the frequency region where the ear is most sensitive.

The noise-shaping transfer function is designed to follow closely the Fletcher-Munson curves; consequently, the noise may rise by perhaps as much as 40dB at the very high frequencies, but because your ears are very insensitive in that area you cannot hear it. It is also important to realize that in a properly designed SBM system the noise is of constant level, and there should be no intermodulation with the signal. Also, the signal-transfer function is constant. So, provided that your DAC has at least 18-bit accuracy, you can perceive a subjective resolution of around 18 bit. And at its core, again, is a concept that you would recognize as an error-correction amplifier!

Your use of that AD844 current conveyor in your error-correction amplifier does remind me of a similar topology that I developed with two of my research students, Paul Mills and Richard Bews. This design, which led to the LFD moving-coil preamp, was published in *HiFi News* in May 1988. Richard subsequently developed this conceptual LFD pre-pre that used floating power supply circuitry by optimizing component selection and overall construction to achieve a very high level of performance. The reasoning behind the circuit is as follows: In a simple, single-ended emitter follower (**Fig. 11A**) the transconductance of the stage $G_m = 1/(r_e + R_E)$ where r_e is the intrinsic base resistance.

Since $r_e = 25/I_E$, you see that because r_e changes with signal current, this introduces distortion. You can improve on this (**Fig. 11B**), and now $G_m = 1/(r_{e1} + r_{e2} + R_E)$, where, for example, when r_{e1} increases, r_{e2} falls. There is not perfect cancellation because the transistors of the long-tail pair are effectively connected in series in the AC-equivalent circuit, but it is much more linear than the previous case. You can further improve on that with **Fig. 11C**, where complementary transistors are now effectively in parallel for AC, so the changes in the respective r_es due to signal current are almost perfectly complementary such that the transconductance of the combined transistors is almost independent of signal current; that is, the circuit is linear.



Slope distortion reduction using feedback topology. (a) Conventional cascode. (b) Enhanced cascode. (c) Illustration of signal current paths i_{ce}, i_{cb} in z_{ce}, z_{cb}.

FIGURE 10: Enhanced cascode concept.

If you plot the nonlinearity (as an error function) versus the value of R_E and signal current (**Fig. 12**), you see that there is a point, with very low R_E, where the **Fig. 11C** stage is almost perfectly linear. So this is a valuable property, but as you can see there are some challenges in biasing it, especially with those very low-value emitter resistors. However, you can rework the circuit to retain the linearity yet make biasing somewhat easier.

Another most important aspect of the topology is the use of truly floating power supplies because even if the supply voltage were to vary or to exhibit noise, there is no signal path linking to the RIAA impedance, as related currents can only circulate in closed loops. Consequently, power-supply imperfections are dramatically reduced, which is very critical in MC applications where small signals can be sub microvolt in level.

Under large signal conditions, you have transistor slope resistances and slope capacitances which are being modulated by the signal, and that's potentially bad news. Some people call it phase modulation, going back to something Otala brought up many years ago.

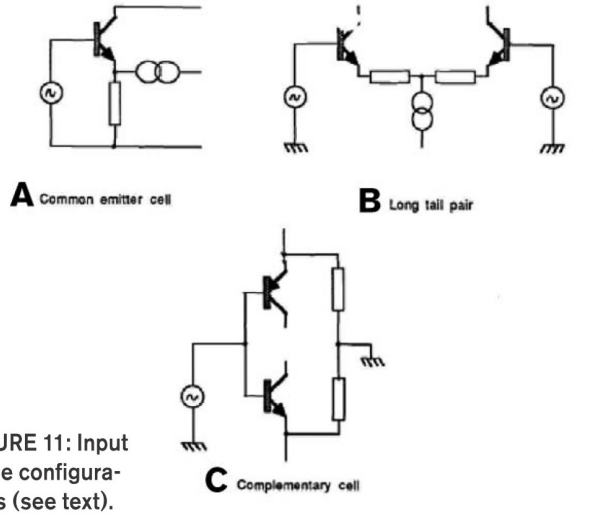
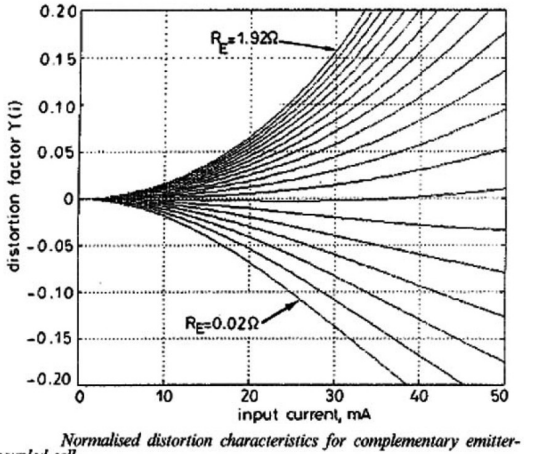


FIGURE 11: Input stage configurations (see text).



Normalised distortion characteristics for complementary emitter-coupled cell
Circuit data: I_{BQ} = 25mA; ΔR_E = 0.1Ω; input current 0–50mA

FIGURE 12: Input stage linearity versus RE.

It's more like a gain-bandwidth modulation, and I prefer to think of it as a time-domain modulation. For instance, in a feedback amplifier, this would slightly modulate the open-loop gain-bandwidth product and you can then calculate what it does to the closed-loop phase shift. It's like a signal-dependent phase shift, which manifests itself as jitter. It is analogous to a signal-dependent jitter, and it basically happens in all analog amplifiers. So, you have jitter in digital systems, you have jitter in I/V converters due to finite slew rate leading to slight modulation of the loop gain-bandwidth product, and you have these signal-dependent jitter-like phenomena in analog amplifiers in general, albeit that the modulation is time continuous rather than being instigated at discrete instants.

PHOTO 3: A younger Malcolm Hawksford showing off his speaker building skills.

You know, if you start to design a system, you need to have some sort of philosophy that drives you. For me, it is often the minimization of these timing errors, and I think that large-signal nonlinearity is less of a big deal than sometimes is believed. Most of the time you listen to low-level signals anyway, where linearity is very good. So then, you ask, what distinguishes one system from another, right in these low-level regions?

Now, I don't have any magic number, but let's assume that 100pS is the magic number for digital jitter, and suppose that you find similar numbers for what I call "dynamic timing errors" in analog amplifiers, the picture sort of comes together. It just *might* be that simple, open-loop circuits, while having higher large-signal level distortion, potentially have less of these timing nonlinearities, which could explain their very good sound. I would need to get the sums together, but it just might be possible that this is one of the reasons why people prefer those simple, low-feedback amplifiers. Especially in transistor circuits, where the transistor parameters themselves are modulated by changing voltage and current.

So having simple circuits that minimize these changes and are designed to minimize power supply influences clearly helps. Of course, feedback can help in many ways and there is no fundamental reason that a feedback amplifier cannot exhibit exemplary results, providing care is taken to minimize modulation of the amplifier loop transfer function.

Another example: When Paul Mills was still at Essex, he was working on an amplifier design using a cascode stage (**Fig. 10A**) that had reasonably low distortion. Then I told him, "Look, Paul, I will make one modification to your circuit that lowers the nonlinearity by an order of magnitude!" What I did was re-locate the biasing for the cascode to its emitter rather than to the supply (**Fig. 10B**).

It doesn't look like much, but it is a very significant change, and I can explain it with **Fig. 10C**. Why is the Z_{out} of a cascode not infinitely high and its distortion zero? It has to do with transistor slope parameters and their modulation with signal level.

PHOTO 4: Professor Hawksford and PhD student Adam Hill in the university audio lab.

You can see that an error current that is the difference between the ideal output (collector) current and the actual one is a result of the non-infinite impedances between emitter-collector and base-collector of the cascode transistor, where in **Fig. 10C** these two impedances are modeled by Z_{ce} and Z_{cb}. The modulation of transistor slope parameters with signal level I mentioned can be described as modulation of Z_{ce} and Z_{cb}. So, if you could find a way to prevent these error currents from ending up in the output (collector) current, then their bad influence would be eliminated.

Now, what is the effect of re-locating the bias to the emitter instead of the supply? For example, the i_{cb} error current now no longer comes from the supply but from the emitter of the top transistor. It is subtracted from its emitter current, which is basically the same as the cascode collector output current. So when i_{cb} is added to the cascode output current, it is no longer an error but makes up for the current that was subtracted in the first place! For i_{ce} a similar reasoning can be made. So the error currents now circulate locally in the stage and don't contribute to the output. It doesn't work perfectly, because there are some minor errors due to base currents, but it is, nevertheless, a huge improvement. The output impedance goes up typically by a factor of 10, and the distortion goes down by a factor of 10!

Note that it does not matter whether these error currents have a nonlinear relationship to the signal, as they do not contribute to the output current. This technique therefore works well in large signal amplifiers. I just picture this process in my mind, and I "see" what's going on, and then the solution pops up.

JD: You need to make the mental leap to model this modulation as an error current, and then find a way to shunt that error current away.

MH: Yes, indeed. There are some issues involving stability, as there is some form of regeneration in the circuit, but that's the gist of it. Now, I often wonder whether I would have seen that if I had

plugged it into a simulator and run a distortion analysis. I like to think that I might *not* have made that connection. I also believe that you should lay out the PC board, build your designs, and think about the topology at the same time. The days of a light box and black tape were great and very intuitive, very human. You move the layout around, changing this and that and in some way that connects back to the circuit again and you may then end up improving the circuit. It's an iterative process that can give you just that extra bit of quality or performance that you don't get when doing a sim and then saying, well, that's it.

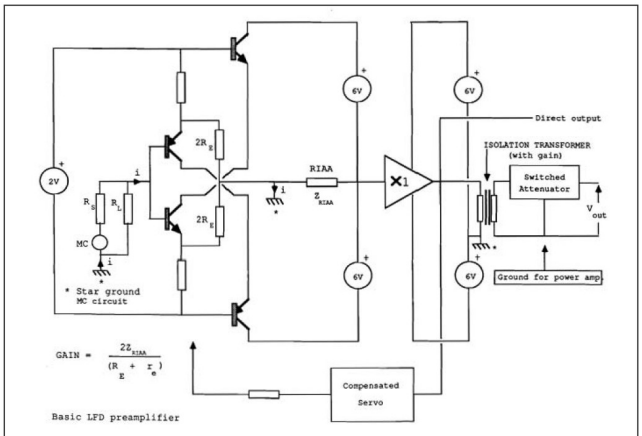


FIGURE 13: LFD preamp simplified diagram.

Anyway, this particular enhancement then appeared in my enhanced cascode paper[10]. Also, Richard Bews and I used this concept in the LFD preamp (**Fig. 13**), which, as previously mentioned, employed a true floating power supply system. And even if those batteries were to introduce some supply voltage nonlinearity, this doesn't show up in the output signal. There are no grounding problems because of the floating supplies. The floating-bias input pair is coupled to a cascode stage.

It's clear that any changes in that bias voltage do not have any influence on the output signal. So this will have high output impedance which drives current into the passive RIAA network to convert that current to voltage.

Now, if you look at which components determine the sound quality, it's only the input transistor emitter resistances and the components of the RIAA network. The cascodes don't do anything; the power supplies don't do anything, so it's an extremely linear circuit overall. And because it is only those few components, Richard was able to optimize component selection, ending up with a truly world-class preamp. Richard really is extraordinarily good at tuning and laying out circuits, and the battery-powered preamp worked extremely well. Also, this is why LFD Audio now enjoys almost cult status with its amplifier products.

JD: That **Fig. 13** circuit looks deceptively simple, but it is a very intricate circuit, isn't it?

MH: Yes, it is very simple, yet has a lot of interesting points: low noise, low distortion, almost no supply interaction, virtually no ground-rail current, very insensitive to transistor parameters, accurate RIAA correction, yet only a few active devices.

Often manufacturers have a good basic topology, but then they need to work in the power supply and grounding as well as the electrolytics and the other components in the signal path, and it all tends to blur the final sound. If you have many components in the signal chain, individual optimizations have relatively small impacts. But with this simple circuit, the components that determine the quality are few, and thus optimization has a relative large effect as well.

The absence of power supply interaction, however, is key to its performance. I find that at least as important as the topology itself, not only in preamps, but also in DACs and power amplifiers, for that matter. A lot of the differences between equipment in terms of clarity and cleanliness have to do with internal EMI issues and the power supply interactions and ground contamination.

JD: Well, we've already covered a lot of ground, but perhaps I can ask you about your views of switch-mode amplifiers.

MH: As you know, I've done a lot of work on Sigma-Delta (SD) modulation over the years. There is one proposal using an SD modulator driving an output stage with a pulse-density modulated signal. Now, the switching frequency would generally be higher than in the case of a PWM stage.

As you mentioned before, there is a basic problem with these types of circuits with EMI, and a higher switching frequency doesn't help. Do you remember our discussion with raised-cosine modulation in a DAC? Well, in this particular idea I used something similar. Instead of supplying the switching output stage with a stiff supply, you use a resonant supply synchronized to the switching frequency of the amplifier. The supply voltage would, in effect, be a raised cosine, so that at each switching instant the supply voltage would be zero, and would then smoothly rise toward the full value (**Fig. 14A**).

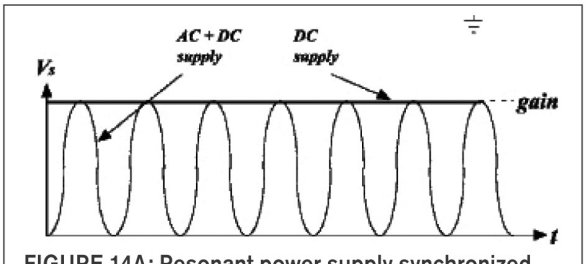


FIGURE 14A: Resonant power supply synchronized to sample rate outputs raised-cosine voltage.

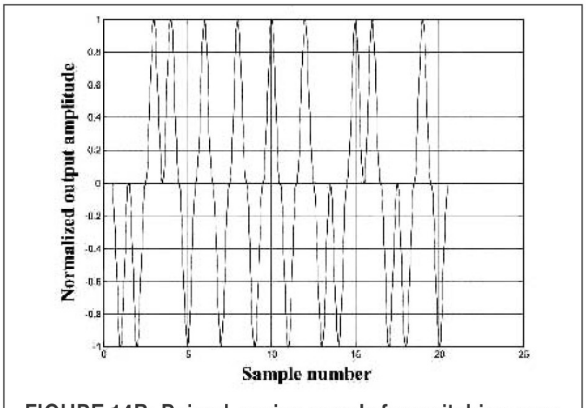


FIGURE 14B: Raised-cosine supply for switching amp dramatically reduces output signal bandwidth.

The result is that EMI problems are greatly reduced because the switching effectively occurs at zero voltage, and the harmonics are both lower in level as well as much lower in bandwidth. The output voltage of the amplifier is now no longer rectangular but somewhat sine-shaped (**Fig. 14B**). Switching efficiency of the output stage is im-

proved as well, and not only are those switches still either fully on or fully off, but because switching occurs with zero voltage across the device, power dissipation in the finite switching transition region is reduced. The average output level of this scheme is somewhat lower than a regular PWM amplifier, but that can be compensated for as described in the paper.

JD: Do you think that these switched-mode amplifiers can reach the quality levels of a good analog amplifier?
MH: Well, I've heard some commercial systems with B&O IcePower modules, which seemed to work really well, so I would say it's getting there, yes. It's an interesting technology, and even if the samples I've listened to were not always very low distortion, they did have a certain cleanliness and transparency to them. I'm not absolutely sure, but it may be related to the absence of low-level analog problems like dynamic modulation of device characteristics in an analog amplifier.

So, I'm fairly optimistic, also because it brings the digital signal closer and closer to the loudspeaker, skipping analog preamps and the like. Of course, you need to distinguish between "analog" switching amplifiers and "digital" switching amplifiers where the power amplifier is, in effect, the DAC. I have always been more interested in the latter class, especially the signal processing needed to achieve good linearity[11,12]. Just because an amplifier uses switching techniques does not necessarily make it a digital amplifier. This is an important distinction which is often misunderstood.

JD: Bruno Putzeys, a well-known designer of switching amplifiers, maintains that switching amps are analog amps: they work with voltage, current, and time—all analog quantities.
MH: Indeed. So, there are still a lot of problems to overcome, but they have a philosophical "rightness" about it.

JD: Not the least because of the high efficiency!
MH: Of course. And even if you want ultimate quality, running your amp in class-A with a 500W idle dissipation doesn't solve your quality issues either. There's much more to amplifier quality than just the choice between class-A or class-AB/B topology. An AB/B amplifier, properly implemented, with attention to all the often misunderstood issues of biasing, power distribution, grounding, and so forth, can sound so good that there is nothing to be gained by going to class-A. It's better to go for a simple system, with as few stages as possible because an additional stage cannot fully undo any damage done by a previous stage.

Now, a great-looking box with lots of dials and lights certainly may play music well, but for ultimate quality, get the best DAC you can afford (preferably a networked DAC linked to a NAS drive!), followed by a passive volume control and a great power amplifier and, of course, keep the cables short. Nothing can beat that, in my opinion.

JD: Professor Hawksford, thank you very much indeed for many hours of your time, for most interesting and illuminating discussions. In particular, I was intrigued by the correspondence between seemingly disparate phenomena, like noise shaping versus error correction and jitter versus analog phase modulation. I hope this will inspire readers to do their own experiments and come up with yet other interesting configurations. *aX*

REFERENCES

8. A MOSFET power amp with error correction, presented at the 72 AES Convention, Anaheim, 23-27 Oct. 1982, revised 25 Jul. and 27 Oct., 1985. Also on www.cordellaudio.com/papers/MOSFET_Power_Amp.pdf

9. Relationships between digital noise shaping and nested differentiating feedback loops, presented at the 93rd AES Convention, San Francisco, 1-4 Oct., 1992, revised 9 Nov. 1999.

10. "Reduction in Transistor Slope Impedance-dependent Distortion in Large Signal amplifiers," *J*AES, Vol. 36, no. 4, April 1988.

11. "Dynamic model-based linearization of quantized pulse-width modulation for applications in DA-converters and digital power amplifier systems," *J*AES, Vol. 40, no. 4, pp. 235-252, April 1992.

12. "Linearization of multi-level, multi-width digital PWM with applications in DA conversion," *J*AES, vol. 43, no. 10, pp. 787-798, October 1995.

Note: All papers referenced here and many more are available at www.essex.ac.uk/csee/research/audio_lab/malcolms_publications.html

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Ascend Acoustics Sierra-1 Speaker

By Howard Ferstler

	Ascend Acoustics, Inc.
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	Fax: (310) 388-1500
	www.ascendacoustics.com Price: $848 per natural color pair; $898 in piano black; $40 additional for magnetic shielding; $26 flat-rate shipping cost.

In my review of the small, stand-mountable Ascend Acoustics CBM-170 loudspeaker system (*$ensible Sound*, Dec./Jan. '03), I stated that it was a terrific speaker at the asking price of $348 per pair (today's rate, with $20 flat-rate shipping), or even at twice that. I still feel that way (and perhaps more so, because the new CBM-170SE version has replaced the Audax tweeter with a more refined SEAS model) but now we have the company's new, more upscale, Sierra-1 model, which is similar in size, much more refined in terms of cabinet finish, and just happens to sell for a tad more than twice what the CBM-170 cost.

The CBM-170—for all of its manifest positive acoustic attributes—was still a somewhat basic-looking satellite-style speaker system. It had—and still has—a very advanced crossover network and a pair of drivers that are bang-for-the-buck items in the extreme. The system was "basic," in that the enclosure had a nice standard black finish, but hardly triggered the "wow" factor based upon its looks.

> "This is one good-looking small speaker that can cosmetically hold its own with the best."

FEATURES

This is not the case with the Sierra-1, which will elicit a favorable wow from just about anyone (including spouses). This is one good-looking small speaker that can cosmetically hold its own with the best.

The cabinet is a tad more than 14″ high, 7.5″ wide, and just 10.5″ deep (including the grille). It is light, too, coming in at only 19 lbs. The drivers include a 1″ dome tweeter (with a shallow, soft-surface wave guide) and a 5.25″ mid/woofer, with its low-end output augmented by a flared port on the backside of the enclosure.

Initially, the crossover network is electrically second order both in the high-pass and low-pass sections, but ultimately, in order to take the characteristics of the drivers into consideration, it attenuates as a fourth-order network. The result allows the system to better simulate a point source at all frequencies, even in the vertical plane. Indeed, the design allows you to place a tweeter-reconfigured version of the system (with the tweeter's wave-guide baffle rotated 90°) horizontally and use it as a uniform-dispersing center-channel speaker.

The cabinet backside also includes two very high quality, gold-plated five-way binding posts, which are considerably wider apart than usual (even wider than typical European-spec hookups), because designer David Fabrikant thought that wider spacing would better allow hookups from the assorted super-duper cable options available these days. Obviously, double-banana connectors will not work, but single bananas work fine, as will spades, pins, and tinned and bare-wire ends.

The speaker is rated as an 8Ω load, with a pretty standard 87dB sensitivity (2.83V input measured at 1m), and the minimum recommended power is 45W per channel. The unit supposedly can handle unclipped, momentary peaks of 400W, which attests to a robust pair of drivers, and promises reasonably high average output in small- and medium-size rooms. The system comes with a seven-year parts and labor warranty, by the way.

Oh, yes, that "wow factor" I mentioned previously. Well, that involves a cabinet made entirely of bamboo. Yep, bamboo. And this bamboo box is beautifully done, indeed, with a perfectly smooth finish and a thick cabinet stiffness that holds its own with the best.

Bamboo is a terrific material for speaker cabinets, as well as floors, not only because of its strength and beauty, but also because of the ecological advantages. It grows as fast as kudzu and using it just has to save wear and tear on our forests. The downside is that it is expensive and difficult to assemble into cabinet form (over 200 strips of ¼″ bamboo were used to build a cabinet ¾″ thick), but with an upscale speaker like this—and believe me, the Sierra-1 is an upscale speaker—that is no big deal.

MEASUREMENTS

I first ran a standard set of room/power curves in my main room. The results (**Fig. 1**) show exemplary response from 500Hz on up to above 12.5kHz. This is above-average reverberant-field performance, and is notable for no serious dip artifact in the mid-to-tweeter crossover range. The gradual, and typical, rolloff above 12.5kHz is of no audible significance.

Between 500Hz and 200Hz (a boundary-dependent region where reverberant-field response curves can sometimes give false impressions of speaker quality), the response becomes attenuated somewhat, with a dip centered between 300 and 400Hz. This is partially due to the "Allison Effect" suckout phenomenon, which occurs when a woofer interacts with a large room boundary (or boundaries), and is difficult to eliminate completely in any normal room.

At some point within the bass range, the rarefaction part of the reflected signal (the negative side of a wave) arrives back at the driver at just the right time to partially null out the compression part (the positive side of the wave) generated a moment afterward. The impact is gradual and not abrupt, because a wall boundary is not the same distance from the woofer over its whole area, so you get a slope instead of a notch. Because of the time it takes for the signals to be bounced back to the source, the frequency where this happens to the maximum extent will depend upon the distance to the boundary—or boundaries, because you will have more than one in typical rooms. The closer to the boundary the speaker is located, the higher in frequency the cancellation null will exist.

You can counter the effect in either (or both) of two ways. One is to locate the speaker so that woofer-driver distances to room boundaries (floor, front wall, and sidewall) are different enough to keep the three primary cancellation nulls from all stacking up at the same frequency. The second is to employ a good equalizer to level out the dip.

Note that a good equalizer should not need to apply more than 5 or 6dB of correction. Any more than this and the reverberant-field and first-arrival balance of the signals involved will be screwed up. (Incidentally, the danger from overkill equalization also exists when correcting for other, more speaker-related aberrations.) Not all of the dip shown here is attributable to cancellation artifacts, however. Most of those should exist at somewhat lower frequencies that have here been counterbalanced somewhat by the bass elevation culminating in the peak at 80Hz (more later), and it is possible that the speaker itself has a slightly attenuated output in the dip area.

For most of my equalization work, I use Rane THX-22 or THX-44 combination one-third-octave/parametric units, and with the Sierra-1 one of those equalizers absolutely flattened out that dip with only minor slide work. Note that I have yet to find a speaker system that did not benefit at least somewhat from the use of a decent equalizer, including the Dunlavy and NHT speakers I had installed in my other systems for years. Even my own Allison IC-20 main system units benefited. This is the case even if the corrections are only applied to areas other than those related to boundary cancellations.

The curve (**Fig. 1**) illustrates two plots in the bass range. One, the solid line, involves the speaker operating full bandwidth down there, and you can see that there is that previously mentioned peak at 80Hz, with a rolloff below that point. In other rooms the peak will shift somewhat, but I believe that it is pretty much inherent in this system, and most other small satellite speakers designed for full-range operation that I have reviewed exhibited similar peaks in the bass range. Even the large, $2500 a pair Dunlavy SC-II systems I reviewed had the artifact, as did the smaller but still equally expensive Waveform MC satellites.

It is just about impossible for a full-range, small-woofer, two-way speaker to have decent low bass without including an elevation of this kind at a slightly higher-up frequency. It goes with the territory, although boundary-related dips a bit higher up in frequency tend to make

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the peak more pronounced than it really is in practical terms. The peak will rarely show up with an on-axis anechoic plot, but with a curve run in an enclosed space *room gain* will emphasize things that will never show up in a dead-space measurement, and room absorption will attenuate output somewhat above the bass range. Consequently, slopes displaying characteristics such as this in real listening rooms are not uncommon, even with speakers that are exemplary under anechoic test conditions.

Of course, room contributions vary, so it is important to look at other curves. With this curve the slight leveling off in frequency response down to 50Hz probably involves the contribution from the port, and the comfortable low-end reach of this speaker is about 45Hz. Test sweeps had it delivering audibly significant bass levels down to 35Hz, but with also audible port-noise doubling at anything above near-whisper levels. You can only squeeze so much out of even the best 5.25″ woofer, even with the assistance of a well-designed reflex port.

The dotted additional line within the bass range plot shows the impact of a small SVS SB12-plus subwoofer. When I used the sub with its on-board, electronic crossover network, its 80Hz high-pass circuit had only a slight impact on the Sierra-1's 80Hz bass peak. However, when I employed an outboard Paradigm X-30 electronic crossover, selected its 150Hz high-pass option, and then carefully dialed in the low pass at 70Hz, I got the curve in **Fig. 1**. With that crossover and careful speaker placement, the curve was very good, and when I used the Rane THX-22 equalizer to deal with what was left of the suckout dip, the curve was right up there with the very best speakers I have worked with in the past. In a proper environment, and with a bit of equalization and sane placement, I believe these speakers will hold their own with the other best systems available, at least up to intelligent listening levels.

Some individuals do not like subwoofers, so I gave the system an additional run with the sub disengaged, but with the THX-22 still in operation. With some easy twiddling I flattened the dip and pulled down the room-gain emphasized 80Hz peak a few dB and got a sensational curve that lacked only deep-bass extension. While some might prefer to have the peak completely eliminated, it is best to have some bass elevation below about 100Hz, because in most rooms, with most recordings, this results in a better sense of acoustic realism.

Even without outboard EQ, those who prefer to work without subwoofers will get the kind of mini-monitor performance from the Sierra-1 systems that some knowledgeable enthusiasts dream about, even if the package will not show off demo-grade organ works or happily handle the pyrotechnics of home theater. For the latter, I definitely advise adding a decent subwoofer, as well as the center-speaker version of the Sierra, plus some good surround speakers. Ascend's own CBM-170 models are more than adequate for however many surround channels you care to employ. The combination would make a home-theater and surround-music package that would be second to very few other systems available, at least in any room short of mansion-home size.

LISTENING TESTS

Measurements out of the way, I proceeded to do some serious listening. I kept the SVS subwoofer and X-30 crossover involved, and became very impressed with what this combination could do. (Note that you could get results just as good or better with one of the bigger SVS subs available, or with any number of other good subs built by outfits such as Hsu Research, Paradigm, or Velodyne.) These are premium-grade speakers, to say the least, and with a good sub they will deliver the full-bandwidth goods.

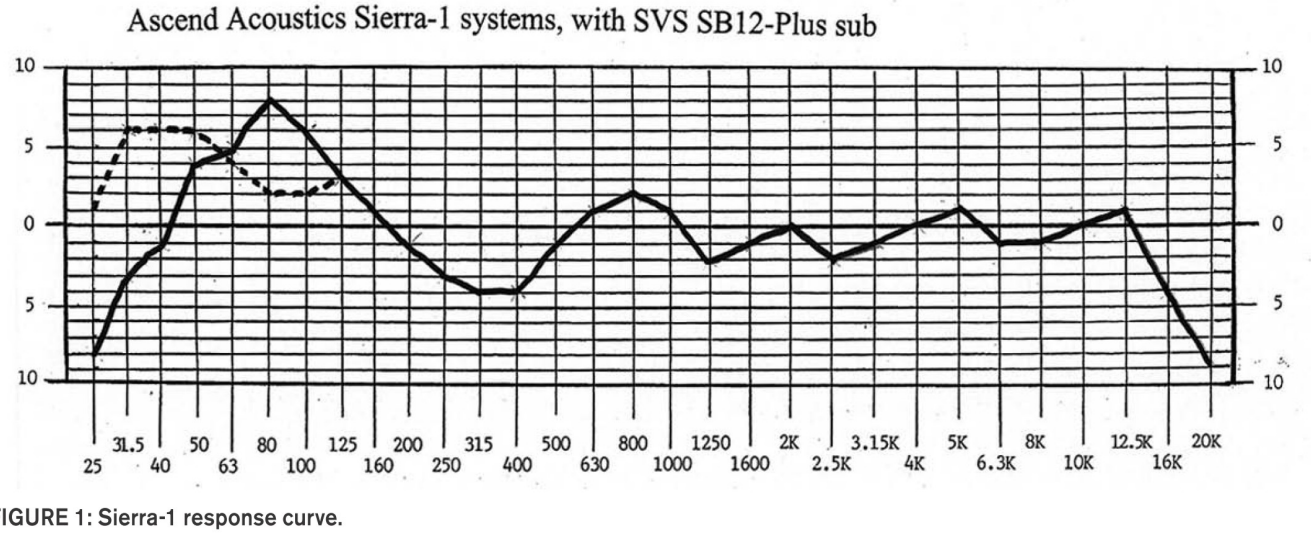

FIGURE 1: Sierra-1 response curve.

Normally, I avoid level-matched A/B comparisons between review items and the large, 100-pound each Allison IC-20 units in my main system—located in a 3,400ft^3 room roughly 18.5 × 22′. The tall, triangular-shaped cabinets and ultra-wide dispersion characteristics of the Allison models simply will not allow large, box-shaped speakers to be placed close enough to them to do fair comparisons. However, the Sierra-1 units are so small that I could successfully place them on slender stands, 4′ out in front of the against-the-wall Allison systems, and staggered a tad to the side AB/AB style to keep soundstage widths similar.

With the Allison units not crowded from the sides or blocked up front, the Ascend models were in a position to not chop up the Allison soundstage during the comparison. While the big Allison speakers, with two 10″ woofers apiece, are more than bass potent enough to match, and even surpass, the small SVS SB12-Plus subwoofer at normal levels, none of the recordings I auditioned probed the low-bass depths. I was interested mainly in comparing soundstaging width and depth, detail, and focus.

During this session I was happy to discover just how good the Ascend speakers performed in that somewhat large room. One work that I love to listen to is Händel's *12 Concerti Grossi* (op. 6), performed by the I Solisti, Italiani, a three-disc set on the Denon label (81757 6705), recorded back in 1990. In terms of engineering perfection, it is one of the best small-ensemble baroque transcriptions I have ever heard, and the performance is outstanding, too. Overall, the Sierra-1 and subwoofer combo sounded a tad thinner in the mid-bass range (that measured mid-bass dip, which the IC-20s, due to their cabinet and woofer arrangement, simply do not exhibit), with the big Allison units (ten drivers in each cabinet, mounted in dual, angled-panel MTTM vertical arrays) having a generally richer, somewhat more spacious sound. However, above the dip range the Sierra-1 systems were competitive, particularly when it came to violin string sound and overall clarity.

My usual comparison CD is the Delos *Engineer's Choice* disc (DE 3512), engineered by the late John Eargle a number of years ago. While the excerpts are not exceptional in themselves, they are at the very least very good, and it is easy to use the disc to do A/B speaker comparisons with a variety of instrumental sounds. During the comparison session it was evident that the Allison systems had the edge when it came to spaciousness, soundstage blend, and mid-bass richness. However, in some cases, notably those involving centered soloists and violin string tone, the tighter focus and clarity of the Ascend units had them holding their own.

FINAL ASSESSMENT

As an experiment, I replaced the IC-20s with the Sierra-1s in the main system itself. After setting the balances and reconfiguring the main channels to have the normal Velodyne F1800RII subwoofer feed, I then listened to a variety of materials with assorted Yamaha hall-simulation modes engaged, courtesy of a RX-Z1 receiver.

Yes, the surround speakers were not Ascend items (they were wall-mounted, very wide-dispersion Allison Model Fours), but with orchestral materials the sound was still outstanding. The surround channels delivered a nice sense of space (which, no doubt, speakers such as the Ascend CBM-170 systems could also do), and the ability of the Sierra-1 units to present a nicely focused phantom-center image resulted in the best of both worlds: spaciousness combined with soundstage precision. (Note that when listening with my more diffuse-sounding IC-20 systems for pleasure, in addition to engaging the surround-ambiance circuitry, I also engage the center steering for my six-driver, home-built center speaker, and so I get the best of both worlds then, too.)

The next day I moved the Sierra-1 units to my middle system (in a 2,900ft^3 room, roughly 17 × 22′), where I at first did a level-matched comparison between them and the two 85-pound-each Dunlavy Cantata systems running full bandwidth. To level the playing field in the bass range (where the Cantatas would obviously have a clear low-bass advantage if the small Sierra-1 speakers were themselves running full bandwidth), this time I used a Hsu VTF-2 subwoofer with the Ascend models. The Sierra-1 units were on 28″ stands, to keep the tweeter heights similar. The spacing arrangement was still AB/AB, which kept the soundstage outputs equally wide.

The source materials were the same as before, and in this case the Sierra-1 units held their own very well. With some materials, such as the guitar excerpts on the disk, as well as massed violins, it was difficult to tell the systems apart. The Cantatas do not have much of a mid-bass suckout notch, so they were a bit richer sounding than the Sierra-1 units, and they also exhibited a bit better front-to-back depth with some tracks. This varied, because different recording techniques were used with each track, with the Sierra-1 units sometimes appearing superior to the Cantatas.

With some baroque orchestral ensemble sections and the quartet selection of the Delos disc, it was difficult at times to tell which pair was playing, which says a lot for proper level matching when doing A/B comparisons. With the centered vocal selection the Sierras were as tightly focused and controlled as the Cantatas, and remember that a tight central image is one of the Cantata's supposedly strong points. I think that many buffs would be hard pressed to choose which system package was best, at least with a good subwoofer aiding the Sierra-1 pair.

Finally, I replaced the Cantatas with three-way NHT ST4 systems (which cost $1000 a pair when I reviewed them in 2002). I ran the Sierra-1 units full bandwidth, without subwoofer assistance. With the systems side by side, and with the Ascend models on 18″ stands to keep

the tweeter heights similar, it looked like a David and Goliath contest. The levels were matched carefully, and the comparison began.

The contest was closer this time. The guitar sections sounded nearly identical, as did the small- and large-ensemble orchestral works. The Sierra-1 systems were a tad more spacious sounding with some tracks—no doubt because of their smaller, wider-dispersing midrange-handling drivers (5.25″ versus 6.5″). The upper midrange was also a bit more detailed with the Ascend models, perhaps because of their slightly flatter response in the broad mid-to-tweeter transition region.

The NHT units were a tad richer sounding due to their not having quite as much mid-bass suckout as the stand-mounted Ascend units, at least as positioned in this room, but overall I would call the contest a draw. Well, almost a draw, because with the organ-demo track the three-way ST4s, with their 8″ woofers, were the clear winners in the low-bass sweepstakes. The ST4s had fewer (but still easily audible) doubling artifacts below 80Hz, in spite of their almost equally steep rolloff—their reflex port contributions notwithstanding.

Actually, neither pair of systems could get to the lowest octaves decently, so I suggest using a subwoofer with either pair for pipe organ or home-theater operations. With the centered vocal selections, the Sierra-1 units were a bit better focused, but with clarity essentially the same.

I also tried the Denon Händel material and found the Sierra-1s to have a slight edge in spaciousness and upper-midrange clarity, with the ST4s having a slight edge in mid-bass richness. These differences were so subtle that I am pretty sure the two speaker pairs could each be moderately equalized to sound nearly identical.

Finally (and just for kicks), I played some tracks from Dire Straits *On Every Street* (Warner 26680). Here, I again judged this to be a near tie, with the Ascends doing better at keeping the centered soloist tightly focused from the sweet-spot listening position, and with the NHT models delivering a bit more richness and bass depth. Both system pairs would do better with the assistance of a good subwoofer with ultra-punchy material of this kind.

Overall, the Ascend Acoustics Sierra-1 speaker systems were exemplary performers. Running full bandwidth, they managed to hold their own with the much larger, floor-standing NHT ST4 systems. With proper subwoofer assistance they did equally well against yet even larger Dunlavy Cantata systems, and would probably then outclass non-subwoofer-assisted ST4s. This would be the case with even a small sub such as the $300 Hsu Research STF-1.

With a good subwoofer backing things up, I would not hesitate to use the Sierra-1 speakers in any small- or medium-size listening room, including the one once occupied by my Cantatas. Their superb cabinet finish is a bonus that should endear them to spouses who think that most speaker boxes are ugly. These are not. ***rr***

In celebration of the return of *$ensible Sound* equipment reviews as the Reliable Reviews section in *audioXpress*, readers can enjoy 10% off a pair of Sierra-1s. Please call 949-366-1455 and mention *audioXpress* to receive the discount offer.

CONTRIBUTORS

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OPPO DV-980H Universal Player

by Tom Lyle

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I offer these brief, mostly subjective comments regarding the sound of this player as a supplement to the commentary by David Rich in *$ensible Sound* (Nov./Dec. '07, p. 10). I am speaking here only of the unit's audio performance playing CDs, SACDs, and DVD-As through a two-channel system.

FORM AND FUNCTION

On the CD-Check disc (Digital Recordings) it read levels 4 out of 5 of the red book tracks without error, which is outstanding. My "reference" Arcam DiVA CD 192 ($ 1499) could not read past track 2. The OPPO did not reject any disc fed it.

This player has a very slim (in height) design. The front panel is pretty bare, with almost all its functions controlled by the remote whose layout is just okay. There are a lot of buttons for a lot of functions. The disc play, stop, and pause buttons are larger than most of the others, so you can quickly learn basic operation. A pleasant surprise was a volume control, which is a nice feature.

During my rather long audition period, I found that its red book CD sound was more than acceptable. If you believe that there is little difference in CD players regardless of price, then this player should suit you just fine. For the rest of us, its treble sounded a little tipped up when directly compared to the Arcam. Its bass did not go as deep, nor was it as authoritative. When performing A/B and informal A/B/X tests versus the Arcam, I found the bass a tad deficient, and the upper-bass, too, was lacking a bit.

The last Nick Cave CD I bought (not including his *Grinderman* project), 2004's *Abattoir Blues*, is not the best recording in the world. Nor is it the last word in low-end power. It needs all the help it can get.

But otherwise, especially in the all-important midrange, the disc sounded very good. The instrumentation becomes pretty dense at times—there are all sorts of acoustic instruments mixed with the standard rock band of guitar, bass, organ, and drums. But the vocals are front and center, and are reproduced very realistically. The OPPO sounded very close to the Arcam in the midrange, and if judged solely on its midrange performance, I think there is very little doubt that you could not tell one player from the other.

Some might say that the transients of the OPPO's treble were sharply drawn, which might lead you to think that the high frequencies were being spotlighted unnaturally. But after living with the player for a while and frequent comparisons to the very neutral-sounding Arcam, I determined that it just ended up sounding a bit brighter. As usual, I'm exaggerating to make a point—the Oppo wasn't wimpy sounding or anything, and judged on its own it sounded perfectly fine. When all is said and done, the differences between the players were small, but I definitely could hear those differences.

STELLAR PLAYER

The OPPO's reproduction of SACD and DVD-A discs was a completely different story. I now need to take back *everything* I said about its sound in the previous paragraphs. An SACD or DVD-A in the OPPO blew away the performance of the Arcam in red book mode on every disc I played. On the SACD of the 1970's beloved *Future Days* by the German band Can, Holger Czukay's bass and Jaki Liebzeit's drums shook the room with extremely well-balanced bass. The soundstage was also impressive, and the separation of each instrument in space was extraordinary. I could "see" into the recording session, at the same time separating the live to two-track performance from overdubs.

One thing I love about SACD versus regular CD is its seemingly limitless treble extension. The OPPO flaunted this quality. The treble on everything from Damo Suzuki's natural vocal sibilants to Jaki's cymbals (along with a good measure of tape hiss) sounded as though I was listening to the analog master tape.

The Beatles' *Love* is one of my favorite CDs at the time of this writing. The plain vanilla CD is impressive enough. George and Miles Martin's re-mix of songs I've heard ten thousand times before brings these Beatles recordings into the 21st century; I get shivers (in a good way) every time I listen to it. The stereo tracks of the DVD-A bring it quite a few notches above this—the OPPO elevates this disc to new heights. If only the original albums from the Beatles catalog were remastered in a hi-res format! (But I digress.)

At this price, and considering its compact size, you can justify the purchase of this player solely for use as an SACD and DVD-A player. It's that good.

Manufacturer's Comment:

On behalf of all staff at OPPO Digital, we thank Mr. Lyle for this comprehensive and insightful review. Mr. Lyle apparently spent a lot of time and effort with a lengthy audition period using the DV-980H Universal Player in a two-channel system. Although the DV-980H is primarily designed to be an affordable up-converting DVD player, we put a significant amount of effort into optimizing the audio performance.

With the ability to play DVD-Audio, SACD, Redbook CD and its affordable cost, we hope to introduce more users into the wonderful world of high fidelity audio and high-resolution audio formats. This review is an indication that we have achieved this goal. The experience and know-how from the DV-980H have been migrated into our new BDP-83 Blu-ray Disc player, which also supports CD, DVD-Audio, and SACD with audio performance as one of the design goals. Thanks again for the great review, and enjoy the music!

Jason Liao
VP Product Development
OPPO Digital, Inc.

rr

Crib Death

Our birth announcement for a new English-manufactured tube in the November issue ("New Approach to Tube Design," p. 6) became a victim of the worldwide financial downturn. Micro Tech Solutions of Blackburn, England, has declared bankruptcy (www.blackburntech.com). The news arrived after the issue of *audioXpress* had gone on press. Deep sadness prevails in the town of Blackburn, since the old Mullard plant had been a major employer in the town with more than six-thousand employees at peak production in 1961. The remnant company was still a minor employer as a supplier of CRT guns even though that technology is now disappearing. This is nothing new in Blackburn, which was once the largest source of cotton cloth in the world.

At first glance this may look to most people as just another tragedy of the downturn which is closing businesses of all sizes everywhere. I am told that Teradyne in Boston, one of the world's primary chip testing companies, laid off half its workforce after one quarter in 2009 produced no sales at all, none. But this closure in England snuffs out a possibility that had the Blackburn enterprise succeeded, it would have produced not only 12AX7/ECC83 type triodes, but would have been followed by the 12AX cousins, as well as EL34s, EL84s, and eventually KT88s, even possibly the GZ34s with their slow turn-on time.

Those enterprising few fortunate enough to have paid their £50 for one of the new E8133C tubes have posted glowing, incredulous reports of reduced hum levels and quality of the sound as "the ultimate 'tubeness," to quote one (http://www.head-fi.org/forums/f5/mini-review-blackburn-microtech-e813cc-vacuum-tube-443651/). Other sources have reported problems of microphony, despite high fidelity in reproduction.—ETD

reliable reviews

Reliable Reviews

More Mono Matters

By Tom Krehbeil

Here's an idea for headphone manufacturers. I'm offering it free of any obligation.

We need a music-quality single on-ear phone or in-ear bud that uses a stereo plug and blends the two channels into a mono signal for one-ear listening. I know this because I see MP3 player-toting office workers putting one earpiece in place for listening and tucking the other one into the neckline of blouse or shirt.

They're solving a problem as best they can: listening with both ears blocks conversation and makes using stereo earphones extremely inconvenient in a work (or, for that matter, social or recreational) environment.

That ad-hoc one-ear solution has an obvious flaw. The listener must do without the music carried by one stereo channel. Mono earbuds and mono earphones exist, but they don't combine two stereo channels. A few sport three-contact stereo plugs, but those may or may not have a connection between the ring (right channel in a stereo set) contact and the driver. One website, in fact, specifically states that for mono use the ring contact is left unconnected.

So I believe that there's a place in the market for a proper music-quality stereo-to-mono earphone or earbud.

> "There's a place in the market. . . for a proper music-quality single-channel earbud or earphone that has a fully connected stereo plug that feeds a combined signal to the one driver."

In the meantime, you can accomplish pretty much the same thing by inserting a stereo-to-mono 1/8" headphone adapter between your stereo phones and the jack on your personal audio device. Radio Shack's item number 274-374 for $2.69 would do the job.

With such an adapter in place, you'll get a combined left- and right-channel signal from the left earpiece, and the right earpiece will be mute. It will, however, still be hanging around and will need the usual neckline tuck. Or, you could just snip it off.

CAVEAT: Since drafting this piece a year or so ago, I've found at least one MP3 player that craps out when the channels are combined at the jack. It mutes or quits, probably an impedance mismatch issue of some sort. No damage was done to the player or to the earbuds, but a manufacturer who decides to design a stereo-to-mono solution may need to do more than simply parallel the channels in the plug to guarantee full compatibility with all MP3 players. (Portable CD players seem to be immune, but who uses those these days?) *rr*

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Audio Aid: Tube Amp Mods

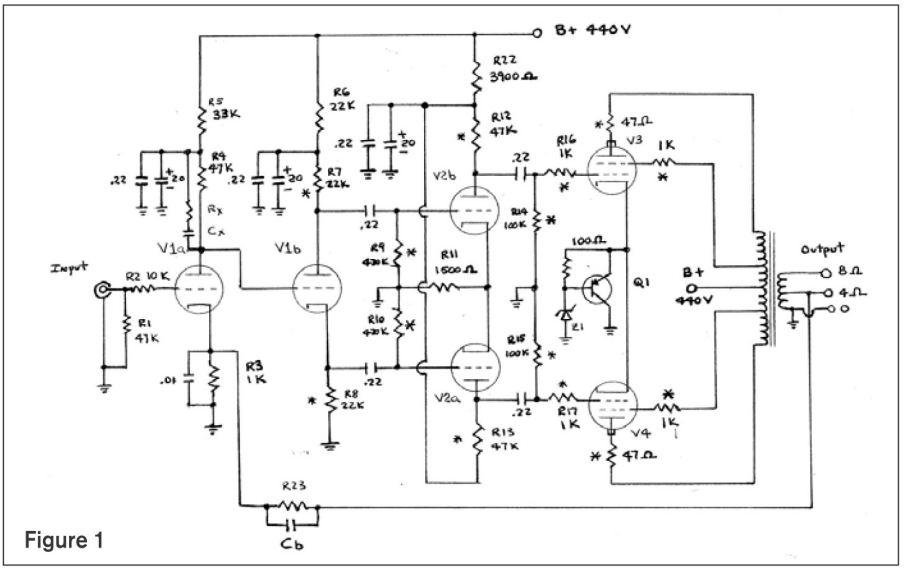

Figure 1

As a longtime reader of *audioXpress*, I look forward to each issue. Of special interest was an article in May 2009 on the Stereomatic 602S amplifier.

First, I would like to compliment John Badalamenti on his article, the *excellent* quality of construction, and superb workmanship as displayed in the pictures. The Williamson circuit topology is a very good time-tested approach, and over many years has lent itself to justifiable improvement. After reading the article and studying the schematic diagram, I have a few suggestions that you may want to consider.

You should note that much of the Williamson circuit is balanced in nature. The chosen value of certain components may not be that critical, but I believe that a close match of corresponding components is important. To facilitate this you should obtain a pack of resistors of a given value, and find two that most closely match by using an ohmmeter. I denote these matched pairs with an asterisk (*), as in **Fig. 1**.

Second, one of the themes in the article is instability. A few suggestions to help in that area are as follows:

A. Add plate stopper resistors. Use one 33 to 68Ω 2W carbon resistor in each of the plate leads of the 807s. You should solder the resistor in as close to the plate cap as possible. Then use some heat shrink to cover up the resistor and the exposed solder connection. This addition may help prevent amplifier runaway, and also eliminate ultrasonic and subsonic parasitic oscillations. Elimination of parasitic oscillations will noticeably—and dare I say drastically—improve the headroom of the amplifier! Your amplifier components will run cooler as well. Remember, just because you may not be able to hear or see the effects of something doesn't mean that it is not affecting the performance and stability of your amplifier.

B. Change the values of screen-grid stopper resistors R20 and R21. The value of 100Ω, in my opinion, is far too low for ultralinear operation. Usually, 100Ω resistors are used when the 807s are connected for triode operation. I recommend 1000Ω 1W carbon resistors.

C. Try a value between 47kΩ and 100kΩ for grid resistor R1. I think that 470kΩ is too high a value.

Third, I recommend changing the bias arrangement in the following ways:

1. A value of 1000Ω for R11 is OK, but you may want to try 1200Ω or 1500Ω. It has been suggested that this change to the operating point of V2 reduces distortion and helps with its ability to drive V3 and V4 to higher output levels before distortion.
2. For cathode bias of V3 and V4, I recommend removing R18 and R19, C9 and C10 and replacing with the simple circuit shown in **Figs. 1** and **4**. See that Q1 is a TIP107 Darlington transistor, which comes in a TO-220 package that you can bolt directly to

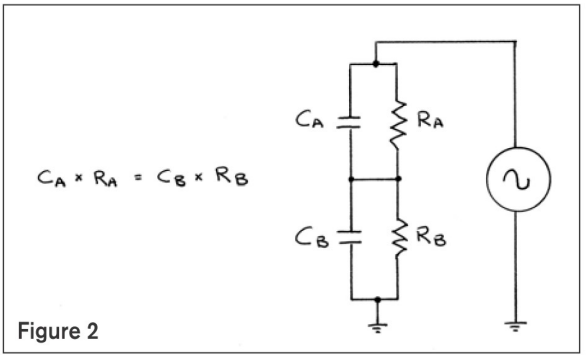

Figure 2

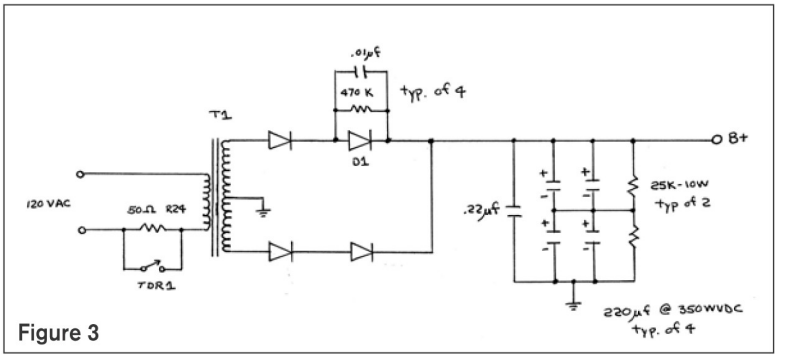

Figure 3

the chassis which is used as a heatsink. Diode Z1 is a 1W (or higher) zener diode selected for proper bias voltage for the 807s. A usual value is 30V, but that can vary depending upon the final operating point of the output tubes.

The regulating factor of the zener diode will be improved by the current gain of the transistor. The TIP-107 is rated for 80W dissipation with a current gain of 1K minimum and a voltage rating of 100V DC. You can substitute a TIP-147 if you need higher dissipation. The cathode bias voltage will stay *very* stable with changing plate and screen currents typically found in Class AB amplifiers.

One of the advantages of this circuit is that an effective capacitance of many thousands of microfarads is developed in the cathode circuit, resulting in a deep low end. I have used this circuit for many years with EL34s, 6L6s, 811s, and 805s, all with good results. Whether you use two 807s in push-pull or four in push-pull parallel, the bias voltage will hold rock solid.

Fourth, consider feedback and frequency compensation in the following ways:

a. Add a frequency compensation network across R4 with a series-connected 270pF silver mica capacitor and 4700Ω ½W carbon resistor. This combination helps to control high-frequency response, and has a turnover frequency of about 125kHz. So it will not cause any audible high-frequency loss but will help attenuate ultrasonic oscillations.
b. I recommend changing C3 to 0.01µF and keeping R3 at 1000Ω. This combination will be one section of a frequency-compensated voltage divider and part of the overall feedback network.
c. Feedback resistor R23 seems too high in value. You may prefer to try values between 8.2kΩ and 33kΩ. The value of this resistor will also depend upon which taps are used on the output transformer. Unfortunately, 100kΩ results in very little feedback, only about 1%. The 807 is a good tube but can be prone to oscillation. That is why the plate and grid stoppers are so very important (and should be soldered in as close to the tube as possible).

The basic formula for a frequency-compensated voltage divider is (Ra) (Ca) = (Rb) (Cb). If Ra = 1000Ω and Ca = 0.01µF, and Rb = 8200Ω, then Cb = 0.0012µF. Because 0.0012µF is not a standard value, you could use a 0.001µF here. This is shown in **Fig. 2**. Some experimentation in this area is warranted, so try different values of Rb and Cb to obtain different levels of feedback, high-frequency response, and amplifier stability.

Fifth, take a look at the power supply. Decoupling among the low-level stages is important for stability and fidelity. For a star-type circuit, I suggest a 33kΩ, 2W resistor for R5, and a 22kΩ, 2W resistor for R6. I recommend removing R22. Put a 3900Ω, 2W resistor between the B+ and resistors R12 and R13. Then connect C4 at the junction of R12 and this new resistor.

In addition, you should bypass electrolytic capacitors C1, C2, C4, and C17 with 0.22µF, or greater, high-quality audio-grade coupling capacitors of sufficient voltage rating. Also, use high-quality coupling capacitors for C5, C6, C7, and C8. Again, I recommend capacitors such as Solen Fast (or equal), which make a noticeable difference in sound quality throughout the entire audio spectrum. Solens are easily obtainable, and reasonably priced.

You can operate the 807 at a plate voltage of up to 750V DC. The power output will correspondingly rise, so pick an appropriate output transformer to handle the power. By selecting the correct plate to plate load impedance, you can easily obtain 100W output from push-pull parallel 807s.

Try any modifications one at a time, and then listen to the results. The order I suggest is:

1. Change the grid stopper resistors, add plate stopper resistors.
2. Make power supply and decoupling modifications.
3. Change the V2 cathode bias resistor R11.
4. Install a new cathode bias circuit for V3 and V4.
5. Install the high-quality coupling and bypass capacitors.
6. Improve feedback and stability with R1, R23, C3, and Cb.
7. Install a frequency compensation network across R4, if needed.

Finally—experiment!

Postscript

1. If there were to be major changes in the power supply, I would recommend two items. First, use 3A at 1000V PIV diodes, with suppression capacitors and equalization resistors, for rectification. Second, replace the filter capacitors with series parallel connected 220µF at 350WV DC Ruby capacitors. This will give you more energy storage and "punch" to your audio (**Fig. 3**).

$\acute{\varepsilon} = ½ C E^2$

Energy = $(.5) \times (220) \times (10^{-6}) \times (440)^2$ = 21.3 joules

2. Tubes that work well in a Williamson circuit are: 6L6GC, EL34/6CA7, 6550, KT66, KT77, KT88, KT90, 5881, 7591, 7027, 6146, and if you want to try some triodes: SV811-10.

I would like to thank my son, Bruce S. Kasprzyk, a freelance professional recording and mastering engineer with a number of projects to his credit, currently residing in Madison, WI, for his critique and revision of this response. Bruce has constructed a similar amplifier based on 12AU7s and EL34 output tubes. *aX*

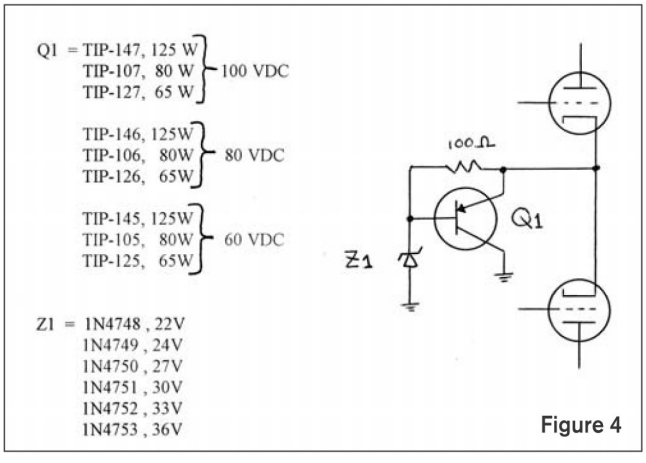

Figure 4

TUBE FACTORY UPDATE

Since the publication of our lead article on the rebirth of the Blackburn tube company ("New Approach to Tube Design," *aX*, Nov. 2009, p. 6), we've received reports that the factory has closed its doors. The historic—now bankrupt—outfit (formerly Mullard and once the largest tube assembly plant in the world) had planned to produce a series of tube replacements for audio amplification. As further details about the status of the company become available, we will post them on our website (www.audioXpress.com).—Eds.

REVIEWS

I just finished reading the Sept '09 issue for the second time and frankly don't get the change in direction toward more "reliable reviews," which took up 13 of 48 pages. I also don't "understand" the Pioneer X-20 review (but more on that later).

I understand that times are tough, but I also think you must (re)examine where you want to go. Your enterprise was established to serve home constructors. Along the way, for some reason, you got into product reviewing. Nothing wrong with that, if the reviewed products in some way had a tie-in to the home constructor. Please show me some home-constructor tie-in to a $1800 Pioneer compact stereo system—will there be a followup tweak article? (I did appreciate the Echo Buster review and there *is* a tie-in here.)

I don't see *audioXpress* competing or attracting new subscribers with an enhanced review presence. There are already too many entities, especially online, trying to review things. In fact, if your trend continues, I would be prompted to stop subscribing. Unless, maybe, your actual idea is to use *audioXpress* as a forerunner of some new publication like *$ensible Sound* with the intent to ditch constructors if the review magazine is successful.

Regarding the Pioneer review: what I read was some kind of "engineering overview" of some kind of product that I doubt *any* reader would ever even *entertain* the thought of purchasing. There were no comparisons to other equipment, nor any suggestion as to why I might want to "rush right out and buy it." And I did not come away with any sense that it fits in your characterization of "reasonably priced, quality audio equipment"—compared to what? A *complete* waste of my time—and valuable *space* in your magazine.

Charles King
Stellavox1@aim.com

I really enjoyed David A. Rich's review of the Sony XDR-F1HD FMHD tuner in *aX* August 2009. He really lays it all out.

Many tuners and receivers have been manufactured that sound great, but unfortunately don't perform well as they age. In his article, David points out the XDR-F1HD runs hot. This is the case with most tuners and receivers. The resulting heat causes the components to fail in time. Over the years I've owned several tuners and receivers, and thus far, none have enjoyed what most would consider a long life.

The failures I've experienced include everything from the audio going mute after several minutes, to loss of the stereo multiplex. I own one that lost its ability to store stations in its eight presets.

The one thing these troublesome tuners and receivers have in common is that they are AC powered units containing a handful of TO-220 style voltage regulators, which run super hot, and raise the temperature inside considerably. With this in mind, perhaps David might try running the XDR-F1HD with an outside (or external) power supply and report to us the difference in operating temperature. My belief is that if you can equip a tuner with a cool running power supply, it will perform as well and as long as a car radio. I am in total agreement with the reviewer in that car radios blow home radios out of the water when it comes to performance and reliability. How many folks out there have had to throw out their car radio?

Neal A. Haight
Tracy, Calif.

RCA POWER SUPPLY

It's not surprising that Aren van Waarde experienced overheating of potentiometer R9 in his high-voltage regulator (**Fig. 1**, "Regulated Power Supply for Tube Projects," *audioXpress*, 8/09). I'll bet R5 is also hot when R9 (which is connected as a rheostat) is set near its minimum resistance.

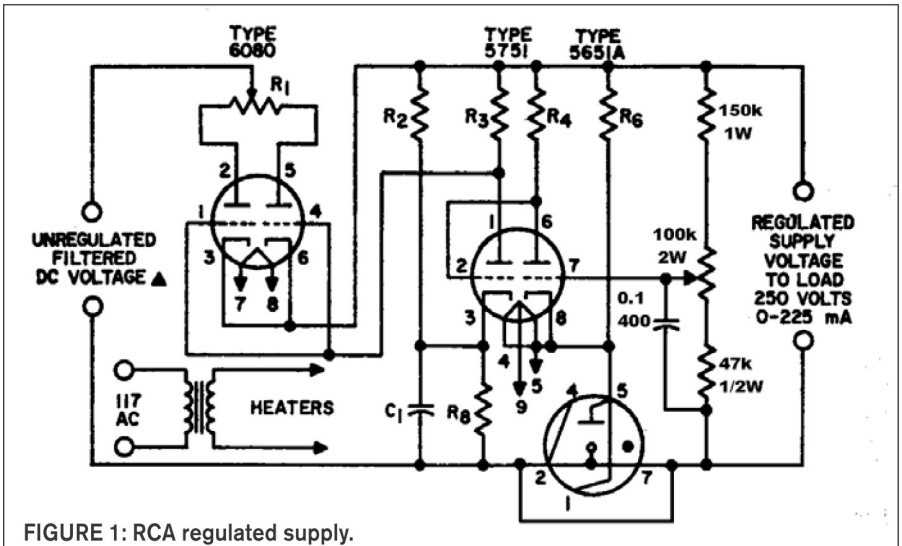

FIGURE 1: RCA regulated supply.

Furthermore, the socket wiring of the 5651/85A2 voltage reference tube as given in the RCA schematic (**Fig. 1**) is such that removing that tube opens the connection to the output ground terminal, while at the same time the regulator pass tube is no longer controlled, allowing the maximum supply voltage to reach the positive output terminal. Not only could the attached equipment receive excessive voltage from the now-unregulated supply (with a ground return path possibly being established via three-wire grounded power cords and the building's power ground system), but a potential shock hazard exists if an *ungrounded* power cord is used on either the supply or the attached powered unit—a person touching both chassis simultaneously could complete the high voltage ground return path through his or her body. This design seems questionable, but the following suggestions should eliminate these problems.

One thing about potentiometers or other variable resistors: Their power rating is for the *entire* resistance element; that is, between the end points. When used as a rheostat where only part of the resistance element is in the circuit (i.e., from one end point to the slider), the power rating must be reduced to the percentage of the total resistance actually being used. For example, if only half of the resistance is in the circuit, the power capacity is reduced to half of what it would be if the entire element was in the circuit; one-tenth the resistance, one-tenth the power capacity, and so on. Unawareness of this has resulted in many a fried pot or adjustable resistor! Unfortunately, the RCA regulator design confronts Aren with this very problem.

Power dissipation in smaller variable resistances may be minimized by using them as true potentiometers instead of as rheostats. This can be accomplished in the RCA circuit by moving the 5651/85A2 regulator tube from the grid of the first triode (right side) of the 5751 amplifier to its cathode (replacing R9), and moving the output adjustment control to the grid circuit. A few circuit changes are necessary in order to do this:

1. Remove R9 (10k pot), R5 (12k, 2W), and R7 (1M, ½W) from the circuit and put them in your junkbox. Be sure pins 4, 5, and 8 of the 5751 remain connected together. Use an ohmmeter to measure between these pins and chassis ground—you should read an open circuit. The filaments in the supply must float above ground and be isolated from all external circuits.
2. Connect anode pin 5 of the 5651/85A2 reference tube to cathode pin 8 of the 5751, leaving R6 (68k, 1W) in place (aside from providing a stable reference voltage for the regulator, this tube also places a +85V bias on the filaments, thereby reducing the potential for cathode to filament breakdowns).
3. Install an external jumper between pins 2 and 7 of the 5651/85A2. This removes the potential shock hazard mentioned above, provided a ground connection exists between the supply and the attached equipment.
4. Construct a voltage divider using three resistances connected in series between the positive regulated output

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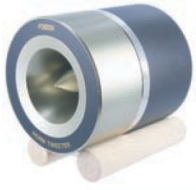

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line and ground, as follows:

a. Connect one end of a 150k, 1W resistor to the positive regulated output line. Connect the other end of this resistor to the right-most terminal (as viewed from the rear) of a 100k, 2W potentiometer (Mouser 785-53C1100K, about $7). Other quality pots rated at ½W or more may also serve.
b. Connect the center terminal (slider) of the potentiometer to grid pin 7 of the 5751. Bypass this pin to ground with a 0.1µF, 400V capacitor.
c. Connect the left-most terminal of the potentiometer to one end of a 47k, ½W resistor. Connect the other end of this resistor to ground.

Everything should now be cool with the possible exception of the 68k resistor when the output is around 300V, although it will still be within its power rating. I might be inclined to use a 75k, 2W resistor here if I had one. If the desired output range is not spread across the full rotation of the 100k pot, change the two fixed resistors in the divider as necessary while keeping the total divider resistance in the vicinity of 300kΩ.

Note that removal or failure of the 5651/85A2 reference tube will now cause the output of the supply to drop to its minimum value.

Keith Kunde
Independence, Ohio

Aren van Waarde responds:

Thanks for your warning regarding potentiometer ratings and your concern for the good health and electrical safety of the readership of *audioXpress*. In my homemade copy of the RCA psu, I used an ungrounded power cord and a cabinet made of insulating material. Thus, it is not possible to touch the PSU chassis directly. Moreover, the output meter will provide a warning sign if the output voltage turns to maximum because of catastrophic failure of the 85A2 voltage regulator tube.

Yet, your suggestions for improvement of the RCA schematic are most welcome. The figure which you have added indicates clearly which amendments you propose. I have not yet tried this modification (because of the fact that my junkbox does not contain a 100k 2W potentiometer), but I expect that the modified circuit will work well.

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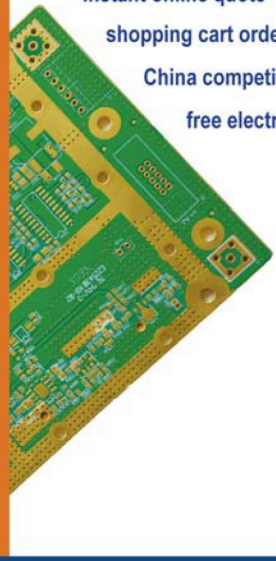

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TURNTABLE TESTING

I think that the turntable-strobe devices reviewed by Gary Galo (*aX* 7/09, p. 24) represent overpriced foolishness. The classic solution to this problem is, of course, a printed cardboard strobe disc, which used to be available in the US for about a dollar, plus a cheap neon lamp and resistor on an AC cord, costing perhaps another dollar. Galo observes more or less correctly (although, see below) that the lamp and turntable motor can both be affected by line-frequency variations, but exactly how big are these? Unless you listen to records using a dodgy backyard diesel generator, the answer seems to be: not big enough to matter.

Years ago, I became curious enough to chat up the chief engineer of my local electric power utility (I then lived in a large midwestern metropolitan area) who told me that prevailing US electric utility standards then called for regulating the frequency within ±0.3%. Note that there's only one significant digit of accuracy in this figure. It amounts to

slightly more than ±0.2 rpm in a nominally 78 rpm turntable, and it's less than the variation in standard tuning pitch (the frequency of "A" above middle "C") used by many of the world's major orchestras. Unless your sense of pitch is acute, a listener is unlikely to notice such a small difference. Of course, rapid pitch changes ("wow" or "flutter") are a different story, but stroboscopes can't cure those anyway.

This engineer pointed out to me that there are at least two good reasons to keep the power line frequency under reasonably tight control:

1. When a generator is switched on or off a power grid, which happens more often than you might think, its frequency and phase must be closely matched to the grid in order to avoid a huge switching transient. History has shown that a small error can trigger a big, expensive disaster!
2. The US power grid still supports an enormous number of clocks and timers powered by synchronous motors, many of which are operated by the utilities themselves, so they can't allow much drift. If the line frequency is low for a period of time, then it must be run high for an equal period in order to cancel the error. Practically speaking, the goal is to keep these averaging periods short enough (just a few minutes) that a human with an electric clock won't notice the difference.

If your turntable has a reasonably heavy platter and well-oiled bearings, it should have enough inertia to maintain adequately constant speed over such periods of a minute or two, so I think that readers should save their money for some other hi-fi non problems!

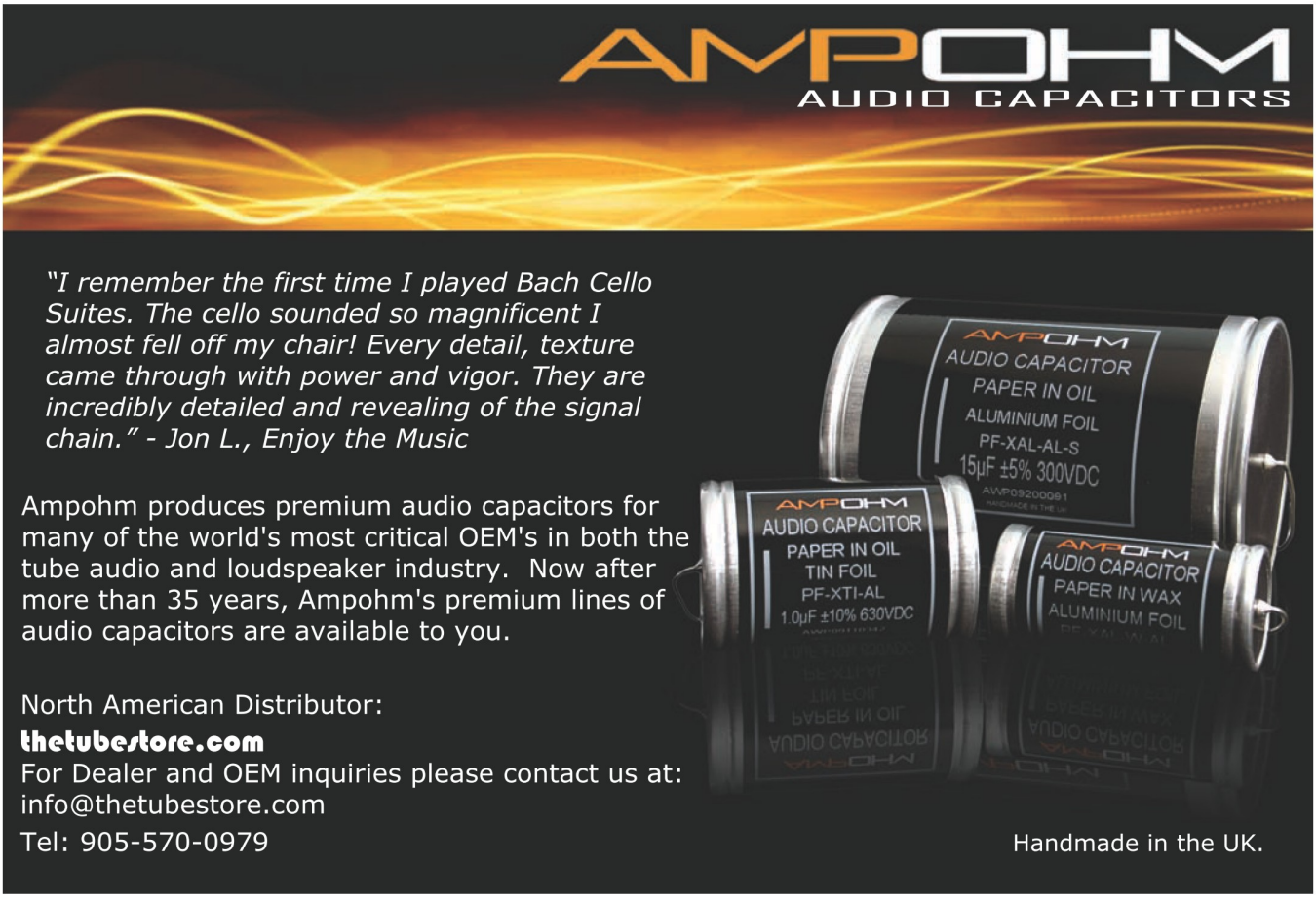
Scott B. Marovich
East Palo Alto, Calif.

Gary Galo responds:

I disagree with Mr. Marovich regarding the KAB Speed Strobe being overpriced. At $99 I consider it to be a fair value for what it offers, particularly the ability to measure the 12 most common "78-rpm" speeds. No other currently available strobe that I'm aware of offers this capability. There are a number of 78-rpm turntables available that do not offer digital readouts, so the only way to get repeatable settings is with the KAB Speed Strobe.

Regarding the unimportance of a 0.3% error, to musically sensitive listeners with acute pitch sensitivity, such an error is audible. There's more to be concerned with than just pitch. For those who collect historic vocal recordings, even very small changes in pitch can affect the timbre of a singer's voice. A 0.3% error represents the difference between A = 440Hz and A = 438Hz (European vocal recordings made early in the 20th century often have a pitch reference a bit lower than A = 440). Many serious vocal collectors, and I count myself among them, will tell you that such a difference is not without consequence.

If an expensive amplifier sounds the same to you as a cheap amplifier, by all means purchase the cheap one. If a conventional strobe illuminated with the AC supplied by your power company is accurate enough for your needs, don't spend another dime. But, if the additional accuracy offered by the KAB makes a difference to you and/or you have the need to measure common "78-rpm" speeds, the KAB is fairly priced, in my opinion.



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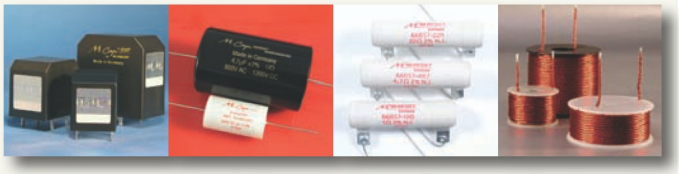

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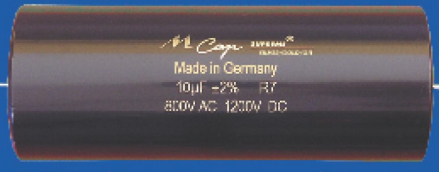

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SOUND DIFFERENCES (CONT.)

Kent Smith owes the AES an apology for his cheap shot in "Amplifier Differences and Lack of Tests (*aX* 7/09, p. 5)." There are no secrets being withheld from the common masses except for the willfully ignorant cheapskates choosing not to become members. I presume Mr. Smith pays for *audioXpress*, and he should expect to pay the membership fee that supports the publication of AES research papers.

Furthermore, musicians often are the worst judges of audio performance because they hear their instruments from a drastically different proximity and direction in comparison to the audience's perspective that recordings attempt to capture. A drum up close produces ear-damaging painful transients that cause tension for the listener that could trigger a stapedius response that partially mutes the sound. Perhaps the clipping in his grossly underpowered amp simulates the same effect at a comfortable loudness, but it is not surprising that he would prefer the softer clipping of the tube amp. On the other hand, compressed music with vastly more of the music at the maximum level would be most damaged by a lack of amplifier power, so it is no surprise that the 60W IC would beat a 2W tube.

There is no lack of material from AES regarding amplifier overload, but it becomes moot if you use an adequately powerful amplifier. Frankly, I find the comparison of two inadequate amplifiers about as exciting as a drag race between a Smart Car and a Tata. I've been doing live recordings (including the Ypsilanti Symphony) for many years with a coincident pair of AKG C414B-ULS microphones and no signal doctoring, and my recordings always sound most accurate when the amplifier is not clipping. Since my budget is not unlimited, a test for Watts per Dollar will do in most cases.

Dennis L. Green
d7green@yahoo.com

Kent Smith responds:

I would like to thank Mr. Green for the invitation to join AES. For some reason, I had the idea years ago that the AES was only for degreed professionals who worked in audio. I assumed I wasn't qualified until I looked at their website. Truth is, they appear very interested in including us all as Associate Members! For this, I do indeed apologize to the AES and to Mr. Green personally as well. It does not appear to be the secret society I once thought it was. But I must admit I'm cheap; I'm still thinking about whether I want to spend over $200 to get the membership and access to the e-library for one year.

Of course, I agree with Mr. Green's other comments as well. I know some musicians with horrible playback systems, and I'm still amazed how they can hear subtle differences that I notice only when they point it out to me. I guess quality isn't everything! Fortunately, my son is only 23 and hasn't blown his ears out yet.

Of course, when doing a comparison listening test, we avoided levels that would clip the amplifiers, kept the gain about the same, and used a listening level that was below where ear distortions begin, so maximum power levels of the amplifiers are of no consequence, of course. The 2W amp into my high-efficiency

two-ways (Eminence Pro 8″ in vented box at 50Hz and Hi-Vi Planar tweeter) produces up to 95dB at 1m, which is quite adequate for me (I don't like loud anymore). Most important, this little amp produces acoustic instruments correctly, including the timpani (I go to classical concerts), where the LM3886 chip does not. So I must stand by my other non-AES-related comments.

Weird, huh? Sometimes, I still can't believe that I have become one of those weird tube people. Would the AES still welcome me?

TUBE CALCULATIONS

Regarding the article by Pierre Touzelet ("A Graphical Evaluation of Harmonic Distortion in Valves," July '08, p. 42), it is possible to determine up to the fourth harmonic with this method:

$$\%2^{nd}\text{ harmonic} = \frac{300(A + B - 2P)}{4(A + C - D - B}$$

$$\%3^{rd}\text{ harmonic} = \frac{50(A + 2D - 2C - B)}{(A + C - D - B)}$$

$$\%4^{th}\text{ harmonic} = \frac{25(A + 6P - 4C - 4D + B)}{(A + C - D - B)}$$

It is also possible to use this method for a class A push-pull amplifier. Let A C D B P be the anode current for a single tube. Let A' = A – B, B' = B - A, C' = C – D, D' = D – C, P' = 0. Put in the values for A' B' C' D' P' in the formulas above and you get the distortion from a class A push-pull amplifier.

It is also possible to use this method for a class AB push-pull amplifier. A' = A, B' = -A, C' = P, D' = -P, P' = 0. Put in the values for A', B', C', D', P' in the formulas above and you get the distortion from a class AB push-pull amplifier.

Of course, you can also use the formula for transistors or different combinations of transistors. The A B C D P must not be a current, it can also be a voltage. For example: Calculate the distortion from an MJE 340 emitter follower with a 1kΩ emitter resistor. Input voltage swing 20Vpp:

Vb	Ve
5V	A = 4393.7mV
10V	C = 9373.9mV
15V	P = 14362.8mV
20V	D = 19355.0mV
25V	B = 24349.1mV

$$\%2^{nd}\text{ harmonic} = \frac{300 \times 17.2}{4 \times 29936.5} = 0.043\%$$

$$\%3^{rd}\text{ harmonic} = \frac{50 \times 6.8}{29936.5} = 0.011\%$$

$$\%4^{th}\text{ harmonic} = \frac{25 \times 4}{29936.5} = 0.0033\%$$

Rickard Berglund
Sweden

HELP WANTED

I'm trying to replace a speaker design book that I bought at a walk-in Heath (kit) store. I think it was published by McGraw-Hill. It had "classic" old designs such as Paragons, corner Klipschs, and so on. My copy ended up in a box that went to a group known as "Friends-of-the-Library."

Is it possible that someone may remember the title? This book, with designs from the 50s and 60s, and printed in the 80s, is most likely out of print.

Austin M. Gillette
La Pine, Oreg.
aX