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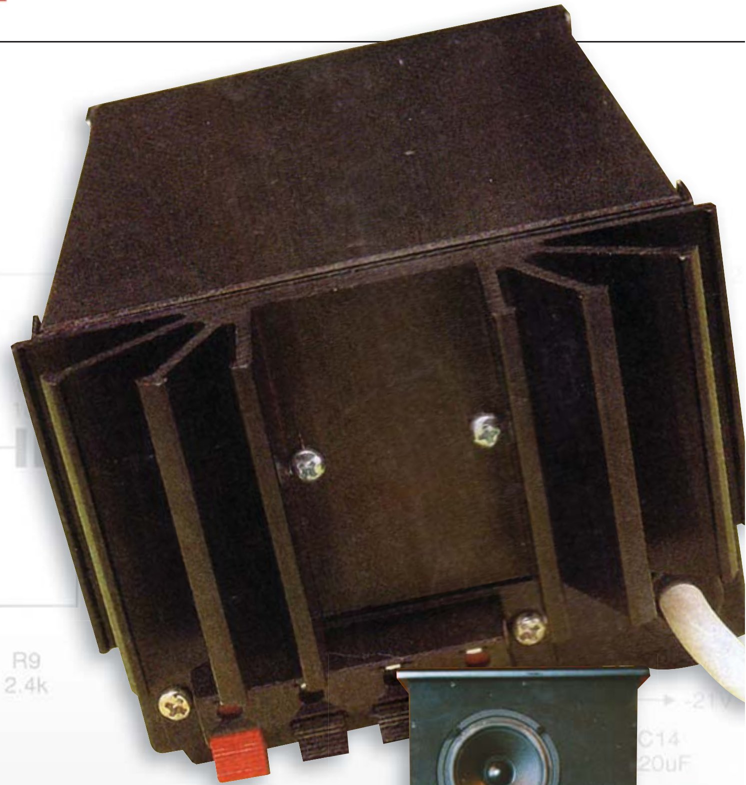

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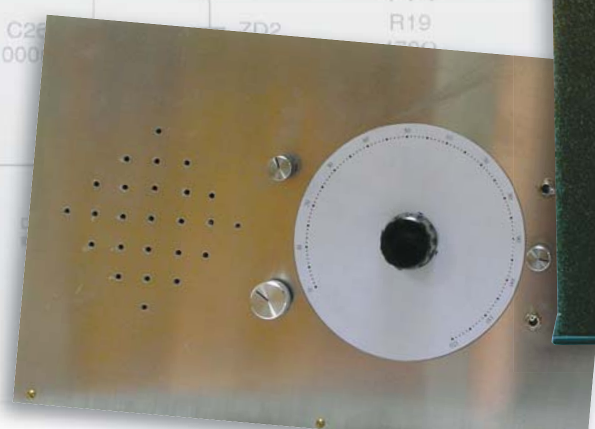

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Satori is a Buddhist term for "enlightenment." The word literally means "understanding." "Satori" translates as a flash of sudden awareness, or individual enlightenment, and while satori is from the Zen Buddhist tradition, enlightenment can be simultaneously considered "the first step" or embarkation toward nirvana.

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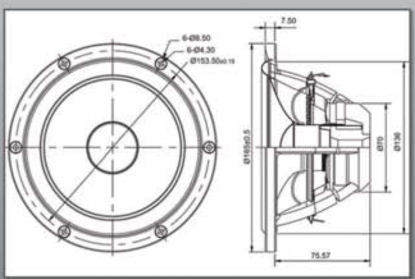

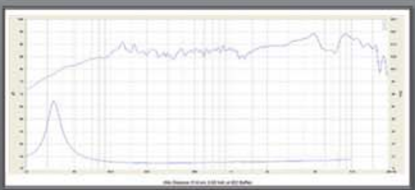

Preliminary Data

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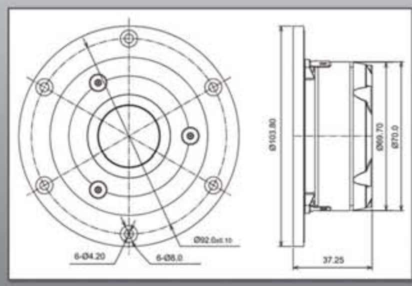

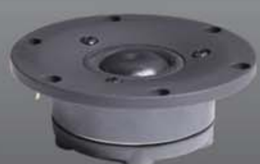

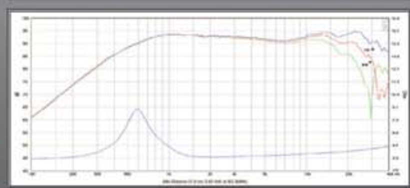

Preliminary Data

Specs :

Nominal Impedance	8 Ω	Free air resonance, Fs	29 Hz
DC resistance, Re	6.2 Ω	Sensitivity (2.83 V/1m)	87.5 dB
Voice coil inductance, Le	0.16 mH	Mechanical Q-factor, Qms	4.9
Effective piston area, Sd	119 cm2	Electrical Q-factor, Qes	0.36
Voice coil diameter	35.5 mm	Total Q-factor, Qts	0.34
Voice coil height	17 mm	Moving mass incl.air, md	13.4 g
Air gap height	5 mm	Force factor, Bl	6.5Tm
Linear coil travel (p-p)	12 mm	Equivalent volume, Vas	44.8 ltr
Magnetic flux density	1.17 T	Compliance, Cms	2.23 mm/N
Magnet weight (NEO)	0.13 kg	Mechanical loss, Rm	0.5 kg/s
Net weight	1.01 kg	Rated power handling	60 watt

The parameter are measured on drive units that are broken in

Specs :

Nominal Impedance	4 Ω	Free air resonance, Fs	590 Hz
DC resistance, Re	3.0 Ω	Sensitivity (2.83 V/1m)	93 dB
Voice coil inductance, Le	0.02 mH	Mechanical Q-factor, Qms	2.5
Effective piston area, Sd	9.6 m^2	Electrical Q-factor, Qes	1.5
Voice coil diameter	29.0 mm	Total Q-factor, Qts	0.9
Voice coil height	2.0 mm	Force factor, Bl	1.8 Tm
Air gap height	2.5 mm	Rated power handling*	100 Watt
Linear coil travel (p-p)	0.5 mm	Magnetic flux density	1.0 T
Moving mass incl. air, Mms	0.45 g	Magnet weight	0.22 kg
		Net weight	0.5 kg

*(IEC 268-5, Via High Pass Butterworth Filter 2500Hz 12 dB/oct.

We have spent a lot of time trimming the fine details of these magnificent drive units, and we are certain you will achieve

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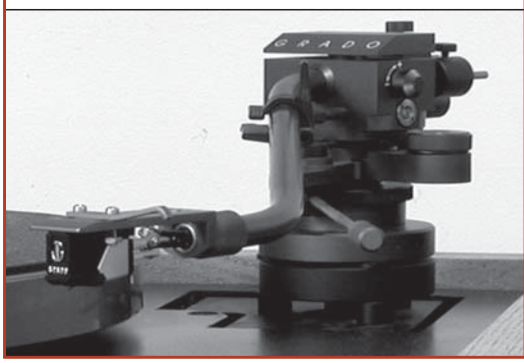
CONTENTS

VOLUME 41 NUMBER 10 OCTOBER 2010

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

FEATURES

RELIABLE REVIEWS



DEPARTMENTS

36

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Printed in the USA.

By Raj.K.Gorkhali

The Brick

This compact, low-cost, add-on amplifier will deliver a cool 15W/channel in stereo mode or about 29W into a single speaker in bridged mono mode—all from an unassuming little box no larger than a house brick!

Like many popular audio projects, this one arose from a clear personal need. After getting a new DVD player, I soon realized that I would need to upgrade my modest "home theater" amplifier and speaker setup in order to take full advantage of the Dolby Digital (AC-3) surround sound on most DVDs.

Up until now I'd been "making do" with a simple analog-type surround-sound setup, using the el-cheapo matrix-type decoder. I hadn't even been using all of the outputs from this. I'd been getting quite satisfying sound from most video movies using just the front left and right speaker (driven by my main stereo amp) and a single rear-channel speaker, driven (in bridged mono mode) by one of the stereo amps.

My family and I didn't really miss the center-front speaker, which, after all, handles only a "sum" of L+R signal with this type of analog surround decoder. We didn't miss out on much of the surround-sound effect either, because there's really only one "rear" signal with this type of decoding anyway, derived from the "L-R" or "difference" signal. Nor did we miss a subwoofer, because my main speaker has pretty good bass response.

UPGRADE NEEDS

But when I tried playing Dolby Digital signals through this "economy" surround-sound setup, it soon became clear that I would need to make some changes. For starters, all of the Dolby Digital "5.1 channels" are discrete and separate, and the center–front channel signal is quite different from a mere sum of the front L and R signals. But as it happens, many DVDs actually have the main screen dialog on the center-front track, so you really do need a center-front amp and speaker if you want to hear most of the dialog! (Unless you use the two-channel mix down outputs, an analog decoder.)

Also, with Dolby Digital, as with MPEG and DTS encoding, there are two true and separate rear surrounding channel signals rather than a single derived "difference" signal. So if you want to hear all of the directional surround information, you really need two separate amplifier channels and speakers.

Finally, with Dolby Digital encoding, the subwoofer track (actually called the "LFE," or "low frequency effects," track) again seems to be quite discrete. In fact, on many action blockbuster DVDs it's used to deliver most of the "grunt" for explosions and other effects. So without a subwoofer, you can again miss most of the impact of these effects.

All of which meant that in order to really enjoy Dolby Digital movies from DVD, I was going to need at least two—possibly three—more speakers and matching amplifiers. I suspect many readers are going to be in a similar position when they upgrade to *digital* surround sound as part of DVD-based home theater.

The extra speakers aren't likely to be too much of a problem, of course; many of us have a few lying around from discarded stereo systems. But I soon found myself looking for a low-cost add-on amplifier, for either stereo or bridged mono use.

What I really needed was a kind of "bare essentials" version of the Shoestring Amp. Just the basic power channels with all of the frills removed, housed in a compact box, and designed so you can build it for a rock-bottom price. The concept of the Brick was born.

Why the name? Well, it's roughly the same size and shape as a house brick, and almost as unattractive. Functional rather than pretty, if you like.

Introducing the

DEQX HDP-Express

There's never been anything like DEQX's HDP-3 DSP processor. Its -140dB THD digital transparency provides DEQX's unique speaker and room correction and active linear-phase crossovers.

It also provides Preamp and DAC features to avoid unnecessary conversions. And yet, despite being modestly priced from just US$3950, it's been out of reach for many serious music lover DIYs.

Enter the HDP-Express™.

Designed for DIY budgets from **US$1950** (plus cal mic and freight) if purchased factory-direct, the HDP-Express™ retains the key performance features of the HDP-3, using the same DEQX-cal setup software for Windows (or Mac with Windows).

For maximum convenience, all preamp functions are controllable by the remote.

Active linear-phase crossover features:

- Steeper than 6dB/octave crossover filters required to reduce distortion don't maintain linear-phase using traditional crossovers. Now they do.
- Traditional active Butterworth and Linkwitz Riley, mild to steep crossovers and four 'Profile' presets lets you directly compare linear-phase advantages.
- Using 48dB/octave or steeper linear-phase crossovers distortion vanishes, resolution increases, while volume can double or quadruple, 3 to 6dB.
- When used with speaker correction, stereo 3D imaging and dispersion also improves to such a degree that a centre speaker is unnecessary.

Speaker correction features:

- Anechoically measure and correct your passive or active speakers using a simple measurement technique, even in your listening room.
- Corrects anechoic (native) speaker frequency-response from typically plus/minus 3dB to plus/minus 0.3dB*: about a 6dB improvement!
- Corrects Group-delay (phase/timing errors at different frequencies) about tenfold* e.g. 1ms reduces to 0.1ms —especially noticeable in midrange.

Room Correction features and preference EQ:

- One or more room measurements displayed graphically, allow manual, real-time, 7-band parametric EQ settings
- Time-domain correction of subwoofers/bass speakers measured in-room.
- Adjust delay between main speakers and subwoofers in real-time.

Media correction—forensic tone control:

- Remote controlled Low-shelf, Midband-fully parametric and High-shelf
- Low, mid and Hi frequencies adjustable in octaves and semitones
- Adjustable Q from one semitone to four-octave wide
- 99-memory storage for instant recall

Pre-amp and processing features

- Four profiles for instant selection of crossovers, correction and EQ
- Four inputs: S/PDIF, AES3, analogue unbalanced and balanced.
- Integrate one or two subwoofers
- Remote controlled Input and profile selection.
- Six unbalanced outs: Stereo low, mid, Hi (mid used for passive speakers)
- Optional balanced outs (6 x XLRs) transformer or active.
- Dual 32-bit SHARC DSPs provide minus 140dB THD digital transparency

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*Anechoic correction may have limited resolution at bass frequencies (20Hz - 200Hz), where room measurement and correction can be used.

All the same, it provides two 15W RMS power amplifier channels, with low distortion and noise, and good frequency response. You can use the two channels for two separate signals, if you wish, or combine them with the flick of a switch into a single bridge-mode mono channel, with an output of about 29W RMS. So it's equally suitable for driving a pair of rear–channel speakers, a single center-front channel, or even a reasonably sensitive subwoofer.

The input sensitivity is pretty good, too. You need only around 200mV input to produce full output, so it's very suitable for use with "line" output on DVD players with built-in Dolby Digital decoders. In fact, you could use one or two of the Bricks and the new surround decoder to produce a complete and quite effective low-cost analog surround sound system. Of course, the Brick would also be quite suitable for enhancing a multimedia computer setup, allowing you to drive a pair of decent external speakers.

OPERATION

There's nothing very novel about the Brick's circuitry; it's basically just the power supply and power amps with a "front end" added. The only real twist is a simple scheme that allows the two channels to be driven from one of the inputs with opposite polarity, for hassle-free bridge-mode mono operation. The schematic (**Fig. 1**) shows how it's done. All of the front-end circuitry is based on a TL072 dual FET input op amp (IC1), while each channel's power stage uses a familiar and low-cost LM1875 power op amp.

In stereo mode, each half of the TL072 operates as a non-inverting input amplifier with a gain of about 2.8 times—as determined by feedback resistors R7 and R5, for example. At the front of each input amp there's a low-pass filter for RF suppression (R1/C1 and R2/C2, followed by the volume/gain control RV1a/b) and blocking capacitors C3/C4.

Switch SW1 is used to configure the circuit for bridge-mode mono operation. When it's switched to the mono position, the non-inverting input of IC1a is connected to ground, so the "left" input of the Brick becomes ineffective. However, the "cold" end of resistor R6, used as that stage's lower feedback resistor in

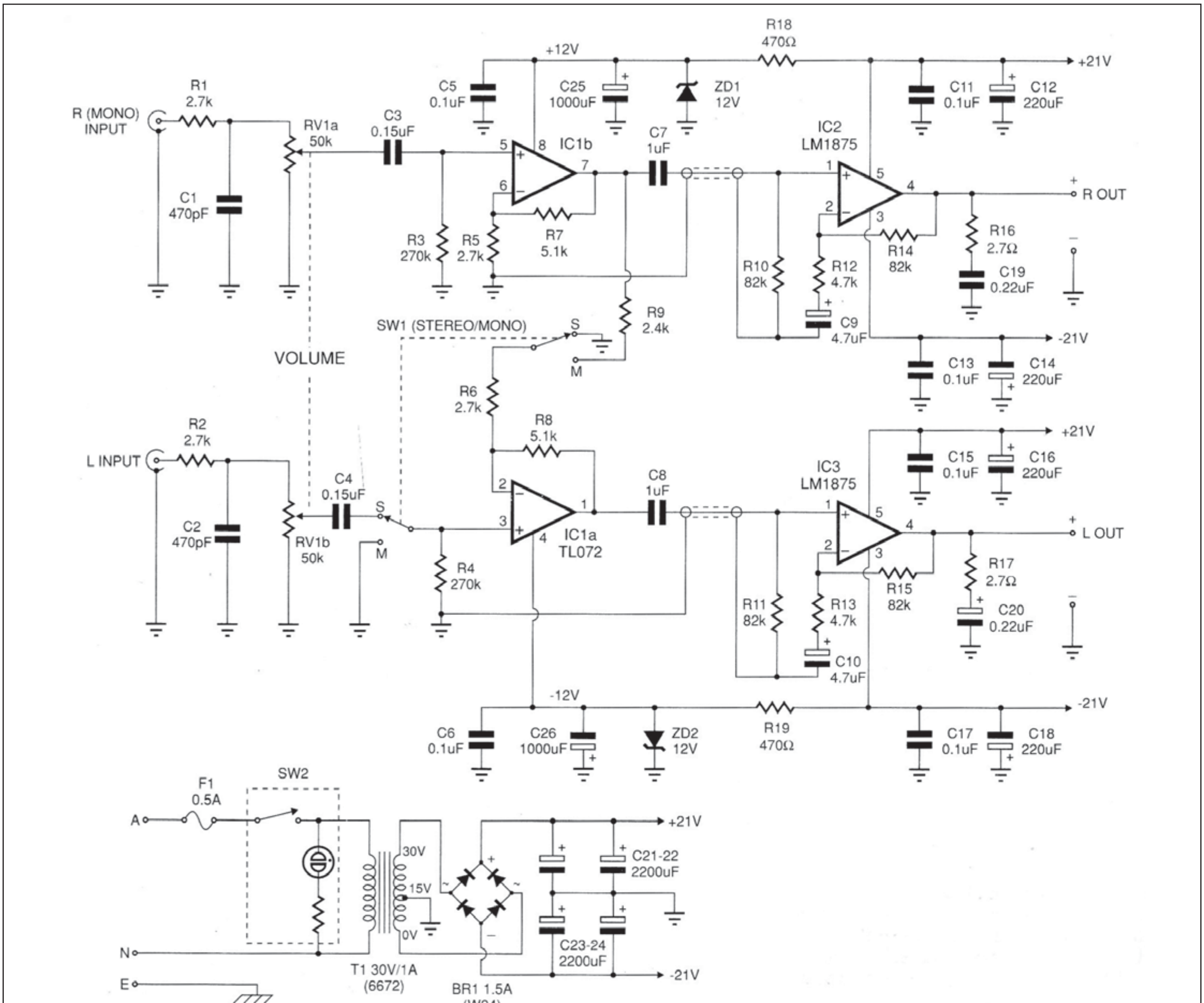

FIGURE 1: The Brick amp circuit.

stereo mode, is now disconnected from ground and connected instead to R9, running to the output of IC1b, the right channel's input stage. As a result, IC1a is now converted into a unity-gain inverting stage, taking the signal from IC1b and producing an opposite-polarity replica at its own output. This converts the front end so that it now produces dual-phase drive signals from the right channel input.

Each input stage drives one of the LM1875 power stages (IC2 and IC3), with very traditional circuitry. Coupling caps C7 and C8 provide DC blocking, with the power stage voltage gains set to around 18 by the feedback network R14/R12 and R15/R13. Resistors R10 and R11 provide biasing for the non-inverting inputs, while C9/C10 provide AC grounding for the bottom of the feedback networks, yet allow the power stages to operate as DC voltage followers for maximum thermal stability.

The earth returns for the output amp input circuits are taken back to the front-end circuit earth to maximize AC stability. There's also a zobel impedence stabilizing circuit across each power amp's output (R16/C19 and R17/C20) to again ensure maximum AC stability. In stereo mode, each of the output stages drives its own speaker, which is connected between the "+" output terminals to receive the combined output of both stages.

The output stages operate from DC rails of ±21V, which is provided from a simple power supply using a 30V/1A center-tapped transformer driving a bridge rectifier (or, if you prefer, two full-wave rectifiers), with two 2200µF reservoir capacitors for each output rail. The input stages operate from ±21V rails via simple dropping circuits using resistors R18/19, zener diodes ZD1/2, and decoupling capacitors C25/26.

CONSTRUCTION

The electronics side of the Brick is pretty conventional. The only real trick is to squeeze everything into a small, low-cost brick-shaped aluminum utility box, measuring only 125 × 100 × 75mm (**Fig. 2**). This wasn't easy, especially as the 30V/1A power transformer takes up over half of the internal volume.

The power supply and input stages are mounted on a small PCB just behind the front panel—in this case, mounted vertically—while the power amp stages are mounted on a small narrow PCB mounted just behind the rear panel and heatsink. Because of the smaller box and the need to use a low-cost finned heatsink for the output devices, even the power amp PCB had to be smaller than that in the amp. I achieved this by moving the input coupling capacitors to the front end board, and also by using separate power supply connections for each channel. As with the corresponding board, this one is supported purely by the two power amp chips. This causes no problems as the board assembly is very light.

The front end/power supply PCB is automatically much smaller than in the

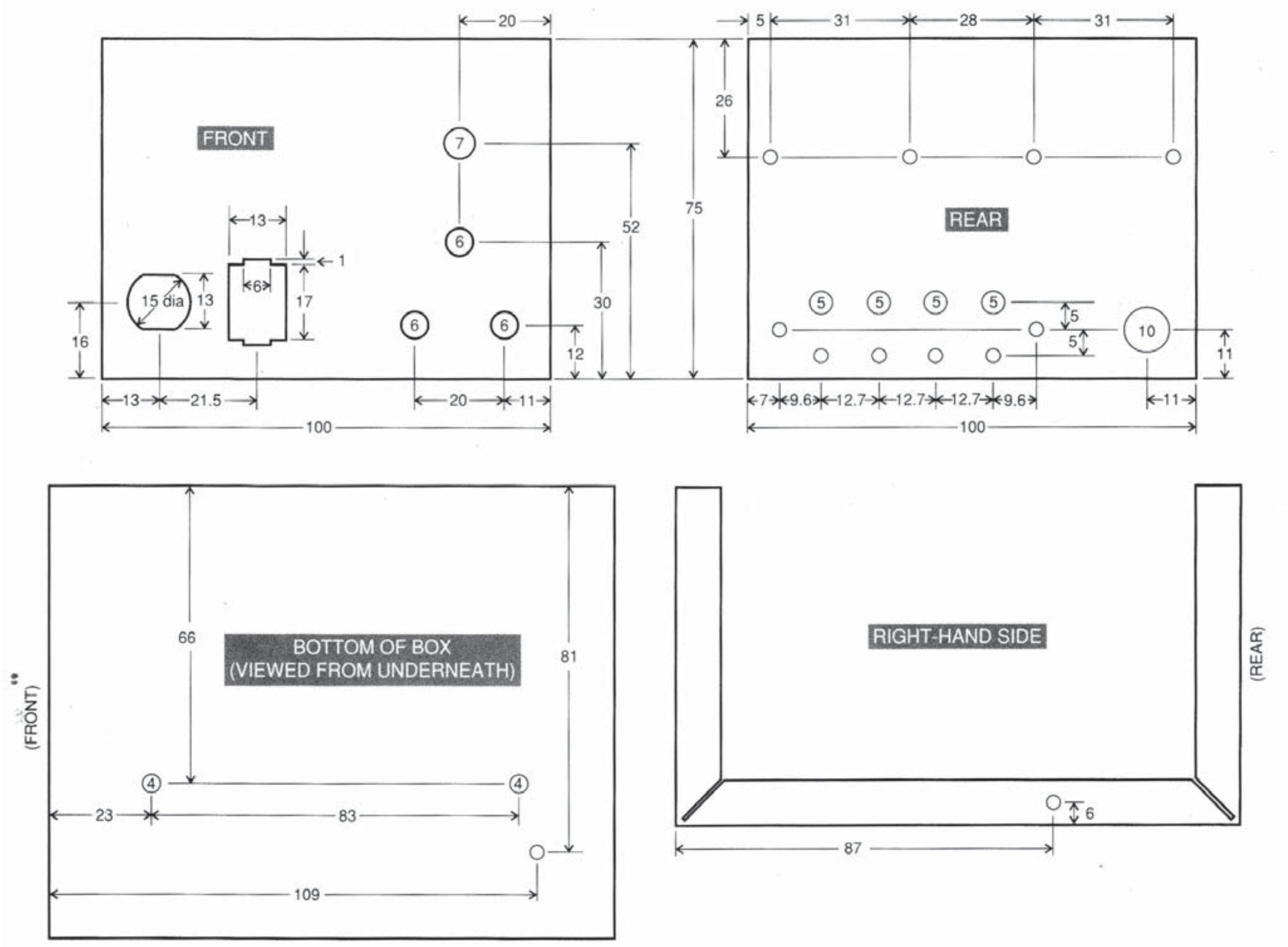

FIGURE 2: Box dimensions (all in metric). Note: All unmarked holes are 3mm diameter.

previous design, because it has much less circuitry in the front-end section. In this case it measures only 89 × 69mm, and is supported mainly by the dual-gang volume control RV1 and a single machine screw and spacer at the lower rear corner. This is, again, quite satisfactory because there's little weight involved.

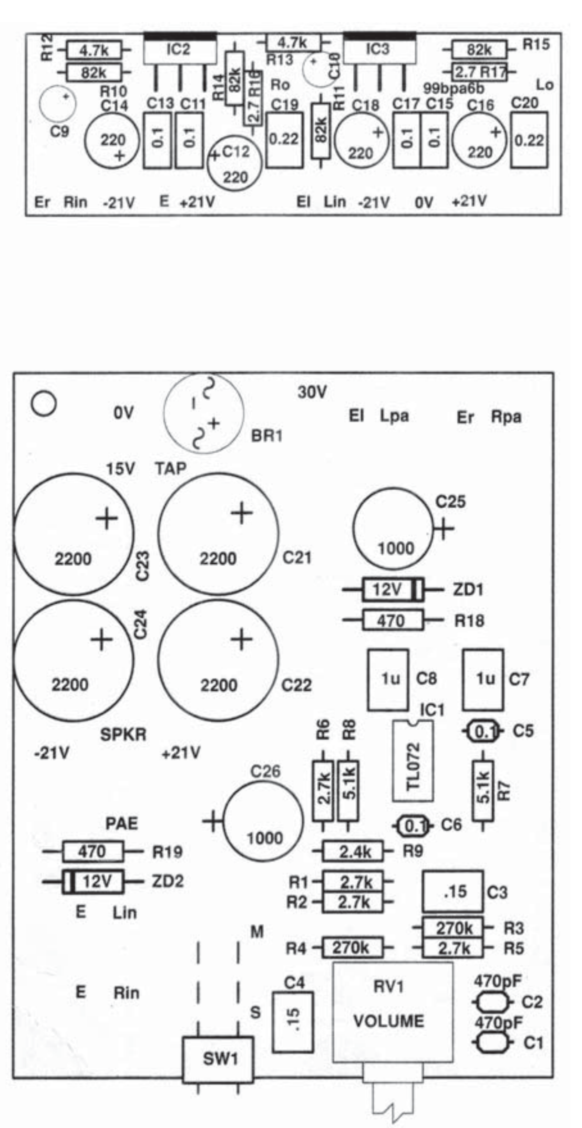

FIGURE 3: Power supply and power amp PCBs.

Note that this PCB mounts vertically near the right-hand side of the box in order to clear the power transformer. Everything fits with room to spare, but you must assemble the parts in the right order.

ASSEMBLY

I suggest you start assembly by fitting all of the minor components onto the two PCBs (**Fig. 3**). As usual, it's best to fit the PCB terminal pins first, then the low-profile passive components (resistors and zeners), followed by the smaller capacitors, and finally the larger parts and zener diodes with the correct polarities, as shown in the PCB overlay diagrams (**Fig. 4**).

Next you can fit the rectifier bridge BR1 to the larger board, taking care to match the orientation with that shown on the overlay (the longest lead is the "+" output). Then you can fit the power chips IC2 and IC3 to the smaller board, mounting them so their leads just pass through the PCB by about 1.5mm—just enough for a solid solder joint.

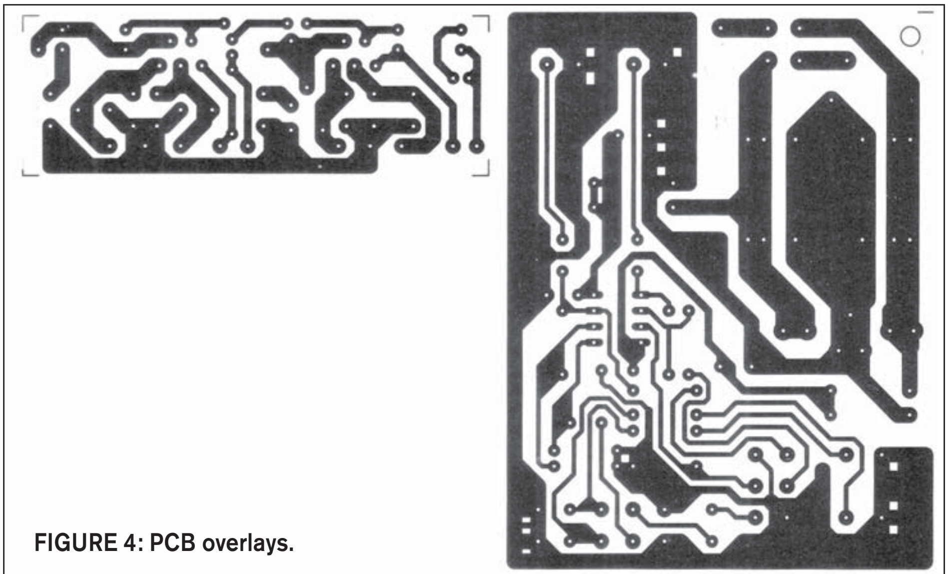

FIGURE 4: PCB overlays.

Don't fit IC1 to the larger board at this point. In fact, you may want to fit a good-quality 8-pin DIL socket on the board instead, to allow you to conveniently plug in the dual op amp later on.

To complete the PCB assembly for the present, you can fit dual-ganged pot RV1 on the front of the larger board, pushing its six pins through the PCB holes until the wider sections are all resting against the top surface of the board, and the spindle axis is parallel to the board plane. Then solder all six pins to the board copper to provide a solid bond. Note that neither the mono/stereo switch nor the two RCA input sockets are mounted on the larger PCB; these components mount on the front of the box, with short leads connecting them to the PCB when it's fitted behind them.

Place both board assemblies aside, while you prepare the box. This mainly involves drilling, reaming, and cutting holes in the front and rear panels—plus three in the base and another in the right-hand side flange. All of the hole locations and dimensions are shown in the metalwork drawing (**Fig. 2**).

Note that in addition to the mounting holes in the rear panel for the speaker terminal block, there are also clearance holes for the "tail" of the terminals, and also locating holes for the plastic spigots (grommets) beneath the tails. Also note the holes in the rear panel for mounting the finned heatsink and the two output devices; you also need to drill matching holes in the heatsink itself. These are all on the heatsink's center axis (26mm in from either end), and at 28mm and 90mm spacing.

On the front panel, the holes for the volume control, mono/stereo switch, and RCA input sockets are all circular, so you just need to drill and ream them. Note that the holes for RCA sockets need to be large enough to accept plastic insulating sleeves.

The larger holes for the mains fuse holder and power switch are both irregular, so you may need to prepare them by hand, which will involve careful work with a drill, nibbling tool, or small jeweler's files.

With all holes prepared in the lower half of the box, you can also cut the small clearance notch in the bottom of the right-hand side of the top, to clear the support screw and pillar. Then your two box halves should be ready to paint.

I prepared the prototype's box by lightly rubbing the outside of both halves with fine glasspaper, wiping clean and then giving them a thin coating of spray-on etch primer. When this dried, I applied a coating of spray-on matte black enamel, which gave it a presentable finish (**Fig. 5**).

After the enamel fully dried, I applied a stick-on dress label made from the artwork shown. This very thin aluminum with an adhesive backing is sold in A4 sheets so you can print them in a standard laser printer. It isn't cheap, but it produces a nice dress label. To provide guidance when the label was applied to the front panel, I punched 5mm guide holes on each of the holes' centers, using a hand-held leather punch. Then when the label was in place, I carefully enlarged the label holes to match those in the panel, using a very sharp hobby knife (always cutting toward the rear of the box, to prevent lifting and tearing).

With the box finished and label applied it's time to start the final assembly operation. I recommend that you do things in the following order; otherwise, you may find some aspect of the assembly much harder than necessary.

MAINS WIRING

First, I suggest that you fit the mains cord entry grommet and bring the mains cord into the case through it, threading the "P" clamp onto the cable but not attempting to bolt the clamp into position just yet. Then carefully strip back about 70mm of the cable's outer sleeving from the end to reveal the three insulated wires. Then poke all three wires through the hole in the

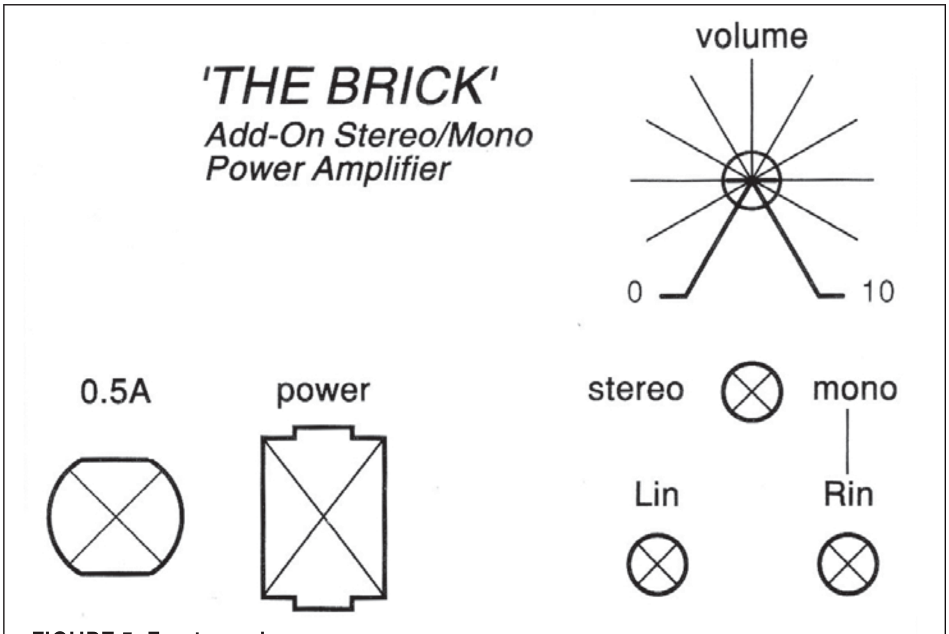

FIGURE 5: Front panel.

front panel intended for the power switch to make it easier to fit the switch wiring.

Bare (strip) about 5mm of insulation from the end of the blue (green in US) (neutral) wire, and also prepare another short (about 60mm) length of similar wire with blue main-rated insulation, by baring about 5mm of the insulation at one end of it as well. Then slip both bared ends through a 20mm length of 8mm diameter heatshrink tubing and finally solder them both to the "neutral" connection lug on the back of the switch—this is the outer lug with a small copper-colored wire welded to it. When the solder has cooled, push the sleeving over both the solder joint and lug until it reaches the switch body, and then heat it all around with the soldering iron barrel until it shrinks to snugly cover all exposed metal.

Now prepare two short lengths of wire with brown or red mains-rated insulation, one 70mm long and the other about 90mm long. Again bare about 5mm from one end of each, and slip a 20mm length of heat shrink sleeving over each before soldering them to the two remaining switch lugs. The *longer* wire connects to the center lug of the switch, while the shorter one goes to the remaining outer lug. After the solder has cooled in each case, slide the sleeving up against the switch body to cover all exposed metal and heat it to shrink and hold it reliably in position.

Now you can move the switch and cord back through the holes in the box panel to fit the switch into place. Note that the switch is oriented so that the lug with the mains cord's blue neutral wire attached is toward the bottom of the box.

Mount the fuse holder to the adjoining hole in the front panel with its side solder lug facing outward (i.e., away from the switch). You should then be able to make the fuse holder connections in much the same way as those for the switch. The shortened red wire from the top switch lug connects to the *side* lug of the fuse holder, while the wire connected to the rear lug is the brown wire of the mains cord (i.e., the incoming mains active)—which is first shortened to about 40mm long, before baring the 5mm for soldering. As before, each wire should have a 20mm length of heat shrink sleeving slipped over it before soldering, and then brought forward and shrunk down to cover all exposed metal.

With the switch and fuse holder wiring complete, you're left with red/brown and blue wires ready to connect to the transformer primary and the green/yellow mains cord earth. I'll deal with these presently. For now, you can gently pull the mains cord back through the rear grommet, until there's just a small amount of slack in the active wire connecting to the rear of the fuse holder. Then you can clamp it in position via the P-clamp, using a 3mm × 12mm machine screw passing through the hole nearest the grommet, with flat washer, star lock washer, and nut.

Next, I suggest that you attach the finned heatsink to the upper rear of the box, using 3mm × 5mm machine screws, lock washers, and nuts through the end-holes—making sure that the two holes near the center are carefully lined up in the process. Similarly, you can fit the four-way loudspeaker terminal block underneath, again using 3mm × 5mm screws, lock washer, and nuts through the end holes.

With the terminal block fitted, cut two 90mm lengths of red-insulated hookup wire, and a single 150mm length of black insulated wire. Solder on the end of each red wire to an end lug; these will be used to connect to the power amp PCB when it's fitted above. The longer black wire is bared by about 18mm, and connects them to *both* of the center lugs—it will be used to connect them to the power supply earth. For the present, simply run the wires out the sides of the box, after perhaps baring about 5mm of their ultimate connections.

At this stage it's probably best to fit the RCA input sockets and the mono/stereo switch to the front panel. Note that the RCA sockets are fitted with insulation sleeves and washer to avoid earth loops. The sleeves are punched through the panel holes first, from the front, and then trimmed to length behind the panel with a sharp knife. The sockets are then fitted with the insulating fiber washers and finally the large solder lugs and nuts.

After fitting the sockets and switch, I suggest that you prepare 10 × 35mm lengths of tinned copper wire, and carefully solder one to each of the sockets and switch lugs—preparing them for connection to the front end PCB, when it's finally fitted. After soldering the wires, you can fit each with 25mm lengths of insulating sleeving, if you wish, and carefully bend them outwards by 90° so that they're roughly ready to mate with the PCB.

The next step is to fit the power transformer into the case (240V primary winding lugs facing outward). Using 4mm × 10mm machine screws, flat and star washer, and nuts, you'll find that the transformer body will just make contact with the sleeved wires at the rear of the mains switch, but without causing any problems. Similarly, the washers and nuts for the front-most mounting screw will be under the same sleeved wires, but it should be relatively easy to hold the nut with a spanner or needle-nose pliers while you're tightening the screw.

By the way, ensure that the *rear* of the transformer body is 26mm from the inside surface of the box, before you finally tighten the screws. This is to make sure that there will be room to mount the power amp PCB.

With the transformer firmly held in place, you can then fit another lock washer, solder lug, and nut to the front mounting screws for the mains earth (ground) connection. The mains cord earth wire (green/yellow) is soldered to this solder lug.

You should now also be able to complete the mains wiring by fitting the remaining red and blue insulated wires to the transformer primary lugs. As before, they're fitted with 20mm lengths of heat shrink sleeving, which cover all exposed metal after soldering.

ADDING THE PCBS

You should now be ready for the final stages of assembly. To begin, I suggest that you prepare three 150mm lengths of red, black, and blue insulated hookup wire and another three lengths about 120mm long. In each case, strip 5mm of conductor at each end and tin them carefully. Also prepare two lengths of light-duty shielded audio cable—one 150mm long and the other 120mm long—in this case removing the outer sleeving to about 10mm at each end, and twisting the braid together before baring 5mm of the inner conductors and then tinning them all, ready to make joints with PCB pins.

Solder one end of each of these wires/cables to the terminal pins along the "front" of the smaller power amp PCB to make all of its interconnections (apart from the speaker wires). The longer 150mm wires are used for the "right" channel connections (IC2 side), with the 120mm wires used for the IC3 side. I used the red wires for the +12V leads, the black for the power earth leads, and blue wires for -12V leads. The shielded cables are used for the signal connections, of course, with the braids connected to the Er and El terminal pins.

At this stage it's probably best to mount the completed power amp board assembly into the rear of the box, where it just fits behind the power transformer. It's held in place by the mounting screws for IC2 and IC3, but this operation is a little tricky because the ICs must be fitted via the usual mica insulation washers and sleeves—together with thermal conducting compound to ensure that their heat dissipation is conducted to the box rearwall and finned heatsink.

I tackled this by first cleaning any paint from the inside of the box rear, around the mounting holes, and then smearing a thin layer of compound over one side of the two mica washers and finally placed each washer carefully into position, so that its screw hole lined up with the panel holes—and with its "unsmeared" side toward the rear, so that it contacted the compound already on the panel.

The compound then held the washer in place while I introduced the PCB assembly and lined up the IC mounting holes with the washer/panel holes, before passing 3mm × 10mm machine screws through from the rear. Then it was a matter of slipping on the insulation sleeves, flat washers, lock washers and nuts, before tightening both screws carefully to mount everything in place.

With the power amp PCB fitted, you can bring the two red speaker wires around from underneath, and solder them to the Ro and Lo output pins. This essentially completes the power amp wiring. Next, I suggest that you lay the front end PCB assembly on the table alongside the right-hand side of the box (looking from the front), with the volume pot toward the front. This should allow you to make the connection between this board and the power transformer secondary, so you can test the power supply.

POWER TESTING

It's a good idea to do a quick functional check of the power supply at this stage, before connecting the power amp and plugging in the front-end IC. So with only the three wires from the transformer connected to the front-end PCB near the rectifier bridge, try plugging in the mains cord, turning on the power, and then measuring the voltage between the "PAE" (power amp earth) pins and the "+21V" and "-21V" output pins (between C22/24 and R19). If these voltages check out correctly, try checking the voltages at pin 8 and 4 of the socket for IC1—again, with respect to the "PAE" pins they should measure very close to +12V and -12V, respectively.

If all four voltages are OK, turn off the power, unplug the mains cord, and you're ready for the final stage of assembly. This involves first connecting the various wires from the power amp PCB over to the power supply and front-end circuitry—and also connecting the black wire from the speaker terminal block to the pin near C22/C24 marked "SPKR" on the overlay.

Once you've connected all these wires, you can swing the front-end PCB up into the vertical position, and carefully introduce it into the box so that the spindle and sleeves of the volume control slip into its holes in the front panel. (You may need to slip a couple of thin spacer washers behind to prevent it protruding too far.) Then you can fit the nut, but tighten it up only lightly for now.

You should now be able to introduce a 3mm × 10mm machine screw through the hole in the bottom side flange of the box, threading it with a 5mm-long spacer, and then passing it through the mounting hole in the lower back corner of the front-end PCB. You can then fit the screw with a star washer and nut, after which you can tighten up both this screw and the volume pot mounting nut, to fasten the PCB into its correct final position.

The second-last step of all is to take a pair of tweezers or needle-nose pliers and carefully push the end of each wire "tail" from the RCA input sockets and the stereo/mono switch through its corresponding hole in the lower front of the PCB—making sure that the wires are neither twisted together, nor unduly strained. This done, you should be able to solder each one to its PCB pad and clip off any excess. Your Brick amplifier should now be finished and ready for final testing and assembly.

There are no adjustments to be made, so testing should largely be a matter of connecting it up to a pair of speakers and a source of mono or stereo signals, applying power, and listening to the music. If all seems well, you can switch it off and fit the top of the box, with the usual four mounting screws. If you're concerned about the proximity of the sleeved transformer primary connections to the side of the case, you might want to attach a protective inverted-L cover of "elephant hide" or similar fireproof insulating sheet over that side of the transformer, before fitting the case top.

As an enhancement, you might also want to fit four self-adhesive rubber mounting feet to the bottom of the box, to prevent it from scratching the surface it rests on. Otherwise, your Brick amplifier should now be complete and ready for use. Happy surround-sound listening and home theater viewing! *aX*

PARTS LIST

Resistors

All 0.25W 1% metal film unless specified

R1, 2, 5, 6	2.7K
R3, 4	270K
R7, 8	5.1K
R9	2.4K
R10, 11, 14, 15	82K
R12, 13	4.7K
R16, 17	2.7Ω
R18, 19	470Ω
RV1	dual 50K log taper, 16mm diameter

Capacitor

C1, 2	470pf
C3, 4	0.15μF MKT
C5, 6, 11, 13, 15, 17	0.1μF MKT
C7, 8	1μF MKT
C9, 10	4.7μF/16VW RB electrolytic
C12, 14, 16, 18	220μF/25VW
C19, 20	0.22μF
C21-24	2200μF/25V
C25-26	1000μF/16VW

Semiconductor

IC1	TL072
IC2, 3	LM1875 power amplifiers
ZD1, 2	12V1W zener diode
BR1	W04 1.5 rectifier bridge

Switches

SW1	DPDT miniature
SW2	SPDT mini mains +rated switch with indicator

Miscellaneous

Two PC boards, one 89 × 69mm, the other 75 × 25mm
30V/1A power transformer, tapped at 15V
metal case, 125 × 100 × 75mm
finned heatsink, 52 × 100mm

By Aren van Waarde

The “Modern Homodyne”

How to build a vintage broadcast band receiver clone.

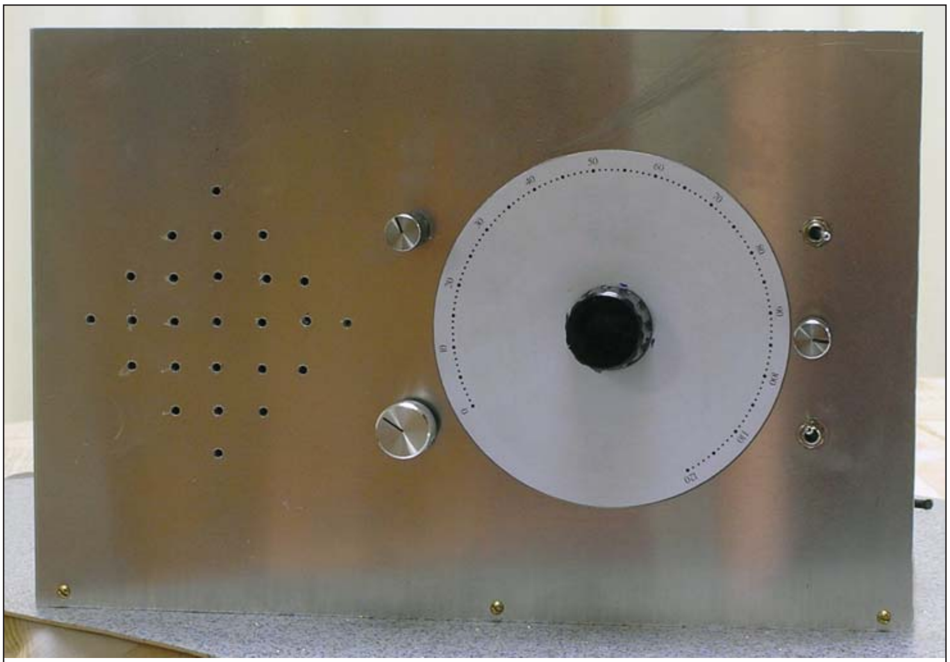

PHOTO 1: Front panel of the Homodyne receiver.

How were you infected with the DIY virus? Possibly we caught it in a similar way. In the 1960s, Dutch children listened to new songs, such as those from an obscure band named The Beatles, on Radio Luxemburg. A powerful pirate transmitter (Radio Veronica) broadcast from a ship in international waters and could be received throughout The Netherlands. All Dutch children wanted to hear these programs, but most parents (including mine) did not allow the use of the home receiver to play such modern “trash.”

Thus, my brother and I mounted a long wire between our bedroom and our barn. This served as an antenna, and with a homemade crystal set and a reasonable ground, Radio Veronica came in “loud and clear.” We built the set in an empty marmalade jar with the tuning capacitor mounted on its metal lid, and the tank coil wound on a cardboard tube fitting snugly in the jar. We hid this miracle of technology behind the books on our shelf, and settled back for bedtime listening[1].

That crystal set started it all. In subsequent years, I built several radios of increasing complexity—most of them from kits—and with varying results. From there I progressed to stereo sets and loudspeakers. But I never forgot the fun of building a simple receiver, and using it to listen to programs from foreign countries. The finest signals come from a homebrew rig!

My interest in constructing a simple broadcast band (BCB) receiver was revived when I discovered the “Vintage Radio and Electronics” website of Maurice Woodhead (http://vintageradio.me.uk). Maurice has scanned many construction articles from defunct British magazines, such as *Radio Constructor*, *Practical Wireless*, *Radio and Electronics Constructor*, and *Wireless World*. Projects from the 1960s and 1970s are well represented. Reading this material brought back fond memories. In the Dutch town where I was born, we were not familiar with these UK magazines, but we had periodicals of our own, such as *Radio Bulletin*, *Radio Elektronica*, *Radio Blan*, and *Hobbyskoop*. Sadly, none of them survived.

One article on Maurice’s site describes a simple radio that employs an unusual method of demodulation, called “homodyne” or “synchrodyne” reception[2]. A receiver using the same technique was published in 1973 in the Dutch magazine *Elektuur*. The Dutch radio did not employ any coil, because it used RC rather than LC filters for tuning[3]. Its construction was very easy, but it was difficult to tune. You could clearly receive local medium-wave (MW) stations during the daytime. But after sunset, its poor selectivity became evident and you heard many stations at the same time as a confused jumble (I know because I made one!).

Later I discovered that US ham radio operators built copies or modified versions of the circuit on Maurice’s site[4]. This fueled my interest even further and I decided to give the circuit a try (**Photo 1**). As you will learn, the “Modern Homodyne” has an excellent performance/complexity (or performance/price) ratio, and is great fun to build and use.

HOMODYNE RECEPTION

So, what is a homodyne? You are probably familiar with the superheterodyne method of reception (**Fig. 1**), in which the incoming signal from the antenna is mixed with the signal of a local oscillator. That oscillator is tuned to a frequency higher (or lower) than the desired signal; that is, to a *different* frequency. For this reason the radio is called a heterodyne (in the Greek language, “heteros” means different).

The distance between the signal frequency and the frequency of the local oscillator is kept constant during receiver tuning, and the difference frequency is greatly amplified by an “intermediate frequency” (IF) amplifier. A detector extracts the audio information from the IF signal, and after additional AF amplification, you hear the radio program. Adequate selectivity is provided by tuned LC-filters in the IF amplifier.

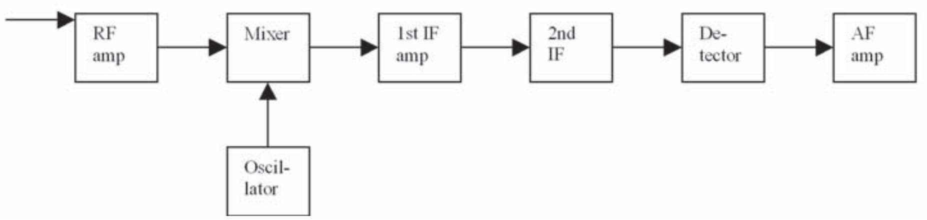

FIGURE 1: Basic setup of a superheterodyne receiver. The desired signal may be an AM station at 1008kHz. The local oscillator is tuned to 1478kHz. The difference frequency of 1478 – 1008 = 470kHz is amplified by two IF stages. A detector extracts the audio information from the amplified 470kHz IF signal.

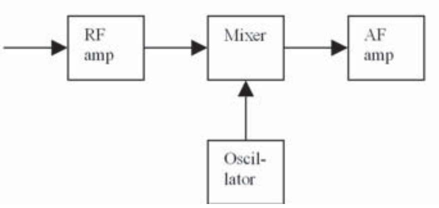

FIGURE 2: Basic setup of a homodyne receiver. If the desired signal is the same AM station at 1008kHz, the local oscillator is tuned to 1008kHz. The difference frequency is 0Hz for the carrier, but 20 to 4000Hz for the sidebands. The mixing process results directly in the required audio information.

A homodyne, or "direct conversion," receiver operates differently (**Fig. 2**). Here, the incoming signal is mixed with the signal from a local oscillator, which is tuned to the *same* frequency as the desired signal. For this reason, the radio is called a homodyne ("homoios" means same or similar).

The difference between the signal and oscillator frequencies is zero cycles per second, or DC. However, an AM signal consists of the carrier (i.e., the transmitter) frequency *plus* sidebands. These sidebands are on slightly different frequencies than the carrier, and after mixing with the signal of the oscillator which is tuned to the frequency of the carrier, difference frequencies are generated, which happen to be exactly identical to the original audio.

Suppose the station is at 1008kHz and is amplitude-modulated by a 1.5kHz sine wave. Two sideband frequencies are then present in the radio signal—one at 1006.5kHz and the other at 1009.5kHz. The difference between the carrier and sideband frequencies is 1.5kHz in both cases, i.e., the original modulation. A station on an adjacent channel will be on 1017 or 999kHz, i.e., at 9kHz difference from the frequency of the local oscillator. You can suppress the resulting high-frequency whistle of that undesired station with a steep low-pass filter (cutoff frequency 5kHz or less).

So, after just a single mixing stage you immediately acquire the desired audio, and receiver selectivity is provided not by two or three IF transformers, but by a simple low-pass filter. "A neat approach," you may think. "Could save a lot of parts."

So, why isn't every commercial receiver a homodyne? Well, there are trade-offs, as usual. The IF amplifier in a superheterodyne receiver consists of multiple stages resulting in a large overall gain, but the homodyne in its simplest form consists of only a single stage (combined mixer and local oscillator) and can provide only a limited amount of amplification.

Thus, a simple homodyne will be less sensitive than a superhet. Also, you should tune the local oscillator to *exactly* the same frequency as the station carrier, and its frequency should not drift. You can only achieve this by synchronizing the oscillator to the station carrier; i.e., when the oscillator is tuned and approaches the station frequency, the carrier should "pull" the oscillator so that the oscillator and station frequencies become equal. This isn't easily achieved. For good detection, the signal of the local oscillator should be much stronger than the station signal, but increasing the oscillator amplitude makes synchronization more difficult.

Detailed information on homodyne reception, as well as literature references to original sources, is provided on various websites[5].

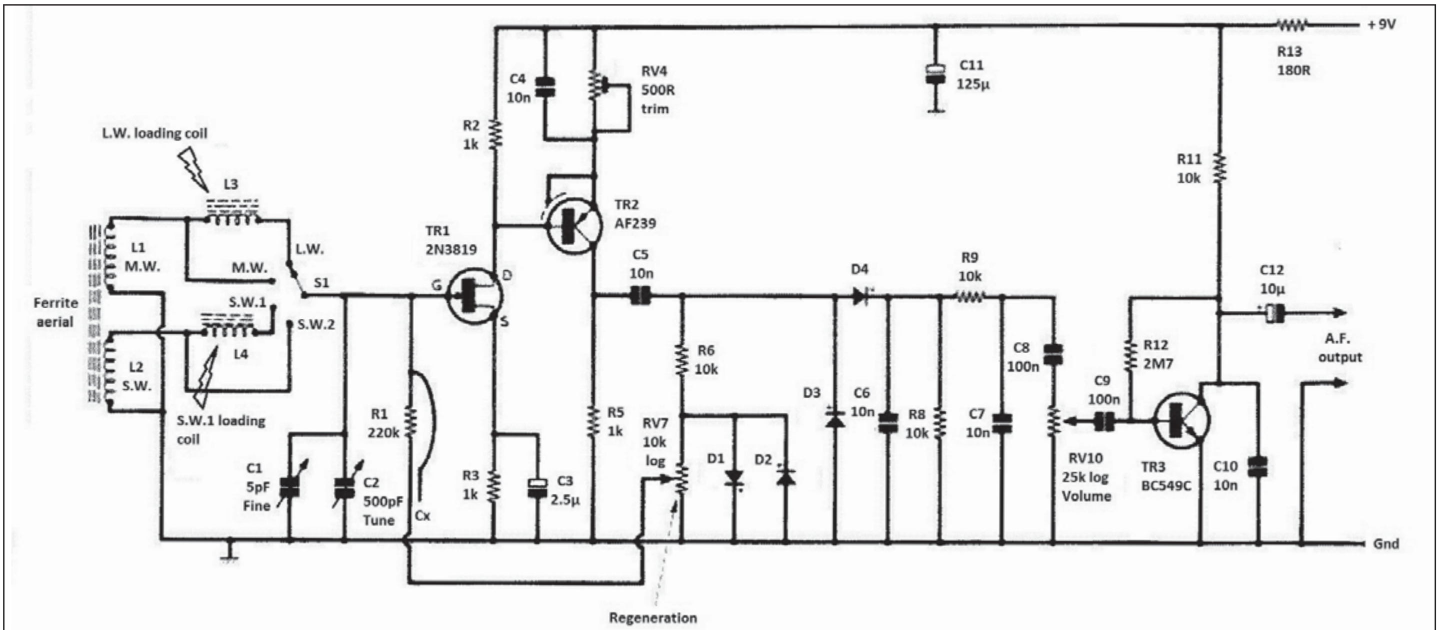

FIGURE 3: Schematic of the front end of G.W. Short's homodyne receiver.

SHORT'S HOMODYNE

The homodyne presented on Maurice Woodhead's website was designed by George W. Short and was published in the March and April 1972 issues of *Radio Constructor*. The schematic of the receiver front-end is shown in **Fig. 3.** As you will notice, it is extremely simple, consisting of just three transistors of which only two operate at radio frequencies.

The signal is picked up by a ferrite rod antenna which is tuned by C2, an air variable of 500 (or 365) pF. FET TR1 serves as a buffer stage providing a high input impedance. Thus, the LC circuit is not loaded much and the antenna coil does not need any tap.

Short's receiver contained a band switch S1 and could be tuned to long-wave, medium-wave, and two shortwave bands. Its rod antenna was made of a special type of ferrite (designed for use at high frequencies) to allow shortwave reception. Because I did not have such an antenna, I limited myself to MW reception and used an old rod antenna salvaged from a damaged tube radio.

RF gain is provided by TR2, a germanium transistor which is still available from several sources. AF239s from large mail-order companies can be expensive (up to $8), so I purchased ten of these transistors from a small company in Germany for only 20 cents each, through eBay. If you don't like the idea of using germanium transistors, read on: a modern alternative will be provided.

You should initially adjust RV4 for 3mA collector current of TR2. You can later vary its setting (between 1 and 4mA collector current) for optimal quality of reception. You can easily determine the collector current of TR2 by measuring the voltage drop across R5 (drop in volts equals current in milliamperes).

RV7 is the "regen control." This potentiometer controls the amount of positive ("regenerative") feedback from the output of the RF amplifier TR2 to the input of buffer stage TR1. When RV7 is turned up, a point will be reached at which the RF section (TR1 plus TR2) will start oscillating. Cx is a gimmick capacitor (a few turns of insulated wire twisted around the lead of R1 which is connected to the RV7 wiper). It compensates for phase shift in TR2 at the higher frequencies.

D1 and D2 limit the maximal amplitude of oscillation when the regen control is turned up. D3 and D4 form the detector, followed by a simple low-pass filter (C6, R9, C7). R8 provides a rather heavy load to the detector, resulting in smooth regeneration. TR3 serves as an audio preamplifier with high gain (50- to 100-fold). RV10 is the audio volume control.

AUDIO OUTPUT STAGE

Short's Homodyne employs a very simple AF amplifier using just four discrete transistors (1 × BC168B, 1 × BC169C, 2 × 2N4289). The main advantage of this amplifier was its low battery consumption (less than 1mA quiescent current), but it had, in my opinion, two important disadvantages: a special high-impedance loudspeaker was required (80Ω), and the quality of the audio was rather poor. High impedance loudspeakers are hard to find nowadays.

Because I did not have an 80Ω loudspeaker, I decided to build another AF amplifier based on a TBA800 IC. The schematic is shown in **Fig. 4**. This amplifier was once published in the (long defunct) magazine *Radio Bulletin*[5]. It provides very pleasant audio, can drive an 8Ω speaker, and at the low supply voltages used here, its quiescent current is less than 4mA. The TBA800 was once common in TV sets, but is now obsolete, although it can still be purchased from mail-order companies. A replacement item (NTE1116) is available from Mouser Electronics (www.mouser.com).

You can, of course, use any other audio IC. I am not a fan of the ubiquitous LM386, which is rather noisy and low-powered, but there are many alternatives, such as the TDA7052, TDA2003, and LM380. The manufacturer's datasheets will provide all information required to build an amplifier. If you appreciate good audio, don't use a small loudspeaker, but, instead, choose a speaker with a diameter of 4″ or greater which can handle at least 1W.

CONSTRUCTION

Particularly with simple radios (but, I think, with electronic equipment in general), the mechanical layout and construction are equally important, as is the electrical circuit. You can, of course, design your own PC board for the homodyne receiver. I have built a prototype of Short's radio that has worked well using an ancient breadboard system from Philips (EE 2013). But your Homodyne is more likely to be a successful receiver if you build it according to the layout recommended by *Radio Constructor* (**Fig. 5**). I have employed that layout for the final incarnation of my own radio (**Photo 2**).

You should cut a 4.5 × 2″ piece of plywood (thickness 12mm or greater). Drive small copper nails in this board at the locations indicated in **Fig. 5**. These will become the anchor points to which you solder component lead-outs. If you don't have copper nails, you may also use domestic pins driven into the wood by hand. Then you cut off the heads, leaving about ⅜″ of stem projecting. Tin the nails (or pins) before soldering any component and place a copper shield below this wooden RF board.

According to Mr. Short, you should place another shield made of copper foil above the area marked ABCD and connect it to circuit ground. In my two proto-

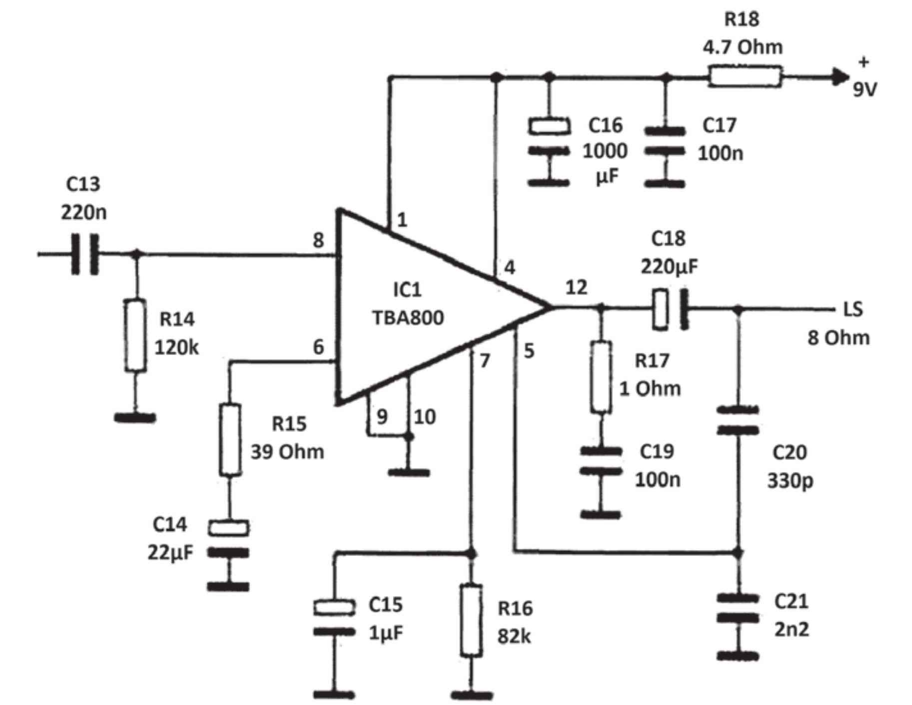

FIGURE 4: Schematic of the audio output stage (the loudspeaker is connected between the negative pole of C18 and ground).

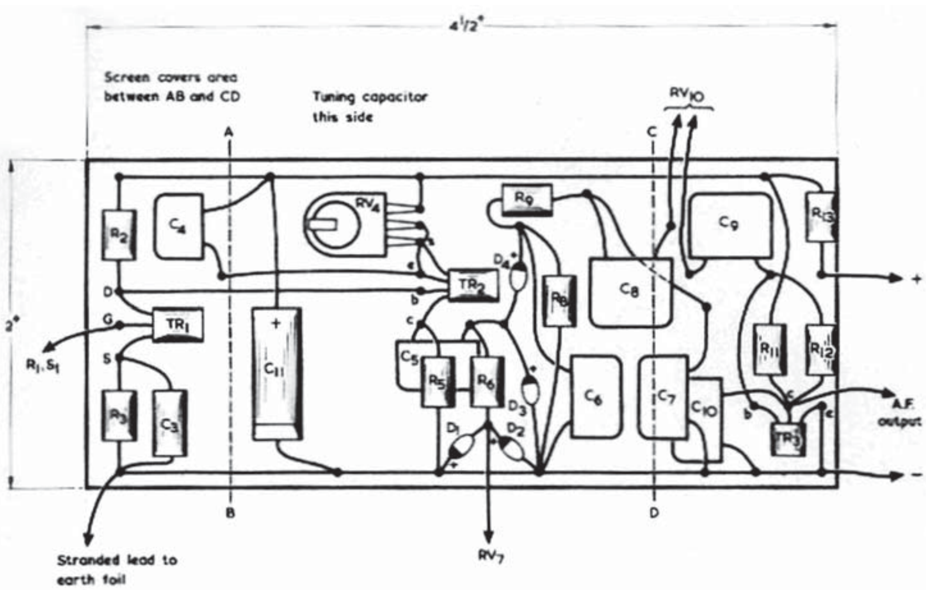

FIGURE 5: Layout of the receiver board (from *The Radio Constructor*, April 1972, p.556).

types, such a shield was not necessary, but for shortwave reception it may be required.

The receiver cabinet consists of plywood bottom, top, and side panels with an aluminum front (**Photo 1**). The (dur) aluminum front panel is 2mm thick and therefore quite sturdy. It is connected to circuit ground in order to avoid capacitive hand effects. Controls are: upper left regeneration, lower left audio volume, center main tuning, upper right long-wave/medium-wave switch (not yet functional), middle right fine tuning, bottom right on/off. The speaker is mounted at the left-hand side of the front panel, behind 25 drilled holes (3mm diameter).

Unfortunately, duraluminum is quite hard (in contrast to normal aluminum), and my drill was old and somewhat blunt, so the holes are rather untidy. Keep the wiring to controls as short as possible, and use wire that is thick and stiff. What you are building is basically a RF oscillator, which should be stable. The tuning scale is made of an old CD. You can download circular scales with the proper sizes from ham radio sites[7], print them on paper, and glue them to the CD.

ACQUISITION OF PARTS

If you don't have a well-stocked junkbox filled with parts salvaged from ancient tube radios, it may be difficult to acquire a suitable ferrite rod antenna and air variables. In the corner of Europe where I live, we have several mail-order companies which can (often) provide them. See AK Modul Bus (www.ak-modul-bus.de), Baco Army Goods (www.baco-army-goods.nl), ElektroDump, Frits Meuris Electronics (www.frimu.nl), Oppermann Electronic (www.oppermann-electronic.de), Pollin electronic (www.pollin.de), or Van Dijken Elektronica (www.vandijkenelektronika.nl). In the US, you could try All Electronics (www.allelectronics.com), Alltronics, Antique Electronic Supply (www.tubesandmore.com), Circuit Specialists, Dan's Small Parts (www.danssmallpartsandkits.net), or Play Things of Past (www.oldradioparts.com) (to mention just a few). I bought my AF239 transistors from a friendly Internet shop in Bavaria, RH Electronic (www.rhelectronic.tradoria.de).

For those of you who hate germanium

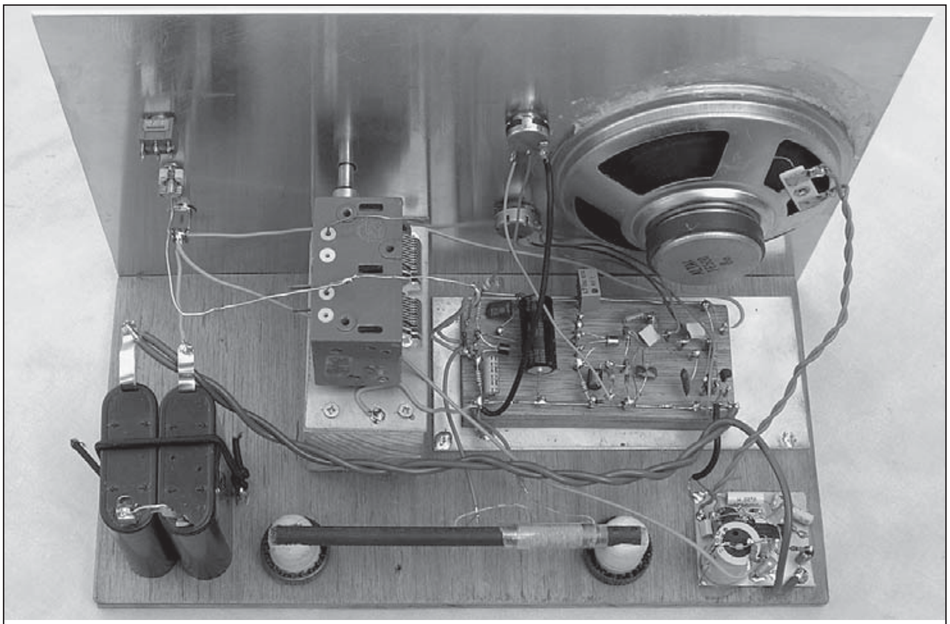

PHOTO 2: Back view of the Homodyne prototype. The RF board is directly behind the loudspeaker, the TBA800 power amplifier board is at the lower right. I used the rear section of a 2-gang 500+500pF variable capacitor, but a single-gang 500 or 365pF capacitor could have done quite as well.

transistors, Jim Kearman (ham radio operator KR1S, from Stuart, FL) has provided a solution: replacing the AF239 with a MPSH10[4]. Slight modification of the circuit is then necessary because the AF239 is a PNP transistor, whereas the MPSH10 is a NPN. Thus, the positions of emitter and collector with its associated components should be reversed (**Fig. 6**). In all other respects, the Kearman circuit is identical to G.W. Short's original.

WHY BUILD IT?

Nostalgia apart, why would you build a minimalist receiver from 1972? The reason is simple: It provides very nice audio after you have learned how to operate it. So nice, in fact, that the "Modern Homodyne" has become the favorite AM BCB receiver in our home. It is mainly tuned to the BBC World Service (648kHz), which comes in "loud and clear." Other favorites are Colourful Radio 747 AM (an extremely powerful local transmitter), Groot Nieuws (Good News) Radio at 1008kHz (a commercial station run by Protestants), Radio Maria at 675kHz (a religious station run by Catholics), BBC Live and Talk Sports (both British), RTBF1 Bruxelles (Belgium), NDR Info/Funkhaus Europa, and Deutschlandfunk (Germany).

All of these can be heard at various frequencies during the day. More stations are heard after dark (e.g., Kossuth Radio from Hungary). During daytime, reception is exceptionally clear and clean. At night, powerful neighboring stations may cause slight interference when a weaker station is being received (it remains a TRF with a single tank circuit, after all!).

After switching on the radio and setting it to moderate audio volume, turn up the regen control until you hear a slight hiss. Then turn the main tuning knob to search for a station. When you approach the station carrier, you'll hear a loud howl. Now turn the tuning knob further, very slowly, in the proper direction. When the point of exact tuning is virtually reached, the howl becomes a low-frequency grumble. Suddenly the oscillator is synchronized with the station carrier and you'll hear the program. Finally, you can adjust the receiver for optimal reception using the knob for fine-tuning. All of this sounds complicated, but is, in fact, very easy after you've got the feel of it. When you find the optimal setting of the regen knob, only slight adjustments are necessary during further tuning.

In his original article, Mr. Short gave the following additional advice: "If the circuit is detuned a little, but not far enough to lose synchronization, the audio output falls and becomes distorted. The best tuning point is usually nearer

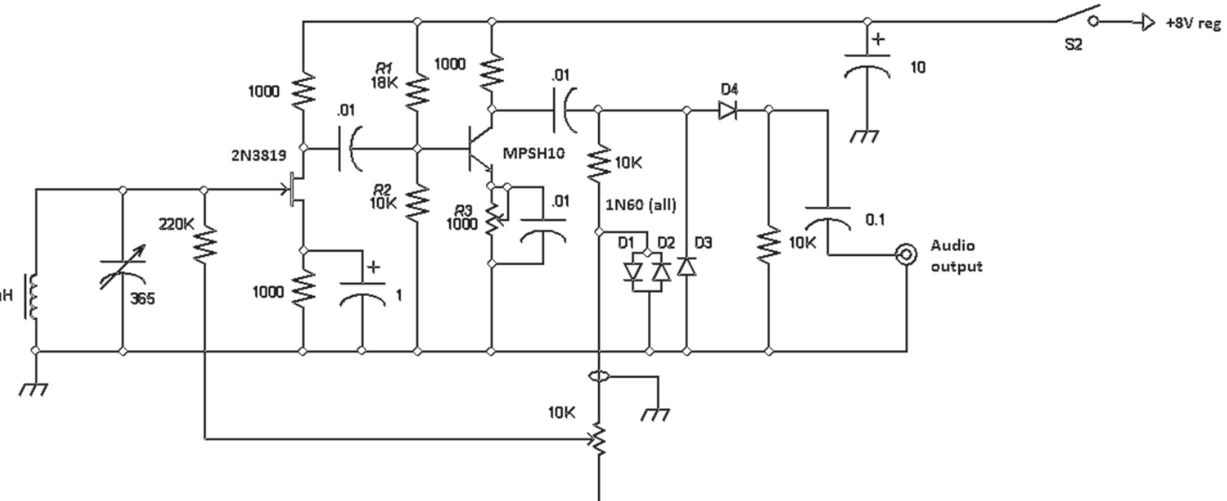

FIGURE 6: Jim Kearman's Homodyne. Identical to the front-end of G.W. Short's circuit, but the AF239 has been replaced by a MPSH10 and emitter and collector of TR2 have been reversed.

one edge of the locking band than the other. Loss of sync may occur momentarily during deep modulation troughs, giving a rasping quality. This is a sign that the circuit is slightly off tune. . . It is an advantage to use the lowest practicable level of oscillation. If you have to increase the level to override a strong signal in the next channel, turn up the reaction, until the intelligible breakthrough turns into unintelligible 'monkey chatter.' This is as far as you need go."

Construction of the "Modern Homodyne" gave me the same kick as building the marmalade jar crystal set or a three-transistor RE2 reflex radio in the 1960s. Perhaps I have never grown up. *aX*

REFERENCES

1. For the sake of historic accuracy, note that the construction plans for the crystal set, plus the tools required to drill holes in the lid of the marmalade jar, were provided by our dad!

2. G.W. Short, "A Modern Homodyne Receiver," *The Radio Constructor*, March 1972, p.492-495 and April 1972, p.552-557. You can download scans of these articles from various websites, e.g., the "Vintage Radio and Electronics" site of Maurice Woodhead, http://vintageradio.me.uk, and the "Selection of Circuits by G.W.Short" website, http://www.radioconstructors.info/gws/gws.html.

3. "Gevoelige spoelloze MG en LG synchrodyne ontvanger," *Elektuur*, juli/augustus 1973, p.760.

4. Mike Tuggle has built a homodyne with basket coil, http://crystalradio.us/1adradios/1ad-2007-2.htm, Dave Schmarder a homodyne with spiderweb coil, http://makearadio.com/solidstate/homodyne.php, and Jim Kearman a homodyne with modern RF transistors http://qrp.kearman.com/html/synchrodyne01.html.

5. In the April 1954 issue of the *Journal* of the British Institution of Radio Engineers, an article was published entitled "The History of the Homodyne and Synchrodyne." You can download the full text of this article from Jon Evans' site, http://www.thevalvepage.com/radtech/synchro/synchro.htm. PDF versions of several articles (in German) are available from the forum of the Europäisches Radiomuseum Luzern, http://www.radiomuseum.org (search in "Forum" for "Homodyne"). These include: Prof.Dr.Berthold Bosch, "Synchron-Homodyn-Empfang," Funkgeschichte Nr.180:104-109,2008, and ibid., Nr.181,148-150, 2008. Werner Häusle, "Homodyn-Demodulation und der Empfänger dazu," Funkgeschichte Nr.180:100-103,2008. If you can't find the PDFs, drop me a line (at a.van.waarde@hetnet.nl) and I will send them to you.

6. G.J.M.van de Werff, "Signaalzoeker en -gever," Radio Bulletin, mei 1981, p.27-28.

7. E.g. Gary Johansen's site, at http://www.qsl.net/wd4nka/TEXTS/Tubepadf.html.

TABLE 1: PARTS LIST (RF SECTION AND AUDIO PREAMP, AS PUBLISHED BY G.W.SHORT)

Schematic ID	Value
TR1	2N3819 JFET
TR2	AF239 Germanium PNP transistor (UHF type, see text)
TR3	BC549C (was originally BC169C)
D1-D4	1N60 (were originally OA90)
R13	180Ω 0.25W 5%
R2,R3,R5	1k 0.25W 5%
R6,R8,R9,R11	10k 0.25W 5%
R1	220k 0.25W 5%
R12	2M7 0.25W 5%
RV4	500Ω trimpot
RV7	10k potentiometer, logarithmic, metal case (Regeneration control)
RV10	22k potentiometer, logarithmic (Audio volume)
C4,C5,C6,C7,C10	10nF metallized polyester
C8,C9	100nF metallized polyester
C3	2.2 or 3.3μF electrolytic 16V
C11	220μF electrolytic 16V
C12	not mounted (since the power amplifier has an input capacitor)
Cx	gimmick capacitor, wire with solid core and plastic insulation wound a few turns around the lead of R1
C1	5pF air variable (fine tuning, convenient but not strictly necessary)
C2	365 or 500pF air variable (good quality, if possible reduction gear)
L1	Ferrite rod antenna with BCB coil
L2	In Short's original, SW coil on ferrite rod antenna (missing in my version—if multiband reception including 2 shortwave bands is required, a ferrite rod specially designed for use at high frequencies should be employed)
L3	2.5mH r.f. choke (available from Antique Electronic Supply)
L4	Shortwave loading coil (missing in my version—it consisted of 20 turns of enamelled wire (0.7mm diameter) close-wound on a small ferrite rod, 45mm $\times$ 10mm, also special ferrite for high frequencies)

TABLE 2: PARTS LIST (OUTPUT AMPLIFIER)

Schematic ID	Value
IC1	TBA800
R17	1Ω 0.5W 5%
R18	4.7Ω 0.5W 5%
R15	39Ω 0.25W 5%
R16	82k 0.25W 5%
R14	120k 0.25W 5%
C20	330pF styroflex, polypropylene or metallized polyester
C21	2n2 metallized polyester
C17	100nF ceramic
C19	100nF metallized polyester
C13	220nF metallized polyester
C15	1μF metallized polyester
C14	22μF electrolytic 16V
C18	220μF electrolytic 16V
C16	1000μF electrolytic 16V

Eliminating Speaker Reflections with Digital Filters

Discover how digital filters can improve your audio system.

During my years of speaker building, I have consistently encountered one problem, especially for tweeters: unwanted driver reflections originating from the speaker cabinet. You can reduce cabinet reflections by smoothing the edges of the enclosure; however, even the proximity of other drivers mounted on the same baffle can cause reflections. I had become resigned to live with cabinet reflections, because I didn't see any way to eliminate them. However, digital filters seem to offer endless possibilities to do things that were never possible in the analog domain, and when I started working with them it occurred to me that they might be useful for reducing these reflections.

I have read a variety of opinions—most of them negative—regarding the use of digital filters for the purpose of "room correction." It is very tempting to try to correct every possible anomaly with digital filters, to obtain a perfect response. However, this "perfect response" is obtained only for one listening position; if you move away from the sweet spot, digital filters can actually make things much worse.

In this article I attempt a more modest goal. I am not trying to correct for any anomalies introduced by the room in which the speaker is placed; rather, I am only addressing the early reflections that arise from the speaker cabinet itself. This seems to me a promising approach since the cabinet reflections are relatively constant and do not depend on such factors as the distance of the speaker from walls. However, I should mention that even with this approach there is still a directional issue, because cabinet reflections are different depending upon the angle of measurement.

Before going into a specific example of reducing tweeter reflections, it is valid to first ask the question whether it is possible even in theory to eliminate reflections using a digital filter.

THEORY

Figure 1 shows an example I used for a theoretical analysis. In this figure, I calculated the pulses using functions of the following form:

$$f(t) = \frac{A}{1+\alpha\left(t - t_{offset}\right)^2}, \qquad (1)$$

where A controls the amplitude of the pulse, α the pulse width, and t_{offset} the location of the pulse. In this particular example, the reflection amplitude is 20% of the primary response, and the reflection width is a little over four times broader than the primary pulse width. The upper trace first shows the primary response of a hypothetical driver, that would be obtained if there were no reflections, followed by a single reflection. The lower trace shows the result I would like to achieve; namely, the driver's primary response alone.

Figure 2 shows part of the magnitude of the frequency response calculated for the upper trace of **Fig. 1**; the ripple evi-

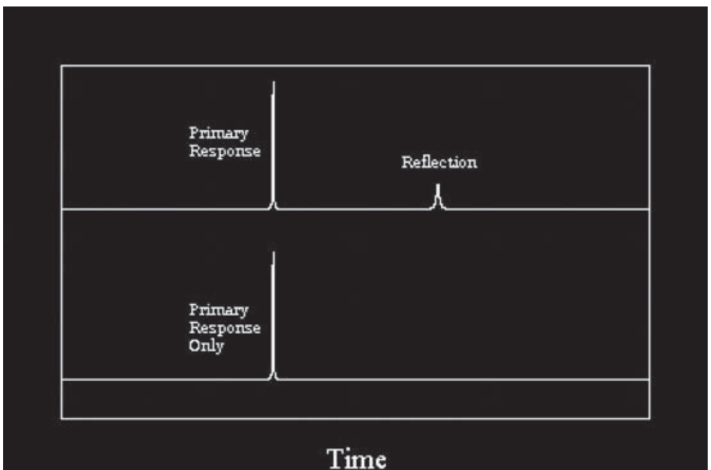

FIGURE 1: Simple primary response and reflection for theoretical analysis.

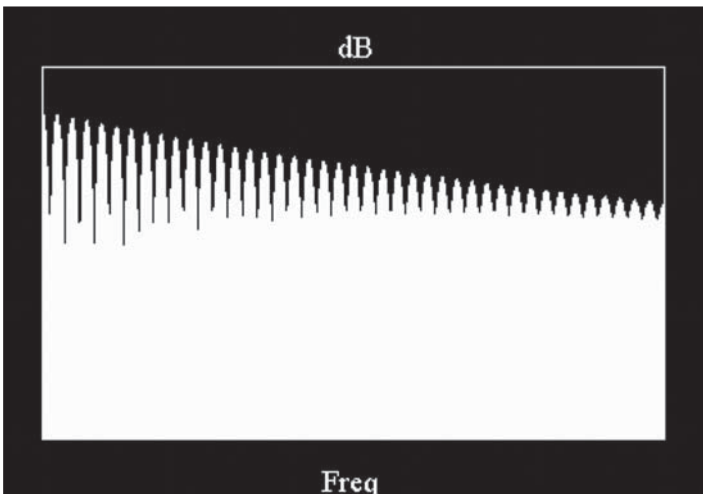

FIGURE 2: Magnitude vs. frequency for the upper trace in Fig. 1.

dent in the magnitude is a hallmark of reflections. The magnitude of the ripple depends on the amplitude of the reflection, and its frequency depends on the time delay between the primary response and the reflection.

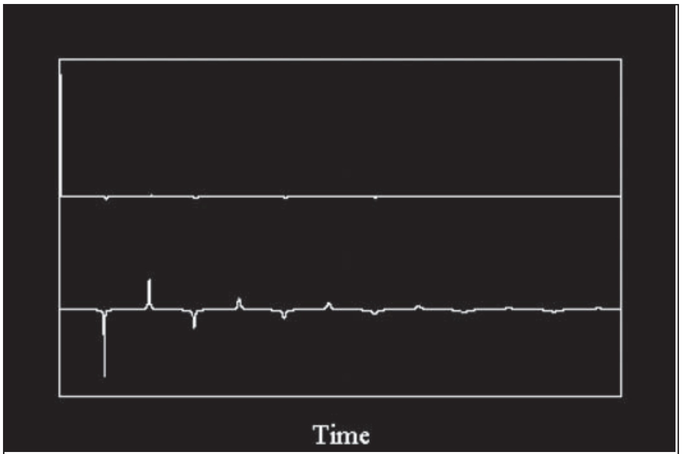

FIGURE 3: Time-domain filter h(t) that removes the reflection in *Fig. 1*.

Is it possible to construct a filter that will transform the upper trace in **Fig. 1** into the lower one? Let P(ω) denote the primary speaker response, and $\exp(^{-j\omega\tau})$ R(ω) the reflection, both in the frequency domain (ω = 2πf). (The $\exp(^{-j\omega\tau})$ factor is the delay operator in the frequency domain, where t is the time delay. It is assumed that R(ω) is the transform of the reflection centered at t = 0).

I want a filter characteristic, denoted by H(ω), such that:

$$(P(\omega) + \exp(^{-j\omega\tau})\, R(\omega)) \times H(\omega) = P(\omega) \quad (2)$$

or, re-arranging this equation:

$$H(\omega) = \frac{1}{1 + \exp^{(-j\omega\tau)} R(\omega)/P(\omega)} \quad (3)$$

Since the ratio R(ω)/P(ω) is generally small, you can write the solution to eq. 3 as an infinite series of terms that are progressively more delayed and progressively smaller in magnitude, as follows:

$$H(\omega) = 1 - \exp^{(-j\omega\tau)}\left(\frac{R(\omega)}{P(\omega)}\right) + \exp^{(-j2\omega\tau)}\left(\frac{R(\omega)}{P(\omega)}\right)^2 - \exp^{(-j3\omega\tau)}\left(\frac{R(\omega)}{P(\omega)}\right)^3 + \ldots$$
(4)

The filter described by eq. 4 will completely remove a single reflection, R(ω), located at t = τ. Because the terms in eq. 4 quickly become small, even if you use a truncated series, it can still remove the reflection to a very good approximation.

The top trace in **Fig. 3** shows a portion of the calculated result for the time-domain filter h(t), the transform of H(ω), which removes the reflection in **Fig. 1**. The filter h(t) begins with a unit impulse function, which reproduces the primary response in **Fig. 1**. The small positive and negative peaks following the initial unit impulse correspond to the terms in the series of eq. 4; the second trace in **Fig. 3** shows a magnification of these terms by a factor of 20.

Of course, a real situation will be much more complicated than the ideal case treated here. In general there will be more than a single reflection, and the form of the reflections may be more spread out than the localized pulse I have assumed. However, using digital processing techniques, you can still obtain a filter h(t) that significantly reduces the reflections. The next section

PHOTO 1: Speaker enclosure used for the measurements of *Fig. 4*.

presents a practical application of this idea.

EXAMPLE

I obtained the results in this section using the Virtual Crossover, which is a much improved version of the device I first presented in *audioXpress*[1-3]. The Virtual Crossover is a measuring device, but it also allows crossover filters to be placed in memory, either for measurements or for playing musi c. Here I use the Virtual Crossover to first measure a tweeter, and in the final step I load a reflection-correcting filter and measure the tweeter again, to verify that speaker cabinet reflections can be eliminated. If you would like more information about the Virtual Crossover, I am working on some documentation for my website at www.acoustic-instruments.com.

The example in **Fig. 1** assumes that the exact nature of the reflection was known. In a practical situation, it would require extensive analysis to determine the exact form of the reflections to expect from a speaker enclosure. Rather than attempting such a complicated analysis, I tried a much simpler approach.

Figure 4 shows a portion of the impulse response of a Vifa H26TG-35 tweeter mounted in the speaker enclosure (**Photo 1**). I obtained this impulse response using a MLS sequence of order 16 (65535 points) with a sampling rate of 96kHz. I placed the microphone on-axis about 6′ away from the tweeter. After the primary tweeter response, you can see evidence of early reflections from the speaker cabinet, and perhaps also from the adjacent midrange drivers mounted in the enclosure. Also in **Fig. 4** you can see the first prominent room reflection that comes from the floor, a little before 3ms in the figure.

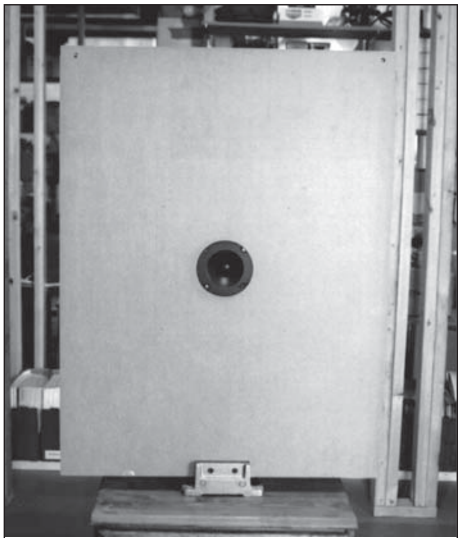

PHOTO 2: 24 × 32″ baffle used for reference measurements.

Truncating the impulse response in **Fig. 4** at the vertical line to remove the effects of this and subsequent reflections and performing an FFT, you obtain the frequency response of the driver plus early cabinet reflections shown in **Fig. 5**. The significant ripple in **Fig. 5** illustrates the problem, showing clear evidence of reflections.

Next comes the task of removing the reflections evident in **Figs. 4** and **5**. As mentioned previously, I did not attempt to identify the reflections from first principles. Instead, I mounted the same tweeter in an extended 24 × 32″ baffle shown in **Photo 2**, and I took

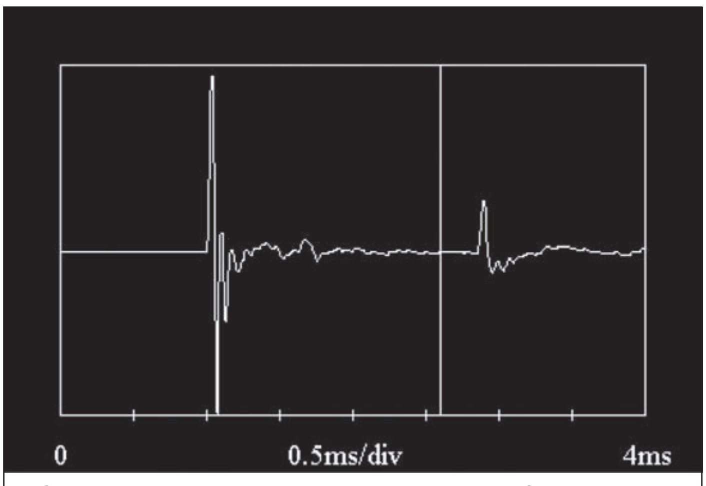

FIGURE 4: Measured impulse response of H26TG-35 Vifa tweeter mounted in speaker enclosure.

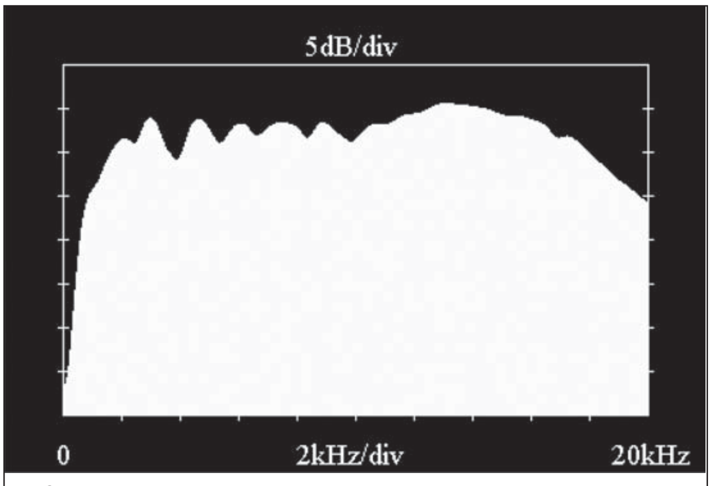

FIGURE 5: Portion of frequency response magnitude of Fig. 4 truncated at the vertical line.

measurements with the microphone in the same relative position with respect to the tweeter. **Figure 6** shows the impulse response of the Vifa tweeter mounted in this baffle.

The impulse response was truncated at the vertical line, which, you hope, removes any effects due to reflections from the perimeter of the baffle. **Figure 7** shows the measured impulse response out to 20kHz. It is evident that the response is much smoother and more reflection-free in the extended baffle.

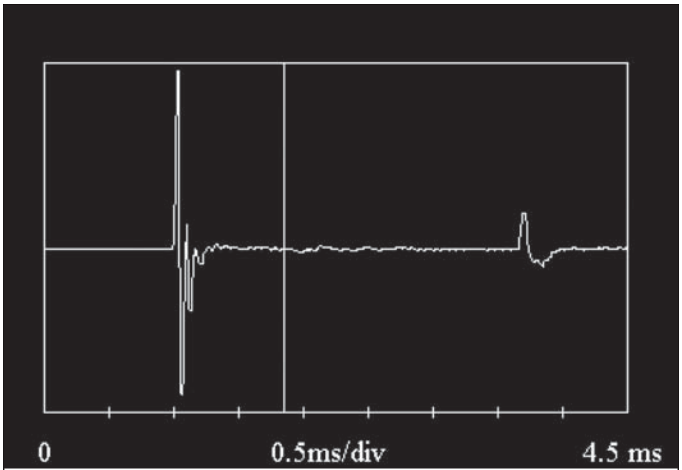

FIGURE 6: Portion of impulse response of Vifa tweeter in baffle of *Photo 2*.

My method was to assume that the primary tweeter response plus the reflections, analogous to P(ω)+ exp$^{(-j\omega\tau)}$ R(ω) for a single reflection in eq. 2, is given by the impulse response from the speaker enclosure in **Figs. 4** and **5**; and that the desired primary response without reflections is given by the impulse response in the baffle in **Figs. 6** and **7**. This is not strictly the case, because the impulse response in the baffle also includes interactions of the tweeter with the baffle; however, these interactions are over the uniform baffle surface and do not show up as localized reflections.

Perhaps a more accurate way to say it is this: I was trying to get the tweeter to behave as though it was mounted in a large uniform baffle, instead of in the speaker enclosure. The required H(ω) to do this is simply given by the ratio of the baffle frequency response indicated in **Fig. 7** to the frequency response in the speaker enclosure of **Fig. 5**. (Of course, in computing H(ω) the phase of the frequency responses must also be taken into account, not just the magnitudes; H(ω) is computed as the ratio of complex numbers.)

Figure 8 shows the result of carrying out this calculation for H(ω) and then transforming to the time domain to obtain h(t). At first this result may be confusing because according to the theoretical result in **Fig. 3**, you expect the waveform to start with a unit impulse. In fact, the initial portion of **Fig. 8**, the part to the left of the vertical line, may be considered as an impulse. This is because I used a 30kHz filter for anti-aliasing, and the initial part of the waveform in **Fig. 8** is just the impulse response of the 30kHz filter. (The vertical line in **Fig. 8** is not a truncation point. It is meant to draw attention to the part of the waveform corresponding to the 30kHz filter impulse response.)

In order to verify that h(t) of **Fig. 8** does indeed transform the tweeter response in the speaker enclosure to that of the baffle, I loaded the waveform of **Fig. 8** into the Virtual Crossover as a crossover filter for the tweeter, and then carried out a measurement with the tweeter in the speaker cabinet. **Figure 9** shows the initial portion of the impulse response that was obtained.

Truncating the impulse response at the vertical line and taking the FFT yields the resulting magnitude response of **Fig. 10**. Comparison with **Figs. 4** and **5** shows that essentially all of the reflections from the speaker cabinet have been removed.

CONCLUSION

Digital filters offer tremendous power, but with power comes responsibility. I know from first-hand experience that you can really screw up an audio system with digital filters if you are not careful. The method I have presented in this article is not perfect, because it corrects only for reflections on-axis, where the measurements were made; off-axis, reflections are still observed.

And there is another imperfection: Because the crossover filter is based on measurements, it contains some low-level noise that is a by-product of the measurement process. It would be preferable to have an analytical model for h(t) that would yield a noiseless correction filter. I am working on this project, but so far I haven't made much progress.

I have incorporated the correction filter presented in this article into the digital crossover for my speaker system; however, it is only one of many components such as low-pass and band-pass filters, analog correction networks, and time delays. The overall system sounds excellent, but I can't really say how much of an audible difference the correction filter itself makes. It certainly makes the measurements much nicer, though.

I believe that digital filters offer much potential for improving audio systems in ways that were never possible in the analog domain, provided they are applied properly. I would welcome hearing about readers' opinions or experiences with digital filters, either through the pages of *audioXpress* or my e-mail address, rkm@acoustic-instruments.com. aX

REFERENCES

1. Mains, Richard, "The Virtual Crossover, Part 1," *audioXpress* May 2001.
2. Mains, Richard, "The Virtual Crossover, Part 2," *audioXpress* June 2001.
3. Mains, Richard, "The Virtual Crossover, Part 3," *audioXpress* July 2001.

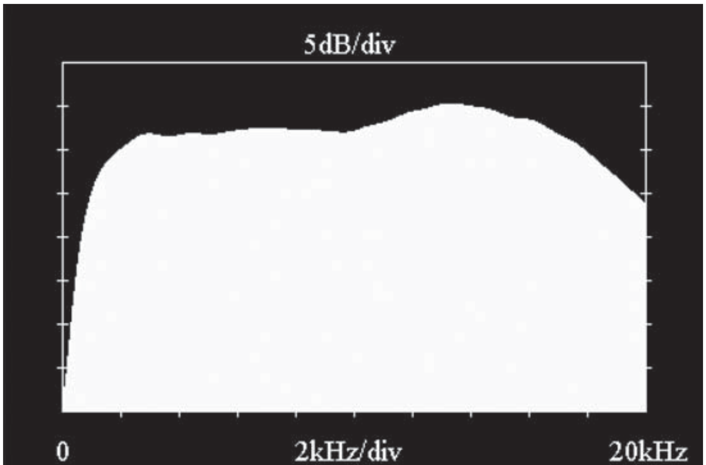

FIGURE 7: Portion of frequency response magnitude of *Fig. 6* truncated at the vertical line.

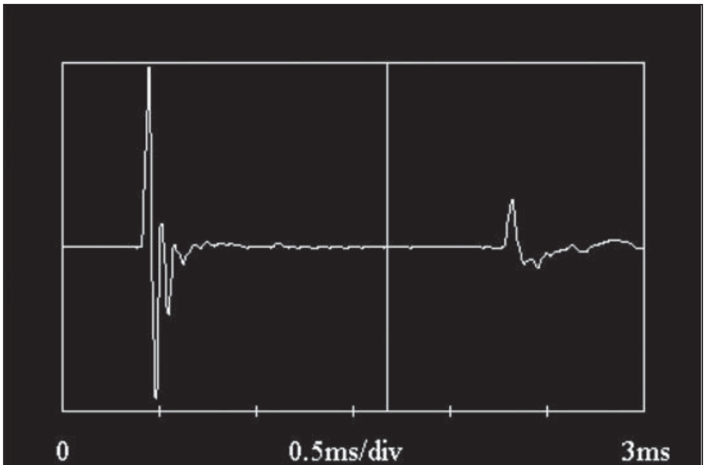

FIGURE 9: Initial part of impulse response with tweeter mounted in speaker cabinet of *Photo 1*, but using *Fig. 8* as Virtual Crossover filter.

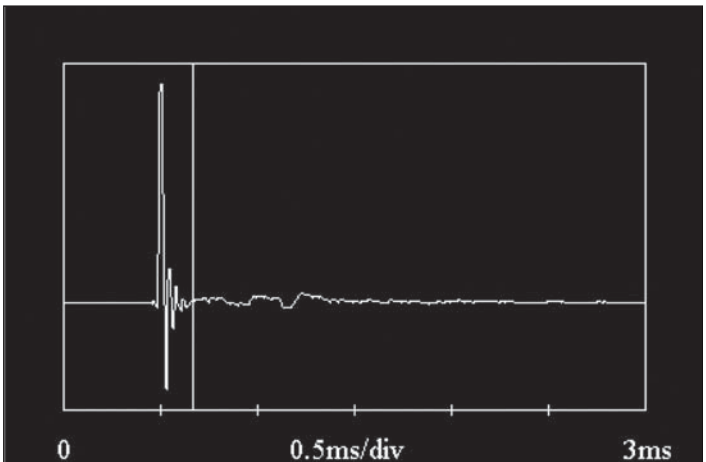

FIGURE 8: Transform filter that makes tweeter mounted in speaker cabinet appear to be in extended baffle.

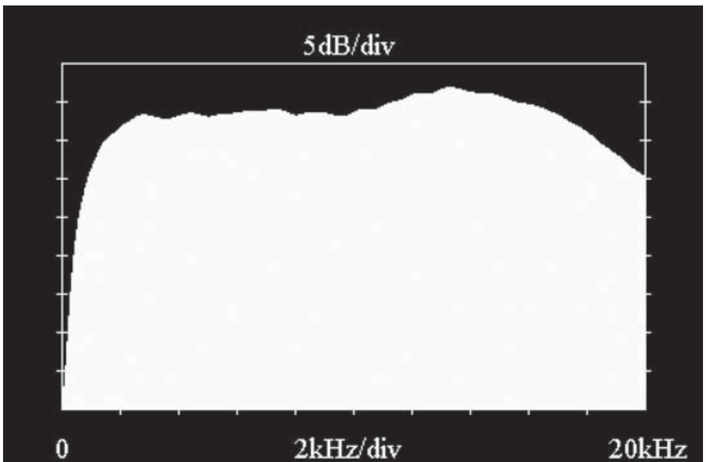

FIGURE 10: Frequency response magnitude to 20kHz from FFT of *Fig. 9* truncated at the vertical line.

Jelco JL-45 Cueing Mechanism

By Gary Galo

Jelco JL-45 Cueing Mechanism
www.jelco-ichikawa.co.jp
mail@jelco-ichikawa.co.jp
$45 US from Audio-Dragon on www.eBay.com

Finding a replacement cueing device for a vintage tonearm can be a problem. I'm still using the Grado Signature Tonearm that I purchased back in 1985. The Grado Signature (www.gradolabs.com) remains one of the finest high-end tonearms ever made, and original owners rarely part with them. I've seen only one for sale on eBay for as long as I've been active on that site.

The Grado Signature arm had only one serious weakness: the cueing mechanism. The design itself was awkward (**Photo 1**). The device had a protruding rod that you turned in order to raise and lower the arm, rather than

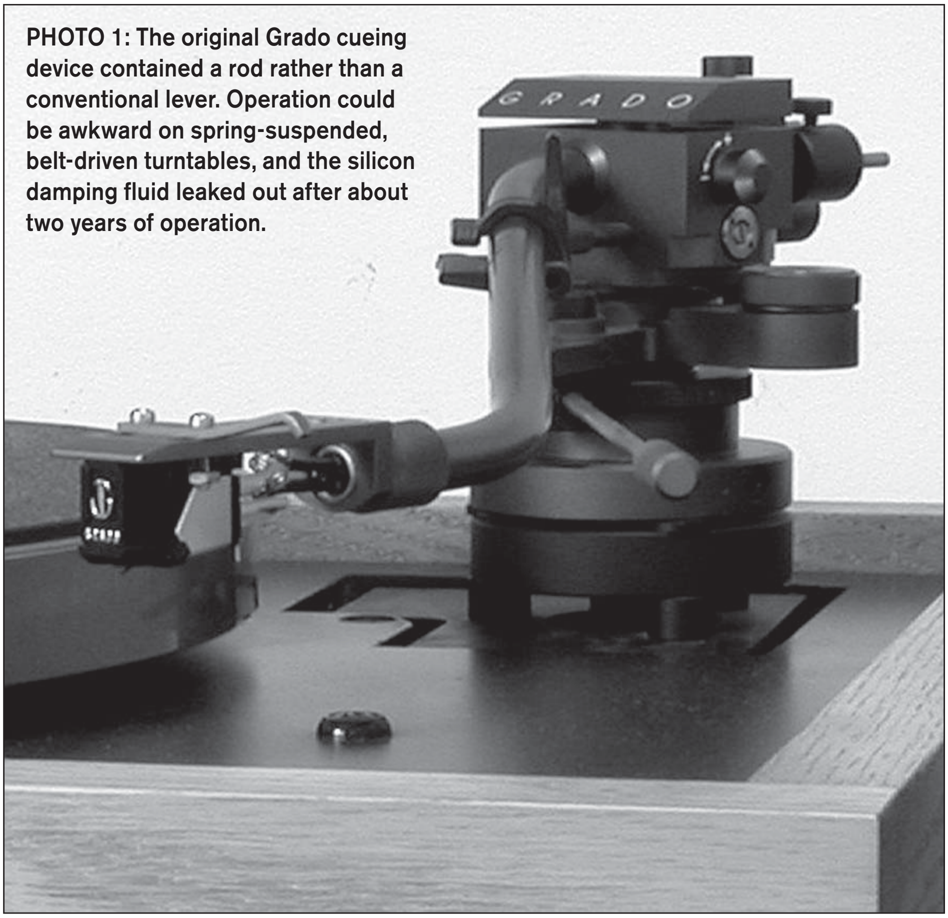

PHOTO 1: The original Grado cueing device contained a rod rather than a conventional lever. Operation could be awkward on spring-suspended, belt-driven turntables, and the silicon damping fluid leaked out after about two years of operation.

the conventional lift lever found on nearly all others. If the arm was mounted on a belt-driven turntable with a three-spring suspension, you had to be very careful when raising and lowering the arm in order to avoid disturbing the suspension.

The silicone damping fluid in mine had leaked out after about two years, so the cueing device had no vertical damping. For 22 years I lowered the arm by hand, and used the cueing device only for raising it. In the early 1990s I asked Joe Grado whether he had any replacement cueing mechanisms for sale as replacement parts, but by then his stock had been depleted.

The Grado Signature Tonearm was actually manufactured in Japan by Jelco, a firm that has made tonearms for many high-end manufacturers including Audioquest, Graham, Sumiko, and Koetsu. Last year, Jelco began listing the model JL-45 replacement cueing device on its website (**Photo 2**; www.jelco-ichikawa.co.jp/e_accessory.htm). Jelco also has a drawing of the JL-45 with precise dimensions, so it wasn't too difficult to determine whether it would fit my Grado arm (**Fig. 1**). Jelco's drawing gives 12.8mm as the diameter of the tonearm lift cylinder. I measured the Grado cylinder with my digital dial calipers and found it to be 12.76mm, so I was quite certain that the Jelco would fit the Grado arm.

Jelco normally sells only to dealers and manufacturers, and my inquiry about purchasing a JL-45 directly from them went unanswered. Fortunately, there's a dealer on eBay who carries the entire Jelco line, selling the JL-45 for $45. The dealer calls himself "Audio-Dragon" and is located in Fremont, Calif. Just go to www.eBay.com, select Electronics as the category, and search for Jelco. As of this writing, Audio-Dragon has received feedback from 494 buyers, 100% positive. My order was shipped within 24 hours and arrived within a week.

Installing the JL-45 in the Grado arm is quite simple. First, remove the lift platform from the Grado cueing device—it's held in place with a small screw. Next, loosen the Allen set screw that holds the cueing device's lift cylinder in place (Allen wrenches were supplied with the Grado arm). The cylinder will drop down, but it won't clear the tonearm base. Loosen the two Tonearm Locking Screws (#14 in the Grado manual) and the VTA Lock Screw (#9). This will allow you to raise the arm enough to drop the tonearm lift cylinder out of its mount.

Next, remove the lift platform from the Jelco cueing device. Be very careful not to lose the mounting screw! Raise the tonearm enough to slide the Jelco cylinder into its mounting hole. Tighten the two Tonearm Locking Screws (#14) and the VTA Lock Screw (#9) on the Grado arm. Position the Jelco JL-45 so the lift lever

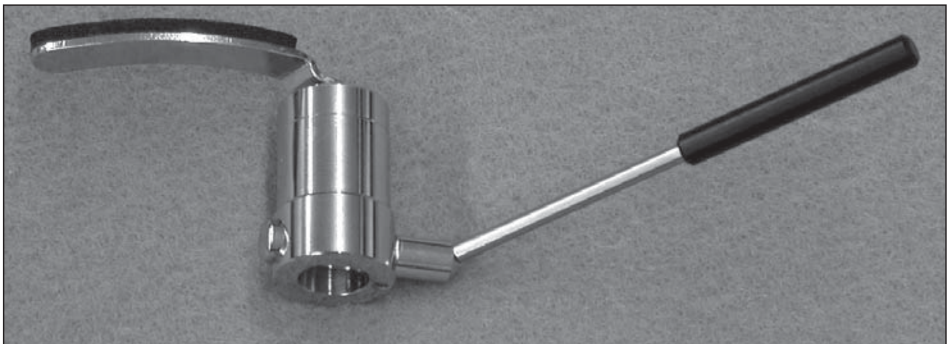

PHOTO 2: The Jelco JL-45 Cueing Mechanism. With a cylinder diameter of 12.8mm, the JL-45 is an exact fit for the Grado Signature Tonearm.

is about ¼″ from the Grado anti-skate platform when the lever is raised. There's a ring around the Jelco cylinder about ⅛″ from the top. This ring should be barely visible when the cylinder is in the correct vertical position.

Once you've got the cylinder positioned correctly, tighten the Allen set screw. Replace the Jelco lift platform—be very careful not to drop the mounting screw. After the arm is lowered onto the record, there should be about 1/8″ of clearance between the tonearm's Arm Lift Support (#3) and the cueing lift platform. The Arm Lift Support should follow the curvature of the lift platform across the entire surface of the record. **Photo 3** shows the Jelco JL-45 installed in the Grado arm.

I anticipated one interference problem when I ordered the Jelco JL-45: When the Grado Arm Rest Lock is flipped open, it obstructs movement of the JL-45 lift lever. I made a small metal clip from a piece of a small paper clip, bent into a "J" shape (one side is longer than the other), and slipped it over the vertical arm rest support on the Grado arm (**Photo 3**). Close the "J" on the clip so it fits reasonably tightly over the arm rest support. The long side keeps the Arm Rest Lock from falling too far back, and getting in the way of the cueing lever. The short side won't obstruct the

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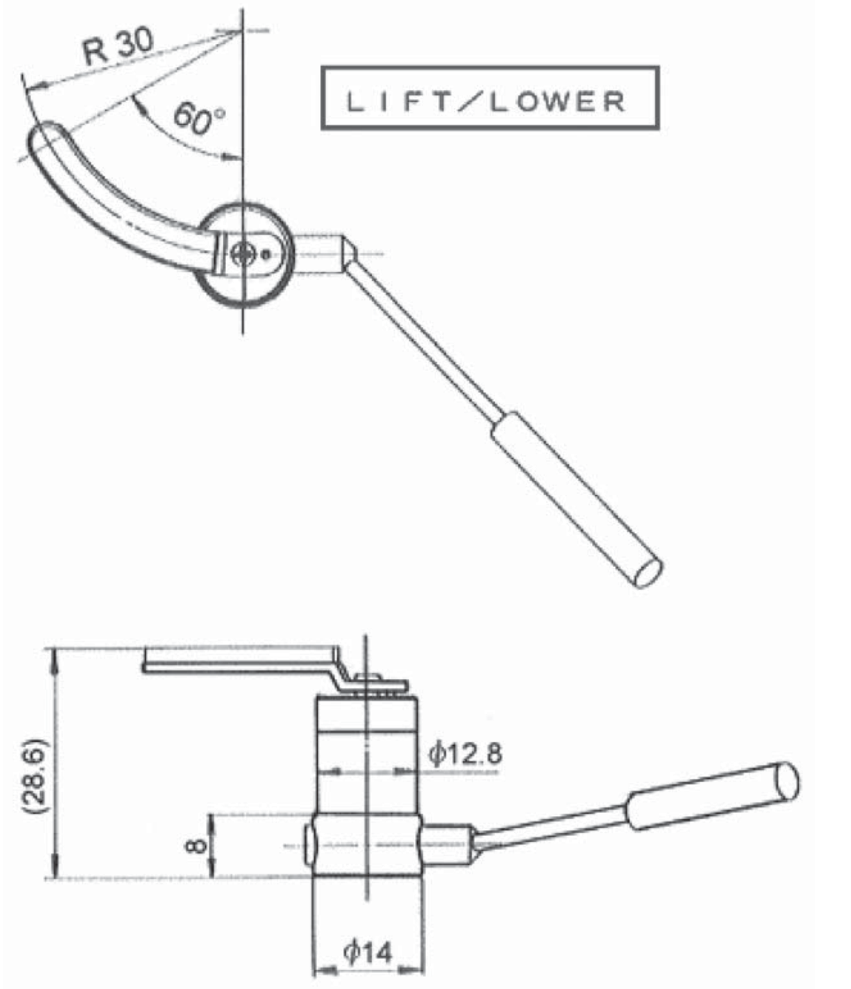

FIGURE 1: Mechanical drawing of the JL-45 posted by Jelco on their website (courtesy of Jelco).

[In Xpress Mail last month, we inadvertently omitted two figures that should have accompanied Thomas Bohley's letter. We reprint the letter in its entirety with the two missing figures.—Eds.]

GOOD DESIGN

In Bard Kallestad's otherwise excellent article "Designing for Everyone" (July '10, p. 14), there is an error in calculating the output voltage under a 5mA load, starting with his Fig. 1. For easy analysis, it is best to replace the two 47K resistors with their Thevenin equivalent as shown in my **Fig. 1**.

Now, when 5mA is drawn from the emitter of the 2N3904, Mr. Kallestad correctly calculated the base current at 50μA, but this current is drawn through 23.5K, not 47K. **Figure 2** shows the correct output voltage under a 5mA load.

In this analysis I have assumed 0.7V for Vbe. (0.6 V or 0.7V is used by different designers; at 5mA I think 0.7V is more accurate.) The actual output voltage is closer to 2.6V than Mr. Kallestad's 1.55V. This is still very poor regulation, just not quite as poor.

I thought the balance of the article was excellent and clearly stated many valid and valuable goals for robust and repeatable designs.

Thomas Bohley
Colorado Springs, Colo.

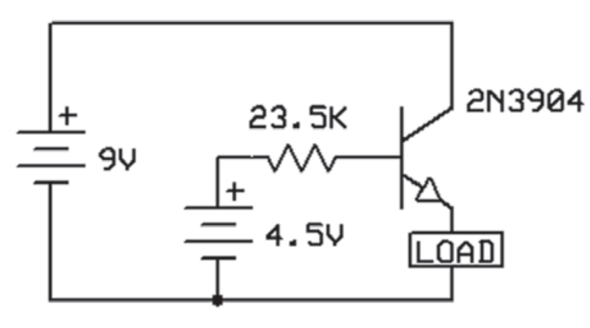

FIGURE 1: Replacing the 47k resistors with Thevenin equivalents.

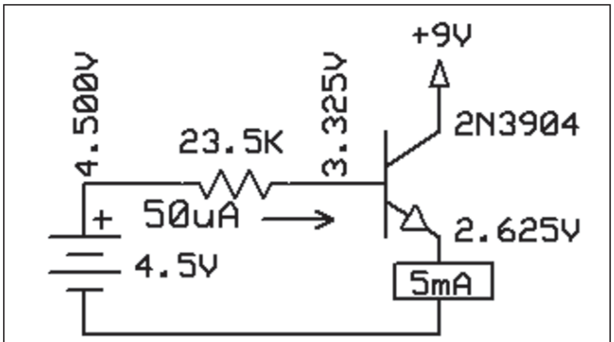

FIGURE 2: Correct output voltage under a 5mA load.

BOX MEASUREMENTS

I am a long-time speaker builder who has been putting them together since 1963, when I started with a 12″ full-range Calrad speaker in a 4ft^3 box made out of half-inch plywood. I was pleased to see Anastasios Fiakas' article ("Laminated MDF Three-Way Speaker System," Feb. '10, p. 6) featuring that really gorgeous and elaborately built multilayered MDF box. I was amazed at the attention he paid to reducing vibration and standing waves in the cabinet. I may need to borrow some of his design ideas for my next "big" living room speaker.

However, I was left a little unsatisfied that there were no measurements taken on the finished unit. The subjective assessments, though somewhat helpful, are not as good an index of performance as measurements. I was curious as to the ultimate low-frequency capability of the KEF B139 in this sort of box, because I have only heard this speaker in a transmission line system, where its transient response was really terrific, but ultimate lows were rolled off.

So, I took out my trusty TI-30 hand calculator and decided to calculate the F3 for the system as described, just to get an idea of the low-frequency cutoff of the KEF B139 in that 70 ltr box. Mr. Fiakas says the low-frequency box Q is .82. This would give an F3 of about 76Hz [box Q (.82) divided by the Qts of the KEF (.269) times the Fs]. That's not a very low F3, particularly when you consider the low Fs of the KEF.

Also, a box Q of .8 will give a very slightly elevated response characteristic around, and a little above, F3. This means you will have a slightly "fat" response in the lower end of the instrumental fullness range. It will also compromise transients some. This is not the result Mr. Fiakas describes. It would certainly be a sexy, salable full-sounding response curve, but transients and real lows would be compromised. Also, with that box Q and that speaker, the theoretical box comes in at about 21 ltr, not the 70 ltr specified.

In working the various theoretical options, I determined that if the low-frequency box Q is set at .42, where Mr. Fiakas says the mid-frequency box is set, then you get a gradually rolled off curve and very tight response, with an F3 of about 40Hz, which is the fundamental of the low E on a Fender bass. While not infrasonic, it certainly is respectable, and because the response curve will be rolled off some, you can really slam it with bass boost, or just turn up the woofer amp, and get a solid-sounding low end. Also, when you work out the theoretical volume for box Q of .42 with the KEF B139, it comes in at 67 ltr.

It looks to me as though Mr. Fiakas simply made a mistake and switched the box Qs for the low-frequency box and the mid-frequency box. That would get to the sonic result he describes, and the numbers make more sense.

I always thought that the KEF B139 was better suited to a bass reflex than a closed box because of its very low Qts and relatively low Fs, but that is just my opinion and Mr. Fiakas certainly appears to have gotten an excellent result. The bass reflex alternative gives an f3 of 39Hz, a box frequency of 34Hz, and a theoretical volume of 56 ltr. This is using the Keele formula, as listed in Ray Alden's *Advanced Speaker Systems* (Master Publishing Co., 1995, www.amazon.com/Advanced-Speaker-Systems). The difference would be a reduction in cone excursion in the bottom octave and an elevation in the response curve, reducing the need for a bass boost. But, negatively, transients might not be as good.

Thank you for publishing this well-thought-out and well-crafted design, and Mr. Fiakas should certainly not be bothered with my minor quibbles.

Joseph A. Zannieri
Norwalk, Ohio

Anastasios Fiakas responds:

I thank Mr. Zannieri for his interest in my article. I always welcome comments and the opportunity to present some measurements. KEF B139 drivers came in various models throughout the years of production. I have the model B139B SP1044, and, after the break-in procedure, I measured the following parameters: Vas = 160 ltr, Qts = .44, Fs = 30Hz.

For a target Qtc = .8 during the initial calculations: a = $(Qtc/Qts)^2 - 1 = 2.3$, so the box volume should be: Vb = Vas/a = 69.4 ltr. Theoretically, Fc = Qtc/Qts times Fs = 55.9Hz and F3 = Fc times .9 = 50Hz.

In **Fig. 1** you can see the measured impedance and phase curves of the speaker in the low-frequency box. Maximum impedance (Zmax) = 61.2Ω. In order to find Qtc you must find the frequencies f1 and f2 where the impedance Zmax is equal to 43.3Ω (Zmax times .707). So f1 = 51Hz, which is also the F3,

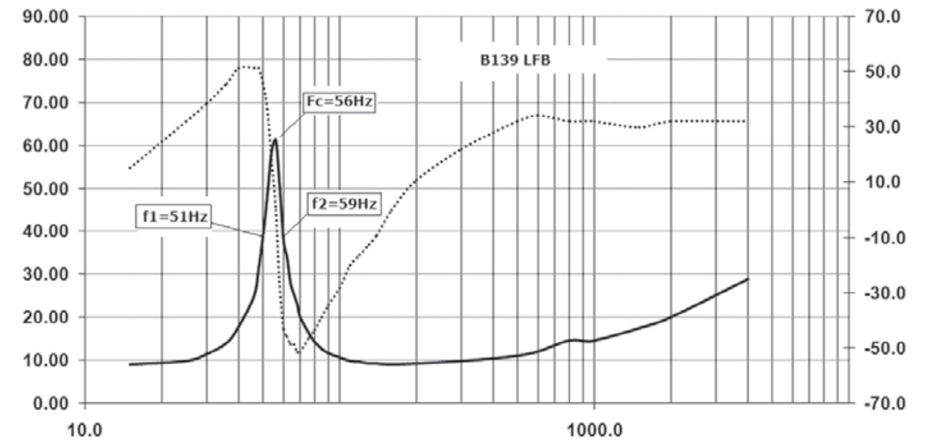

FIGURE 1: LFB impedance and phase curves.

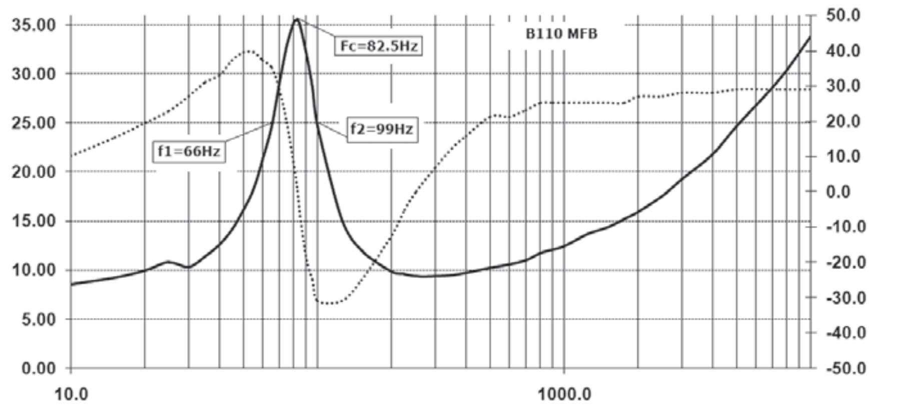

FIGURE 2: MFB impedance and phase curves.

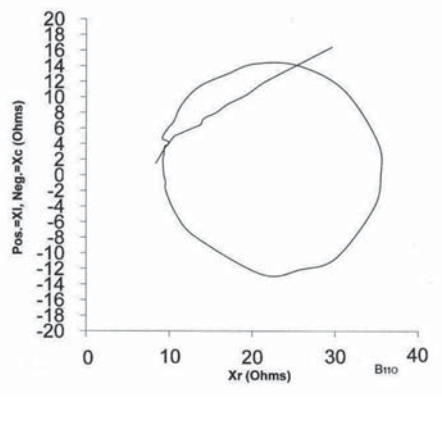

FIGURE 3: LFB Nyquist plot.

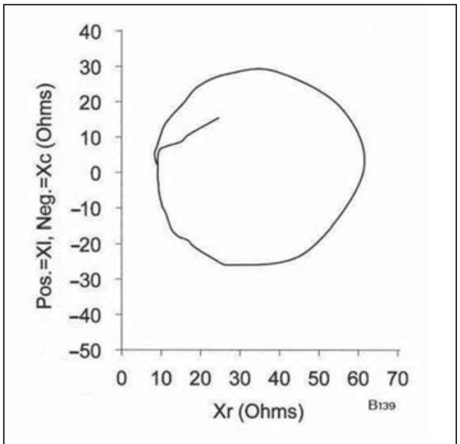

FIGURE 4: MFB Nyquist plot.

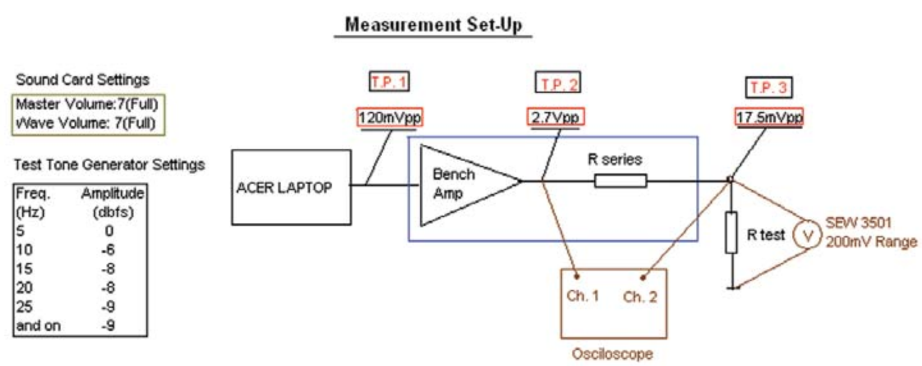

FIGURE 5: Measurement setup.

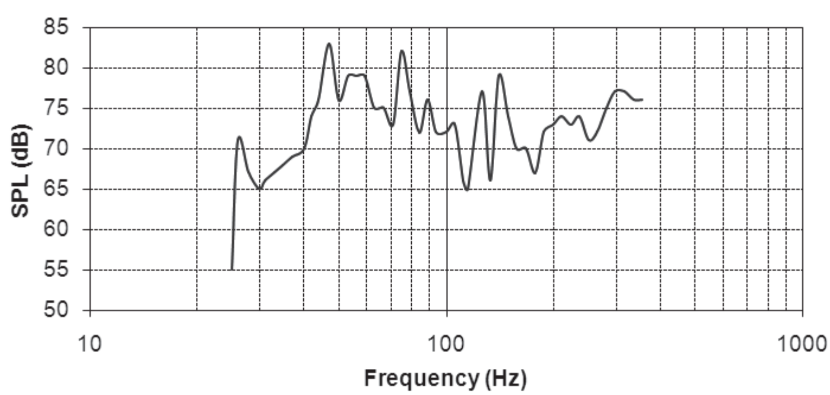

FIGURE 6: In-room response.

and f2 = 59Hz. The DC resistance of the B139 is Re = 7.2Ω. Then Qtc = (Fc/f2 - f1)(Re/Zmax) = .824.

Doing the same for the mid-frequency box (MFB) in **Fig. 2**, you have Zmax = 35.52Ω, f1 = 66Hz, f2 = 99Hz, Re = 6.6Ω, and Qtc = .465. In **Fig. 3** you can see the Nyquist plot of the low-frequency box, in **Fig. 4** the Nyquist plot of the MFB, and in **Fig. 5** the measurement setup.

From the beginning of the project, knowing that the woofer's F3 was 50Hz, I was also skeptical. Certainly it is not very loud at 40 or 30Hz, although it sounds adequate.

In **Fig. 6** you can see the low-frequency response of both loudspeakers operating simultaneously, measured from the listening position with a Radio Shack digital SPL meter. Instrument readings are corrected according to the chart at: http://www.subwoofer-builder.com/SPL-corrections.htm.

The listening position and the loudspeakers' location are near optimum, regarding the control of modal resonances of the room, which has the following dimensions: L = 6.9m, W = 3.63m, and H = 2.94m. The axial mode fundamentals and harmonics are (Hz): 25, 47, 49, 58, 74, 94, 99, 116, 140, 173, 187, and 231.

Regarding the paragraph about the listening tests, I tried to make a brief summary of opinions of the ordinary and experienced listeners who evaluated the speaker system. *aX*

CONTRIBUTORS

Raj Gorkhali ("The Brick," p. 6) resides in Kathmandu, Nepal, and is an audio hobbyist who enjoys working in his private workshop. He has graduated from Siddhartha Vanasthali College.

Aren van Waarde ("The Modern Homodyne," p. 16) is a biochemist working in the field of medical imaging (positron emission tomography.) He has worked as a Ph.D. student and a postdoc at several universities (including Leiden and Yale Universities) before accepting tenure at the University of Groningen. His passion for audio started on his ninth birthday when his parents gave him a Philips kit. Most of his current audio equipment (loudspeakers, radios, tuners, pre- and power amps, both tube and solid-state) is homemade.

Richard K. Mains ("Eliminating Speaker Reflections with Digital Filters," p. 24) obtained a Bachelor of Science degree in electrical engineering and a Master of Science degree from the Ohio State University in 1972 and 1974, and a Ph.D. in electrical engineering from the University of Michigan in 1979. He worked for several years in the area of high-frequency semiconductor device modeling at the University of Michigan. Since the early 1990s, he has worked in the area of power electronics circuit design and microprocessor programming at McCleer Power, Inc., located in Jackson, Mich. His interests have included working on several speaker systems, developing measurement systems for loudspeakers, and designing power amplifiers and other electronics for music reproduction.

Gary Galo (Review: Jelco JL-45 Cueing Mechanism, p. 29) is Audio Engineer at The Crane School of Music, SUNY Potsdam, where he also teaches courses in music literature. A contributor to AAC since 1982, he has authored over 230 articles and reviews on audio technology, music, and recordings. He has been the Sound Recording Reviews Editor of the *ARSC Journal* (Association for Recorded Sound Collections) since 1995, was co-chair of the ARSC Technical Committee from 1996 to 2004, and has given numerous presentations at ARSC conferences (www.arsc-audio.org). Mr. Galo is also a frequent book reviewer for *Notes: Quarterly Journal of the Music Library Association*, has written for the *Newsletter of the Wilhelm Furtwängler Society of America*, and is the author of the "Loudspeaker" entry in *The Encyclopedia of Recorded Sound in the United States*, 1st edition.

Bill Fitzmaurice ("Designing Enclosures with SketchUp," p. 36) has been a professional musician since 1966 and has been constructing instruments, amplifiers, and speakers for just as long. Vice president of DeltaSounds Loudspeakers Inc., Fitzmaurice is the author of *Speaker Builder's Loudspeakers for Musicians* and over 30 magazine articles dealing with speakers and electric instruments, as well as the action-adventure novel *Operation: Sergeant York*. Bill and his wife of over 30 years reside in Laconia, NH.

Designing Enclosures with **SketchUp**

You can take the guesswork and tedium out of building speaker boxes with this handy design program.

I've been using Google SketchUp (http://sketchup.google.com/download.html) to design my enclosures for about five years, and it's totally changed the process. What used to take a lot of paper—and a lot more plywood—now is done on my computer, with far more accurate results. Learning how to use SketchUp to its full potential can take literally years, but even if all you know is the basics it speeds the design process considerably. I'm going to show you how to use it to design a basic box, using the fundamental tricks you need to know to get started.

GETTING STARTED

When you open the program, choose the style you want for the background. The default Simple Style has green ground and blue sky, which is fine if you're designing a house, but not so much for a speaker. I use Shaded with Textures. Click on the Window tool, Style, and Shaded with Textures. The next option to define is the measurement units, found in Window/Model Info/Units. I prefer to use inches only, as opposed to feet and inches, so I choose the Fractional option, with 1/16″ resolution.

The Toolbar has graphic representations of each function, such as draw line (a pencil), erase (an eraser), draw a rectangle (a box), and so forth. To use a function, click on its icon, move the pointer to the screen, and click again to initiate. To build my example cabinet I'll start with the bottom. You can use the rectangle tool, but I find the line tool easier to work with. The two horizontal and one vertical axes initially show up as red, green, and blue lines.

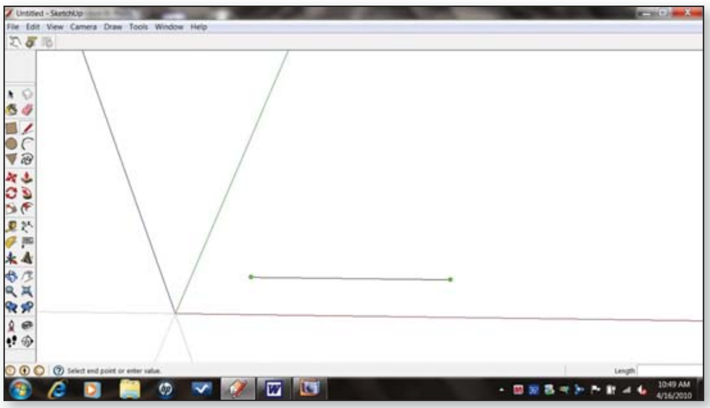

FIGURE 1: A line.

I start with a 24″ line on the red axis (**Fig. 1**). Click on Line, click where you want the line to start,

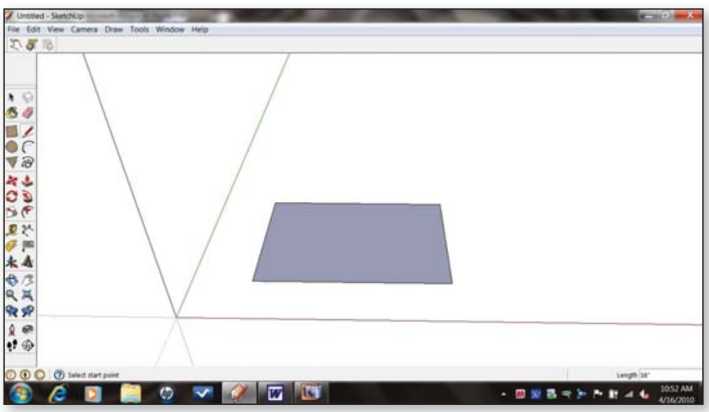

FIGURE 2: A rectangle.

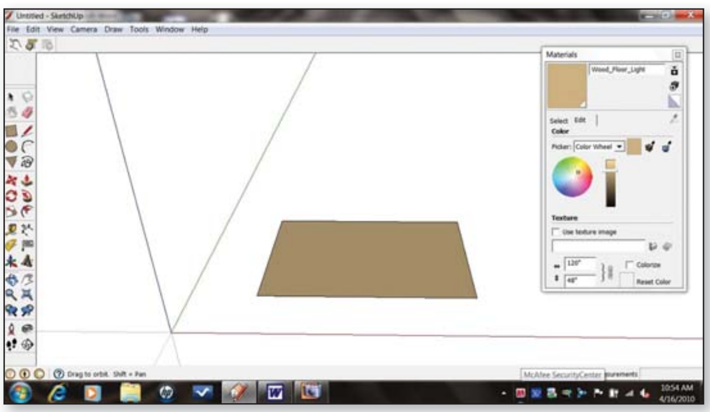

FIGURE 3: A painted rectangle.

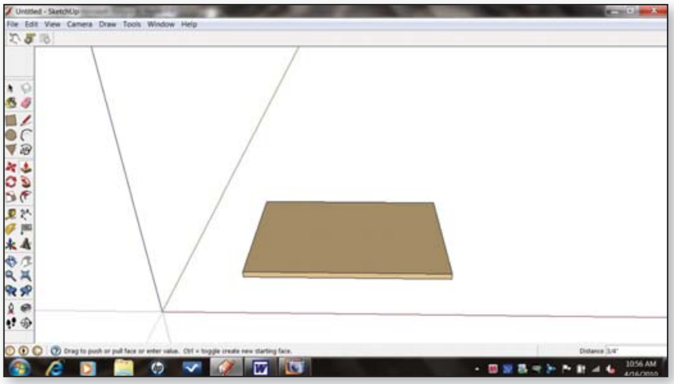

FIGURE 4: A bottom panel.

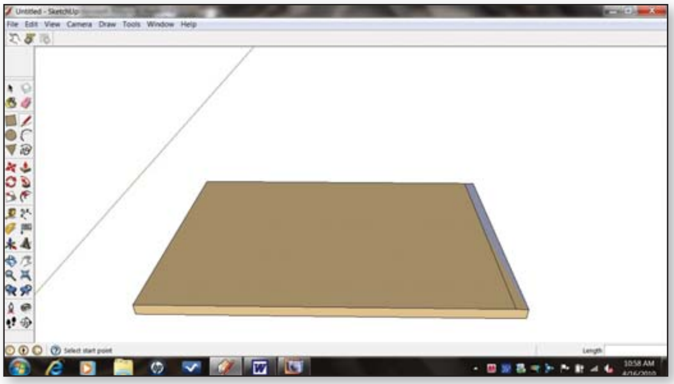

FIGURE 5: Adding a side, part one.

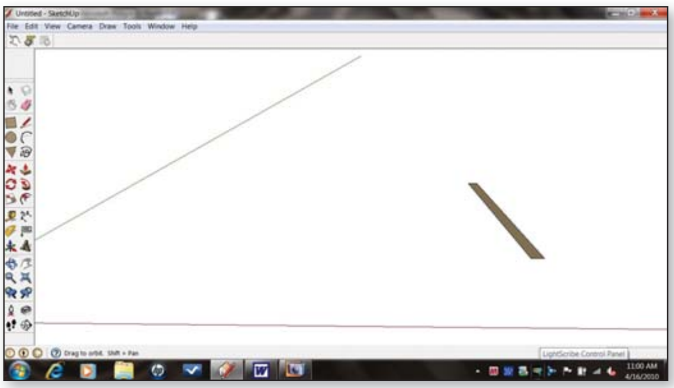

FIGURE 6: Adding a side, part two.

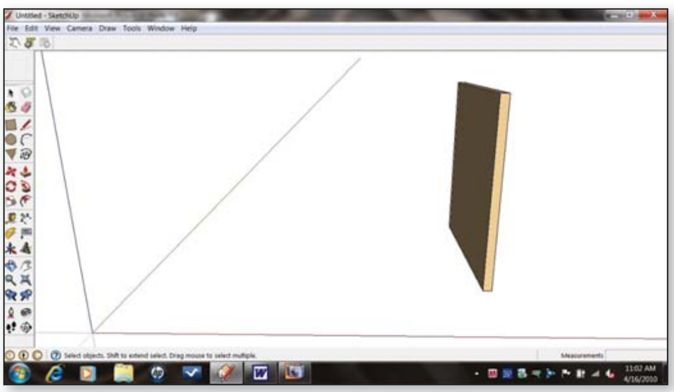

FIGURE 7: Adding a side, part three.

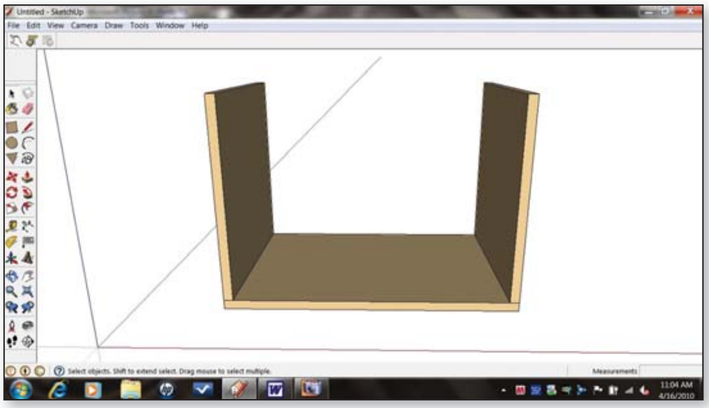

FIGURE 8: Adding the second side.

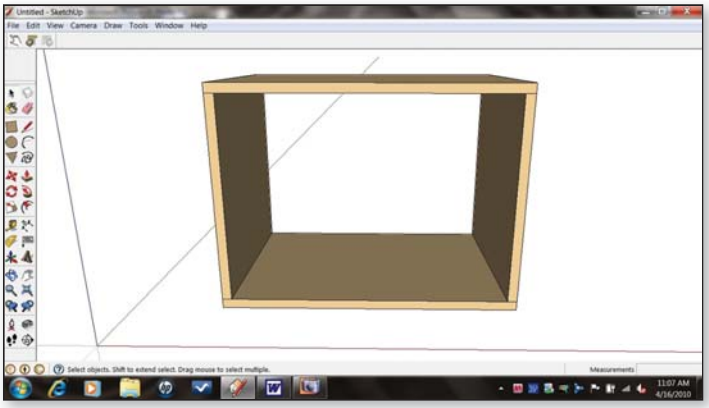

FIGURE 9: Adding the top.

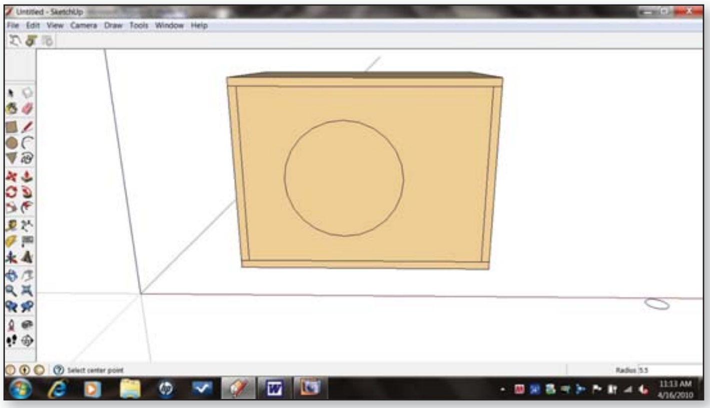

FIGURE 10: Adding the baffle, marked for the driver.

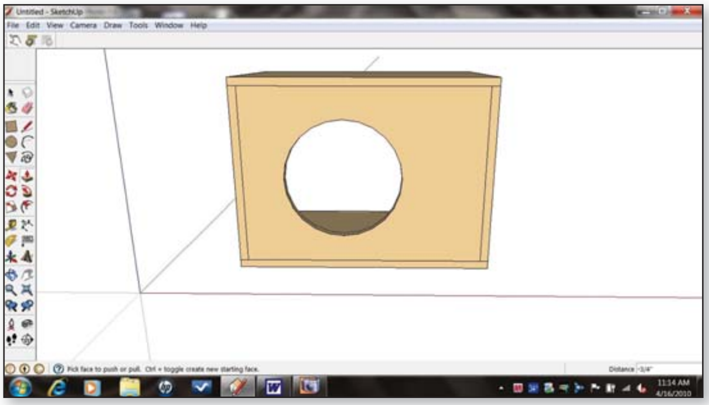

FIGURE 11: The driver hole "cut."

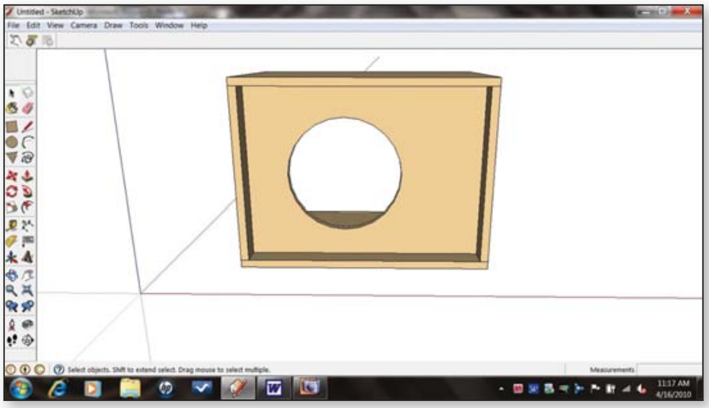

FIGURE 12: The baffle recessed.

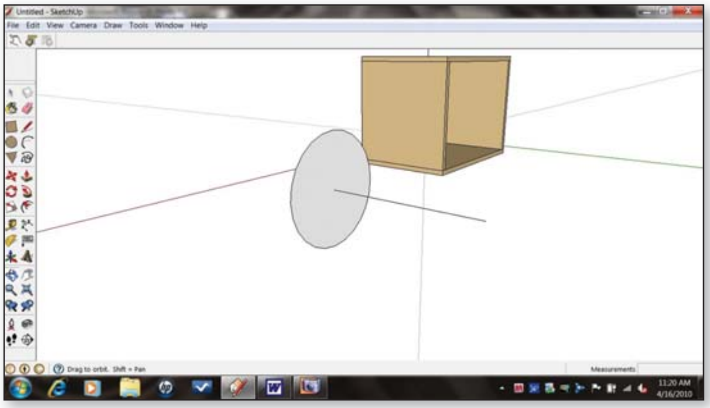

FIGURE 13: A driver, step one.

and drag along the desired axis. The length drawn will display on the lower right corner of the screen. You can drag the line to the desired length, or, to start the line, type in the desired length and hit "enter." Next I draw the 16″ dimension on the green axis, another 24″ line on the red axis in the opposite direction of the first, and the last 16″ line on the green axis. After you finish the last line, the program automatically fills in the rectangle to produce a solid surface (**Fig. 2**).

Immediately color the part. Click on the Paint Bucket icon to display the material choice box (**Fig. 3**), from which you can choose hundreds of materials and colors. Many of the materials have texture, which looks nice, but also eats up memory and processor speed. So if your computer is anemic, use a color that has no texture. To paint a part, click on the paint color and then the part. You'll want to paint both sides of the part, so click on the Orbit icon to view the opposite side and paint it.

TAKING SHAPE

To change the flat 16 × 24 rectangle into a ¾″ thick cabinet bottom (**Fig. 4**), use the Push/Pull function. Left-click on its icon, then left-click on the surface you want to pull and drag it to the desired thickness. You will see the thickness in the lower right corner; it's easiest to start to pull the surface, type in the desired dimension, and hit "enter."

Once you have the part fully sized and painted, you want to make it into a Component. To do so, left-

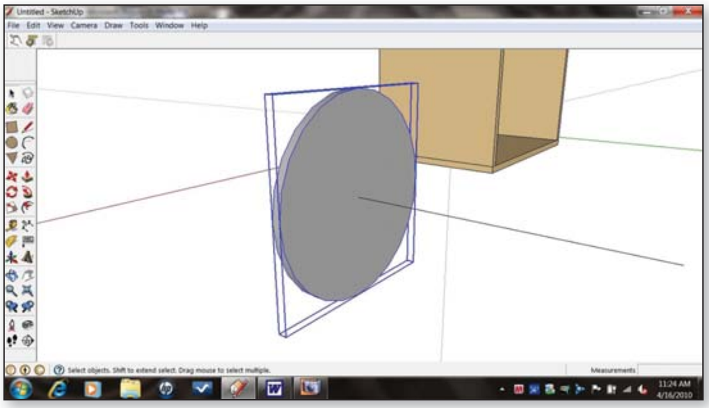

FIGURE 14: The driver flange.

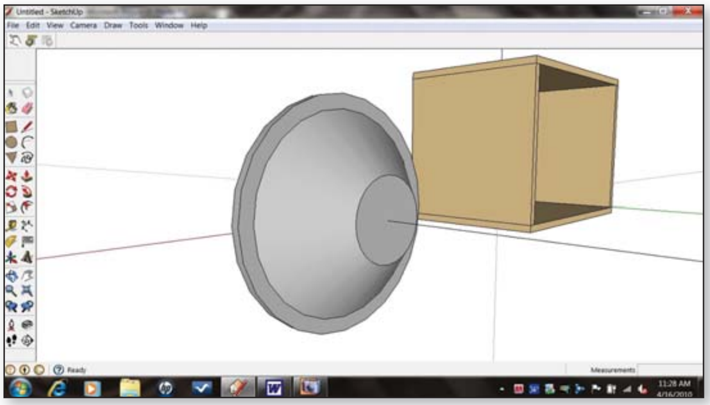

FIGURE 15: The driver frame.

click on the screen, hold the button down and drag the pointer, producing a box around the part. Release the button, right-click to bring the menu on-screen, left-click "Make Component." Now you can add more parts without affecting those already componentized.

To add a side, draw a ¾ × 16″ rectangle on one edge of the bottom (**Fig. 5**). Left-click on the bottom to highlight it, right-click to bring the menu on screen, left-click "Hide," and the bottom will disappear from view (**Fig. 6**). Now paint and pull the rectangle to form the completed side (**Fig. 7**). Make it a Component. Left-click the Edit toolbar; left-click "Unhide Last" on this menu and the bottom will reappear.

To add a second side, copy the first. Left-click on the part to highlight it, then left-click on the Edit toolbar, Copy, and Paste in Place. This produces a second side occupying the same space as the first. Left-click on the "Move/Copy" icon, left-click on the part, and drag the duplicate part to its new location (**Fig. 8**). When you drag on a horizontal or vertical axis, that axis color will appear on the screen. Instead of dragging, you can always type in the distance, and hit "enter." Use the same technique to add the top (**Fig. 9**).

Add the front and back using the same technique as with the sides. To cut a hole in the front for your driver, double-left-click on the front, which puts that component into Edit Component mode. Left-click on the Circle icon, left-click on the face of

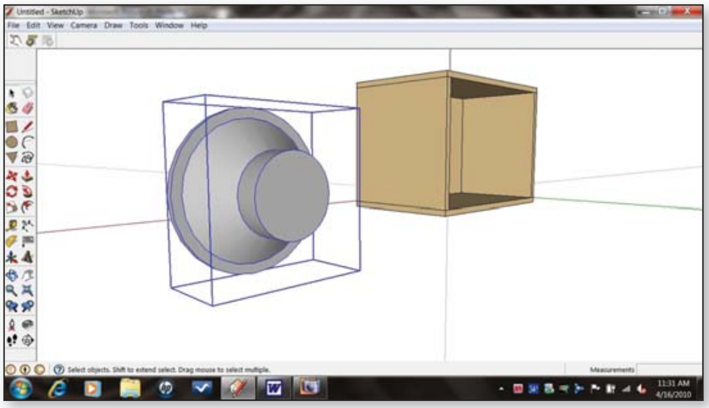

FIGURE 16: The completed driver.

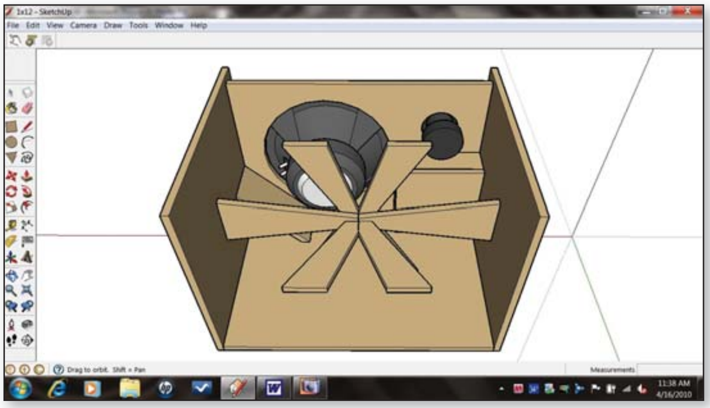

FIGURE 18: Three-way speaker cut-away view.

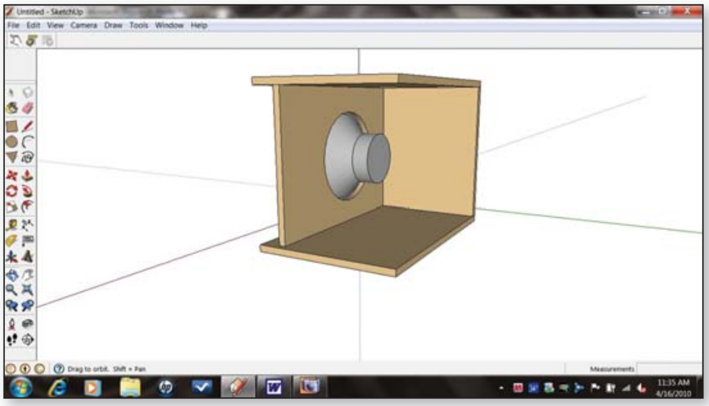

FIGURE 17: Mounting the driver.

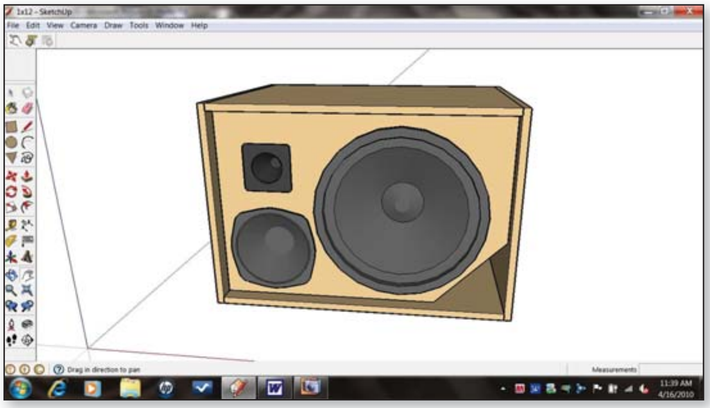

FIGURE 19: Completed three-way speaker.

the baffle, and drag the circle to the desired size (**Fig. 10**). Left-click on the Push/Pull icon, and left-click on the interior of the circle, which will highlight it.

"Push" on the circle and it will retract into the baffle. Push it until the "On Face" message appears next to the pointer, release the button, and you have a hole in the baffle (**Fig. 11**). Left-click anywhere on the screen outside of the highlighted area of the component to leave the Edit Component mode. If you want to recess the baffle from the front, left-click on it, left-click on "Move/Copy," and left-click on the baffle again and move it back (**Fig. 12**).

FINISHING TOUCHES

To make a driver, start by producing a circle, with an axial line extending from its center (**Fig. 13**). Color it, pull it, and component it, producing the driver mounting flange (**Fig. 14**). Draw a circle on the face the diameter of the frame, and color and pull it. To make it cone-shaped left-click on the rear edge, highlighting that circle. In the Tools menu left-click on "Scale." A box will appear around the highlighted circle; dragging on its pointers will shrink the circle, producing a cone shape (**Fig. 15**).

When done, make the frame a component, then add the magnet to the back of it. After the components are all done, left-click on the screen and drag the working box to highlight all of the driver components. Right-click to bring up the menu and left-click on "Make Group" (**Fig. 16**). Now you can move the completed driver about, to mount it into the cabinet. That's made easier by hiding a side so you can see where the driver's going (**Fig. 17**).

Where SketchUp proves invaluable isn't in producing external parts, it's with internals, such as braces. Figuring out their exact measurements is a snap, as is figuring out whether the drivers you're using will actually fit into the box. When you're designing cabs like the one shown in **Figs. 18** and **19**, the time saved makes learning how to use the program well worth the effort. *aX*

Continued from p.31

Arm Rest Lock when it's used to lock the tonearm in the Arm Rest.

The JL-45 works beautifully in the Grado Signature Tonearm—after 24 years, I finally have a "normal" cueing device on my favorite tonearm. The gentle vertical damping is just right and easy on the stylus. Operation of the cueing lever is smooth and never disturbs the turntable suspension.

Since Jelco manufactured so many high-end tonearms, the JL-45 should fit many other arms in need of a replacement cueing device. Jelco also manufactures the model HS-20 Headshell, which features the dual locking pins used in the Grado Headshell. It is also sold by Audio-Dragon and should make an excellent replacement or upgrade for standard Japanese headshells. ***rr***

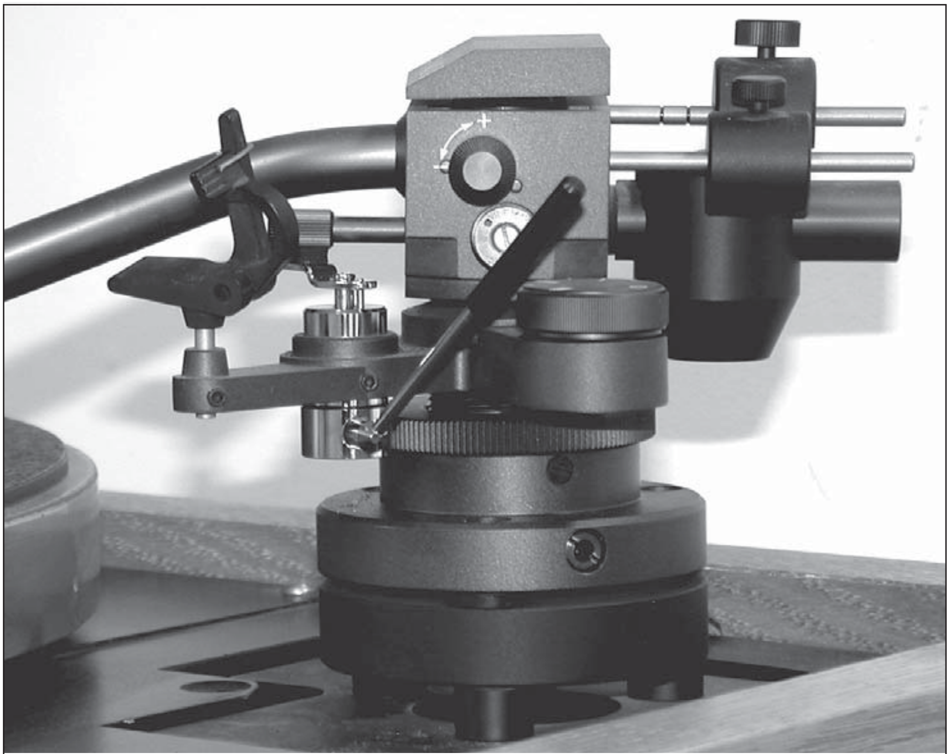

PHOTO 3: The Jelco JL-45 installed in the Grado Signature Tonearm. The small clip fitted to keep the Arm Rest Lock from interfering with the cueing lever can be seen in the upper left.

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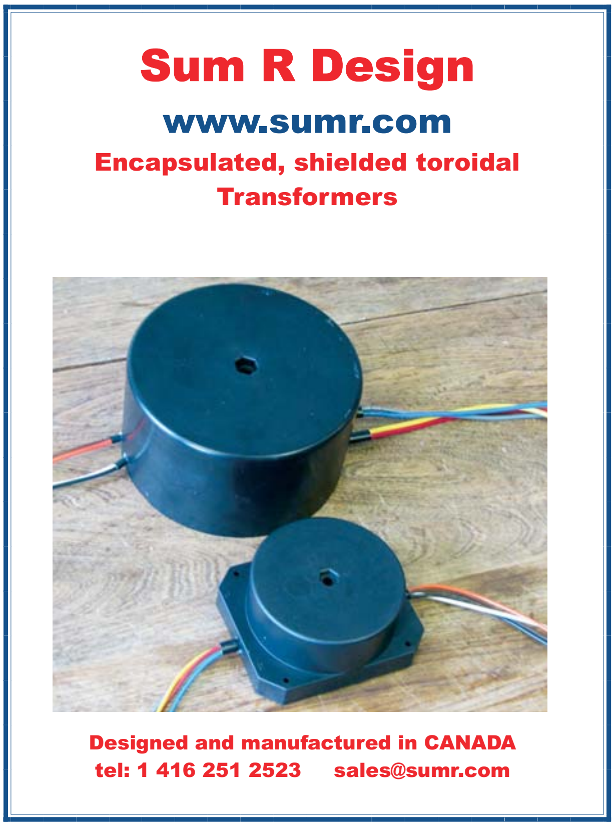

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By J.R. Laughlin

The LM12 Amplifier

This father-son project takes on a life of its own: a half-dozen amps to drive this speaker.

One day, one of my boys, taking a woodworking class in middle school, asked me whether he could build a speaker cabinet in that class. "Well," I said, "that sounds like a good idea." Based on the modest amount of money he asked for this project, I really had no idea what he would come up with.

He called me from school one day and asked whether I would come to help him carry his speaker cabinet home. I said, "Will they not let you carry it home on the bus?" He replied, "No, pa." So as I drove to the school and approached the building, I saw this large plywood box sitting outside. I thought, "I hope they are discarding that crate, which perhaps they will let me have for the plywood."

Good grief! It turned out to be my boy's speaker cabinet, approximately 4′ × 5′ × 3′. I asked him why he made one so big? "Well, pa, it was just an idea," he replied.

Later, I found out it had something to do with his discussion with the woodworking teacher about me and my love for unusual things. Later I found this to be the cabinet design for an Electro-Voice "Patrician 800" using a 30″ woofer. Contacting them, I was told that the 30″ model is no longer available. There were no holes cut in the cabinet yet, so I thought I might substitute some 12″ drivers for the 30″ type.

A bit of calculation determined that the area of this 30″ driver is approximately 706 in^2. Now, 6-12″ drivers have approximately 678 in^2. I say approximately because I was not sure of the exact effective cone area. So, I cut six holes to accommodate a standard 12″ driver into the front of the cabinet with a bottom vent cutout. I decided that a large mid-range horn—and, of course, a tweeter—would be appropriate to add to the structure. The immediate question was how to match to the woofers. I decided to parallel two each and drive each of the three pairs with a separate LM12 amp.

The LM12 is one heck of an operational amplifier:

1. 80W of sine wave power into 4Ω with 0.01% distortion
2. Power bandwidth of 60kHz
3. Input protection, controlled turn-on, thermal limiting, overvoltage shutdown, output current limiting
4. Slew rate 9V/µs

   Its datasheet lists other nice characteristics.

I also decided to drive the midrange, tweeter, and second speaker with their own separate LM12 amp. I enjoyed the idea of an individual LM12 amp for each speaker (why work one LM12 so hard?). I got into the habit of referring to the big speaker with all the woofers as the "main speaker" and the other stereo speaker as the "second speaker," which should be a good quality speaker, but is otherwise not defined here.

The total LM12 usage for this proj-

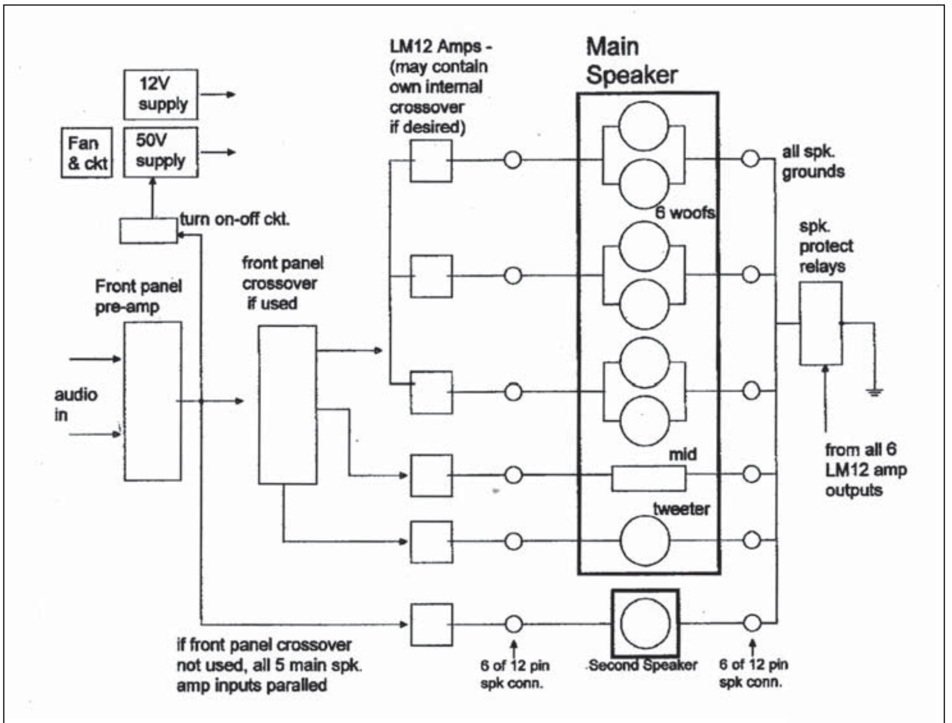

FIGURE 1: LM12 block diagram.

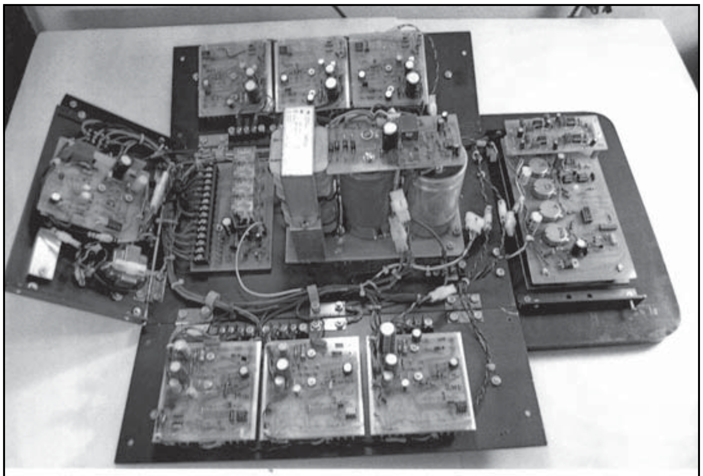

PHOTO 1: Completed amp with sides down.

PHOTO 2: Front panel.

ect is seven: three LM12 amps for the woofers and one each for the midrange, the tweeter, the other (second) stereo speaker, and the voltage regulator to power the LM12 amps. **Figure 1** shows connections between items. **Photo 1** shows installed components with all sides down. **Photo 2** is a view of the front panel. There is no negative power supply, so all amps are biased for proper operation.

One LM12 regulator to power all six LM12 amps seems like a hardship on the regulator. However, for my particular application, a relatively small amount of power is required from each LM12 audio amp because of their use in my home (I am not one of the boom-bangers). The woofer speaker cabinet design is very efficient, requiring a small amount of power to shake the walls if you so desire (I do not).

This circuit design uses a lot of separate amps for the job, but I thought it was a neat approach. Other circuit design features include:

1. Front panel preamp
2. Speaker protective relay circuit
3. Automatic turn on/off circuit on same board with power rectifier/filters
4. 50V regulator and automatic fan control circuit
5. Optional active op amp bandpass filter for woofer, mid, and tweeter
6. 75V and small 12V power supply

The first cabinet design utilized a formed U shaped chassis to hold the circuits. As you can see from **Photo 1**, I quickly needed to redesign the chassis with hinged front, back, and side panels!

TRANSFORMERS AND RECTIFIERS

Look at the transformers used in the power supplies (**Fig. 2**). There is a small 12V transformer connected to the 120V line at all times. This provides continuous operating power for the automatic turn-on/off circuitry. There are two 24V transformers connected in series to power the LM12 amps. Note that power to them is applied through contacts of the "turn-on" relay.

Shown here also is the cooling fan controlled by the "fan turn-on" relay. Rectifier-filter circuits are shown for each transformer with a front panel LED to indicate when the unit is operational. There was no real need for such a high voltage supply for the LM12 amps. I provided it just in case I might use the amp for some other—as of now unknown—application.

One 24V transformer would have been been more than sufficient for my application. **Photo 3** shows the 75V diodes (left) and the 12V supply (center of board). The turn-on/off circuit (**Fig. 3**) is on the right-hand side of the board.

THE TURN-ON/OFF CIRCUIT

This circuit board is mounted on the terminals of C1 (**Photo 3**). Incoming audio through C3 is applied to the LT1006 op amp, whose output is clamped, rectified, and filtered by D3,2 and C7. This positive DC is sufficient to turn on the FET Q1, which energizes the relay, which, then, turns on the LM12 regulated power supply.

An LED mounted on the board indicates that the relay is energized. If audio is not present for a pre-determined time, then the voltage of C7

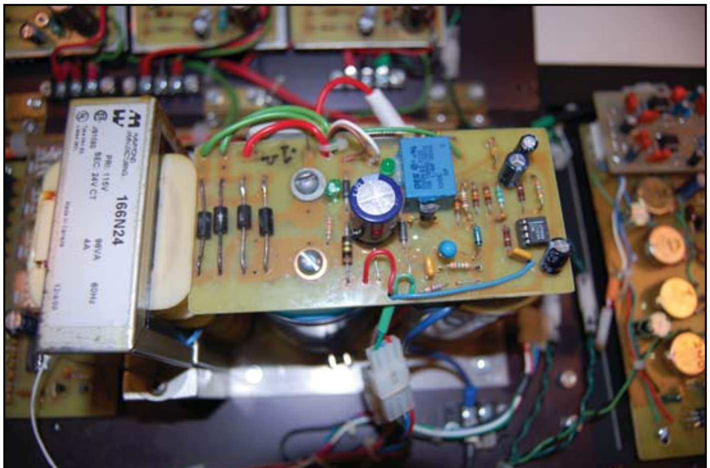

PHOTO 3: Auto turn-on/off circuit and power supplies.

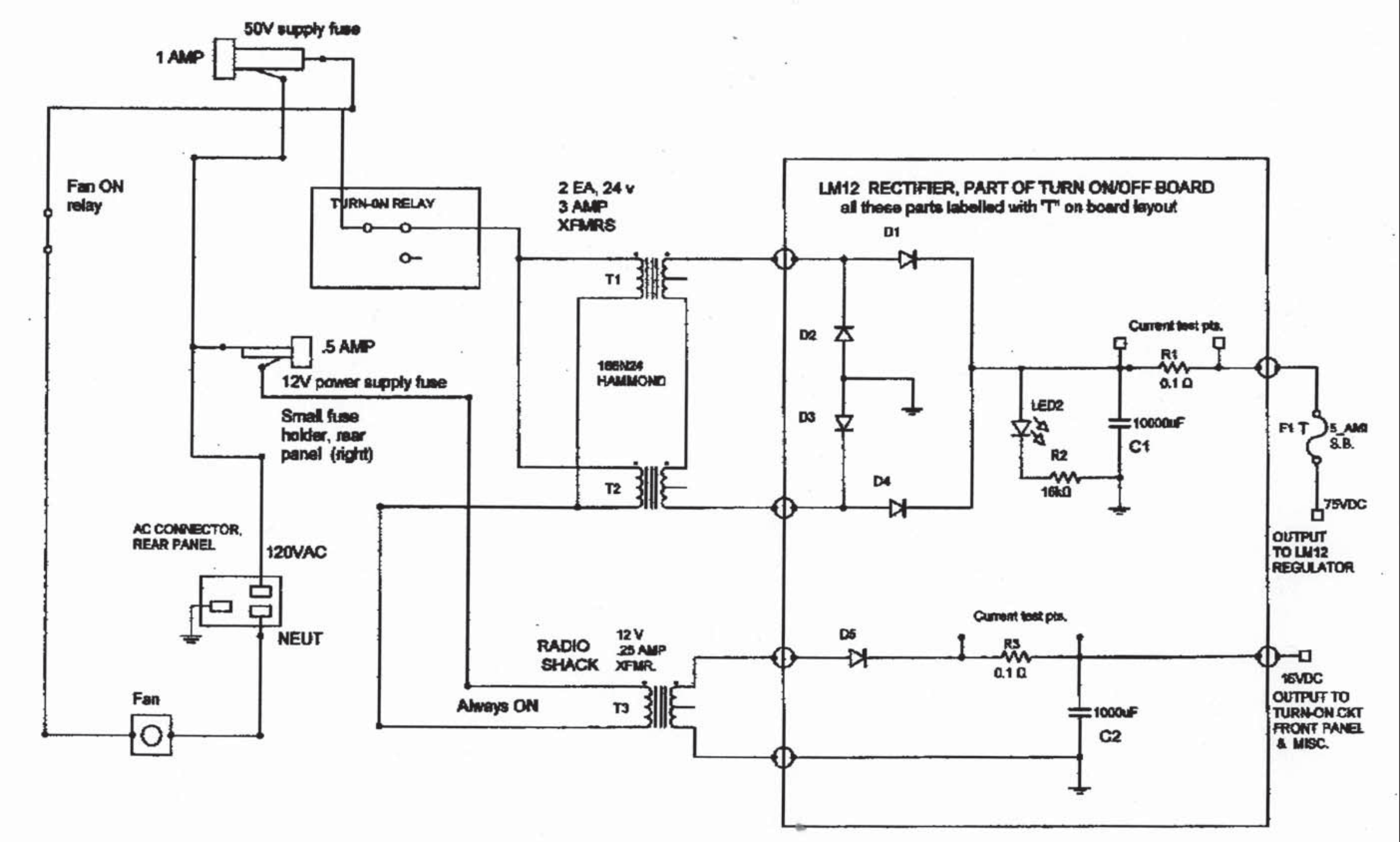

FIGURE 2: LM12 power transformers.

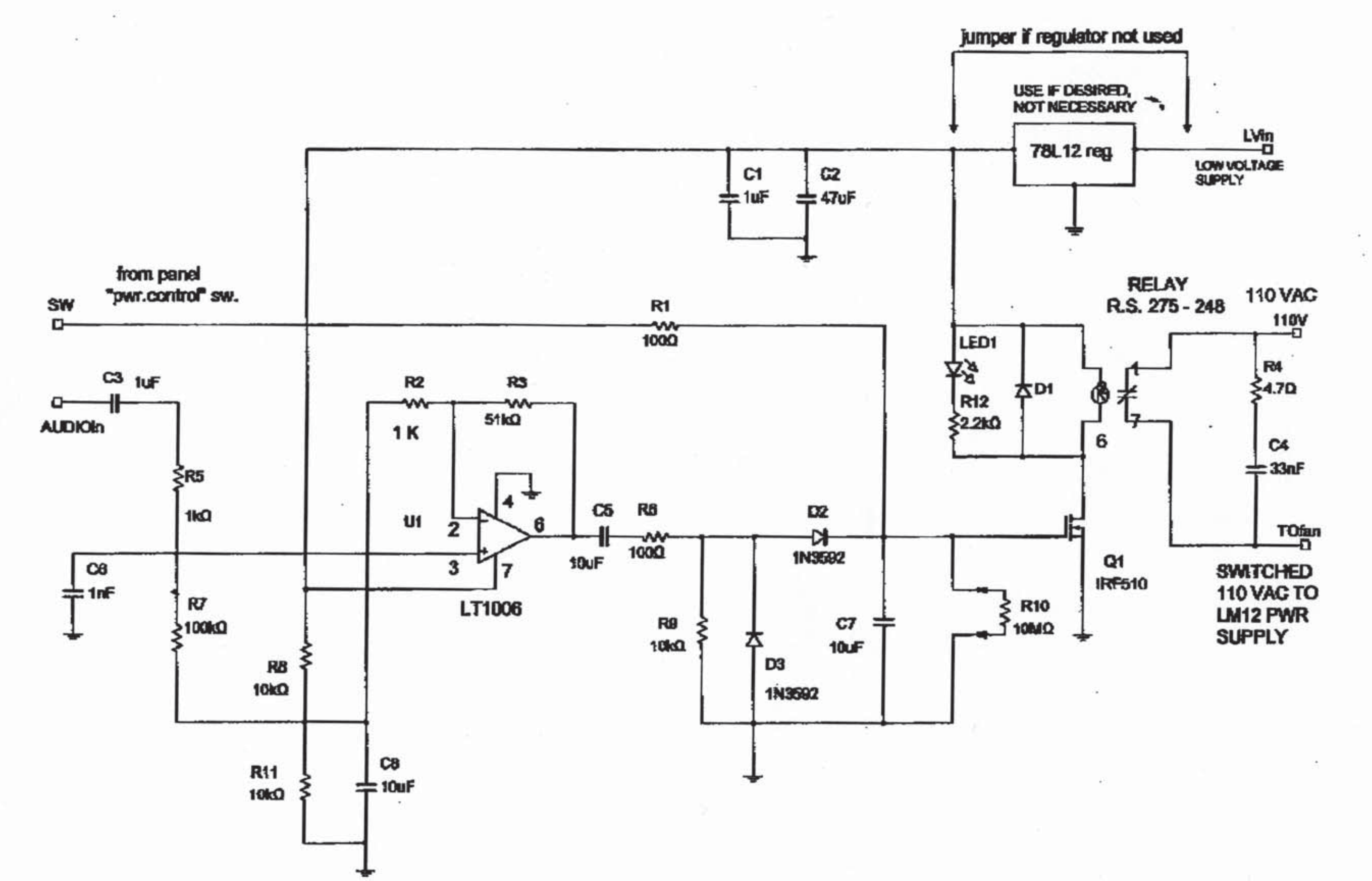

FIGURE 3: Auto turn-on/off circuit.

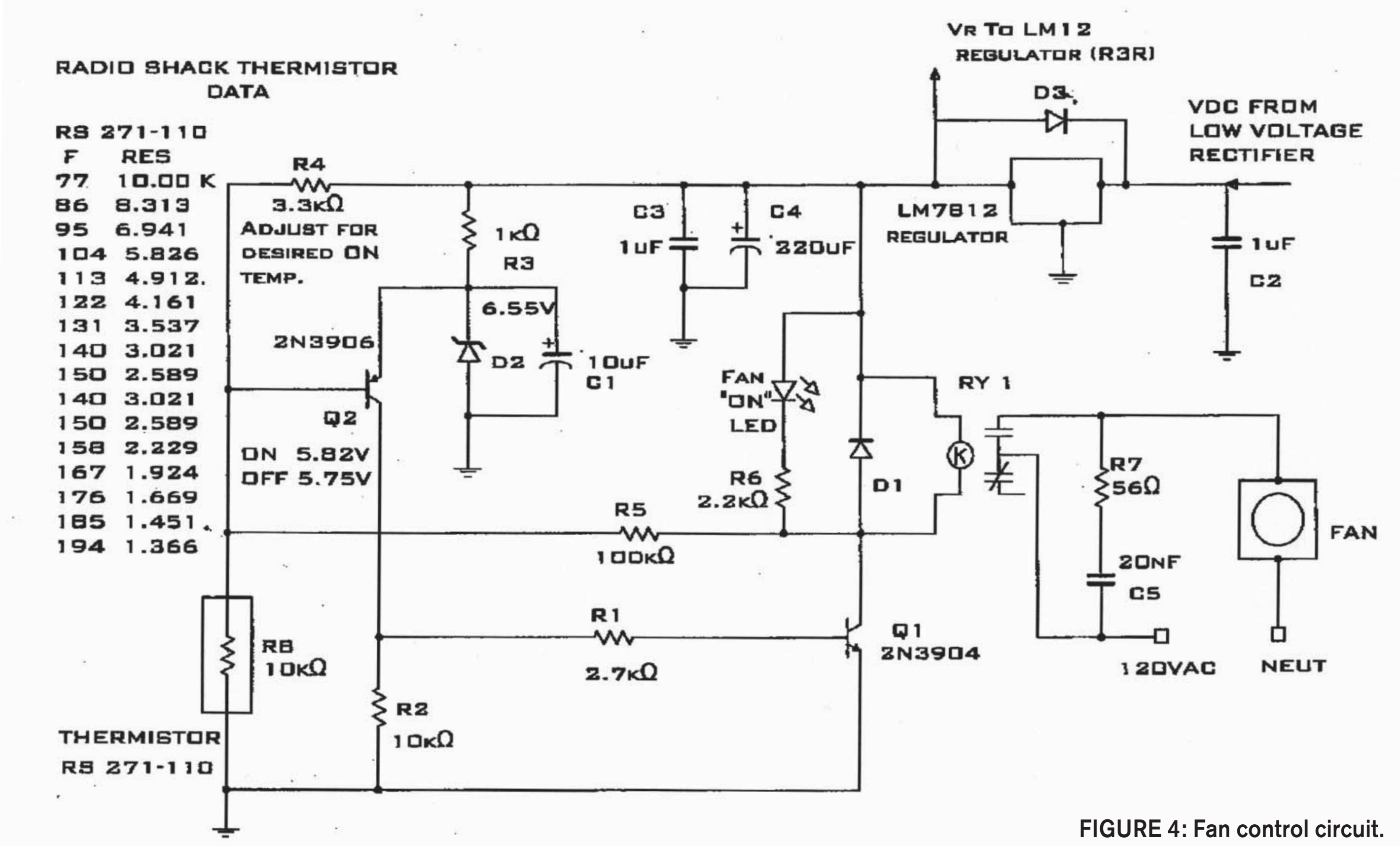

FIGURE 4: Fan control circuit.

declines to a point where the relay is de-energized turning everything off (except the 12V supply). You can set this time period by changing the value of R10. I made this resistor "pluggable" for ease of change if needed.

You use a switch on the front panel to manually turn the amp on or off or to select auto on/off. To turn the amp off, the gate of Q1 is grounded by the switch through R1. To keep the amp permanently on, a positive voltage is applied to the Q1 gate. For the auto mode, R1 is left disconnected. You can apply power to this circuit through the 12V regulator chip, but regulation is not necessary (or really needed).

FAN CONTROL CIRCUIT

The fan control circuit (**Fig. 4**) is located on the same board with the 50V regulator on the top half of the board (**Photo 4**). I just as well could have simply let the fan run continuously, but I had fun devising a thermistor control circuit to measure the regulator LM12 temperature and turn on the fan at a determined heat level. Notice that Q2 here will turn on if the value of R8 compared to R4 causes its base voltage to drop to about 6V DC. R8 is a temperature variable resistor (thermistor) whose resistive value decreases with increases in temperature. I mounted this item on the heatsink using extension wiring from the board and used a Radio Shack thermistor with the listed resistive versus temperature values.

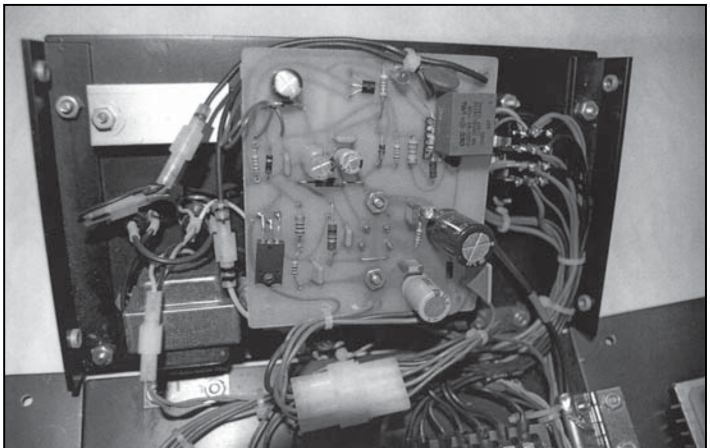

PHOTO 4: Fan control on same board with 50V regulator.

If Q2 turns on, this will cause Q1 to turn on, energizing RY1 and turning the fan on to cool the LM12 regulator. This fan air can also be useful to put fresh air into the cabinet. A bit of positive feedback between the two transistors through R5 provides some hysteresis, giving the turn-on and -off action a good "snap" and also providing a small temperature difference between turn-on and -off so the circuit provides a decent cooling effect to the regulator before turning the fan back off.

This same action is present in your home thermostat. For instance, it might turn on the air conditioner at a room temperature of 80° and cool the room down to 75° before turning it off, then waiting for another 80° before repeating the same. This characteristic keeps the temperature controller from "jiggling" on and off with

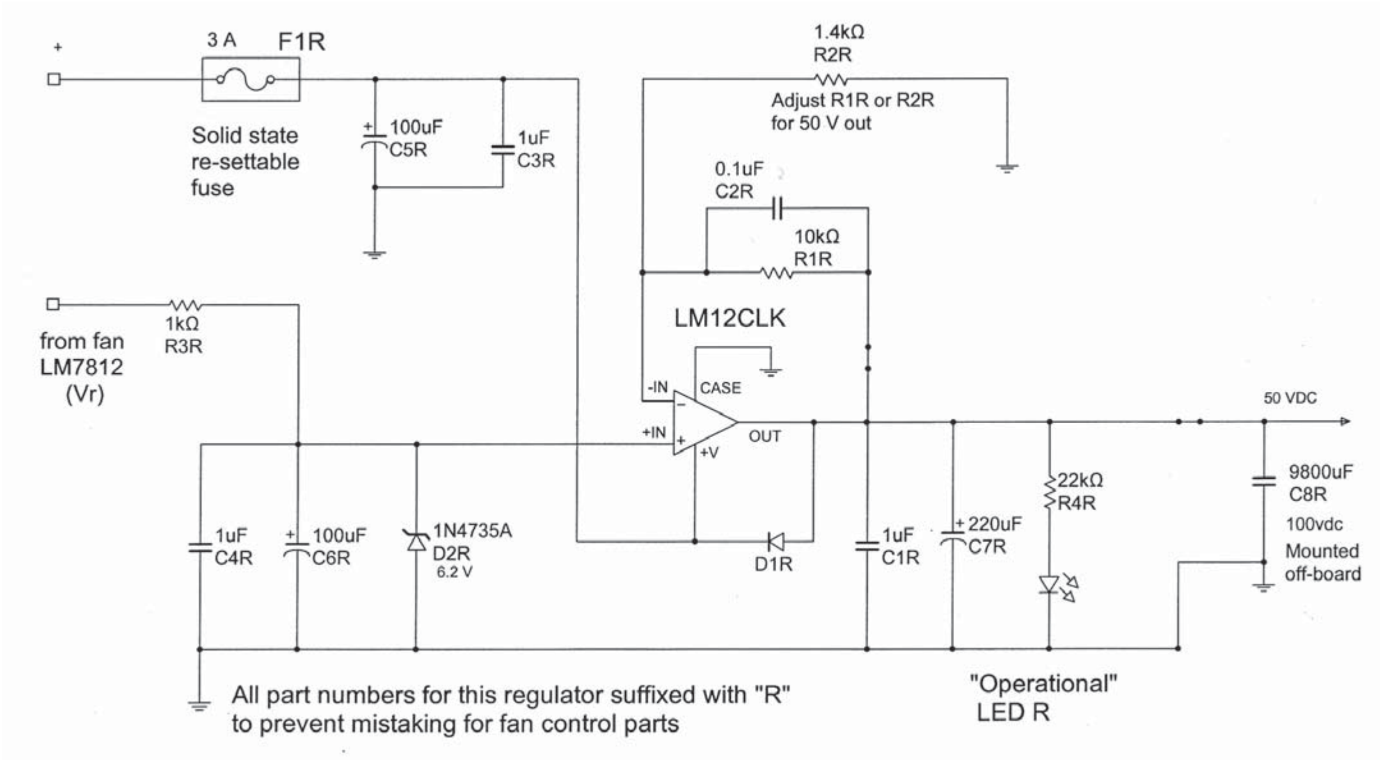

FIGURE 5: LM12 50V regulator.

very slight changes in temperature.

I provided an LED on-board to show the fan on/off action. I also provided the necessary position for a voltage regulator chip. With R4 being 3.3K, Q2 will turn on at a thermistor temperature of approximately 140° F; you can change the value of R4 to adjust this if desired. The circuit with heatsink is mounted inside the cabinet with the fan on the outside blowing air through an appropriate hole cut in the panel.

THE 50V REGULATOR

The 50V regulator (**Fig. 5, Photo 4**) is located on the lower half of the board. You can see the four LM12 pins and case bolts projecting through the board from its bottom side. Because this regulator is on the same circuit board with the fan control, I used an "R" following any components related to the regulator. This easily prevents confusion between components used for the fan and regulator.

This circuit is just a simple op amp amplifier, except that, compared to the usual, it has a relatively huge power and voltage capability. Input bias voltage to the LM12 is provided by a regulated 6.2V DC by D2R. The gain, set by R1R and R2R, amplifies the bias voltage to an output voltage of 50V. A bit of frequency compensation is provided by C2R. Specifications provided with the LM12 state that heavy capacitive loading on the output is OK.

Again, an LED is onboard to indicate output. D1R prevents reverse polarity between the output and the +V terminal as the circuit is turned off. I was pretty generous with filter capacitor values—a whole lot is not too much! The heatsink and LM12 are mounted the same way as on the LM12 amplifier boards. Note that the position of the LM12 regulator IC is different, so you will need to prepare the heatsink holes differently, but using the same technique.

SPEAKER RELAYS

Photo 5 and **Fig. 6** show this circuit board, which does the following:

1. Observes the output of each LM12 amp, detecting any DC output from coupling cap failure that might be applied to the speakers. This condition will quickly de-energize the speaker-connecting relays.
2. It will de-energize the relays in case of lower than normal voltage from the 50V power supply.
3. This same circuit provides a short delay before connecting the speakers at turn-on, preventing any possible "thump" sound from them.

Relay disconnect will be indicated by a front panel LED. When Q4 turns off, base voltage to Q3 goes high enough to turn on Q3 and the front panel LED.

The U1 NOR chip output will go low with any of its inputs being held high. These inputs are connected to each LM12 amplifier output. Normal AC from the amps is filtered at the inputs by the 1 meg and 1µF RC combination. When 50V is applied to the circuit, Q2

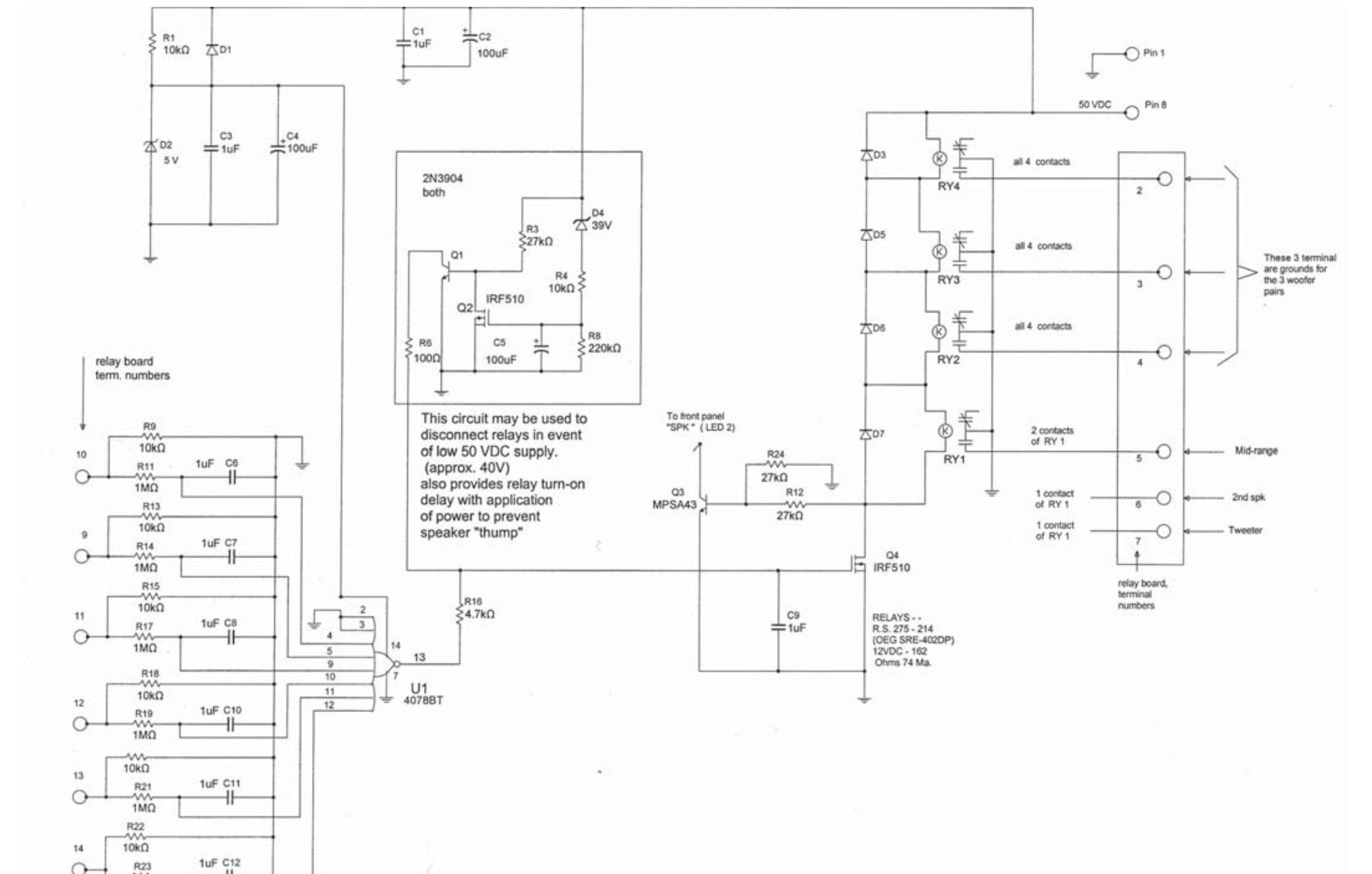

FIGURE 6: Speaker relays.

turns on after a short delay provided by R4 and C5. This turns off Q1 disconnecting R6 from the Q4 gate. Should the 50V supply drop to approximately 40V, then Q2 turns off and Q1 turns on grounding the gate of Q4 and turning it off; the speakers are disconnected.

THE FRONT PANEL PREAMP

Photo 6 and **Fig. 7** show the front panel preamp and also the front panel crossover board plugged into the right side. R4 and R14 can be a double pot to provide single knob balance for the two channels. If so, the pots are "reverse wired" so that rotating the shaft increases the audio to one of the channels and decreases the audio to the other. You will need to hand-wire the center tap of the pot providing audio to the "main speaker" to the "Main Audio input." Connect the input to this pot to the wire that normally connects to the "Main Audio input," and the other terminal goes to ground.

Audio from the "Main Audio input" is applied to U1. Its output feeds three pots for setting the individual levels to the three LM12 speaker amps and R4, if it is used. These adjusted levels can go directly to the LM12 amps or to the active filter board if you choose to use it. The audio for the second speaker simply feeds directly to the LM12 amp through U2. I provided four level detecting comparators, which indicate when one of the audio levels has reached a preset level.

This trigger level is adjustable with R17. This circuit can indicate that the audio is reaching the clipping level of the LM12 amps or any other level you might be interested in. You can adjust LM12 amp audio with R1, R4, 5, 6, 7, 11 to provide the desired operation of this circuit. There is a switch position provided to control the operation of the automatic on/off circuit. Note that it is a three-position switch: "on-auto-off."

There is a "speaker disconnect" LED driven from the speaker relay board to indicate activation of the speaker

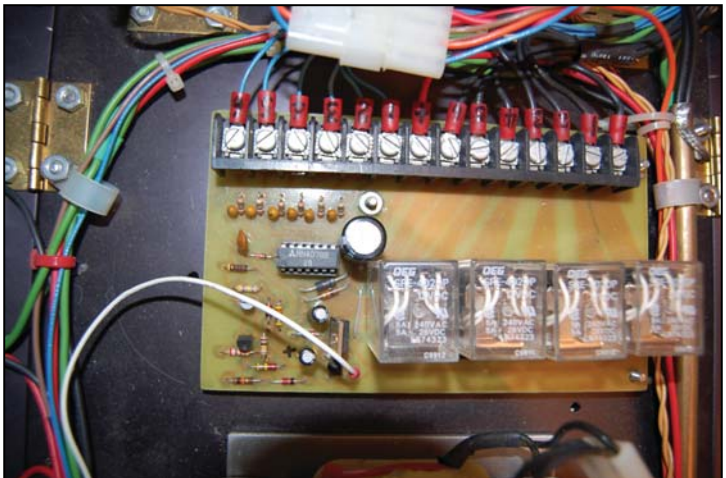

PHOTO 5: Speaker relays.

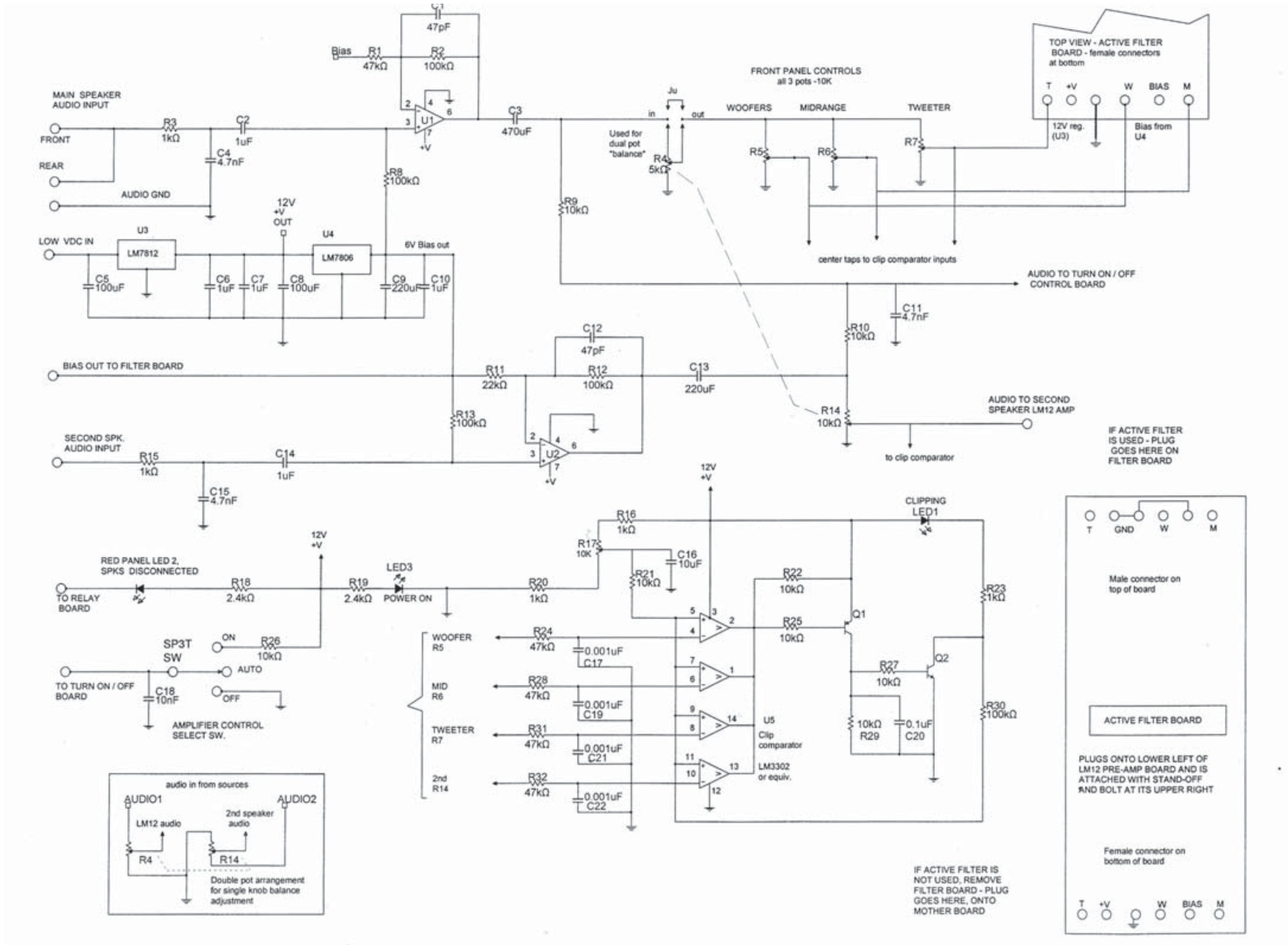

FIGURE 7: LM12 front panel preamp.

disconnect relays. Two voltage regulators are included for op amp power and bias. Each audio input is provided with an RC low-pass filter for removal of any possible RF that might be present. Note that R9 and R10 sample audio from U1 and 2 passing it on to the automatic turn-on/off circuit. There are no critical resistive or capacitive values in this circuit.

This board was designed to be made as a two layer or one layer only. For a one-layer board you will need to place four jumpers, one through four as noted on the parts layout and three connecting wires from center of R5, 6, 7 to Tweet, M, and W at the lower left corner of the board.

THE FRONT PANEL CROSSOVER

Photo 6 and **Fig. 8** show this board mounted on the right side of the front panel preamp board. This small board contains three active filters, one low pass for woofer U4; one band pass for mid range U1,2; one high pass for tweeter U3.

This filter, as you can see, is only for the "main speaker." This small board is designed to simply plug onto the front panel preamp board, on the lower left-hand corner. A bolt at its upper right corner fastens it to the preamp board.

The cable that would normally plug onto the preamp board output at the lower left corner will plug onto the active filter board at its top end when the filter board is used. Each active filter has an input buffered by U1a, 3a, or 4a. The filters are simple double pole filters that you can configure as active or not by simply mounting C7, R8, R21, or C21 to ground or as fed back from the amp output. Pads are provided for either choice.

The selection of component values for your particular

PHOTO 6: Front panel crossover filter board and front panel preamp.

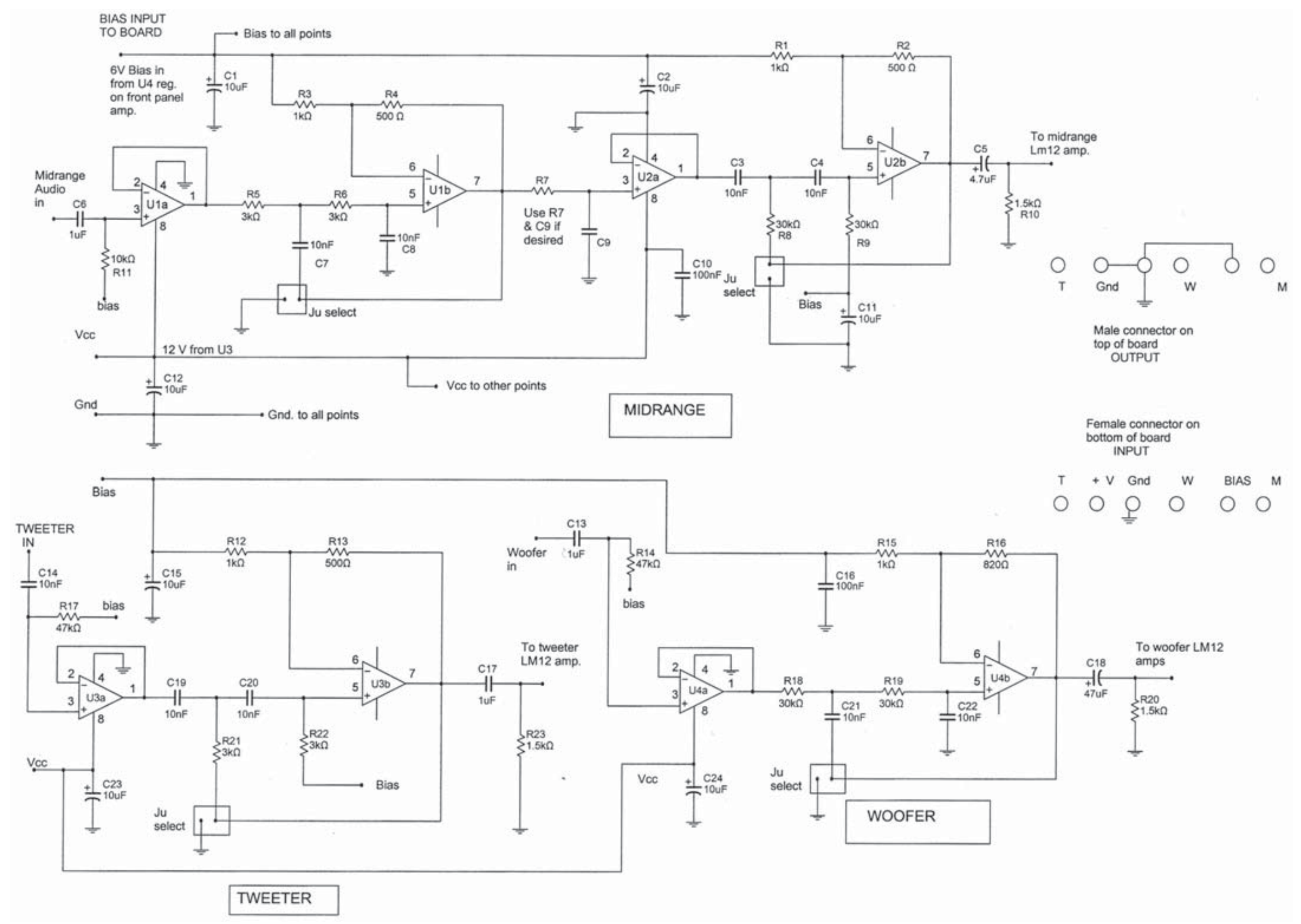

FIGURE 8: LM12 front panel crossover filter board—woofer, midrange, tweeter.

speaker arrangement will, no doubt, be different from mine. As mentioned before, use of a decent circuit design program to help in this design is a tremendous help.

Use of eyelets implanted into the input connector pads on the circuit board is the best way to ensure good connections between the pads and the connectors. Short of these, you can use short pieces of wire placed through the pad holes along with the connectors and bent so that they connect the top and bottom pads when soldered properly. The output pads are on the bottom side only, so they do not need this.

LM12 AMPLIFIER

Photo 7 (and **Fig. 9**) shows three LM12 amps. **Photo 8** shows the heatsink side of the LM12 board.

In the event that a negative and positive swing from the LM12 is needed, you can connect a negative supply voltage to the LM12 case. In this event, positive bias to the LM12 input would not be used by eliminating R10,11 and grounding the bottom of R6. Possible negative supply is indicated on the lower left side of the schematic.

All parts mount on top of the board except the LM12, which mounts from the bottom. After constructing the circuit board, first, drill the six holes with a small bit (.040?) for the LM12. Then, carefully position the heatsink onto the board in the proper position and clamp it in place. Make certain that you clamp the heatsink on the bottom side (foil) of the board. Using the six small holes as a guide, drill through them into the bottom of the heatsink to mark the exact location of the holes to be put into the heatsink for the LM12. Then drill heatsink and board holes to the required sizes.

Mount the LM12 on the heatsink and bolt it on both ends. Use bolts long enough to extend through the circuit board. The nuts on the bottom side of the heatsink are used to connect the LM12 case to the foil pattern of the circuit board by being pressed onto them when you tighten the nuts on the top side of the circuit board holding it to the heatsink. Place eyelets into the LM12's four terminals.

Now, if you plan to use a negative supply voltage to the LM12, you probably will prefer to insulate the LM12 case from the heatsink. I placed a small insulated pad the same thickness of the nut holding the LM12 on each corner

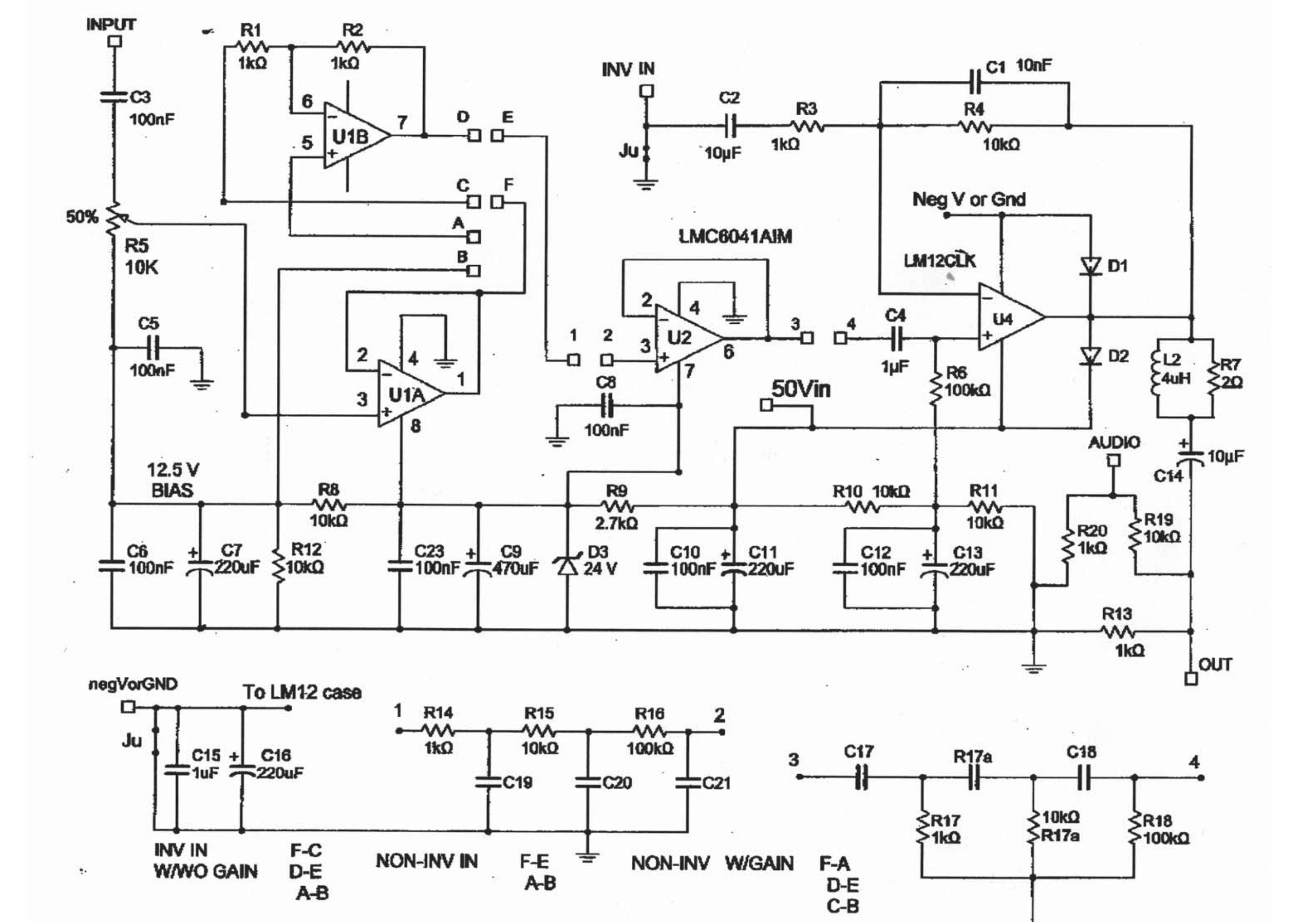

FIGURE 9: LM12 amplifier schematic.

of the heatsink to prevent the heatsink from possibly easing over and shorting to any traces on the circuit board.

There are three ICs—U1A is an input buffer with a gain of one. U1B can be driven by U1A. Note that the schematic shows the jumper arrangement to use three different configurations, inverting with/without gain, non-inverting with/without gain, or without the U1B.

U2 provides for implementing two filters between points 1 and 2 and 3 and 4 at the input and output of U2. These possible filter configurations are shown at the bottom of the schematic. They are simple 1 to 3 pole low- and high-pass configurations. The resistors increase by 10 each time as the capacitors decrease by 10. I prefer this because successive poles have little loading effect on the preceding ones. This feature enables you to drive the woofer and tweeter with proper rolloff. Also, by using both filters together, midrange speaker configuration is easy.

Note that C4 and R6 would probably be used instead of C18 and R18 for the high-pass filter in most cases, but positions for each are provided. U4, the LM12, has adjustable gain possibility as

PHOTO 7: LM12 amp.

PHOTO 8: Amp heatsink.

well as capability to be driven at the inverting input from an external source or even from the output of U2 if desired.

Low- and high-end rolloff is possible using C1 and/or C2 if needed. You can base the value of C14 entirely on the low-end rolloff characteristics desired.

Protective diodes D1 and 2 are provided. L2 and R7 positions are provided as suggested by the datasheet. Audio is available, divided down from the output if desired, by R19 and 20. Note that bias and power for all amps is provided from the 50V input, and D3 regulates the U1 supply voltage. *aX*

Table: The LM12 amp.

LM12 AMPLIFIER PARTS LIST

Resistors	
R1-3, 13, 20	1K
R4, 8, 10-12, 19	10k
R5	10k pot.
R6	100k
R7	2Ω
R9	2.7k
Capacitors	all 25V unless noted
C1	0.01µF
C2	10µF
C3, 5, 6, 8, 10, 12	0.1µF C10 75V
C4, 15	1.0µF
C7, 11, 13, 16	220µF C11 75V
C9	470µF
C14	10-2200µF see text, 75V

NOTE: R14 through 18 and C17 through C21 see text

Semiconductors	
U1	TL082 or equivalent
U2	LMC6041 or equivalent
LM12	CLK
D1, 2	2N4003
D3	1N4749 24V
Mechanical	
Heatsink	approximately 3 × 4″ Digi-Key 345
	1053ND Wakefield 641K
4 each	eyelets for LM12 pins

FRONT PANEL CROSSOVER FILTER PARTS LIST

Resistors	¼W
R1, 3, 12, 15	1K
R2, 4, 13, 16	560
R11	10K
R10, 20, 23	1.5K
R14, 17	47K
R5, 6, 21, 22	3k
R8, 9, 18, 19	30K
R7	not used
Capacitors	15V minimum
C1, 2, 11, 12, 15, 23, 24	10µF ele.
C10, 16	0.1
C3, 4, 7, 8, 14, 19-22	0.01
C6, 13, 17	1µF
C18	47µF
C5	4.7µF
C9	not used
Semiconductors	
U1-4	TL082 or equivalent
Hardware	
Connectors	6 pin - .156 1 male and 1 female
	Jameco KK series
	Digi-Key WM4404-ND MALE
	WM2104ND FEMALE
Eyelets	to fit 0.05″ pin
Stand-off	0.725″ threaded

FAN CONTROL PARTS LIST

Resistors	all 0.25W
R1	2.7K
R2	10K
R3	1K
R4	3.3K
R5	100K
R6	2.2K
R7	56Ω
R8	thermistor
Capacitors	all 15V
C1	10µF
C2, C3	1µF
C4	220µF
C5	0.02µF
Semiconductors	
LM7812	regulator
D1, D3	1N4001
D2	zener 6.55V 1N828 or equivalent
Q1	2N3904
Q2	2n3906
Relay	
Radio Shack	275-248
Fan	
Radio Shack	900-2518 110V AC

50V REGULATOR PARTS LIST

Resistors	all 0.25W
R1R	10K
R2R, 3	1.4k
R4R	22k
Capacitors	
C1R, 2, 3	1µF 100V
C5R	100µF 100V
C6R	100µF 15V
C7R	220µF 100V
C8R	9800µF 100V
I used a larger value than needed	
Semiconductors	
D1R	1N4004
D2R	6.2V zener 1N4735
LED	Green LED
LM12	CLK
Fuse	
3A	solid-state, resettable
	can be standard slo-blo fuse

FRONT PANEL PREAMP PARTS LIST

Resistors	all 0.25W
R3, 15, 16, 20, 23	-1K
R2, 8, 12, 13, 30	-100K
R11	-22K
R1, 24, 28, 31, 32	-47K
R9, 10, 14, 21, 22, 25-27, 29	-10K
R18, 19	-2.4K
R5-7, 17	-10K audio POT
R4	-4.7K audio POT
Capacitors	all 15V
C2, 6, 7, 10, 14	1µF
C1, 12	47pF
C4, 11, 15	4.7nF
C9, 13	220µF
C5, 8, 3	100µF
C17, 19, 21, 22	0.001µF
C20	0.1µF
C16	10µF
C18	0.01F
Semiconductors	
Op amps	OPA340, AD820 or equivalent
Comparator	LM3302 or equivalent
LEDs	2 red, 1 green
PNP	2N3906 or equivalent
NPN	2N3904 or equivalent
Regulators	7812, 78L06
Hardware	
Connectors	6 pin - .156 male
	Jameco KK series
	Digi-Key WM4404-ND MALE
Eyelets	to fit 0.05″ pin
Stand-off	0.725″ threaded

50V SPEAKER RELAYS PARTS LIST

Resistors	all 0.25W
R1, 4, 9, 13, 15, 18, 20, 22	10k
R2, 5, 7	not used
R3	27K
R6	100Ω
R8	220K
R12, 24	27K
R11, 14, 17, 19, 21, 22, 23	1meg
R16	4.7k
Capacitors	all caps 15V unless noted
C1	1µF 100V
C3, 6-12	1µF
C2	100µF 100V
C4, 5	100µF
Semiconductors	
U1	4078BT NOR GATE
Q1, 3	2N3904
Q2, 4	IRF510 enhancement FET
Q3	MPSA43
D2	5V zener 1N4734
D4	39V zener 1N5366
D3, 5-7	1N4004
Relays	
OEG	SR402DP
Radio Shack	9009-2361
Terminal Block	
14 pin barrier terminals 0.375″ spacing	
Molex series 72	

LM12 50V TURN ON-OFF PARTS LIST

Resistors	all 0.25W
R1, 6	100Ω
R2, 5	1k
R3	51k
R4	4.7Ω
R7	100k
R8, 9, 11	10k
R10	10 meg
Capacitors	at least 15V
C1, 3	1µF
C2	47µF
C4	0.033µF
C5, 7, 8	10µF
C6	0.001µF
Semiconductors	
U1	LT1006
D1	1N4004
D2, 3	1N3592
regulator	78L12 if used
Q1	IRF510
Relay	
Radio Shack	275-248

TRANSFORMERS AND RECTIFIERS PARTS LIST

Resistors	
R1, 3	0.1Ω 1W
R2	15k 0.25W
Capacitors	
C1	10000µF 75V
C2	1000 25V
Diodes	
All	1N4004
LED	
LED 2	Green
Fuse	
5A SB	
1A SB	
0.5A SB	
Transformers	
T1, 2	166N24 Hammond
	24V 3A
T3	any 12V 0.25A

REAR PANEL SPEAKER CONNECTOR
12-pin heavy duty connector
many types available

audio on the avenue

By David Rich

The author, a long-time reviewer and commentator on the audio scene, will appear from time to time in this space.

BI-AMPLIFICATION

Most speakers are shipped with four terminals: two drive the woofer's passive crossover network (resistors, capacitors, and inductors) and two drive the tweeter's passive crossover network (in the case of a two-way speaker). For a passive three-way speaker, the midrange and tweeter crossover are typically driven from one set of terminals, and the extra terminals enable bi-wiring.

In bi-wiring, a separate cable connects the amplifier to the woofer's passive crossover network and another set of wires connects the same amplifier to the passive crossover of the tweeter for a two-way speaker. The benefits of bi-wiring are questionable, in my opinion, aside from the dealer gaining by selling twice the cable. Given the proliferation of quad-terminal speakers, speaker manufacturers may be acquiescing to dealer pressure. Even products from The Harman Group now have them, although it is doubtful Harman Labs will issue an Audio Engineering Society (AES) paper justifying the practice of bi-wiring on the basis of their advanced double-blind speaker tests...

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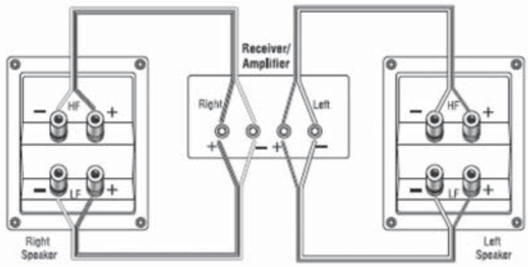
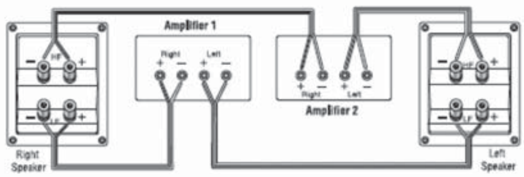


Bi-Wiring of a stereo pair of speakers



Bi-Amplification of a stereo pair of speakers using two amplifiers

Source

Infinity Classia Speaker

Owners Manuel

http://manuals.harman.com/INF/HOM/Owner%27s%20Manual/Classia_C205_C336_CC225_OMrev12-4-07.pdf

DECLINE OF THE LOCAL AUDIO DEALER

In a significant surprise, B&W, which in the past had only sold into independently owned dealers in the US, announced its intention to sell some products in the 350 Magnolia Home Theater entertainment sections inside many Best Buy stores. Magnolia Hi-Fi was a regional chain of audio/video stores until Best Buy purchased it in late 2000. Nine stand-alone Magnolia locations are left. Best Buy closed the rest. All B&W lines will be sold in the stand-alone stores.

An article in the magazine *Twice for Consumer Electronics* details the B&W decision with an interview with B&W Group chairman Joe Atkins.

http://www.twice.com/article/454938-B_W_Signs_Magnolia_Rules_Out_Online_Sales.php

This is a must read for audiophiles to understand the sad state of the independent audio dealer. It supplies more details, including numbers, than I have seen before. Here are a couple of quotes:

"In the past 12 months, the brand's own dealer base has fallen from about 230 retail outlets to about 180 outlets operated by about 145 dealers," Atkins said.

He then added: "Quite a few A/V specialists have closed their retail outlets to open up appointment-only custom-install shops." B&W was unique among larger speaker manufacturers for using independent dealers as their primary distribution channel. Definitive Technology (including the acquired Polk brand), Klipsch (including acquired Canadian Energy and Mirage brands as well as Denmark's Jamo), and Bose have been in Best Buy for many years...

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The Munich High End Audio Fair 2010

The all-digital age has not yet arrived.

By Ward Maas

This year's Munich High End Audio Fair (May 6-9) included 253 exhibitors from 24 countries who showed an overwhelming number of products within 18.373m^3. According to Kurt W. Hecker, chairman of the committee of the High End Society, the organizer of the show, this audio fair has grown over the last years into not only the largest of its kind in Europe, but also into the largest exhibition of specialty consumer electronic products in the world (www.highendsociety.de). He also presented the results of a market survey held in Germany.

This survey revealed how old current stereo systems were, how often people listened to music, and the most important factor in selecting a stereo system:

- The stereo system of a typical German is 10 years old on average. 15.8% of those interviewed had a stereo system up to 11 to 19 years old, and 13.8% had a stereo system more than 20 years old.
- The most important attribute when buying a stereo system is sound quality (62.4%), followed by longevity (29.4%). Design was only the most important consideration for 4.8% of the interviewed.
- Music is an integral part of everyday life for Germans. More than half of them (50.4%) listen more than an hour a day. 15% of those interviewed listen for five hours or more. Only 4.5% go completely without music.

So there seems to be ample opportunities for the business.

PRODUCTS ON DISPLAY

At a show like this, you need to look hard for the real gems. Due to the sheer volume of products displayed, it's easy to overlook the products of Constellation Audio. Their booth was far from spectacular, but their products were. I received a tip from Jan Didden to look at the preamp designed by John Curl. I was grateful that Peter Madnick the VP Engineering from Constellation Audio took the time to show me some of the Audio Precision graphs from the Altair and Orion products. Now you can write an article or even a book on these products, but the noise floor graph sums it up nicely (**Fig. 1**).

If you have any aspirations to design the best preamp in the world regarding specs, just forget it. John Curl put the crown safely out of reach—most impressive! By the way, its looks are not bad either (**Photo 1**, www.constellationaudio.com).

Having said that, when I spoke to VandenHul designer Jürgen Utee, he showed me their "Grail" phono preamp (**Photo 2**). A rather unusual approach of design with inductors instead of capacitors for the RIAA correction. Besides that, it included HF material PCB, HF style PCB layout, custom integrated circuits, and a custom mechanical vibration damper. Again, most impressive. I am looking forward to a listening test (www.vandenhul.com).

In another corner I found the 15″ Kilimanjaro-Series field coil speaker (**Photo 3**). Just a wooden stand with two speakers and a stack of business cards on the floor. It turned out that CEO Wolf von Langa needed to attend two audio fairs at the same time, leaving these wonderful products to be discovered by those who wondered what that power supply connector was doing on a loudspeaker. Field coil loudspeakers do have two coils—one traditional voice coil and one coil to generate the magnetic field. To use such loudspeakers, you need a power supply connected to this field coil.

The focus of the Kilimanjaro-Series speakers is mainly on vintage equipment. But because prices for products like that are becoming more and more acceptable (I did not say cheap), they become more and more open to experimentation. By changing the current through the field coil, you can change the T/S parameters. Of course, they are also ideally suited for listening to music (www.kilimanjaro-series.com).

A different fundamental approach to loudspeaker chassis design was taken by Audio Consequence. Their 190mm inverted coaxial loudspeaker chassis (**Photo 4**), which will be available in the third quarter of this year, is not only remarkable in appearance, but also offers a solution to the design problem in coaxial chassis where an ideal position of the tweeter is less ideal for the woofer

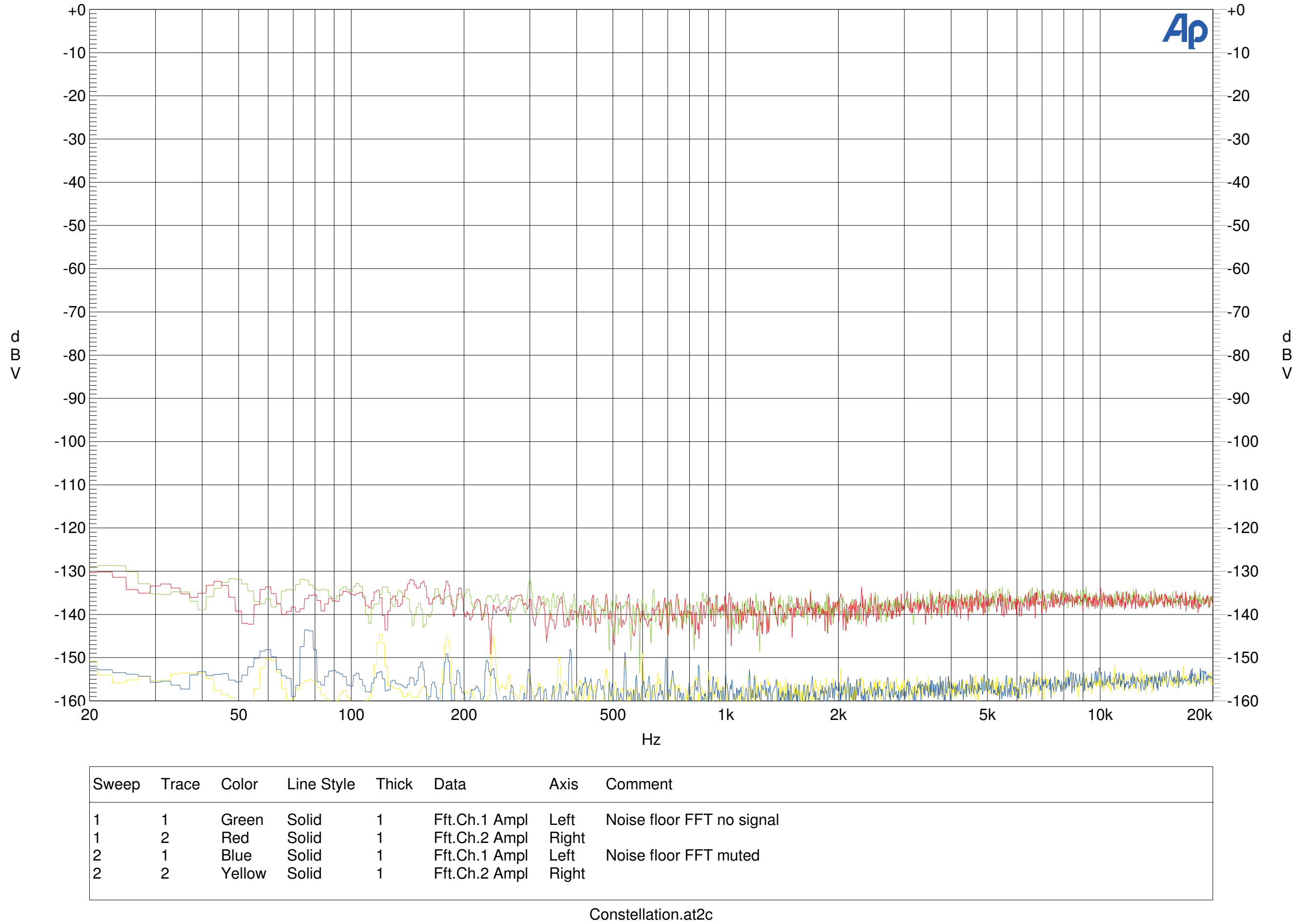

FIGURE 1: Audio Precision low-noise graph.

and vice versa. This loudspeaker chassis is usable in a full-range system (20-25 ltr) going down to 35Hz. Even 20Hz is possible with a limitation regarding output power. And even an Xlin of ±6mm and an Xmax of ±11mm eventually comes to an end. Support of a woofer up to 200Hz can be a welcome addition (www.audioconsequence.de).

For the DIYer, the Ground Sound booth offered a large range of modules for active amplification: digital crossover modules, power amplifier modules from 100-1500W, power supply modules, and transformers (**Photo 5**). With their Xoverwizard II software, they demonstrated the ease in making a decent setup for an active version of the Focal Esprit loudspeaker (www.groundsound.com).

Since 2005, the "Technology Stage" has offered a range of lectures as an integral part of the show, including a variety of lectures and forum discussions ranging from the basics of mains- and power supplies and the ins and outs of (a)symmetrical connections to high-end digital room correction. In particular, the full-hour presentation of Mr. Hans M. Strassner about mains sup-

PHOTO 1: Constellation Audio preamp.

PHOTO 2: The Grail phono preamp from VandenHul.

PHOTO 3: The Kilimanjaro-series field coil speaker.

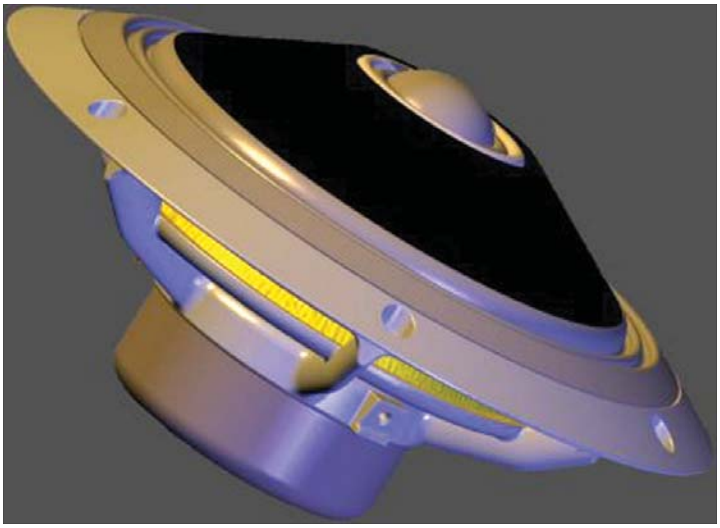

PHOTO 4: The Audio Consequence speaker chassis.

plies, mains filtering, and power supplies did not leave a seat free (**Photo 6**). It was remarkable to see that everything he said was soaked up by a very attentive audience. I hope that this part of the show will be extended, because there is a great need for this type of factual technical audio knowledge.

One of those areas where knowledge and understanding will lead to an improved listening experience is acoustics. While bad room acoustics are often to blame for a less than optimal listening experience, the solutions are often sought in new equipment, cables, or recordings. One of the companies offering a wide range of products to solve acoustic problems is Vicoustic (www.vicoustic.com). From bass traps and absorption panels to diffusion panels and accessories, they have it. So next time you are not satisfied with the overall audio quality of your audio system, you might consider experimenting with the room acoustics.

At a show like this you always find the products and displays that bring a smile to your face, like the Analog Domain Audio Apollo amplifier (analogdomain.eu). Rated 4000W in 8Ω/or 8000W into 4Ω, this is truly a monster with a triple switched class H output stage. Or a product such as the "Magic Flute" loudspeaker from Swspeakers, which looks like a jet engine display that ran away from an air show (**Photo 7**, www.swspeakers.com).

You often see amplifiers displayed with open hoods. Some companies go to great lengths to show their product's innards. A product such as the Ayre KX-R preamp is definitely a beauty (**Photo 8**, www.ayre.com). But what to think of a Mark Levinson amplifier taken apart so completely that it looks as though they're selling the components on the DIY market (**Photo 9**, www.marklevinson.com)?

If you truly are in audio, then you need to look at the products of Silent Wire (**Photo 10**, www.silent-wire.de). There are many discussions regarding cables and how audible the use of ordinary or not so ordinary materials is. I am confident, however, that a lot of discussion will come to an abrupt stop

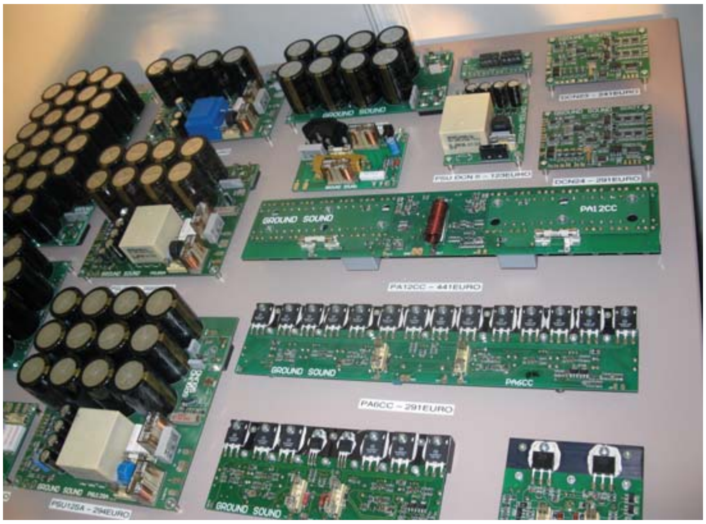

PHOTO 5: An overview of Ground Sound products.

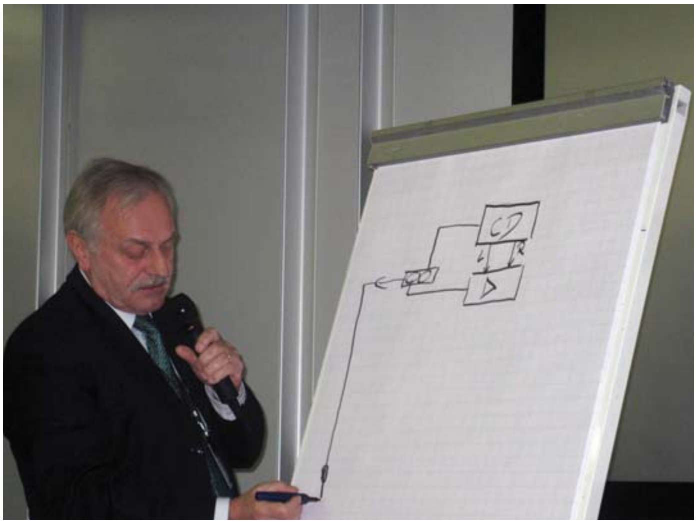

PHOTO 6: Mr. Hans M. Strassner explaining mains cables connection.

PHOTO 7: The Swspeakers "Magic Flute" ready for takeoff.

PHOTO 8: The Ayre KX-R preamp.

with this loudspeaker cable. Prices begin at 25,000 Euro (over $30,000).

WHERE'S THE DIGITAL?

Some time ago, I had the pleasure of attending an AES lecture at the Amsterdam conservatorium by famous recording engineer and producer George Massenburg. As usual, his "lecture" was more a large listening session. One of the highlights was a song by Dawn Langstroth. It's been a rather long time since I've heard a piece of music that gave me goosebumps. I found out that her album was available as a digital download in 24-bit 96kHz FLAC on www.linnrecords.com. I was wondering what you would need to play back such a file in the best quality possible. Or, for that matter, what if the recording was in a 24 bit 192kHz format. You would expect to find the answer to this question at a show like this, with its many products on display. For instance, if you look at the record-breaking number of mastodon-size LP turntables at this show, you see that no effort is spared to get the best out of an "old" LP disc. I was rather disappointed that many efforts to play back digital audio recordings were limited to gadgets. Of course, with some effort you can find equipment that will play back a high-quality Dawn Langstroth track, but it seems that more attention is given to Internet Radio, visual presentation, Wifi, and PC software.

I vented my frustration to Christian Rechenbach, who is responsible for the German digital audio only magazine "Eins-Null" (meaning one-zero). He was expecting more high-quality digital equipment at the show as well. "May be more before Christmas," he said. "Most companies do not want to miss the boat and are looking more to each other than the consumer." But it is a "when" and not an "if."

So at the end of the last day of the show, when all crews were preparing to pack the monster turntables in large crates, some of the workers looked rather happy. The Eyjafjallajokul volcano in Iceland made flying from Munich impossible, but as the sun was still shining after all the heavy lifting, at least there was some extra time to spend in one of those Bavarian beer gardens where the audio discussions went on. *aX*

PHOTO 9: A disassembled Mark Levinson amp.

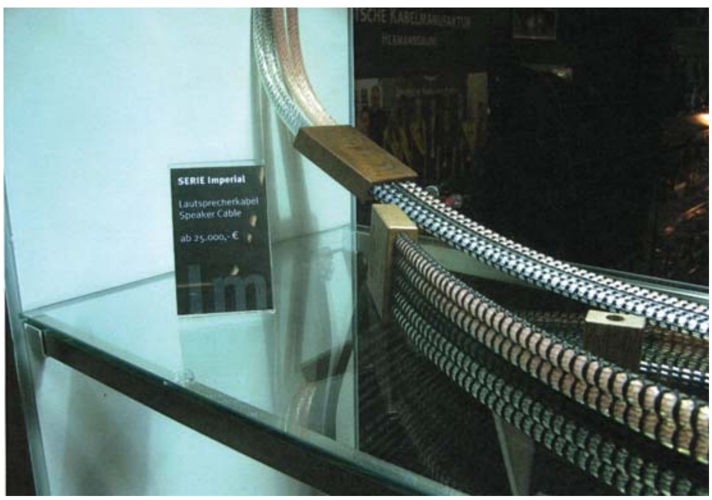

PHOTO 10: Pricey cables from Silent Wire.

audio news

Compiled by Richard Surrette

Belden has recently released a new line of Compression Connectors, with fine-tuned tolerances to ensure maximum signal reliability and performance. Ensuring a secure, positive compression, the Compression Connector Tool results in quicker installations. These solutions include an ergonomically designed tool with increased strength and durability, and two stripping tools (one for standard cable, and a lighter model for more delicate cable). For more information, visit www.belden.com.

Now on clearance from **THIEL**, the CS7.2, once the company's flagship model, has recently been discontinued. It paves the way for Thiel's new high-performance CS3.7 speaker. To get a chance to buy a pair of these classic loudspeakers, contact gdayton@thielaudio.com.

The NNLS speakers, from **NaimNet**, feature high-performance drive units and long-throw bass with pressure-cast aluminum-alloy baskets. In-wall versions of the NNLS speakers require maximum 90mm depth, as the cabinets mount into a simple rectangular cutout with solid aluminum toggles. On-wall models mount with two- or four-point wall fixings. Models include the NNLS02 and NNLS03, which are mid-range units configured to reduce side radiation, as well as the NNLS01, with wider dispersion. To learn more, go to www.naimnet.com.

Audio-Technica (audio-technica.com) is now featuring ribbon microphones, the AT4080 and AT4081. These bidirectional mikes contain N50 neodymium magnets, ultra-fine mesh, high-SPL capability, and phantom-powered active electronics. These handcrafted microphones have 18 patents pending and utilized MicroLinear ribbon imprint.

The **Onkyo** HT-S6300 and HT-S7300 are premium 3D-ready packaged home-theater-in-a-box (HTiB) systems. Both have HDMI-1.4A inputs, 1080p video upscaling, lossless Dolby and DTS high definition audio, and more. The HT-S7300 features 41″ tall floor-standing speakers with dual, vertically arrayed woofers, while the HT-S36300 system consists of six 1′ tall bookshelf speakers and horizontal center speaker with a 4″ two-way driver. Both use a 290W subwoofer working with the receiver's seven 130W 6Ω power amps. For more information, visit www.onkyousa.com.

Using a simple amplifier to provide pure sound in the earpieces, the **ZEM** headphones feature non-electronic noise cancellation. The headphones' only electronics are two miniature electromagnet receivers that convert signals into sound. Its 3.5mm plug is compatible with MP3 players, CD players, cell phones, laptops, and more. To learn more, go to www.SensGard.com.

The KD-VP1250, from **Key Digital**, is a digital video processor, switcher, and distribution center which up-converts/down converts SD, HD, XGA, SXGA, and WXGA resolutions of analog and digital video inputs. Benefits of this device include frame rate conversion from 50-60Hz and from 60 to 50Hz, optical IR, on-screen display, and discrete remote codes. Similarly, the KD-VP750 is a digital video processor and switcher capable of scaling 480I standard definition and 480p high definition component, composite, and S-video sources with VGA output to accommodate component video format with a supplied breakout cable. It features a universal video center to consolidate multiple video formats to a single component, as well as an RS-232 control. For more, visit www.keydigital.com/.

Earthworks Audio's new website is now live, featuring a range of media clips and articles sharing tips, tricks, and more. The site contains a featured artists section, as well as new product categories. Check out www.earthworksaudio.com.

New from **Phase Technology**, the Teatro WL-SURR is a line of wireless surround speakers requiring only an AC power connection. It features an on-wall bipole/dipole switchable design and active digital crossovers along with 100W total amplifier power split between the woofer and tweeters. Its transmitter operates at 2.4GHz and uses frequency hopping. Another new line of speakers from Phase Technology includes the SPF-15, a surface-mount 76mm speaker with a rotating mounting system; the CI 1.5, a round ceiling-mount speaker with 3″ polypropylene full-range driver and providing rapid fixed-wing installation; the CI 15, an in-wall speaker; and the CI-MM3-II, which updates the 3″ Spacia speaker. The CineMicro Series, also from Phase Technology, is a line of audiophile-grade speakers for use with home theaters. The CineMicro One contains four satellite speakers, a dual-woofer center-channel speaker, and an 8″ long-throw subwoofer, and handles 100W per channel. To learn more, go to www.phasetech.com.

The en•Core 300, from **Blue Microphones**, is a performance mike with open architecture design and reinforced build. It is mounted on a rubber suspension system and coupled with a phantom-powered circuit tuned to match the capsule. Its design maximizes air volume, which results in uninterrupted sound, and reduces in-chamber resonances. Other features include a reinforced chassis and a heavy-gauge zinc barrel grip. For more information, visit www.bluemic.com.

Some of **Audience**'s (www.audience-av.com) new products include the ClairAudient 2+2 and ClairAudient LSA 8+8 loudspeakers, which use the full-range A3-S drivers, as well as the AdeptResponse aR6-TS and aR12-TS power conditioners, Wavemaster buffered autoformer preamp, prototype Wavepower class D monoblock power amps, Au24e cables, and Au24 powerChords.

Piccolo, a 50W amplifier kitset, uses the LM3886 amplifier chip as well as low negative feedback. It retains low distortion of 0.03%, and features a power supply of paralleled capacitors integrated on the same compact PCB. For more, go to www.designbuildlisten.com.

OWI Inc. introduces the AMP-1SGRN "Green" amplified speaker amplifier with priority audio override, which integrates audio along with an emergency paging solution through its priority override. This class D power amp has low radiated electromagnetic interference and protects itself from over-temperature, over-current, and over-voltage by shutting itself down under any of these conditions. To learn more, visit www.owi-inc.com.

The **Kicker** HP201 full-range active-style headphones feature open-cell-foam speaker coverings, and use a 53″ Kevlar-reinforced cable. For more on these durable headphones, contact www.kicker.com.


Brass studs isolate speaker from vibration
Pillars can be filled with shot or sand for exceptional stability
Cork damping rings absorb acoustic resonance
Hi-tech poly-mineral composite base reduces vibration
Neoprene gaskets isolate speaker from ambient room vibration
High-strength custom carpet spikes create stability