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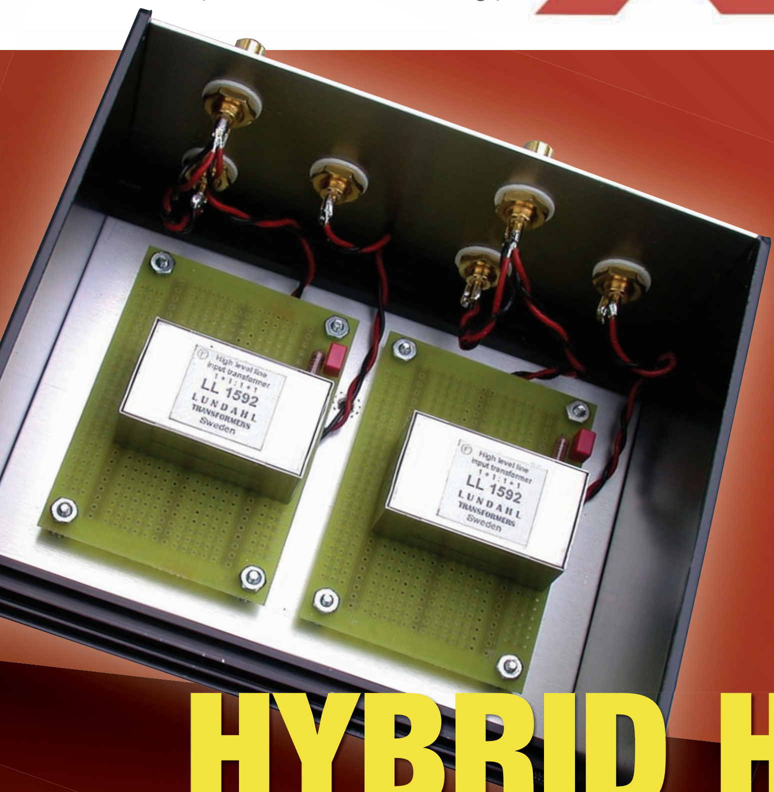

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The CurveArray

This line-array design is ideal for your center-channel home theater setup.

PHOTO 1: The CurveArray.

Line arrays offer a method to achieve high efficiency in a relatively small package. However, conventional line arrays also have a very narrow dispersion angle on their long axis. That's a plus for minimizing early reflections when the cabinet is upright, but a problem in applications where a horizontal placement is the only option, which is very often the case with center-channel home theater boxes.

The CurveArray (**Photo 1**) is intended for horizontal placement, offering the high sensitivity of a line array with the dispersion of a point source. The difference in dispersion compared to a flat array is dramatic. **Figure 1** shows the on-axis and 30° off-axis response of the CurveArray and that of an identically configured cabinet with a flat baffle.

DRIVER SELECTION

When searching for drivers to use in this project, check out the Web for specials and closeouts. The woofers used in the original were 4″ "NSB" closeouts from Parts Express, which I got for 49 cents apiece. Specs—fs 105Hz, Qts .76, vas .1ft^3, Z 8Ω—are fairly common in many 3 to 5″ drivers in the nominal \$5 to \$10 range. Be sure that you choose full-range drivers, with response to at least 8kHz.

The tweeters I used were 3/8″ dome Onkyo closeouts, Parts Express # 269-780, at an astounding 24 cents apiece (**Photo 2**). These tweeters have no mounting flanges, so they must be glued in place. There are many variations of this Onkyo tweeter available, some with coaxial mounting frames that can be discarded. The nominal crossover is 8kHz, so there's no need for larger than a ½″ dome. However, while a small frame size is also desirable, it's not an absolute necessity. A tweeter frame size up to 2″ in diameter is acceptable.

PHOTO 2: Frameless Onkyo tweeters.

TWEETER PLACEMENT

The CurveArray (CA) design spaces tweeters far wider than usual in a line array. The reason is shown in **Fig. 2**. On the left is a straight array; the shaded area represents the typical 15° high-frequency dispersion angle you may expect from a horizontal array. "X" is the distance from the array where the individual wave fronts integrate at a sample frequency. Minimizing dis-

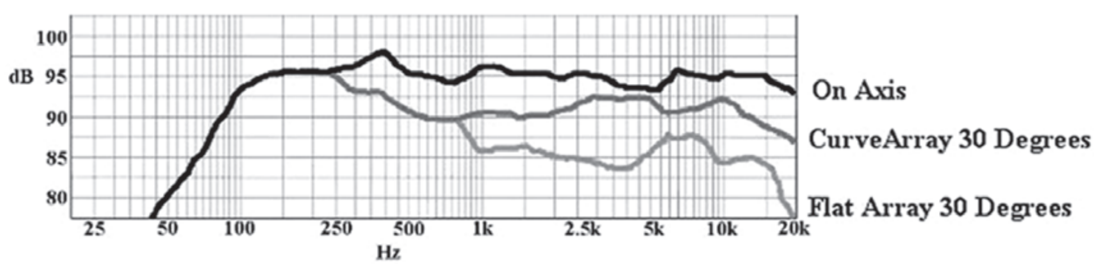

FIGURE 1: Response, on- and off-axis.

tance "X" is necessary to prevent lobing and comb filtering. The usual method of doing so is to space drivers with their acoustic centers no more than a wavelength apart.

The center cabinet is an array that incorporates an outward splay angle. The overall dispersion angle is increased, but so is distance "X," so both lobing and comb filtering are worse than in the straight-faced box. The cabinet on the right shows what happens when the drivers are splayed inward, cross firing. The dispersion angle is wider than with outwardly splayed drivers, yet the "X" distance is less than that of the flat array.

I tested the CA configuration with tweeter spacing half that of the final version, using two tweeters per woofer, and found no improvement below 12kHz. Use two tweeters per woofer if you wish, but considering the bandwidth of home theater (HT) center-channel program material, it's probably not worth the added expense. For that matter, you can make life much simpler by not using tweeters at all. **Figure 3** shows the response of the prototype without them, and the HF response is adequate for most center-channel program material.

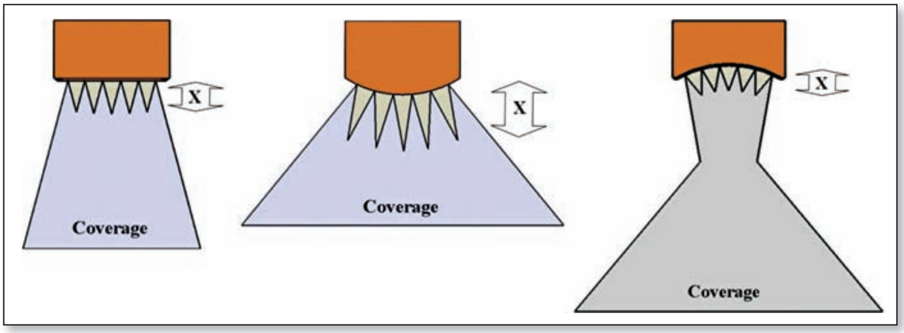

FIGURE 2: Dispersion versus wave-front integration.

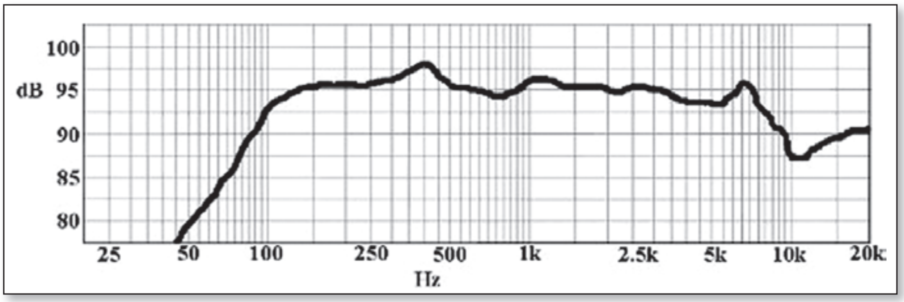

FIGURE 3: Response, woofers only.

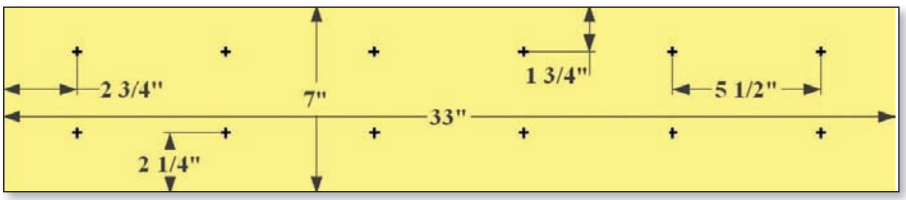

FIGURE 4: Initial baffle layout.

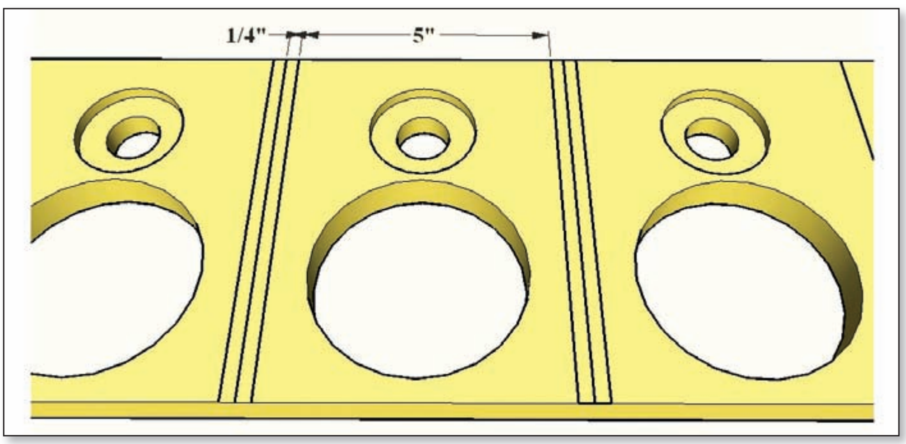

FIGURE 5: Secondary baffle layout.

CONSTRUCTION

You may build the cabinet from plywood or MDF, but solid wood is a viable option, and greatly simplifies the finishing process. I built the original from red oak, using 1 × 8 for the baffle and sides and edge-glued 1 × 6s for the top and bottom. Following these dimensions, you'll need a 6′ 1 × 8 and two 8′ 1 × 6s, while you can use plywood scraps that you may have in your shop or a 1 × 8 pine. The dimensions of the cabinet height and driver spacing shown are for the 4″ midbass drivers used in the original; you may alter them as desired for larger or smaller drivers.

Start construction by cutting out the baffle, 33″ long. The height of a standard 1 × 8 is nominally 7.25″. Trim that height based on your driver sizes; I made mine 7″. Draw two horizontal lines on it—one the radius of the tweeter plus ¼″ from one edge, the other the radius of the woofer plus ¼″ from the opposite edge. Draw intersects on these lines as shown in **Fig. 4**. These are the centers for the holes for the woofers and tweeters, respectively.

Drill holes for the drivers, sized for your drivers and their mounting schemes. For the tweeters I first drilled a 2″ diameter recess in the rear of the baffle, ¼″ deep, then a 1″ hole for the tweeter to fire through, chamfering the exit of that hole to a ½″ radius with a quarter-round router bit. For my woofers, I first routed a 4″ diameter circle 1/8″ deep on the baffle face, drilled a 3¾″ diameter hole and then chamfered the hole on the rear of the baffle, again with a ½″ radius quarter-round router bit.

After drilling the driver holes, draw cutting lines on the baffle per **Fig. 5.** With 4″ drivers, first draw a line equidistant to the driver centers, then draw lines ¼″ to either side of the first, for 5″ baffle segment widths. Cut the baffle segments—four with two 5° edges, two with one 5°, and one 25° edge, per **Fig. 6**. Be sure the angles are facing the correct direction, and that the two end segments

PHOTO 3: Polyurethane glue "fizz-out."

are mirror-imaged. Sand the baffle segments in preparation for finishing before assembling them, because it is difficult to do so after.

ASSEMBLY

You may construct a jig to hold the baffle segments in place for gluing, but there's an easier way. I used "PL Turbo," a three-minute setting polyurethane glue. Dampen one mating surface with water, apply a few thin beads of glue to the other mating surface, then hold the two parts together for three minutes. You can tell the glue has set when it foams out of the joint while curing (**Photo 3**). In this fashion you can glue the six baffle segments together in about 20 minutes, which is probably less time than it would take to fashion a jig. Be careful handling the baffle assembly, because the glue won't fully cure for a few hours. When the glue has fully cured, scrape off the excess with a utility knife and sand the assembly in preparation for applying a finish, taking care not to break any joints.

Measure the finished baffle (**Photo 4**) end to end. Initially cut the top and bottom that distance plus 1¾ to 2″ wide, and no less than 8″ deep. Cut the back the same width as the baffle. You may make the cab more than 8″ deep as desired; I made mine 15″ deep, to double as a shelf in my HT unit. You can go even deeper; for example, if you want to use the cab as a TV stand. You also can go wider, adding wings to either side of the baffle, should you wish to match the width of the cab to your TV.

Should you decide to make your top and bottom from solid wood, the easiest way to joint them is with biscuits, clamping them with clamps and cauls to hold them flat (**Photo 5**). Cut the sides the same depth as the top/bottom. The outboard woofer magnets will come close to the sides (**Photo 6**), and with some woofers the magnet could actually hit the sides, so trial-fit the baffle, sides, and a woofer before assembling the cabinet, routing recesses in the sides if required. Glue the baffle assembly to the cabinet top, with the woofer holes being closer to it.

If you have enough clamps, you may glue the baffle, sides, and back to each other and to the top simultaneously (**Photo 7**). Leave an equal amount hanging over the top at either end. If you have only a few clamps, glue the baffle in place first, adding the sides and back after the glue has set. "PL Turbo" is an option if you have no clamps, but standard urethane glue is a better choice if

PHOTO 4: Assembled baffle.

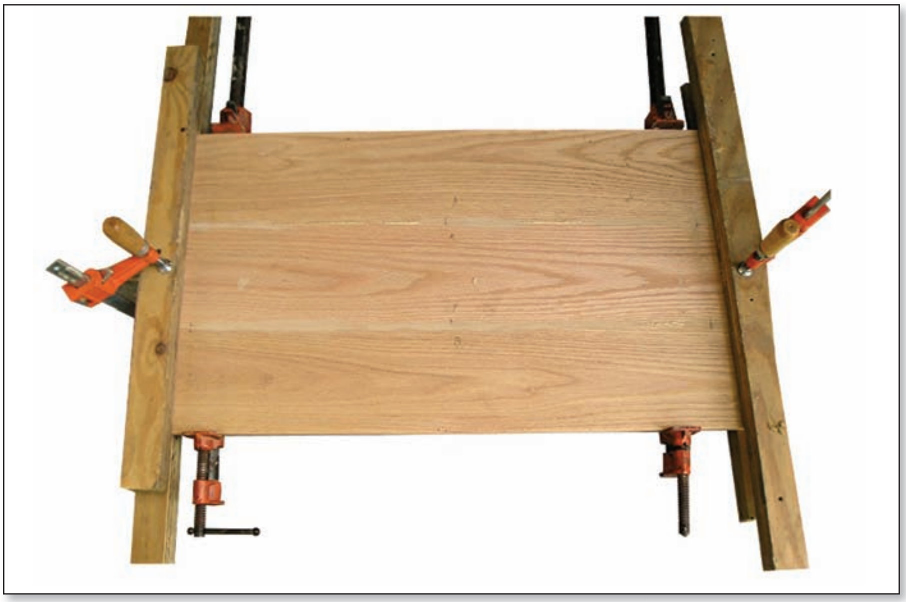

PHOTO 5: Top glue-up.

PHOTO 6: Drivers installed, interior view.

you do have clamps, allowing you plenty of working time.

If you make the cab 8″ deep, mount the back flush to the rear of the box, to maximize internal volume. With a deeper cab, you can inset the back from the rear of the cab by 1″ or so, to hide connectors and to allow "invisible" mounting of an external crossover.

After the glue has cured, screw the bottom in place temporarily. Use a router trim bit and/or sander to trim the top and bottom flush to the sides. Sand the exterior in preparation for finishing, smoothing the joints of the baffle ends and sides. Remove the bottom, fine-sand as required, and apply a finish. My preference is tung oil, which does a professional job with the least amount of effort.

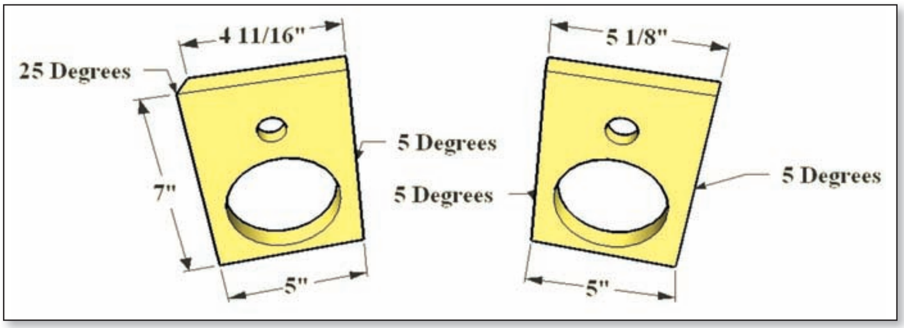

FIGURE 6: Individual baffle segments.


PHOTO 7: There's no such thing as owning too many clamps.

DRIVER INSTALLATION

Install the drivers. For an airtight seal, use a driver sealing gasket tape, such as Parts Express #260-540. With a frameless tweeter such as the Onkyo, glue it in place with a hot melt glue gun, first with a thin bead around the frame perimeter, then thoroughly filling the joint to ensure an airtight seal.

The drivers are wired to give impedance between 4 and 8Ω. I wired those of the prototype in banks of three, each bank parallel wired, the two banks in series. With my 8Ω woofers this gave a 5.3Ω impedance, while the 6Ω tweeters came out to 4Ω. I crossed them over second order at 6kHz, using component values from the calculator contained in my WinISD software (www.linearteam.org).

Install binding posts or a jack through the back, and function-test the cab before loosely filling it with poly batting. Screw the bottom in place, using the driver gasket tape on the joint for an airtight fit. Hook it up and be prepared to lower the center-channel volume on your HT receiver, because it's probably going to be considerably louder than your old center speaker, especially off-axis. *aX*

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Of Tweaks, Mods, and Evidence

This freeware is a quick and easy way to detect differences in audio signals. . . and thus the effect of components changes on your music system.

PHOTO 1: Nipper dispersing the bad vibrations.

Many audiophiles are interested in making changes to their audio equipment to improve performance. Modifications, or "mods," can range from changes that border on redesign, such as replacing a tweeter with a different type, to less fundamental changes, such as rewiring the innards of a power amplifier with silver leads.

Even less invasive changes to an audio system—in which no screws or covers get removed during installation—are usually called "tweaks." These could be old standbys such as contact treatments put on connectors, or newer, more radical changes involving special woods that are said to make the system sound better just by being in the room. Tweaking has become almost a separate hobby of its own these days, with recommendations of countless "free" tweaks easily found on the Internet and an entire industry of not-so-free (and sometimes downright expensive) tweaks for more adventurous audiophiles. Tweaks usually seem to be controversial, often without generally accepted explanations—and sometimes no explanations at all—about how they could work.

COLLECTING EVIDENCE

I am by profession an analog/RF electrical engineer at a large corporation and also develop audio test equipment and software with Liberty Instruments. You might assume, then, that I have an extreme "objectivist" point of view, discounting anything that is not predicted by known theory. But that's not quite true.

Like many, I trust engineering concepts. But I won't declare an idea to be bogus just because it lacks undeniable proof. It's always at least remotely possible that even seemingly off-the-wall ideas could work. Who knows, maybe there are non-obvious or yet-to-be-explained mechanisms operating?

I'm even using some tweaks right now in my own system. I have a special mat placed under my power amp, and a painted *statue* sitting on its chassis, and I have not the least doubt that they greatly improve the listenability of my system.

Wait, don't go away shaking your head! Let me explain why they're there. The amp is an older, rebuilt, modified Hafler DH200. I keep it sitting on the floor near a corner, driving my main speakers.

I recently noticed a weak but bothersome hum coming from the amplifier. The noise wasn't induced into the speakers, but came from the amplifier chassis itself. Apparently, the laminated core of the power transformer was vibrating with the AC power waveform, and the vibration was getting into the chassis and from there to the floor, projecting a low humming sound into the room. I tried tightening the bolts that compressed the laminations, but it didn't help. I also came up empty after a brief search for an inexpensive replacement transformer.

So, I tried putting some folded foam padding under the amplifier to isolate it from the floor. That helped, but the chassis still was radiating hum directly through the air into the room. I noticed that if I pressed down a little on the chassis, the hum level would drop. And if I pressed down *hard* on the chassis, the hum was essentially gone.

Conveniently nearby on the fireplace hearth sat a heavy statue of Nipper (RCA's mascot, **Photo 1**) that was given to me by a neighbor. Well, soon Nipper was standing guard on top of the power amplifier—and the hum problem is now gone. It's not an audiophile delusion after all, it's just that those tweaks help in ways that aren't immediately obvious.

But the possibility of an unknown operational mechanism exists for any wild concept you might imagine, and there isn't infinite time or interest to check out very many of them. There are plenty of real, known issues to pursue in audio that have a tremendously higher likelihood—at least for technically educated people—of bearing fruit. Before I become interested in investing time or money and actually trying some tweak or seemingly unlikely mod, I like to see some *reasonable*, competent, nonsense-free theoretical explanation about how it *might* work; or, failing that, some *evidence* that it actually does. Testimonials really don't count for much and are very easy to come by for promoting or discounting just about anything.

With some tweaks demonstrated to me by honest audiophiles, I know I couldn't hear any effect and wasn't convinced that the demonstrator was really hearing anything that the tweak was doing, either.

Audiophiles, after all, are all *about* illusions—they build systems in the hope of convincingly presenting live musicians right there in the room, when any sane person can plainly see that there aren't any! We regularly strive to be deceived by our systems and to give in to our imaginations.

Actual evidence can be hard to come by, though. ABX testing, which aims to offer it, is time consuming to do, and it is complicated to maintain controls that might guarantee a verifiable result. And ABX also seems like a tedious operation that would not be much fun, at least not compared to relaxing and just listening to music. And the results of ABX are too often challenged on the basis that the ABX switchbox itself might be masking differences that could really be there. That argument seems to me to be more than a little like a dodge, but it *could* be possible, I guess.

And, should no evidence of an audible difference be uncovered in an ABX test, the strongest statement that might be made is that no audible differences were revealed while using that given set of associated equipment, by that particular set of listeners, with those particular recordings, at that particular time. At best, it might be suggested, but never be proven, that something is completely inaudible. But the opposite *could* be proven were it to be the case—that something is *audible* under some circumstances. That would be a useful result and might lead to something interesting.

I admit that I'm suspicious of companies and audiophiles who say that improvements made by using (as examples) particular interconnects or power cords are absolutely audibly *obvious*. Yet none of these persons have ever demonstrated at an AES convention, for example, the ability of even one listener to differentiate, by hearing alone, when different components are changed. It seems this would be an easy demo to set up for such obvious improvements, and could result in fame for that listener and a chance to shake up the foundations of audio engineering, while providing excellent press coverage and powerful material for advertisement copy for manufacturers. Cable companies might choose to consider doing this if they really believe in their products.

With the lack of interest in conducting time-consuming verifiable ABX tests, or for making clear and incontestable public demonstrations of differences being heard, it would be good to have some other ways to get evidence that tweaks or mods might be worth checking out.

SETTLING DIFFERENCES

Particular skepticism toward some of the tweaks or mods results when they lack any apparent way that they could even make *any* difference, much less one detectable by ear. If a tweak or mod changes nothing, then I at least would conclude that there's no difference to hear. If something in an audio signal does change, then there's at least a chance. So a good first step toward giving evidence for a tweak or mod might be to show that it does *something* to an electrical signal in equipment or to sound in the air.

You need a way to show when audio signals have changed; that is, when one audio signal made one way (with-

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out tweak or mod) is different from one made another way (with tweak or mod). It would be better still if you could listen only to the part of the signal that was different, removing the unchanged part of the signal, so you could tell whether those changes all by themselves might register to your ears.

Flash back to your school days when you recall that "difference" in math has a particular meaning: subtracting the subtrahend from the minuend resulted in the "difference." When two numbers or expressions are different, then their difference won't be zero, while a "zero difference" would mean that there wasn't any difference between the expressions before subtraction; i.e., that both expressions were the same. When you talk about differences in audio signals, the concept is very similar.

You can also subtract the signals representing sound in a sound system. In analog electronics, subtraction is done by inverting the polarity of the second signal and then mixing (summing) the two together. If the signals were exactly the same, they would cancel each other out and leave a silent signal, a difference of zero (**Fig. 1**). If they didn't leave anything (except maybe the noise which intrudes into any analog process), then the signals weren't different. If they leave something other than noise, then the signals must have been different in some way.

In the digital domain, an audio signal exists in a series of values of the voltages in the signal at successive time intervals (samples). If you had two digitally recorded signals, you could see whether they were different by essentially the same method, by subtracting the values of matched samples at each point through the series. If they were exactly the same, you'd get a series of zeros, a recording of silence. If the recording equipment had some distorton, you'd still get just silence because the same distortion acting on the same signals would give the same result. If they differed only by noise that was produced when the signals were produced (such as from an analog to digital converter), then you would get that noise in the "difference signal," because noise is always changing.

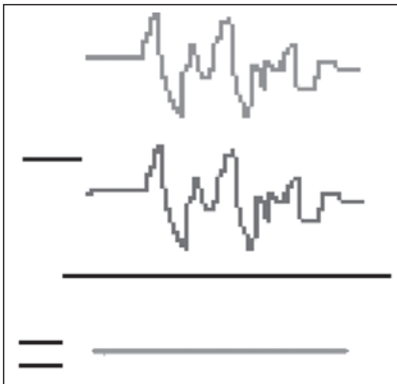


FIGURE 1: Subtracting signals.

You can use this process to see whether two signals are different, testing whether use of a tweak or mod has changed anything. If you obtain from that a result that seems significant, you will have some *evidence*. You won't have proof, though. The test could have been done erroneously, for instance.

If that possibility is dispelled, then it is still possible that the change that was uncovered may not be large enough to be heard. You can probably make an estimate (subjective) of that by listening to the level of the "difference signal" in comparison to the level of the original signal. That way you could rather easily tell for yourself whether any difference that may be there could be significant for you. Though again, you won't ever prove that someone with perhaps better hearing abilities wouldn't hear something he/she believes is significant there.

Even though the signal subtraction process by itself won't give you definite proof either way, discovery that there is some difference might be enough to warrant looking further into the tweak or mod. Lack of a significant difference would—to me at least—imply that the tweak or mod isn't worth putting time into.

AUDIO DIFFMAKER

There is a complication in using signal subtraction to indicate possible changes when a tweak, mod, or accessory has been applied. The two recordings that will be subtracted must usually be made at different times if the system is to have been changed only by the tweak, in a before and after comparison. Some way is needed to shift time in the recordings so that equivalent time instants will be properly compared. That's where computer processing can help.

"Audio DiffMaker" is a freeware software program I have written to provide this ability. I made it to help obtain and share evidence about whether a given tweak or mod has actually made a difference to an audio signal passing through the electrical parts of an audio system on the way to the loudspeaker terminals. DiffMaker won't likely be able to deal with what happens to the signal after the speaker has turned it into sound, because even minuscule movements of a microphone or changes of reflecting objects (or even air movements) in the room are often enough to result in an audible difference track. Nor will DiffMaker be able to address tweaks or mods that might affect a listener directly rather than through the music signal.

But DiffMaker should be able to help demonstrate differences made by cables, component parts changes, CD treatments, and many others, if they exist. DiffMaker is intended to be convenient and reasonably fast to use, and even people at different times or places can participate by listening to shared difference tracks on their own, evaluating for themselves with little effort. While it can't provide proof of audibility, like ABX conceivably might, DiffMaker could give strong evidence that a tweak or mod may at least be doing *something,* and within a short time. Or it could make a pretty convincing demonstration, which you could provide to others for them to witness as well, that perhaps nothing has really changed.

So, how does this test work? You need a computer and a high-quality soundcard with line-level inputs. The basic process is pretty easy to understand.

First, get a copy of the Audio DiffMaker software, which you can download for free from www.libinst.com. It runs in Windows 2000, XP (most recommended), or Vista. Install the application, then find "Liberty Instruments" in your start menu so you know where to start DiffMaker.

Try to use a stereo system in which you expect or it's been reported that the tweak, mod, or cable makes a significant difference to the sound. Try to use the same program material or musical cut that seems to exhibit the difference well.

Set up the computer so that its soundcard can record electrical signals at a point along the hi-fi system chain where a tweak or mod is expected to make a change. For example, if you are testing a CD treatment, you can make your record from the output of the CD player. If testing an interconnect cable, test perhaps at the output end of the changed cables or further downstream.

If the point operates at a line level of a volt or two, you may be able to just use some y-adaptors to pick off the signal to supply it to your soundcard's line inputs. If at higher level (like the output of a power amplifier), you should put some resistors (20kΩ or more) between the "hot" center conductor wires feeding the soundcard line inputs and the large signal voltage. Don't connect your power amplifier's big output signal directly to your soundcard's low voltage input unless you enjoy replacing burned-out soundcards or computers!

Next, start the DiffMaker software. The first form that appears on the screen, "Settings" (**Fig. 2**), lets you select the recording and playback soundcard(s) you want to use, should the computer have more than one. You can also select an optional "Source" soundcard, to be described later.

When using DiffMaker, you will play the same section of music at least twice, with and without the tweak or mod. You can source the music signal either by playing a CD, SACD, and so on, from your deck or from having the DiffMaker program play a musical track through the line-output of your "Source" soundcard (**Fig. 3**). DiffMaker has several utility

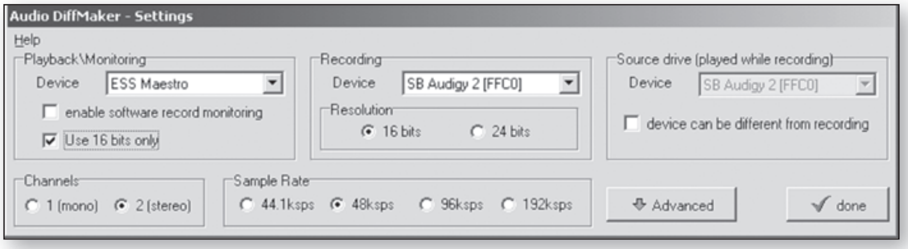


FIGURE 2: The settings form.

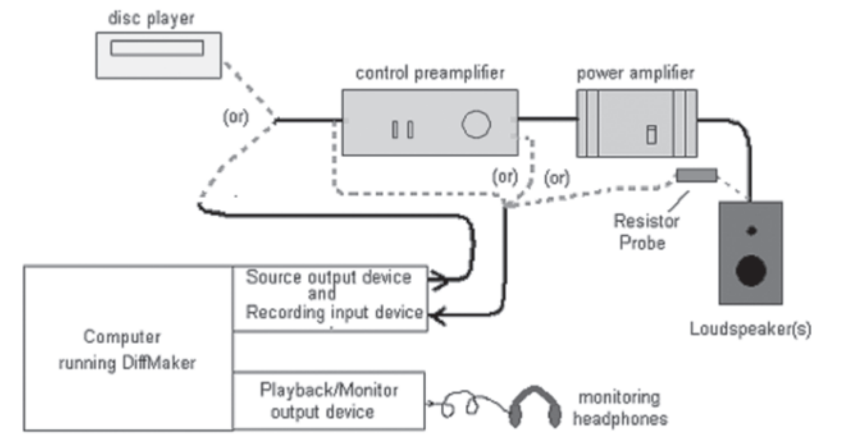
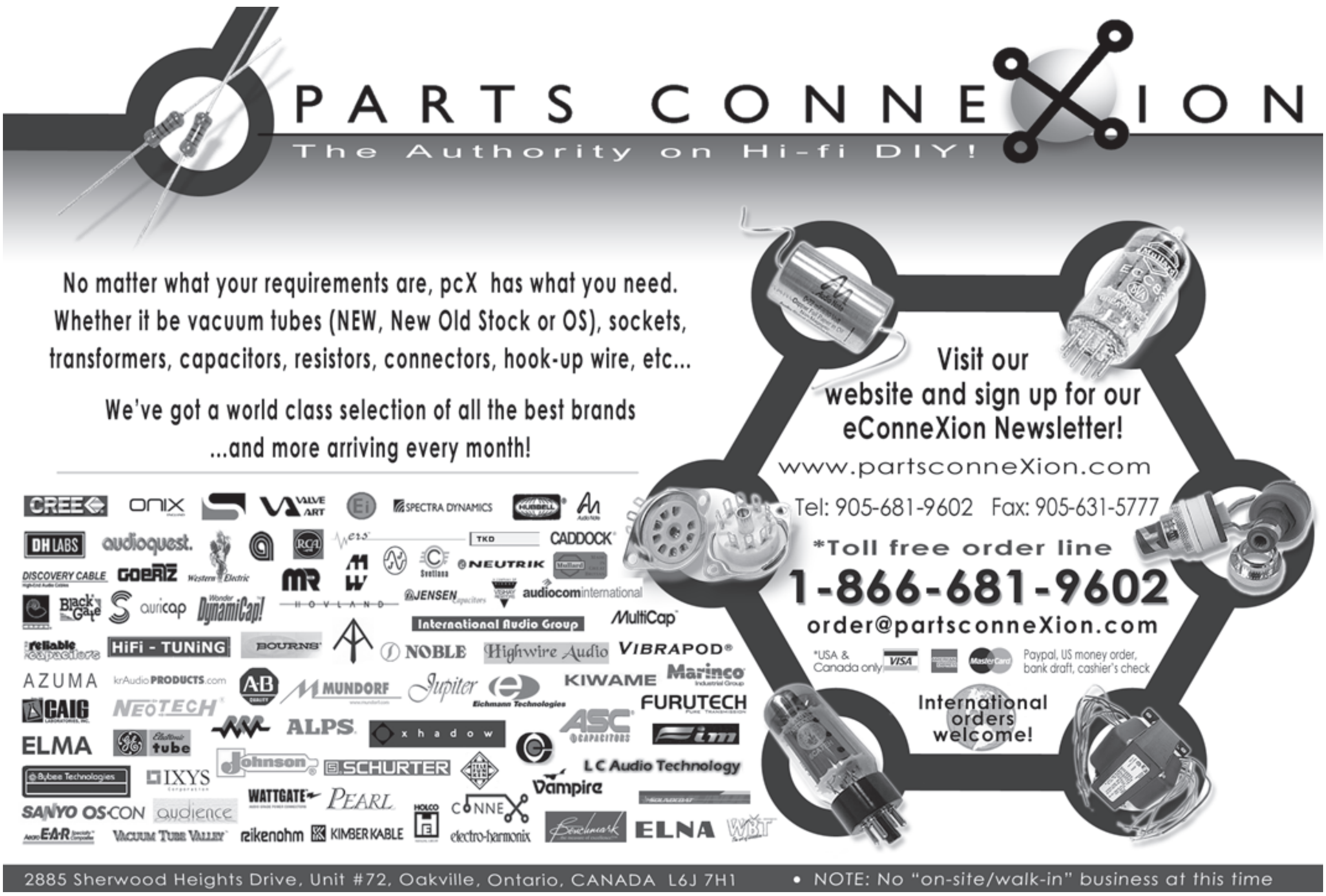


FIGURE 3: Setup for making recordings.



programs built-in, and, if desired, you can use one of them, the "Ripper/Trimmer" (**Fig. 4**), to rip part of an audio track off a CD into a WAV file for use as source test music.

The program's "Settings" form has a number of other controls, but usually you can leave them in the default settings—at least at first. When the right soundcards have been selected, click "Done" and you'll see the Main Form (**Fig. 5**), which is where most of the activity happens. It has quite a few menus and buttons, but isn't complicated to understand. Most of the menus at the top are used to get to other forms. Most of the buttons let you Record, Load, or Play audio tracks, including the three main tracks:

- The "Reference Track": one of the tracks (recordings) that you will be comparing, usually the "before" track.
- The "Compared Track": this will be aligned with and then subtracted from the "Reference Track" to leave
- the "Difference Track."

There is also the optional "Source" track, which you can use to play out to the stereo system, as I mentioned.

To make the first recording (probably the one before a tweak has been applied), click on the "Record" button in the box marked "Reference Track." This will make a "Recorder" form appear (**Fig. 6**). Using this Recorder is easy, and the help menu at the top of the form will show you detailed instructions. The Reference Track (and other tracks) that you record should be relatively short, under 30 seconds usually. That's so the computer won't take too long in processing them later and also to minimize effects of any "gain drift" or "time drift" (which are described in the help file).

You need to record a Compared Track that is recorded when substantially the same section of music is playing. The starting positions of the two tracks, with respect to the test music, should be within about one second and should end within the same second or two.

Then you will subtract the recorded signals, one from the other. Do this by clicking the "Extract" button and waiting for the process to complete. You may expect that subtraction should be a fast process, but there are some complications that actually make it rather time consuming. The Reference and Compared tracks need to be in the same relative time position together, in lockstep, in order for subtraction to be able to possibly leave a nearly silent result. The signals from equivalent time instants should be subtracted, because any small time shifts or mismatches will degrade results (i.e., show a difference that might not be there).

Taking two very similar recordings and aligning them to within one sound sample (tens of microseconds) isn't difficult, using an FFT-based process called "correlation." But being within one sample isn't nearly close enough. Time adjustment between recordings needs to be matched much closer than to within one sample to cancel signals to below audibility. DiffMaker uses a frequency domain process to bring time adjustments between mostly identical, but shifted, signals even to within nanoseconds. The adjustment is done on the Compared Track, which then is shifted to best match the time position of the Reference Track.

Because very small and simple amplitude ("volume," "gain," or "level") differences are usually of no interest when looking for tweak differences, DiffMaker also optimizes the amplitude match between the two recordings. It does this by making the Compared Track louder or weaker as needed, until it finds the best match to the Reference Track.

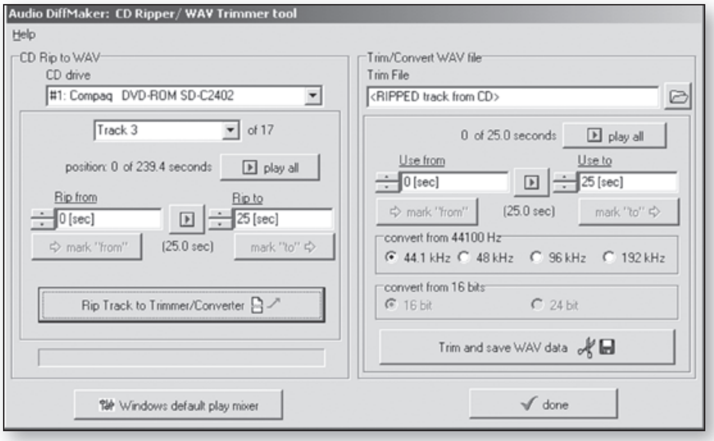


FIGURE 4: Ripper/trimmer form.

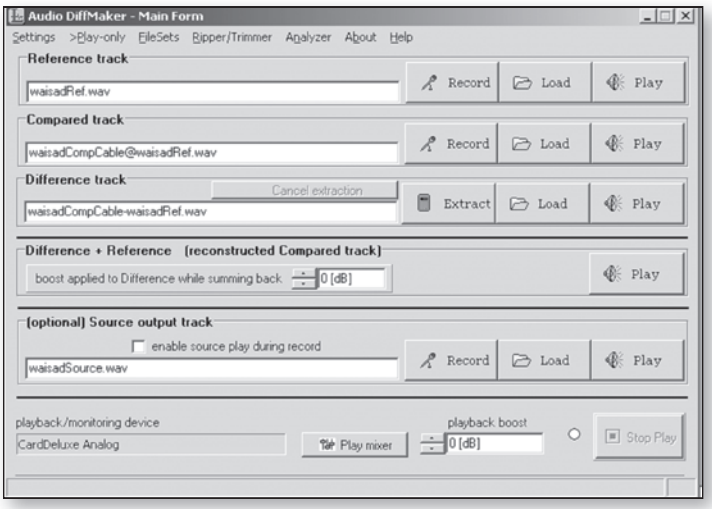


FIGURE 5: The main form.

HEAR THE DIFFERENCE

When the adjustment and subtraction are done, the Compared Track will be replaced by its time-shifted version, and the Difference Track will be presented.

An important ability of DiffMaker is to play several tracks in matching time, allowing the operator to shift between hearing one of the original signals and the difference sound that results from subtracting. This allows a fair comparison of the level of any revealed difference (or of noise) to the level of the original music track. When you are playing any of the tracks, you can instantly switch to listening to a different track (Reference, Compared, or Difference) by clicking its play button.

The multiple playback ability is also important because it allows results of a comparison to be experienced by others. You can use DiffMaker to allow others to listen to results of tests you make. To simplify the program operation for them, change the Main Form into a simple-to-operate "Play Only" form (**Fig. 7**) using the menu option of that name.

After you have made a set of recordings and a difference track, you can use DiffMaker to bundle them, along with written comments, into a compacted (but lossless) "dyf" file that you can e-mail or post for others who have DiffMaker installed on their machines. Select "File Sets" in the menu to do this.

When you are testing some component parts, there will be small, but easily explained, differences made to frequency responses. For instance, if the system drives a cable being tested from a relatively high impedance, differences in cable capacitance will slightly change the high-frequency response and phase shift. Even very small and inaudible response differences can greatly degrade signal subtraction results.

DiffMaker is exceedingly sensitive to outrageously tiny changes, which is generally good, but sometimes frustrating to deal with. Any error in setup, or equipment drift, will result in an erroneous difference being detected. Everything

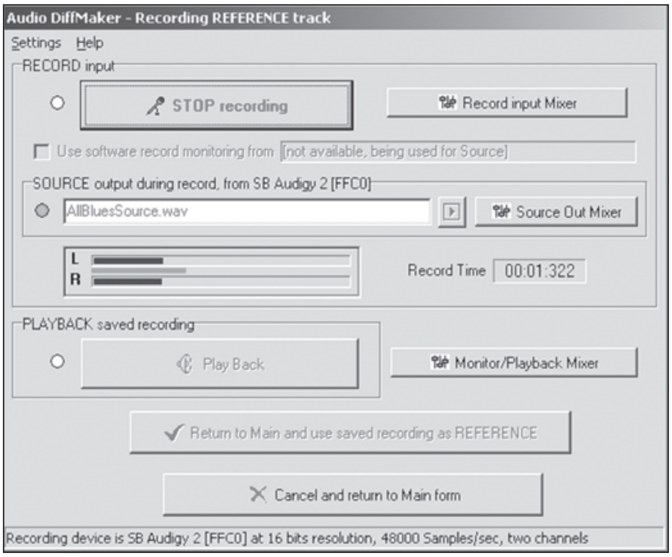


FIGURE 6: The recorder form.

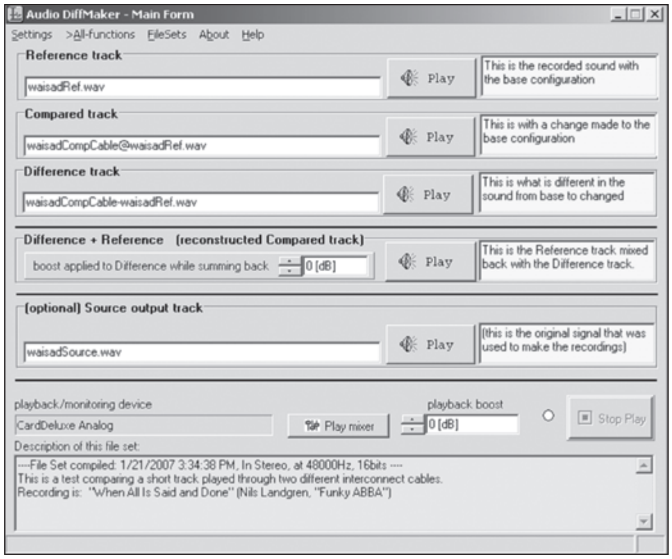


FIGURE 7: Play-only presentation of main form.

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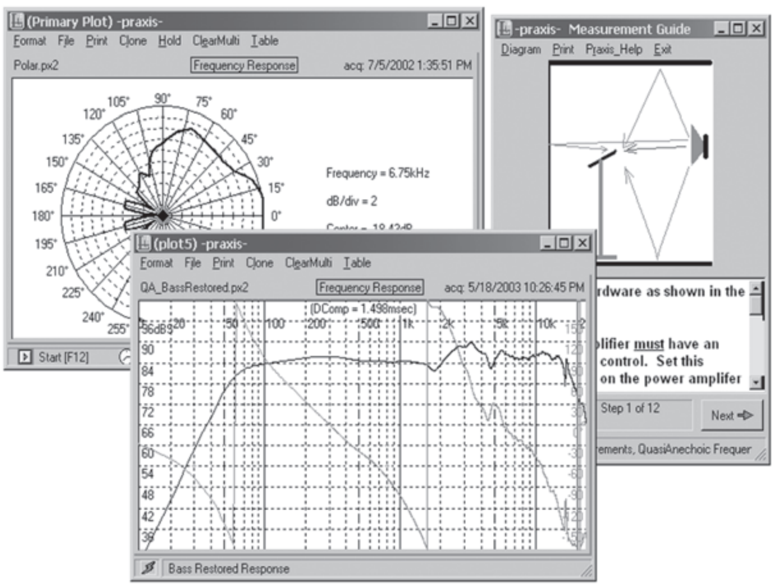

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must go correctly to approach a full cancellation.

If there are any already known differences, there will be a very noticeable signal level in the difference track. If you want to find out whether there are any differences other than in frequency response, you will need to first compensate for or avoid these differences in the responses. In some cases, such as with rolloff from cable capacitance, that may be relatively easy to do: add a fixed capacitor to the cable that has lower capacitance, or add resistive loading on the one with higher capacitance.

First you need to be able to identify a response difference when you are adjusting it. So the DiffMaker application includes an "analyzer" form (**Fig. 8**), a measurement tool capable of resolving very small frequency response differences. The analyzer outputs a special noise-like sequence that sounds like pink noise, but isn't quite. The advantage of using this sequence rather than real pink noise is that it is keyed to the analyzer's operation so that no settling time is needed—response differences can be found quickly and compensated in near real time. The operation of the analyzer is described in the help file in the "analyzer" section, and an example of its use is given in a detailed Usage Example, also in the program's help file.

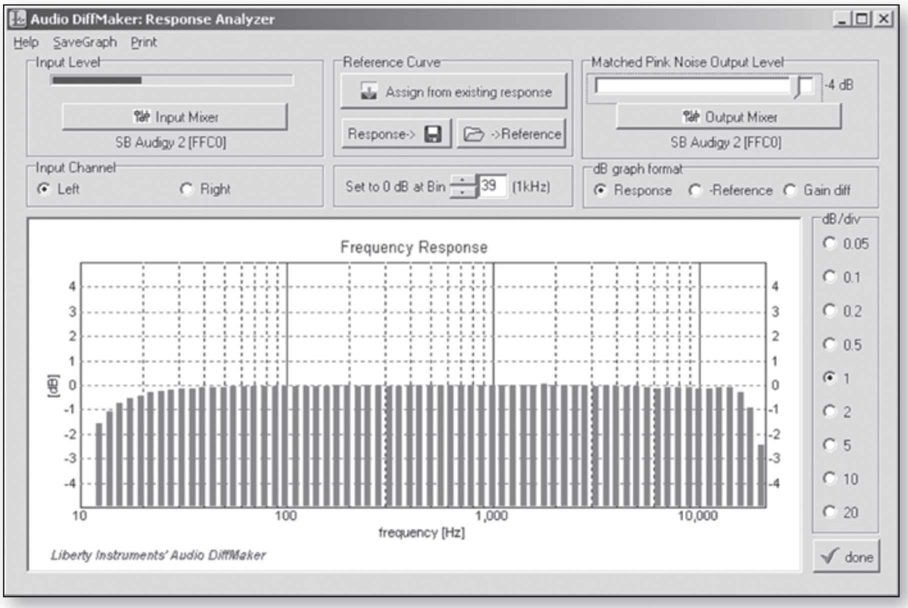


FIGURE 8: The analyzer form.

DIFFERENCES UN-MASKED

When you compare the before and after sound of a system by ear to evaluate a tweak or mod, you face a tricky task. One problem is the time lag that it usually takes to get from one state to the other. The sound character must be remembered during that lag and the conditions of the system, the room, or even the listener can change in that short period.

An even bigger complication is audible masking. The presence of some louder or more prominent sounds can make it difficult for an ear to detect effects of related but subtle sound character that effectively becomes covered up.

There is also the issue of what is sometimes called "resolving power." In a high-end system, an audiophile listens for what evokes feelings of real music. In a clean, good-performing system, changes to subtle details or imaging qualities might be noticed. But if everything were to be routed, instead, through a clock radio, those small nuances wouldn't register afterwards—not because the signal nuances aren't there anymore, but because they just don't matter to the ear anymore if everything already sounds so artificial and distracting.

Subtle effects may be undetectable in a system lacking sufficient resolving power, and one grungy component somewhere in the chain might make a system unsuitable for critical listening. Recall that this is one objection that has been made about ABX testing: that the switchbox might lack enough resolution to allow differences to be heard.

Masking and lack of audible resolution are problems during listening comparisons because people are listening for differences between separate situations. But a subtractive test such as the one with DiffMaker doesn't detect *audible* differences, it looks for something more fundamental and simple. It isolates *absolute* differences into one track. Anything different—not just what you might recognize while listening—is brought to light. Masking isn't relevant or even meaningful, because it just doesn't apply to subtraction.

A "masked" difference is still different, when signals are subtracted. Nor does resolution of each component in the chain matter in non-absurd cases, because subtraction isn't sensitive to or bothered by how much a presentation resembles music. When a difference track leaves nothing, you can be sure that the tracks being subtracted were the same, not that distortions or louder sounds somehow fooled basic arithmetic. In the first evaluation, the difference track is not analyzed for sound quality, but for *whether it has anything in it.*

Unless a component could somehow strip off all traces of some "change" from another signal—to make an output that is the same as it would had the "change" part never been there at all—the component's output will still have effects of that change. It could happen that some component along the way adds enough noise to obscure details that might otherwise be heard. In a differencing test, though, noise *won't* subtract out, because noise at different times is not the same as at other times. In fact, it will even appear at a slightly higher level in a difference track, and a listener can readily hear whether there is too much noise to trust the result. But added noise can't cause an erroneously silent difference track.

Insufficient bandwidth in a component, such as in a soundcard's recorder, might hide the presence or absence of a signal that changed previously in the chain, if that signal were completely beyond the soundcard's frequency range. If you suspect this odd situation, operate DiffMaker with sample rates up to 192kHz (and at 24 bits), as high as commercial audio recordings are made. Extremely high-quality soundcards (better than were being used for commercial recording not long ago) have recently become available at very affordable prices.

Even if a component or soundcard were to have a dead zone near zero volts, able to block small signals, so that only voltages above a certain level would get to the output, that still wouldn't make the component blind to changes in small musical details (**Fig. 9**). Those small signals would still have an effect, when they are present with (and therefore superimposed on) normal level voltage swings that *were* large enough to appear at the output. They'd sound terrible, but would still exist, unless, of course, no signals at *all* were to get through. But it doesn't seem unreasonable to require that you not use dead soundcards to use DiffMaker.

TRUST YOUR EARS

The result of a DiffMaker test isn't a number or graph (though one could be derived from it), but is instead a "WAV" file or audio track. It doesn't so much say, "Yes," "No," "good," or "bad," but rather, "Here's how loud the difference is and what it sounds like." That, I hope, may minimize arguments about the significance of results.

It also somewhat addresses the common assertion that evaluations of audio product effectiveness should be done using ears rather than test equipment. Evaluations in DiffMaker *are* made by ear, though with computer assistance. Any interested persons can personally make their own by-ear evaluation of whether the tweak or mod might be effective if they are sent the "dyf" file and have loaded the free program.

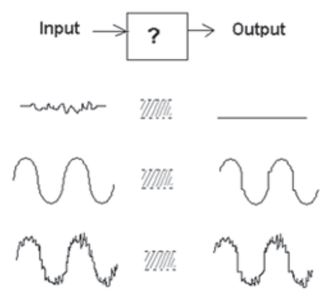


FIGURE 9: A device with a dead zone: small signal details ride on the large signals. Top: a small signal alone. Middle: a large signal. Bottom: mixed large and small signals.

I hope to see evidence for something new in audio research and understanding and that DiffMaker may perhaps help break such new ground. Or, if not those benefits, that it will at least provide you some fun. *aX*

REFERENCES

There have been several similar analog type difference type tests that have been proposed:

1. Baxandall, Peter J., "Audible Amplifier Distortion is Not a Mystery," *Wireless World*, November 1977.

2. Hafler, David, "A Listening Test for Amplifier Distortion," *HiFi News & RR*, November 1986.

There are also discussions of these and also a suggested digital difference test in:

3. Dunn, Chris and Hawksford, Malcolm, "Towards a Definitive Analysis of Audio System Errors," presented at the 91st AES Convention, October 1991, Preprint # 3137.

In Praise of Wall Warts

These often-overlooked, but vital, little boxes are capable of more than you think in your electronic projects.

You know them—the small, squarish plastic boxes that plug into a wall socket or power strip and send power to a multitude of electronic products. They're often objects of contempt, derisively branded "wall warts"; their presence in a design is often considered a herald of cheapjack design and construction, even though they've been used in such acclaimed audio products as FMR Audio's "Real Nice Compressor."

Why should serious audio constructors bother with wall warts? To answer that question, you first must ask why manufacturers use them in the first place. The answer, in one word, is insurance.

Take a look at the electrical and electronic equipment in your home. If you live in the US, it's almost sure to have a printed, engraved, or stuck-on label reading "UL Approved." "UL" stands for Underwriters' Laboratories, an organization set up by the insurance industry to test and certify electrical equipment for safety. (If you live outside the US, you'll find similar logos for parallel certification programs, such as the European "CE.")

Equipment with a UL label is presumed safe to plug into a wall socket. Conversely, gear without the label is sometimes presumed by insurance companies not to be safe. In extreme cases, if a house catches fire and a piece of non-UL equipment is found in the ruins, insurance companies may be reluctant to pay the claims.

So why don't manufacturers have everything approved by UL? Money. Getting a piece of equipment tested and certified by UL can cost thousands of dollars, especially if they demand design changes. For a small company, that's prohibitive.

Many choose to punt. UL's chief concern is what actually plugs into the wall and connects to the AC mains, which in electronic equipment is the power supply. Instead of designing their own supplies and submitting them for UL approval, manufacturers will buy prebuilt supplies from a company that has already gotten the magic sticker. (How can that company afford it? Well, they're not building just a few thousand a year. Instead, they're building hundreds of thousands, for a multitude of products, and selling them to all sorts of companies, from Joe Blow's ElectroWidgets to Radio Shack and Sony.)

In other words, they're using a wall wart, or its snake-that-swallowed-a-brick cousin, a line lump.

Does this matter to you? If you're building DIY gear for your home stereo it might or might not. Different insurance companies have varying policies (pardon the expression) about UL approval. But if you're running a home studio, for example, where outside clients come in to perform and pay for your recording services, it's an issue. If you use DIY equipment in a home-studio context, you should probably get to know and love the wall wart, and learn to use it effectively.

AC/DC

In this article I mostly talk about AC wall warts (basically transformers in a box) rather than the more common DC devices, which are problematic for many applications.

In the first place, most solid-state circuits these days use bipolar supplies. To power them from a two-terminal wall wart requires the designer to produce some sort of artificial ground halfway between the wart's + and - voltages, and to keep that ground stable. DC wall warts usually come in a maximum voltage of $24V_{DC}$, which means the most you can get from them in a split supply is $\pm 12V_{DC}$, and that's without benefit of regulation. This could limit headroom.

The warts could also power DC-DC converters, but these incorporate switching circuits that can generate all sorts of high-frequency garbage, not something you want floating around inside your chassis. Some DC wall warts, in fact, are switching circuits themselves.

Besides, most DC wall warts are mediocre; they lack the good-design touches you've come to expect in high-quality gear. Instead of large, high-quality, well-bypassed filter capacitors, they use small, cheap caps with no bypasses. They don't use fast-switching, low-noise diodes, nor do they incorporate snubber circuits to minimize diode switching noise. Typically they incorporate no spike-filtering protection.

AC wall transformers offer surprising flexibility, and can be made to sit up and do tricks, including multiple supplies from a single wart. You can even derive plate voltages for vacuum tube circuits from a low-voltage AC wall wart. Because the user chooses rectifiers and filter components, they can be as good as those in a more conventional circuit.

BASICS

First, consider connections. The near-universal connector for a wall wart's umbilical cord is a round plug, with an inner diameter of 2.1mm or 2.5mm and an outer diameter of 5.5mm. The cord connector is normally female, and the chassis connector is normally male. (This is the opposite of the convention for signal connectors such as XLRs; the idea of making the cord connector female is that it's safer to carry live voltage back inside a hole than on a prong. The same principle applies to AC wall sockets.)

Where can you find chassis jacks? (When I talk about specific parts, I'll limit myself to devices available in the US, because those are what I know about. Similar parts are made and sold all over the world.) Switchcraft sells the 712A and 722A connectors (for 2.5 × 5.5mm and 2.1 × 5.5mm plugs, respectively). Both have metal sleeves, so they must be insulated from the chassis by 5/16″ I.D. shoulder and flat washers. Mouser, Jameco, and Digi-Key all stock these jacks for about $2.40 apiece.

How about the warts themselves? Digi-Key has a few, but Jameco Electronics has the mother lode. Jameco's catalog 273, pp. 197-8, lists 37 different "AC-to-AC Wall Adapters," all but two carrying UL approval. Output voltages range from $9V_{AC}$ to $24V_{AC}$, with current ratings up to 1.9A. Prices—under $15—are very reasonable.

BALANCED RAILS

For a simple example, consider a $\pm 15V_{DC}$ regulated supply for use with op amps or similar solid-state circuits. That's a 30V spread, even ignoring the extra needed for regulators. Will a wall wart provide enough voltage?

One possibility which I don't like is to use a voltage doubler (**Fig. 1**). They're notorious for poor load regulation, and because each leg of the rectifier is basically a half-wave design, ripple is high and equal to the line frequency, rather than 2× the line frequency as found in a full-wave rectifier. This means a given filter capacitor is only half as effective in removing ripple.

Instead, I'll do something more practical for a DIY designer than a commercial manufacturer: I'll use two identical

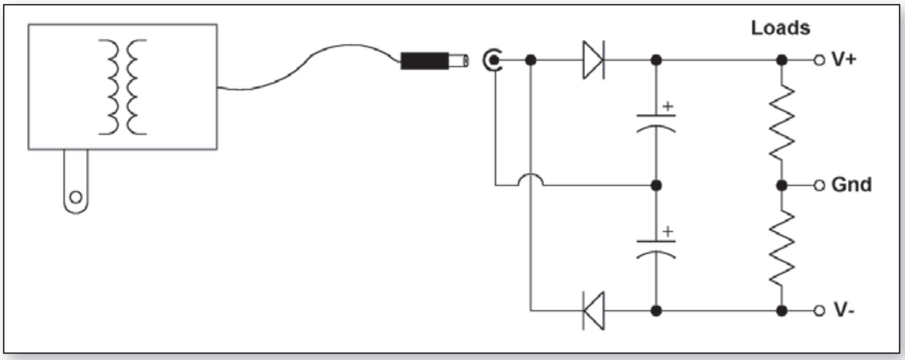


FIGURE 1: Wall wart used with voltage doubler. In this and subsequent drawings, resistors hanging on the DC outputs of the supplies represent their loads. Note that the ground is produced at the junction of the two legs of the load, not within the supply itself.

wall warts.

Without going through the math, once I allow for diode, filter, and regulator drops, plus provide some margin for low line voltage, I find that I want $18V_{AC}$ on each leg of the supply. (You can just squeeze by with $16V_{AC}$, but at the cost of poorer filtering and less undervoltage margin.) Jameco's catalog lists model #121216 from ReliaPro, giving $18V_{AC}$ at 1000mA. They're $9.85 each, or $19.70 for a bipolar supply.

How to hook them up? One possibility is shown in **Fig. 2**: the two transformers' secondaries are wired in series to form a center-tapped $36V_{AC}$ transformer. With wall warts, this means connecting the tip terminal of one chassis jack to the ring terminal of the other, and connecting that junction to the main ground point of the power supply. That works fine as long as the two wall warts operate with the same polarity—in other words, the voltages on tip connections go up and down together, rather than in opposite directions.

Can you rely on that? These are mass-market items, normally used singly, and there's no particular reason for a manufacturer to care about polarity. Besides, not every wart comes from the same manufacturer.

Because of this uncertainty, I'd rather use the two warts as shown in **Fig. 3**, connected to separate diode bridges. In this circuit, wart polarity doesn't matter, and the cost of four extra diodes seems pretty modest for the added peace of mind.

(I'm planning another article on power supply design that will describe a triple supply board designed to allow either center-tapped or separate windings. Stay tuned!)

One drawback to wall warts is their lack of built-in protection against major overvoltages and spikes in the line AC. A .01µF 1kV ceramic disc capacitor in the power strip will help absorb RF, but it's possible to also add surge absorption to the output of a transformer, including a wall wart. I assume ±10% variation in line voltage, so adding a metal-oxide varistor (MOV) or similar clamping device with an AC rating 20% over the nominal voltage is reasonable. (Before you choose the MOV, though, check the actual voltage of the wall wart under full load from the supply and audio circuits; if you're drawing only a fraction of the transformer's rated current, its output voltage may already be higher than nominal. Pick an MOV 20% higher than the wall wart's real output under load when the wall socket is at nominal voltage.)

You can install Hagerman-style snubbers on the outputs of wall warts, just as you can on the secondaries of conventional transformers,[1] and you can place fuses in the leads from the wall warts' chassis jacks to the supply, rated at the transformers' current rating.

There's another way to make wall warts produce ± supplies; I'll work my way around to that.

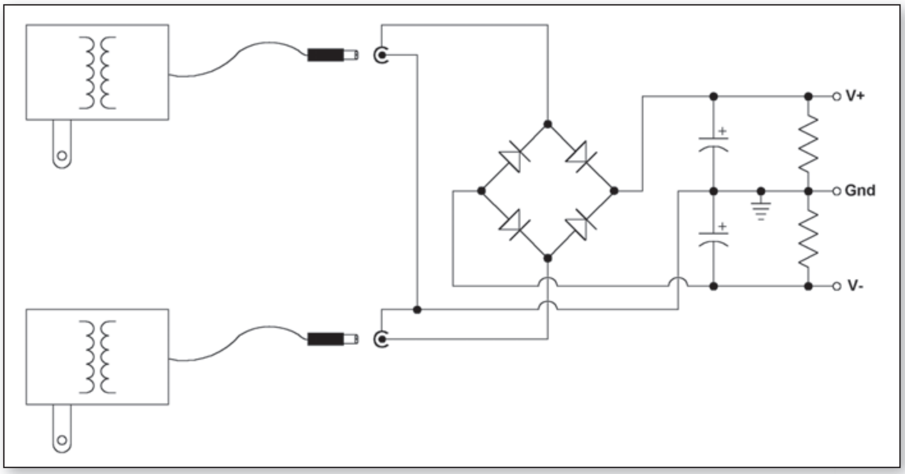


FIGURE 2: Two wall warts connected in series to simulate a center-tapped transformer.

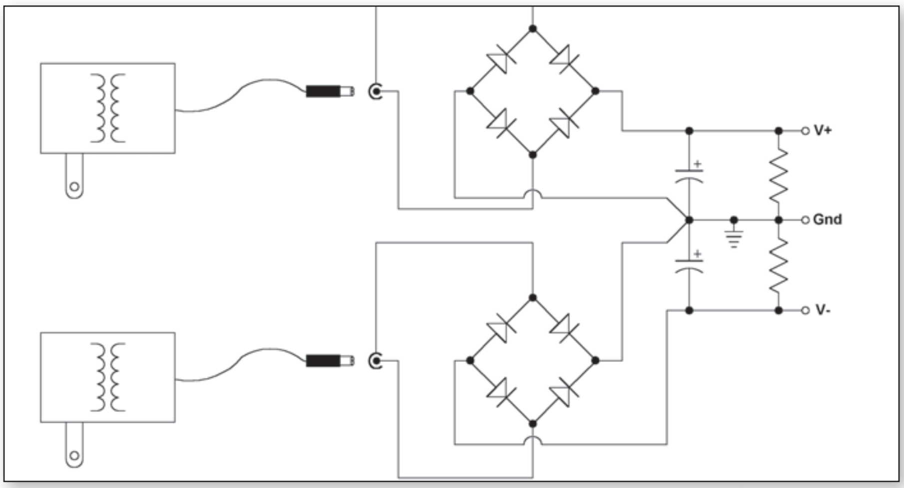


FIGURE 3: Two wall warts with separate diode bridges.

FILAMENTS AND FLOATING

Next, consider a pair of regulated DC supplies for tube filaments, each $6.3V_{DC}$, 1.2A. Voltage amplifier tubes normally have their filaments referenced to ground, but cathode followers and similar circuits lift their cathodes as high as $+160V_{DC}$. This can cause problems; every tube type has a maximum heater-to-cathode voltage rating, typically 90V, and the tube can misbehave seriously if that rating is exceeded.

The solution is to hoist the filament supply's reference up $60\text{-}80V_{DC}$ above ground, usually by hanging a voltage divider on the plate supply. With wall warts, this is easy, as shown in **Fig. 4**. Here the first wart drives the filaments of voltage amplifier tubes; its supply has its common terminal grounded. The second wart drives the filaments of cathode-follower tubes; its common terminal is connected to a voltage divider, similar to that found in my Gamp supply.[2] Note that while this illustration shows a negative common, the method works just as well with a circuit having a positive common.

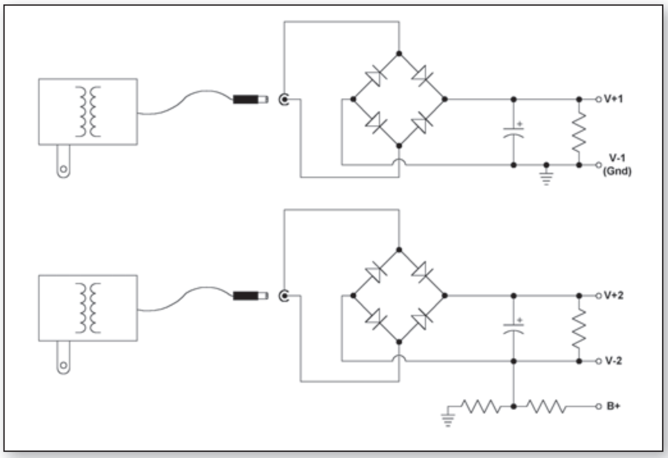


FIGURE 4: Two filament supplies for tubed circuits; the first is referenced to ground, the second to a voltage divider from the plate supply, scaled to hoist the filament about 80V above ground.

OK, which wall warts? Again without going through the math, I use a $12V_{AC}$ transformer to generate $6.3V_{DC}$, regulated. Jameco lists several $12V_{AC}$ devices; I'd go for cat. # 167151, rated at 1500mA. A pair of these will cost $15.18.

PHANTOM POWER

Time to get fancy with a $+48V_{DC}$ 30mA phantom power supply, used to supply juice to condenser microphones. Immediately you're faced with a problem: This requires about $45V_{AC}$ from the power transformer, and I haven't found any such wall warts on the market.

What to do? One solution is a voltage doubler, which would work with a $24V_{AC}$ wart, but I've already said I don't like those. Besides, this raises a practical question.

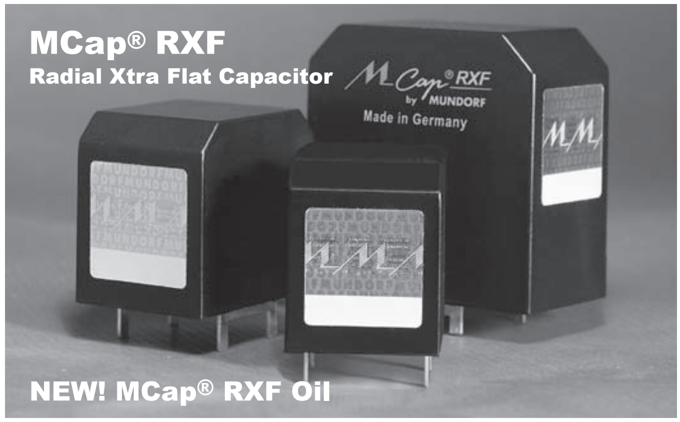
I have a strong prejudice in favor of making equipment idiotproof. This is because I sometimes work in situations where I'm sleep-deprived, and my higher cerebral functions become—shall we say—dulled (I've called this "chimp mode" in previous articles). In a wall wart context, this translates into a design limitation: If a device is powered by multiple wall warts, it should be impossible to plug the wrong wart into the wrong jack. Insurance against Murphy's Law, if you will.

One way to do this is to make the jacks different—use 2.5 × 5.5mm jacks for one or two warts, and a 2.1 × 5.5mm jack for another. That certainly works, but it requires a bit of thinking, and also opens the possibility of an operator in chimp mode (me) trying to shove the wrong plug in the wrong hole and bollixing up the whole works.

A more elegant solution is to design so that all the wall warts feeding a box are identical—any wart can plug into any jack without problems. How do you do that?

THINKING INSIDE THE BOX

Say you wanted to add the $+48V_{DC}$ phantom supply to the two tube filament supplies described in the previous section. They use $12V_{AC}$ warts, so you'll need to use the same for the phantom supply. To turn $12V_{AC}$ into the $45V_{AC}$ needed for the phantom supply, use an additional transformer located



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inside the audio chassis, connected to the external wall wart. Is this kosher? Yes—UL cares about what's plugged into the wall, not what's on the other end of the cord.

I said before you'd need $45V_{AC}$ on the phantom supply; to get that from $12V_{AC}$ you'll need a stepup transformer with a ratio of at least 12:45 = 1:3.75. One easy way to do that is to use a standard power transformer, hooking it up "backwards" with the normal secondary connected to the wall wart and the primary connected to the rest of the supply (**Fig. 5**).

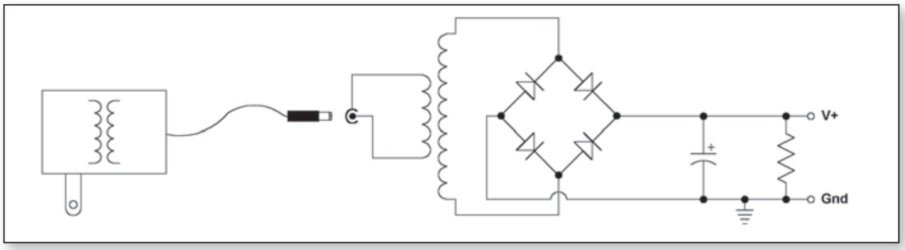

FIGURE 5: Using a second transformer to step up the voltage of a wall wart.

Allied Electronics has a decent variety of power transformers, and the prices of their "house brand" parts are lower than those of many name-brand transformers. Pulling out an Allied catalog, I find that the primaries of most of their transformers are defined as a nominal $117V_{AC}$, and 117/3.75 = $31.2V_{AC}$. That means that any Allied power transformer with a nominal secondary voltage of $31.2V_{AC}$ or lower will step up the wart's voltage adequately. ("Lower?" Yes; a lower nominal secondary voltage means a *greater* turns ratio in the transformer, which means more stepup for the wall wart's voltage.)

In practice, the next-lowest standard transformer voltage is $28V_{AC}$; Allied's cat. # 227-2025 is rated at 28VCT/1A, and costs $12.53. This has a turns ratio of 117:28, or about 4.18:1. Turned backwards, that will step up the $12V_{AC}$ wall wart's voltage to $50V_{AC}$, higher than you need, but it's easy to drop the excess voltage in the filter section. Leave the center tap unconnected.

What about current? An ideal transformer steps down the current when it steps up the voltage, and vice versa. I specified that the phantom power circuit will draw 30mA; it's easiest to see what happens to the current requirements by thinking backwards. The $117V_{AC}$ side of the Allied transformer is pulling 30mA, or .030A; the other side is rated at $28V_{AC}$, a lower voltage, so the current must be higher by the same ratio, 117:28, or 4.18×. That works out to 4.18 × .030A = .125A. That's how much current will be passing through the Allied transformer's 28V winding, and because it's rated for 1A there's plenty of margin. Similarly, the wall wart is rated at 1.5A, so it's nowhere near its limits.

Wait a minute, though. This design already has two filament supplies, each drawing 1.2A from warts rated for 1.5A. Would it be possible to piggyback the Allied transformer onto the first filament supply (with common grounded)? Yes; total draw would then be 1.325A, still within the wall wart's rated capacity. This means you'll have gotten three supplies out of only two wall warts.

How about adding the phantom supply to the balanced-rail ± supply described earlier? That has $18V_{AC}$ wall warts, so the in-the-box transformer needs to have a ratio of 18:45, or 1:2.5. This suggests a transformer with a secondary voltage of 117/2.5 = $46.8V_{AC}$ or lower, again connected backwards (the $117V_{AC}$ primary connects to the diode bridge, the secondary to the wall wart).

Flipping through the Allied catalog, I find that none of their house-brand transformers will do the job, but the Triad FP88-65 (Allied cat. # 967-8240) is rated at $44V_{AC}$ 0.13A with its primaries and secondaries paralleled. The primary is nominally $115V_{AC}$, so the turns ratio is 2.6:1, just about perfect, and it costs $10.58. It's a "flatpack" transformer, intended for mounting on a circuit board, but a piece of perfboard will work fine.

The internal transformer's output voltage will be 18 × 2.6 = $47V_{AC}$, again a bit high but easy to drop in the filter. Current requirement from the internal transformer will be .030 × 2.6 = .078A, so it'll pull very little from the wall wart. It would certainly make sense to piggyback this transformer onto one of the wall warts from the ± supply.

BLUE PLATE SPECIAL

Can you use a wall wart to power a tubed circuit? Yes, but for this you'll need a more drastic step-up transformer.

Say you want a plate supply in the +250-$300V_{DC}$ range that will provide 50mA of current. The easiest way to achieve that is to use a transformer with an "international" primary (dual $120V_{AC}$ windings). On this type of transformer, the primaries are hooked in parallel for nominal 115-$120V_{AC}$ countries, in series for nominal 230-$240V_{AC}$.

In this case, choose a transformer with a secondary voltage equal to the wall wart's voltage, and hook it up in reverse. To add a plate supply circuit to the previously described filament/phantom supplies, I'd use another $12V_{AC}$ wart (to keep Murphy out of the works).

Allied cat. # 227-0102 ($15.55) has a secondary rated at $12V_{AC}$/2A and a primary rated at $115/230V_{AC}$. Will that be enough juice?

Look at current first. If I draw 50mA (.05A) from the $230V_{AC}$ primary, I'll be pulling .05A × 230/12 = 0.96A through the $12V_{AC}$ secondary. This is about half the transformer's rated current, a nice conservative margin. It'll require a wall wart of its own, though; this is too much to be piggybacking.

What kind of voltage will this provide? Nominally, the supply should produce about $322V_{PK}$, or about $320V_{DC}$ with a 100µF input cap. With the line voltage 10% below normal, this will drop to about $288V_{DC}$. Allowing a few volts to drop in a second filter stage, regulated 270-$280V_{DC}$ seems reasonable, depending on the dropout voltage of your chosen regulator circuit. When the line voltage goes up to 10% over nominal, expect $353V_{PK}$, or about $350V_{DC}$ on the first filter stage; I'd use filter and bypass capacitors rated for at least $400V_{DC}$.

THREE IN A STACK?

In earlier sections I described methods for piggybacking multiple circuits onto wall warts to obtain three supplies from two warts. Can you get three from *one* wart?

As long as the current draw is reasonable, you can. With commercial parts, there's not enough juice to do filament, plate, and phantom supplies, but (say) ±15V_{DC} for op amps and +48V_{DC} for phantom is doable. The part that makes it possible is a transformer with dual primaries and single or dual secondaries of identical voltage. The Hammond 185*230 series is a good example, available from Allied (substitute letters C, D, E, F, or G for the asterisk to choose current rating). Triad makes similar transformers, either in wire-lead or printed-circuit-mounting packages.

Take a look at **Fig. 6**. The two primaries of the top transformer are connected in parallel; the two secondaries are connected in series. The nominal voltages are 115V_{AC} and 230V_{AC}, but in reality this transformer is a general-purpose converter that changes a voltage of X to a voltage of 2X, with a center tap.

In the example, the Hammond 185F230 transformer (Allied cat. # 836-1881) is connected to an 18V_{AC} wall wart to produce a 36VCT transformer, perfect for feeding a ±15V_{DC} regulated supply. Hanging on the same wart is the Triad FP88-65 to provide phantom power, as discussed above.

How much current can you draw from this supply? The wall wart is rated at 1A; the phantom circuit will pull 0.078A. This means you can draw an additional 922mA from the wart. Because the transformer has an effective 1:2 voltage ratio, the current ratio is the inverse of that, or 1:0.5; you can draw 461mA from the Hammond transformer's secondary. The Hammond transformer is rated at 570mA in series mode, so it will be happy.

Three outputs from one wall wart; what's the catch? Cost—the Hammond transformer runs $25.26, while an additional wall wart plus chassis jack would cost about $12.20. The trade-off is paying about $13 more for the simplicity (and potential Murphy-proofing) of a single wart. (Note that using the hookup described here also puts a higher-current transformer inside the box, which makes it more likely to cause coupling problems to audio circuits.) Electronics is the art of choice and compromise.

WINDING IT UP

I hope this article has demonstrated that wall warts don't deserve the junky reputation they've attracted. They're actually highly useful gadgets, and can help make your projects easy to build and thoroughly legal. Use them in good health! *aX*

REFERENCES

1. Jim Hagerman, "Calculating Optimum Snubbers," 1/1998 *Audio Electronics*, p. 26.

2. Paul J. Stamler, "The Gamp: Adaptable Power for Tube Amps," 7/2007 *audioXpress*, p. 20.

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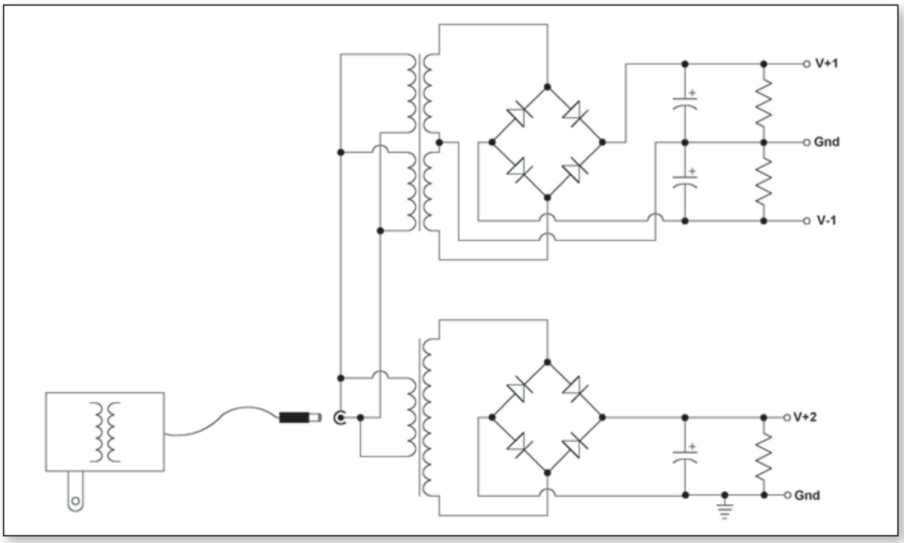

FIGURE 6: Using two additional transformers to derive three DC voltages from a single wall wart.

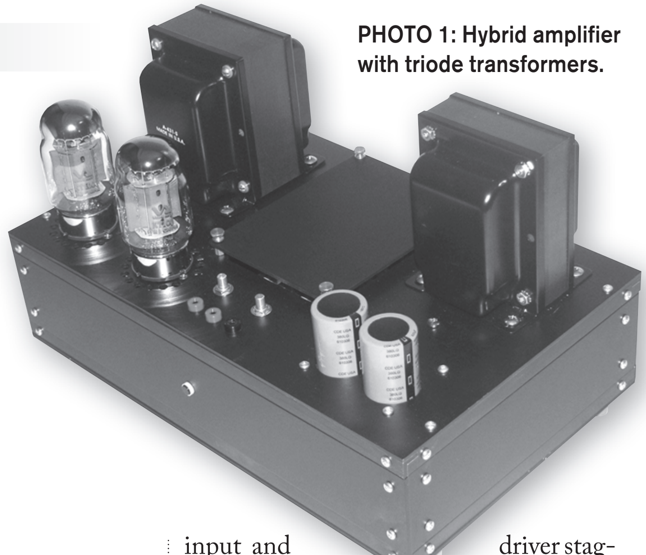

PHOTO 1: Hybrid amplifier with triode transformers.

A Hybrid **High-Fidelity Power Amplifier**

This patent-pending design features a National Semiconductor IC in a vacuum tube topology.

For years I have thought about building an amplifier with output tubes and transformer driven by a solid-state front end and driver stage. But I put the idea aside because I wanted to use integrated circuit operational amps to avoid the complexity of designing the driver stage with discrete components, and, to my knowledge, there wasn't a suitable IC op amp available. About a year and a half ago, that changed when National Semiconductor introduced the LM4702, a high voltage dual op amp intended for use in high-fidelity amps with solid-state output stages. In the commercial grade you can operate it from a ±75V power supply, which is more than adequate for the driver stage of a tube amp.

When the output devices of a class AB power amp are properly biased, there is a small overlap in the signals delivered by these devices. Class AB amps that employ solid-state devices in their output stages generate extremely fast switching transients when one device switches off and the other switches on. Traditionally these transients are nulled out with large amounts of negative feedback.

But this can cause an effect called transient intermodulation distortion. This is because the input stage where the error correction takes place has a difficult time handling the slew rate of these transients. This effect is difficult to measure, but it is audible. It is my opinion that this phenomenon is what makes tube amps sound different from solid-state amps.

CIRCUIT DESIGN

Using the LM4702, I designed a hybrid audio amp that uses a solid-state input and phase-splitting driver stage feeding a push-pull vacuum tube output stage typical of conventional vacuum tube audio amps. The input stage is composed of a non-inverting operational amp that accepts the input signal and negative feedback signal. The driver stage uses the LM4702 with one amp inverting and the other non-inverting, thereby providing the phase-splitting function with sufficient voltage swing needed to drive the control grids of vacuum tubes in a push-pull output stage. The combined gain of the input and driver amps is set to emulate their counterparts in a conventional vacuum tube amp. The gain is set higher than the overall closed loop gain of the amp by a factor determined by the amount of negative feedback that you desire.

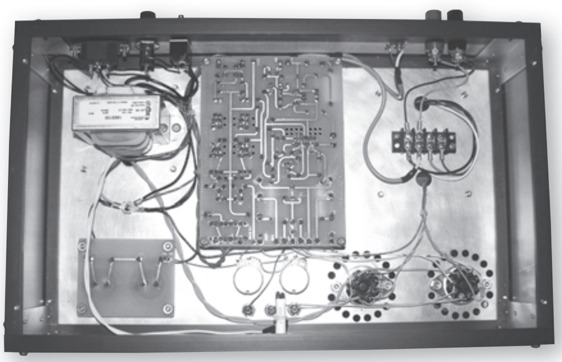

PHOTO 2: Hybrid amplifier chassis wiring.

For the solid-state operational amps I used Burr-Brown's (Texas Instruments) OPA134 for the input stage and National Instruments' LM4702 high voltage dual op amp for the driver stage. You can achieve much lower distortion in the input and driver stages with these amps than you can typically achieve with input and driver stages using vacuum tubes. This combination takes advantage of the strengths of both technologies.

The basic topology of the hybrid amp is shown in **Fig. 1**. It is typical of the topology found in many conventional vacuum tube amps, with the exception that the input amp and phase-splitting amps are solid-state. The input is connected to the non-inverting input of op amp U1, and the output of the feedback network of R7 and C3 is connected to the inverting input of U1. Resistors R1 and R2 form a local feedback loop that sets the gain and stabilizes this stage of the amp.

The output of this stage feeds two operational amplifiers U2a and U2b. U2a is configured as an inverting amp with resistors R3 and R4 setting its gain. U2B is configured as a non-inverting amp with resistors R5 and R6 setting its gain. The gains of U2a and U2b are equal in magnitude but opposite in polarity, thereby producing the required phase-splitting function.

The DC voltage of this stage is isolated from the output stage by capacitors C1 and C2. This is necessary because the control grids of the output tubes must be negatively biased with respect to their cathodes. The DC bias network performs this function. The output tubes are connected to the primary winding of the output transformer.

The secondary winding of the transformer is connected to the load and to a feedback network of resistor R7 and capacitor C3. Resistor R7 in conjunction with resistor R1 sets the overall closed loop gain of the amp in the audio range. Capacitor C3 is not always necessary, but when it is employed, it will reduce the gain of the amp above the audio range and in some cases may be necessary to stabilize the amp.

THREE EXAMPLES

I've built three amps with this topology. The first, a mono amp, uses the transformers out of an Eico HF-20 integrated amp that I bought at a flea market years ago. The Eico used 6L6GC output tubes, but I was able to upgrade to EL34s because I didn't need to provide filament current to preamp and driver tubes. The result was great. I achieved nearly 40W versus 20 that Eico specified and with less than 0.1% IM distortion. And it sounded great.

The second is a stereo amp that I built around Acrosound TO-330 output transformers, which I had from amps I built using Eico's HF-60 circuitry. I powered the amp with a Toroid of Maryland Transformer, P/N 326.5072, which was large enough to power the two channels. Again, the results were great. I achieved about 70W per channel.

The only problem these amps have is that the output transformers would be nearly impossible to duplicate. Therefore, I decided to build a third amp using readily available parts—transformers from Triode Electronics that are clones of transformers used in the Dyna Mark III. This is the amp that I am describing here. The schematic is shown in **Fig. 2** with the power supply in **Fig. 3**.

Combined gain for the input and driver stage is set to 180. The gain of the driver stage is about 32 or 30dB (National Semiconductor specifies a minimum gain of 26dB for stability). The

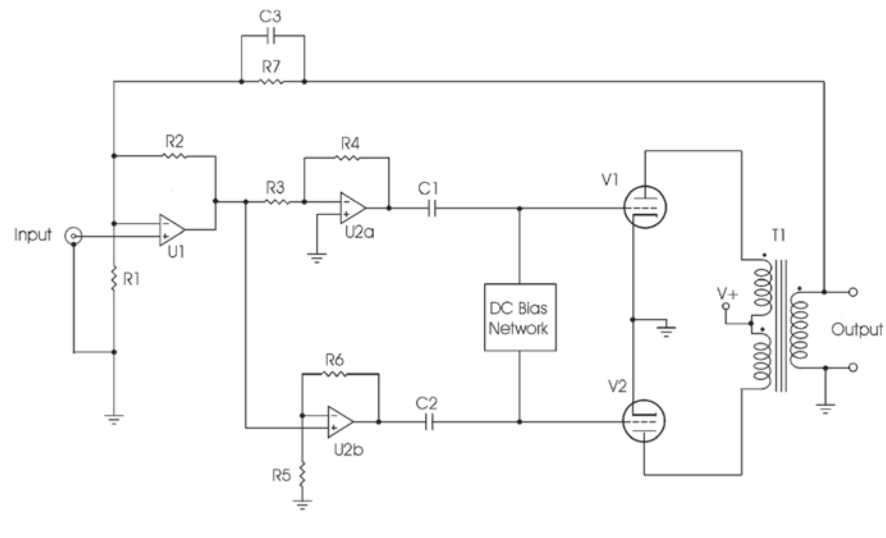

FIGURE 1: Simplified schematic.

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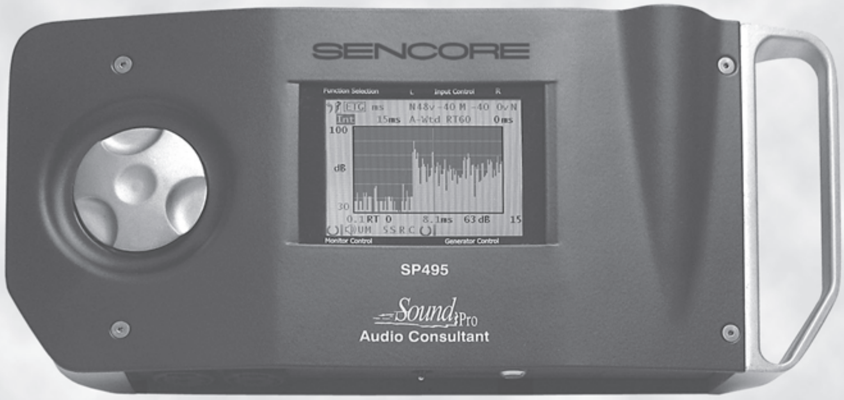

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closed loop gain of the complete amp with a 16Ω load is set by resistors R3 and R4 which turns out to be about 32. This gives roughly 15dB of negative feedback. Raising the value of R5 would increase the negative feedback and lower the distortion, but the amp sounds so good that I decided not to change it.

IC CONSIDERATIONS

Since it was designed for amps with a solid-state output, the LM4702 has a source and a sink output to facilitate biasing the output transistors. This is not needed in this application, so I've just tied the source and sink together. The LM4702 requires external compensation. C6 and C7 fulfill this requirement.

The LM4702 will dissipate approximately 3W with the power supply used here. Without a heatsink, the temperature rise would be 30° C per watt. You should use a heatsink with a thermal resistance of 7° C per watt or less to keep the rise within reason. I've also put ventilation holes in the PC board under the heatsink.

Provision for wiring a mute switch is provided on the PC board. If you're not going to use one, you should put a jumper on the board in place of the wiring to the switch.

The output stage is pretty typical. It is wired for ultralinear operation to maximize the power output. I put a ¼A fuse between ground and the cathode resistors of the tubes. Tubes do short on occasion and this affords a degree of protection for the high voltage supply and output transformer.

The 10Ω cathode resistors provide a convenient place to measure the bias and balance of the output stage. I included an AC balance pot R8 in series with resistor R8a. This allows the gains of the two driver amps to be set exactly equal. You could substitute a 51k resistor for R8 and R8a with satisfactory results.

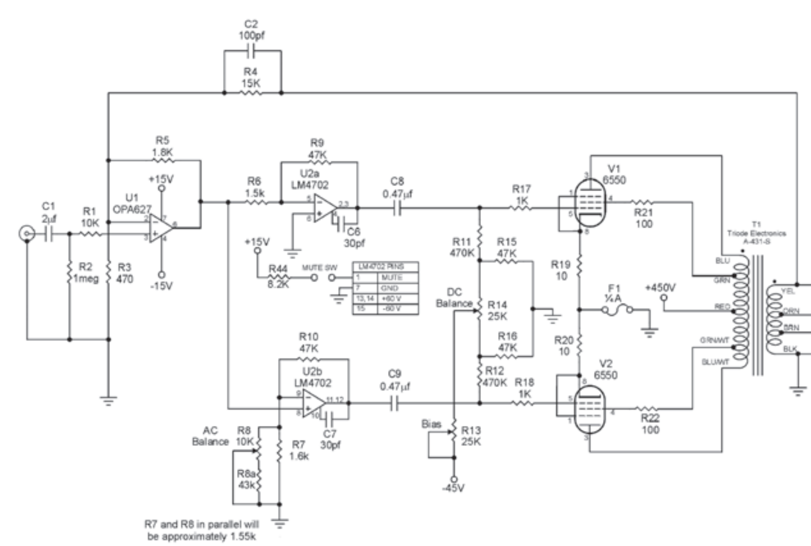

FIGURE 2: Hybrid power amplifier schematic.

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POWER SUPPLY DESIGN

The high voltage and bias supplies are not regulated but are highly filtered. I needed to add a second power supply transformer (Hammond 166 G100) for the solid-state input and driver stages. Power to the LM4702s is regulated using adjustable 3-terminal regulators.

I set the driver stage voltage to ±60V, which is higher than the maximum allowed for the LM317 and LM337 regulators, but National Semiconductor outlines a clever technique for overcoming the limitation in its linear brief, LB-47. The zener diodes ZD1 and ZD3 keep the voltages across the regulators well within their voltage limits. The ±15V supplies for the input op amp are derived from the ±60V supplies with simple shunt regulators.

I left space on the PC board for resistors R29 and R36 to dissipate some of the heat that would otherwise be handled by the transistors and regulators. It turned out they weren't necessary, so I just put jumpers in their place. The output voltage of the positive regulator is determined by the equation:

VCC+ = 1.25 * (1+R32/R33) + 0.00005*R32

Similarly, the voltage from the negative regulator is:

VCC- = -1.25 * (1+R40/R39) - 0.00005*R40

The large filter capacitors preceding these regulators are necessary to keep ripple voltage low enough to ensure adequate headroom for the regulators.

MECHANICAL ASSEMBLY

I used a 1/8˝ thick aluminum plate for mounting everything. This is thick enough to drill and tap, which facilitates mounting parts under the transformers. I drilled a ring of ventilation holes around each output tube and cut a large hole in the plate where the main printed circuit is mounted. I mounted the lower voltage power transformer under the main power transformer and a terminal block to select the output impedance of the amp under the output transformer. If you do the same, make sure the screws are flush with the top of the plate before mounting the main power and output transformers.

I like using dual banana plugs for connecting the speaker to a pair of 5-way binding posts, and using the terminal block to select the output impedance means that I have only two wires going to the back panel. The large hole has two advantages. First, it allows access to both sides of the PC board, and, second, it allows for ample ventilation.

The PC board is mounted under the hole with spacers, and a cover plate is mounted with spacers above the hole. Mount a smaller PC board containing

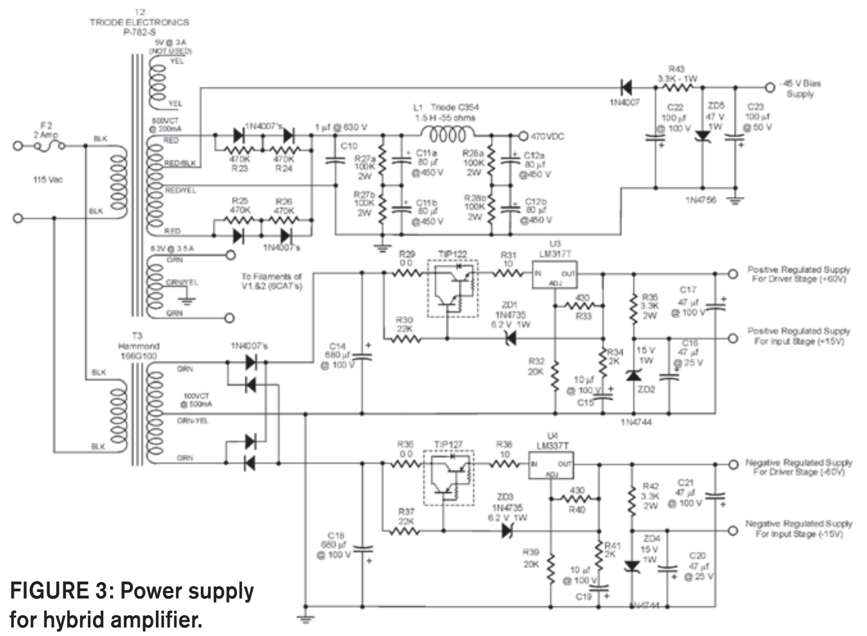

FIGURE 3: Power supply for hybrid amplifier.

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the high voltage filter capacitors in front of the power transformer and also mount a number 8 screw next to it to serve as a connection point for a star ground. All grounds are tied to this point.

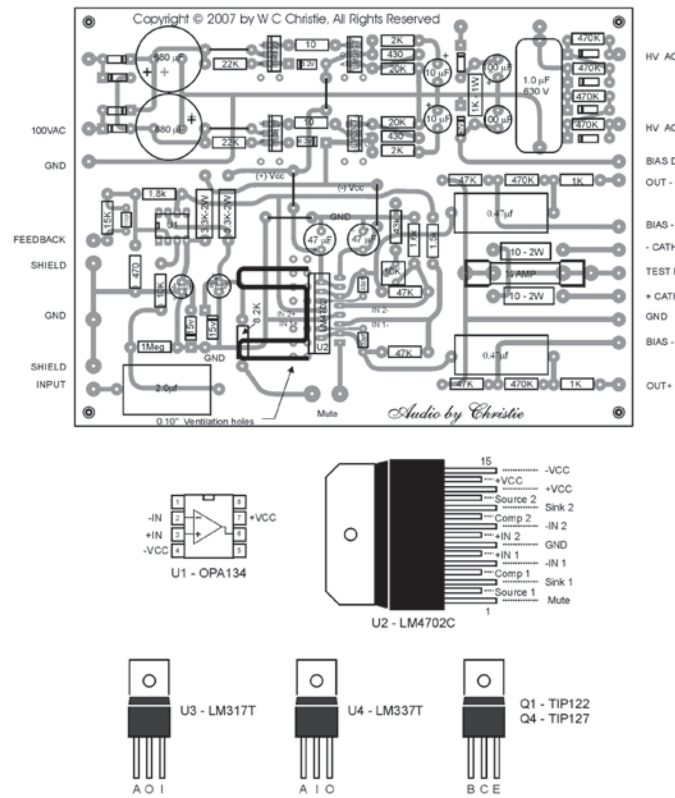
You can mount this assembly in either a wood or metal base. A back panel is used for the main power connector, fuse, switch, and input and output connectors. I made the bottom plate out of 1/8″ thick aluminum and put holes under the tubes and PC board.

FINAL ASSEMBLY AND ADJUSTMENTS

You should use a socket for the input op amp U1, and you need to solder the LM4702 driver IC directly to the PC board. Fully populate the board with the exception of jumpers J1 and J2 and don't insert the input op amp. This will allow checking out the power supplies without risking damage to the op amps.

With the amp otherwise fully wired and the output tubes not yet inserted,

FIGURE 4: PC board with component layout.



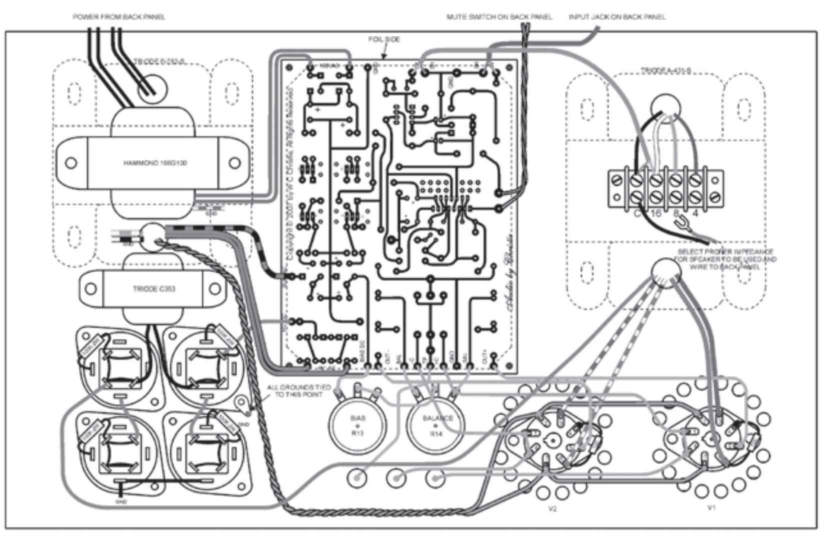


FIGURE 5: Chassis wiring for hybrid amplifier.

you can turn on the amp and measure the voltages. **Table 1** gives the voltages that you should find. Use extreme care when measuring high voltages, which, in this case, are high enough to be lethal.

Once you've checked out the power supplies, remove power and wait about five minutes for the filter capacitors to

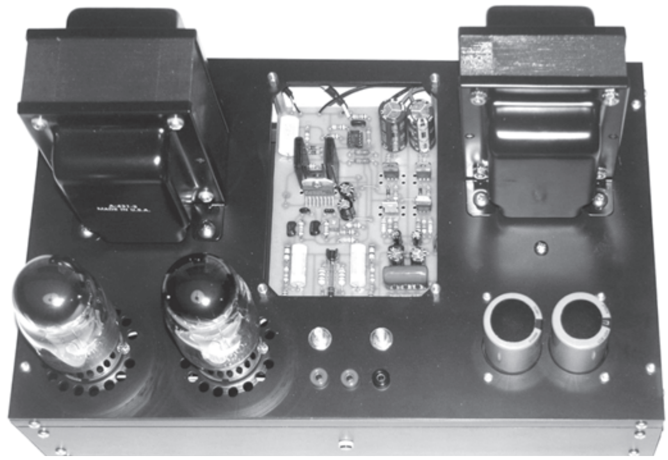


PHOTO 3: Hybrid amplifier with cover off.

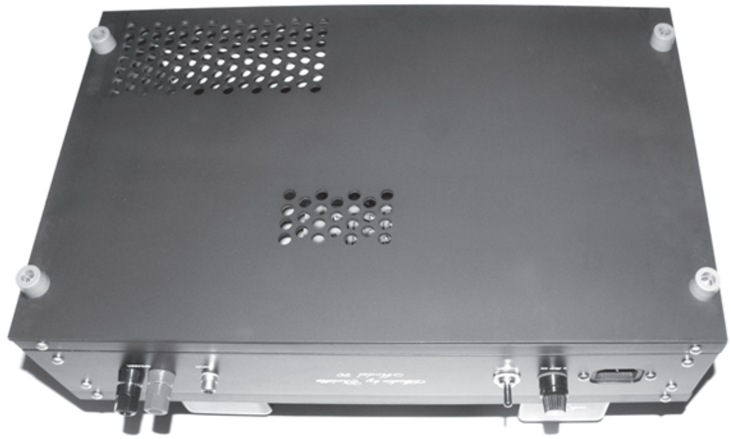


PHOTO 4: Hybrid amplifier, bottom view.

TABLE 1

Preliminary measurements before installing tubes		
High voltage DC	+600V	From C10 to ground
+Vcc	+60V	From J1 to ground
-Vcc	-60V	From J2 to ground
U1+ supply	+15V	From U1 pin 7 to ground
U1- supply	-15V	From U1 pin 4 to ground
Raw bias voltage	-78V	From pot R13 to ground
Bias voltage at V1	-69V	From V1 pin 5 to ground (R13 ccw)
Bias voltage at V2	-69V	From V2 pin 5 to ground (R13 ccw)
Measurement with tubes installed and bias adjustments made		
High voltage DC	+527V	From C10 to ground
+Vcc	+60V	From J1 to ground
-Vcc	-60V	From J2 to ground
U1+ supply	+15V	From U1 pin 7 to ground
U1- supply	-15V	From U1 pin 4 to ground
Raw bias voltage	-75V	From pot R13 to ground
Balance adjust	0V	From TP2 to TP3
V1 plate current	+70mA	+700mV from TP2 to TP1
V2 plate current	+70mA	+700mV from TP3 to TP1

Note: High voltage DC and raw bias voltage will vary with line voltage. Measurements in these tables were with line voltage at 120V AC and no input signal.

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TABLE 2

AMPLIFIER PERFORMANCE
Power output: 60W
IM distortion: Less than 0.1% at 1W
Less than 1.0% at 60W

TABLE 3 PARTS LIST

RESISTORS	
R1	10K ½W
R2	1 Meg
R3	470 ½W 1%
R4	15K ½W 1%
R5	3K3 ½W 1%
R6	1K5 ½W 1%
R7	1K6 ½W
R8	10K trimpot
R8a	43k
R9, R10	47K ½W 1%
R11,12,R24-27	470K ½W
R13,R14	25K chassis mount potentiometer
R15,R16	180K ½W
R17,R18	1K ½W
R19,R20	10 1W
R21,R22	100 ½W
R23	8K2 ½W
R28,R29	100K 2W
R30,R37	Jumper
R31,R38	22K ½W
R32,R39	10 ½W
R33,R40	20K ½W
R34,R41	430 ½W 1%
R35,R42	2K ½W 1%
R36,R43	3K3 - 2W
R44	3K3 1W
CAPACITORS	
C1	2µF
C2	100pF
C3,C4	15pF
C5,C6	33pF
C7,C8	470pF
C9	1µF at 630V
C10,C11	820µF at 400V
C12,C16	680µF at 100V
C13,C17	10µF at 100V
C14,C18	47µF at 100V
C15,C19	47µF at 25V
C20,C21	100µF at 100V
TRANSFORMERS	
T1	Triode Electronics A-431-S
T2	Triode Electronics P-782-S
T3	Hammond 546-166G100
DIODES	
ZD1,ZD3	1N4735 (6.2V - 1W)
ZD2,ZD4	1N4744 (15V - 1W)
D1,D2,. . .	1N4007 (1A - 1000 PIV)
TRANSISTORS	
T1	TIP122 (Darlington) (NPN 100V, 3A, 40W)
T2	TIP127 (Darlington) (PNP 100V, 3A, 40W)
ICs	
U1	OPA134PA (Single OP amplifier)
U2	LM4702C (Dual HV OP amplifier)
U3	LM317T (Pos. Adj.,. 1.5A)
U4	LM337T (Neg. Adj., 1.5A)
VACUUM TUBES	
V1,V2	KT88 (Power Tetrode)
FUSES	
F1	¼A
F2	3A Slow Blow

discharge. Plug in the input op amp, install the jumpers installed, and then you can test the input and driver stages. With a small signal applied to the input—say 0.1V RMS at 1kHz—you can adjust the AC balance pot to get equal AC voltages from the output of the amps in the driver stage. (Note that the mute switch must be closed to get anything out of the LM4702.)

Once this is done, you should remove the signal and short the input. Prior to inserting the output tubes, preset the bias and balance pot. You should set the balance pot at its mid point and the bias pot fully counterclockwise (zero ohms). Pins 5 of all the output tube sockets should read –45V or more negative. If they don't, then something is wired wrong and needs to be fixed before proceeding. Output tubes, which are expensive, can easily be damaged if the bias voltage is not negative enough.

With the power supplies checked out and the input and driver stages working properly, and again waiting for the filter capacitors to discharge, you can install the output tubes. You need to put a dummy load across the output terminals before turning on the amp. Without it the amp will be unstable and damage to the output transformer can occur.

Now you can adjust the bias and balance pots. You should short the input jack for this procedure. While the tubes are warming up, monitor the voltage across one of the 10Ω cathode resistors (test points 1 and 2 or test points 1 and 3). If it exceeds 0.8V, turn the amp off and look for wiring problems.

After the tubes have had time to warm up—say 3 or 4 minutes—adjust the balance pot for equal voltage on the cathodes (zero volts between test points 2 and 3). Next measure the voltage between test points 1 and 2 and slowly turn the bias pot clockwise to get a 0.6V reading. Let the amp set this way for an hour or so and then readjust the balance as before, and then adjust the bias for a reading of 0.75V between test points 1 and 2.

The amp should now be functioning. The output tubes will change slightly during a burn-in period. After a few days or a week of use, recheck and adjust the bias and balance if necessary. *aX*

Lundahl 1592 Transformer

Lundahl Transformers
Tibeliusgatan 7
SE-761 50 Norrtalje, Sweden
Phone: +46 – 176 13930
www.lundahl.se
Dimensions: 47 × 28 × 20mm
Net weight: 83gm

The LL1592 is a four-winding (two each, equal primaries and secondaries) high-level audio input transformer with mu-metal laminations. The windings are arranged to give a high degree of symmetry if you use the transformer for phase splitting. The transformer primary and secondary are separated by an electrostatic (Faraday) shield. Like all audio transformers, the LL1592 eliminates the inherent ground noise coupling mechanism.

A schematic of the LL1592 is shown in **Fig. 1** in the recommended configuration with the Faraday shield winding connected to the low side of the secondary. The secondary winding also shows the series RC termination network I used for measurements—a 390pF polystyrene capacitor and a 7k15 metal film resistor. The manufacturer-suggested values for best square-wave response are 400pF and 7k, which are non-standard. My RC termination network yields virtually the same time constant as the non-standard values recommended by Lundahl.

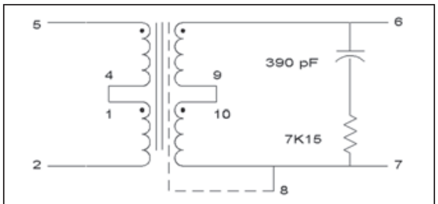

FIGURE 1: LL1592 schematic.

MEASUREMENTS

There is quite a bit of opportunity for "operator error" when testing the LL1592 because each specification calls for a different source and/or load resistance, and the absence or presence of the suggested RC terminating network.

The LL1592 maintains normal polarity. The input impedance measured approximately 37k at 1kHz, with the four DC winding resistances quite closely matched, between 276Ω and 281Ω. The winding-to-winding capacitance was 730pF, and the winding inductance was 614H, which should enhance low-frequency response. The leakage inductance (secondary shorted) was less than 1μH.

I recorded the output frequency response for resistive loads of 100k, 40k, and 10k with an input source impedance of 600Ω as specified in the datasheet. This response is shown in **Fig. 2**. At the nominal specified load of 40k with the termination network connected (solid lines), the frequency response for the LL1592 was within ±1dBu to 60kHz, with an input signal of +4dBu (1.228V RMS) at 1kHz. I also took data for loads of 600Ω and an IHF line load, but the response was down -12dB at 600Ω and is off the graph. There is about 0.5dB response peak at about 50kHz. Insertion loss at 1kHz with a 40k load was -0.38dBu.

Without the termination network (dashed lines) the high-frequency response peak increased to +2.5dB at the higher secondary loads, and a bit less at the lower loads. These peaks also occurred just above 50kHz at 100k load and increased in frequency as the load resistance decreased.

I was initially confused by the phase splitting balance specification of >46dB, and asked Per Lundahl for some clarification. The test calls for measuring the difference between the two secondary voltages with the primaries still in series, a 20k load on each secondary (a 2:1+1 connection), and the RC termination network across the two series-connected secondaries. He defined this test as the ratio of the secondary output (5V RMS for this test) to the difference between the voltage at each secondary. I saw 46.4dB at 50kHz using the prescribed measurement technique, but it was a higher 48.9dB without the RC network, perhaps due to some advantage gained by a bit more HF peaking.

The phase shift with frequency is shown in **Fig. 3**, output with respect to input, referenced to 0dB at 1kHz (lower curve using left side vertical legend). The source impedance is 600Ω, and the output load is 10k. The deviation from linear phase[1], using the right side vertical legend, is within the specified 2° out to 29kHz without the RC terminating network. With the network (solid lines), the phase shift—and thus the deviation from linear phase—increases more rapidly.

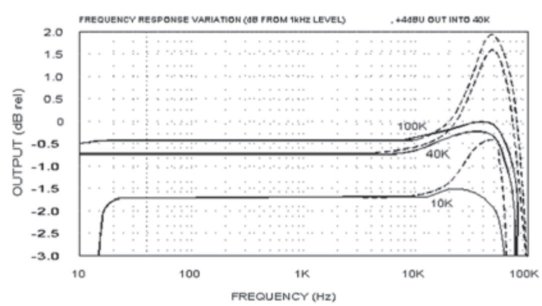

FIGURE 2: LL1592 frequency response.

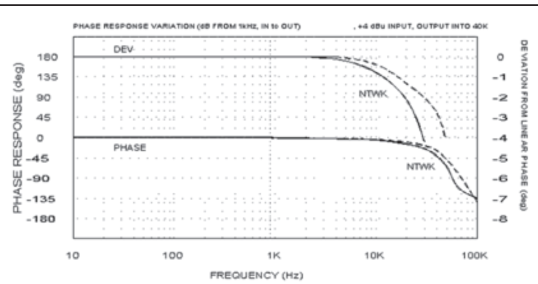

FIGURE 3: LL1592 phase response and deviation from linear phase.

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Figure 4 shows THD+N vs. frequency for the LL1592. I engaged the test set 80kHz low-pass filter to limit the out-of-band noise. In all instances the monitor output of the distortion test set, after the fundamental notch filter, showed the third harmonic, as would be expected of a magnetic core transformer. The LL1592 THD+N is relatively insensitive to output loading. There is just a hint of a dip in distortion at 60Hz at the higher values of load resistor, which may have been the result of the transformer being mounted on a prototype PC board without audio connectors and a metal case. The Lundahl transformer is fairly immune to external magnetic fields.

Figure 5 shows output THD+N versus input voltage into 40k at 20Hz, 1kHz, and 20kHz. Using a 40Hz input signal (not shown), the LL1592 endured a huge 25V RMS (+30.2dBu) before "clipping" at 1% THD+N. Beyond this point the increasing 3rd harmonic distortion due to impending core saturation caused the sine wave peak to flatten. Distortion was not affected by adding the RC termination network.

If you want to convert the graph voltages to dBu, the formula is

$$\text{dBu} = 20 \cdot \log(\text{Vin}/774.597)$$

I viewed the response of the LL1592 to three 2.5Vpp square-wave test frequencies on an analog scope using a 40k load and 600Ω source impedance. The response at 40Hz showed a reasonable amount of tilt with about 0.7Vpk of critically damped ringing on the leading edge of each transition. The ringing was at roughly 77kHz. The 1kHz and 10kHz square waves were nearly perfect except for the ringing. Changing the secondary load did not materially affect either the shape of the square waves or the ringing frequency. Removing the RC termination network increased the peak of the ringing about 16%.

I did find that the square-wave ac-

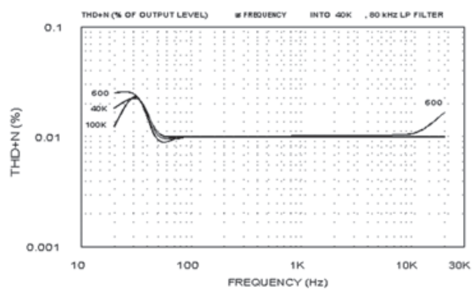


FIGURE 4: LL1592 THD+N vs. frequency.

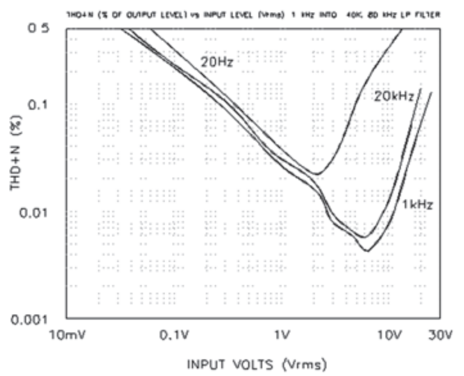


FIGURE 5: LL1592 THD+N vs. input voltage.

TABLE 1 MEASURED PERFORMANCE

Parameter	Manufacturer's Rating	Measured Results
Static Resistance, Pri and Sec:	**270Ω nominal**	**276Ω to 281Ω**
Distortion (series connection):		
+20dBU (10.94V RMS), 40Hz	0.1% typical	0.14%
+29dBU (21.83V RMS), 40Hz	<1%	0.93%
Self Resonance Point	**>120kHz**	
Frequency Response, ref 1kHz:		
Source 600Ω, Load 20k, no network	10Hz – 50kHz, ±1.0dB	10Hz – 62kHz, ±1.0dB
Source 600Ω, Load 100k, with network	10Hz – 100kHz, ±1.0dB	10Hz – 88kHz, ±1.0dB
Phase Splitting Balance, Source		
600Ω, Load 20k + 20k, with network	**>46dB, 10Hz – 50kHz**	**46.4dB at 50kHz**
Deviation from Linear Phase (DLP)[1]:		
Source 600Ω, Load 10k	10Hz – 20kHz, <2°	2° at 29kHz
Input Impedance, Zi, 1kHz, +4dBu:	**NS**	**Approximately 37k**
Insertion Loss, 1kHz, +4dBu:	**NS**	**-0.38dBu**
Capacitance, 1kHz, input to output:	**NS**	**30pF, no network**
Isolation:		Not Tested
Between Windings	3kV	
Windings and Shield	1.5kV	

curacy and the peak level of the HF ringing very much depends on the input impedance of the signal source. With a 50Ω source impedance, the 10kHz square wave had the highest peak on the leading edge. Increasing the source resistance to 2k while maintaining the same output level caused the leading edge of the 10kHz square wave to round over and the peak of the ringing to decrease. The best compromise between squareness and the peak added by ringing was at 1k7, but the number of cycles of ringing increased to cover almost the entire width of the top of the square wave. This indicates that the high-frequency response will be sensitive to the input source impedance.

Table 1 shows the manufacturer's specifications and my measured results for comparison.

COMPARISON

The Jensen[2] ISO-MAX CI-2RR is designed primarily as an isolation transformer for solving ground loop problems. The Lundahl has dual primary and secondary windings that make it more versatile. The Jensen has flatter frequency response and its built-in RC termination network does a better job of removing any HF peaking. I did not experiment at all with alternate RC time constants on the Lundahl termination network to try to improve the HF peaking.

Both transformers have very good deviation from linear phase performance. I would say they have similar THD versus frequency curves, with the Jensen showing a bit lower midrange distortion, and the Lundahl a bit flatter at the low end. You can drive the larger Lundahl transformer harder (higher input voltage and better handling of low output impedances), especially by taking advantage of the series winding configuration.

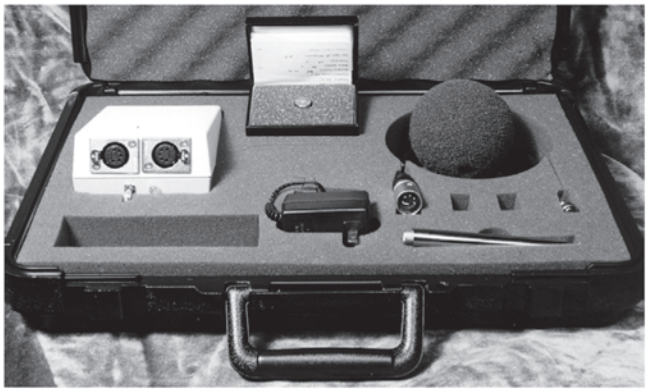
Both transformers gave a good account of themselves. Your choice may depend on whether you need the added versatility of the Lundahl dual windings. *aX*

REFERENCES

1. "High-Frequency Phase Specifications—Useful or Misleading?" Deane Jensen, 1986 AES Paper 2398 (E-8).

2. "Inside the ISO-MAX," Hansen, C., *audioXpress*, pp. 44-49, Oct. 2006.

Precision Acoustic Measurements Require Precision Microphones



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Lundahl and Jensen Meet the Monarchy M24

In the Oct. '07 issue of *aX* (p. 36) I reviewed Monarchy Audio's excellent M24 D/A converter. Charles Hansen published a thorough battery of measurements for the M24 in the June '07 issue (p. 20).

Readers will recall that the M24 has two analog outputs, tube and solid-state. The solid-state outputs are taken directly off the current-to-voltage (I/V) converter stage. Because they are minimally filtered, you need some type of bandwidth-limiting device to couple them to your preamplifier. For my review, I used the Monarchy-recommended Jensen CI-2RR isolation transformer, which I found gave excellent results. See Charles Hansen's measurements on the CI-2RR in *aX* Oct. '06 (p. 44), and my listening evaluations on this device in Jan. '07 (p. 26).

Hansen's measurement review of the M24 also included some tests of the Monarchy DAC used with the CI-2RR. Monarchy also recommended the Lundahl LL1592 line input transformer as an alternative to the Jensen. In this article, and the previous one (p. 37), Charles Hansen and I offer an evaluation of the Lundahl LL1592, including comparisons to the Jensen CI-2RR (our thanks to Per Lundahl, president of Lundahl Transformers, for providing samples for these articles).

Both the I/V and tube outputs on the M24 are polarity-inverted. You can reverse your speaker leads to correct for this, but that will make your system wrong for other sources, such as a turntable and phono preamp. A transformer offers an easy way to change polarity, by simply reversing the primary or secondary connections. Jensen and Lundahl both connect the electrostatic (Faraday) shield and the metal transformer case on the secondary side. So, you *must* reverse polarity on the primary side of the transformer. Later in this article, I'll show how to wire the transformer with both polarity-correct and polarity-inverted inputs.

FEATURES

The Lundahl LL1592 has dual primary and secondary windings, so you can use the transformer in both balanced and unbalanced applications. The following is from the LL1592 datasheet: "LL1592 is a high-level line input transformer with a mu-metal lamination core. The transformer is designed for high-end pro audio line input applications with or without phase splitting. The windings are arranged to give a high degree of symmetry if the transformer is used for phase splitting. The dual-coil structure also greatly improves immunity to external magnetic fields from, e.g., power supplies and motors. Primary and secondary windings are separated by electrostatic shields. The transformer is housed in a mu-metal can."

The Lundahl LL1592 datasheet recommends a series R/C damping network across the secondary of 7k + 400pF for the best square-wave response. As Chuck Hansen points out in his article, these are non-standard values, but any series combination that yields the same time constant should work. This 7k + 400pF combo yields a time constant of 2.8µs, corresponding to a frequency of 56.8kHz. I chose off-the-shelf values of 6.19k + 470pF, which gives a TC of 2.9µs, or 54.7kHz.

Per Lundahl notes that some audiophiles prefer the LL1592 without the R/C damping network, but I don't agree. Both measurements and listening evaluations indicate that the R/C network is beneficial, and that the 2.8µs time constant is correct. For the square-wave measurements described below, I used a 25k load, because that is the load provided by my preamp.

With a 1kHz, 1V peak-to-peak square wave and the R/C network connected, the LL1592 showed two cycles of well-damped ringing at 0.4V peak-to-peak amplitude. With the network removed, I observed five cycles of ringing at 0.8V peak-to-peak amplitude. A 10kHz, 1V peak-to-peak square wave with the network connected showed less than one cycle of ringing at 0.4V peak-to-peak amplitude. Removing the network increased the ringing to three cycles at 0.9V peak-to-peak amplitude.

I also checked Jensen's square-wave performance with and without the R/C network. A 1kHz, 1V peak-to-peak square wave with the network connected showed barely 0.5 cycles of ringing at a peak-to-peak amplitude of 0.05V. Removing the network increased the ringing to five cycles at 1V peak-to-peak amplitude. A 10kHz, 1V peak-to-peak square wave showed barely 0.5 cycles of ringing at a peak-to-peak amplitude of 0.1V, with the network in place. Removing the network increased the ringing to three cycles at a peak-to-peak amplitude of 1.0V.

With both the Jensen and the Lundahl transformers, changing the R/C time constants substantially above or below the specified values only degraded square-wave performance. Both manufacturers have worked out these values carefully, and I don't recommend changing them or removing the network. On the other hand, a slight deviation from the factory-recommended R/C values is not a problem. In fact, Jensen shows a 620pF capacitor in the datasheet for the JT-11P-1HPC transformer, though 680pF (a standard E12 value) is supplied with the CI-2RR.

MONARCHY FILTERING

In my Monarchy M24 review, I noted that there were visible high-frequency digital by-products riding on the distortion waveform at the I/V outputs. This is because of the I/V converter's minimal filtering. The bandwidth limiting of the transformer *used with the recommended R/C damping network* is necessary to remove these high-frequency by-products. The effect of the transformer coupling is also apparent in the actual distortion measurements.

I used my Denon Audio Technical CD (38C39-7147), a modified Sony CDP-XE500 CD player as a transport (I added a coax S/PDIF digital output to this player), and my Parts Connection D2D-1 sampling rate converter fed to the coax digital input of the M24. I ran distortion measurements with the D2D-1 in the transparent mode, and with the upsampling set to 48kHz and 96kHz. The distortion analyzer is my Sound Technology 1700B operated wide-band (without the 80kHz low-pass filter). I used a 25k load for all of these tests. The results are instructive.

First, Denon CD track 49, a 1kHz sine wave:

	Transparent	48kHz	96kHz
Lundahl LL1592	0.25%	0.23%	0.23%
Jensen CI-2RR	0.25%	0.23%	0.23%

Next, track 55, a 19.999kHz sine wave:

	Transparent	48kHz	96kHz
Lundahl LL1592	0.085%	0.065%	0.05%
Jensen CI-2RR	0.0.068%	0.06%	0.037%

Both transformers provide adequate filtering of the high-frequency digital by-products at 1kHz. Upsampling has virtually no effect on the 1kHz distortion measurements, but the 19.999kHz measurements improve with upsampling. Jensen's slightly better THD performance at this frequency simply indicates more effective filtering of the M-24's digital by-products. This measurement *does not* indicate that the Jensen has lower THD at audio frequencies.

CONSTRUCTION

The Lundahl LL1592 transformer is intended for PC board mounting, and also requires an enclosure. You can mount each LL1592 and R/C network on an Old Colony D-4 prototyping board, but you'll need to enlarge the holes in the board slightly to accommodate the rather large pins on the transformer (**Photo 1**). I recommend premium-quality components for the R/C network, such as the Vishay/Dale RN60 metal film resistors and Wima FKP2 polypropylene capacitors.

My favorite audio enclosures are still the modular units carried by Sescom. Unfortunately, Sescom has become a very unreliable dealer under their current ownership arrangement. Fortunately, there's now an alternate source: Wolgram Engineered Plastics, a company specializing in injection molding, which expanded into the metal enclosure business. Wolgram bought the metal fabrication equipment from Sescom and became Sescom's supplier for the MC-series metal cabinets and rack chassis. Wolgram now sells these enclosures themselves, and seems committed to providing good service. I now recommend Wolgram as the source for these excellent enclosures.

The MC-5A metal cabinet is just the right size for this project. I prefer the

PHOTO 1: A pair of Lundahl LL1592 line input transformers mounted on Old Colony D-4 prototyping boards in a Wolgram MC-5A enclosure with gold-plated, Teflon-insulated connectors.

Hum or Buzz Problems?

Don't defeat safety grounds (it's dangerous and illegal) and don't waste money on new wiring, ground rods, exotic cables, or power conditioning! *ISO-MAX® isolators solve ground loop problems safely while preserving pristine signal quality.*

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all-black version, but these enclosures are also available with natural aluminum front and rear panels. Specify your choice when ordering. I also used DH Labs' excellent CM-R1 chassis-mount RCA jacks, which make an excellent mate for their Ultimate RCA Connector. These premium RCA jacks are Teflon-insulated and feature DH Labs' Ultra-Gold plating. You must use the supplied insulators with these jacks in this project. DH Labs says that they require a 7/16″ hole, but I find that 3/8″ is just right. Product descriptions for DH Labs products are available on their website, but they don't normally sell direct; Parts Connexion is an excellent source.

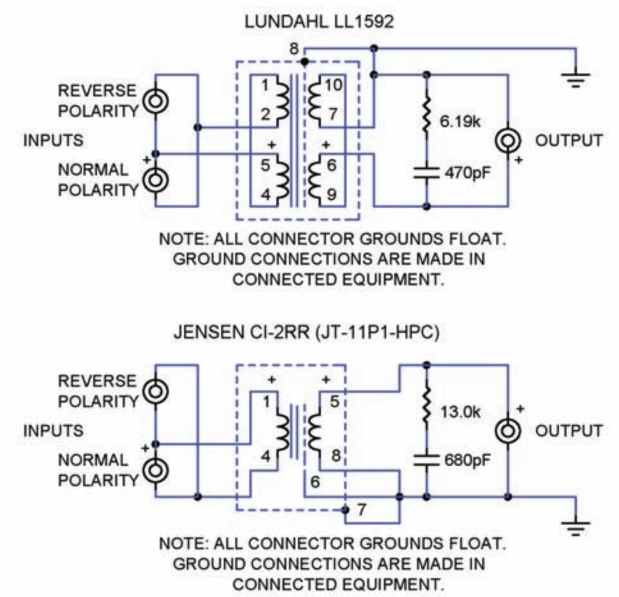


FIGURE 1: Connection diagrams for the Lundahl LL1592 and Jensen CI-2RR isolation transformers, including normal and reverse-polarity input connections. All jacks must be insulated from the chassis, with input and output ground connections made in the connected equipment.

Follow the wiring diagram in **Fig. 1**, and be especially careful of the wiring of the normal and reverse-polarity input jacks. **Photo 2** shows the rear-panel layout. I used center

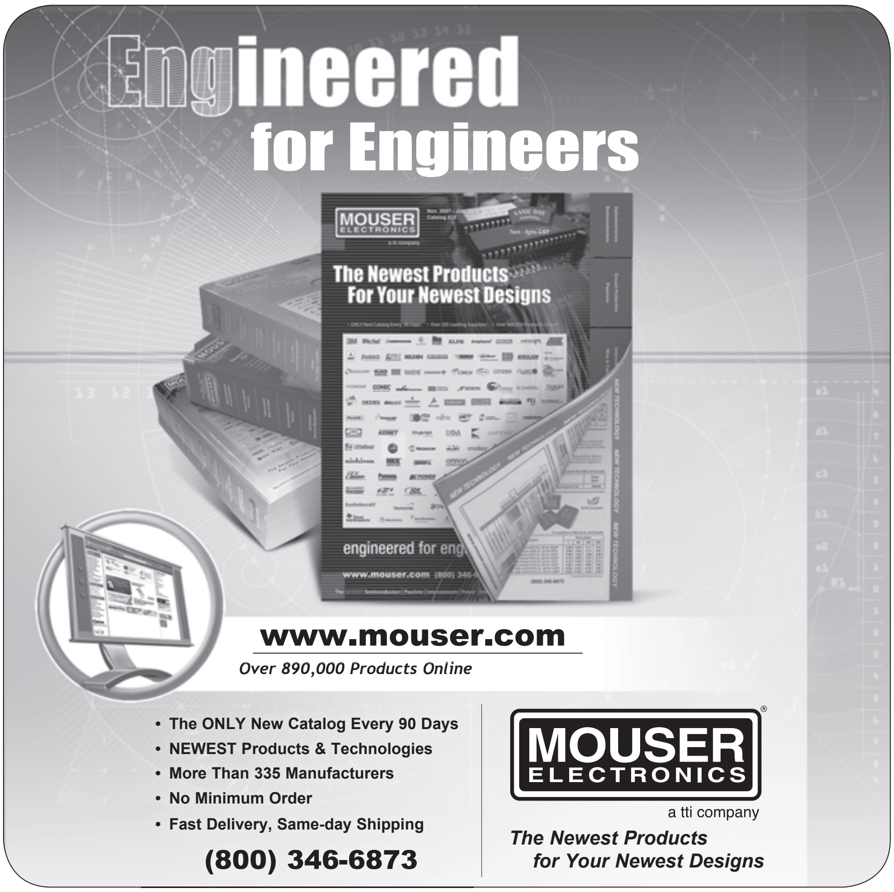
PHOTO 2: Rear panel of the transformer enclosure, showing DH Labs' excellent CM-R1 chassis-mount RCA jacks. Normal and reverse-polarity inputs are included.

conductors from DH Labs BL-1 interconnect cable for internal wiring, which I happened to have on hand, but you can also use their 18AWG hookup wire, which is similar.

The Jensen CI-2RR contains a pair of their JT-11P-1HPC line input transformers mounted on a PC board and housed in a small steel cabinet. If you wish to add the reverse-polarity input option to the Jensen, I recommend mounting the Jensen PC board in the MC-5A enclosure (**Photo 3**). Remove the PC board from the steel enclosure and remove the PC-mount input and output jacks. Add the necessary internal wiring and connect the board to the new input and output connectors (**Fig. 1**).

PHOTO 3: Jensen CI-2RR, audiophile version. The Jensen PC board is mounted in the Wolgram MC-5A enclosure. Vishay/Dale RN60 metal film resistors and Wima FKP2 polypropylene capacitors replace the R/C components supplied by Jensen.

With high-end internal wiring and the DH Labs RCA jacks, you are one step away from an "audiophile" version of the CI-2RR. Jensen supplied high-quality parts for the R/C network—Panasonic P-series polypropylene capacitors and Yageo metal film resistors, both readily available from Digi-Key.



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Both parts have steel leads. As a finishing touch, I recommend replacing these with the Vishay/Dale RN60 resistors and Wima FKP2 capacitors. These parts both feature non-ferrous construction, which I have found to make a subtle, but noticeable sonic improvement.

If you don't already have a CI-2RR, you may wish to purchase the JT-11P-1HPC transformers separately and mount them on a pair of Old Colony D-4 boards. You may be able to squeeze two of the Jensen transformers onto a single D-4 board, but it will be a tight fit. A CI-2RR retails for $178. A pair of JT-11P-1HPC transformers will cost $140, so you'll save some money by doing it from scratch.

TESTING

When you've finished building your transformer box, test it with a CD player and an oscilloscope. The scope should be externally triggered from one of your CD player's outputs—I suggest using the CD player's right channel as the source. Connect the left channel of the CD player to the oscilloscope's input (channel 1 if it's dual-trace). Play a 1kHz tone from your favorite test CD and observe the polarity on the scope. If your CD player maintains absolute polarity (and the test CD is also correct), the displayed waveform should begin in the positive direction.

Now, insert the transformer box in the signal path between the CD player and the scope, using the transformer box's NORMAL input. The polarity should be the same as it was with the CD player connected directly. Then, move the input cable to the INVERTED input. The polarity displayed on the scope should now be inverted. Be sure to check both channels of the transformer box. Once you've finished the transformer box, you can use the Monarchy M24 without reversing your speaker leads.

LISTENING EVALUATIONS

I built only one MC-5A transformer enclosure, drilled for both the D-4 prototyping boards, and the Jensen CI-2RR PC board (you can see the extra holes in **Photo 3**). For my listening comparisons, I used the Lundahl transformers mounted in the enclosure (**Photo 1**) and left the Jensen in its own enclosure. This might place the Jensen at a slight disadvantage, because the RCA connectors are not Teflon-insulated types. My Jensen CI-2RR PC board did have the Vishay/Dale and Wima R/C components for these tests. In its own enclosure, there's no way to reverse polarity on the Jensen transformers, so for these tests, I reversed my speaker leads, and used the Lundahl transformer box with the normal (non-inverted) inputs. I also compensated for the 1dB volume drop with the Jensen transformer—the Lundahl is unity gain.

Both transformers offered incredibly transparent, detailed sound. By most listening criteria, it is very difficult to hear the effects of either the Jensen or the Lundahl transformers in the signal path. Imaging and soundstaging were comparable, and bass performance was excellent with both.

However, with the Lundahl LL1592 used with the Monarchy M24's I/V outputs, I consistently observed some treble brightness. This is consistent with Charles Hansen's measurements, which show a slightly rising treble response even

with the damping network connected. On some program material, the treble brightness became fatiguing. I found the Jensen to be sonically neutral in the treble region. For this reason, I prefer the Jensen to the Lundahl in this application.

However, I also tried the Lundahl with the Monarchy M24's tube DAC outputs, which are already adequately filtered. Here, I found the Lundahl to be much more neutral sounding in the treble region on CD material, compared to its performance with the I/V output. I observed a slight treble brightness on 96kHz DVD material, including the Classic Records DVD of the Rachmaninoff *Symphonic Dances* with Donald Johanos and the Dallas Symphony (DAD 1004). But, the slight brightness did not cause listening fatigue. I can only conclude that much of the difference between the Lundahl and the Jensen, used with the M24's I/V outputs, is due to more effective filtering of digital byproducts by the Jensen. Some audiophiles might consider coupling tube equipment with a Lundahl transformer to compensate for the treble softness inherent in some tube designs.

If you ask a dozen audio engineers who makes the best audio transformers, two manufacturer's names will figure prominently in their answers: Jensen and Lundahl. The differences between the Jensen and Lundahl transformers evaluated here reflect somewhat different design priorities. Jensen opts for flattest frequency response and minimal ringing on square waves, while Lundahl opts for wider bandwidth. Both approaches have applications well suited to their respective design philosophies.

MANUFACTURER'S RESPONSE:

I always feel uncomfortable when getting requests for transformers with filtering characteristics (and in this particular case I cannot even recall being contacted by the manufacturer), as our goal is normally to build transformers with the greatest possible bandwidth. I have not always understood this feeling of discomfort, but after reading this review I understand my reactions better.

The transformer rated the best/most suitable for this particular application (which in my "applications dictionary" is primarily a line output application), the Jensen JT-11P-1HPC, is a high level high impedance line input transformer, designed to handle high signal levels in a limited size unit normally placed close to the input amplifier.

The limited size requires many turns of thin wire (to handle high signal levels at low THD), which results in high copper resistance and relatively high internal capacitance. These characteristics result in the limited bandwidth suitable for this application, but also in an output impedance above 3kΩ, which in my opinion is not appropriate for either a studio or a home environment. But this is *not* a criticism of the Jensen transformer. It is an excellent bridging input transformer with characteristics similar to our LL1540 in terms of size, impedance, and bandwidth.

The fact that the LL1592 (designed to be an audiophile alternative to the "standard" pro audio workhorse LL1540) performed well with the filtered signal but was less successful with the unfiltered signal is thus not surprising, given its greater bandwidth. And thanks to

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this article I can now better understand my feelings of discomfort (and have better formulated my cautions) when receiving similar requests in the future.

Per Lundahl
Managing Director of
Lundahl Transformers

I would like to clarify that the transformers are applicable for the new version of the Model 24 only, hereinafter called NM24. The older generation does not have a solid-state DAC output to be used with the transformers.

While it is only appropriate that the comments be left to the transformer manufacturers, I wish to point out there are two other types of coupling transformers that I have tried, with very satisfying results.

One is made by a company called Electra-Print Audio Co. in Las Vegas, and readers may contact Jack Elliano at electaudio@cox.net.

Their PVA-2N series actually has a higher winding ratio that yields a significant amount of gain, and Jack recommends just connecting a 10K pot at the secondary as a passive volume control, to drive any power amp with good sensitivity inputs (such as Monarchy's SM-70 PRO). Jack claims he uses a Nickel core material.

The other is made by Dignity Audio in Hong Kong. Readers can contact Franky Wong at: "hitekxmr" hitekxmr@biznetvigator.com. Franky uses a Permalloy core that is supposed to couple the signal from Input to Output much more efficiently.

The same core, in a much larger size, is used in his 300B power amp, DA08SE, that won "The Most Wanted Component of the Year 2007" from the www.stereotimes.com audio journal.

C.C. Poon, Monarchy Audio
aX

Parts List

Either Option:

MC-5A Black Enclosure (Wolgram)
(6) DH Labs CM-R1 Chassis-Mount RCA Jacks (Parts Connexion #68098)
DH Labs SH-18 Red and Black 18AWG Hookup Wire (Parts Connexion #54927 and #51968)

Lundahl Option:

(2) Lundahl LL1592 Line Input Transformers (Lundahl or K & K Audio)
Vishay/Dale RN60 Series 6.19k, 1% Metal Film Resistor (Mouser #71-RN60D-F- 6.19k)
Wima FKP2 Series 470pF/100V, 2.5% Polypropylene Capacitor (Mouser #505-FKP2-470/100/2.5, or Welborne Labs)

Jensen Option:

Jensen CI-2RR IsoMax Dual Audio Isolation Transformer (Jensen)
–or–
(2) Jensen JT-11P-1HPC Line Input Transformers (Jensen) and (2) Old Colony D-4 Prototyping Boards (Old Colony)
Vishay/Dale RN60 Series 13.0k, 1% Metal Film Resistor (Mouser #71-RN60D-F-13k)
Wima FKP2 Series 680pF/100V, 2.5% Polypropylene Capacitor (Mouser #505-FKP2-680/100/2.5, or Welborne Labs)

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Ad Index

FERRARI VS. GEO

After reading most of the article "Intrinsic Fidelity Testing" by Dennis Colin in the 11/07 issue, I was compelled to write a response:

The conclusion, "the claims of audibility of resistors, film caps, wire and the best op amps are unfounded," I believe is not correctly worded. It should read: "with my particular audio system components, speakers, power system and cables, I was unable to hear any difference when comparing these particular resistors, capacitors, wires and op amps."

The point I am making is that no definitive conclusions can be drawn that are of any use to anyone, unless there is a way to normalize the other variables. These variables—namely, system components, cables, acoustics, and your ears—are unique to you and your system, and so your conclusions should be your own. They should not be offered as general conclusions.

I continually get into disputes on the web discussion groups about audibility of cables, jitter, and other phenomena. The problem is always the same. We are talking apples and oranges. If all systems were equally resolving and had excellent room acoustics and all of our ears were equally good, then maybe we could compare notes and reach conclusions. However, the reality is that 99% of so-called "audiophile" systems are not capable of decent imaging or even have the capability to deliver the audible difference between a Steinway and a Baldwin piano or a realistic cymbal crash. This is reserved for the few systems in the world that are truly world-class, usually due to a *lot* of work, tens of years of upgrades and intelligent decisions on the part of their owners, and sometimes due to a large budget as well, but not always.

It is like the Formula Race Car. It runs on high-performance fuel and tires and is designed for these. If someone takes this high-performance fuel and these tires and puts them in/on his Geo, it probably won't drive any faster or better than with regular fuel and tires. It may even perform worse. This does not prove that these fuels and tires are not better. It just means that with his particular car, they don't make any difference.

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I can assure you that all of these things: "resistors, capacitors, wires, and op amps" make a big difference in a truly high-end component/system when listened to in an acoustically optimum listening room. All of these devices demand a premium price and they are not on shelves gathering dust. They are usually on back-order and in short supply.

If they didn't perform better than the average parts, then there would not be dozens of "modders" all over the world improving stock components for consumers by replacing these parts, some of them having 3-6 months backlogs.

Steve Nugent
nugent@empiricalaudio.com

Dennis Colin responds:
You said you read *most* of the article. If you read it completely, you'd see that what you referred to as my "conclusion," I stated to be "my personal opinions—open to debate." But I agree with your qualification "with this particular audio system. . . " However, I never described my observations as "definitive conclusions"—please read the entire article before you draw conclusions.

My audio system's resolution, naturalness, and acoustic environment are "truly world class," as you call that 1% of audio systems. The system includes my 250W "Mad Katy" tube amp (*aX* June/July '06), "Venue" speaker (*aX* Nov. '06), Swans M1 ribbon monitors with Scan-Speak subs (*aX* Sept. '05), and Sennheiser HD650 headphones and custom amp (*aX* Nov. '05).

Also, as described on p. 38 of the Intrinsic Fidelity article, my hearing is excellent. Perhaps more important, it's trained with decades of experience as a musician, symphonic concert attendee, studio engineer, psychoacoustic researcher, and speaker reviewer. And I'm very familiar with component quality issues, having worked over 40 years as an analog circuit designer. As I wrote, I welcome debate, but not condescension!

Your statement about some audio systems not resolving the difference between a Steinway and a Baldwin piano is puzzling, coming from one who can hear wires and resistors—I can hear the difference between a Steinway and a Baldwin over a telephone. (I dare say that most other pianists can too!)

In my Intrinsic Fidelity or conventional A/B comparisons, I like to listen through (in addition to a variety of excellent speakers) the aforementioned Sennheiser phones, which are probably more revealing, coherent, and neutral than any "world class" speaker. Sometimes I use live-miked voices and instruments as the audio source; a far cry from today's computer sound!

Therefore, I find your proposed analogy of "racing fuel and Formula tires on a Geo" as mildly insulting and totally irrelevant. Nevertheless, the focus of my article is not the audibility or inaudibility of specific devices, but rather the need for proper, intensive, and repeatable testing.

On that note, *Stereophile* has for years condemned blind testing as "flawed," citing everything from "mental testing stress" to quantum physics, to avoid the kind of serious testing whose results would offend many advertisers ("What do you mean you can't hear my battery-polarized cable dielectric?"). But now, lo and behold, on p. 3 of the Nov. '07 *Stereophile*, J. Gordon Holt, the magazine's retired founder, said, ". . . high-end audio lost its credibility during the 1980s, when it flatly refused to submit to the kind of basic honesty controls (double-blind testing, for example) that had legitimized every other serious scientific endeavor since Pascal." He also cited "the mindless acceptance of voodoo science."

Considering the above, your claim that, "If they (premium resistors, and so on) didn't perform better. . . there would not be dozens of "modders". . . improving stock components (with them). . ." does not prove that they produce an audible difference! Does the fact that many people buy those "magnetic therapy" bracelets prove that they actually work? Of course not, but we know that the placebo effect does.

I see from your website that you're a manufacturer of exotic cables for computer-processed audio. Now that could be analogous to using a Ferrari transmission with a Geo engine.

Three cheers for Dennis Colin's article, "Intrinsic Fidelity Testing" (*audioXpress* 11/07), which should be required reading for all audiophiles and audio engineers, and especially those equipment reviewers whose statements imply a superhuman ability to remember perfectly a particular sound heard in the past. As for engineers, there is a natural tendency to favor the products of one's own efforts, but there are audio checks and balances which can be employed to help eliminate self-delusion.

Doug Pomeroy
POMEROY AUDIO
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CORRECTION

In the article "Build a New Tone-Control Circuit" in the November 2007 issue, there are two missing connections in the schematic:

Junction of R8, R10, C15 to U1:B output

Junction of R15, R17, C12 to U1-:D output

Jon Lark
jd_lark@yahoo.com

AMBIOPHONICS

I read Roger Russell's article "Sounds and Hearing, Pt. 1" (Oct. '07, p. 28) with great interest and I thank him for mentioning Ambiophonics.

I urge you to go to the www.ambiophonics.org website and read the AES paper I recently presented at the 123rd convention entitled "360 degree Localization via 4.x RACE Processing." The R stands for recursive, and the paper describes how even the crosstalk correction signals are canceled at each ear until all the crosstalk is inaudible. Your statement, "However, the cancellation component from each speaker is heard by both ears" is simply not correct. This recursive nature of the Ambiophonic protocol has always distinguished it from Sonic Holography, VMAX, Lexicon Panorama mode, and so on.

You also state, "Ambiophonics claims that further computer processing produces very realistic effects even with two widely spaced front speakers." I don't know where this comes from, but in Ambiophonics the direct sound front speakers must be relatively close together—normally about 26° apart—although the exact spacing is not critical. Putting them on either side of a large TV screen is usually fine. For surround sound movies a rear pair of similarly closely spaced speakers enables a full 360° circle of direct sound to be localized, which is awesome if the movie really has a rear stage.

For music, concert hall impulse responses can be convolved in a computer, and purely hall reflection signals sent to ambience surround speakers whose precise location and number are not critical. Variety in direction is what counts. The idea is not that Ambiophonics is as good as being at a live performance, only that it reproduces CDs, LPs, DVDs, and so on much better than the traditional stereo triangle and comes closer to a live concert hall experience than any other home reproduction method demonstrated so far.

If you use the Ambiophone to make recordings, then I would claim a comparison with the real live experience is valid.

Ralph Glasgal
glasgal@ambiophonics.org

Roger Russell responds:

The double crosstalk should have been referenced separately and was not meant to be part of the Ambiophonics process. The source is from *Scientific American* 2/24/2002, "Surrounded by Sound," by Fiona Harvey.

"The crosstalk from one speaker would

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be cancelled out by an equal but opposite sound wave from another speaker. Of course, this method creates its own difficulties: the canceling wave also reaches both ears and interferes with the original sounds as well as the cross talk. Sensaura had to devise a set of overlapping signals that cancels out the cancellations."

You may recall my e-mail to you on 2/12/2007 when I wrote: "I first thought your method of eliminating stereo crosstalk was for widely-spaced speakers but now you indicate this is a waste of time. Does this mean that left and right speakers must be placed close together for your system to work?

"I am comparing this to the Hughes SRS AK-100 system. This was demonstrated to me several years ago and the speakers were placed 1 to 2 feet apart."

Your answer was: "The Ambiophonic algorithm which is now being included in TACT products or can be downloaded from the web is very tolerant of speaker spacing. It has two adjustments that compensate for weird speaker angles. I use about 20 degrees between the speakers but TACT users report that they like it just fine even without moving their speakers. I think they are just lazy and are probably missing a lot. There is no fixed rule, just move the speakers and your seat until you are happy. Ambiophonics is a golden-eared tweaker's delight and high-enders will spend years adjusting Ambiophonic speaker pairs, cables and tube amplifiers. The things that change are the maximum stage width and size of the sweet area; these will change with each recording just as in stereo."

I concluded from your e-mail that a wide speaker spacing can be acceptable, but perhaps this is only a matter of semantics between what is wide and what is narrow. In my original question, I am comparing the Hughes SRS system having a spacing of 1 to 2´. Now you say that spacing is normally 26°. At 10´ from either speaker, this comes out to be a speaker spacing of about 4½´.

With respect to the Hughes system that has a normal angle of 6 to 12°, 26° is wide, and perhaps you imply that it could be even more. However, the optimum effect for the Hughes SRS system deteriorates as the angle increases, and as it increases even

CONTRIBUTORS

Bill Fitzmaurice ("The CurveArray," p. 8) 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.

Bill Waslo ("Of Tweaks, Mods, and Evidence," p. 16) is an RF Design Engineer with a midwest engineering firm. With his wife Carol, he also runs Liberty Instruments, developer of the IMP, IMP/M, and Liberty Audiosuite analyzer systems. His interests include music, loudspeaker design, signal processing, reading, and gardening.

Paul J. Stamler ("In Praise of Wall Warts," p. 24) is a recording engineer/producer, musician, and technical writer; he also hosts a radio program, "No Time to Tarry Here," featuring traditional folk music and related stuff. He has delighted in 78s since he was a boy, when they were still being made.

Bill Christie ("A Hybrid Hi-Fidelity Power Amplifier," p. 30) developed an interest in electronics at the age of 6 after his father helped build his first crystal set. Since then he has built numerous kits and started building amplifiers from scratch while still in high school, starting with a mono tube amp designed by Mullard. He graduated from the University of Akron with a bachelor's degree in physics. He is a retired electrical engineer from Goodyear Tire & Rubber Company after serving 35 years designing controls and instrumentation for manufacturing.

Charles Hansen (Product Review: Lundahl 1592 Transformers, p. 37) is an electrical engineer and holds five patents in his field of engineering. He began building vacuum-tube audio equipment in college. He plays jazz guitar and enjoys modifying guitar amplifiers and effects to reduce noise and distortion, as well as building and restoring audio test equipment. He enjoys sailing and has over 150 magazine articles to his credit.

Gary Galo ("Lundahl and Jensen Meet the Monarchy M24," p. 40) 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.

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further, up to 60° at a 10´ distance, the sound not only loses more benefits of SRS, but also further alters the audible frequency balance. I pointed this out in my article.

From what you say, it would seem that the adjustments you mention can produce a variety of alternate realities that are very subjective and vary significantly from person to person.

PHONO PREAMP

I have a couple of questions for Dennis Colin about his "LP Ultra Low Distortion Phono Preamp" that appeared in the Sept. '07 issue.

1. Is the preamp suitable for a Dynavector 10X5 with an output of 0.28mV and minimum impedance of 30Ω?
2. The MM/MC switch S1A in Fig. 2 does not connect to anything. How do you switch in the extra gain needed for the MC cartridge?

Larry Cleven
lcleven@shaw.ca

Dennis Colin responds:

Thank you for your interest. Please note the corrections on p. 49 of the Oct. '07 issue. In particular, the top terminal of the MM/MC switch (labeled "MC") should be grounded.

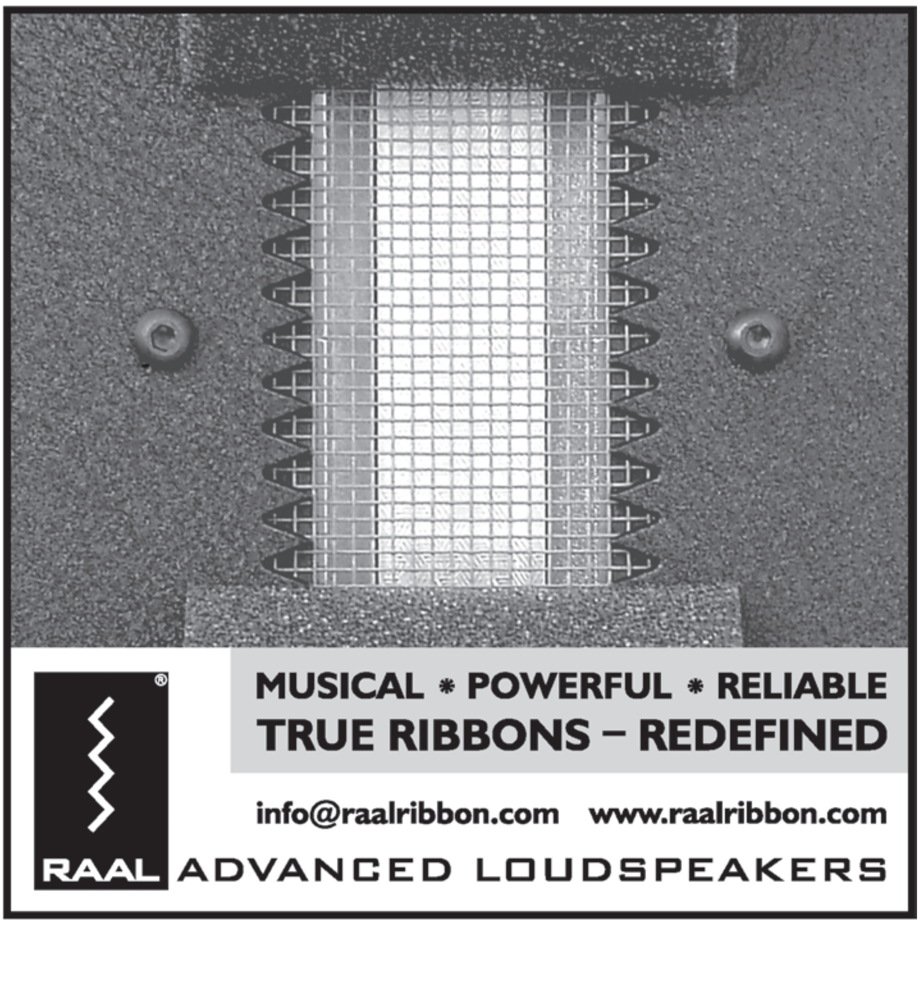
In the article I mentioned the Dynavector 10×5 (top of p. 13). On p. 12, Fig. 7 (curve 3) is plotted the preamp's noise spectrum with this cartridge. This is a high-output MC, specified at 2.5mV at 5cm/sec. The figure of 0.28mV that you stated does not apply to the 10×5; 0.28mV would be representative of a medium-to-low output MC.

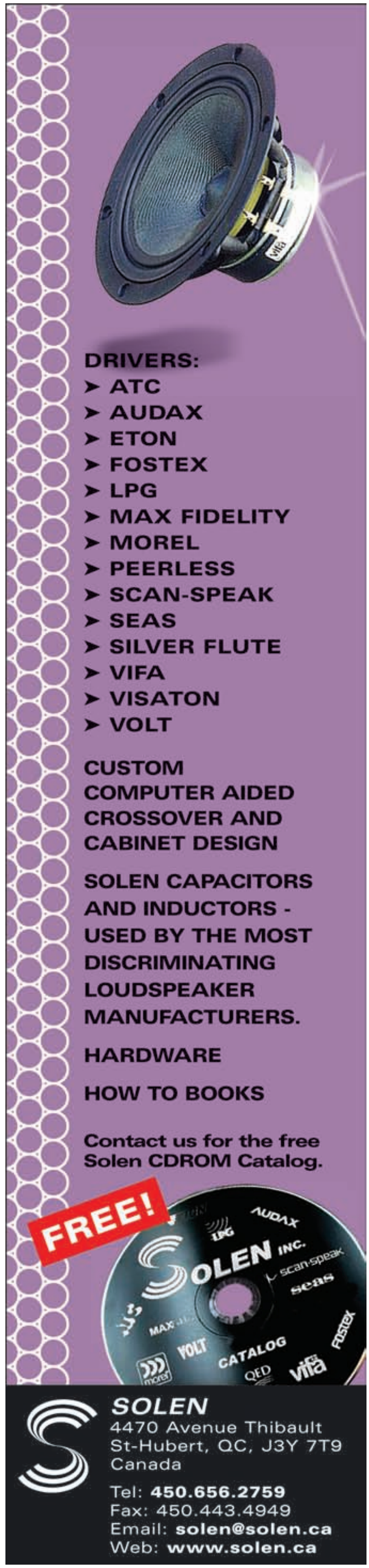
The 10×5 is most suitable with the LP797 preamp (use it with the switch in the MM position, *not* MC). This combination will provide a peak dynamic range of 103dB, headroom (re 5cm/sec) of 25dB at 1kHz and 42dB at 20kHz, noise that's likely inaudible, and very high quality sound.

The cartridge is very insensitive to load valves; Dynavector recommends 1000Ω, and you don't need to add any capacitance load.

I might be offering a kit and/or PC board; it will use the circuit shown in my article plus selectable R and C loading and gain (36 to 70dB). You can see the prototype on colinelectronics.com; click on "Groove Master."

aX





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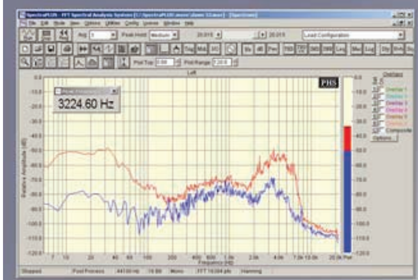

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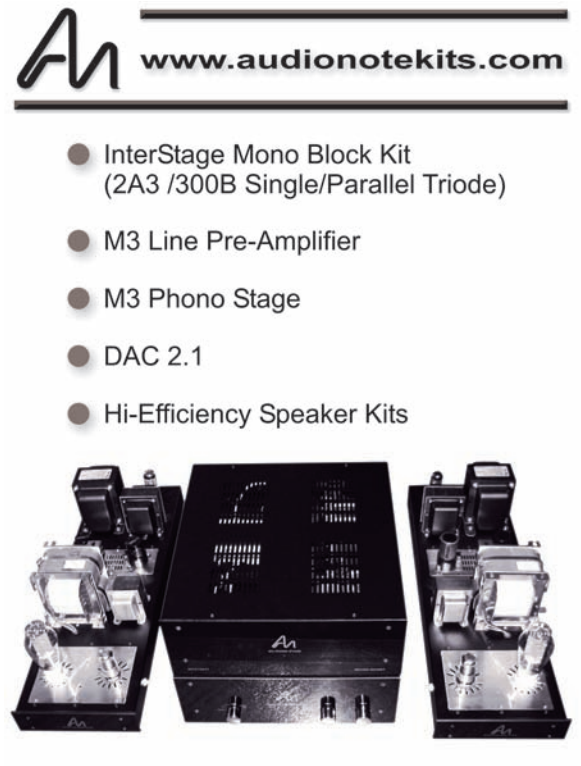

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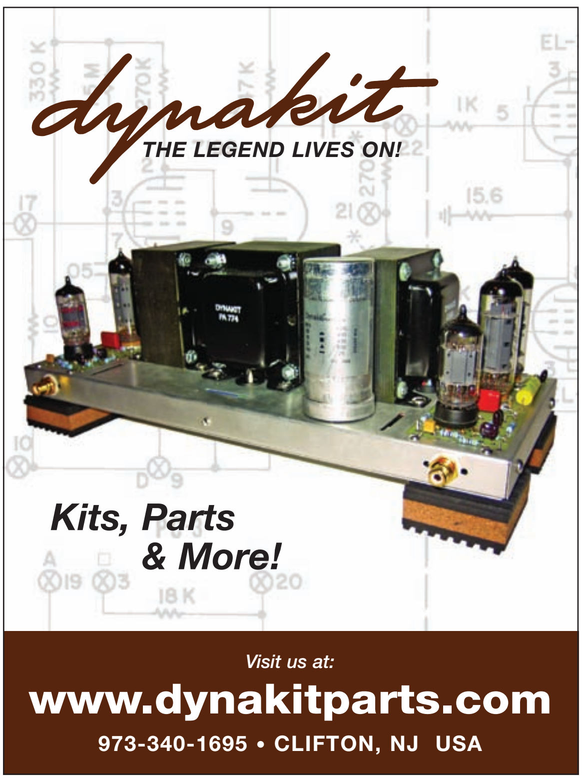

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