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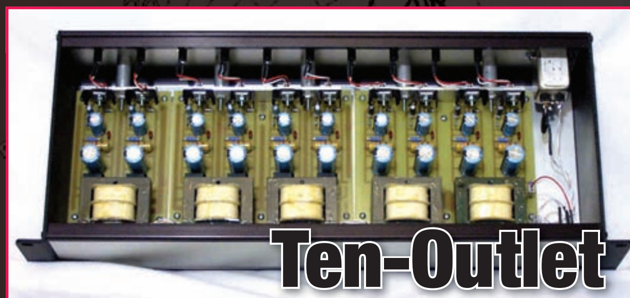

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Longevity

Speculation about whether a magazine dedicated to hands-on audio gear can survive seems to be widespread these days. Gene Pitts, distinguished editor of the occasional magazine, prophesied on the subject not too long ago talking to Boston Audio Society members. He suggested that our magazines would disappear when the publisher does. Maybe, maybe not.

The publisher is certainly older, but not an old person. He gave that up some time ago, deciding against old age. He has a strict diabetic diet, works out three days per week at the local Wellness Center (weights, treadmills, all manner of exercise machines), is careful with medications, and has multi-checkups. All reports are positive so far and he intends to keep it that way. Fortunately, he chose the right parents with excellent genes, and just for good measure wears a jewel in his left ear.

No one asks what he/she can do to help ensure the survival of our publications. But I do not believe that is a barrier to my discussing the matter with you. First of all, we have a great opportunity in that like so much else in our time, we have become international. *audioXpress* is the only English-language periodical in the whole world dealing with audio as a hands-on-enterprise.

And we are helped enormously by the cheap US dollar. The British are able to buy two of our dollars with one of their pounds sterling. So we are half price to our potential readers in the sterling zone. Europeans are able to acquire a dollar and four dimes with one of their Euros. We are a cheap buy even with the extra we must charge for overseas postage.

So those of you in our rapidly growing list of overseas readers are invited to spread the word: *audioXpress* is alive and well, and cheap. Grab a three-year sub while the dollar is weak and we will send you an English version of Jean Hiraga's classic audio obstacle course on CD. We even have a few in the original French if any of you citizens of that Republic would like one.

The other good news about our own little version of globalization is that our author roster is growing by leaps and bounds—now above 850 total. Most of the growth is from overseas: New Zealand, France, Spain, Brazil, Greece, India, to name just a few. Many are professionals—as has always been the case with those who write for *aX*. We do not recruit authors; we welcome articles, of course. We do not pay very much, far less than other mags we could mention who regularly offer anything from 65 cents to $1 per word.

The question of the magazine's survival is larger than the publisher's health, however.

The presence of advertisers in our pages is the result of concerted efforts on the part of the persons in our sales company who get up early daily over in Gloucester, Mass., to invite audio companies all over the world to place their offerings in the pages of *aX*. Some days they sleep in the mornings to talk to advertisers in the evening because the clients live in time zones halfway around the world.

You could give them, and us, a very large assist in a rather simple way. After each issue of one of our publications comes out, I sit down at my computer keyboard and leaf through looking at the ads. I am usually happily surprised by newcomers and wonder how you will respond to them. I am also unhappily surprised sometimes at the absence of some.

Would you consider a simple request from me? Whenever your issue arrives, set aside some time, perhaps an hour, to look through and consider the ads seriously. Click on their web addresses and see what they are about. If you need more information about their product, give them a call. Many of these enterprises are run by people just like yourself who are turning a hobby interest into a small business, usually specializing in a product or two.

Others of our advertisers are quite large, thank God, and believe in DIY enough that they allocate some of their budgets to contracts for our space. All of them are interesting and worth knowing about. If they know, from your response, that they interest you, then their impulse to renew their contracts is likely to be stronger. You have a direct influence on the effectiveness of advertising in *aX*. I do not expect your decisions about purchases to be influenced by anything other than the quality and usefulness of the products offered. Your responsiveness, however, is one of the most important ways, beyond subscription, which can ensure the survival of this periodical.—E.T.D.

Passive Crossover Linear Phase Speakers, Part 1

Extensive study of linear-phase passive crossover designs.

The focus here is on a speaker system that when fed an electrical square wave will output an acoustic square wave. Unfortunately, such systems are referred to by a variety of names. It is a system that is "transient perfect," one that produces no "delay dispersion" and no "phase distortion." Articles on any of these topics are of interest to anyone involved in linear-phase systems.

This article discusses ways to implement such systems with conventional cone drivers. The scope is limited to systems using passive crossovers (COs) with passing references to active CO approaches. I also discuss the effect of linear phase on the sound of a multi-way multi-driver system that I built last summer. Finally, I discuss what I learned during that work that may be advantageous to the design and construction of any speaker system.

WHAT IS A LINEAR-PHASE SYSTEM?

The subjects of time delay, group delay, and phase shift can become a bit confusing. Waslo[1] covered this topic very well, and I highly recommend his article. In defining a linear-phase speaker, I will talk only about phase shift and time delay, terms with which I believe the reader is comfortable.

My definition of a linear-phase speaker is one whose system phase shift can be plotted as a straight line at 0°. If it plots at ±180° the input terminals are reversed. How did I arrive at this definition?

To maintain the wave shape from electrical input to acoustic output, the speaker must be a pure time delay. This means it delays all frequencies the same amount of time. On a linear-phase-shift versus linear-frequency grid, a time delay plots as a straight line descending from the origin, 0° at DC. This straight line can be rotated to 0° by subtracting a fixed amount of time.

Once plotted at 0°, it makes no difference whether the frequency axis is linear or logarithmic. No real speaker system goes to DC or infinity, thus you are looking for speaker systems whose phase shift plots at 0° over a major portion of the audio band. The wave shape of signals whose fundamental and vital harmonics are contained within this portion of the band will be preserved.

IMPLEMENTATION POSSIBILITIES

If you had a single driver that covered the full audio frequency range with a stable acoustic origin[2] position, it could produce a linear-phase system. There are systems using a single driver, usually with the back of the cone horn loaded to limit displacement (minimizing Doppler distortion) and boosting the bass response. From what I read, most of these systems suffer from some frequency anomalies possibly because they generally have auxiliary cones (whizzers) that cause movement of the acoustic origin or other problems. They are of no interest to this article.

A good approximation to a linear-phase system can be implemented by a subwoofer-satellite system if the CO frequency is sufficiently low. Small drivers that cover the range from about 100Hz up are now available, and if they have a flat response with stable acoustic origin, they could yield a linear-phase system over a major portion of the band, but with limited power capability. These systems are also outside the scope of this article.

There are numerous linear-phase CO designs that can only be implemented in an active format. If the digital domain is used, then time delay can be included to correct for differences in the driver's acoustic origins. While this is probably the best approach to building a linear-phase speaker, it is not the topic of this article. This article is restricted to fully passive COs; i.e., the speaker system can be driven by a single amplifier channel with the entire CO function after the amplifier.

When you build with multiple drivers and passive COs, the positioning of the drivers is critical. The best case involves using a coaxial woofer-tweeter that has its acoustic origins properly aligned. With independent drivers, the spacing between them introduces additional problems. A multiple driver system shows linear phase over only a small region of the possible listening area. Keeping the drivers vertically aligned yields a wide area in the horizontal plane, but a rather limited area in the vertical plane. All systems studied in this article use vertically aligned drivers, so you need to study only the effects of vertical angle.

PASSIVE CROSSOVER CANDIDATES

The CO that most people know can produce linear phase is the first-order with all drivers connected with the same polarity. You can, indeed, build a linear phase system with this CO, but it is not easy. The first-order crossover requires that the drivers have a wide region where both drivers have a useful response, optimally two octaves each way from the CO frequency. The slow attenuation slopes of the first-order CO also make it susceptible to trouble from impedance anomalies such as the tweeter's resonance. Also, as demonstrated by Carlsson[3], the first-order CO is very sensitive to offsets in the driver's acoustic origins. This article will investigate this CO configuration.

It is well known that when you use a two-way second-order CO with both drivers the same polarity, a notch develops at the CO frequency. It turns out that you can add a third driver to "fill" in this notch, and this configuration can be phase linear. This was first discussed by Baekgaard[4] and expanded on by Stokes[5], who shows two different second-order and one third-order passive configurations. Stokes gives design equations to implement these with passive components. It was reported that someone commented, "If you have three drivers, why not do a three-way system?" I can only conclude this person had never tried to build a phase-linear system. I will look at this passive approach. Active implementations are discussed by Conniff[6] and Koonce[7].

Sandrik[8] offers an interesting approach to curing the problems of the gradual slopes of the first-order CO. He starts with a first-order CO and then increases the attenuation slopes some distance from the CO frequency. He does not claim to produce zero phase shift, only a minimum phase shift. Sandrik gives recommendations but does not show a general approach to circuit implementation. I will cover this approach showing one method of implementation.

Kreskovsky[9] offers a technique for building a phase-linear CO by overlapping the high-pass (HP) and low-pass (LP) responses and then adding an equalization network ahead of the CO to correct the magnitude and phase. For fully passive COs, the equalization must be done at the amplifier output. With this requirement, his approach shows something like a 4dB loss in system sensitivity, which I consider unacceptable and thus will not cover. If you are willing to put the equalization circuit between the preamp and amplifier or use Kreskovsky's[10] full active implementation, this approach looks much better.

THE IDEAL DRIVER

I start the investigation with those drivers you would love to have—but can't buy—the ideal driver. Throughout the audio band these drivers radiate a constant level (which is adjustable) with 0° phase shift and exhibit an input impedance of 8Ω at 0° phase. In addition, the acoustic origin is at the front of the mounting face so if they are flush mounted in a baffle the acoustic origins are in line. They are also point sources, thus radiating equally in all directions.

The only real-world consideration I will give to these drivers is they must be mounted in a vertical array rather than located at the same point in space. I assume a 6½″ woofer and a 4″ tweeter in overall diameter, thus requiring about

6″ center-to-center (CTC) spacing between drivers.

THE SPECIAL PLOTS PRESENTED

There are several items I choose to investigate about each approach. To plot each item independently would greatly increase the length of the article, and I have found that approach confusing when I read such articles. **Figure 1** shows the combination plot developed for this work. The plots are small with limited readability, so you will not be able to read "to the nearest dB." They do, however, show you how the system behaves in one figure. **Figure 1** shows modeled performance for an actual two-way system that I constructed.

The top left shows the response of the CO sections. The plot may be the electrical responses of the CO alone or, as in **Fig. 1**, the on-axis acoustic response of each driver driven by its CO section. The acoustic CO frequency is clearly just above 2kHz.

The second magnitude versus frequency plot is the system response at some point in space. While you don't listen there, the summation point normally used is infinity. If any other distance is being used, that will be noted in the text.

For **Fig. 1** it is the response on the tweeter's axis, the 0° reference, and back 10′ from the speaker's front panel. The lower left plot is the system phase shift at this same spot in space minus the phase shift caused by the sound traveling to that point. The two-way system in **Fig. 1** clearly is not minimum phase.

The top-right polar plot shows how the system magnitude response varies with vertical listening angle relative to the tweeter's axis. You can move the curves on this plot to place the main response at 0dB, so the level shown may not agree with the level shown in the system magnitude versus frequency plot. **Figure 1** shows polar plot curves at six frequencies, which are about 500, 1000, 2000, 3000, 5000, and 10,000Hz. You can't tell the curves apart in B&W printing; it is how the various curves interact that is of interest, not the individual curves. In **Fig. 1** it is clear the system performs best in magnitude at a listening angle between 0 and +10°, +5° being a good choice.

The bottom polar plot shows the system phase shift variation with vertical listening angle for the same frequencies. It is clear the system in **Fig. 1** is not linear phase at any listening angle. Keep in mind there is always an ambiguity when talking about phase shift.

On this polar plot –360° is the same as 0°, which is the same as +360°. When the phase shift was about to go off the plot, it was jumped up or down as needed by 720°. These long jumps have been blanked for clarity on the plot. Shorter jumps of 180° represent where the system has gone through a phase reversal at that frequency. You will see these points correspond in angle with nulls in the magnitude response.

The one item not covered in the figures is the system power response. Refer to the various references for discussion on this topic.

THE TWO-WAY FIRST-ORDER CROSSOVER

Figure 2 shows the performance of a two-way system using ideal drivers and a first-order CO. The drivers are spaced vertically at 6″ CTC and the CO frequency is 2kHz. The plot of electrical CO section outputs is identical to the acoustic driver responses due to the ideal drivers. The 0° reference is the tweeter and the listener is back 10′. The system frequency magnitude is an ideal 0dB and the system phase shift 0° at all frequencies, just as desired. The lower

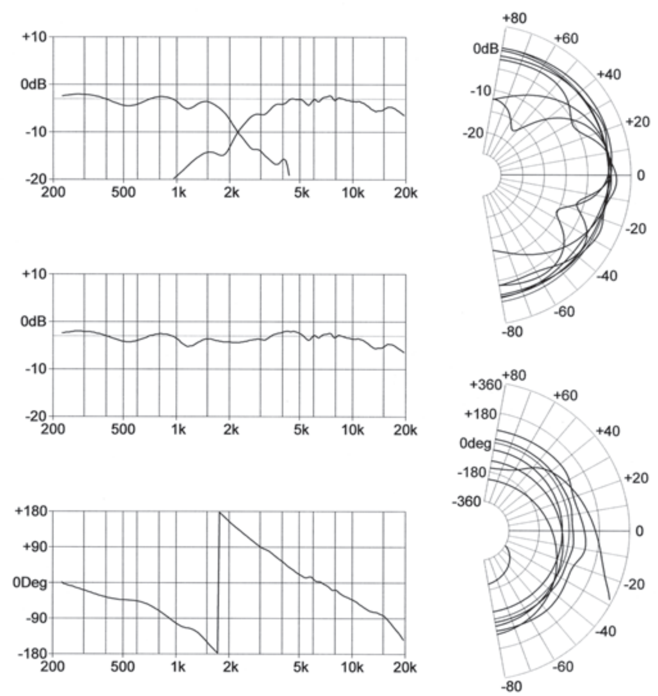

FIGURE 1: The special plot developed for this article, see text.

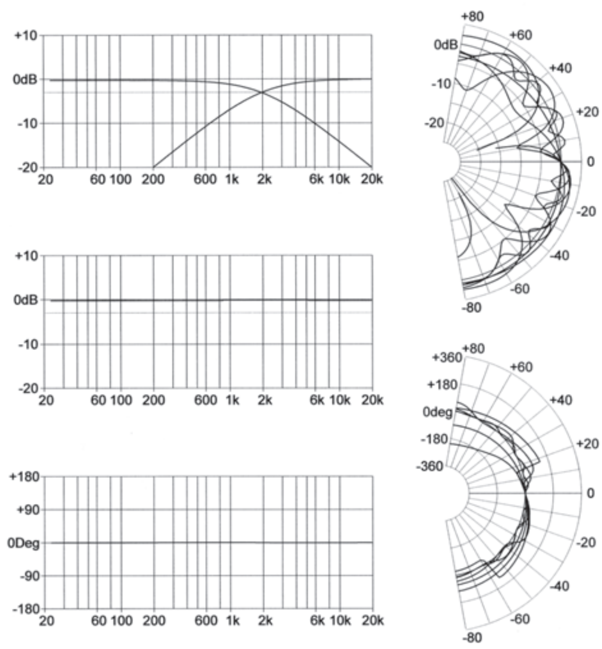

FIGURE 2: Performance of two-way system with ideal drivers and first-order crossover.

polar plot shows the system at 0° is phase linear and varies from linear fairly slowly with vertical angle.

It is the upper polar plot that indicates trouble. Yes, the 0° response is 0dB; however, at numerous other angles certain frequencies show a magnitude of +3dB. These give floor, ceiling, and other echo producers a level advantage over the on-axis response. Note also that at angles slightly above the listening axis certain frequencies show deep magnitude nulls, making performance for a standing listener questionable. These effects are referred to as lobing problems and can result in systems whose sound varies considerably with the location of the speaker and the listener and also the local environment.

To get the system to go phase linear at the listening point, you must move the ideal tweeter 0.12″ behind the ideal woofer. Thus, even with ideal drivers the first-order CO has some problems in a two-way system. A slight step is required in the front panel to focus the response to a fixed listening distance, and the response has mild lobing problems.

When you use real drivers the situation becomes worse. **Figure 3** shows the responses for the two drivers used in the two-driver systems in this article. They show considerable overlap about the 2kHz CO frequency. Real drivers always have zobels to flatten their high-frequency impedance to an approximate resistance.

Both drivers are flush-mounted in a flat front panel. Any change in one driver's acoustic origin position requires a step in the front panel. Also, real drivers have directivity that causes their off-axis magnitude response to roll off at the upper end of their frequency response band.

Figure 4 shows the performance of the two-way system with real drivers flush-mounted on a flat front panel and using a first-order CO. The top-left plot shows the electrical outputs of the two CO sections and a problem is clearly evident. The tweeter's impedance rise at its resonance overrides the slow rolloff of the first-order HP, causing the tweeter input to rise to 0dB. This also causes the notch in the system magnitude response and wiggle in the phase response.

You may also have the real-world problem of possible

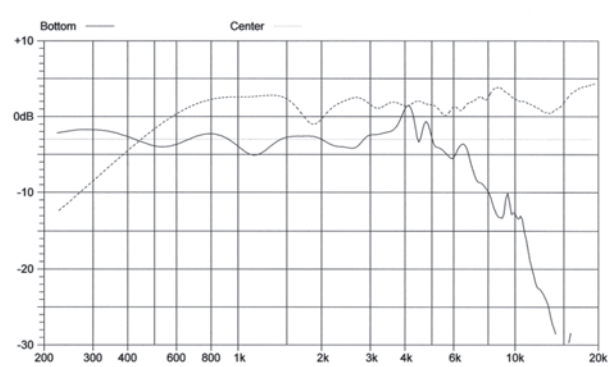

FIGURE 3: Responses of woofer and tweeter used in two-driver systems.

Doppler distortion or destruction of the tweeter. Note also that the point of linear phase is several degrees below the tweeter axis. The magnitude and phase shift versus frequency plots are at a vertical angle of –12°, which you can possibly correct by putting a step in the front panel. The magnitude polar plot is also a mess and does not show an optimum response at the angle of best phase performance.

The cure for the peak in the HP response at the tweeter's resonance is to place a series R-L-C network across the tweeter in an effort to cancel the impedance rise. **Figure 5** shows the same system with this network added and a step in the front panel to move the nearly linear phase point to 0°. The response plots are now on the tweeter's axis with summation at infinity. The system still has lobing problems where the response at 0° is not the largest response relative to other angles.

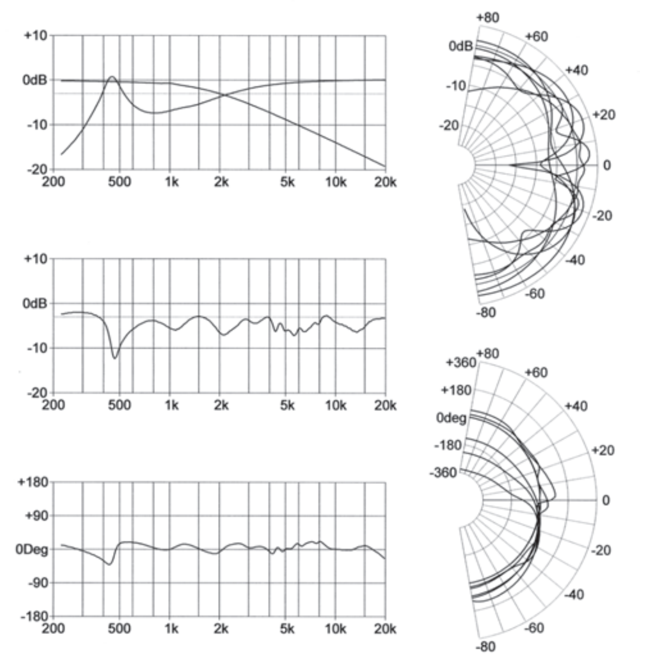

FIGURE 4: Performance of two-way system with real drivers and first-order crossover.

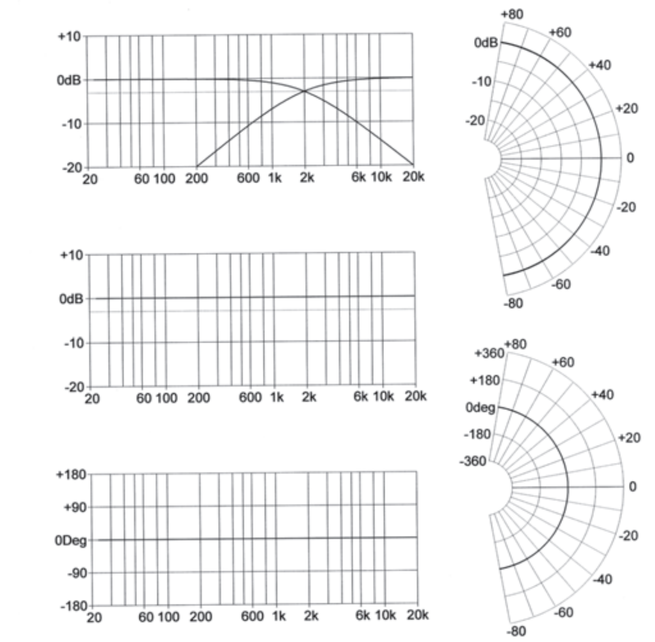

FIGURE 6: Performance of ideal coaxial driver with first-order crossover.

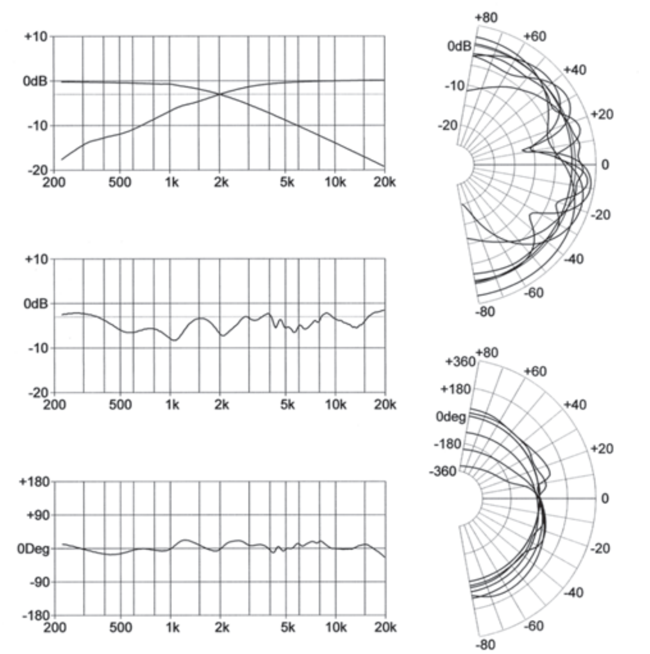

FIGURE 5: Performance of modified two-way system with real drivers and first-order crossover.

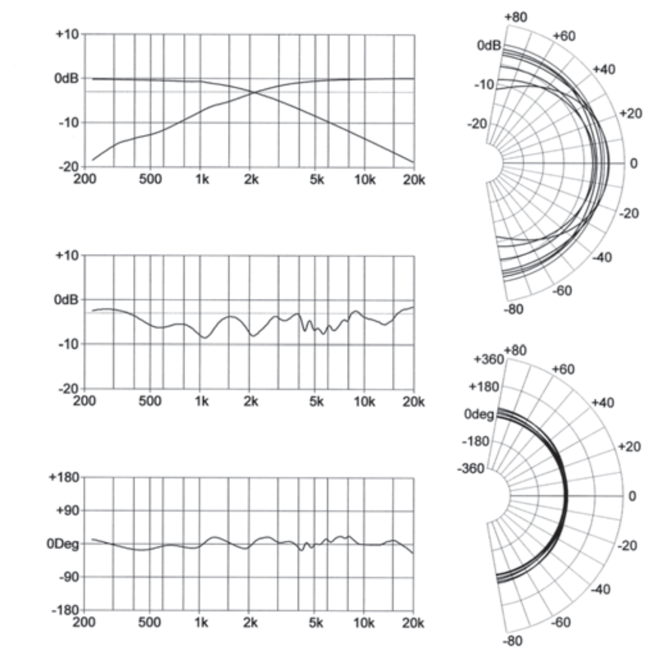

FIGURE 7: Performance of real coaxial driver with first-order crossover.

In summary, the first-order CO with two drivers is not a great approach to building a linear-phase system. Is there any cure? As noted earlier, the coaxial driver has a major advantage, and **Fig. 6** shows the response of an ideal coaxial driver with first-order CO. This looks beautiful. It is phase linear at all angles and has no lobes to cause trouble.

When a real coaxial driver is modeled (the same two drivers located at the same location), the performance is still quite good (**Fig. 7**). Note this still requires the network to cancel the tweeter-resonance-impedance rise and that the manufacturer has properly offset the two acoustic origins. Clearly you have a better shot at building a two-way phase-linear system with a good coaxial driver.

Another helpful consideration is to build with a symmetrical (D'Appolito) configuration of woofer over tweeter over woofer (WTW). Note that this requires a tweeter with 6dB higher voltage sensitivity than the woofers. **Figure 8** shows the performance of a WTW system with ideal drivers all flush-mounted in a flat front panel with 6″ CTC spac-

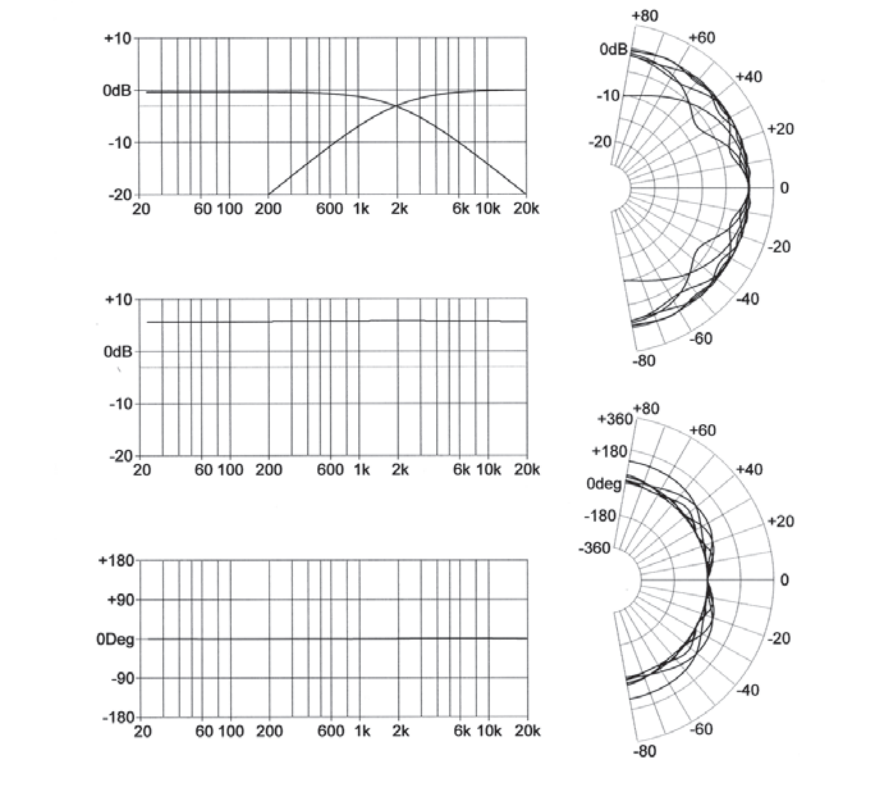

FIGURE 8: Performance of WTW symmetrical system with ideal drivers and first-order crossover.

ing between drivers. The system is linear phase at 0° relative to the tweeter axis and goes bad slowly with vertical angle change. The magnitude response is best at 0° and no lobes exceed the on-axis response. This is a good-looking system if only you could build it.

Figure 9 shows the performance of the WTW system with first-order CO when I install the real drivers. Note this system requires the network to remove the tweeter resonance impedance rise and a stepped front panel (tweeter set back slightly) to achieve the indicated performance. This approach has linear phase at 0° relative to the tweeter axis, which goes bad slowly. It also has less severe magnitude peaks than the nonsymmetrical approach; i.e., reduced lobing problems.

If you plan to build a two-way linear-phase system with a first-order CO, you must look for a good coaxial driver or use the symmetrical WTW approach.

THE VARIOUS "FILL DRIVER" CROSSOVERS

The article by Stokes shows three passive implementations of the "fill driver" approach. There are second-order designs by both Yamanaka and Baekgaard and a third-order design by Baekgaard. I will investigate all these approaches, which require three drivers. **Figure 10** shows responses of the drivers that will be used in all real driver investigations requiring three drivers.

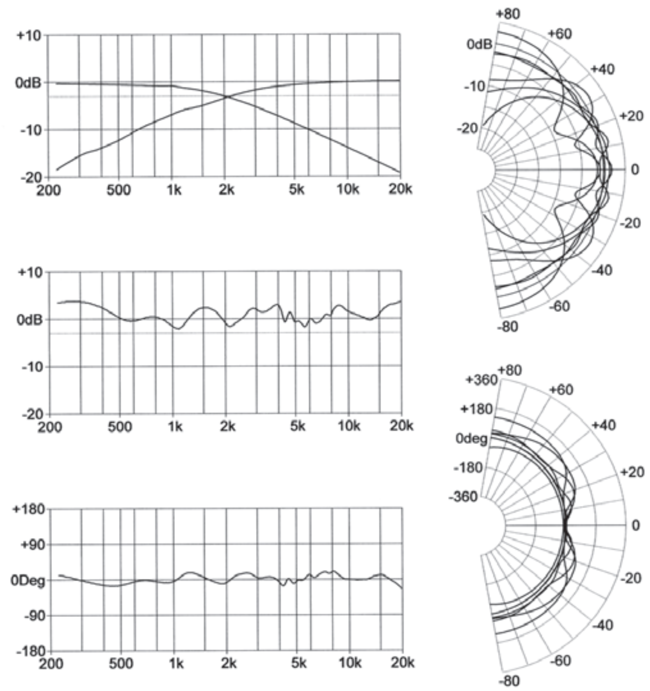

FIGURE 9: Performance of WTW symmetrical system with real drivers and first-order crossover.

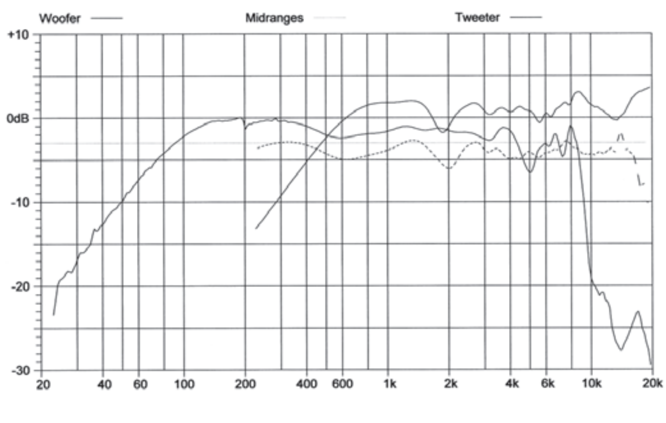

FIGURE 10: Responses of three drivers used in three-driver investigations.

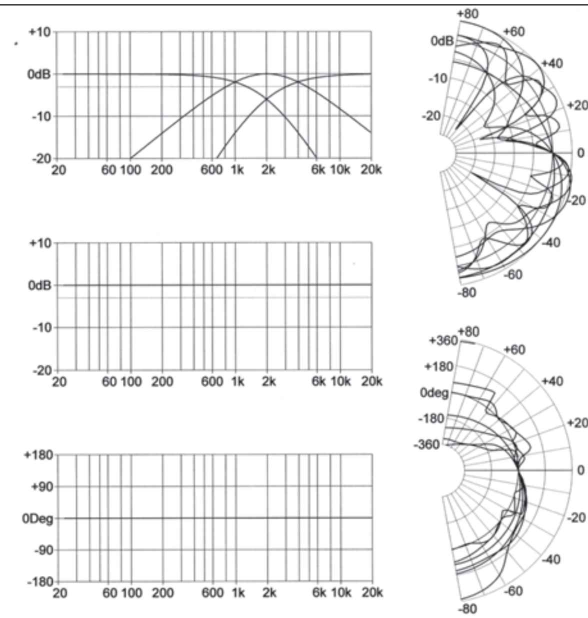

FIGURE 11: Performance of Yamanaka second-order crossover with ideal drivers.

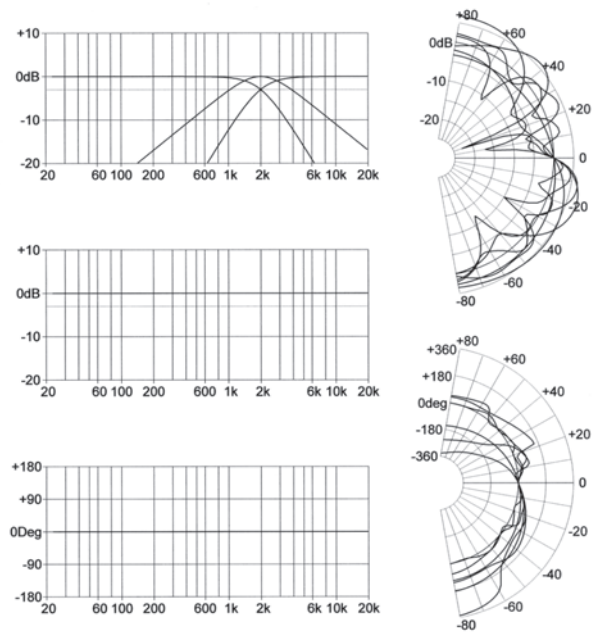

FIGURE 12: Performance of Baekgaard second-order crossover with ideal drivers.

Wideband drivers are identified in **Table 1**. I obtained these drivers from Madisound Speaker Components, who advised me some of them are of limited availability. Note that this tweeter seems very sensitive to its physical environment. Measured in the test baffle, it is very flat, while the curve in **Fig. 10** shows its response mounted in an enclosure with other drivers and grille cloth.

For the fill speaker study, the 6½″ woofer is physically at the bottom, while the 4″ midrange used as the fill speaker is located between the woofer and tweeter. The vertical CTC woofer to midrange spacing is 6″, and the vertical CTC midrange to tweeter spacing is 4.25″. I will use the same spacing when considering ideal drivers.

Figure 11 shows the performance of the second-order Yamanaka CO with ideal drivers. It sums at infinity to 0dB magnitude and 0° phase shift as predicted. The phase shift polar plot shows that it leaves linear phase slowly. However, the magnitude polar plot indicates very bad lobing problems with +5dB lobes off the listening axis.

Figure 12 illustrates the performance for the ideal-driver second-order Baekgaard CO with even worse lobing problems than the Yamanaka approach. **Figure 13** shows the performance with the ideal-driver third-order Baekgaard approach. Note that the fill speaker must have a 6dB higher voltage sensitivity than the other drivers. The lobing problem is even worse, with lobes of about +10dB relative to the 0° listening axis. The performance of these three-driver approaches was so poor that I did not study real drivers.

To give you an idea of what the lobing looks like on a typical speaker design, **Fig. 14** displays the performance of a two-way system with third-order Butterworth CO and ideal drivers. A couple of frequencies near the CO frequency have lobes slightly above the listening axis response, but nothing like what is shown for the fill speaker approaches. In **Fig. 14** both drivers have the same polarity. This CO also works with the tweeter inverted (**Fig. 15**). While neither approach is linear phase, the inverted tweeter has far less total phase shift over the audio band.

I can't recommend any of the fill driver approaches because of their severe lobing problems. Perhaps whoever said if you have three drivers you should build a three-way system was wise after all. You will later see whether a symmetrical array helps with these COs, noting that five drivers are required.

SANDRIK'S MINIMUM-PHASE CROSSOVER

Sandrik's approach is to design a normal first-order CO and then add additional components to increase the rolloff slope at some distance from the actual CO frequency. **Table 2** shows his recommendation of where to place the slope increase based on how much you increase the slope. He recommends that you end up with a final slope of -12dB/octave for the woofer LP, meaning you add an additional first-order CO ahead of the original first-order but designed for twice the CO frequency. For the tweeter HP he recommends a final 24dB/octave slope, meaning you add a third-order CO ahead of the original first-order CO but designed at one-fourth the CO frequency. Sandrik did not show a basic implementation.

I have developed the approach shown in **Figure 16**. For the HP you zobel the tweeter to a resistance and then add the capacitor (C1) for the first-order CO. You then add a series Rt-L1 to "zobel" the capacitor and driver. This results in a resistive load on the pre-CO equal to that of the driver and its zobel. The approach is the same for the woofer LP except you need a series Rw-C2 to compensate for the driver, its zobel, and the first-order CO inductor (L2).

I investigated several versions of Sandrik's approach. Here the ideal woofer and tweeter are located 6″ CTC in a vertical array. **Figure 17** illustrates the performance with both first-order pre-

TABLE 1: DRIVERS USED IN THREE-DRIVER SYSTEMS

Driver	Size	Make and Model*	Notes
Woofer	6½″	Vifa P17WJ00-08	Poly cone mid/bass
Midrange	4″	Vifa MG10SD09-08	Shielded, fiberglass cone
Tweeter	1″	Vifa XT25TG30-04	Dual concentric ring radiator

*The Vifa line is now known as the Peerless V-Line

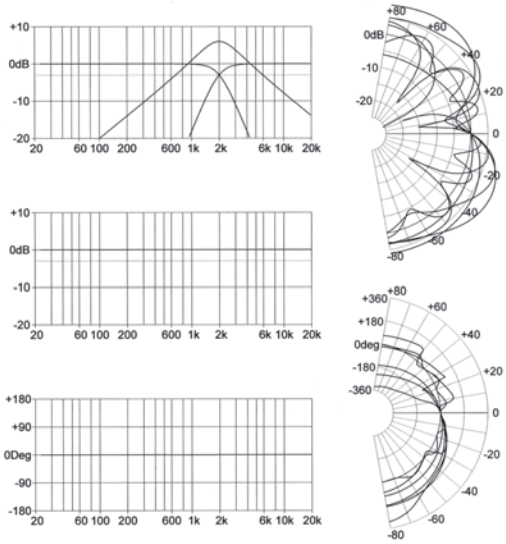

FIGURE 13: Performance of Baekgaard third-order crossover with ideal drivers.

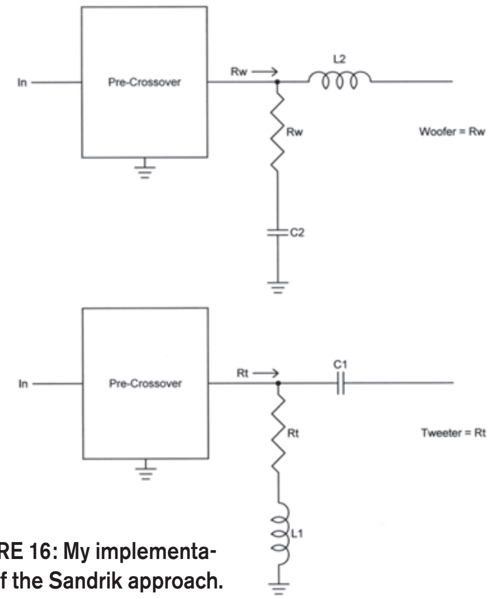

FIGURE 16: My implementation of the Sandrik approach.

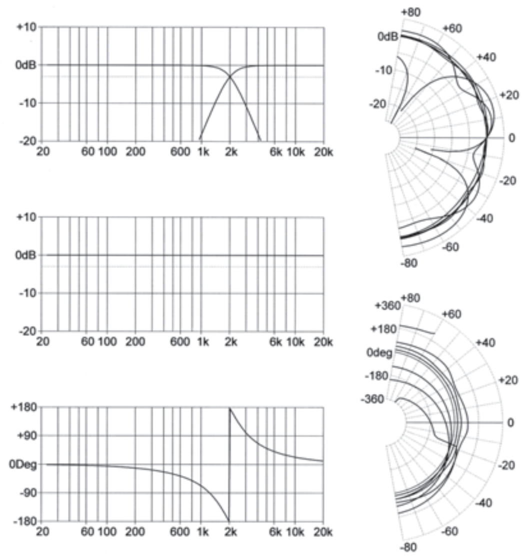

FIGURE 14: Performance of Butterworth third-order crossover with ideal drivers wired with the same polarity.

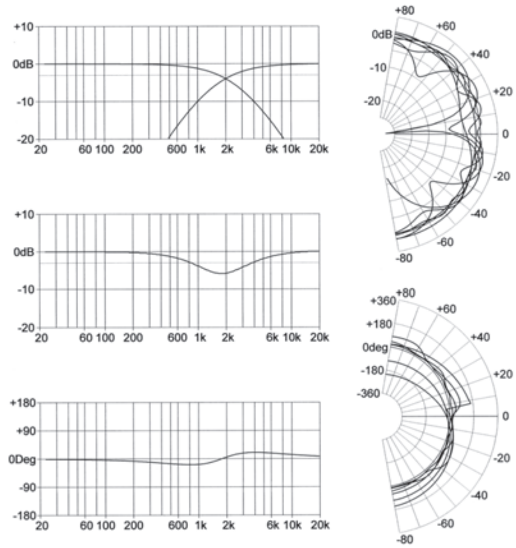

FIGURE 17: Performance of Sandrik approach with both first-order pre-crossover sections and ideal drivers.

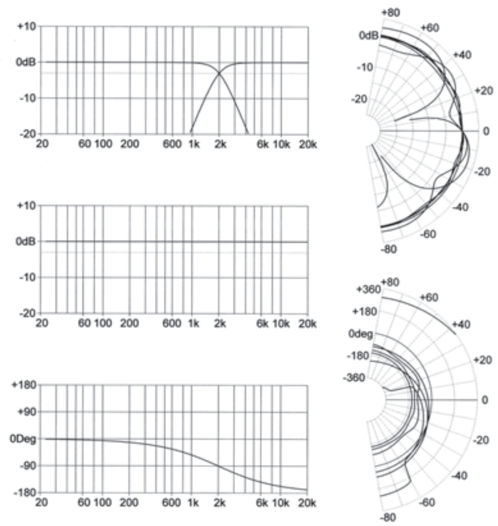

FIGURE 15: Performance of Butterworth third-order crossover with ideal drivers wired with inverted tweeter polarity.

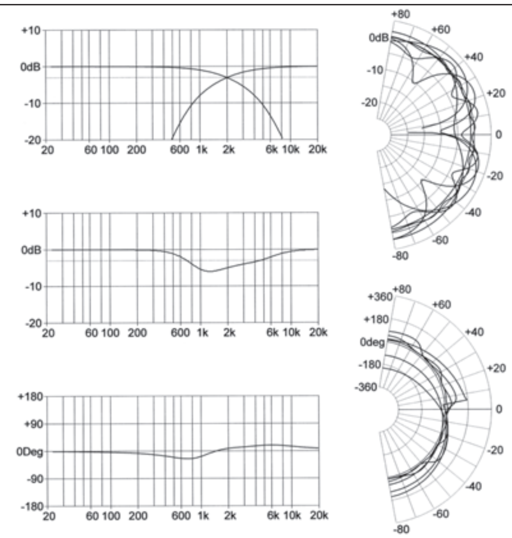

FIGURE 18: Performance of Sandrik approach with both second-order pre-crossover sections and ideal drivers.

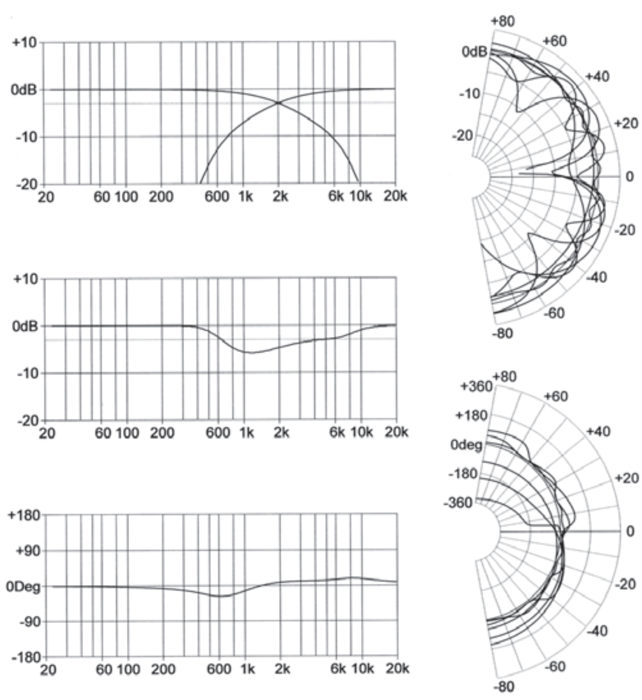

FIGURE 19: Performance of Sandrik approach with both third-order pre-crossover sections and ideal drivers.

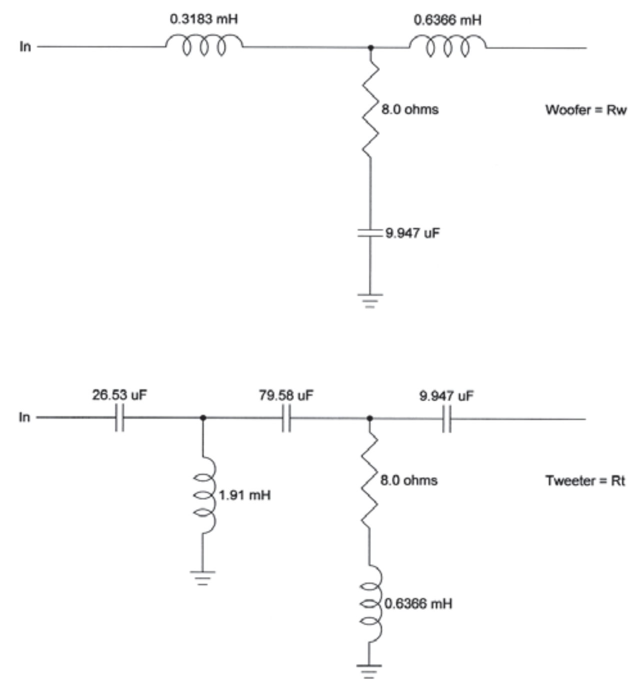

FIGURE 20: Schematic for my implementation of the Sandrik approach with first-order LP pre-crossover, third-order HP pre-crossover, and ideal drivers.

CO sections. The approach is minimum phase, but has a magnitude dip of about 6dB near the CO frequency. The magnitude polar plot shows that just above 0° one frequency nulls badly. **Figure 18** displays both second-order pre-CO sections with about the same result. The same performance is shown in **Fig. 19** for both pre-CO sections third-order.

Figure 20 is the schematic for my approach to Sandrik's recommendation of 12dB/octave final LP slope and +24dB/octave initial HP slope. The component count is becoming rather high. **Figure 21** shows the performance of this configuration with, again, good control of the phase, but still with the magnitude dip and null just above 0°.

Perhaps I do not understand Sandrik's approach, or my implementation is poor. Sandrik states you need to experiment, so I tried playing with acoustic origin

TABLE 2: Sandrik's Recommendations of Where to Add Slope Increase

Final Slope	Frequency for LP Slope Increase	Frequency for HP Slope Increase
±12dB/octave	2 · fco	fco/2
±18 dB/octave	2.828 · fco	fco/2.828
±24 dB/octave	4 · fco	fco/4

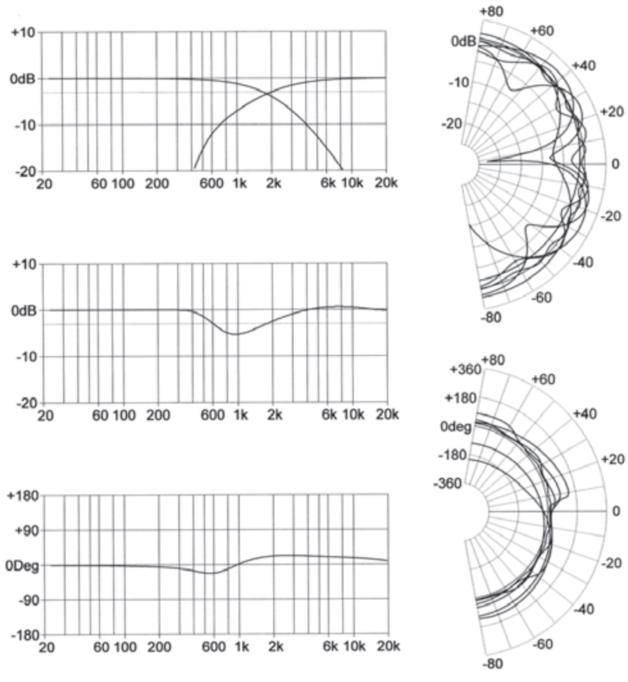

FIGURE 21: Performance of Sandrik approach with first-order LP pre-crossover, third-order HP pre-crossover, and ideal drivers.

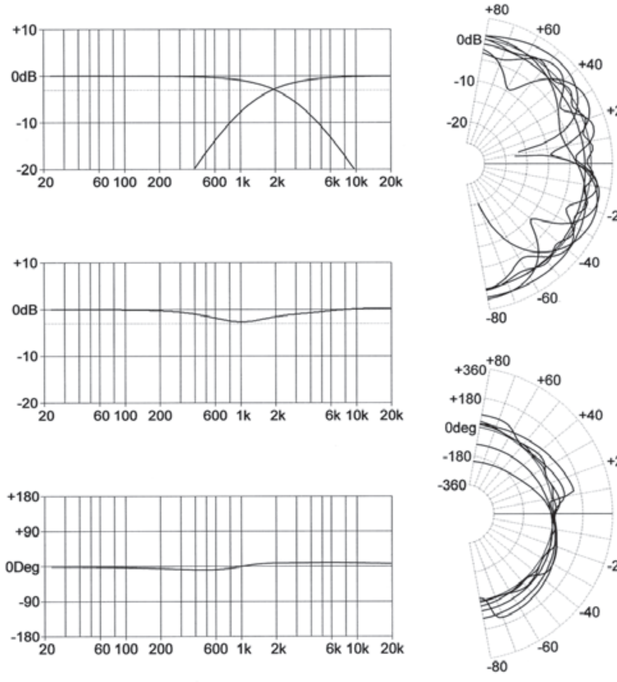

FIGURE 22: Performance of Sandrik approach with both first-order pre-crossover sections, ideal drivers, and 30% overlap.

locations, CO component values, and so on, with not much improvement. What seemed to help was putting some overlap in the original first-order CO sections, but not changing the pre-CO sections. Using 30% overlap (LP designed at 2600Hz, HP at 1400Hz) reduced the dip with both first-order pre-CO sections to under 3dB (**Fig. 22**). Note the lobing performance is not great. Finally, I tried the recommended first-order LP pre-CO and third-order HP pre-CO with the 30% overlap (**Fig. 23**). The magnitude dip is still excessive, and the null just above 0° remains.

I did not have much luck with the Sandrik approach, so I did not bother modeling real drivers. Perhaps if you have drivers that tend to peak near 2kHz (or whatever CO frequency you use), this approach has merit, but be ready to experiment.

Next month the author continues his study of passive CO approaches for building linear-phase systems—Eds. *aX*

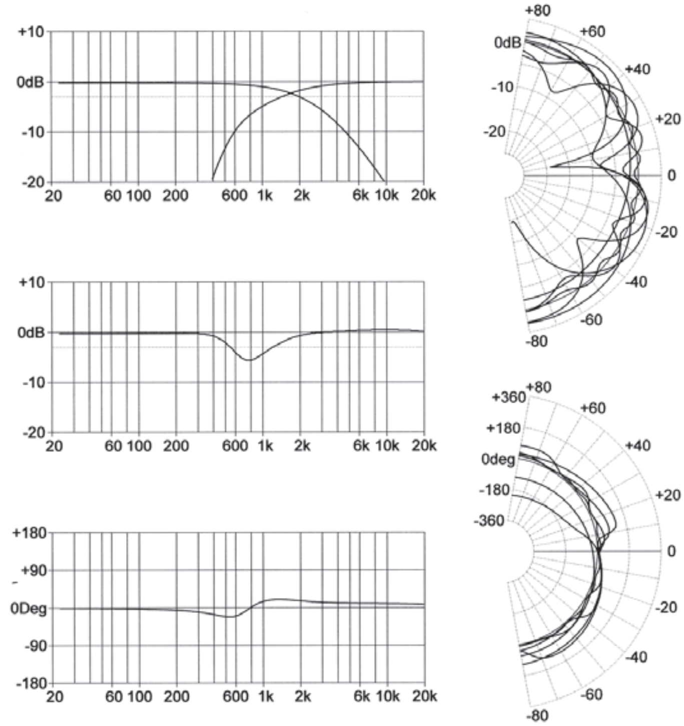

FIGURE 23: Performance of Sandrik approach with first-order LP pre-crossover, third-order HP pre-crossover, ideal drivers, and 30% overlap.

REFERENCES

1. Bill Waslo, "Time, Frequency, Phase, and Delay," *SB* 7/96, p. 12.

2. Many of the references will call the point where the sound appears to radiate from a driver the acoustic center. The currently preferred term is the acoustic origin. For this article you may consider the two terms interchangeable.

3. Bruno Carlsson, "Driver-Offset-Related Phase Shifts in Crossover Design," *SB* 1/95, p. 26, and 2/95, p. 26.

4. E. Baekgaard, "A Novel Approach to Linear Phase Loudspeakers using Passive Crossover Networks," *J*AES (May 1977), pp. 284-294.

5. Steve Stokes, "A Unique Crossover Design with Waveform Fidelity," *aX* 1/07, p. 42.

6. J. F. Conniff, "Baekgaard Revisited," *SB* 2/96, p. 30.

7. G. R. Koonce, "The Baekgaard Crossover Technique," *SB* 2/95, p. 20.

8. T. E. Sandrik, "Minimum Phase, High-Order Filtering," *SB* 4/98, p. 16.

9. John Kreskovsky, "A Transient-Perfect Second-Order Passive Crossover," *aX* 5/01, p. 22.

10. John Kreskovsky, "Designing an Active Transient-Perfect Second-Order Crossover," *aX* 12/02, p. 36.

By Ron Tipton

A Multiple "Wall-Wart" Power Supply

Get organized with this handy device that can reduce the rat's nest of plugs and wires.

A power supply is perhaps not a very glamorous or exciting project, but this one is very useful. It replaces ten DC wall-wart supplies, thereby freeing up nine AC outlets. And this, in itself, is a worthwhile accomplishment.

Each output voltage is isolated from the others so there is no common return or ground. I have operated the prototype unit continuously for over three years with no problems. I chose to make nine of the outputs 24V DC, with the tenth one at 14V. The output voltage is easily settable with two 1% resistors, so you can choose whatever voltages you need (up to about 30V) with the transformers in the parts list. If you set all the outputs to a lower voltage, using a transformer with a lower secondary voltage would minimize power dissipation in the regulators.

The circuit diagram is shown in **Fig. 1**. All ten circuits are identical, with the exception of the voltage set resistor values R3 and R4. The needed values are calculated from:

Vout = 1.25 + (1.25 * (R3 + R4)/240)

For example, for 24V (R3 + R4) = 4368Ω. This is closely approximated by choosing standard 1% resistors of 4120 and 249Ω.

Because there are two secondary windings on each transformer, two supplies share a transformer. I considered using one large circuit board, but on second thought this didn't seem like a practical idea. I settled for two different boards: one for four supplies (two boards needed), and the other single board for

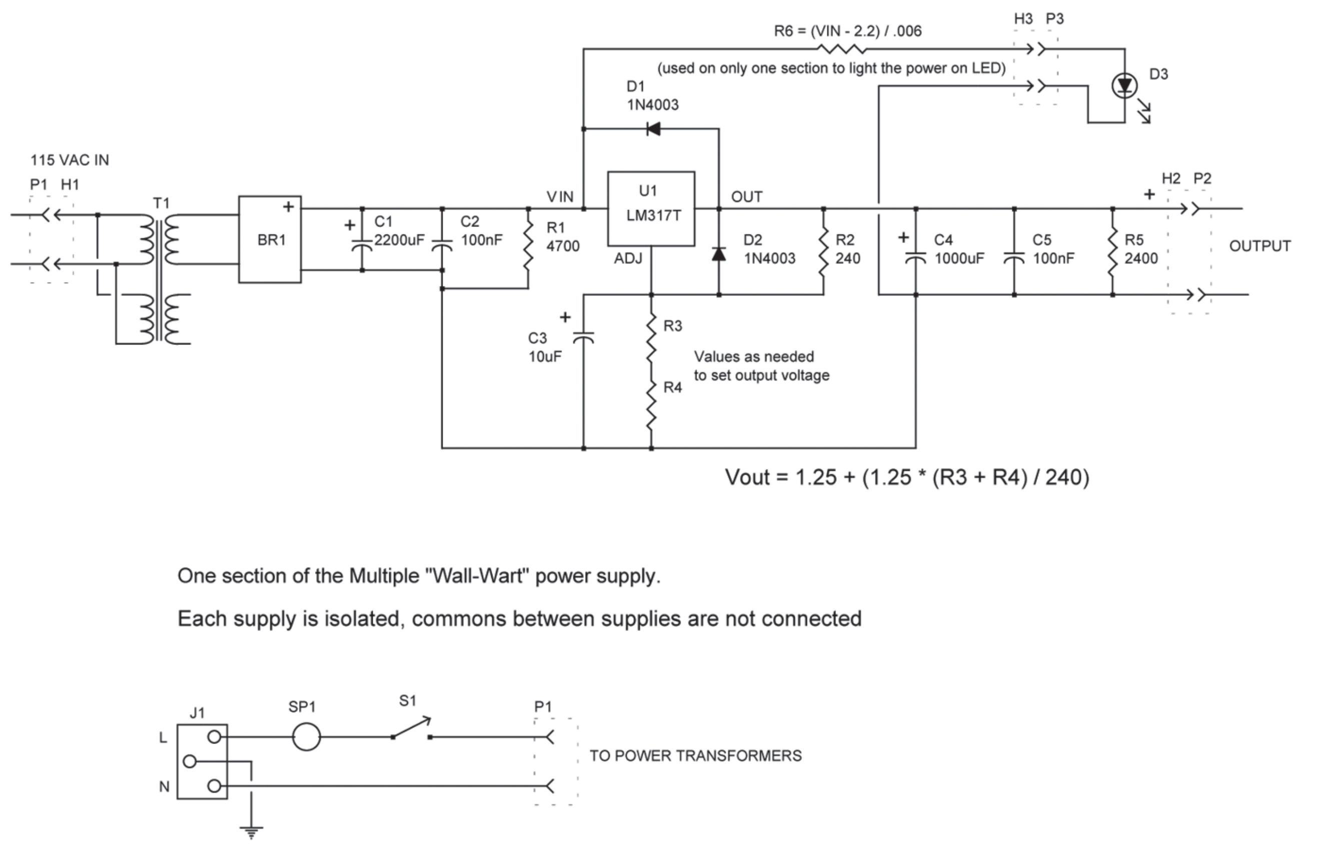

FIGURE 1: Multiple "wall-wart" power supply circuit diagram.

two supplies. The circuit board layouts (single-sided) are shown in **Figs. 2** and **3**. (You can download the layout files from the TDL® website.[1]) I named this supply a Model 701 (just to keep track of circuit boards and parts), but it is not a product, and I have built only the one prototype. Hence, the front panel has no labels.

CONSTRUCTION

The construction is straightforward and not critical. But because the circuit boards almost fill the enclosure, there isn't much room for variation (**Photos 1** and **2**).

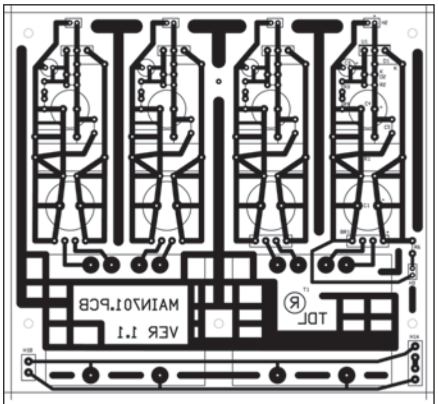

FIGURE 2: Four section supply circuit board. Note that H1B is used to connect AC power from board to board.

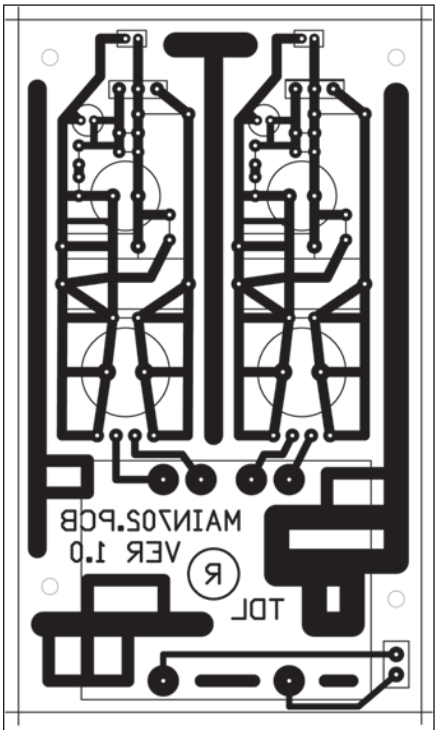

FIGURE 3: Two section supply circuit board.

PHOTO 1: Multiple wall-wart power supply inside view. The inrush current limiter, SP1, is directly wired to the fused, IEC AC power input connector.

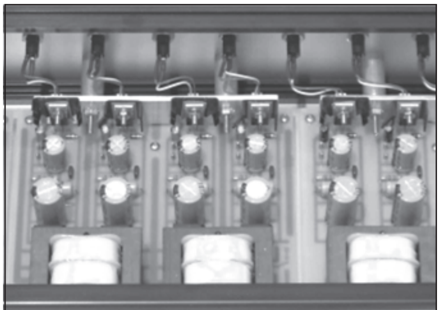

PHOTO 2: Multiple wall-wart power supply inside view. After mounting the circuit boards, measure the distance from the aluminum heatsink bar to the rear panel and cut the ½″ diameter aluminum rod to this length.

PARTS LIST

Reference	value	description	source
R1	4700	5%, 3W carbon film	Xicon (Mouser) 283-4.7K-RC
R2	240	1%, ¼W metal film	Xicon (Mouser) 271-value-RC
R3, R4		1%, ¼W metal film	values set output voltage for example: for 24V out R3 = 4120 and R4 = 249, for 14V out R3 = 2400 and R4 = 47.5
R5	2400	5%, 1W metal oxide	Xicon (Mouser) 286-2.4K-RC
R6		1%, ½W metal film	value depends on input voltage R6 = (Vin – 2.2)/0.006 (medium bright)
C1	2200µF	50V radial electrolytic	Xicon (Mouser) 140-XRL50C2200-RC
C2, C5	100nF	50V polyester film	Panasonic (Digi-Key) P4525
C3	10µF	25V radial electrolytic	Xicon (Mouser) 140-XRL25V10-RC
C4	1000µF	50V radial electrolytic	Xicon (Mouser) 140-XRL50V1000-RC
D1, D2	1N4003	silicon diode	
D3		red LED, panel mount	Lumex (Mouser) SSI-LXR48161D
BR1		bridge rectifier, 2A 100 piv	MCC (Mouser) RS402L-BP
U1	LM317T	adjustable 3-terminal voltage regulator, TO-220	
J1		IEC power connector with built-in line fuse	
H1A		4-pin header, 0.156″	Molex KK-series 26-60-2040
H1B		2-pin header, 0.156″	Molex KK-series 26-60-2020
H2, H3		2-pin header, 0.1″	Molex KK-series 22-03-2021
P1A		(2) 2-pin shell with terminal pins	Molex KK-series 09-50-7021
P2, P3		2-pin shell with terminal pins	Molex KK-series 22-01-2027
SP1	CL40	inrush current limiter (thermistor)	Digi-Key or Mouser
T1		power transformer	depends on output voltage, for example: for 14 and 24V outputs, a good choice is Jameco # 209015 rCR, each secondary is 24V AC at 400mA. If lower output voltages are needed, using a transformer with lower secondary voltages minimizes power dissipation in the regulators.
		inside heatsinks	for example: Avid (Mouser) 530613B00000G
		rear panel heatsinks	for example: Avid (Mouser) 507002B00000G
		enclosure	rack mount 2RU7 (3.5″ high) from Wolgram Engineered Plastics, LLC., 913-491-5414. www.wolgrammfg.com.

The Jameco 209015 is not in their current catalog (#281) but it is on their website, www.jameco.com. The single-quantity price is $2.59

For reliable operation it's important to get the heat out of the regulators and into the air, preferably outside of the enclosure. Each group of regulators (either two or four) share a common 1″ wide × 1/8″ thick aluminum bar, which is thermally connected to the rear panel through a length of ½″ aluminum rod drilled to pass a 6-32 machine screw. As you can see from the photos, each regulator has its own small finned aluminum heat dissipater between the regulator and the aluminum bar. These may not be necessary because the supply operates very cool. However, I prefer to err on the side of "overkill."

The rear panel (**Photo 3**) also has a finned dissipater held in place by each of the five machine screws passing through the aluminum rods. (The regulator mounting tabs are not isolated, so you also need a silicon rubber insulator between each tab and the aluminum bar as well as insulating washers.)

It is important not to omit the inrush current limiter, SP1, because all five transformers are powered on when S1 is closed and must charge a total of 22,000µF (ten times the value of C1) at turn-on. I used

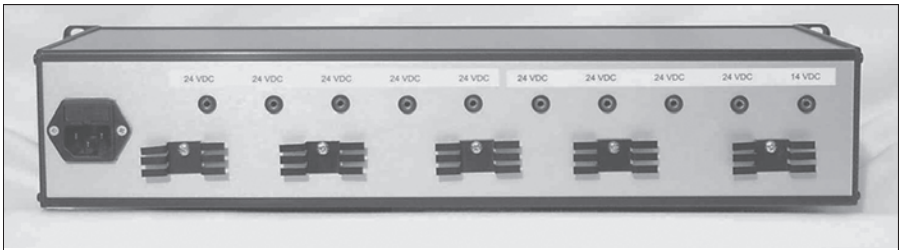

PHOTO 3: Multiple wall-wart power supply rear panel. During continuous operation, the panel is only slightly warm to the touch.

PHOTO 4: Multiple Wall-Wart power supply front panel.

3.5mm insulated mono phone jacks for output connectors because they provide a positive, locking connection and have sufficient current capacity. You could also use two-pin DIN or mini-DIN as well as ¼″ insulated mono phone jacks. The circuit boards are attached to the enclosure bottom panel using ½″ long nylon spacers and 4-40 × ¾″ screws and 4-40 nuts.

If all the output voltages are the same, no rear panel labels would necessarily be needed. But I thought it would be useful to keep track of which one was 14V. I used a word processor to print two rows of voltages on an 8½ × 11″ sheet of sticky-back label, cut them into ½″ wide strips, and stuck them to the rear panel over the output connectors. If you are going to build one of these with several different output voltages, it might also be helpful to use a different type output connector for each voltage.

FINAL WORDS

This supply is better than the wall-warts it replaces because all the outputs are regulated and have much less hum and noise. I would build another one if needed, but so far the one prototype has been sufficient, although there are times when all ten outputs are in use.

REFERENCE

1. Download wallwart.zip from the downloads page at www.tdl-tech.com. They are in .pcb format and can be printed with a free version of CIRCAD from www.holophase.com. *aX*

Old Colony Sound Lab often makes available printed circuit boards in support of magazine projects if there is sufficient reader interest. To indicate your interest level in PCBs for this power supply project, contact Old Colony Sound Lab at 603-924-9464, 603-924-9467 (fax), or custserv@audioXpress.com (e-mail). Availability decisions are usually made in about two-three months—watch Old Colony ads for further developments.

 By Gary Galo, Regular Contributor

Is SACD Doomed?

New York AES 2007: A Convention Notable for What Wasn't There

The Audio Engineering Society held its annual US convention at the Javits Convention Center in New York City over the long Columbus Day weekend, Oct. 5-8, 2007. Unlike many previous AES conventions, this one didn't promote any particular theme, with the program cover simply noting "Where Audio Comes Alive." As usual, the program was huge and diverse, way too much for one person to canvas thoroughly. Like most attendees, I focused on presentations and exhibits of particular interest to me and, I hope, readers of this magazine.

SACD SUPPORT?

In the September 1999 version of the New York AES, Sony introduced the Super Audio Compact Disc (SACD) with considerable fanfare. A special five-channel demonstration room was fitted with Sony's top-of-the-line SACD player, five Sony SS-M9ED loudspeakers (at 16 grand a pair), each powered by a Pass X600 class-A mono amplifier. In my guest editorial, "The High End at AES," I declared the demonstration to be ". . . without a doubt, the finest sound I have ever heard at an AES convention" (*Audio Electronics*, 2/2000, p. 8).

Sony also gave a special presentation in a large room filled with several hundred attendees. Panelists at this presentation included the well-respected recording engineer Tom Jung, President of DMP Records, who offered a most enthusiastic endorsement of the SACD and its Direct Stream Digital recording system. Jung noted that, in his opinion, the SACD system yields recordings closer to the direct feed from the mixing console than anything he had previously heard. At three subsequent New York AES conventions—2001, 2003, and 2005—Sony's flagship SACD players were featured prominently in their exhibit.

The October 2007 AES convention was notable for what *wasn't* there. Namely, any trace of Direct Stream Digital or the SACD in the Sony exhibit area (Sony didn't even have a demonstration room this year). Indeed, it was difficult to find anyone promoting high-resolution digital audio. I asked a Sony representative whether the company had given up on SACD. He said that while Sony had not dropped its support for the format, there has been no clear support in the industry for a replacement for the conventional CD. Sadly, the Sony rep is correct.

The industry has simply followed the demands of the marketplace and, with most music lovers delighted with the sound of Apple's iPod and other compressed digital formats, interest in DVD-Audio and SACD is confined to a fringe corner of the audio marketplace. Granted, the iPod doesn't need to contain compressed digital audio, but that's how the vast majority of Apple's enthusiastic customers use it. Indeed, papers on various coding schemes for Internet and portable audio occupy a significant place on AES programs.

The rejection of high-resolution digital audio by the mainstream audio community is due to one of two things. Either most people who listen to music can't hear the difference between the CD and high-resolution digital, or the difference is not significant enough to matter. Convenience is often more important than sound quality, which certainly accounts for the success of various portable music formats, from the analog cassette to MP3 players and the iPod ("I have 12,000 tunes on my iPod—why should I buy SACDs?").

The inability of many to perceive the differences between the CD and high resolution formats was documented in an article recently published in the *AES Journal*, "Audibility of a CD-Standard A/D/A Loop Inserted into a High-Resolution Audio Playback."[1] Authors E. Brad Meyer and David R. Moran, both members of the Boston Audio Society (as well as AES), conducted a series of double-blind listening tests over the course of a year using an A/B/X box. These tests compared the direct output of an SACD player to an A/D/A conversion at the 16-bit/44.1kHz CD standard. In other words, the analog output of the SACD player was fed through the CD-standard conversion, and an A/B/X box was used to switch between the direct output from the SACD player and the output from the CD-standard converters. Out of 554 trials, there were 276 correct answers—no better than "coin flip" results. To those of us who have recorded in both high-res and CD-standard digital, the conclusions seem ridiculous.

Several recording engineers and audio professionals I spoke with found the conclusions baffling—either the participants can't hear, or there's something seriously wrong with the test. I know a number of people who work in the audio field and seem to fit the "can't hear" category. One recently told me that he did not believe that the Sony PCM-F1 digital recorder had any audible shortcomings. In other words, digital audio today doesn't *sound* any better than it did in 1981.

In their conclusion, Meyer and Moran admit that "virtually all" of the SACDs and DVD-Audio recordings they heard sounded better than most CDs. But, they don't believe this can be attributed to the inherent superiority of high-res digital formats. From discussions they've had with engineers who work on commercial releases in high-res formats, they conclude that the extra care taken in the production of SACD and DVD-Audio releases accounts for their superior sound (I'd be willing to bet that much of this "extra care" involves things that A/B/X testing would fail to substantiate).

But, this doesn't account for the proliferation of dual-layer SACDs in which the CD layer is identical in production to the SACD layer, except for a conversion from DSD to 16-bit/44.1kHz PCM. Such SACD discs include the RCA Victor Living Stereo series and the Telarc SACD catalog (the Mercury SACDs are an exception, because the CD layers are Wilma Cozart Fine's older CD transfers, and are thus different from the newly transferred SACD layers).

On my own equipment, the differences between the SACD and CD layers are readily audible. Yet the conclusion of Meyer and Moran is that "Our test results indicate that all of these recordings could be released on conventional CDs with no audible difference." In their view, "Further claims that careful 16/44.1 encoding audibly degrades high resolution signals must be supported by properly controlled double-blind tests."

Apparently they believe that recording engineers such as Tom Jung are simply deluding themselves. Perhaps we also need a series of A/B/X tests to verify whether there are any visible differences between HDTV and NTSC.

DIGITAL DOMAIN

One manufacturer who was demonstrating DSD recordings was ATC Loudspeakers (www.lasvegasproaudio.com). Their SCM150ASL 3-way active loudspeaker was used in a 5-channel surround system with a full-range center channel. Digital hardware included DSD playback equipment from EMM Labs, a company run by renowned digital hardware designer Ed Meitner (www.emmlabs.com). The jazz recordings featured in this demonstration were made on a Sonoma DSD Recording and Editing Workstation, using Meitner-designed DSD converters. Unlike some editing systems used in the production of SACDs, Sonoma's "DSD Pure" system keeps the audio in the DSD domain at all times (**Photo 1**, www.superaudiocenter.com). Although the recordings were impressive enough, I don't think the playback system revealed their full potential.

In Sonoma's product literature, Tom Jung offers the following endorsement: "The Sonoma system, when optically interfaced with Ed Meitner's DSD converters, is the most musically accurate recording and editing system available today at any price." Sonoma boasts endorsements from several other renowned engineers and producers, including Telarc engineer Michael Bishop, who states: "I've recorded to analog tape for well over 30 years and to digital nearly as long. The Sonoma 32-track DSD workstation gives me the quality of analog—without the drawbacks—plus the convenience and speed of a DAW."

Two manufacturers were promoting Digital eXtreme Definition, or DXD, converters. DXD is a 24-bit PCM system operating at a sampling frequency of 352.8kHz, which claims to offer the resolution of DSD with the ease of editing of PCM systems. One of the disadvantages of native DSD editing is the limited signal processing capabilities compared to PCM editing systems. With the DXD system, you can make a recording in DSD format, convert it to DXD for editing on a DXD workstation, and then convert it back to DSD to make the finished SACD master. Or, you can record the original in DXD, edit it, and convert it to DSD for the finished SACD product.

Digital Audio Denmark (DAD, www.digitalaudio.dk) exhibited its AX24 multi-channel converter, noting that "24-bit DXD at 352.8kHz is the perfect work format when producing SACDs (**Photo 2**). It offers the editing and processing advantages known from PCM and converts smoothly to DSD." Digital Audio Denmark further notes that the DXD format is supported by Merging Technology's Pyramix 5.0 workstation.

Merging Technology's exhibit featured the Pyramix workstation (**Photo 3**), now in version 5.1, along with its own Sphynx 2 DXD multi-channel converters (www.merging.com). The Sphynx 2 was jointly developed with Digital Audio Denmark, and appears to be virtually identical to the DAD AX24. On their website, Merging Technologies makes the following case for the Pyramix system in DSD/SACD production:

"For those who want to work in the DSD domain at 2.82MHz sampling rates for SACD production, Pyramix has a unique answer. . . It is the first system in the world to offer multi-track record/editing and mixing as well as mastering while maintaining all the real-time audio processing (effects) such as EQ, Dynamics, Reverb and Surround Sound in a DSD compatible quality level up to the final SACD master. Quite simply, to have only a mastering capability for DSD is not enough. Unless you can record, edit and process the signals in a true multi-track configuration, mastering makes little sense.

"Pyramix is the first system with enough processing power and resolution to handle in real time a complete digital mixing console capable of operating in 32 bit floating point at 352.8kHz, which is the minimum requirement to preserve all the intrinsic original quality of 1 bit 2.8MHz DSD signals. With real-time EQ, Dynamics, Reverb, etc., and with full SACD scarlet book specification for mastering, we can truly state that 'Pyramix is the only commercially available system on the market that is capable of a complete multi-track source to master project for Super Audio CD production.' "

Regarding the DXD format, they note: "A new format, DXD (Digital eXtreme Definition for high quality and low noise recording and editing for SACD), has recently been acknowledged by Philips and Sony. DXD was initially developed for Merging's Pyramix DSD workstation and recognized as one of the best formats for DSD source recording."

The Sphynx 2/AX24 converter was used on an excellent SACD of Mozart Violin Concerti Nos. 3, 4, and 5, performed by Marianne Thorsen with the Trondheimsolistene led by principal cellist Øyvind Gimse, on the Norwegian 2L label (www.2l.no). Digital Audio Denmark handed out a sample copy of this recording containing two discs with identical programs, an SACD and a conventional CD, thus making it easy for the listener to compare the two formats without going into the SACD player's setup menu. Levels between the two discs appear to have been precisely matched.

To my ears, the SACD is superior to the CD—more spacious, with a silky-smooth treble region that becomes slightly grainy on the CD. The CD also sounds as though it were recorded in a slightly drier venue than the SACD. A single-disc, dual-layer version of this recording is available from a number of Internet dealers, including ArkivMusic (www.arkivmusic.com/classical/album.jsp?album_id=146118). The SACD version makes a strong case for the transparency of the DXD conversion process, though I admittedly have no way of hearing the original, prior to DSD conversion.

The audio community is fortunate that there are still a few manufacturers committed to advancing the state-of-the-art in audio recording. Those of us who care about high-quality sound can only hope that room will remain in the marketplace for such companies. They face an uphill battle against those who claim that it makes no difference, and those who simply don't care.

Keith O. Johnson, the recording engineer responsible for the material on the Reference Recordings label, gave a master class titled "The Art and Science of Making and Playing Great Recordings—The High Resolution Experience." Johnson is also the co-designer of the HDCD system, which Reference Recordings is still promoting. They have not issued any SACD or DVD-Audio discs, but their website notes: "We record two-channel masters at 176.4kHz, 24-bits, and discrete five-channel masters at 88.2kHz, 24-bits for future release in high-resolution surround sound" (www.referencerecordings.com).

PHOTO 1: Sonoma's DSD-Pure multi-track recording and editing system operates entirely in the DSD domain. Leading recording engineers and producers have proclaimed it the most transparent recording and editing system available. (Courtesy of Super Audio Center, LLC)

PHOTO 2: Digital Audio Denmark's state-of-the-art AX24 D/A converter offers 24-bit DXD at 352.8kHz, as well as DSD. (Courtesy of Digital Audio Denmark)

The large room provided for this 90-minute presentation contained a multichannel surround system with Parasound Halo-series electronics driving PMC BB5 loudspeakers that appeared to be quite capable of providing high-definition sound in a large venue (PMC is a UK-based company specializing in high-resolution professional loudspeakers; www.pmc-speakers.com). Unfortunately, Johnson's presentation was not well timed. He spent too much time at the beginning discussing the hearing mechanism, and left little time at the end for playback of recordings. This was unfortunate because his recording of the Rachmaninoff *Symphonic Dances*—what little of it we heard—was very impressive. His presentation would have been more meaningful if he had interspersed recorded examples throughout his talk, to illustrate the points he was making along the way.

THAT SEMINAR

THAT Corporation gave an excellent seminar titled "Analog Secrets Your Mother Never Told You," a presentation by four representatives of the company, each focusing on a specific product line and application. All of the presentations were excellent, but I was particularly interested in the one given by Gary Hebert, Chief Technology Officer (gkh@thatcorp.com), titled "Balanced Outputs." Hebert described THAT's OutSmarts® 1606 and 1646 balanced line drivers.

In 2000, at the 108th AES convention, Hebert presented the paper "An Improved Balanced, Floating Output Driver IC." As stated in the paper abstract, "The design and implementation of an improved balanced, floating output driver IC for professional audio applications is described. It is shown that when the most common existing designs are used to drive ground-referred loads, the grounded output is forced into current limiting whenever the active output clips. This results in large current spikes flowing into the ground of the receiving device. Techniques used to eliminate this problem as well as the overall performance of the resulting design are described." The OutSmarts devices are based on the concepts described in that paper, and are an improvement over conventional cross-coupled balanced line drivers.

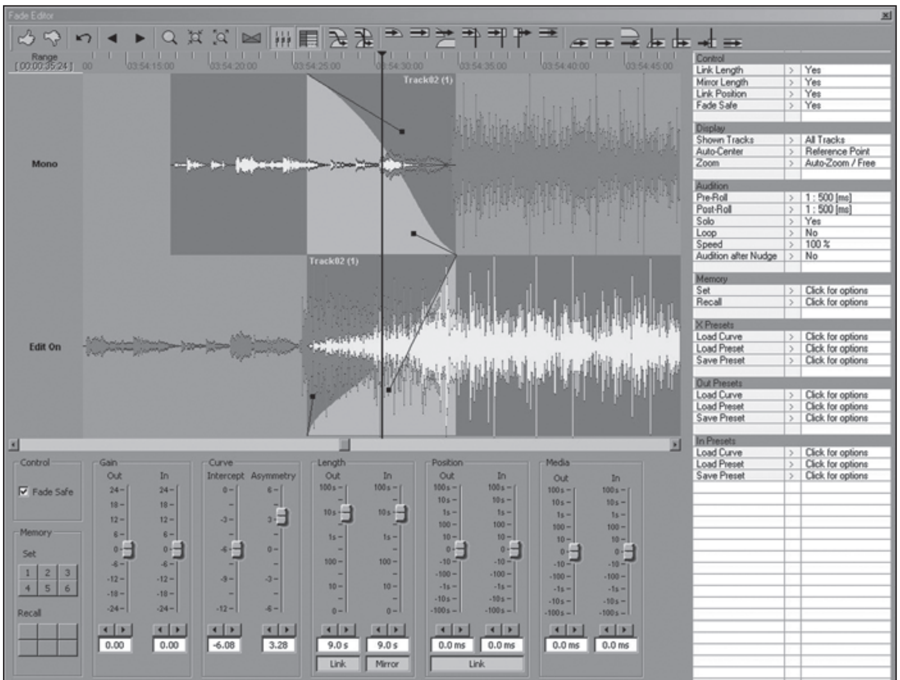

PHOTO 3: The Pyramix editing system operates in 24-bit/382kHz DXD mode, and offers exceptional flexibility and performance. All editing operations, including cross-fades, are 100% real-time—no rendering time is lost. The Pyramix system also includes a multi-track recording system with 32-bit floating/382kHz performance. (Courtesy of Merging Technologies)

Balanced line drivers are often connected to unbalanced loads, with pin 2 on the XLR connector used as signal, and pin 3—the "negative" audio leg—connected to ground pin 1. Traditional designs can lose control over output current if clipped when one output is grounded. Under such conditions, common-mode feedback is lost. Output current in the grounded leg increases to current limit, which can lead to distorted crosstalk. The OutSmarts common-mode feedback loop maintains control, with no current limiting. The OutSmarts devices are also less sensitive to PCB layouts.

Hebert's presentation focused on measurements that show how the 1606 and 1646 line drivers exhibit much better behavior than their competitors' devices when driven into clipping while feeding single-ended loads. Hebert demonstrated the superior performance of the OutSmarts line drivers using the test setup shown in **Fig. 1**. The test rig allows easy comparison of four balanced line drivers, a cross-coupled circuit using "discrete" 5534 op amps, an Analog Devices' SSM2142, a TI/Burr-Brown DRV134, and a THAT 1646. The SSM2142 and DRV134 are cross-coupled designs in a single, self-contained IC, and have been widely used in the pro-audio industry. All of the devices were driven into clipping into a 10k, single-ended load (one of the outputs is grounded while the other drives the 10k load).

The current sensors in **Fig. 1** monitor the output current of the ± outputs of the line driver, though the current sensor for the ungrounded output was not used for these tests. The test setup has two outputs that can be monitored on an oscilloscope: the voltage output of the active, non-grounded output, and the current output of the grounded output. In **Figs. 2-4** the left trace is the output voltage of the active side and the right is the current output of the grounded side. Scope calibration is 5V/div for the left trace and 20mA/div for the right.

Hebert notes: "The current into ground in the case of the 1646 is an inverted replica of the output voltage (**Fig. 2**). In the case of the other parts (**Figs. 3** and **4**), as soon as the output voltage clips, the output current driven into ground increases to the current limit of the device (50-60mA in the case of these devices). Depending on where the output pin is grounded, these currents may contaminate other circuitry leading to crosstalk of a very nasty sort as they return to the supply pins of the device."

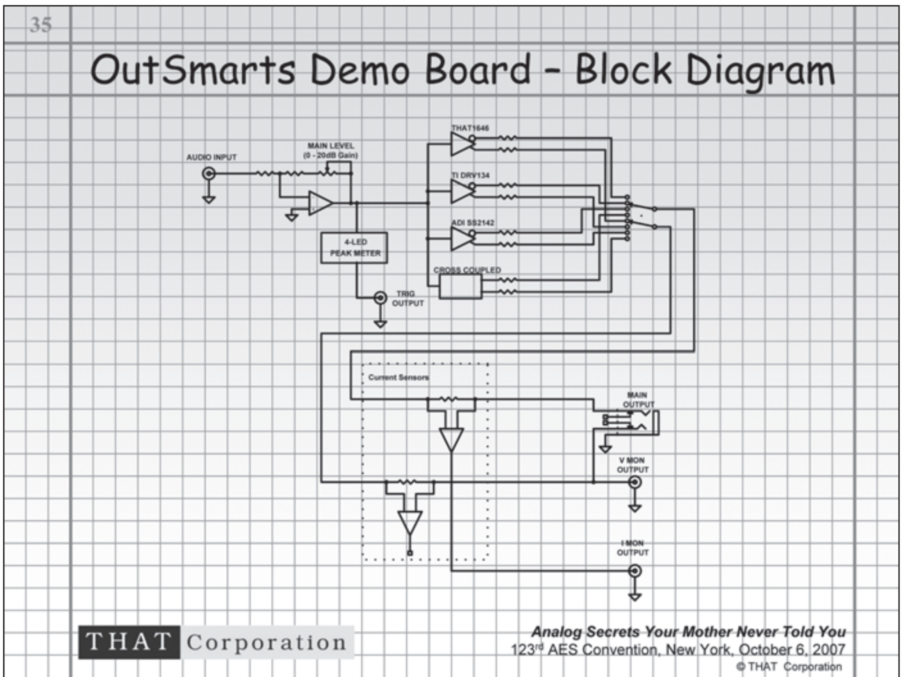

FIGURE 1: Demo board used by THAT Corporation to compare the performance of various balanced line drivers. The board's output allows monitoring of voltage in the active output and current in the grounded output. (Courtesy of THAT Corporation)

The performance of the "discrete" cross-coupled circuit using 5534 op amps shows instability similar to the SSM2142 (**Fig. 3**). Hebert offered the following comment for this report: "This behavior is not uncommon in op amps when they are in current limiting. Often there's a need to enclose more circuitry in the feedback loop created by the current limit circuitry for one polarity than the other. This makes the stability of such loops tricky. Since current limiting is not 'normal operation,' this is usually considered acceptable as long as the part is protected. In the case of the floating, balanced output driver driving a single-ended load, the cross-coupled op amp topology goes into current limit every time it's clipped. In audio, clipping can occur a lot in 'normal operation.'"

The OutSmarts line drivers were designed to imitate some of the more important characteristics of output transformers, including high common-mode output impedance (several kΩ) and low differential output impedance. Feedback minimizes common-mode output current (Iout+ = Iout-). The output appears across two output terminals even if one is grounded. In fact, THAT recommends grounding one output whenever unbalanced loads are driven. Doing so yields lower noise and distortion, without stability issues. Like the InGenius® line receivers, the OutSmarts devices use bipolar rather than JFET technology.

Hebert also reinforced a few points made by Bill Whitlock in his 2005 AES presentation *Grounding and System Interfacing*, which I reported on the Jan. 2007 issue of *aX*3. Most important is that common-mode rejection is determined by impedance matching, and has nothing to do with signal balance in the two conductors of a balanced line. Signal balance affects headroom, and may even affect crosstalk in multi-pair cables, but it does not affect common-mode rejection. Trimming of differential line drivers to obtain matched output impedances, and hence the best CMRR, is complex.

THAT's 1606 and 1646 devices have been internally trimmed, eliminating the need for user adjustments. When used with their InGenius differential input receivers, optimum CMRR is easily achieved, along with excellent signal symmetry. The OutSmarts line drivers are clearly a significant contribution to the analog audio IC market.

Other presentations given at THAT's seminar are also worth mentioning. Bob Moses, IC Program Manager for THAT (rwm@thatcorp.com), presented "New ICs," which include the 2162 dual VCA and the 1280-series of dual line receivers. The 2162 devices are "pre-trimmed" and feature two completely independent VCAs in a QSOP-16 package. The 1280-series devices are manufactured in three gain versions, and are pin-compatible with the TI/Burr-Brown INA2134 and INA2137 devices. They are clearly intended for applications in which the cost of the InGenius products is prohibitive.

Rosalfonso "Ros" Bortoni, Applications Engineer (rb@thatcorp.com), presented "Mic Preamps," focusing on THAT's 1510-series of mike preamp ICs. Bortoni revealed a method for eliminating pops caused by fluctuating DC offsets when the gain is changed, and also showed an implementation with DC servo control. He also illustrated a method for implementing digital gain control using CMOS switches.

Leslie B. Tyler, President of THAT (lbt@thatcorp.com), presented "VCA/RMS & Log Math," which showed designs of Basic Voltage Controlled Amplifiers (VCAs), RMS Detectors, and Analog Engines® using THAT's 2181, 2252, and 4315 chips. He also discussed "Cool 'log math' " to simplify designs using these chips. The excellent Power

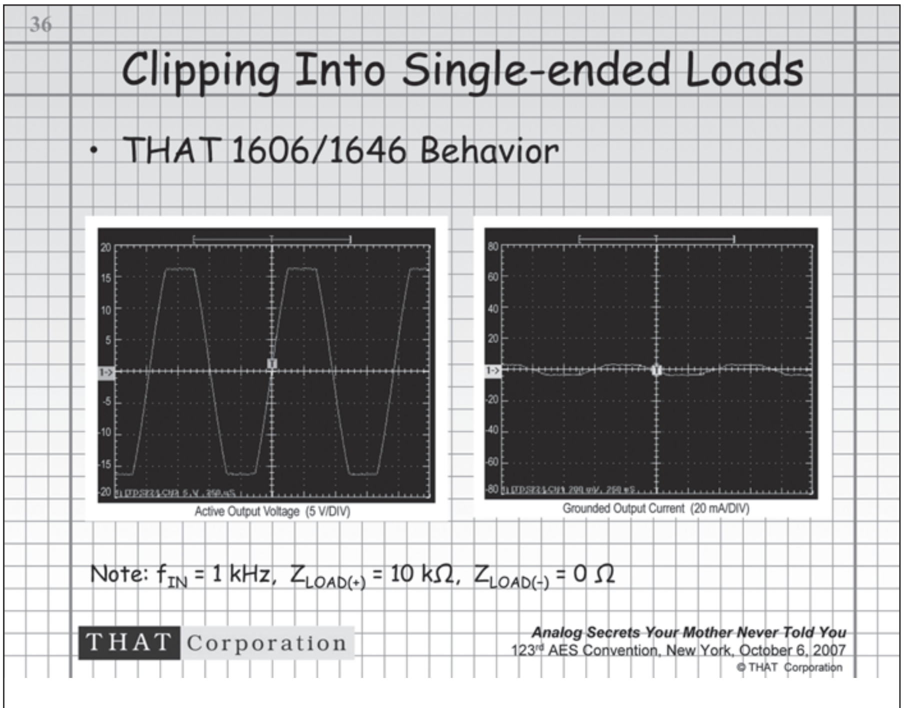

FIGURE 2: THAT 1606 and 1646 balanced line drivers clipping into grounded, single-ended 10k loads. The current into ground, shown on the right, is an inverted replica of the output voltage, and shows no signs of instability. (Courtesy of THAT Corporation)

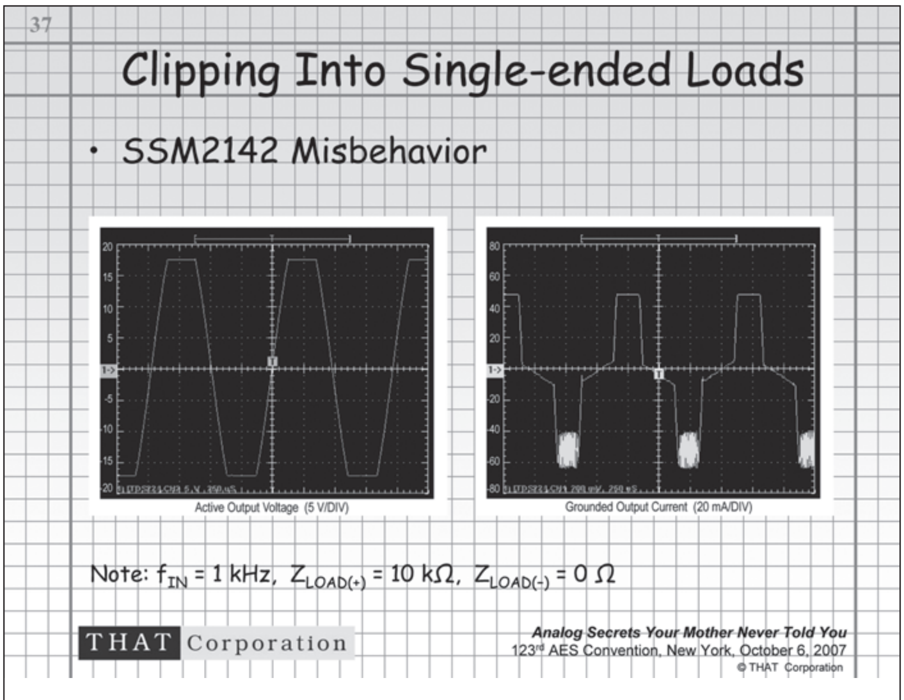

FIGURE 3: The well-known SSM2142 cross-coupled balanced line drivers. The output current driven into ground increases to the current limit of the device. (Courtesy of THAT Corporation)

Point presentation that accompanied this seminar is available under "Tradeshow News" on THAT's website, www.thatcorp.com. **Figures 2-4** in this report are taken from that slide show, and I thank Denise M. Waterhouse, Marketing Manager (dmw@thatcorp.com) for granting permission to reproduce them here. Thanks also to Gary Hebert for clarifying a few points while this article was in preparation.

THAT's seminar was one of the most worthwhile sessions I've attended at AES. Feel free to contact the various presenters at the e-mail address given—I have found the entire staff at THAT to be helpful and enthusiastic in answering questions about products and applications.

NATIONAL CHIPS

National Semiconductor also offered several audio design seminars. Unfortunately, I seemed to be tied up with other convention activities during the times they were given, but a visit to their booth was well worth my time. National seems to have increased their commitment to audio applications and high-performance audio op amps in recent years, and I urge anyone involved with analog design to visit their website (www.national.com), which has an enormous assortment of product data and application notes. In particular, National's LME 49000 series of bipolar-input op amps boasts THD of 0.00003% driving 600Ω loads. National notes that the LME 49870 op amp has been "optimized for superior audio fidelity" and features a slew rate of ±20V/µs, a gain-bandwidth product of 55MHz, and a low input noise density of 2.7nV/√Hz. The SOIC-packaged chip also can operate over a very wide range of supply voltages, from ±2.5 to ±22V.

The op amp can also be driven to within 1V of either supply rail when driving 2k loads or higher; 1.4V for 600Ω loads. Unlike many low-noise op amps suitable for high-gain applications, the LME49870 is unity-gain stable. A dual version, the LME49860, is also available in both SOIC and DIP packages. One suggested application is a passively-equalized RIAA phono preamp, shown on the front page of the 49860 datasheet (**Fig. 5**). The 49000 series op amps are well worth investigating.

Texas Instruments was promoting their own impressive lineup of high-performance ICs for professional audio use, including those manufactured under the Burr-Brown name (www.ti.com). These included some of the best op amps available for audio applications, as well as A/D and D/A converter chips operating at 24-bit resolution and sampling frequencies of 192kHz, including the PCM1792/4. Their *Audio Solutions Guide*, which they were distributing in printed form at the convention, is available in .PDF form on their website, as are numerous other documents of interest to audio builders and designers. *aX*

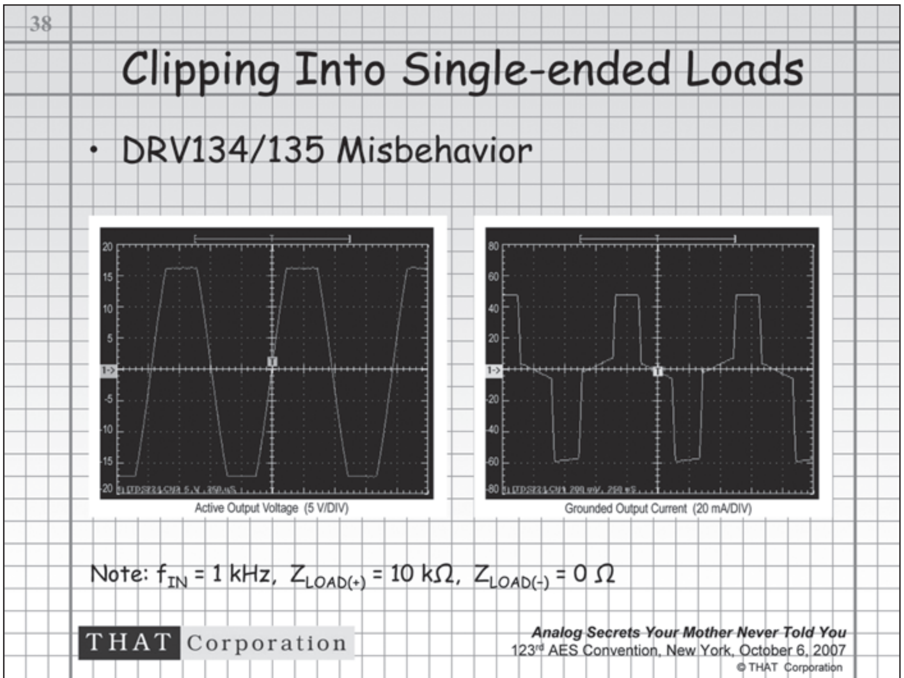

FIGURE 4: The DRV134/135 cross-coupled balanced line drivers. They exhibit the same ground-current problem as the SSM2142, though without the instability seen in Fig. 3. (Courtesy of THAT Corporation)

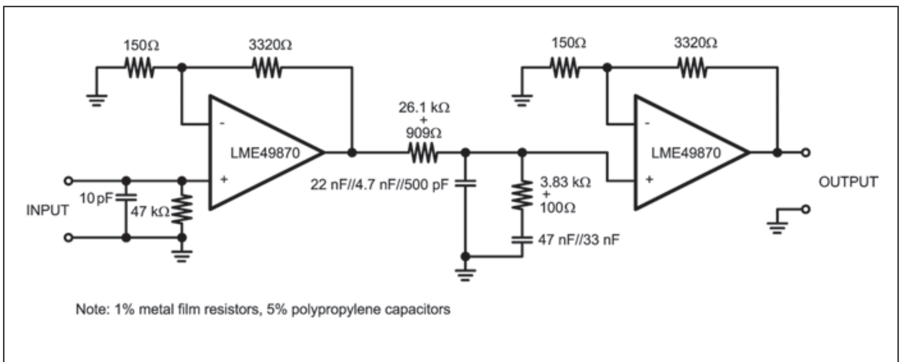

FIGURE 5: Passively-equalized RIAA phono preamplifier using National Semiconductor's high-performance LME49680 dual op amp. (Courtesy of National Semiconductor and THAT Corporation)

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2. Herbert, Gary. "An Improved Balanced, Floating Output Driver IC." Presented at the 108th convention of the Audio Engineering Society, Paris, France, Feb. 19-22, 2000. AES Pre-print 5152, www.aes.org.

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HD-2: A Single-Ended Class-A Headphone Amp

This vacuum tube headphone includes a 6SL7GT for the input stage and a 6AS7G for the driver.

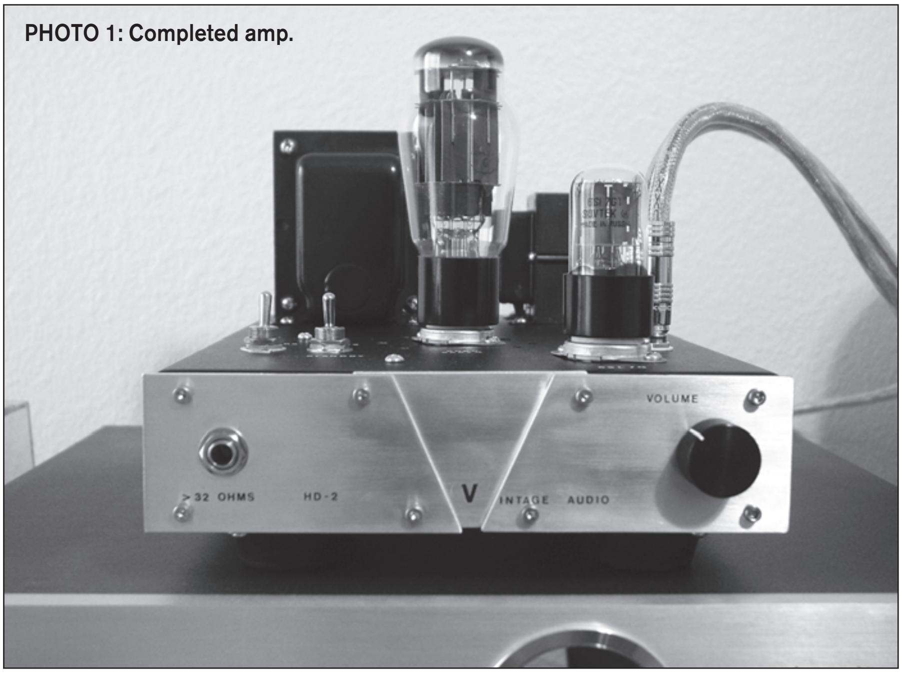

PHOTO 1: Completed amp.

I often enjoy listening to music with my headphones. My family also likes when I'm using headphones.

I began searching for a vacuum tube headphone amp, because solid-state circuits were too boring. The glow of vacuum tubes gives a relaxing ambience to a room in the evening. I was also looking for a gift for my nephew, who moved off to college, and I thought a headphone amp would be ideal for a college dorm. He could listen to what he wants—whenever and however loud—without disturbing anyone. He was interested in the tube sound system I have at home, as well as the old technology that can rival newer solid-state systems.

PROS AND CONS

I decided on a class-A, single-ended output for several reasons. A headphone amp can operate with very low wattage. A class-AB amp would be too powerful and, anyway, would linger in the class-A portion nearly all the time with such a low output. The power demand of headphones would never reach class B. Class A has more distortion than class AB, but the power demand of headphones makes the load-line so short that it lies on a linear part of the tube's characteristic curve. Class A second-order distortion increases proportionately with the output power.

Unless you've suffered major hearing loss, you'll never drive this amp into audible distortion. The disadvantages of class A are the wasted heat and poor power supply rejection. The latter is significant but can be overcome. Headphones can be unforgiving with hum or noise, so you need good power supply filtering.

First, a few words about safety. This project has potentials near 300V. Get some help if you are unfamiliar with high potential tube circuits. Perhaps someone you know can inspect and do the initial power-up for you. I always use a Variac to slowly power up new equipment.

I also use a three-prong power cord. Ground the chassis so the fuse will blow if there is a major fault or if the chassis becomes energized. Fuses, which are very cheap, are the first thing the hot line goes to when it enters the chassis.

I've seen other DIY designs with the neutral fused instead of the hot line. This will not offer protection if there is a hot chassis or a loss of neutral. I've also seen DIY projects that switch the neutral instead of the hot line. You should always place the power switch on the hot. I service industrial equipment in my dayjob, and the hot line is always fused and switched.

Add up all the volt-amps used in all the transformers. Divide this by 120V and the result will be the current consumed from the line. You will need a little lagniappe [extra measure of safety—Eds.] to get the fuse through turn-on surge without blowing, so multiply the steady-state current by 1.5. I ended up using about 0.5A, so I used a 0.75A fuse.

SPECS

I wanted to experiment with a headphone driver with no transformer between the output tube and headphones. My nephew uses a pair of Sennheiser HD-280s with an impedance of 64Ω, 102dB sound pressure level at 1.0V RMS. I use a pair of Sennheiser HD-600s with an impedance of 300Ω, 97dB at 1.0V RMS. I would need a low output impedance to drive these headphones. A cathode follower would work well in this application.

I began browsing my RCA tube manual[1] for a tube with certain characteristics. The output tube would need high transconductance, low plate resistance, and the capability of relatively high plate current. I looked at common audio power tubes such as 6L6G and 6550, which didn't fit all the criteria because of plate resistance or current limitations. Then I remembered the 6AS7G, which was designed for regulated power supplies but has also been successfully used for audio power OTL output stages.

The 6AS7G is designed to pass lots of current with little resistance. Its typical plate current is 125mA and plate resistance a mere 280Ω. A great caveat: the 6AS7G is a twin triode; one envelope is good for stereo and it's in current production at a reasonable price.

I like the choice of class-A triodes because the distortion contains mostly second-order harmonics with virtually no odd harmonics. Also, a cathode follower contributes virtually no Miller effect (the loss of high frequencies) because the gain is less than one. My headphones have impedances at or above 64Ω, which is a good impedance match, but you should avoid using headphones below 64Ω for this project.

I again referred to my RCA manual[2], and began designing the output stage. Typical operating parameters are 135V on the plate, 250Ω for the cathode resistor, and 125mA plate current. There are no plate curves in my RCA manual for the 6AS7G, but I found the curves in my 1998 Fall Svetlana Audio Tubes book[3]. I included this in **Fig. 1**.

I wanted to operate this hot-running tube conservatively. I chose a 100-0-100V plate transformer from the Hammond catalog to provide 120V plate supply. The next transformer option was too high. RCA recommends a cathode resistor of 250Ω[4], which I tried, but found it a little low for my application.

The 6AS7G ran 125mA as specified. I ended up using 270Ω because I wanted 100mA plate current. You may prefer to run your tube a little hotter to achieve lower distortion. The 270Ω cathode resistor will bias the tube within its maximum operating conditions.

I could not tell the difference in listening tests, so I used a lower current bias for reliability and to extend the life of the tube. Fixed bias is not recommended because of potential thermal runaway on this power-hungry tube. Self-biasing also provided a small amount of degenerative feedback to help reduce distortion.

The Red Book standard for a CD maximum output signal is 2.0V RMS, or 2.83V peak-to-peak. **Table 1** indicates the signal level as it passes through the amp. An input of 2.0V yields a grid-drive of 9.12V at the 6AS7G. The peak-to-peak swing is 1.4 × 9.12 = 12.8V.

Therefore, the swing from quiescent current is ±6.4V. **Figure 1** indicates this grid drive yields plate current of 65 to 145mA. The center of this swing is 105mA, but the quiescent current is 100. This is from the nonlinearity of the tube.

The distortion at this level is (105-100)/100, or 5%. The RMS voltage-swing at the cathode resistor is 7.64V. This is at maximum CD drive and 120dB sound pressure level (SPL) at the headphones! But music is dynamic and the average SPL will be much lower. At 84dB SPL, the distortion will be 1/64 that, or 0.078%.

The apparent loudness is halved for every -6dB SPL. 120dB-84dB is -36dB, which is a -6dB drop—six times—so the level is halved six times, or 1/64. The 2.0V RMS specification accommodates the huge dynamic headroom of CDs. The average level is much lower. I don't have a distortion analyzer, so these figures are purely theoretical.

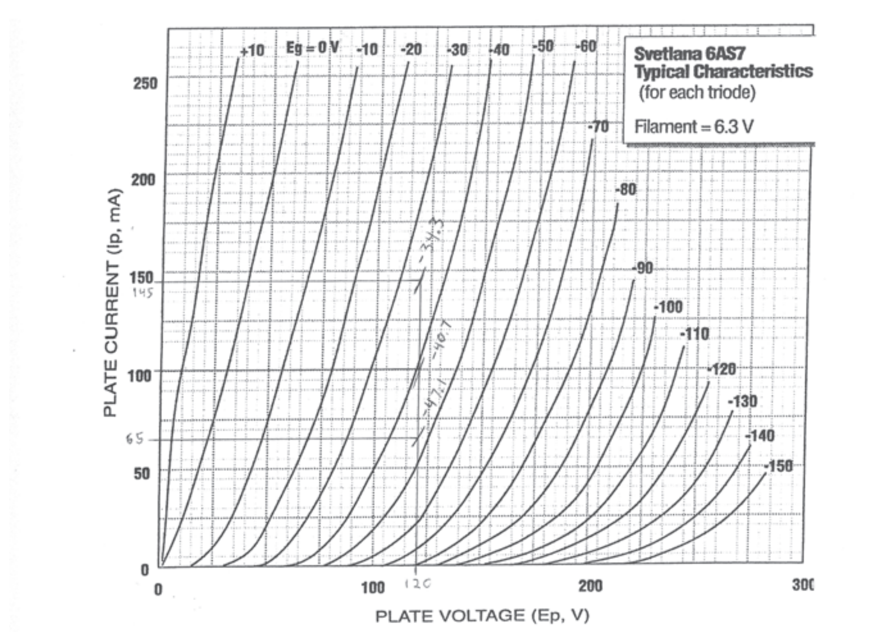

FIGURE 1: Svetlana 6AS7 typical characteristics (for each triode). Filament = 6.3V

COMPONENT SELECTION

In order to achieve unconditional output stability, I used R14 and R15 at the output, no global negative feedback, the grid resistors R10 and R11, and less than unity gain of the 6AS7G. The output of the cathode follower has a DC voltage, so I needed to compromise, using an electrolytic capacitor to couple the output with the headphones. Don't be alarmed because the capacitor needs to block less than 30V DC.

Higher voltage caps can be expensive, but you can use your money here on quality instead of voltage. The output capacitor needs to be large enough to go down to 20Hz. The minimum load impedance is 64Ω. The –3dB formula for capacitance is $C = 1000000/(f * 2 * \pi * R)$, where C is capacitance in microfarads, f is the cutoff frequency in Hz, and R is the headphone impedance.

$C = 1000000/(f * 2 * \pi * R)$
$C = 1000000/(20 * 2 * 3.14 * 64)$
$C = 124\mu F$

I used quality film bypass caps C14 and C16 paralled with the output electrolytics C13 and C15. Resistors R14 and R15 bleed the output caps to keep the phone jack at zero DC potential when there is no load attached. This will keep your headphones from popping when you plug them into the jack.

I began designing the input stage next. **Figure 3** is the complete schematic. I wanted to design the unit to operate at comfortable listening levels with the volume knob at about 25%. My intention here is to get good gain first, then throttle it at the input volume control.

I used a volume pot that Dennis Colin recommended in his solid-state headphone amp[5] in an earlier *audioXpress* issue. I have used this Alps pot, which is sold by Radio Shack, on other projects with very good results. It is very quiet and has good tracking between channels. When the volume pot is at 25%, or the 9 o'clock position, it is at 75% attenuation. The volume attenuation is a log function, so log (3/4) = -0.12, or 12%.

I looked at common voltage gain tubes such as 12AT7 and 6SN7, but their gains were too low. 12AX7 looked OK but I prefer octal tubes, which have larger cathodes and anodes, to miniatures. I think this provides better imaging and presence.

This narrowed my search down to the 6SL7G, whose gain is in the upper 30s. This is a little high, so, again, I compromised. You may need the extra gain for sources other than a CD player, and the lagniappe places my volume knob at 10 o'clock for comfortable listening. I used the data in the RCA manual[6], which has an excellent section on resistance-coupled amplifiers, to design the 6SL7G input stage. I had a 250V plate supply transformer I wanted to use, but I needed to utilize some interpolation of the data between 180V and 300V.

This is how I arrived at the resistor values for the 6SL7G input stage. I didn't use a bypass cap on the cathode resistor. This provided a small amount of local degenerative feedback and also reduced the excess gain in order to place the volume control position in a good location. Sometimes things just come together. I did not like the RCA manual's coupling cap recommendation with a high pass 100Hz cutoff[7].

This was probably for AM tabletop radios. I wanted the frequency response to go down to 20Hz. Again, the 3dB point for capacitance is:

$C = 1000000/(f * 2 * \pi * R)$, where R is the following stage's grid resistance

$C = 1000000/(20 * 2 * 3.14 * 100{,}000)$

$C = 0.08$

But this is the 3dB cutoff, so I elevated this by one decade to 0.8µF. A 1.0µF is the closest value I could find from Mouser that was a film capacitor with a 250V rating.

Table 1 shows the signal level as it passes through passive and active components in the amplifier. The CD player's maximum output is 2.0V RMS. This first goes to the volume pot and reduces the signal by a factor of 0.12 with the volume at 25%. The signal leaves the pot at 0.24V and goes to the grid of the 6SL7G. The gain of this stage is 38, so the signal leaves

TABLE 1

CD player	→	volume pot	→	6SL7G	→	6AS7G	→	headphones
	2.0V	×0.12	0.24V	×38	9.12V	×0.838	7.64V	120dB

the plate resistor at 9.12V and then goes to the grid of the 6AS7G. The gain here is about 0.838, less than one because it is a cathode follower. The signal leaves the cathode resistor at 7.64V to drive the output jack.

OPERATION

The headphones have an efficiency of 102dB/1V RMS. The dB level will be 20log (7.64V/1.00V) = +18dB above 102dB, or 120dB for maximum CD output. 2.0V RMS equals 2.83V peak.

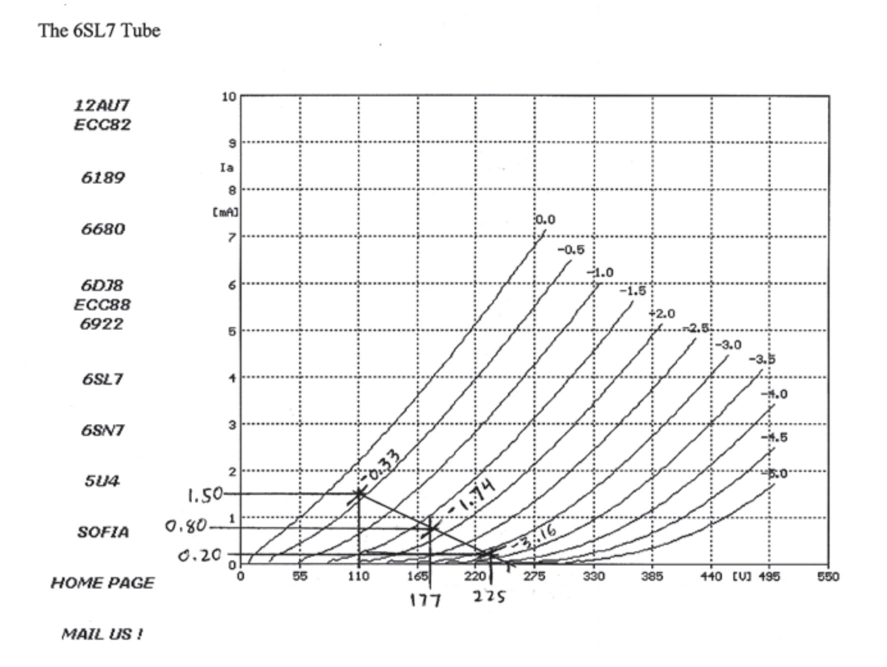

FIGURE 2: 6SL7G characteristic curves.

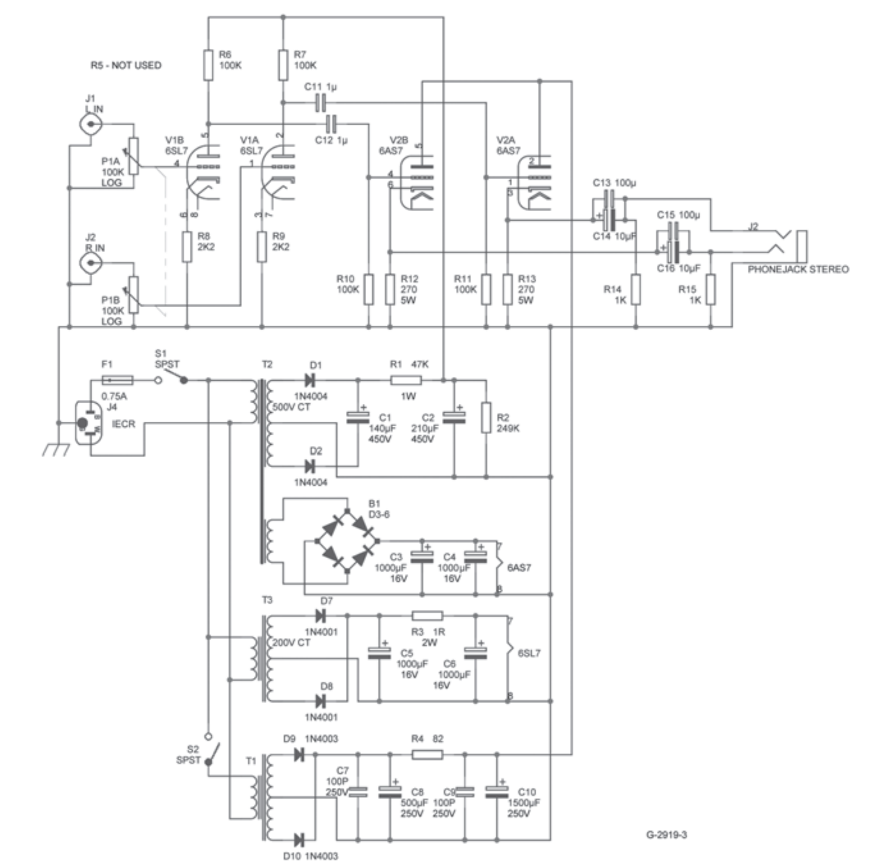

FIGURE 3: Amp circuit.

From the characteristic curves in **Fig. 2**, a 2.83 grid drive yields a 1.50mA to 0.20mA plate current swing and 110V to 225V plate voltage swing. The middle of the plate voltage swing is 167.5, but the quiescent plate voltage is 177.

The distortion calculates as (177-167.5)/167.5 = 5.67%. At 25% volume, the signal level is 20log (3/4) * 5.67 = 0.71%. This distortion level in the first stage is for a SPL of 120dB at the headphones with maximum CD output. As I mentioned earlier about the output stage, this is for transient dynamic headroom excursions. Typical average SPL will be around 84dB. Therefore, the distortion will be 1/64 of 0.71%, or 0.011%.

I could have chosen a more linear place on the characteristic curves to operate, but the gain would have been too high. Besides, headphones use very little drive, and I'm not driving loudspeakers to fill a room.

I performed an experiment with my Radio Shack sound level meter. I set the amp's volume to 9 o'clock and measured C-weighted, slow response with the SPL meter. I tried several different discs and noted SPL values mostly around 66dB, but some discs were hotter and measured 74dB.

I observed 84dB rarely on peaks. I raised the volume to get an average 84dB and listening was unbearably loud. This raised my confidence level about the distortion of the amp because actual listening SPL levels will be much lower than the values I used to calculate. I believe the calculated levels are worst-case figures. For me, it was an educational exercise measuring real dynamic music sound levels.

Also of interest, the action level for noise exposure is 85dB. Hearing protection is required for workers who enter a zone at or above 85dB. How many of you wear hearing protection around the home mowing the lawn, for example?

POWER SUPPLY

The power supply is in **Fig. 3**. Switch S1 energizes T2 and T3. This is turned on first to energize all tube heaters and the 6SL7G plates. I wasn't too concerned about cathode stripping in the 6SL7G. I've serviced many old tube amps, and the driver tubes in the voltage gain stages always outlast the power tubes. I believe power tubes are more susceptible to cathode stripping because of the higher voltage and plate current. The 6SL7Gs operate at very low current and have a 250V plate supply.

In contrast, most output power tubes operate nearly twice that potential. Although the 6AS7G operates at only 120V in this design, it draws a tremendous amount of current from the cathodes. Therefore, good thermionic emission is needed to avoid cathode stripping. This is why I chose to delay the plate supply to only the 6AS7Gs.

T2 provides the plate supply voltage to the 6SL7G. The voltage is full-wave rectified, and then a simple pi filter is used with plenty of capacitance. R2 is there to bleed the stored voltage down below 25V in one minute after the power is switched off. This is for your safety and convenience if you need to open the chassis for modifications or service. T2 also powers the 6AS7G heaters. This is also rectified, but more about that later.

T3 supplies power to the 6SL7G heaters. I always rectify heaters used in line-level gain stages. Eliminating hum is critical in the design of a headphone amp because your eardrum is only a few inches from the diaphragm. R3 was meticulously chosen by experimentation to arrive at 6.3V DC.

I am confident you will have success using the parts listed in **Table 3**. You may have different results if you use different diodes or transformer, but this should get you within heater specifications. T1 supplies the plate voltage for the 6AS7G. R4 drops the rectified voltage to 120V DC. R4 also provides a time constant to charge C10. Ninety percent of voltage takes about one-half second and gives a smooth application of the B+.

You may have noticed by now that T1 cannot be energized without S1 turned on. This allows you to manually delay the application of the 6AS7G plate supply. It's impossible to turn on the 6AS7G plate supply without the heaters. If you turn on S2 first, then nothing happens.

The proper power-up order is first turn on S1, then after 45 seconds, turn on S2. You can incorporate an automated delay for S2 with an on-delay timer-relay. I prefer the manual switch because I also use S2 for a mute/standby switch if I want to unplug interconnect cables when I change input sources. I thought about adding a source selector input switch but I wanted to keep this simple.

You may have noticed that there is no bleeder resistor for the 6AS7G B+. This tube draws so much current that it will dump C10 quicker than the cathode can cool down when the amp is shut down. I used solid-state rectifiers on this project because I could not find a tube rectifier that could handle the high plate current demand from the 6AS7G.

On completion, I tried several types of headphones of different impedances to ensure compatibility. I originally used AC voltage on the 6AS7G heaters because there is no gain in this stage to amplify the hum. I didn't think any hum would be picked up in this stage but I was very wrong. My 25-year-old Yamaha headphones sounded very nice and dead quiet, but the Sennheiser HD-600 had very slight hum. The Sennheiser HD-280 hummed even louder. The correlation was increase in hum with decrease in impedance.

I decided to try rectifying the filament voltage on the 6AS7G. I researched the Mouser catalog for high current diodes and used a conventional silicon rectifier. I was on the right track because the hum was gone in all three types of headphones but the filament voltage dropped to 5.8V. The forward voltage was too high from the silicon rectifiers.

I consulted the Mouser catalog again for a high current, ultra fast switching, soft-recovery diode and arrived at the 512-RURP860 rectifier. This was better but a little low at 6.0V. My Svetlana publication gave 5.7V to 6.9V for filament tolerance[8]. This was the best I could do, so I compromised and accepted what I had. **Table 2** gives a list of voltages to look for when you power up for the first time. I adjusted my Variac

TABLE 2

TUBE	6SL7G	6AS7G
FILAMENT	6.30V	6.30V (see text)
PLATE SUPPLY	250V	120V
PLATE	177V	120V
CATHODE	1.73	26.0V

PHOTO 2: Top view.

TABLE 3

ITEM	SOURCE	DESCRIPTION	P/N	$/UNIT	EXT$
R1	Mouser	47K, 1W	281-47K-RC	$0.13	$0.13
R2	Mouser	249K, ½W	71-CCF60-249K	$0.13	$0.13
R3	Mouser	1.0Ω, 2W	282-1.0-RC	$0.19	$0.19
R4	Mouser	82Ω, 5W	280-CR5-82-RC	$0.39	$0.39
P1	Radio Shack	100k Stereo	271-1732	$3.29	$3.29
R6, R7, R10, R11	Mouser	100k, ½W	71-CCF60-100k	$0.13	$0.52
R8, R9	Mouser	2.2K, ½W	71-CCF60-2.21K	$0.13	$0.26
R12, R13	Mouser	270, 5W	286-270-RC	$0.49	$0.98
R14, R15	Mouser	1K, ½W	71-CCF60-1K	$0.13	$0.26
C1	Mouser	140µF, 450V	539-CGS450V140	$15.24	$15.24
C2	Mouser	210µF, 450V	539-CGS450V210	$16.87	$16.87
C3, C5	Mouser	1000µF, 16V	647-UVZ1C103MHD	$2.38	$4.76
C4, C6	Mouser	1000µF, 16V	647-UVZ1C102MPD	$0.64	$1.28
C7, C9	Mouser	100pF, 250V	539-GM101K	$0.26	$0.52
C8	Mouser	500µF, 250V	75-36DX501F250AB2A	$10.90	$10.90
C10	Mouser	1500µF, 250V	75-36DY152F250AC2A	$19.78	$19.78
C11, C12	AES	1.0µF, 260V	C-FT1-250	$2.21	$4.42
C13, C15	Mouser	100µF, 50V	75-515D107M050BB6AE3	$0.36	$0.72
C14, C16	Mouser	10µF, 100V	5989-100V10.0-F	$4.22	$8.44
CAPMOUNT	Mouser	CAPMOUNT (4)	539-VR3	$1.57	$6.28
D1, D2	Mouser	400V, 1A	512-1N4004	$0.06	$0.12
D3, D4, D5, D6	Mouser	FRED	512-RURP860	$1.11	$4.44
D7, D8	Mouser	50V, 1A	512-1N4001	$0.06	$0.12
D9, D10	Mouser	200V, 1A	512-1N4003	$0.06	$0.12
T1	AES	6AS7 B+	167H200	$35.06	$35.06
T2	AES	6AS7 heater, 6SL7 B+	P-269JX	$30.56	$30.56
T3	Radio Shack	6SL7 heater	273-1365	$6.29	$6.29
F1	Mouser	0.75A FUSE	576-0313.750HXP	$0.76	$0.76
F1H	AES	FUSE HOLDER	S-H201	$1.58	$1.58
IEC	AES	POWER CORD	S-W123	$2.66	$2.66
IECR	AES	CORD RECEPTACLE	P-SP2-106	$1.35	$1.35
KN	Mouser	VOLUME KNOB	450MPN92	$5.63	$5.63
SW1, SW2	Mouser	SWITCH (2)	103-0132-EVX	$2.60	$5.20
CH1	AES	12 × 8 × 2 BLACK CHASSIS	P-H1441-22BK3	$17.78	$17.78
CH2	AES	12 × 8 × 2 BLACK COVER	P-H1431-22BK3	$7.70	$7.70
FT1	AES	FEET, PACK OF 4	P-H9107	$2.72	$2.72
J1	AES	BLACK JACK	S-H267B	$3.56	$3.56
J2	AES	RED JACK	S-H267R	$3.56	$3.56
J3	Mouser	PHONE JACK	568-NYS230	$1.09	$1.09
SKT	AES	OCTAL SOCKET (2)	P-ST8-807	$2.03	$4.06
TB	Radio Shack	TAG BOARDS (4)	274-688B	$1.49	$5.96
V1	Parts ConneXion	6SL7, Sovtek	6SL7G	$7.95	$7.95
V2	Parts ConneXion	6AS7, Sovtek	6AS7G	$19.95	$19.95
					$263.58

to 120V AC; your measurements should be ±5%.

CONSTRUCTION

I was very careful about the grounding scheme. A star ground is a soldering nightmare, so I made a ground bus from stripped wire. Each component in the circuit that goes to ground has its own lead to this ground bus. I kept the ground bus as short as possible, and close to the input jacks. When soldering a new wire to the ground bus, the heat will melt the solder joints of the other ground leads and the leads will separate from the bus.

I shaped the ends of the lead wires into hooks so they would stay attached when soldering other leads to the ground bus. The chassis is connected to the ground at only one point, the headphone common lug. Be sure to scrape the paint and bond to the metal. The ground bus is ultimately connected to chassis ground at this point and only at this point.

Ground the shields of shielded input cables only at one end. Tightly twist all A/C power leads and keep them a good distance from the audio path. The A/C neutral is not connected to the audio ground or the chassis ground. In the US the neutral is connected to ground outside at your service meter.

I used a Hammond steel chassis because steel will offer some protection against electromagnetic interference (EMI). I've included some figures to help in construction. You may use your own imagination to achieve more personalized results. **Figure 4** is a hole-pattern for the top and sides of the chassis.

All bolts are 6-32 machine screws. Some are tapped but the size B holes are for mounting hardware on top as well as solder tag-boards underneath with a 6-32 nut fastener. I suggest purchasing all components first, then verifying bolt patterns and sizes before drilling or punching. I like to cover the entire chassis with masking tape, and then I mark the holes before using a center punch.

This will protect the chassis' finish while you're drilling, punching, and cutting. You're welcome to use the information provided for personal use, but the faceplate design as seen in **Photo 2** is for Vintage Audio. You can be very creative with aluminum flat-bar to make your projects a personal statement and aesthetically pleasing.

I drilled a pattern of vent holes on the top and the bottom cover plate to allow the chassis to breathe. The bottom feet are tall and give plenty of space between the bottom and the surface where the amp is placed. I use rub-on lettering to mark the chassis in all my projects.

I have some that are over five years old. I am careful when handling them, but if you prefer, you can apply polyurethane clear-coat, which will last forever. I placed the RCA input jacks on top of the chassis for convenience. I have always hated blindly reaching behind equipment to plug in cables.

My nephew used this amp for a few months before it blew a fuse. I measured more than 50V on one of the 6AS7G cathode resistors. Failure analysis indicated the plate current was much too high. I unsoldered the coupling capacitor C11 and the plate current dropped to the correct value. The coupling capacitor was leaking DC voltage to the 6AS7G grid. On inspection, the outer wrapping of both C11 and C12 were peeling back from excessive heat.

I had placed these delicate polypropylene caps too close to R12 and R13. These cathode resistors each dissipate 2.5W and cooked the coupling caps, so be careful of the component layout. I relocated two new coupling caps and replaced the cathode resistors, too. I checked the 6AS7G tube on my Hickok transconductance tester and it checked out good and matched on both sides. This was an old RCA New Old Stock, a very rugged tube. The fuse did a good job, too.

This tube and the 6SL7G are still in current production at very reasonable prices. You can also obtain a NOS version of it. You may try substituting the 6AS7G with a 6080. This is the industrial equivalent, but I like the curvy shape of the 6AS7G better than the straight bottle 6080.

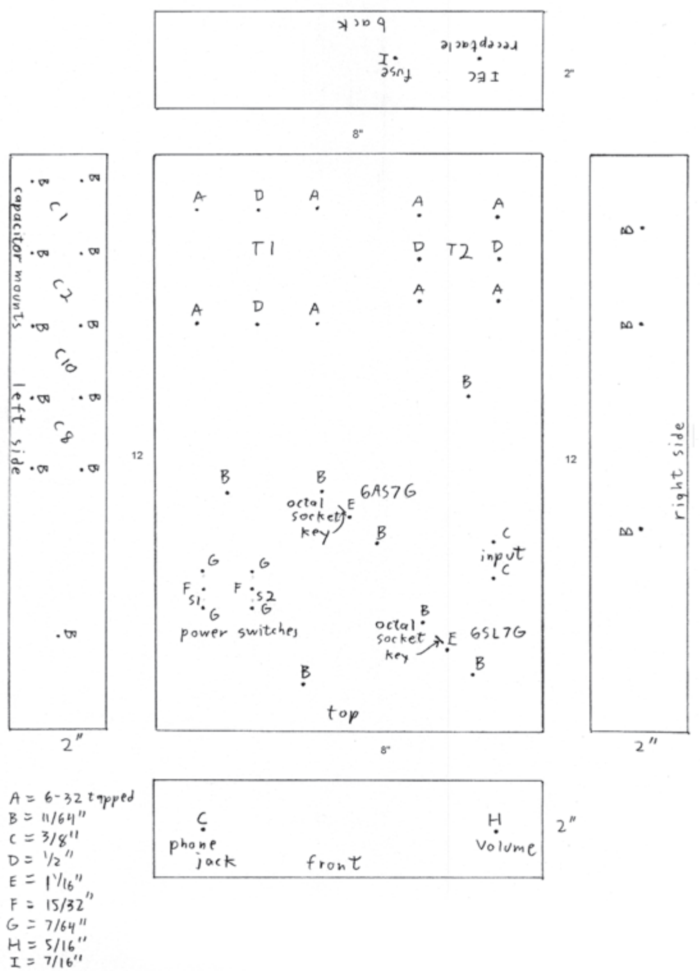

FIGURE 4: Chassis hole pattern

CONCLUSION

This small amp is designed to drive headphones with impedance at or above 64Ω. It will drive lower impedances but will lose the deep bass. I tried to design this amp to give 20-20kHz, -3dB, and the finished unit performs quite well with low distortion, too. Some compromises were made, but I think this is a satisfying headphone amp that will provide years of enjoyment. All the parts I chose are common, available, current production, and reasonably priced. I achieved the major goals of zero global feedback and class A in both stages. The output coupling capacitors' value is large enough to drive 64Ω headphones down to 20Hz.

A larger electrolytic capacitance could drive lower impedances, but would contribute nonlinearities. I'm amazed at the detail in small jazz band recordings, especially listening to plucked upright bass. The texture of brushes on drum skin is much more detailed than any loudspeaker I've heard. Larger orchestral pieces do not sound congested because of zero global negative feedback. This provides good imaging as well. I hope to hear from you with any comments, suggestions, or critique.

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1. RCA Receiving Tube Manual RC-30, 1975.
2. RCA Receiving Tube Manual RC-30, 1975, p. 171.
3. 1998 Fall Svetlana Audio Tubes, p. 2.
4. RCA Receiving Tube Manual RC-30, 1975, p. 171.
5. *audioXpress*, Nov. 2005, p. 38.
6. RCA Receiving Tube Manual RC-30, 1975, p. 645.
7. RCA Receiving Tube Manual RC-30, 1975, p. 642.
8. 1998 Fall Svetlana Audio Tubes, p. 2.

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aX Visits SEAS

SEAS (Scandinavian Electro-Acoustic Systems; www.seas.no) is a household name in the audioworld, known for its high-performance speaker driver units. In this age of massive outsourcing to low-wage areas, SEAS (pronounced sā'az) maintains a full R & D, production, and quality assurance facility in Moss, Norway, alongside the impressive Oslo Fjord.

The history of SEAS has been closely tied to other well-known Scandinavian acoustics companies. The company's technical roots go back to a driver designed by Vebjorn Tandberg in the 1920s. SEAS was formed in 1950 as an offshoot of the Radionette company that produced complete radio sets just after World War II. SEAS sold speaker units to other companies as well as to their parent company.

During the 60s, SEAS acquired the Videbaek Højttalerfabrik (later ViFa) speaker company in Denmark. Both companies enjoyed strong growth, with about 450 employees in the mid-70s. In the early 80s, economic conditions became less favorable, and the two companies split. SEAS became a fully employee-owned company in 1986 and proceeded to win the prestigious Norwegian Export Price award in 1997 and has enjoyed a D&B AAA rating since 1999.

Many of today's technological advances in today's drive units were either pioneered or first applied by SEAS: closed and shielded magnet drivers for use in TV sets (1952), woofers with rubber surrounds and soft dome tweeters for hi-fi (1968), and the patented DD-rings as part of the dynamic damping concept (1977). The accumulating capabilities and know-how led to the design of the Dynaco-A25 speaker in 1969, one of the most successful speakers ever produced with over a million units sold.

In 1988, SEAS developed coaxial drivers with neodymium magnets almost simultaneously with KEF, with neither company apparently aware of the other's accomplishments. KEF beat SEAS with a patent application by a few weeks, partly because SEAS was not sure it would be patentable at all. In 1993, SEAS introduced magnesium cast and machined cones. Another patent was granted in 1997 for the Hexadym magnet system based on an assembly of six neodymium magnets.

KEYS TO SUCCESS

Norway is a high-wage nation with almost zero unemployment. SEAS management knows that in such a situation it is impossible to compete with mass products and low prices. As Managing Director Jan Paus explains, the company focuses on a clearly defined portfolio of technically advanced drivers for demanding high-fidelity applications: "We realize that consistent quality is king in our business. So, even if we outsource parts and subassemblies, we keep the critical capability in the areas of design and development, final assembly, and production as well as quality control. We try to make our subcontractors part of the family. We keep a close relationship with them and we actually go out to train their employees to produce the quality we require." SEAS is now in the position where it receives regular offerings from subcontractors and can pick and choose the best ones.

Continued success also comes from constant improvements to construction and production methods. One example is the new spider ring assembly. Most drivers feature the lead-out wires coming out through or along the cone and then attached to the connector plate. That means several steps during assembly to route and connect the wires while the cone is already attached, increasing the risk of damage.

PRODUCT LINE

SEAS now uses a new subassembly that makes the voice coil, the spider ring that keeps the voice coil centered, as well as the wire connection plate together in a single assembly (**Photos 1** and **2**). Driver assembly is now much faster and of better quality because the spider ring assembly can be tested dynamically in a magnet system before the loudspeaker is built.

"We have basically three product ranges," says Paus. "For high fidelity ap-

PHOTO 1: New voice coil and spider ring subassembly.

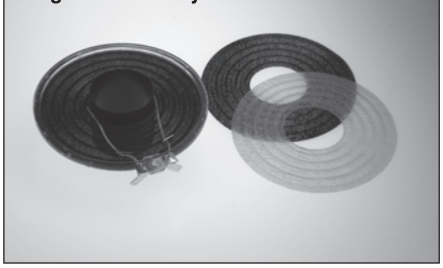


PHOTO 2: SEAS driver featuring spider ring.

plications there are the Excel and Prestige ranges, and for car sound applications we have the Lotus range. We are also introducing a new range, the Exotic drivers. Initially we will offer an 8″ full range driver with a papyrus cone and an hf sub-cone[1], the F8 (**Photo 3**). There is increasing demand for such a unit, which will be a perfect complement to relatively low-power tube amplifiers, especially for reproducing LP records. The Exotic range will be extended with a woofer and a cone tweeter in the very near future."

DESIGN PHILOSOPHY

But fine products are not enough to keep a small company successful. The fact that SEAS is a fully employee-owned company is an important factor. Employees identify with their company and its products and are willing to go the extra mile if necessary. There are no middle management layers, and I had the opportunity to attend the weekly company meeting at the factory floor, where anybody can bring up any issue. Management informed the employees on business, turnover, order book and plans, to keep them informed and committed.

So, what is the difference between the top-of-the-line Excel range and the Prestige line? "It is all a matter of when

PHOTO 3: Managing Director Jan Paus (center) and his R&D Manager Olav Mellum Arntzen (left) discuss the new Exotic F8 full-range driver with design Engineer Bjørn Børja.

and when not to compromise," says Olav Mellum Arntzen, SEAS' R&D Manager. "Obviously, you can never avoid compromises completely, but in the Excel range we tried hard. As an example, using the patented Hexadym magnet system with copper damping rings decreased the distortion caused by the motor system. But that brought out the THD and IM distortion of the cone, which we attacked by developing the cast and machined magnesium cone. Interestingly, we found that there was very little difference in cone performance between stamped cones from aluminum or from magnesium sheet. Our own magnesium cones show much more internal damping and therefore less break-up modes. The only explanation we have is that this is realized by casting the magnesium cone."

Another interesting product is the recently introduced DXT tweeter. One of the issues with tweeters is that with increasing frequency you get increased directivity—the beam becomes narrower and narrower. At the same time, you want the beam to remain homogeneous, without interferences from the faceplate, the baffle, or even the mounting screws. That's why you see so many flat faceplates, with recessed mounting screws, with the whole assembly recessed in the enclosure front baffle. In the DXT, the faceplate is deliberately shaped to enhance controlled reflections to improve the directivity and battle the beaming effect. You can see the shaped faceplate in **Photo 4**; the concentric surfaces are not flat but slightly concave. The improved directivity of this design is shown in **Fig. 1**.

TOUR OF FACTORY

Olav Mellum took me on a tour (**Photos 5-7**) of the development labs. Needless to say, SEAS has its own anechoic chamber. As a driver manufacturer, SEAS also invests in testing materials and mechanical behavior; most of that equipment is by Klippel, augmented with Bruel and Kjaer and ATB equipment. "We're really happy with that stuff," Olav explains. "The Klippel equipment gives us the means to really find out what's going on inside the driver, and that is an absolute must for us." A subset of Klippel equipment is available at final production QC; every single driver is tested before leaving the factory and test data is saved. The people at the development lab have access to all the data of all the drivers ever produced. "Tell me," I asked him, "what's the *real* story behind 'speaker break-in'?" It is real, he says. When a new driver is being used, the spider suspension properties change slightly, resulting in perhaps up to a 10% change in driver resonance frequency. The impact on the complete system is, of course, less, and I doubt whether it is actually audible. But if necessary it can be done in as little as one minute, if you

PHOTO 4: The concave and angled surfaces of the DXT driver prevent beaming at higher frequencies.

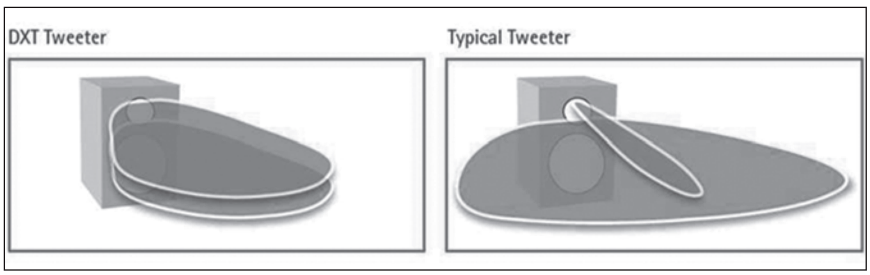


FIGURE 1: The effect of the special faceplate surface of the DXT tweeter provides much better dispersion (left) than common tweeters (right). Company figure.

PHOTOS 5 and 6: Shareholders at work at the employee-owned company.

PHOTO 5

PHOTO 6

would stretch the spider to its limits.

In addition to the "scientific" labs, there is the company listening room. You can make the world's best measuring drivers, but in the end it's the sound from a complete speaker that is the goal. The listening room is where complete speakers and crossover filters are developed and evaluated. "I guess by now everybody knows that the flattest response doesn't mean the best sound," Olav says with a wink. So, a lot of work goes into cabinet and crossover design. Although, with a few exceptions, SEAS does not produce complete speakers, they have developed speaker and crossover sets as an application service to their customers.

Not surprisingly, SEAS' view of the audio scene is that real progress today doesn't come from new amplifiers or sources, but from better speakers. There is still much unlocked potential in the electromagnetic driver. But improved products seldom result from radical breakthroughs. Step-by-step refinements, new materials, and constructions will deliver better and more lifelike audio performance in the future. SEAS clearly intends to be on the forefront of that. *aX*

REFERENCE

1. The new Exotic F8 wideband driver has many of the attributes of the legendary Philips type 9710M wideband driver of the 1960s and is targeted at similar applications.

Author's note: SEAS has recently launched a new website at www.seas.no. The site will be expanded shortly to include complete distortion and linearity data of all drivers as well as application notes.

PHOTO 7: Completed drivers waiting for final QC test before being shipped.

A Greek Hybrid in Blue

A conventional bipolar transistor and tubes amp project done with a Greek flavor.

PHOTO 1: The Hybrid Blue amp.

My first plan was to build an amplifier that would "move" my heavy, monstrous speakers, having very good sonic performance and combining the good bass response offered by the power transistors with the sweet tube middles. But I needed more power, which the tubes were not able to offer—except for maybe some parallel push-pull with six or eight KT88s, heavy and very "hot" constructions I tried before consulting some Philips books. Because the climate conditions have changed here in Greece, I decided to use the transistors, which broke my heart. The best approach was to use MOSFETs or conventional power TO3 transistors such as the Motorola series 15.000 MF SO (credit Douglas Self, who advanced the development of solid-state power stages, explaining how to improve the old diagrams such as Tiger, Ampzilla, and so on).

I made the PC board for the power stage able to drive 3+3 MJ 15024 and 15025 power transistors, both PNP and complementary NPN ones. For the power supply, I used two separate transformers (toroidal), made by my friend Tim Giatras, and two complete separate rectifiers and filters, as in "dual monoblock" constructions. The line preamplifier uses two 6H8Cs (the Russian equivalent of 6SN7), and the phono stage uses two ECC83s (12AX7). The huge metallic chassis was built by George, and the aesthetics belong to Aris. I made all the rest.

DIAGRAMS

The block diagram of one channel is in **Fig. 1**, showing the black boxes as follows:

- the solid-state power amplifier on PC

board
- the line preamplifier, containing one old Russian 6H8C, first working as a preamplifier, and the second as a cathode follower, to match the input of the differential amp of Mr. Self
- the RIAA phono stage (ECC83/12AX7) with a classical passive RC circuit.

Figure 2 shows one of the two power supplies, simplified.

THE POWER STAGE

The setup (**Fig. 3**) included three NPN transistors MJ802 (I used MJ 15024) with their pairs PNP (MJ15025) mounted on a huge aluminum heatsink, along the left and right sides of the metallic blue chassis. This worked easily and almost cold, after I carefully adjusted the bias (10mV). I also screwed the transistor MPSA06, responsible for the bias, on the heatsink. All the stages of this power amp were working perfectly, but I preferred to build the two PC boards, just to be sure.

The amplifier worked fine at ±65V, having considerable power, on both 4 and 8Ω output charge. It was exactly 185W RMS on 8Ω and 285W RMS on 4Ω, and the spectrum was also incredibly good, ranging from about 5Hz to about 98kHz. Of course, I carefully selected all the transistors, not to mention the rest of the parts. I would like to express my gratitude once more to Doug-

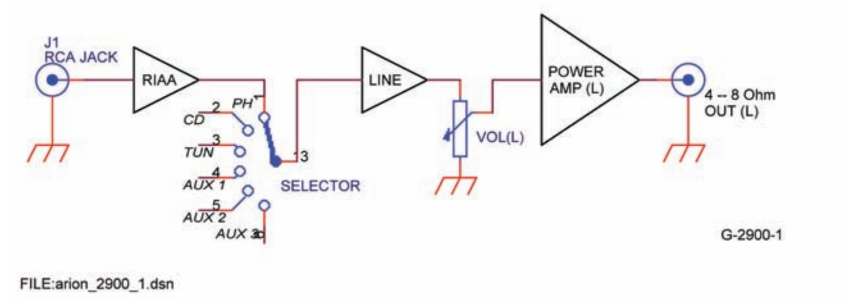


FIGURE 1: Audio chain, left channel.

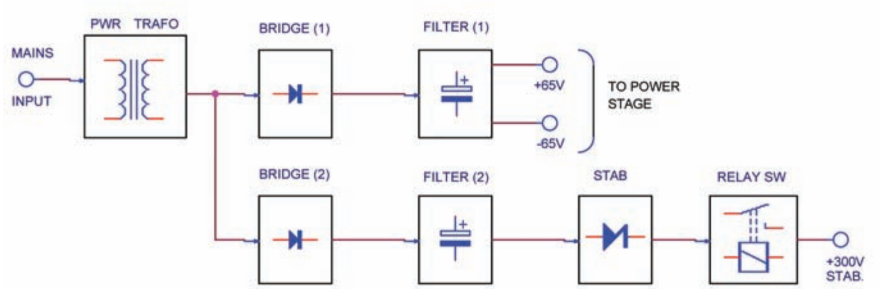


FIGURE 2: Simplified power supply block diagram.

las Self, whose books are a real help for many DIYers and professional solid-state builders, from both a theoretical and practical perspective. Those books made me reconsider my points of view about transistors! Of course, I would not betray my beloved glass tubes.

TABLE 1: Amp Parts List

C1	1μF, 50V
C2, C3, C11	100N, 200V
C4, C10, C12	220μF
C5	15P
C6, C7	47μF, 50V
C8	100P, 50V
C9	1μ, 50V
D1	1N004
JP1	A
JP2	C
K1	RCA IN
L1	zobel, 10mH, 15 turns, 1.5mm copper wire
P1	1K
R7, R14, R18	1K
R1, R9-11	10K
R2, R4, R16, R19, R20	100
R3, R5	68
R6	150
R8	33
R12, R15, R21	220
R13	2K2
R17	10R
R22-27	0.22, 10W
R28, R29	10 2W
T3, T1	56
T2, T4, T9, T10	MPSA06
T5-7	MPSA56
T8	NPN MPSA56
T11	MJE340
T12	MJE350
T13, T14, T17	MJ802
T14, T16, T18	MJE4502

THE PREAMPLIFIER

The line preamp (**Fig. 4**) is a double triode 6H8C (6SN7), with the first working as a class A preamplifier—classic automatic bias, and the input is connected to the volume control and then to the source's switch. Both these important devices are the Alps brands, and you should carefully select all the other components. The first triode is connected with DC with the cathode follower, which ensures low impedance to "attack" the power stage.

TABLE 2: Preamp and Power Stage Parts List

C1	2μ2, 250V
C2, C10	470μF, 100V
C3, C9	470N, 100V
C4, C7, C11	1μ, 400V
C5	6N8, 100V
C6	18N, 100V
C8	100μF, 100V
K1	Phono In
K2	Rec Out
P1	100K Vol. log
P2	47K Master Gain
R1, R3, R7, R12	100K
R2, R8, R11	1K5
R4	56K
R5	6K8
R6	270K
R9, R13, R16	1M
R10	120K
R14	1K
R15	47K
R17	220K
S1A	2P 6 pos, Selector
V1	12AX7 (ECC83)
V2	6SN7 (6H8C)

THE PHONO STAGE

I built the phono stage around the ECC83—TFK type, with the classical RIAA corrector connected between the two triodes. I carefully selected and measured the parts to make sure they were of good quality. The input sensitivity permits connection to MM type heads. I also tried to use 12AX7 and the Russian 6H2P, which needed a little base change (6H2P works only with a 6.3V heater), and the differences were minimal. The RC-RIAA circuit is very simplified, but readers could try many other circuits, such as those published in *Glass Audio*.

THE POWER SUPPLY

Because I am dealing with dual monoblock construction (**Fig. 5**), I used two big toroidal power transformers, built by Tim Giatras, who has helped me for many years to make my crazy valve

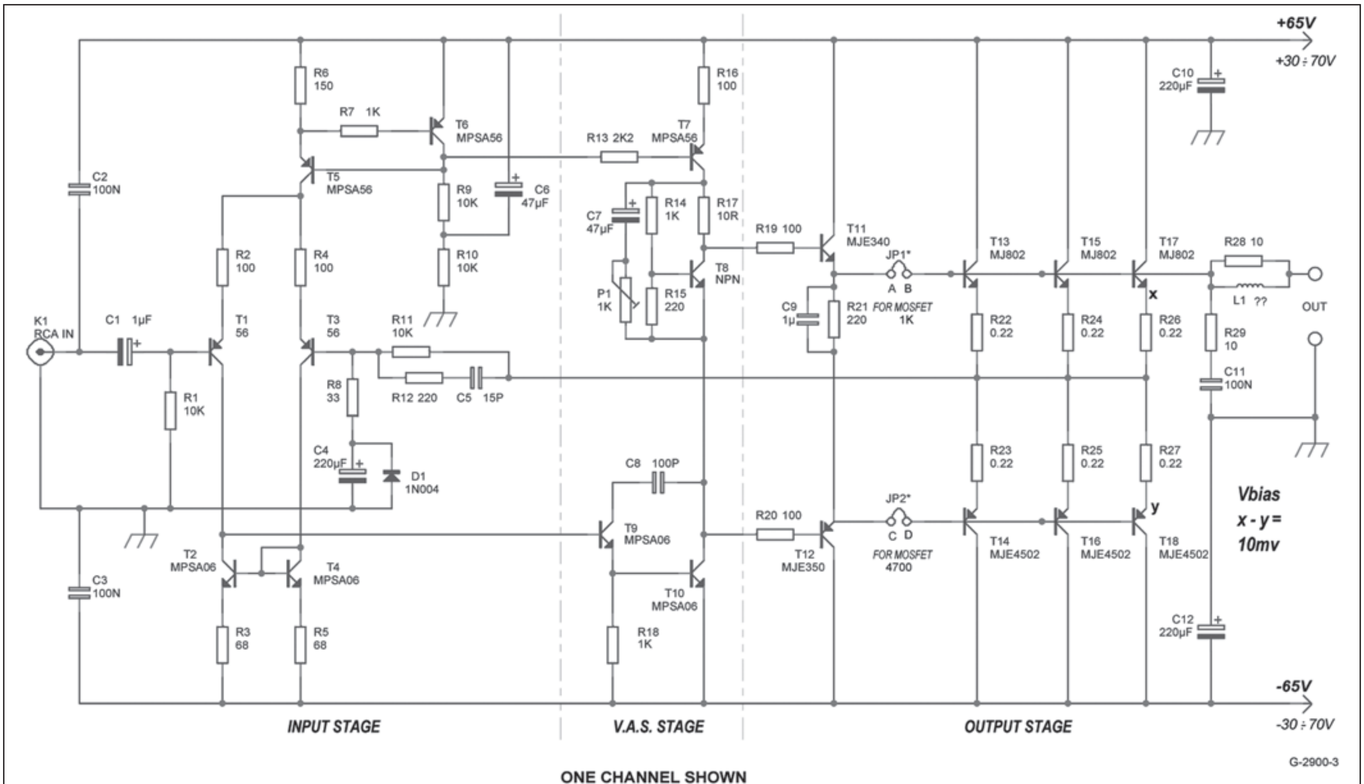


FIGURE 3: Power stage for Blue Hybrid integrated amp.

dreams real. Both transformers deliver the same tensions and current in the secondaries. The power secondary is rectified with a special bridge mounted on the 5mm aluminum base, and the symmetrical tensions are filtered by two Nyppon Chemicon capacitors (27,000MF), shorted with the RC (47k + 0.2MF). The second bridge will deliver the high tension necessary for the tube section, where I used a stabilizer tube (ECL82-6F3P-Russian), a triode/power pentode (the American equivalent is 6BM8). In order to give a little time to the tubes, I used a special relay tube, the same way Ed Dell did in his special construction issue (*Stereophile*) called "The Brute."

It worked well, and I was very sorry I could not find more of these same parts. The third bridge is used for the 6.3V DC for the heaters of all the tubes, including the relay tube. I found it unnecessary to stabilize the heaters, too. In the front panel, I mounted a magic eye (EM84) to show when the mounter is ready to kill. I did

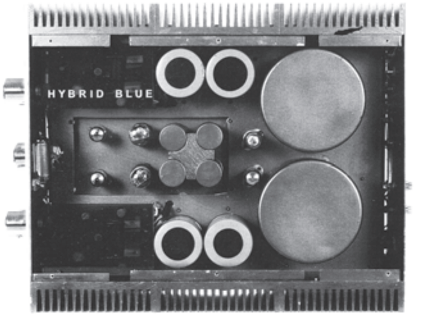


PHOTO 2: Top view.

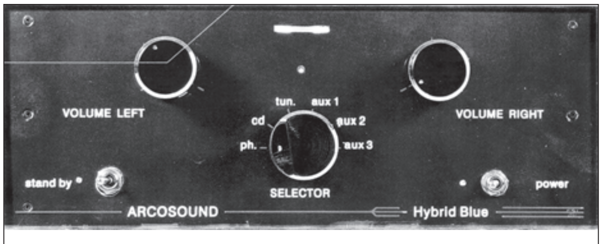


PHOTO 3: Front view.

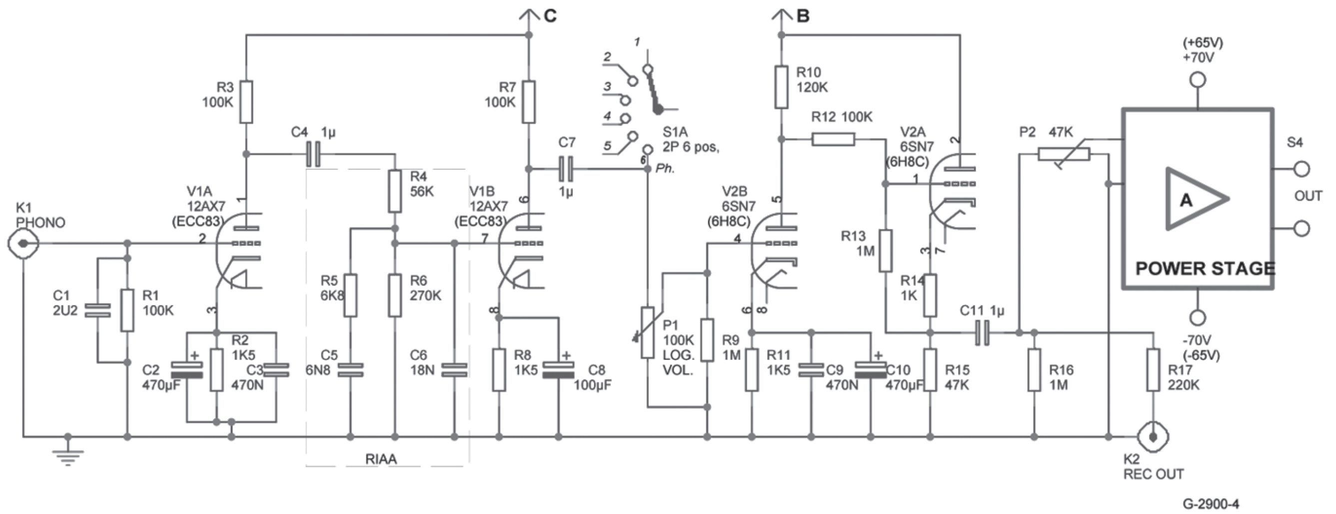


FIGURE 4: Tube preamp and power stage (one channel).

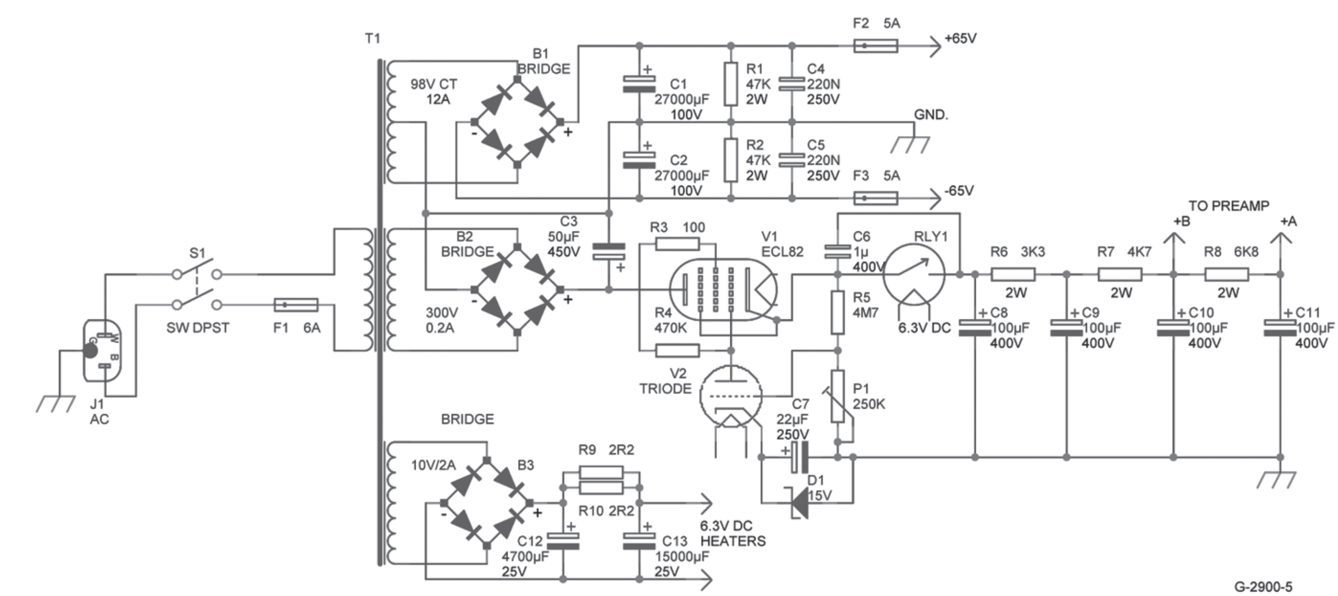


FIGURE 5: Powerw supply.

not want to make the magic eye "dance with the music," so I did not connect the EM84 input to the speaker.

MECHANICAL MOUNTING

As you can see from the photos, the Hybrid Blue was not an easy amplifier to mount. The base of 55/40 cm/5mm aluminum contains the two transformers, the two heatsinks, the mini chassis for the preamps, with their Elko, filter capacitors. In the front on a second chassis are the two PC boards of the power solid-state amplifiers, and in the back are the stabilizers. The power transistors support is Argon soldered to the sinks, same as the heatsinks between them. The massive plexiglass (8mm) cover (engraved back side), the front panel made from aluminum and plexiglass (also engraved), and the custom heavy knobs complete the image of the Hybrid Blue.

PERFORMANCE

The RMS power on an 8Ω load was 185W, and 285W on a 4Ω load. The frequency response was ±0dB from 5Hz to 98kHz, and the THD was under 0.05%. The "Sound and Image" team tested Hybrid Blue, and they put it to work with the most insensitive and heavy speakers such as ATC S171—from Audio Spectrum, and TAOC-Japan, and so on, and the sonic impressions and the positive feedback were the best. If you want to find more about all our Greek amps, please contact me at arcosound@hotmail.com. aX

PHOTO 4: Inside the amp.

TABLE 3: Power Supply Parts List

B1-3	Bridge
C1, C2	27,000µF 100V
C3	50µF 450V
C4, C5	220N 250V
C6	1µ 400V
C7	22µF 250V
C8-11	100µF 400V
C12	4700µF 25V
C13	15,000µF 25V
D1	15V zener
F1	6A
F2, F3	5A
J1	AC Input
P1	250K Lin
RLY1	relay thermal
R1, R2	47K 2W
R3	100
R4	470K
R5	4M7
R6	3K3 2W
R7	4K7 2W
R8	6K8 2W
R9, R10	2R2 10W
S1	SW DPST
T1	98V CT
T2	10V/2A
T3	98V CT
V1	ECL82
V2	Triode

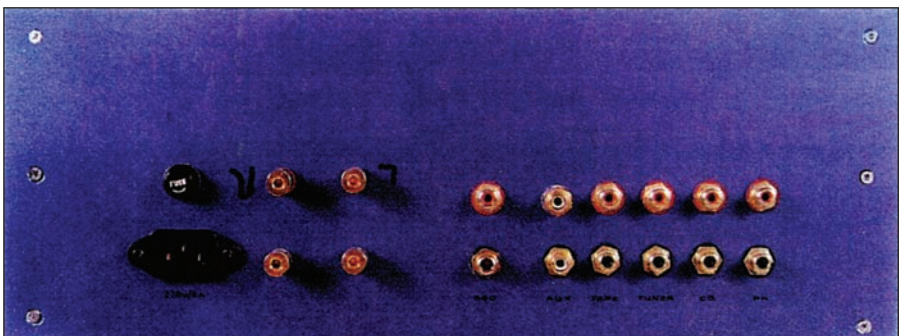


PHOTO 5: Rear view.

 By Darcy Staggs

Copper Tubing **Capacitor Connections**

Copper tubing is not just for plumbing anymore. . .

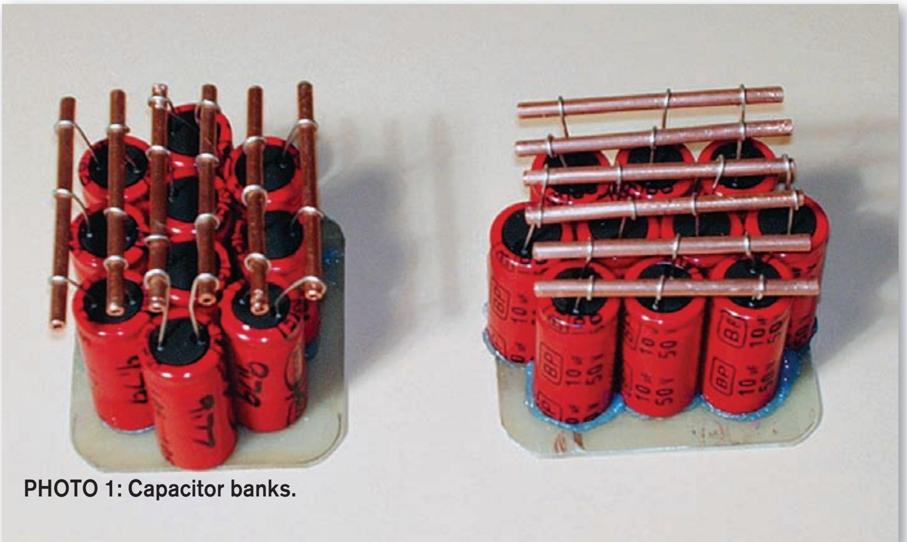


PHOTO 1: Capacitor banks.

Photo 1 shows a pair of capacitor banks I made using copper tubing for interconnects. I use these banks in raw power supplies and crossovers (with non-polarized caps, in this case). I find that the sonics of such assemblies are better than with soldered lead tangles and PC boards, and are easy to build, install, remove, and wire.

DETAILS

Measure each capacitor and mark them for value and for polarity. Then cut the leads to the same length, and form loops in them using "chain nosed" or "round nosed" pliers. The loops should be a friction fit on the tubing.

Before cutting the tubing pieces to length, I sand the tubing stock and clean it with a solvent to achieve a bright, new surface for soldering. Then slide each cut tubing piece through the loops in the capacitor leads, and bond each row thus formed to the PC-board-material platform with silicone and allow it to set.

Soldering requires a large iron because the tubing conducts heat easily, so you should solder all joints on one length of tubing at once, beginning at one end. The solder will follow the wire loop and make a very long, low-resistance joint.

Interconnecting the copper tubing depends on the series/parallel combination of capacitors needed. My preamp raw supply uses a pi-filter design with resistors bridging gaps between tubing runs. The power amp raw supplies use a bank of capacitors in parallel for each voltage. The assemblies in **Photo 1** became crossover components with all capacitors in parallel.

It doesn't take many sketches to design a neat, organized capacitor bank connected with copper tubing, which will perform superbly and look good in the process. aX

OIL CAP CAUTION

The article by Bruce Brown in the April 2008 *aX*, "Upgrading a Classic Pilot Amp," was quite interesting and showed a resourceful use of older tube equipment. I just want to sound a word of caution about old oil capacitors, which are made by winding two metal foils separated by two or more sheets of kraft paper into a compact roll. The wound capacitor is vacuum impregnated with resin, wax, mineral oil, or a synthetic compound. The term "mineral oil" is loosely used to characterize any non-chlorinated liquid, including refined petroleum oils, silicones, fluorocarbons, and synthetic esters.

In the 1950s a new class of synthetic oils called polychlorinated biphenyls (PCB) was introduced by Monsanto and GE for use as dielectric fluids. Because PCBs were relatively expensive, oil capacitors were not the major use for PCBs, but they still may have accounted for 5% of the total usage. The primary use of PCBs was in large utility power transformers.

In the 1970s, PCBs were shown to cause a variety of environmental and health risks, and the manufacture and use of them was either banned or severely restricted in the major industrialized countries. The US and Canada stopped production in 1976. Large PCB-filled transformers and capacitors have mostly been decontaminated or destroyed because of the risk of fire or leakage. By federal law, all new-production non-PCB oil-filled caps must be marked as such.

However, many smaller PCB-filled oil capacitors were used in all kinds of electrical equipment, including home appliances and consumer electronics. If an oil capacitor is not marked as non-PCB, I would assume that it contains PCBs. Small PCB capacitors were still in the distribution pipeline after manufacturing ceased. It may also be possible to purchase used ones by accident (on eBay, for instance). Oils have much higher dissipation factors than more modern dielectrics, and the paper acts as a thermal insulator and holds in the heat. The maximum operating temperature for mineral oil capacitors is +85° C.

A growing number of oil-impregnated paper film caps has been in operation for 20 years or more. A recent IEEE study found that thermal stress over this time period causes deterioration, which increases the amount of dissolved CO_2 in the oil and lowers the partial discharge inception voltage. Partial discharge is a localized dielectric breakdown in the oil insulation system in the presence of high voltage. The dielectric constant of a gas void is considerably less than the surrounding oil, so the electric field voltage stress appearing across the void is significantly higher than the equivalent distance of oil. If the voltage stress across the void is increased above the inception voltage for the gas within the void, partial discharge activity could start within the void, causing progressive deterioration of the paper insulating system, ultimately leading to failure.

The oil caps in Photo 2 in the article do not have a manufacturer logo, just the 2262 date code (1962?) and FAST. LF. This does not refer to the Solen Fast Capacitors that were introduced in 1987. His caps may be 46 years old, and were manufactured during the PCB era. Sprague mineral-oil-filled caps are clearly marked Sprague and after 1976 have non-PCB markings near the date code. The late 1960s Sprague catalog I have shows that all their oil-paper capacitors used purified mineral oil. I hope this caution is useful to readers.

Charles Hansen
Ocean, N.J.

Bruce Brown responds:

I appreciate Mr. Hansen taking time to respond to my article. He is correct in his statements about the potential issues with PCBs in oil capacitors. He is also correct in pointing out the ones in my amp are quite

CORRECTION

In "The Sidewoofer Line Array" (May '08, p. 41), some of the article photos were incorrectly labeled (not the author's fault). The article, of course, makes more sense with the following corrected captions:

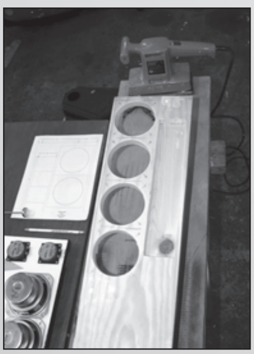


PHOTO 1: Detail of front panel tweeter slot.

PHOTO 2: Installing tweeters in slot.

PHOTO 3: Raw enclosures showing mirror image front panels.

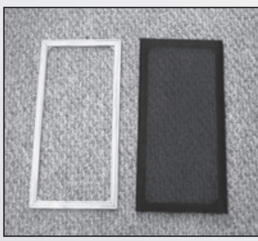


PHOTO 4: Front panel speaker grille made from screen door molding and heavy-duty pet screen material.

PHOTO 5: Crossover installed in enclosure bottom.

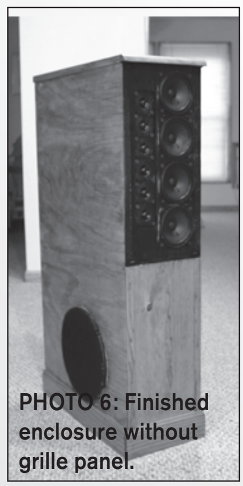


PHOTO 6: Finished enclosure without grille panel.

old. This is no surprise because they were part of the telephone amp I salvaged for the chassis—as stated in the article—which contained 4 and 5 pin tubes.

I would ask Mr. Hansen to consider writing a complete article for *audioXpress* generally on capacitors with comments on PCBs and other oil dielectrics, because I think there are still potentially thousands of sources of oil-filled caps for the consumer. I recently ordered a large number of oil-filled caps from a seller in Russia. I have not received them yet, so I don't know whether they will be marked. Some of the ones I ordered are new production, and a number of them are Russian military surplus. There are also many companies and individuals from Hong Kong, Thailand, Japan, and China selling oil caps on eBay. Given the recent issues with Chinese-made goods, and toxic substances being used, it certainly would not surprise me to find they contain toxic substances.

HYBRID AMP UPDATE

I have the following questions regarding Bill Christie's recent article, "A Hybrid High-Fidelity Power Amplifier" (*audioXpress*, January 2008, pp.30-36):

1. Figure 2 does not show any C3, C4 (15pF) or C5, C6 (33pF) caps as indicated in the Table 3 parts list. C6 and C7 are shown as 30pF in Fig. 1, but in Table 1, C7 is 470pF, C8 is 470pF and C9 is 1µF at 630V. What are the correct values? And where should C3+C4 be placed?
2. Is the chassis custom or off-the-shelf?
3. What is the P/N for U2's heatsink?
4. Subjectively speaking, what have others commented as to the sound quality?
5. Are the output tubes class A, A/B, what?
6. What changes could be made to bring the power down to, say, 25-30W?

Bill Mackenstadt
deltalight77@hotmail.com

Bill Christie responds:

1. The parts list and the schematic for the power supply are in error. The corrected versions of both are on the following page. I sincerely apologize for the confusion. I built several different versions of the amplifier using various transformers that I salvaged from amplifiers I had laying around and I got the documentation mixed up.
2. The chassis is custom built. I made it out of 1/8″ aluminum plates and fastened them together with aluminum angle.
3. The heatsink for the LM4702 is Avid Thermaloy part number 551002B00000G and is stocked in depth at Digi-Key.
4. My audiophile friends all found the sound superb, including one who is currently using Martin Logan Speakers and Bryston amplifiers and has used Magnepan Tympani speakers and Audio Research amps in the past.
5. The driver board is fairly universal and can be used with a variety of output stages. Using output transformers from the Dyna Mark IV amplifier and EL34 output tubes, the resulting amplifier should produce about 35W. And there would be considerable cost savings. At currently advertised prices:

Transformers from Triode:
Mark 4 Bundle (mk4deal) at $169.90
Mark 3 bundle (mk3-xfmrdeal) at $219.90
Tubes from Antique Electronics:
Valve Art EL34B output tubes $23.40/pair
Valve Art KT88s at $42.40/pair

The biasing voltage for the EL34s is different than for the KT88s, so resistors R15 and R16 should be changed to 47kΩ to accommodate the difference. The bias current should be set to 45mA (450mV across the cathode resistors). No other changes should be necessary. The alternates that I added to the parts list for IC U1 and transformer T3 are worth considering. The LME49710 has slightly better specs than the OPA134 and the Avel Lindberg transformer is a toroid and will generate a smaller stray magnetic field. I have used both in a conversion of the Dyna Stereo 70.

AMP DESIGN QUESTIONS

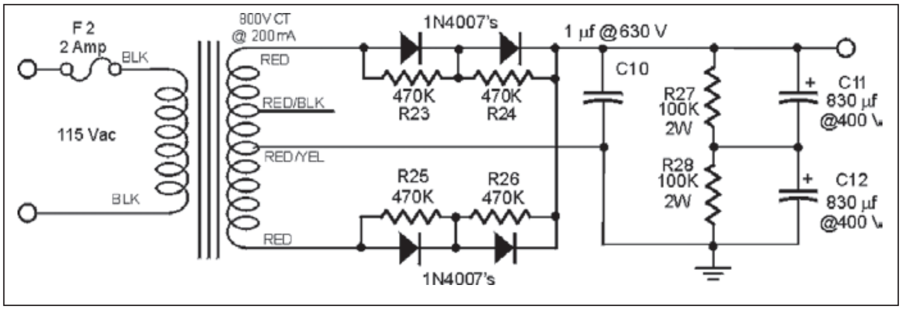
I'm in the process of building an amp using Bill Christie's ("Cascoding National's LM12 for 140W," Aug. '07) design, but I have a few questions.

1. Looking at the chassis wiring diagram pdf, I do not see where the power supply board is tied to the chassis. In fact, the chassis doesn't appear to be tied to ground anywhere. The power supply PCB doesn't have any extra pads to connect it to the chassis.
2. On the chassis wiring diagram pdf,

CORRECTED PARTS LIST FOR LM4702 HYBRID AMPLIFIER

RESISTORS	
R1	10K ½W
R2	1 Meg
R3	470 ½W
R4	15K ½W
R5	**1.8K ½W**
R6	1.5K ½W
R7	1.6K ½W
R8	10K Trim Pot
R8a	43K
R9,R10	47K ½W
R11,R12,R23-**26**	470K ½W
R13,R14	25K Panel Mount Pot
R15,R16	180K ½W
R17,R18	1K ½W
R19,R20	10 1W
R21,R22	100 1W
R27,R28	100K 2W
R29,R36	JUMPER
R30,R37	22K ½W
R31,R38	10 ½W
R32,R39	20K ½W
R33,R40	430 ½W
R34,R41	2K ½W
R35,R42	3.3K - 2W
R43	1K 1W
R44	**8.2K ½W**
CAPACITORS	
C1	2µF
C2	400pF
C3-5	not used
C6,C7	33pF
C8,C9	0.47µF
C10	1µF at 630V
C11,C12	820µF at 400V
C13	not used
C14,C18	680µF at 100V
C15,C19	10µF at 100V
C16,C20	47µF at 25V
C17,C21	47µF at 100V
C22,C23	100µF at 100V
TRANSFORMERS	
T1	Triode A-431-S
T2	Triode P-782-S
T3	Hammond 166G100 (100VCT at 0.50A)
alternate	Avel Lindberg Y236208 (110VCT at 0.45A) Toroid
DIODES	
ZD1,ZD3	1N4735 (6.2V 1W)
ZD2,ZD4	1N4744 (15V 1W)
ZD5	**1N4756 (47V 1W)**
D1,D2,...	1N4007 (1A 1000 PIV)
TRANSISTORS	
Q1	TIP122 Darlington NPN 100V, 3A, 40W)
Q2	TIP127 Darlington PNP 100V, 3A, 40W)
ICS	
U1	OPA134PA Single OP amp
alternate	LME49710
U2	LM4702C Dual HV OP amp
U3	LM317T Pos. Adj.,. 1.5A
U4	LM337T Neg. Adj., 1.5A
VACUUM TUBES	
V1,V1	KT88(6550) Power Tetrode

High Voltage section of the power supply



there is a reference to "See heatsink wiring diagram." Would it be possible to post this pdf to the *audioXpress* website?

3. And most important (to me at least) is locating an appropriate heatsink. If Mr. Christie has any suggestions about where to purchase the Aavid 79815 heatsink, I would much appreciate it, as I have been unable to locate a source.
4. More information on the chassis construction would be welcome. Were the top and bottom panels purchased or made?
5. The input signal to the amplifier appears to be shielded wire in the chassis wiring diagram pdf. I'm assuming that the shield (gnd) is connected to the input connector gnd and amplifier PCB gnd. There is also a wire with a 10Ω resistor connected to the input connectors ground. Why doesn't the last connection to ground cause a ground loop, or to ask another way, what is its function?

Thanks for the detailed drawings, PCB masks, and so on. I'm sure when I finally complete building this beast, it will have been well worth the effort.

D. Heian
St. Paul, Minn.

Bill Christie responds:

Thank you for your interest in the article.

1. The center connection of the power distribution board connects to the ground terminal of the Corcom filter that has a metal body and makes the ground connection to the chassis.
2. Here is a copy of the heatsink wiring (**Fig. 1**).
3. I bought the heatsinks at a flea market years ago. Unfortunately, the only source I know of is Aavid, and with their setup costs the price is unreasonable. I hope you can find a suitable substitute. Sorry about that.
4. I made the chassis (**Fig. 2**) from 1/8″ aluminum that I bought from a local surplus metal supplier. If you contact me, I will forward you detailed dimensioned drawings (christie.bill@gmail.com).
5. You can omit the resistors. I'm not sure what their function is. I've seen them used in other amplifiers, so for no good reason I included them in my design.

I hope these will answer your questions satisfactorily.

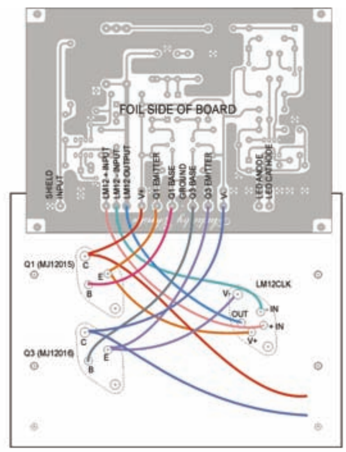


FIGURE 1: Heatsink wiring.

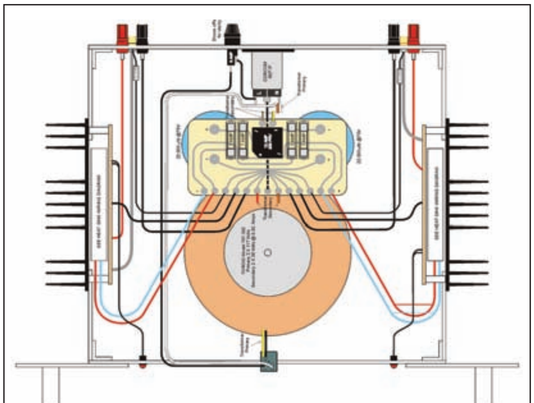


FIGURE 2: Chassis wiring.

CHANGE IN THE AIR

For those who aren't already fully up to date, here is a short rundown of what's happening and going to happen with the pending "switchover" to digital TV next February.

First, the good news: If, like most people, you are receiving your TV channels by cable (such as Comcast) or satellite (DirecTV or DISH), you don't have to do anything! In general, their boxes will receive the local digital channels and convert them to be seen on your old analog TV or new digital TV.

Cable is required to carry your local TV stations (which does *not* include stations from out of state, which you may now be able to get with an antenna).

The satellite providers are not currently delivering digital local channels in all areas, but are working toward doing that. In the meantime, they do carry the CBS, NBC, and ABC programming from East (NYC) or West (LA), but you must petition them to get those. You may need to upgrade your cable box or satellite receiver, if you want to see programs in HDTV, but they all offer deals on those.

For those who are getting their TV reception from an antenna: "Switchover" is in quotes because it is really a "switch-off." All of the local stations (those that you can receive with an antenna attached to your old TV set) are broadcasting on their digital channels already, as well as on their standard or analog channels (which can be tuned on your old TV or VCR without cable or satellite service).

So you can switch over at any time, if you have a digital tuner. You will need to do that anyway, before next February, when all the analog transmitters will be shut off.

The advantages of switching to digital reception—beyond a better and more detailed picture and up to 5.1 channel sound—are that digital can deliver an uncompromised picture with less than a tenth of the signal strength required for an acceptable picture from an analog station, and that the digitally transmitted image will have no "ghosts" or noise (snow). If the digital signal does not have enough signal strength or suffers very bad multipath problems (that would cause strong ghosting of an analog signal), there will be no image on the screen. The only exception being dynamic signal interference, during which the receiver will attempt to adapt to short-term conditions and the picture may momentarily break up.

To its credit, the FCC has mandated that all new TV sets, of any type, must have a built-in digital tuner. All older models, still in retail stock, must be labeled to inform you that they cannot receive digital broadcasts.

The government, realizing that many people can't afford an expensive digital TV receiver/tuner, is offering a $40 coupon that you can use when you buy a digital tuner, which will convert digital broadcasts to analog, so you can continue to use your old TV after next February.

Most major stores that carry TV sets are starting to get those on the shelves now. Here is the website to order coupons (limited to two per household): www.dtv2009.gov/ApplyCoupon.aspx.

There are also other kinds of video equipment, with digital tuners in them, that can receive digital broadcasts, such as DVRs (a digital version of a VCR). The ones that record and output true HDTV quality are more expensive than those that only "down res" to standard analog quality for your older TV.

However, as noted above, any new set, capable of displaying HDTV, will by law already have a digital tuner in it.

Terms and some other technical stuff: NTSC is the TV system we have all been using in the US up to now. It delivers a 480i image (a picture drawn with 480 horizontal scan lines which are produced by the visual interlacing (indicated by the i after 480) of two sequential sets of 240 scan lines. The picture format is four units wide by three units high. Among the unwelcome features of the NTSC system are the presence of a color and a tint control, which are never set just right—hence the popular translation of the acronym NTSC as never twice the same color, rather than National Television Standards Committee. Originally designed for a black and white picture and monaural sound, clever engineering subsequently managed to add color and then stereo sound, while staying within the original bandwidth limits.

ATSC, standing, of course, for Advanced Television Standards Committee, is the new digital TV system. Within these specifications are the ability to deliver many different forms of video, from the same 480i to 1080i or 720p (the p meaning progressive scan, which draws all the scan lines in the picture in a single sweep) HDTV in a wider 16 × 9 (or 1.78:1) format, versus the 4 × 3 (or 1.33:1) format of NTSC. Because it is easy to manipulate the video signal while it is in digital form, digital tuners are able to convert an incoming picture from any one of the allowable forms to any other. Thus the tuner is able to optimally match the incoming video to fit the resolution of the display it is attached to. ATSC also carries sound in the Dolby AC3 format, which allows the audio to be reproduced in mono, stereo, or 5.1 surround.

DTV, or Digital TV, encompasses all of the ATSC-permissible video standards. That can include a single channel carrying four NTSC (standard definition) programs at once, a single, fully detailed HDTV program, or a combination of one HDTV program, with a compromised image quality, along with one or two standard programs. Call the station and tell them you want your HDTV. They see the possibility of more revenue from their one channel when they can send more programs out on it. So DTV and HDTV are not synonymous.

The paranoia of content providers (owners) increases dramatically when the content is in digital form that might be easily copied or sent out (shared) via computers and the Internet. Thus HDTV content is coded with HDCP, or High-bandwidth Digital Content Protection. This mighty and complex defense system actually requires that every individual piece of electronics through which the video signal passes, from source to inside the display, must send out a query to the next piece in line and receive a satisfactory reply of authorizing identification, or the signal stops right there and never reaches your screen. These communications do not always go smoothly, especially because the specs are in their third iteration and there is a lot of first and second iteration gear in use already.

One way of simplifying the interconnection of both video and audio signals between units is the HDMI connector and cable. The HDMI specs include HDCP and have been changing up to now in their third version, including higher video bandwidth and audio signals. Much of the older HDTV equipment used the DVI (Digital Visual Interface) connector (originally created to carry digital video from a computer to its display). DVI connectors can handle the same signals as the HDMI connectors, and you can get cables with DVI at one end and HDMI at the other, as well as adaptors.

Whether you ease into it with a simple tuner/converter and your old TV set, or spring for a new flat screen, you are bound to get a better picture by next February. Happy viewing.

If you are not satisfied or just want to complain about any of this, you can e-mail me at joelco@earthlink.net. The first time you try it, Earthlink will diligently drop your e-mail into my "is this SPAM?" box, but I will fish it out in a day or so.

Joel Cohen, CTO
Crystal View Inc.

TUBE WARNING

I want to thank Charles Hansen so much for writing *A Brief History of Bendix Red Bank Tubes*. I enjoyed reading it, especially the details of the various tubes.

There is one point that may interest you. I have a Bendix tube, TD 36 "Radioactive" that is not referenced in your book. It appears to be used, although not in any obvious fashion. It was a loose tube from a science lab. I believe it to be a gas (cold) voltage rectifier.

Would you perchance have any information on this tube?

Harvey Axelrod
Williamsville, N.Y.

Charles Hansen responds:
The Bendix TD-36 (RETMA type 7416, registered on 23 March 1959) is a gas voltage rectifier as you suspected. **Photo 1** shows the similar TD-82 cold gas rectifier, which, like the TD-36, may have contained tritium gas, an isotope of hydrogen.

Photo 2 is a close-up of another tube, the TG-77 radioactive stabilized spark-gap tube, that shows the box with its radioactive warning label. This tube contains 2 microcuries (μCi) of Cesium-137 (Cs-137). I found it interesting that the label says "Caution," the symbol says, "Warning," around the box to the right, and the tube envelope says "Danger."

Cs-137 has a half-life of 30 years and emits beta and gamma radiation as it decays. Over 200,000 electron tubes containing Cs-137 were produced by all the licensed manufacturers. If you do come across one of these tubes (by way of eBay, for instance), make sure to dispose of it properly because they do contain hazardous radioactive materials. In theory, the glass envelope is thick enough to contain the beta decay particles.

As a means of comparison, the smoke detector in your home contains less than 35 kilobecquerel (kBq), or less than 1 μCi, of Americium-241 (Am-241). The SI conversion factor is 1 μCi = 37 kBq. Am-241 has a half-life of 432 years and emits alpha particles and low energy gamma rays. The alpha particles are absorbed within the detector, while most of the gamma rays escape harmlessly. *aX*

PHOTO 1: The TD-82 cold gas rectifier tube.

PHOTO 2: The Bendix TG-77 radioactive tube.

CONTRIBUTORS

Edward T. Dell, Jr. ("Editorial—"Longevity," p. 6) is editor/publisher of *audioXpress*, *Voice Coil*, and *Multi Media Manufacturer* magazines.

G.R. Koonce ("Passive Crossover Linear Phase Speakers, Part 1," p. 8) is an electrical engineer who has enjoyed the hobby of designing and building audio equipment, test gear, and speaker systems. Professionally, he worked for over 30 years producing audio frequency equipment for military use. He is a member of AES and has been a contributing writer to *Speaker Builder* since 1981.

Ron Tipton ("A Multiple Wall-Wart Power Supply," p. 20) has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at White Sands Missile Range. In 1957 he started Testronic Development Laboratory (now TDL Technology, Inc.) to develop audio electronics. During the 1960s and 70s, TDL built active filters and pseudo-random noise generators for well-known companies such as Bose Corp. and Acoustic Research. He is still the TDL president and principal designer.

Gary Galo ("Is SACD Doomed?," p. 24) 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.

Jim Dungan ("HD-2: A Single-Ended Class-A Headphone Amp," p. 31) has an AAS degree in Instrumentation, and a BS in Industrial Technology. He works for a petrochemical company as an analyzer technician. He owns a very small business repairing guitar amps, jukeboxes, tube radios, and amps.

Jan Didden ("*aX* Visits SEAS," p. 38) lives in The Netherlands and is a longtime contributor to Audio Amateur publications.

The audio hobby for **Alex Arion** ("A Greek Hybrid in Blue," p. 42) began in the magic '60s in Communist Romania in his grandmother's attic, where, together with two friends, he installed a small "illegal" laboratory. Their first attempt was a beginner's guitar amplifier and a professional guitar pickup. The tubes used included ECC40s and EF40s. The rectifier was a 5μ4, Russian equivalent. They mounted all in a hat box that they found in the attic. After finishing his studies and many years of research, Alex moved to Greece, where he continues to build different tube-based and solid-state amps.

Darcy Staggs ("Copper Tubing Capacitor Connections," p. 47) has been interested in electronics and its magic since building a crystal set in grammar school. This elementary model was later equipped with the first commonly available transistor amplifier, a GE 2N107. A college education (MS, engineering) was followed by work on the Apollo Lunar project (simulation of the electrical power system), war gaming, management software for Swedish IBM, weapons system simulation, and successfully raising two sons, who also love good music.

Yard Sale

Wanted

Hi Fi and Stereo equipment, tubes and transistor, kits and factory built (Heathkit, Eico, Scott, Dynaco, and so on) turntables, records, reel-to-reels, tapes. I am a collector, not for resale. Contact Ed at (843) 856-6938 or njvet01@yahoo.com.

"Yard Sale" is published in each issue of *aX*. For guidelines on how subscribers can publish their free ad, see our website.

What's New on the *aX* website?

Articles from Past Issues:

"Horn Theory: An Introduction, Part 2," by Bjørn Kolbrek (*aX* 4/08).

"Everything You Wanted to Know About SPL Meters," by Ethan Winer (*aX* 4/08).

"The Sidewoofer Line Array," by Ken Bird (*aX* 5/08).

"The Mighty Midget," By Pete Millett (*aX* 5/08).

"Valve (Tube) Regulated Power Supplies," by Terry Bicknell (*aX* 5/08).

For information on these and others, visit www.audioxpress.com. And don't forget to check out the links to our other magazines, *Voice Coil* and *Multi Media Manufacturer*!

VENDORS

Ad Index

The MediaGate **MG-450HD** is an entertainment server that allows for wireless streaming of videos, music, and photos directly to HDTV or stereo systems from the PC. Operating on 802.11g wireless networks, the MG-450HD supports HDMI for high-definition output, composite, S-video, and component video outputs. It also acts as a digital media adapter, and can complement the hard drive of choice, letting you customize your media center for your storage needs. For more information, visit www.mediagateusa.com.

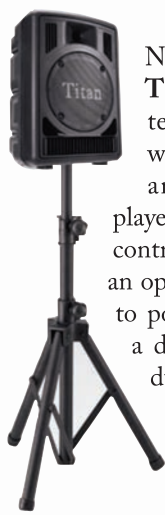
Joseph D'Appolito's definitive work on testing loudspeakers has recently been translated into Italian. *Testing Loudspeakers* is available from Audiomatica S.r.l. Italy, makers of the popular **Clio** analyzer, at www.audiomatica.com, or from www.audioXpress.com.

McFeely's offers an assortment of screws, including the popular flat-heads. In steel there are sizes 6, 8, and 10, from ½″ to 3″, in boxes of 100. There are also clear zinc, yellow zinc, statuary bronze, and black oxide options, designed to match whichever finish you require. Order online at McFeelys.com.

Axiom Audio released a limited-edition slate black version of its **Algonquin** outdoor speakers. The Algonquin is a waterproof, all-weather version of the company's M3 bookself unit, and incorporates a 1″ titanium tweeter and 6.5″ aluminum woofer, housed in a high-impact, tuned plastic cabinet. Featuring anechoic SPL of 88dB, and frequency response of 60Hz-22kHz, the speakers are 13.5″ H × 8.5″ W × 8.25″ D. For more, visit www.axiomaudio.com.

Symwave, Inc.'s "**FirePHY-1600**" solution, based on high speed SERDES technology, enables data transfer rates up to 1.6Gbps, nearly four times faster than the highest speed USB standard today. Target applications include Mac and PC interface ports, PC add-in cards and ExpressCards, external storage devices, high-definition industrial cameras, FireWire hubs, and high-speed Flash card readers. Some advantages of FireWire over other high-speed serial interfaces include the ability to support 45W power over 1394 cables, peer-to-peer networking, and long distance networking over Cat-5, POF, and coax cables. For additional information, go to www.symwave.com.

The DVD Acquisition and Development Group, Inc., has introduced *Digital Video Essentials: HD Basics* on Blu-ray and HD DVD with an upgrade offer available through their website. Designed to benefit HDTV novices and aficionados alike, the disc was created by Joe Kane, HD expert, and it provides easy-to-use, basic calibration tools and setup instructions for getting the most out of HDTVs and offers background information on today's systems. For more information or to take advantage of the upgrade, please visit www.dvdinternational/dveupgrade.



New from TeachLogic, the **Titan** is a portable 50W battery powered sound system with a wireless transmitter and a built-in CD/MP3 player. It features treble and bass controls, a plug-in microphone, an optional auto-link transmitter to powered companion speaker, a dual line input (RCA) and dual line output (RCA), and more. To learn more, go to www.teachlogic.com.