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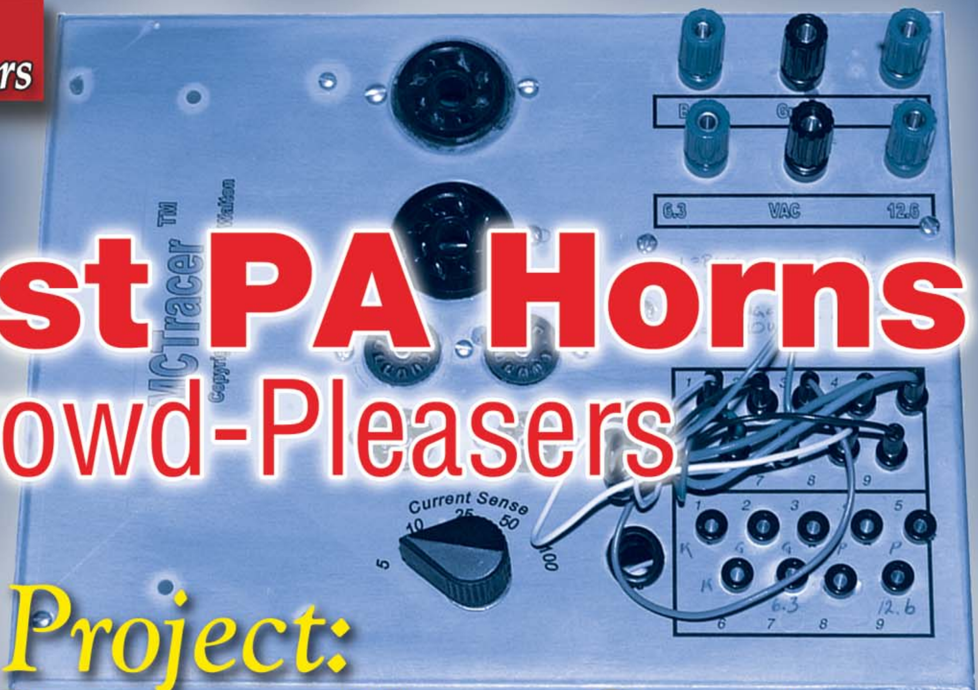

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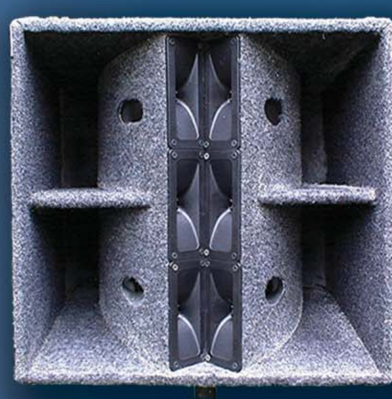

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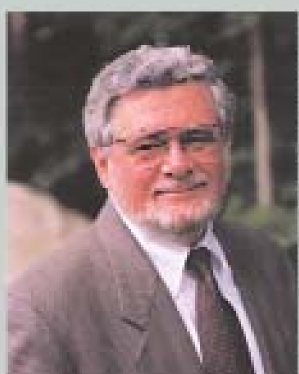

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Dr. Joseph D'Appolito has been working as consultant for Usher Audio since early 2000. A world renown authority in audio and acoustics, Dr. D'Appolito holds BEE, SMEE, EE and Ph.D. degrees from RPI, MIT and the University of Massachusetts, and has published over 30 journal and conference papers. His most popular and influential brain child, however, has to be the MTM loudspeaker geometry, commonly known as the "D'Appolito Configuration," which is now used by dozens of manufacturers throughout Europe and North America.

Dr. D'Appolito designs crossover, specifies cabinet design, and tests prototype drivers for Usher Audio, all from his private lab in Boulder, Colorado. Although consulting to a couple of other companies, Dr. D'Appolito especially enjoys working with Usher Audio and always finds the tremendous value Usher Audio products represent a delightful surprise in today's High End audio world.

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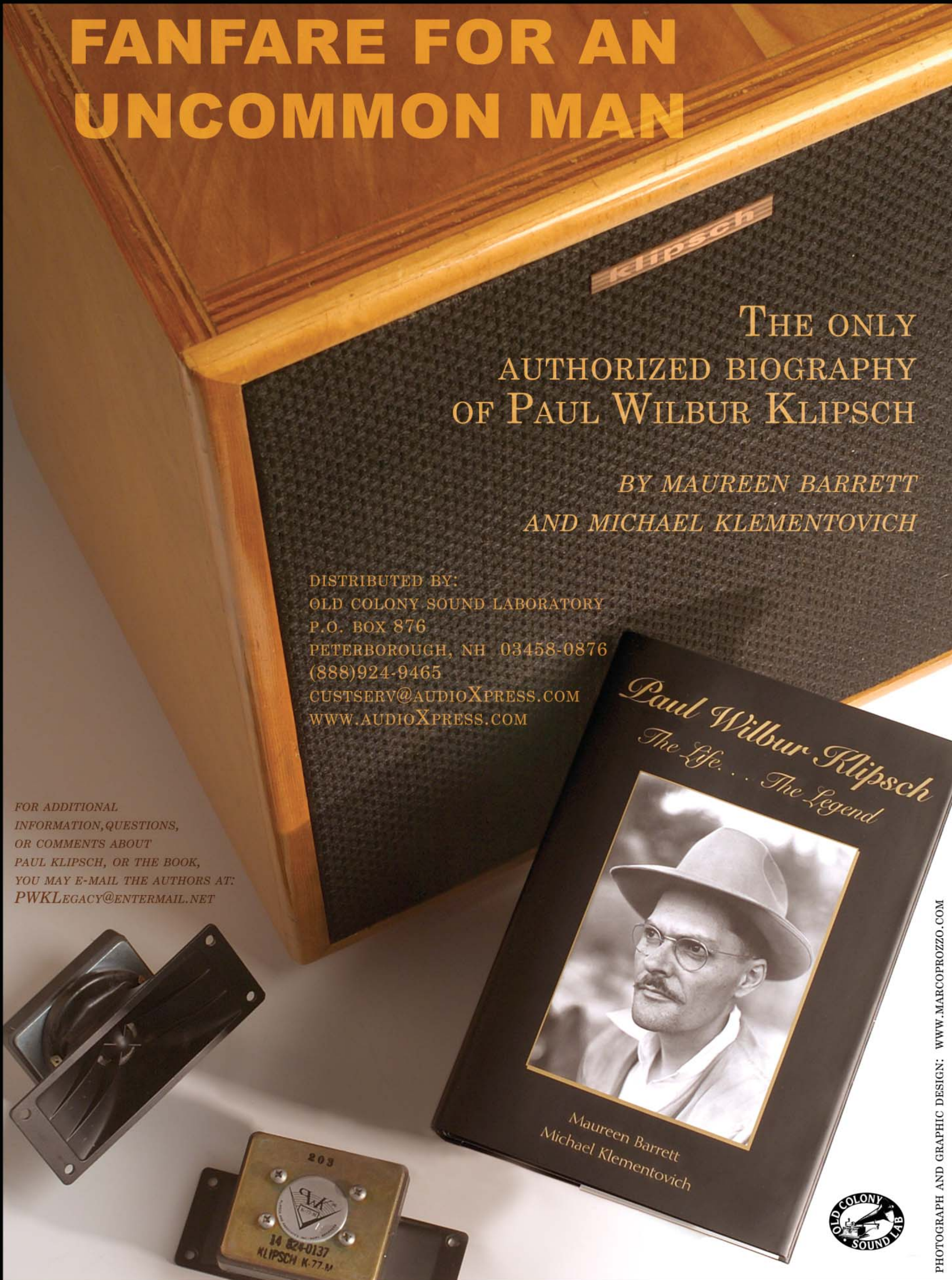
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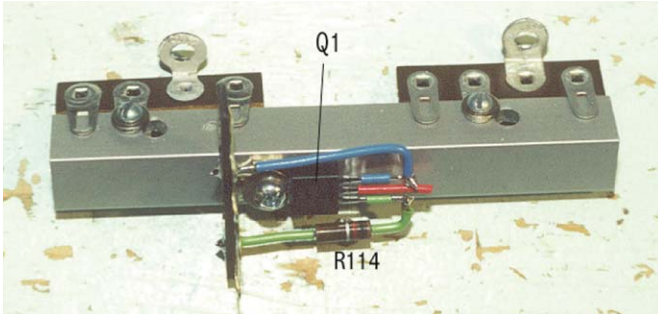

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DEPARTMENTS

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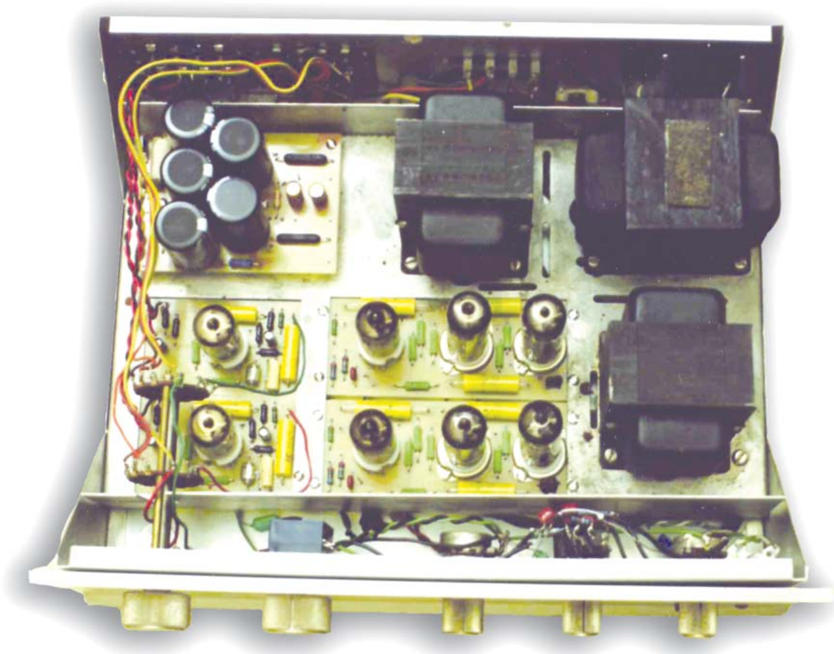

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.

JOHN STUART MILL

Editorial

Where Credit Is Due

We are surrounded every day by evidence of sweeping changes. One of the most pervasive is in forms of communication. Cell phones are spreading like weeds and have become an accessory for driving. Our computers are now the letter and document carriers for all sorts of communication. One effect in our efforts to bring you this magazine is remarkably increased efficiency in assembling and preparing materials for publication. This has produced significant savings in time and money, although a large part of those savings are going into increased delivery costs.

The internet, or the web, is a rising convenience for buying and selling, as well as disseminating advertising and all manner of information. In many ways it is a great blessing whose influence is almost impossible to assess accurately. It is a global phenomenon, and we can surmise that it will change international relationships and penetrate the walls of isolation imposed by despotic governments.

Any new institution brings with it misuses and abuses, however. Junk mail and telemarketing were bad enough. Now we have junk e-mail and worms and viruses. Anything good always has bad uses. We will, I presume, find antidotes for these problems. Meanwhile the web is here to stay. Fully half of retail sales in the US in 2003 were made on the web. You will all be aware that we have also been developing a website for our publications and our product lines. We are happily linked to other sites and are always glad to have others link to us.

The web is affecting us in other significant ways. For the last two years, many of our advertisers have been adjusting their ad budgets to make greater use of the web and smaller spaces in *audioXpress*. This is fully understandable and we realize it is a trend which will continue. It will soon impact you as readers, however. Advertising revenue is linked directly to the number of pages we can publish each month. If ad lineage goes significantly below 40% of the total page count, we will be forced to publish fewer pages and articles. We will be subdividing some articles and placing portions of some of them on our website.

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Since the first issue of *aX* just over three and one half years ago, we have maintained issue sizes from a high of 124 pages to a minimum of 76 pages featuring a "perfect" bound style which includes a printed spine. You will be seeing smaller issues beginning this Fall. Our costs of delivery have risen sharply in the last 42 months, along with other increases. We have, with the cooperation of our printing vendor, effected savings on paper, proofing, and taken advantage of his more efficient presses. Film negatives are now also a thing of the past. We have done our level best to bring you the best technical audio magazine possible with what your subscription dollars and the advertiser's dollars pay for.

I like to believe that what we publish from our worldwide range of authors is fuel for your own creative thoughts. I also hope and believe that this translates into projects that you enjoy constructing for yourselves. Your purchasing power is the reason advertisers pay for space in our pages. The more they realize that *audioXpress* is the engine which stimulates your patronage, the more clear they can be about their priorities in making up their advertising budgets. —E.T.D. ❖

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Room Correction, Part 1

This project to build a DSP preamp will improve the performance of your sound system. **By Rune Aleksandersen**

For a long time, I'd been dreaming about implementing my own digital filter and room correction system. Back in 1998, I sketched out what I wanted to build and named it "Just Awesome," due to the pure thrill of knowing it would be possible to make a digital filter allowing a perfect time response (*Fig. 1*). My main emphasis was a digital loudspeaker filter. But later, I considered a loudspeaker and room correction system.

This design would have been feasible as a hobby project at work, where I am a hardware design engineer, with expertise in both electronics design and digital signal processing in FPGAs. But as always, lack of time is the essential problem in realizing hobby projects. When work is done, there is little time left.

DREAM PROJECT COMES TRUE

Now, years later, the advent of the Signal Wizard (*Photo 1*) has sparked new life into my old project. The Signal Wizard[1] consists of a stereo A/D converter, a DSP processor to process stereo and mono audio signals, and a D/A stereo converter on the output. It has an RS-232 interface for downloading the filters, as well as ready-made software for generating FFT filters from amplitude and phase curves specified by the user. Essentially, I can use the Signal Wizard for making my old dream project come true.

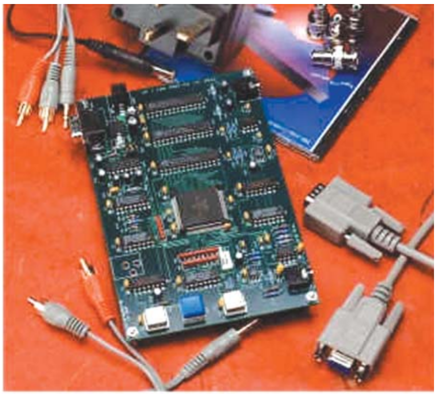

PHOTO 1: Signal Wizard.

Here are some of the features supported by the Signal Wizard:

- FIR filters with a maximum of 1024 coefficients
- Import mode for arbitrary frequency

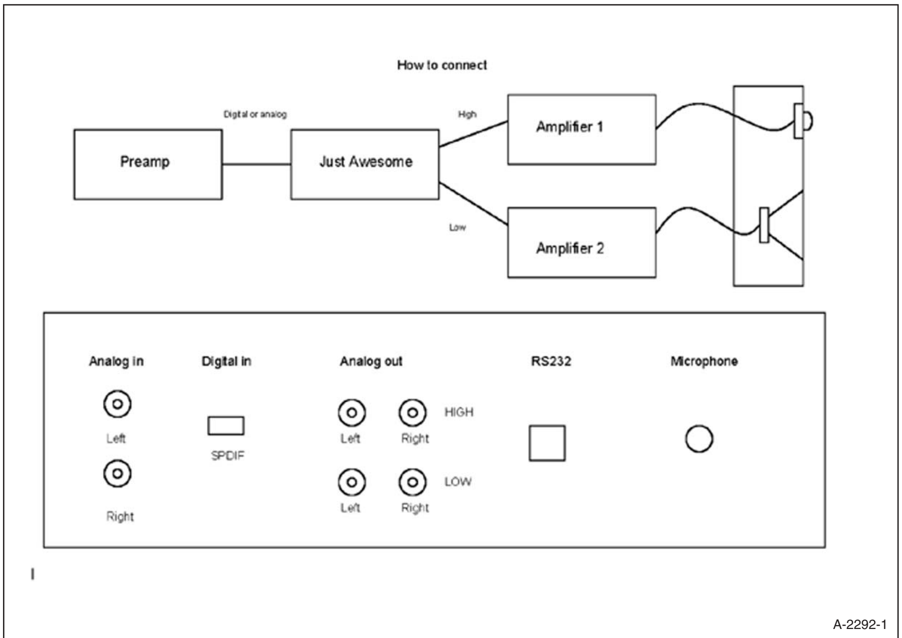

FIGURE 1: Just Awesome system setup.

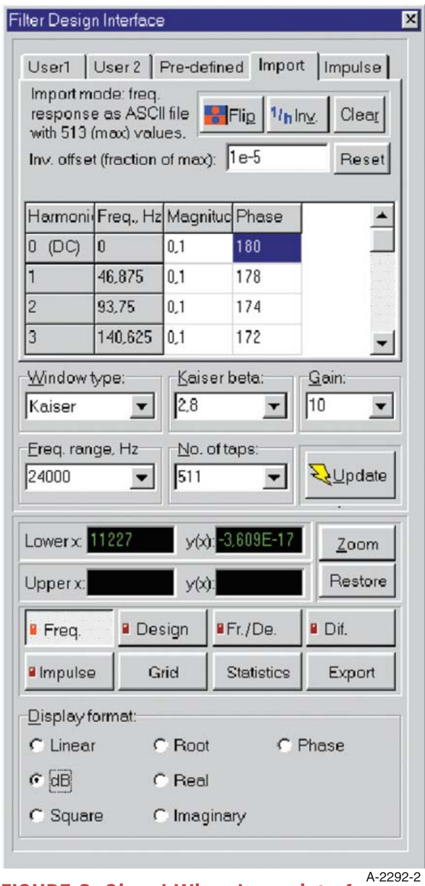

FIGURE 2: Signal Wizard user interface.

ABOUT THE AUTHOR

Rune Aleksandersen lives in Norway, is 36 years old, and has an M.Sc degree in Electrical Engineering and a BA degree in music. He enjoys playing the jazz trumpet and the classical violin. Working as a consultant developing digital and analog electronics, he also enjoys building loudspeakers and audio electronics in his spare time.

or impulse response
- Deconvolution (inverse) or flipped filter options
- Zero-phase distortion or arbitrary phase in the pass, transition, and stopbands
- Choice of rectangular, Bartlett, Hamming, Hanning, Blackman, or Kaiser windows
- Impulse, frequency, and phase response exportable in a variety of formats (dB, power, and so on) as ASCII files for incorporation into standard spreadsheets
- Virtual control panel allowing run-time changes to filter gain and sample rate
- Filter module that holds up to 16 filters, instantly selectable from a click of the mouse
- Single (18-bit) or dual channel (16-bit) modes
- Eleven user selectable sample rates of 48kHz, 24kHz, 16kHz, 12kHz, 9.6kHz, 8kHz, 6kHz, 4.8kHz, 4kHz, 3.2kHz, and 3kHz.
- Normal or special "turbo" mode for high-speed applications.

I have always been fascinated by loudspeaker filters and how they are

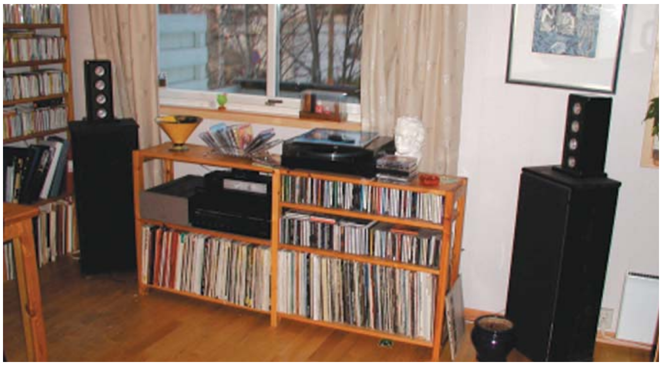

PHOTO 2: Author's current stereo setup.

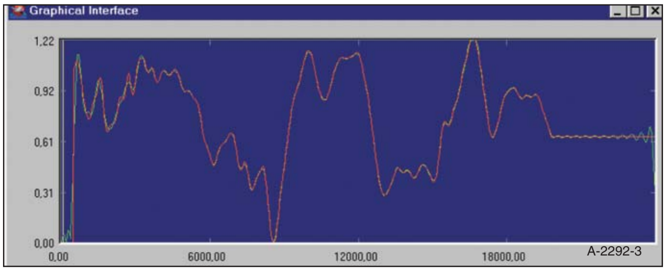

FIGURE 3: Sample frequency response.

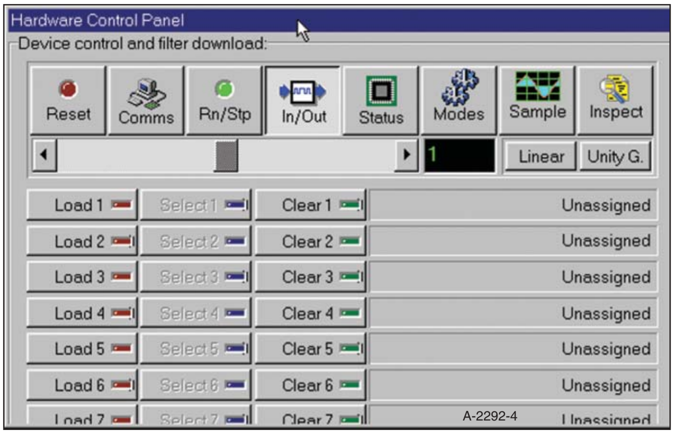

FIGURE 4: Signal Wizard hardware control panel.

implemented. There is also the controversial issue about phase linearity and its audibility that I could now try out on my own. Believing that phase might have some importance, I set up my current stereo system with a first-order crossover and time-adjusted drivers. The system's JX150 wideband Jordan driver is employed in a transmission line setup, while four JX53 tweeters in a closed box are used for the high frequencies. Crossover frequency is 500Hz.

Besides the final target system, I will do lots of testing on my home-built studio monitors. These are closed box speakers featuring the Jordan JX92S driver. In this project, I studied what I could find on the Internet about room correction, theory, and practical solutions applied by manufacturers of such systems.

SIGNAL WIZARD PERFORMANCE

The software interface of the Signal Wizard is very easy to use. There is one filter design interface window where you can specify the actual filter (*Fig. 2*). There are many options, but I will use the import mode where you can

PHOTO 3: Author's home-built studio monitors.

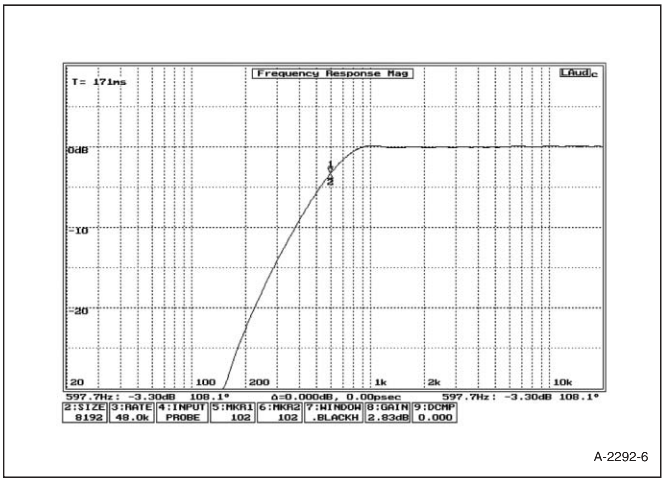

FIGURE 6: Built-in quick filter. High-pass 48kHz, 121 taps.

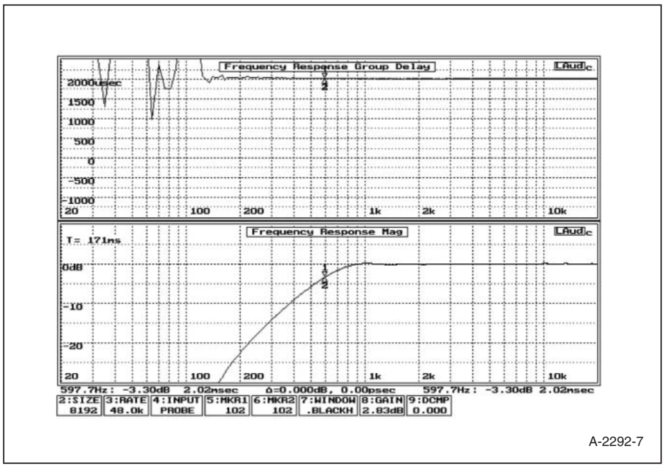

FIGURE 7: Built-in quick filter. High-pass 48kHz, 121 taps.

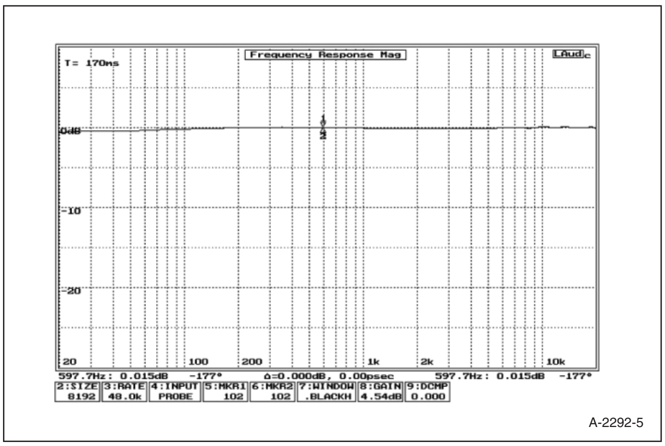

FIGURE 5: Filter frequency response in bypass mode.

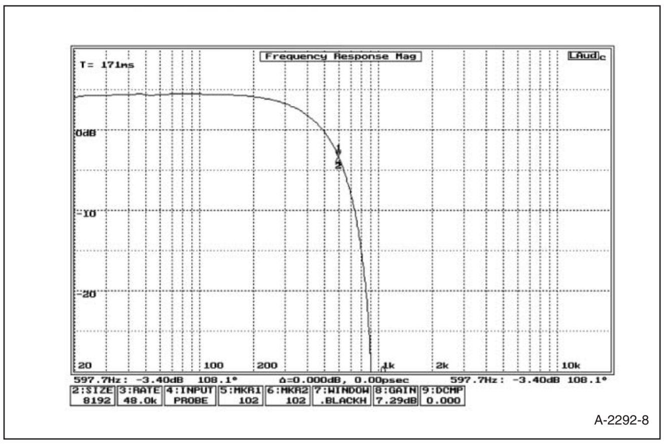

FIGURE 8: Built-in quick filter. Low-pass 48kHz, 121 taps.

enter arbitrary amplitude and phase. Text file formats for importing files are supported.

Figure 3 shows the frequency response window of one of my imported files. The red response is my desired frequency response. Overlaying this is the green curve showing the synthesized filter response. Matching is very close.

The Signal Wizard offers only a linear view of the frequency, which I find a bit difficult to work with. I would have preferred a log view of the frequency data, because this is the format I usually work with. Amplitude, however, can be viewed in dB format.

After designing the filter file, you download the filter using the hardware control panel (*Fig. 4*). This window supports lots of filters in memory, making swapping filters easy. Simply click the select button, and the new filter is activated.

The number of taps the filter can execute (*Table 1*) depends on what you select for sample rate, the mode (dual or single channel), and whether you select the turbo mode.

To check how the Signal Wizard actually performs, I used the LAUD measurement system for some test measurements. First I checked the filter response in bypass mode (*Fig. 5*). The filter is very linear in both phase and amplitude. The phase is 180° inverted for some reason. Because the phase is arbitrarily entered by the user, it can easily be inverted in the software suite that comes with the product.

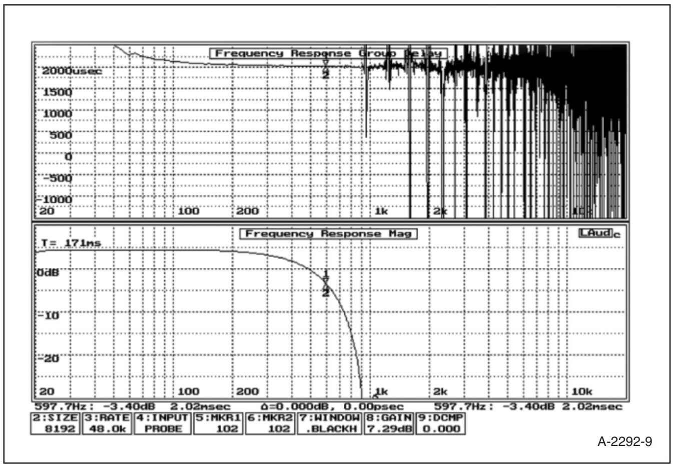

FIGURE 9: Built-in quick filter. Low-pass 48kHz, 121 taps.

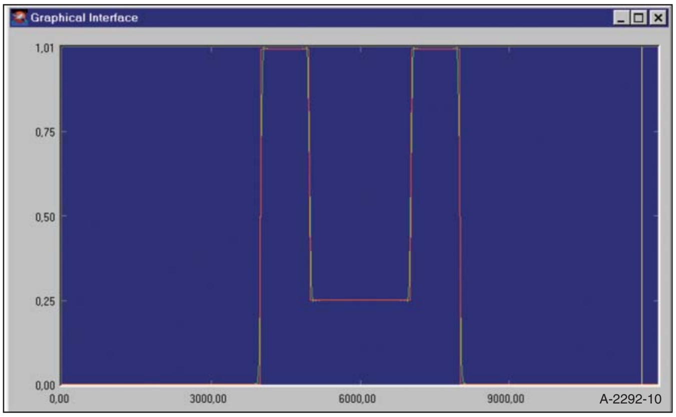

FIGURE 10: Signal Wizard filter in tutorial example.

TABLE 1:
NUMBER OF TAPS THAT THE SIGNAL WIZARD CAN PERFORM

FREQUENCY RANGE	SINGLE CHANNEL NORMAL MODE	SINGLE CHANNEL TURBO MODE	DUAL CHANNEL NORMAL MODE	DUAL CHANNEL TURBO MODE
24000	425	527	141	191
12000	731	937	293	397
8000	1024	1024	447	601
6000	1024	1024	601	805
4800	1024	1024	755	1011
4000	1024	1024	909	1024
3000	1024	1024	1024	1024
2400	1024	1024	1024	1024
2000	1024	1024	1024	1024
1600	1024	1024	1024	1024
1500	1024	1024	1024	1024

After the first check, I ran high- and low-pass brick wall filters using the built-in functions of the Signal Wizard software called Pre-defined filters (*Figs. 6, 7, 8,* and *9*). The Windowing function is Kaiser 1.0 and 127 taps, and the sampling rate is 48kHz. The rate of the filters is steep, and the phase is linear as shown in the LAUD plots.

PHOTO 4: Closed box speaker featuring the Jordan JX92S driver.

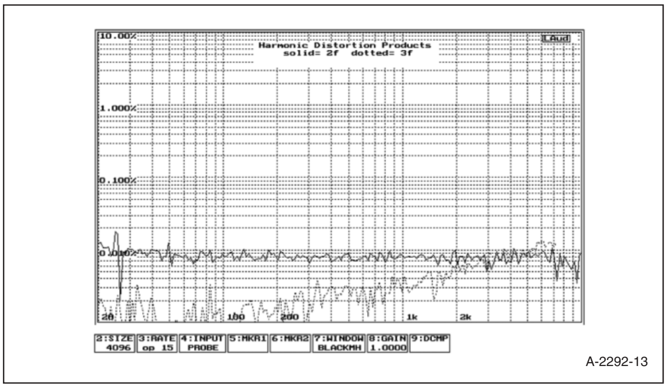

FIGURE 13: Second- and third-order distortion measurement.

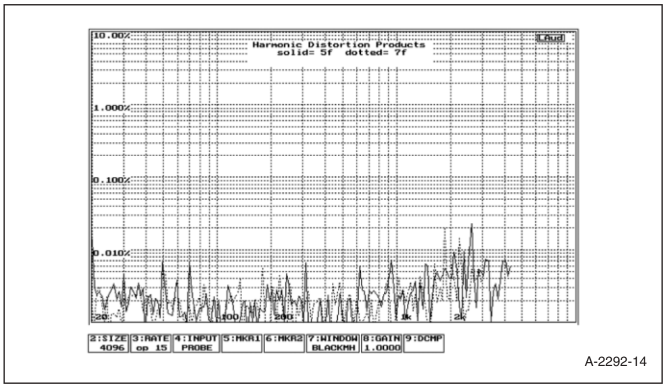

FIGURE 14: Fifth- and seventh-order distortion measurement.

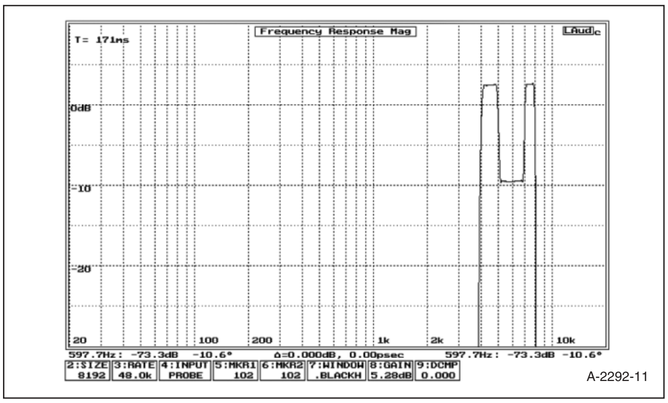

FIGURE 11: Measured filter in tutorial example.

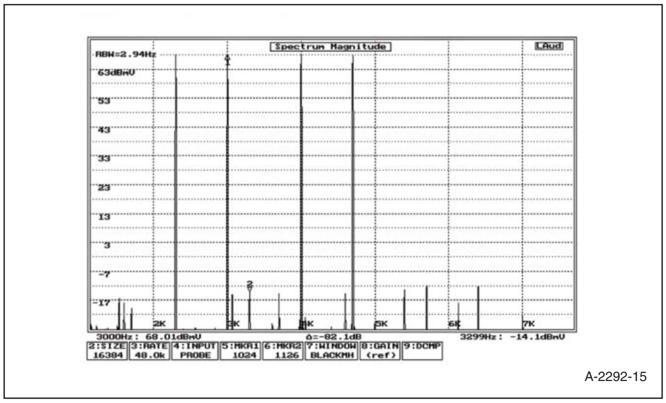

FIGURE 15: IMD measurement.

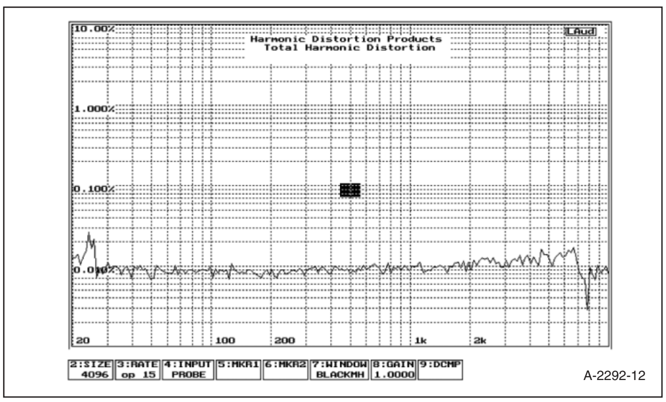

FIGURE 12: THD measurement.

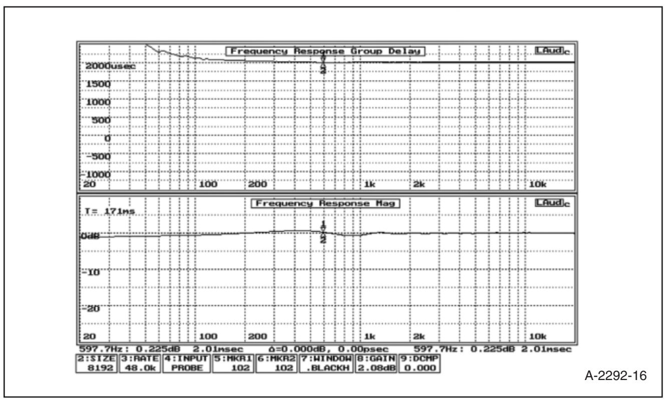

FIGURE 16: Combination of high- and low-pass filters.

Tutorial example 3 was run and implemented with the filter (*Figs. 10* and *11*). The LAUD measurement corresponds exactly to what the software predicts (frequency axis is linear view in Signal Wizard, logarithmic in LAUD). Imagine this implemented with an analog filter, which is not very likely ever to happen.

Distortion measurements show THD at less than 0.01%, which is an excellent number (*Fig. 12*). This is according to what the data sheet specifies for the A/D and D/A converters. The distortion seems to be mainly second order of nature, which is the most ear-friendly sort of distortion (*Figs. 13* and *14*). Intermodulation distortion measurement shows products down below –80dB (*Fig. 15*), which is also a respectable result.

Imaging the use of the filters as loudspeaker filters, I ran the high- and low-pass parts through a Behringer MX802A mixer. The result of the filters looks good (*Fig. 16*), even though the low- and high-pass are not optimized to fit together. The pre-defined filters were used.

With this background material in place, we can move on to addressing the issues of loudspeaker response and room correction in Part 2. ❖

PHOTO 5: Computer setup used to access the Internet for information about room correction theory and practical solutions.

REFERENCE
1. Signal Wizard website: http://www2.umist.ac.uk/dias/PAG/signalwizard.htm

Norman Crowhurst's Twin Coupled Amp … And Beyond

Drawing on the work of a noted amp designer, this project is an exceptional achievement in quality sound reproduction at a low cost. **By John Stewart**

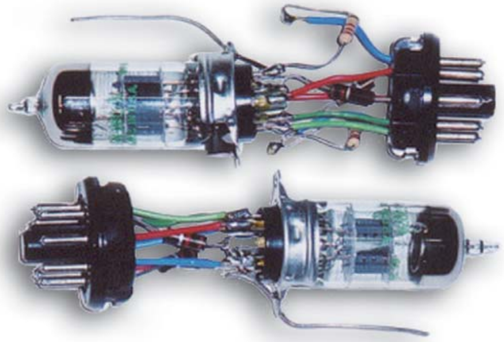

PHOTO 1: 6BQ7 family adapters.

By the late 1950s, Norman Crowhurst had been asked what kind of circuit he would recommend that could provide high-fidelity reproduction while at the same time be possible to build at a reasonable cost. At the time, the top of the heap was populated by amplifiers such as Williamson, McIntosh, Dynaco, Quad, Brooks, and others. All provided good performance, but in some ways were perhaps too expensive or beyond the capabilities of the ordinary experimenter.

Over time, Crowhurst published a series of articles explaining the various circuit configurations available to the designer along with a summary of their strong and weak points[1,2,3]. Out of all this came a number of points worth noting.

ABOUT THE AUTHOR

John Stewart, a professional Engineer (Electrical) has finally retired after 35 years in the sales arena, primarily with Rohde & Schwartz and Hewlett-Packard. He gained design experience with vacuum tubes while working for the University of Toronto, Physics in the '50s and '60s.

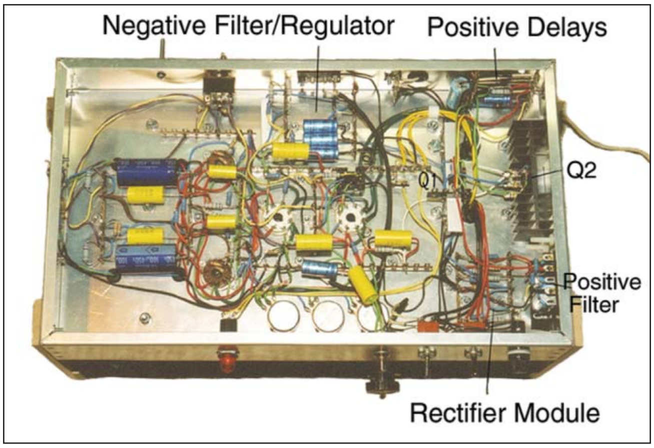

PHOTO 2: Twin-coupled amp bottom view.

- Some amplifiers could become unstable under certain load and signal conditions.
- Higher order harmonics are more objectionable than those of lower order even when at the same level.
- Pentode and beam power tubes result in more high-order distortion and lower damping factors (DF) than triodes.
- Application of negative feedback (NFB) to remedy some of the amplifier shortcomings produces unexpected results.
- In order to apply NFB successfully, you need an output transformer with a large bandwidth.
- Blocking of the signal at the output stage in RC coupled, fixed bias amplifiers is a problem at the clipping point.

The amplifier I will describe here is targetted for those who have had some experience with vacuum tube amplifiers. As usual, you should take great care when building this amp. There are 500V inside the chassis. You must be careful not to touch the heatsinks—a safety issue those with solid-state experience may not be accustomed to.

WHAT TO DO?

In most audio amp designs, the origin of the greatest distortion is in the output stage. In a pentode or beam power output stage the damping factor is for the most part missing, since they are current sources. In the Dynaco, these problems are overcome with a liberal application of NFB, both full loop and by way of the ultralinear (UL) connection. For the Dynaco that is a good fix, since very few stages are within the FB loop, while the output transformer has wide bandwidth.

That kind of remedy doesn't work as well for the Williamson, since there is one more stage of gain (and phase shift) to get in the way of stability. As a result, some Williamsons have what we call "Conditional Stability." In these situations the Williamson is known to have bursts of ultrasonic and sub-audible oscillation.

The same problem occurs in many other amplifiers as well. These may not show up in ordinary testing since for the most part tests are done into a fixed resistive load. In the real world the load is complex, that being a loudspeaker in its box. Pentodes and beam power tubes especially don't like complex loads. All of this is said to result in listener fatigue.

Another problem you have in a pentode or beam power output stage is a large variation in stage gain. The simple formula for stage gain in these amplifiers is Gm × Zl (tube transconductance multiplied by load impedance). For all amplifiers intended to drive a loudspeaker, the load impedance is a function of frequency. Then it is possible the amp gain may be several times as great at some frequencies as it would be at the nominal load. That is OK as

long as you don't try to apply NFB.

If you have set the NFB to be 20dB at midband, it may be 35dB or higher where the loudspeaker is resonant. That is a serious concern at the lower frequencies since the amplifier may already have some of its own phase shift, which will contribute to instability. The same holds true for stability problems at the high end of the audio spectrum.

Beyond connecting beam tubes and pentodes for straight triode operation, a number of output stage configurations have been devised to mitigate these shortcomings. The most recognizable is probably the ultralinear. Here the load is arranged so that part of the output signal is routed back into the screen grids as NFB.

Both 20% and 43% of the output transformer primary winding have been used successfully as NFB in many amp designs over time. There is nothing magical about these numbers; you could choose almost any percentage up to 50% of the primary winding, depending upon the designer. Some call this partial triode operation, because the tubes are run somewhere between triode and pentode mode.

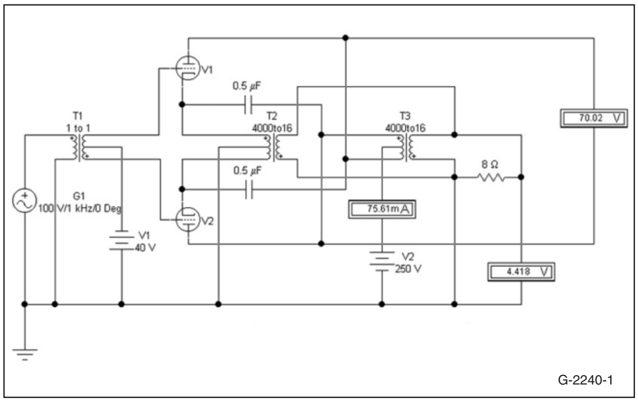

FIGURE 1: Basic circuit—Twin-coupled amp by Norman Crowhurst, Circa 1957.

Distortion is reduced in the output stage while a DF of about one is achieved with the 43% UL connection, before the application of the main NFB connection. The variation of gain in a UL stage with 43% screen NFB when the load impedance is raised from nominal to open circuit is only about 2:1. That is about 3dB change and is most effective in keeping the amplifier stable under normal loudspeaker load conditions.

Local NFB in the output stage is also possible through the cathodes. That is what is done in the McIntosh[4] and Quad. Cathode FB in the McIntosh is 50%, while in the Quad it is about 10%. Both need a specially wound transformer with two center-tapped primaries, which becomes expensive, so you won't find many private experimenters building those circuits. Another form of cathode NFB in the output stage is the Circlotron[5]. However, that circuit needs two separate high voltage (HV) power supplies, so it is not common.

The Circlotron is a good configuration if you want to build an output transformerless (OTL) amp, primarily because it can be made fully symmetrical.

THE TWIN-COUPLED AMP

Crowhurst's solution to most of these problems is unique. His starting point was the McIntosh configuration where the output load was split equally between the anodes and cathodes[6,7,8]. That in itself will provide a DF of 1.5 to

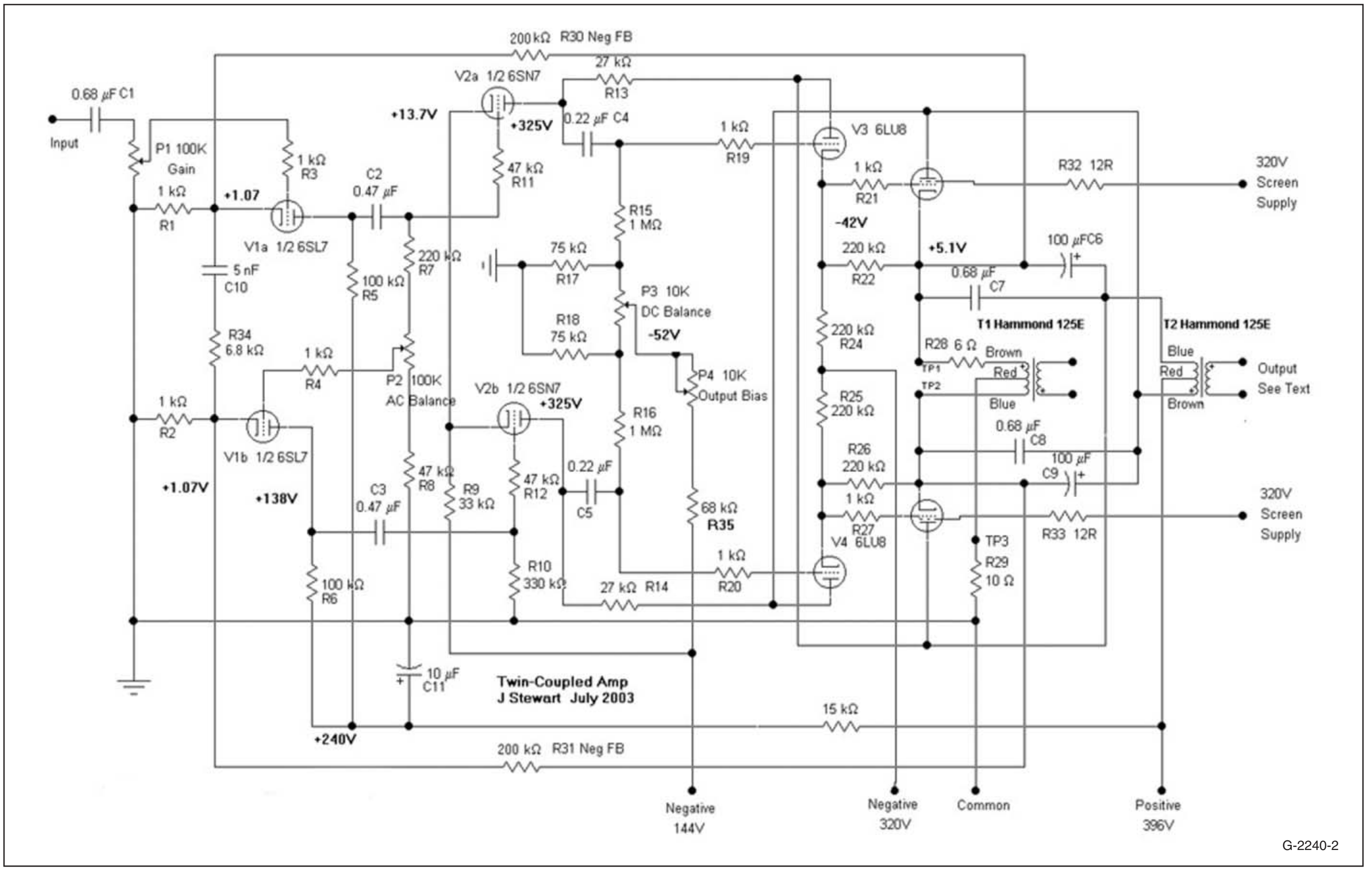

FIGURE 2: Twin-coupled amps.

4 when using pentodes or beam tubes before applying full loop NFB, depending upon circuit constants. In *Fig. 1* I have used triodes for the sake of clarity to illustrate how his circuit worked.

The complete original circuit was published in the May 2003 issue of *audioXpress*[9]. Rather than using an output transformer with a bifilar wound primary of extended bandwidth, he used two ordinary output transformers, which did not require extended bandwidth, since they would not be included inside the NFB loop. However, they did require a special turns ratio. That is because half the load is in the cathode, while the other half is in the plate circuit. In Crowhurst's original circuit, each transformer's secondary delivered half the power to the external load in a parallel connection.

In the simplified schematic, each of V1 and V2 is an imaginary triode with a mu of 5.4. Tubes used in the original circuit required a plate-to-plate load impedance of 8k. That means that each of his output transformers should reflect 4k back to the tubes. The secondaries would each need to be 16Ω to match to an 8Ω loudspeaker.

Crowhurst was unable to find a part to his liking. The universal transformers available to him were too small. As a result, he had some made up for his project. There are suitable universal output transformers on the market today that can do this impedance transformation and are able to handle considerable power.

If a 100V signal is applied to the grids of V1 and V2 as in the schematic, the result at the output transformer primaries will be about 70V. With the impedance ratio of each transformer set at 4000 to 16, you would get about 4.4V into an 8Ω load. All this is based on ideal transformers.

Crowhurst's circuit depends on good coupling of the two secondaries to each of their primaries to distribute the audio load equally between the plate and cathode circuits. At the higher frequencies he has overcome leakage reactance effects by using 500nF capacitors to tie the plate and cathode signals together.

The original circuit used 6BQ5/EL84 pentodes so that each of their screens was wired directly to the opposite plate. That way they still operated as pentodes since the screens were held at a constant potential from their respective cathodes. It was also push-pull throughout, including the driver and predriver. As a result, Crowhurst was able to use

PHOTO 3: Twin-coupled amp front view.

push-pull NFB from the output cathodes to the amplifier input.

Notice also that the output stage needs a large driving voltage. That problem needs to be solved in any amplifier where part or the entire load is in the cathode circuit. Bootstrapping of the driver stage is used successfully in both the McIntosh and Circlotron. Crowhurst uses it in his twin-coupled amp as well.

MY VERSION

I wanted to develop a circuit that would include as many of Crowhurst's ideas as possible. At the same time its cost needed to be reasonable without compromising performance. All of the parts in the resulting design are available as new or on the NOS market. I did not select any tubes with special properties for this project; they are all NOS, right out of the box (*Fig. 2*). The output transformers are a pair of Hammond 125Es, which used together are conservatively rated at 30W, and many impedance ratios are possible.

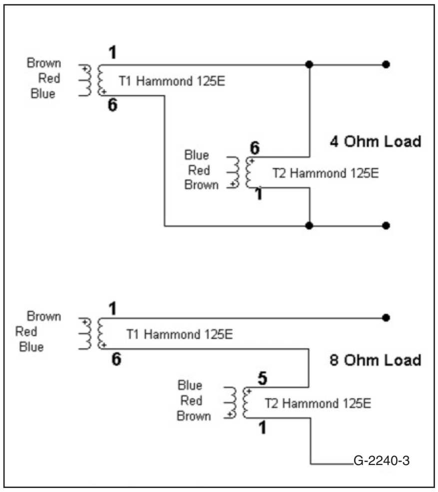

FIGURE 3: Speaker connections.

Based on the experience I had with "The Affordable SE Amp"[10] and its follow-up, "More Power for the SE Amp"[11], I selected a pair of 6LU8s as the output tubes. A pair of 6LR8s would also work in this circuit. Each contains a high mu triode and a beam power tetrode, originally meant as oscillator and amplifier for the vertical deflection systems in color TV. This may seem a bit odd to some.

The beam power tetrode in each has a plate dissipation rating of 14W. However, that is while in a vertical deflection circuit, where there is added stress from large voltage and current excursions. In an audio circuit you can safely increase the maximum plate dissipation by 40%. There are several examples of that in the RCA Receiving Tube manuals, such as the 6K6GT and 6W6GT, which gets us up to just under 20W.

The published ratings are also known as the Design-Maximum Values. These are limits set so that if the power line should go high the tubes would not be in danger. Since this amplifier also has a regulated power supply to look after line voltage variations, you can run safely at close to the maximum specs. With that kind of dissipation possible, you can look forward to considerable audio output from these tubes.

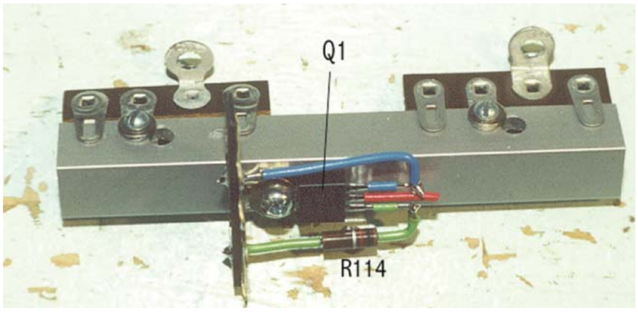

PHOTO 4: Q1 heatsink.

You should set the total cathode current of the output tubes at about 100mA by connecting a voltmeter to the test point TP3 and common. Adjust potentiometer P4 so that a reading of 1V results.

I wanted to avoid the effects of blocking, one of the concerns of Crowhurst mentioned previously. When an RC-coupled stage runs into clipping, the control grid conducts and increases the negative charge on the grid side of the coupling cap. That way the control grid becomes progressively more negative and cathode current decreases.

In a cathode-biased stage the effect is minimized by the automatic bias. However, in a fixed-bias stage it is possible for the control grid to completely cut off the flow of cathode current so that the signal is blocked. This effect is accentuated by NFB as it tries to push the signal through the blockage. The result is serious distortion of the signal for a time during and after the blockage until the bias returns to normal on the control grid.

Crowhurst's solution for blocking in a fixed-bias stage is to direct-drive its control grid with a cathode follower. That is where the high mu triode in

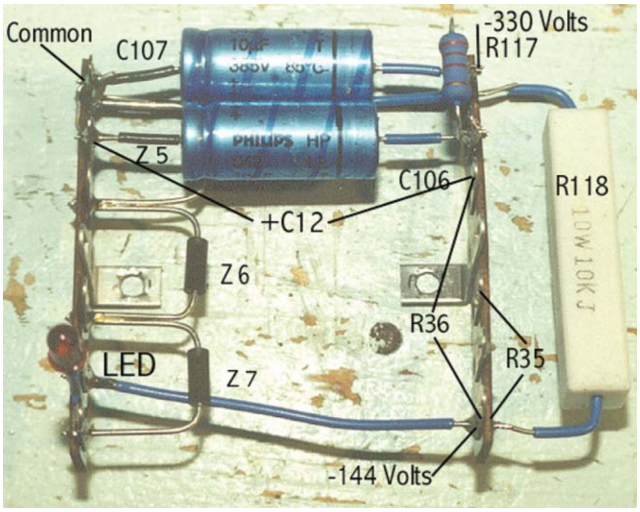

PHOTO 5: Negative filter/regulator.

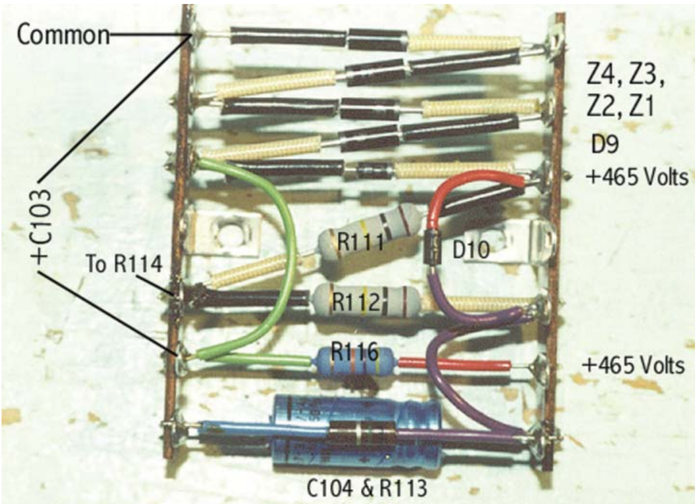

PHOTO 6: Positive delay.

each of the 6LU8s is used. All of this in one tube at a very good price.

When the power stage is generating an output of 25W, there will be about 200V RMS across the transformer primaries. Since the output tubes are running as partial cathode followers, the total signal required at the 6LU8 triode grids is even larger. I measured 250V at this point, which means you will need plenty of gain to result in a reasonable input voltage for the amplifier after NFB is applied. By using a 6SL7GT as the input/phase inverter and a 6SN7GT as the bootstrapped driver, the input required for full output is only about 1V.

I used the same input stage as in Crowhurst's first version. However, I like to use a differential amp with its cathode resistor returned to a reasonably large regulated negative power supply as a driver wherever I can. That minimizes any differences in the drive to the output. In Crowhurst's time the diff amp was well known. However, the negative supply you needed was not as easily implemented as now with our solid- state rectifiers, so was seldom seen in audio amps.

The NFB is taken from the cathodes of the 6LU8 output tubes, so that the Hammond 125E output transformers are not included in the NFB loop. In his original article, Crowhurst states that NFB from the primary of the output transformer is just as effective in reducing distortion due to transformer magnetizing current as it is from the transformer secondary. By taking the NFB from the cathodes rather than the plates, you avoid the use of a pair of DC decoupling caps as well, since the 6LU8 cathodes are barely off ground potential.

You can adjust the output AC balance for minimum distortion by adjusting potentiometer P2. If you do not have equipment for measuring distortion, then it is best to simply connect R4, R7, and R8 at the junction where P2 is located.

I breadboarded an amplifier following the previous steps. It worked well and provided an audio output of about 18W. In his series of articles covering various circuit configurations, Crowhurst mentioned yet another possible hookup for the power stage. Instead of returning the screen grids of the power tubes to the opposite plate as in the McIntosh and the twin coupled amps, you can connect them directly to the HV supply.

That opens the possibility of running the plates at a higher voltage where more audio power will be available. Another result is that you now also have a 50% UL connection. That is exactly what I did here.

The maximum power in the midband at clipping using this arrangement is more than 35W. This is also the circuit arrangement that Crowhurst identified as the best all-round output stage configuration. The third and higher order harmonics are almost eliminated at ordinary levels. Furthermore, in order to suppress parasitic oscillations, I have included resistors in all of the grid leads.

THE OUTPUT TRANSFORMERS

Even though I have followed for the most part Crowhurst's recommendations, there are still some problems that need to be considered. One of these is the high-frequency rolloff in any transformer response. This rolloff is caused by leakage reactance and the self–winding capacity of the primary.

In order to get a reasonable high-frequency output, it is possible to modify the feedback loop. This is accomplished by the combination of C10 and R34 in the circuit of the 6SL7 input cathodes. This CR pair reduces the feedback at the higher frequencies, just as a similar circuit does in Crowhurst's original version of this amplifier.

Another transformer problem that needs to be solved is that of the interwinding capacity, which is between the primary and secondary. This is not the same winding capacity mentioned earlier.

For most low-cost audio transformers, you will find the secondary winding is wound around one end of the primary. Whatever circuit you might connect this to acts as though an additional capacity has been connected to one end of the circuit. In the Hammond 125E that amounts to more than 200pF. If you were to test for this parasitic capacity, you would find it at the end of the winding connected to the brown lead. For a single-ended circuit you would simply connect the primary so that this extra capacity appeared at the power supply lead of the transformer, away from the plate connection.

However, in a PP circuit there will be considerable imbalance, since the extra capacity will show up on one of the signal leads. As a result, you need to connect the output transformers in this circuit so that both their primaries and secondaries are connected in opposite phase to each other. You can see on the schematic (*Fig. 2*) that the primary leads of the transformers are connected in such a way that the parasitic capacities will balance. If you are unfortunate enough to make the other possible connection, you will find response to a square wave to be unsymmetrical.

Output is also considerably reduced before distortion shows up. Because the primaries are connected in opposite phase, you will need to connect the secondaries so the resulting signal adds in order to drive your loudspeakers properly (*Fig. 3*).

The last problem to be solved has to do with the difference of DC resistance between each half of the primary in the cathode circuit. In the case of the Hammond 125E this difference is about 6Ω, so I have paralleled a pair of 12R, ½W resistors to account for this difference.

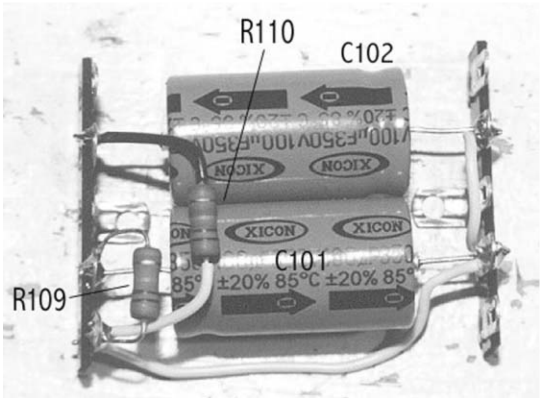

PHOTO 7: Positive filter.

It is now possible to make a measurement of the output tube's DC balance by connecting a voltmeter between the test points TP1 and TP2 and adjusting pot P3 for a zero reading. While doing this, be sure there is no signal present. During normal operation there are very large AC voltages between these two points. It is best to set the gain control to minimum.

THE ZERO FEEDBACK AND OTHER OPTIONS

Some experimenters prefer to run their amps with no full loop feedback. That mode was not possible with Crowhurst's circuit as originally wired, because in his circuit the feedback connections formed part of the phase inversion circuit as he described in his first article on the twin-coupled amp.

There is a way you can reconnect the 6SL7GT stage so that the no feedback option is possible. You can rewire with only a few changes so the first stage is connected in the differential mode. The –144V supply is already available to satisfy the small cathode-current requirement. The other change has to do with the second grid of the 6SL7GT. If left as originally connected, the first stage will form an oscillator.

In the revised circuit shown in *Fig. 4*, I have labeled the components with the same identifiers as used in the main amplifier of this article. The grid which was formerly part of the FB circuit is simply returned to the common bus through its 1k grid stopper resistor. Interestingly, it is also available if you should have an idea about bridging a pair of these amps in order to increase the available audio power. You will have both inverting and non-inverting inputs available. That possible hookup is shown in *Fig. 5*. You could also bridge a pair of these amps at the output since the output transformer connections are floating.

Another option you might choose to try with this differential input stage version of the amp is a feedback circuit similar to that used in the original. Using the same 200k resistors R30 and R31, about 8dB of negative FB results. The distortion is reduced and there is an improvement in the damping factor.

TABLE 1
TWIN-COUPLED AMP PERFORMANCE SUMMARY

FEEDBACK IS CONNECTED

CONDITION	WATTS	2 HARM	3RD	4TH	5TH	THD	DF	INPUT V RMS
6SL7/6SN7 Driver	1	~	0.14	~	0.09	0.14	2.97	
1kHz	10	0.13	0.07	~	0.02	0.16		0.45
	30	0.19	0.10	0.05	0.06	0.25		0.76
100Hz	30	0.26	0.16	0.06	0.08	0.34		
40Hz	1						3.13	
40Hz	30	0.64	0.33	0.14	0.23	0.70		
6BQ7/6BQ7 Driver	1	0.09	0.11	~	~	0.12	2.42	
1kHz	10	0.22	0.21	0.04	0.02	0.22		0.58
	30	0.17	0.19	0.20	0.16	0.34		1.00
Differential Front End								
No Feedback	1	0.20	~	~	~	0.36	1.80	0.12
	10	0.40	0.20	0.10	0.15	0.60		0.40
	30	0.56	0.67	0.14	0.37	1.00		0.70
With 8dB FB	1	0.06	~	~	~	0.15	2.52	0.33
	10	0.12	0.15	~	~	0.20		1.04
	30	0.15	0.23	~	0.16	0.40		1.80
Test Equipment	~	~	0.10	~	0.07	0.055	HP200CD/HP334A	
Residual @ 10V p–p 1kHz	~	~	0.08	~	0.07	0.04	GAG–810/HP334A	

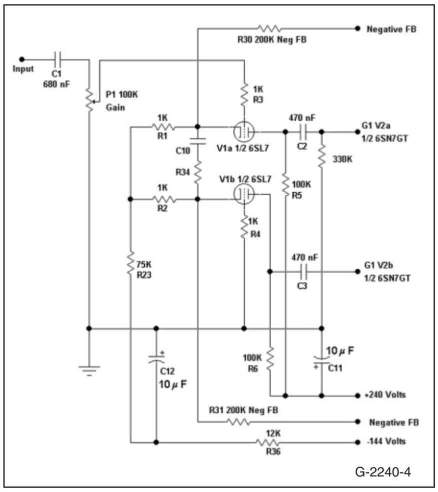

FIGURE 4: Alternative front end for twin-coupled amp.

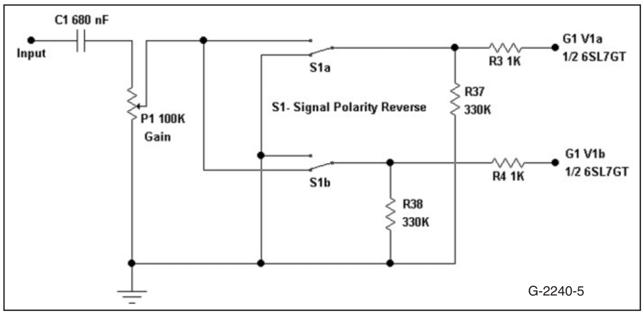

FIGURE 5: Signal polarity reverse.

All of the test results for sensitivity, damping factor, total harmonic distortion, and distortion products for harmonics up to the 5th are given in *Table 1*. Performance of the test equipment is included for comparison.

FURTHER TESTING

I had been curious for some time how tubes of the 6BK7/6BQ7/6BZ7 family would compare as voltage amplifiers and drivers to those you normally use in audio. These are all available on the NOS market at very good prices. To make a comparison using this amplifier as a test set, I wired up a pair of adapters so that I could plug tubes of this family straight in. The octal plugs are made by Amphenol (*Photo 1*).

One of the adapters is wired so that each element of the tube simply goes to its corresponding connection in the octal socket on the chassis. This adapter and tube substitutes for the 6SL7GT. Since these tubes have a somewhat higher mu than the 6SN7GTB, I put a 2.2k resistor in series with each of the cathode leads on the other adapter. That reduced the gain of the other 6BQ7 in test to about the same as the 6SN7GTB that it replaced. It also provided a degree of linearization of the 6BQ7 characteristic in order to reduce distortion.

In order to prevent parasitic oscillation, I also installed 1k, ½W resistors in all of the grid leads on the adapters. Refer to the test results, which are better than I had anticipated.

THE POWER SUPPLY

At first glance the power supply (PS) may appear to be somewhat complicated, but for a good reason. Refer to *Fig. 6*. I always like to include a standby mode in my various designs. However, that can cause a problem when you are working with an RC-coupled fixed-bias amplifier. You must be sure the negative grid bias is applied to the output tube grids before the plate and screen voltage arrives. Otherwise, a large cathode-current pulse of some duration will occur.

This kind of problem is not as serious in a cathode biased power amp. As the cathode-current increases, the bias does also, so that the circuit is self-regulating. I wanted to include this function without resorting to mechanical relays or using any kind of thyristor which could be a source of interfering RF noise.

The remedy may seem simple enough since the positive and negative voltages from the power supply can be readily made to increase at about the same rate. However, that will not solve the problem. The output tube grids are isolated from their bias supply by large-value grid resistors, while at the same time they are being pulled positive by the driver tube's plate resistors through the coupling capacitors. A large cathode current in the output tubes at turn-on is a sure thing and over time will reduce their life.

A possible solution is through the use of a pair of high-voltage power FETs, in this case IRF840s. I had originally tried using a single FET but found that at turn-on there was more than 500V applied to this part of the circuit. That would not be good for reliability since the IRF840 is rated for 500V maximum from source to drain so that there is now a pair of FETs (Q1 and Q2) in a series connection.

The positive high-voltage rise is slowed by the charging of the 22µF capacitor C103 through the 47k resistor R116. By connecting a series of four 100V zener diodes (Z1 to Z4) across C103, the output voltage of the power supply is limited to just under 400V. Regulation and ripple reduction resulting from this combination of parts is very good. To help prevent possible RF oscillation by the FETs I have inserted a 1k ½W resistor into each of their gate leads (R114 and R115).

When the standby switch is closed, more than 500V is applied to the pair of FETs. It is equally divided by the 100k resistors R111 and R112 so that each FET sees about half of the total. However, as C104 charges, the first FET Q1 will absorb a progressively smaller portion of the total voltage drop. A 1M resistor R113 sets the lower limit.

By doing this I was able to use a very small heatsink on Q1, in this case a small strip of aluminum extrusion I pulled out of my junk pile. Q2, the other FET, absorbs most of the voltage drop during normal operation and is mounted on the Wakefield heatsink.

Sometimes a designer will overlook what may happen when power is switched off. In this circuit, what happens to the charge on capacitors C103 and C104? If you are not careful to provide a discharge path, you could destroy both FETs since the excess voltage will be applied to their gates with nowhere to go.

That is the function of diodes D9 and D10. C103 and C104 are safely discharged into the load at switch-off. A 0.68µF cap removes any remaining noise from the HV supply and provides a high-frequency return path for the load. To supply the screen grids of the output tubes, I simply used a 75V zener diode Z8 to drop the HV down to about 320V.

Other aspects of the HV PS are for the most part straightforward. The power transformer is a Hammond 274BX, which is more than adequate for this application. I have included a negative temperature coefficient (NTC) thermistor in the transformer primary circuit to limit the inrush current at switch-on.

Eight 1N4007 power diodes (D1 to D8) provide the raw positive and negative DC potentials. All have 510k resistors (R101 to R108) in parallel to ensure that the reverse voltages are equally shared. Both positive and negative leads include current-limiting resistors (R119 to R122), since the power trans-

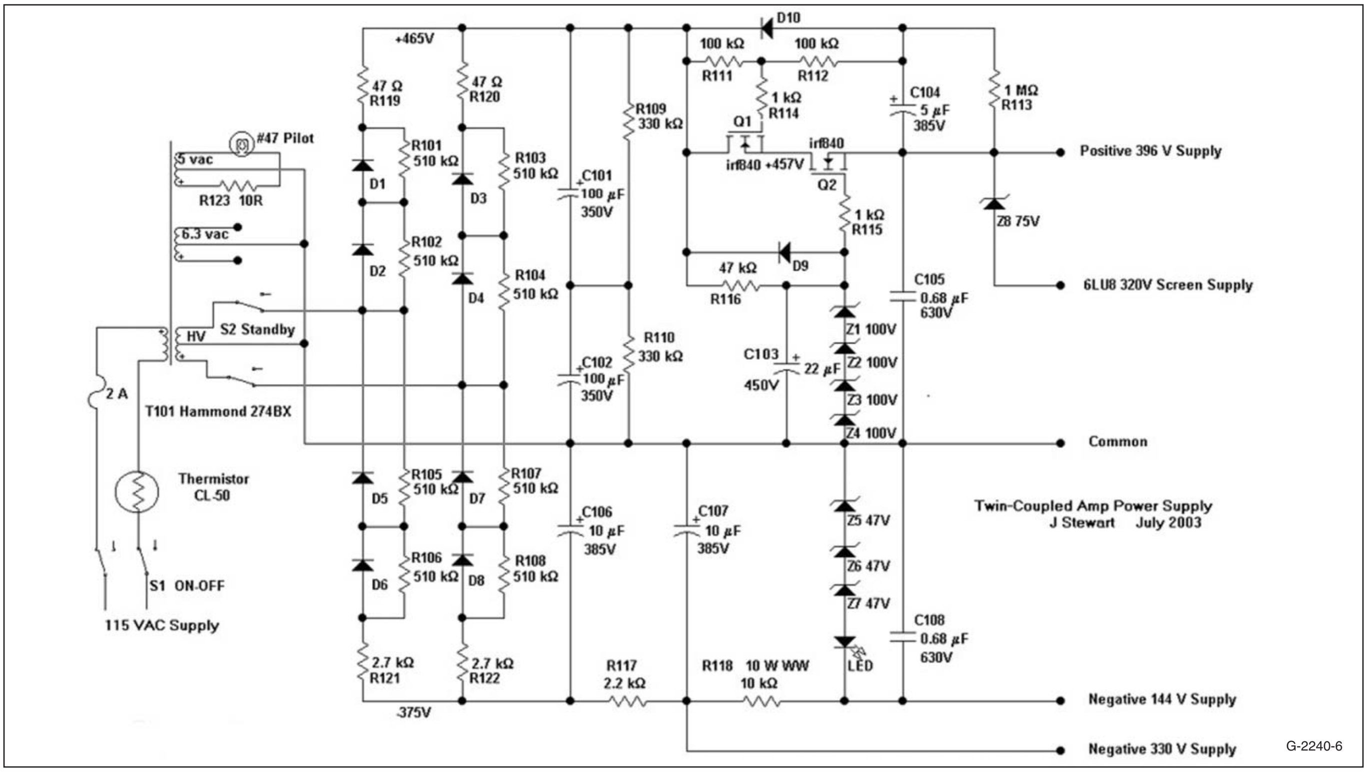

FIGURE 6: Power supply.

former was originally meant to be used with a vacuum tube rectifier.

A pair of 100µF/350V caps (C101 and C102) connected in series provides filtering for the positive HV. Voltage across these caps is equalized by a pair of 330k resistors R109 and R110. Current requirements from the negative supply are minimal. That way you can increase the limiting resistors R121 and R122 in order to reduce the ripple at this part of the circuit and reduce the capacity required for filtering. For the amplifier bias supply and the second stage differential amplifier, the negative supply is regulated by a series of three 47V zener diodes (Z5 to Z7) and a common red LED.

The 6.3V winding on the 274BX supplies all of the filaments, while the 5V winding is left to light the pilot light. When the amplifier audio output is 25W, the power in from the line is 141W or 160VA, with no signal that drops to 106W. In the standby condition the power required is only 38W. The Hammond 274BX is rated for 198VA input to its primary so there is ample reserve.

CONSTRUCTION

The amplifier is built on a 10 × 17 × 3 Hammond chassis. I did not spend much time on cosmetics, which—in general—do not add to the final performance of an amplifier. My primary interest is in the electrical design aspect of the project. I will leave the exterior to you, the builder, to provide what you think is a suitable appearance for their listening environment. Having said that, the inside of my amp is carefully laid out to perform at its very best.

As usual, I have built a set of wooden handles from ¼ × 1.5″ wood strip available from hardware and hobby stores. This allows the amp to spend the beginning of its life inverted during the build and test phase, without damage to the tubes and transformers. When the amp is finished, these become most useful to move the amp from place to place. There is an 8–32 × 1.5″ machine screw protruding from the rear panel so you can easily connect to the many ground leads during testing. I have now built several projects using these two simple but useful ideas.

You can build several of the circuits as subassemblies. To do this you simply attach a pair of terminal strips to a wood base, properly positioned so that you can solder the parts in. When complete, the components will hold the entire assembly together, ready for mounting as a unit in the chassis. Refer to the subassembly photos. The rectifier module contains eight diodes and 12 resistors—everything to provide both positive and negative DC in a very small space. The positive filter as shown is complete as well.

The negative filter/regulator circuit is assembled in a similar way. As shown in the photo, it is not yet complete. I had to determine some of the components required for the circuit to work properly after installation. You can see I have marked the locations of the components C12, R36, and R37.

The positive delay circuit is incomplete as well. I connected the 22µF time delay cap C103 after installing the module in the chassis between the terminals as shown. These subassemblies avoided problems with trying to fit many parts into tight spaces.

The FET Q2 is mounted on a Wake-

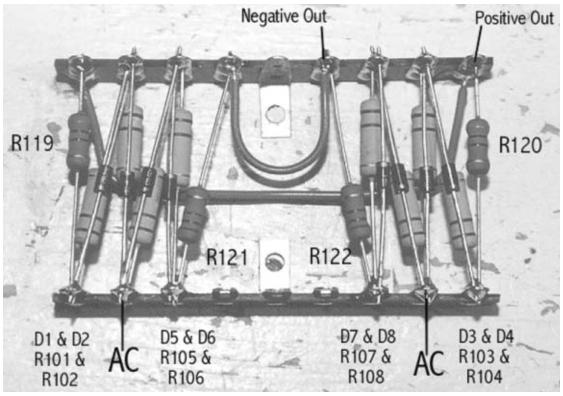

PHOTO 8: Rectifier module.

TABLE 2

PARTS LIST		AMPLIFIER	
C1		680nF, 630V	
C2, C3		470nF, 630V	
C4, C5		220nF, 630V	
C6, C9		100µF, 450V	
C7, C8		680nF, 630V	
C10		6.8nF, 630V	
C11		10µF, 450V	
C12	See Text	10µF, 450V	Diff Amp Input Stage Only
P1		100K Gain	
P2		100K AC Balance	
P3		10K DC Balance	
P4		10K Output Bias	
Knob for Gain Control			
3 Pot. Locks for the Bias, AC and DC Balance Adjustments			
R1, R2, R3, R4, R19, R20, R21, R27		1k, ½W	
R5, R6		100k	
R7, R22, R24, R25, R26		220k	
R8, R11, R12		47k	
R9		33k	
R10		330k	
R13, R14		27k	
R15, R16		1M	
R17, R18		75k	
R23	See Text	75k	Diff Amp Input Stage Only
R28		6R (2 of 12R Paralleled)	
R29		10R, ½W	
R30, R31		200k	
R32, R33		12R, ½W	
R34		3.3k	
R35		68k	
R36	See Text	12k	Diff Amp Input Stage Only
R37, R38	See Text	330k	BiAmp Version Only
All resistors are 2W except as noted			
T1, T2		Hammond 125E Universal Output Transformer	
V1		6SL7GT	
V2		6SN7GTB	
V3, V4		6LU8	
2 Octal Sockets			
2 Compactron Sockets			
RCA Phono Connector (Input) AES p/n S–H310			
One 6–Screw Terminal Strip (Output) AES p/n S–H317			

(continued on 22)

field heatsink, which is attached to the chassis using nylon screws, washers, and nuts. The other FET Q1 is on a small heatsink, just a piece of aluminum extrusion. It is supported on a pair of short terminal strips, but could just as well be mounted using nylon hardware.

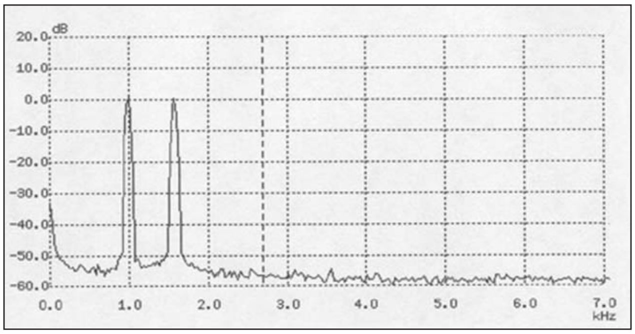

FIGURE 7: Intermodulation test. G-2240-7

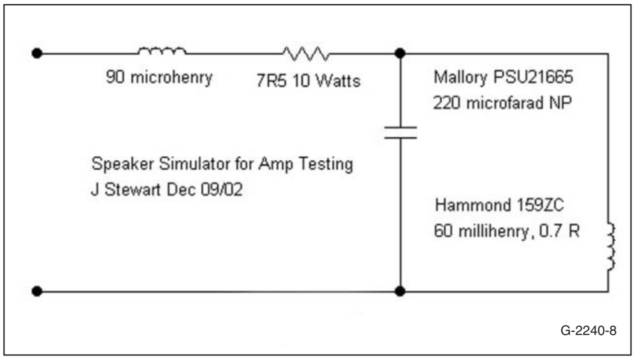

FIGURE 8: Loudspeaker simulator.

DAMPING FACTOR

A look at the test results will tell you that the DF of this amplifier is not particularly high. The reason it is not has to do with the way negative FB, both local and full loop, have been applied. Neither of those FB circuits take into account the internal resistance of the output transformer pair. These resistances, both in the primaries and secondaries of the transformers, are in series between the amplifier and its load, the loudspeaker.

Using both AC and DC methods of measurement, I determined that the resulting resistance in series with the load was 2.32Ω referred to the secondaries. That works out to a DF of 3.45 for an 8Ω load, which is the best you can expect from the circuit as it is implemented here. If I had used purpose-built transformers whose coupling was tighter, then the DF would be higher. However, the primary objective of this project was to build an excellent-sounding amplifier at a minimum cost, just as Crowhurst had set out to do. That is exactly what you have here.

By giving up a little with respect to the DF spec, you have an amplifier with exceptional distortion specs. You also have freedom from low-frequency instability and high-frequency parasitic oscillation under varying load conditions.

TESTS

I wanted these tests to come close to predicting what would happen when the amplifier drove a loudspeaker with real program material. Many of the results you see from amp testing use only a single tone, usually 1kHz into a resistive load. Some builders of amplifiers offer no test results at all. Poor results may simply be an amplifier's inability to drive a particular loudspeaker while being OK with others.

These tests were done with both single and two-tone setups. Many of these were into resistive loads, but I put together a loudspeaker simulator as well. This allowed some tests to look for parasitics and transient response.

For the two-tone tests I connected both an HP 200CD wide range oscillator and a GW GAG-810 oscillator through a resistive network to the input of the amplifier. That allowed for testing of intermodulation (IM) distortion. Simple IM distortion is the sum and difference of a pair of frequencies as they pass through a nonlinear device, such as your amplifier.

For IM tests there are two common methods used in audio. The most common in the 1950s was a combination of 60Hz (or 50Hz) and another signal at 4 to 7kHz in a 4:1 ratio. The low-frequency signal was simply taken from a heater winding on the power transformer of the test set while the fixed

(CONT. FROM 21)

PARTS LIST	POWER SUPPLY
C101, C102	100µF, 350V
C103	22µF, 450V
C104	5µF, 450V
C105, C108	680nF, 630V
C106	10µF, 450V
C107	10µF, 450V
CL–50	NTC Thermistor
D1 to D10	1N4007
LED	A common Red LED
Q1, Q2	IRF840 FET
R101 to R108	510k
R109, R110	330k
R111, R112	100k
R113	1M
R114, R115	1k, ½W
R116	47k
R117	2.2k
R118	10k, 10W Wirewound
R119, R120	47R
R121, R122	2.7k
R123	10R
All resistors are 2W except as noted	
S1, S2	DPDT Switch
T101	Hammond 274BX Power Transformer
Z1, Z2, Z3, Z4	1N5378B, 100V Zeners
Z5, Z6, Z7	1N5368B, 47V Zeners
Z8	1N5374B, 75V Zener
Chassis	Hammond 1444–32, 10 × 17 × 3 Chassis
Bottom Plate	Hammond 1434–30, 10 × 17 Bottom Plate
Heatsink	Wakefield 401k for Q2
Small Aluminum Extrusion	Mounting for Q1
Nylon Screws, Nuts and Washers for Heatsink Mtg	
Pilot Lamp Holder and #47 Bulb	
Terminal Strips	
Fuse Holder and 2A Fuse	
Power Cord	
Various machine screws, nuts, washers, grommets, Etc.	

high frequency was generated by a simple on-board fixed-frequency oscillator. Neither signal source needed to be particularly distortion free since it is the resulting IM sidebands that were measured. If the low frequency used is 60Hz and the high frequency is 5kHz, then the sidebands will be at 4,940 and 5,060Hz. This is the SMPTE test.

The other method sets a pair of signals of equal amplitude somewhere in the midband. Using frequencies of 1kHz and 1.6kHz would produce second-order IM products at 600 and 2,600Hz. The test frequencies should not have a simple harmonic relation to each other. For example, if you used test frequencies of 1 and 2kHz, then the third harmonic of the 1kHz signal (3kHz) would interfere with your measurement of the IM distortion at the same frequency, that being the sum of 1 and 2kHz. Also, the difference frequency would be at 1kHz, which would be completely masked by the 1kHz test signal.

Because I did not have an IM test set that would operate in the SMPTE mode, I used the second method. I measured the resulting IM products with a Pico Technology ADC-100 Dual Channel Virtual Instrument in its Spectrum Analyzer (SA) mode (*Fig. 7*). Any IM products are down more than 50dB from the test signals, which have a combined level equivalent to 30W.

It may come as a surprise to some that their amplifier will not be capable of full-power output to the load if it is driven by a two-tone signal. A quick look at some numbers will illustrate this fact. If your amp can deliver 8W into an 8Ω load at clipping, you measure 8V RMS. That means that clipping occurs at 2.8 × 8V or 22.6V peak-to-peak. The peak-to-peak spec is what will limit the amplifier's output.

Now, if you have an output of a two-tone signal each of 4V, they will add and subtract in such a way that the peak-to-peak limit will be reached regularly at a rate determined by the test frequencies selected. The formula for power is $P = (E^2/R)$, so each signal contributes 2W to the load. That is only 4W total, half of the single tone condition. As the test signal becomes more complex, maximum power delivered by the amplifier may be even less. The actual voltage and power delivered to the load can be confirmed by using one of the new precision DMM/wattmeter instruments.

The loudspeaker simulator (*Fig. 8*) has a nice peak at about 45Hz, similar to a speaker's resonance in its box. If you need to simulate a reflex enclosure you will need a second LC circuit. Other resonant frequencies can be set by changing the Hammond 159ZC choke or its resonating non-polarized (NP) capacitor, which was originally meant as a motor starting cap. The rise of impedance caused by the 90μH choke starts at about 4kHz and is well up at 10kHz, similar to a real speaker.

CONCLUSION

In terms of performance versus cost, this amplifier is easily the best I have ever built. At a cost of about $200 US I'm sure there are few vacuum tube designs that could compete. The sound is quite spectacular when connected to a good loudspeaker system in an adequate listening room. Had I used purpose-built output transformers, the results would have been even better.

I would especially like to thank the people at Hammond Manufacturing for their generous assistance towards the successful completion of this project. Hammond has been a source of transformers, chokes, and chassis for my many projects, both private and work-related over a period of more than 55 years. I still have a working regulated power supply which uses a 25Hz version of their 275× power transformer on a Hammond 8 × 12 × 3 steel chassis. Conversion to a 60Hz power system finally came to Southern Ontario in the late 40s. I still use my Hammond power supply regularly on my test bench as I investigate new circuitry, as I did in this project. ❖

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EQUIPMENT USED IN THE DEVELOPMENT OF THIS AMPLIFIER

HP 334A	Total Harmonic Distortion (THD) Analyzer
HP 200CD	Wide Range Audio Generator
GW GAG–810	Audio Generator
Rohde & Schwarz	Dual Channel, 100MHz Oscilloscope
Metrahit 29S	Precision RMS DMM and Power Meter
Pico Technology ADC–100	Dual Channel Spectrum Analyzer/Oscilloscope
Powerstat	8A Variable Transformer
Loudspeaker Simulator	see text
Burst Gate	Published in *audioXpress*, November 2003 Issue

The DR200 Horn

This author's latest horn design is intended for use as a PA cabinet in small- to medium-size venues. **By Bill Fitzmaurice**

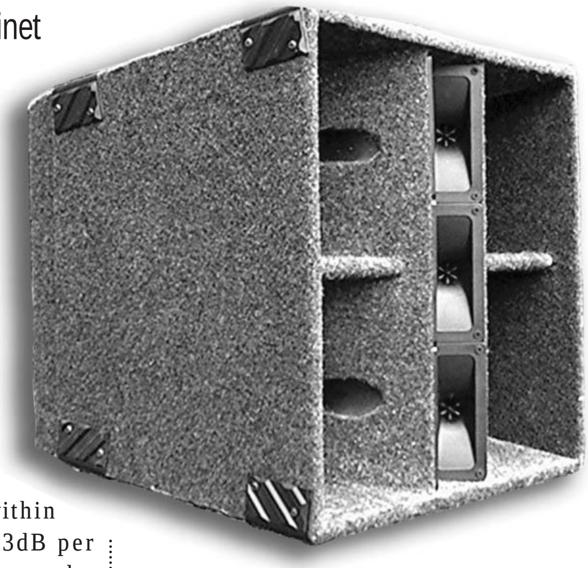

PHOTO 1: The completed DR200 Horn.

This next box in the DR version 2.0 series is the DR200 (*Photo 1*), named after its 200mm (8″) woofer. Its 105dB average sensitivity (*Fig. 1*) gives very high output from a very small package. Similar to the first-generation DR8 (Nov. '01) in design, I modified the external shape for vertical line-array placement of multiple cabinets, though even used singly, line-array technology is utilized via vertically arrayed cross-firing tweeters.

LINE-ARRAYS

Line-arrays are all the rage today for PA, and for good reason. By stacking mid- and high-frequency elements vertically—be they within a single cabinet or in multiple cabinet clusters—comb-filtering effects that occur when elements are clustered on a horizontal plane are eliminated. The horizontal dispersion characteristics of a single element/cabinet remain, while the vertical angle of dispersion is greatly reduced, which is very effective in eliminating early reflections off the ceiling and floor.

Another benefit of the line-array is the production of a near-field radiation zone, within which the SPL level falls at 3dB per doubling of the distance from the source rather than 6dB. This directly correlates with the narrowing of the vertical dispersion angle, and is frequency-dependent according to the height of the array, going lower in frequency as the array increases in height. The relatively small height of the tweeter array in a single DR200 gives a near-field condition from about 3kHz and up, and with every doubling of the array height with additional cabinets the near-field frequency drops by an octave.

Within the near field the higher the frequency the longer the distance from the source before the 3dB/doubling distance loss of the near field returns to the 6dB/doubling distance loss of the far-field. This factor really cleans up the PA in larger rooms, where wave development and boundary effects accentuate the bass, while the treble becomes swamped as the room gets bigger. A well-designed line-array will often sound virtually the same whether you are twenty feet away or two hundred.

This near-field condition for the higher frequencies is a serendipitous circumstance when using a cross-firing tweeter array. Cross-firing tweeters have the benefit of giving off-axis response within a few dB of the on-axis

ABOUT THE AUTHOR

Bill Fitzmaurice has written a book entitled *Loudspeakers for Musicians*, which contains more of his super-performance speaker designs and is available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, custserv@audioXpress.com.

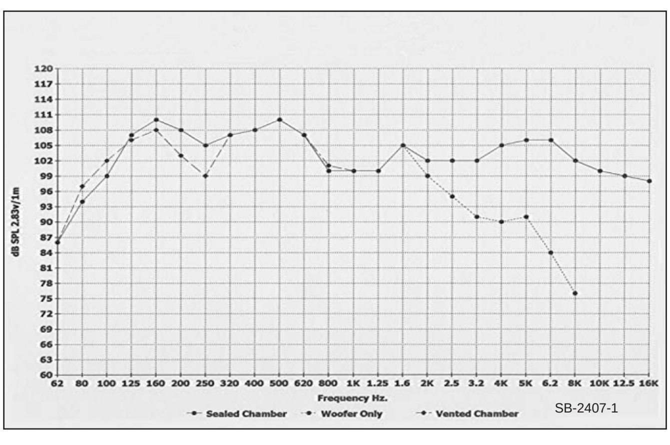

FIGURE 1: DR200 axial response: measured outdoors/half-space.

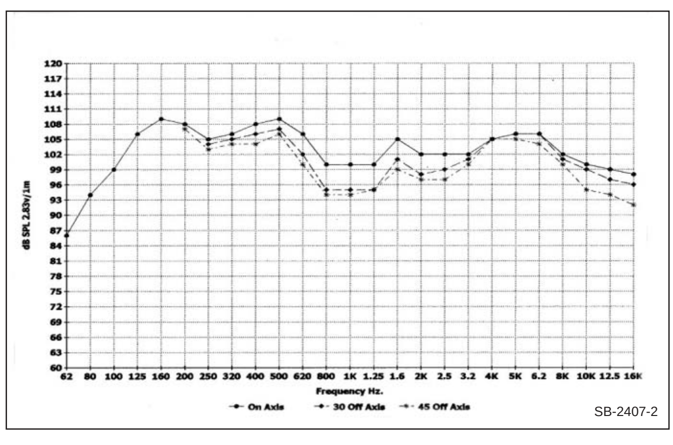

FIGURE 2: DR200 off-axis response: measured outdoors.

response (*Fig. 2*). However, because elements are horizontally aligned, there comes a point at which they are one-half wavelength apart and phase cancellation results in high-frequency rolloff. In the DR200 this occurs at 6.2kHz, but you can easily overcome the rolloff with EQ, as the power requirements above 6.2kHz are quite modest. Since the near-field radiation produced by the vertical tweeter array means that the highest frequencies are rolling off at 3dB/doubling of distance while the lower frequencies are rolling off at 6dB/doubling of distance, the farther you are from the source the less "droop" in the high-end response.

One alteration that you should consider if you're building a fleet of DR200s for an array is to make any intended for long-throw (i.e., over 50′) with a flat tweeter baffle and a single row of tweeters. At long throws the wide dispersion of the cross-firing array becomes less advantageous, while the elimination of the high-end rolloff would be a plus. Therefore, in a three box or higher array I recommend the lowermost two boxes have cross-firing tweeters for maximum horizontal dispersion close to the stage, with all those above having a single row of tweeters for maximum SPL to the rear of the audience. To preserve the

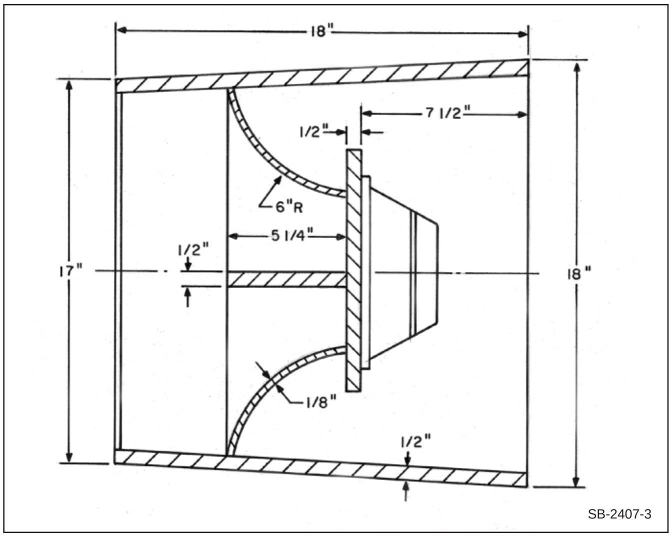

FIGURE 3: DR200 cutaway side view—simplified.

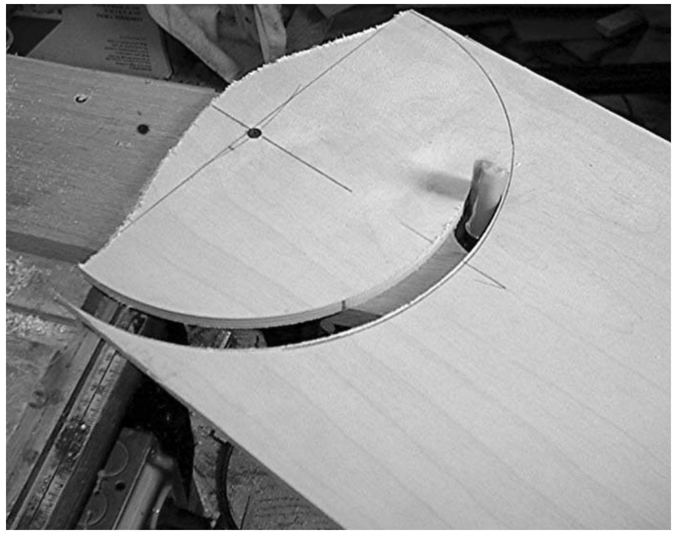

PHOTO 2: Cutting a radius on a router table.

overall tonal balance between the long and short throw versions use the same tweeters in both boxes.

GETTING READY

I built the DR200 prototypes using a neodymium magnet woofer, the PAudio™ SN-8MB. T/S specs for this driver are f_s 70Hz, V_{AS} 24l, Q_{TS} 0.30, SPL 95dB/1W, and Pe 150W. Other MI eights with similar specs will do fine. The main advantage of neodymium woofers is light weight, though in the 8″ size the advantage over ferrite is not huge. Another advantage is that the magnet is quite small, which in the cramped confines of a DR horn is helpful, though again it's not as though an Eminence™ Beta-8 is all that big. I'd recommend a neodymium woofer only if those last two or three pounds are worth the extra cost.

As for the tweeters, I used some that I had on hand, the CTS™ model KSN 1016A. While they work fine, I recommend a different model, the KSN 1167A. The rectangular shape of the KSN 1016A complicates the construction process somewhat over the square-frame KSN 1167A, and the smaller size of the KSN 1167A allows using eight of them as opposed to six KSN 1016A tweeters, giving another dB or two of

MEASURING THE SPL OF LINE ARRAYS

Complications arise when measuring the SPL of line arrays as opposed to point sources. Especially problematic is measuring SPL from vertical tweeter arrays, as multi-path phase distortions make accurate measurements impossible at 1m from the source. I found the minimum distance that gave reasonably uncorrupted readings was 4m, and that brings into play the near-field condition of the higher frequencies when you try to translate a reading taken at 4m to a traditional 1W/1m chart. The measurements shown on my SPL charts are representative of response at 4m, with 12dB added across the board to arrive at a reasonable approximation of the 1m figure.

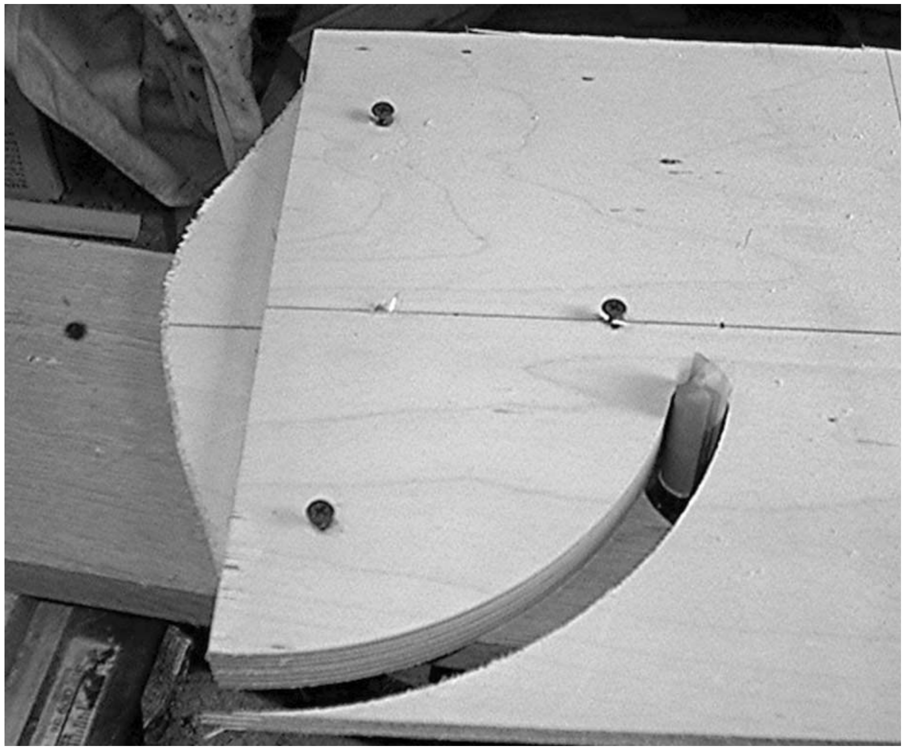

PHOTO 3: Using a pattern bit to duplicate parts; the new part is screwed to the pattern while being cut.

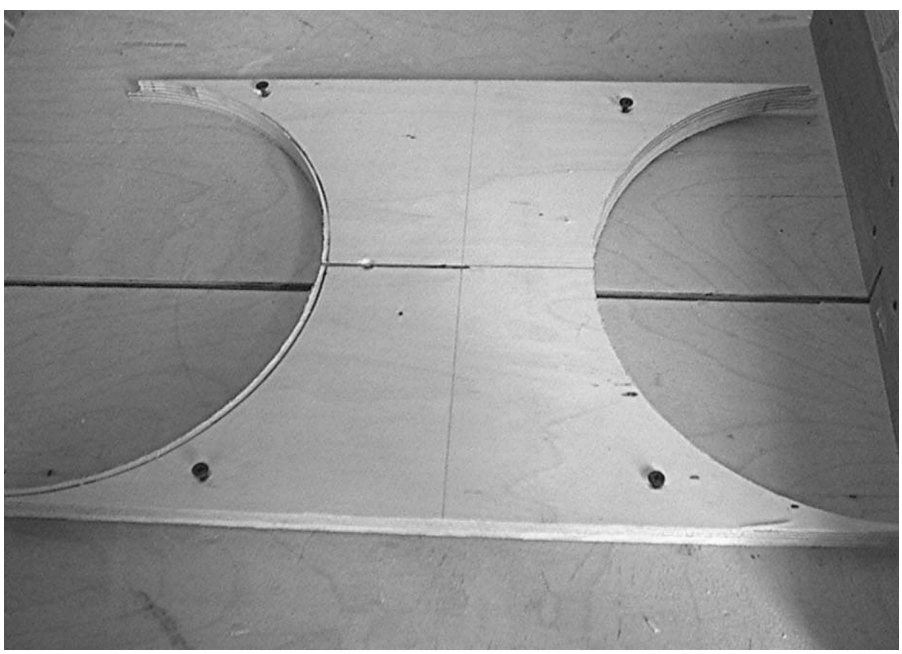

PHOTO 4: Cutting two throat horn sides from a single piece of plywood.

HF sensitivity. The only reason I'd consider the KSN1016A at all is that there is a generic version of that tweeter available from Parts Express (catalog #270-041) for less than a dollar each in quantity, and using them as opposed to the KSN 1167A could save you around $40 per cabinet.

The box is primarily constructed of ½″ plywood; use Baltic birch if you like, but spruce will do. The horn sheaths and back halves are all ⅛″ Baltic birch; determine the easier bending axis of the plywood before laying these out. Most joints are screwed, all screws pilot-holed and counter-sunk, and most joints glued and caulked with construction adhesive, preferably of the polyurethane-based variety. You'll also need a hot-melt glue gun for some of the low-stress joints. Extreme accuracy in measurements and parts cutting is not required, but you're still best off using a table-saw with a panel-cutting jig.

Some of the parts have holes in them to minimize flat internal surfaces. Cut them with hole-saws if you have them, but a saber-saw will do. Also, chances are you're making your DR200 for PA, and that you're building a minimum of

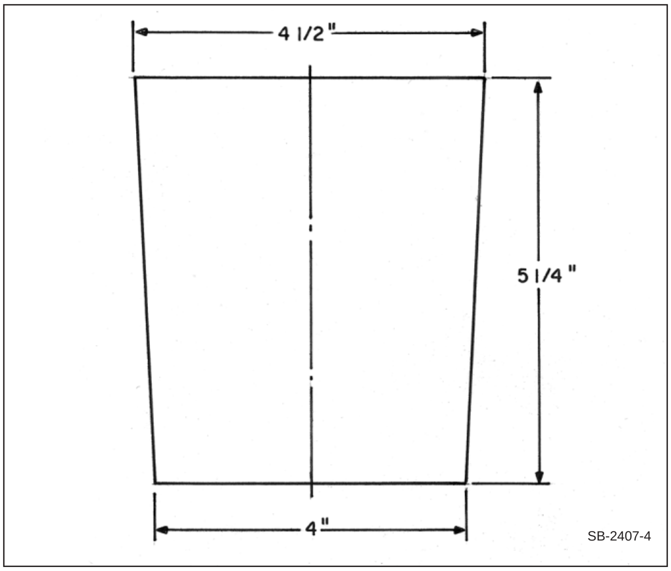

FIGURE 4: Trapezoidal-shaped throat divider.

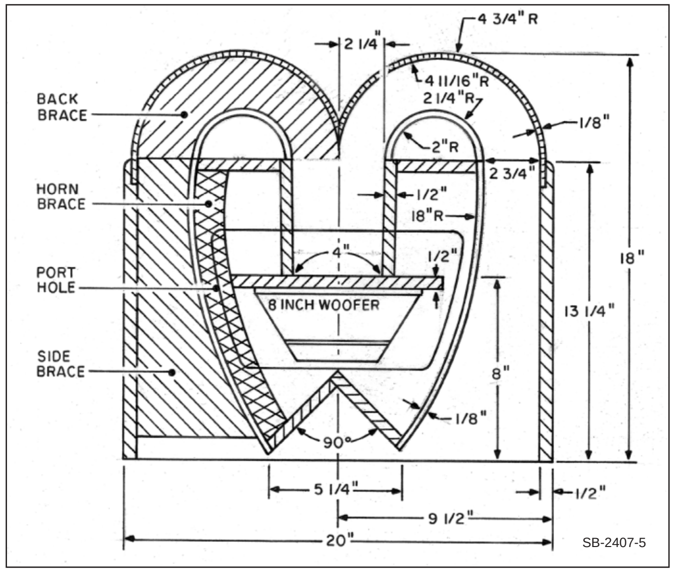

FIGURE 5: DR200 cutaway top view simplified.

two copies, so many of the instructions and photos show using a router table for easy duplication of parts, as well as a high degree of accuracy where rounded cuts are involved. The cabinet has a trapezoidal side profile (*Fig. 3*), so you must cut many of the parts with an approximately 5° deviation from square for proper alignment. You can accomplish the panel cutting with a hand circular saw, but you'll need an accurate guide accessory to keep cuts straight and true.

Fill the cabinet loosely but thoroughly with polyfill to eliminate any internal

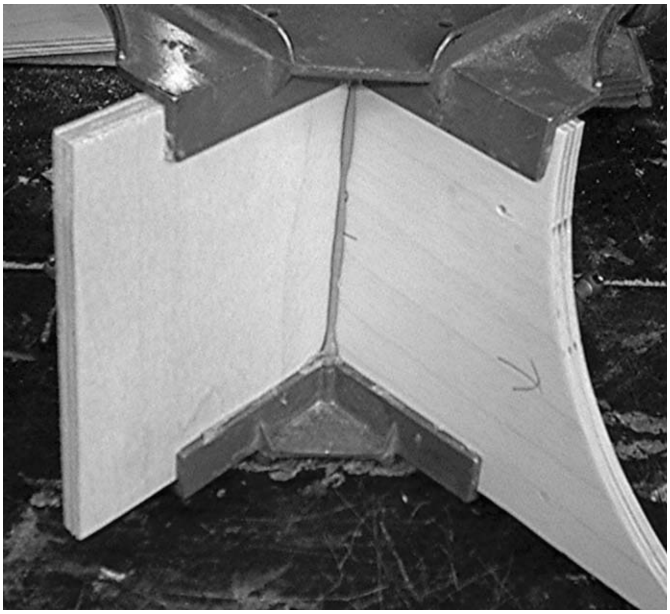

PHOTO 5: Attaching a throat horn side to the throat horn divider.

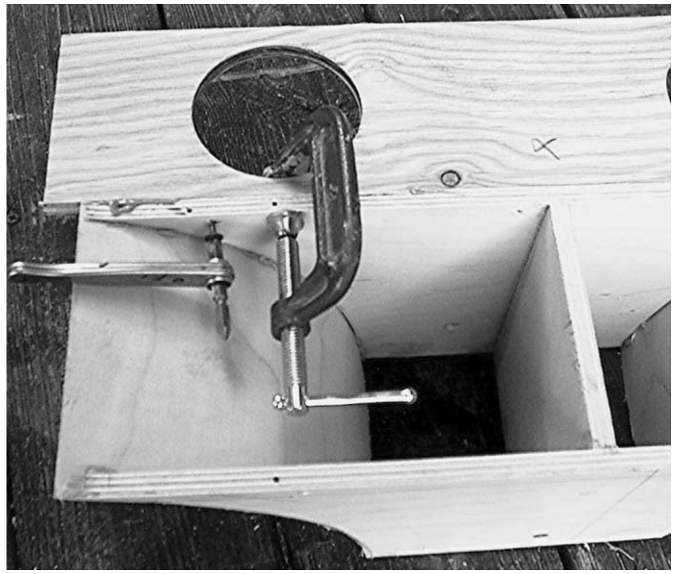

PHOTO 8: Attaching a horn support. Note the right-angle ratchet screwdriver.

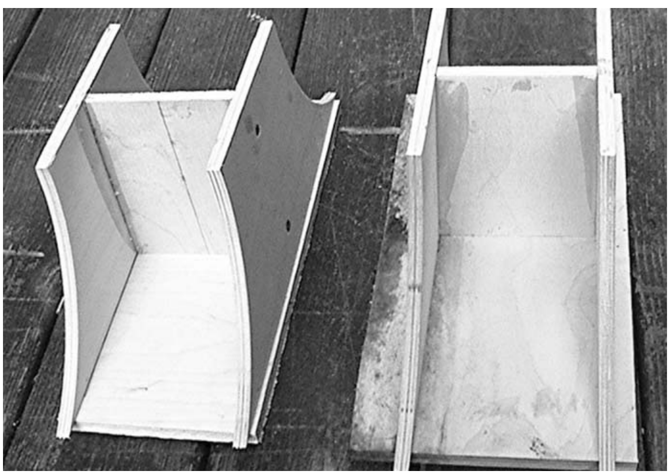

PHOTO 6: Two throat horns on jigs, ready to sheath.

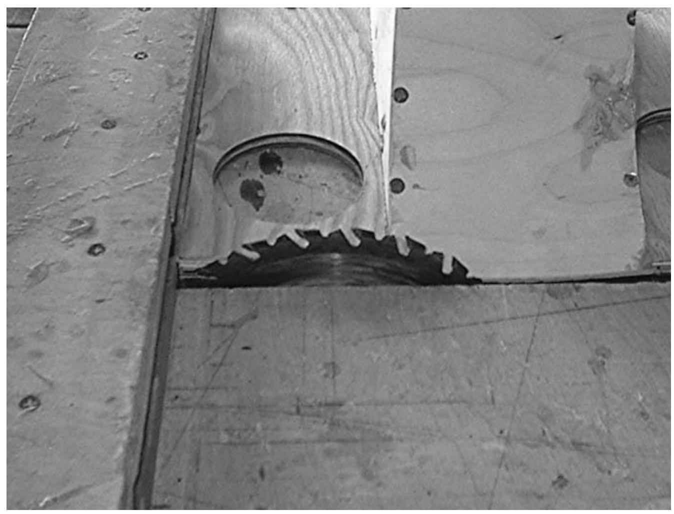

PHOTO 9: Trimming the throat horn assembly to final size on a table saw with panel cutting jig.

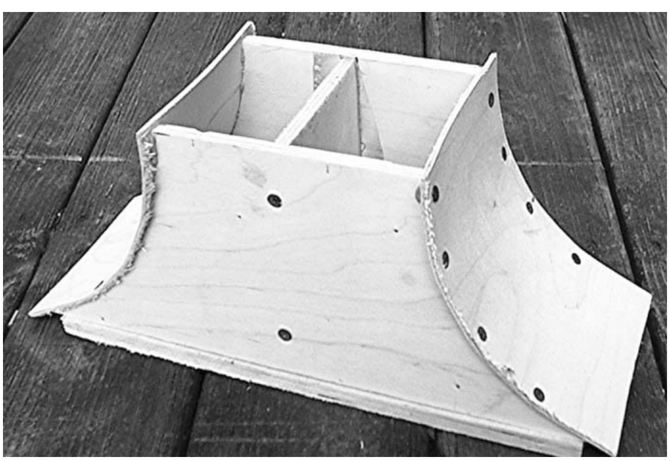

PHOTO 7: A sheathed throat horn; note the excess sheath material to be trimmed.

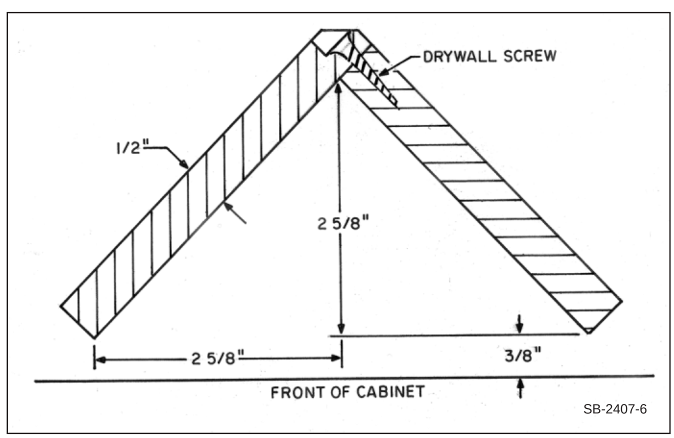

FIGURE 6: DR200 tweeter baffle detail.

midrange reflections. You may opt to vent the rear chamber via ducts or leave it sealed airtight. As you can see in *Fig. 1*, sealing the box gives better sensitivity from 160 to 320Hz, but gives some up from 80 to 100Hz. I recommend ducting the chamber if you don't intend to run your cabs with separate subs, and leaving it sealed if you do.

CONSTRUCTION

Construction starts with the throat horn (*Fig. 3*). Cut out the two throat horn sides. *Photo 2* shows how to use a router table to produce perfect radius cuts on a larger piece of plywood, as the plywood is screwed to the table at the center of the half-circle and the plywood is rotated into the router bit. *Photo 3* shows using a pattern following bit (with a bearing) to produce perfect copies of parts. *Photo 4* shows a piece of plywood with two "half-moon" cutouts being sliced in half to make two throat-horn sides.

First, attach the trapezoidal-shaped throat divider (*Fig. 4*) to one throat horn side and then the other, being offset from the center so that the two halves of the horn are not quite identical; a small picture-frame clamp helps a lot here (*Photo 5*). Screw the throat horn assembly to a scrap piece of plywood that serves as a jig to hold the parts square and true (*Photo 6*) while installing the throat horn sheaths. Cut the throat horn sheaths a couple of inches oversize, to be trimmed later (*Photo 7*). After the adhesive has dried, sand the extra sheath material flush to the sides.

Cut out the horn supports and cut a few holes in them; cut them at least an inch longer than required. Attach the horn supports to the throat horn (*Photo*

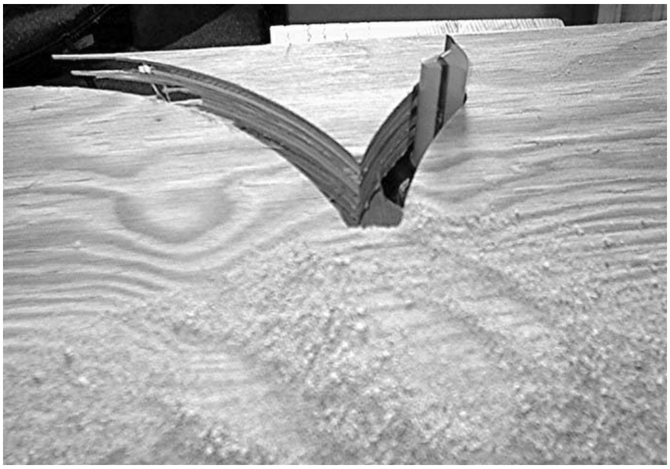

PHOTO 10: Using a router for the radius cuts on the back of the cabinet top and bottom.

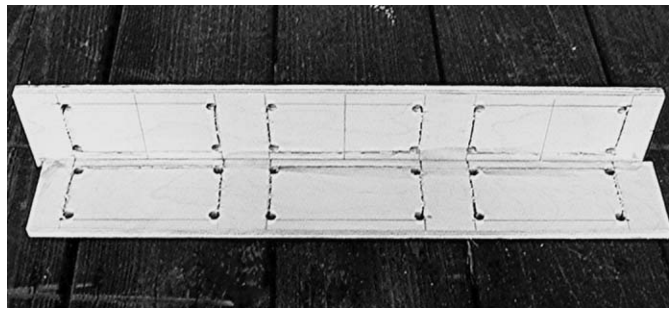

PHOTO 11: The assembled tweeter baffle. Note the rectangular mounting holes for KSN 1016A are left uncut on the outermost edge.

8), a job greatly simplified with a couple of clamps and a right-angle ratcheting screwdriver. After the adhesive has set, run the entire assembly across the table saw, using a panel-cutting jig, to the finished length (*Photo 9*). Note that the cuts at the top and bottom must be made with the blade about 5° off vertical to match the taper of the cabinet, and be careful that the angle is facing the right direction.

Cut out the top and bottom (*Fig. 5*). Here again perfect radius cuts where the back halves join are easy using a router table (*Photo 10*). If you don't have a heavy-duty router, the procedure is easier if you rough-cut the parts with a saber-saw first and use the router to finish it off. Onto the top and bottom draw where the other parts will mate, as per *Fig. 5*.

Using a saber-saw, starting with a plunge-cut, cut out a porthole from the bottom. Cut out the tweeter baffle halves (*Fig. 6*), making them an inch longer than the finished size. If you're using rectangular tweeters, determine their locations and cut out the mounting holes for them, except for the final cut near the horn mouth, and assemble the tweeter baffle (*Photo 11*). [You can cut out round mounting holes for square frame tweeters after assembly with a hole-saw.] With the blade set at 5° off vertical run the tweeter baffle across the table saw to trim it to finished length (*Photo 12*).

[For the non-cross-firing tweeter configuration make a flat tweeter baffle 3½″ wide on the outer face, and alter the positions of the horn braces accordingly. Since accessibility is not a problem you may cut tweeter-mounting holes after the cabinet is assembled.]

Attach the throat horn assembly to the bottom, using a guideboard and clamps to hold it on the joint line (*Photo 13*). The supports will have a bit of splay to them but the angle is slight and they will pull tight to the guideboard easily. In similar fashion attach the tweeter baffle to the bottom, and then the top to the assembly (*Photo 14*).

The baffle is round, 10″ in diameter. Cut it out and hold it in place centered

PHOTO 14: Throat horn and tweeter baffle attached to the top and bottom.

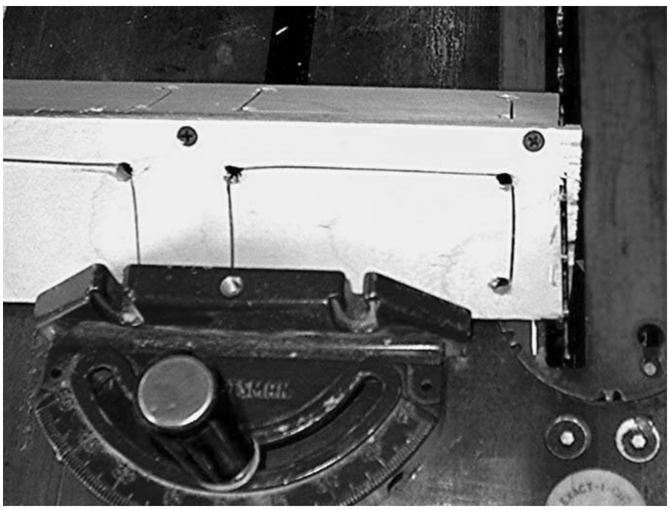

PHOTO 12: Trimming the tweeter baffle to finished size and angle.

PHOTO 13: Using a guideboard and clamps to hold and align parts for jointing.

PHOTO 15: The baffle and horn supports installed.

on the throat horn, and from the inside of the throat horn trace on it the outline of the throat opening. If you wish to use "T" nuts for mounting the driver, install them now, though with a lightweight eight-incher, screws will do.

Cut the hole in the baffle and attach it to the throat, using plenty of adhesive at the joints with the sheathing where screws cannot be used. Cut out one horn brace, using it as a pattern to duplicate as many as you need for your cabinets; with a long duplicating bit the router table can finish-trim three or four duplicates simultaneously. Install four horn braces per cabinet side, one each top and bottom, two others subdividing the assembly; take your tweeter locations into account so that the braces won't interfere with them where they attach to the tweeter baffle (*Photo 15*).

Cut out the back braces. A good way to make them is by using the router technique to form a perfect circle and

PHOTO 16: Installing the horn sheaths.

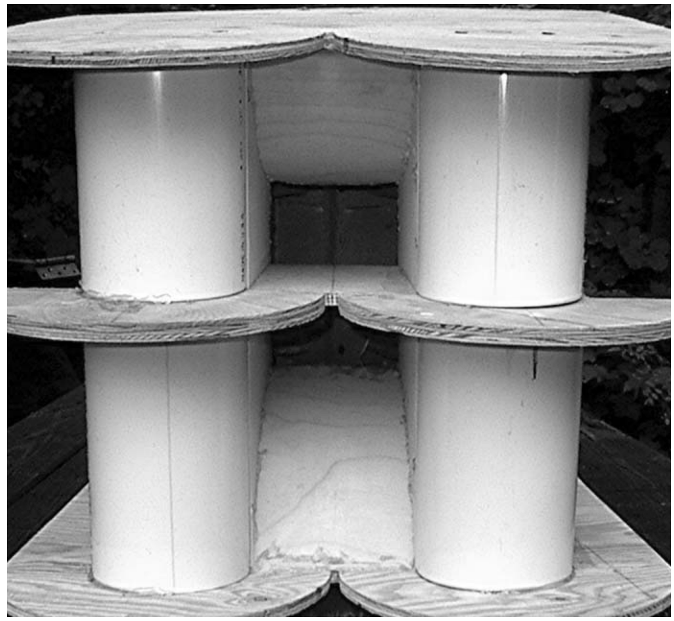

PHOTO 17: The back braces and reflectors installed.

then slice it into two halves. Attach them to the assembly, one end mating with the throat horn divider, screwed to the braces from the inside. Cut out the mouth horn sheaths and install them, starting at the joint with the supports (*Photo 16*). You can use clamps to pull the sheaths into place, but the ⅛″ plywood bends so easily that it can be done by hand.

Using an abrasive blade, halve a couple of feet of 4″ Schedule 40 PVC pipe on the table saw. You need to make four crescent-shaped reflectors from these to mate with the horn supports, with one end cut at about a 5° angle to match where they meet the top and bottom. Fill each reflector with some poly-fill and glue them in place (*Photo 17*), being sure that all joints are absolutely airtight.

Cut the side braces, making them a bit oversize. Temporarily place them in position and use a straightedge to mark their finished sizes (*Photo 18*) and trim them to size, leaving them about an inch short of the mouth of the horn. Attach them to the assembly, driving a couple of screws through the sheaths from the inside. Rough-cut the back halves, making them wide enough to extend at least an inch past the curvature of the back to the flat section of the sides.

When you put the first back half into place, use a small sliver of ⅛″ plywood scrap as a wedge to stabilize it until you drive the first few screws (*Photo 19*). Attaching the back halves is greatly eased with a couple of webbing clamps to pull them into place (*Photo 20*).

FINISHING UP

Now is the time to complete the cutting of the tweeter holes if you are using rectangular tweeters, or drill holes with a hole-saw for square frames. In either case remove any screws from the joint of the sheaths and the tweeter baffle that would be sawn into, filling the holes with adhesive.

If you're venting the box, drill four holes for the vents through the sheaths. The holes go through the sheaths at about a 55° angle, and they are easily cut using a hole-saw mounted on an extra-long drill bit (*Photo 21*). You'll likely be able to cut only partway before having to clear some material from the hole (*Photo 22*) and then finish with a second pass. The ducts are made from 1½″ PVC pipe, cut at a 55° angle at one end, to a length of 1½″ not counting the angled section. Install them with hot melt, most of it being applied from inside the box. When the glue has set sand the ducts flush to the sheaths.

Whatever your choice of finish, now is the time to start applying it, as reaching the inner areas of the horn is easy when

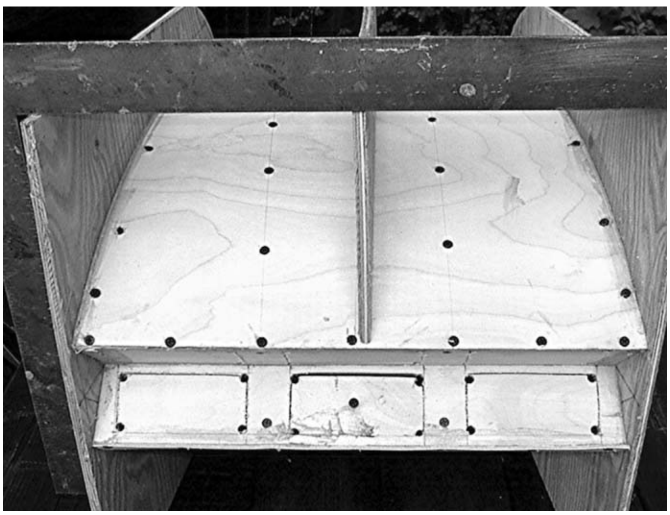

PHOTO 18: Marking a side brace for trimming.

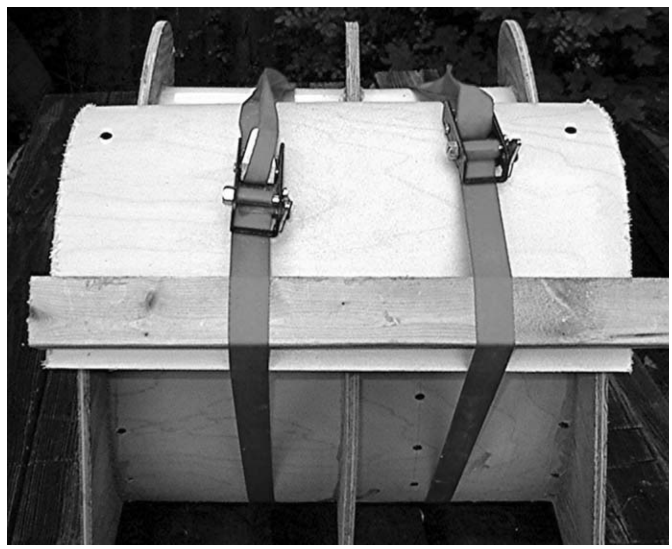

PHOTO 20: Using webbing clamps to pull back halves into place.

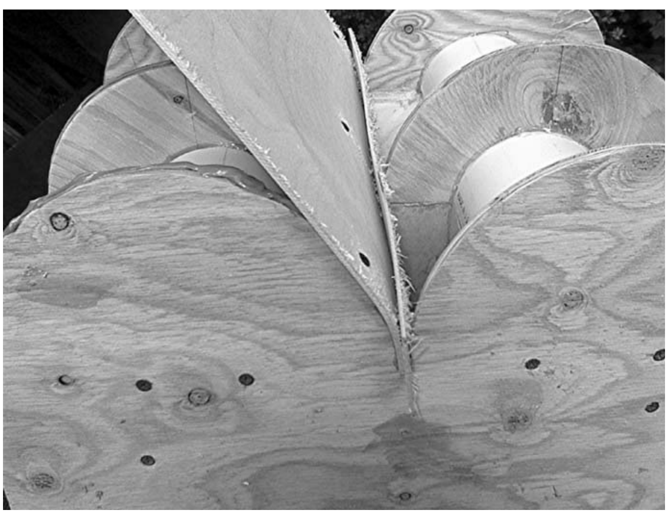

PHOTO 19: Beginning back installation. Note small scrap of ⅛″ plywood at back joint as temporary wedge.

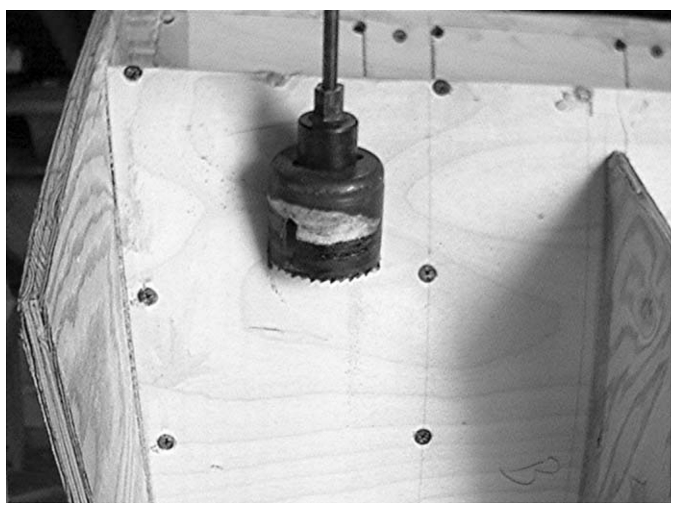

PHOTO 21: Cutting a duct hole.

the sides are not yet in place (*Photo 23*). Then cut the sides, as always a bit oversize; install them and trim the edges flush to the cabinet. Where the sides overlap the back, dado out enough material for a flush fit. Complete applying the finish to the outside of the box, along with any protective hardware and handles as desired (*Photo 24*).

For pole-mounting, a standard "top-hat" socket can't account for the angled cabinet shape, so make your own socket by attaching a 2 × 3 block to the inside of the porthole (*Photo 25*) and drilling a 1⅜″ or 1½″ hole as required through it at the correct angle, capping the block with a piece of plywood.

Install the woofer, reaching through the tweeter holes with a long screwdriver to drive the fasteners. Install the tweeters, weather-stripping their flanges for an airtight fit, though if you use a carpet finish this will double as weatherstrip. With KSN 1016A tweeters you'll need to sand off a smidge from the ends of the frames to fit three on each baffle, though four KSN1167A tweeters have plenty of room, and even five will fit with a bit of sanding of the frames.

Wire the tweeters in each vertical bank together in parallel, and then wire the two banks together in series. Lightly but thoroughly stuff the box with

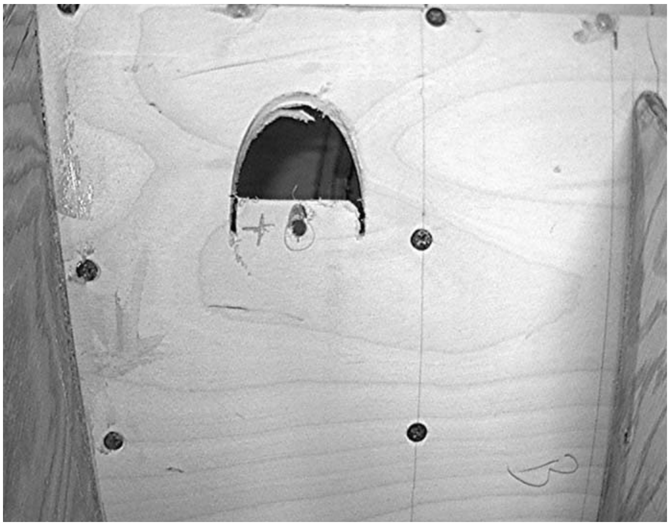

PHOTO 22: Removal of waste allows the duct hole to be completed.

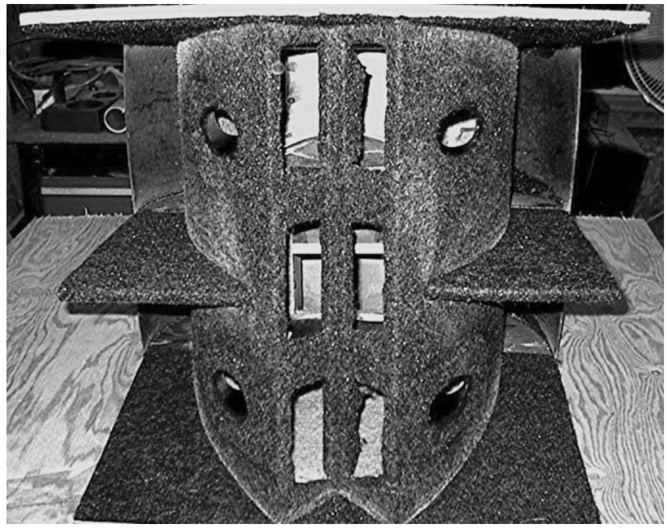

PHOTO 23: Apply a finish to the innards before attaching the sides.

poly-fill, being sure not to block either the rear of the woofer or the duct entrances (*Photo 26*). Install jacks of your choice, sealing airtight where the wire passes through to the inner chamber, and wire the woofer and the tweeters to it in parallel, in phase. Rim the porthole flange with weather-strip and install the porthole.

SETUP

If your band plays clubs of various sizes, you may want to own a few pairs of DR200s as opposed to just one or two pairs of something larger. For small audiences a single DR200 on either side of the stage is very adequate, as it will easily out-power a typical commercial compact 1 × 12 or 1 × 15 with horn cabinet. For larger audiences where a longer throw is required, stack up more DR200s as required, the lowermost boxes angled downward to catch the audience in close, the upper boxes aimed to the rear of the room (*Photo 27*). This same arrangement works quite well in permanent installations such as churches, where a lot of power isn't necessarily needed but a long throw is, and the higher the array the more control you'll have over both early reflections and outright echo.

For smaller rooms where the bass and kick drum aren't run through the PA, or for installations with vocal mi-

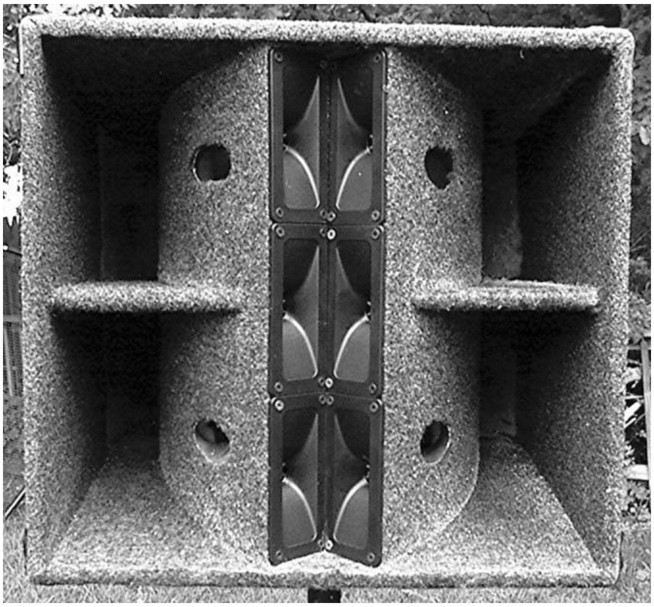

PHOTO 24: Front view of the finished DR200.

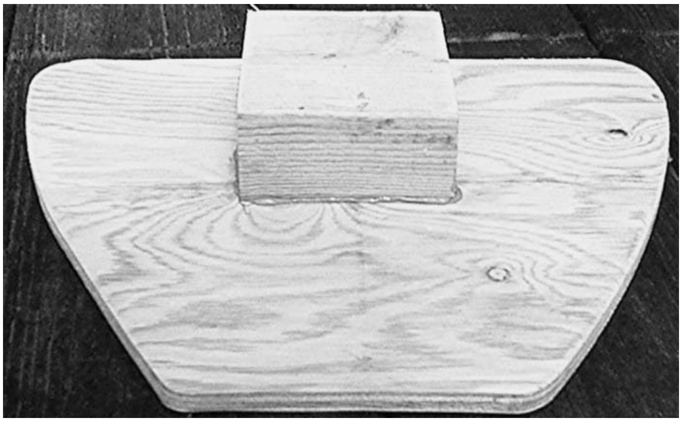

PHOTO 25: A 2 × 3 block on the inside of the porthole cover will serve as a pole socket when drilled out and capped with plywood.

crophones only, DR200s are fine by themselves. In larger venues use subs to cover the bottom end; one to consider would be my Tuba 24 (April ’04), one of which has more than enough output to keep up with two DR200s.

On the other hand, at 2ft^3 Tuba 24 is a bit larger and heavier than many club bands really need, especially those who don’t run at maximum SPL. They might want something like the Tuba, but in a smaller size, say, 18in^3—that could even do double duty as an auto-sound sub when not out on a gig. Guess what: it’s already on the way. My next project: The Tuba 18 Sub. ❖

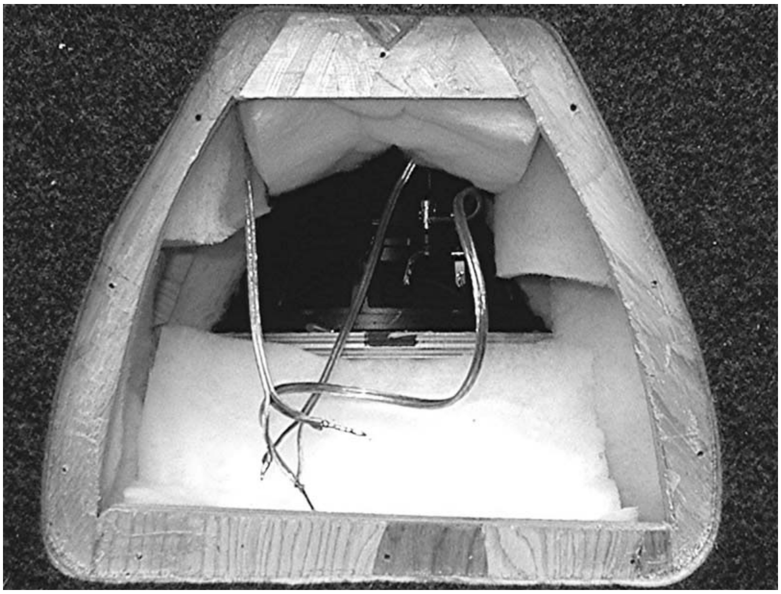

PHOTO 26: Stuffing the cabinet with poly-fill.

PHOTO 27: A pair of DR200s vertically stacked and splayed.

Dual Audio Amps for Biamped Designs

Here's a simple, compact design for dual-amplifier boards intended for biamped loudspeakers.

By John R. Peck and Daniel J. Cyr

Biamping has always been an option for hobby-grade speaker design, with advantages in damping factor, efficiency, and ease of crossover design. Biamped designs are becoming even more attractive as digital filter design for crossovers moves in on the ordinary hobbyist's price range and abilities. The problem is always the need for at least four amplifier channels in the simplest designs. This rapidly adds cost.

As graduate students, we'd like to solve this biamping problem cheaply and easily. Our solution is the amplifiers shown here, where two chip amplifiers support each other on a single-sided circuit board that you can make and stuff in an afternoon.

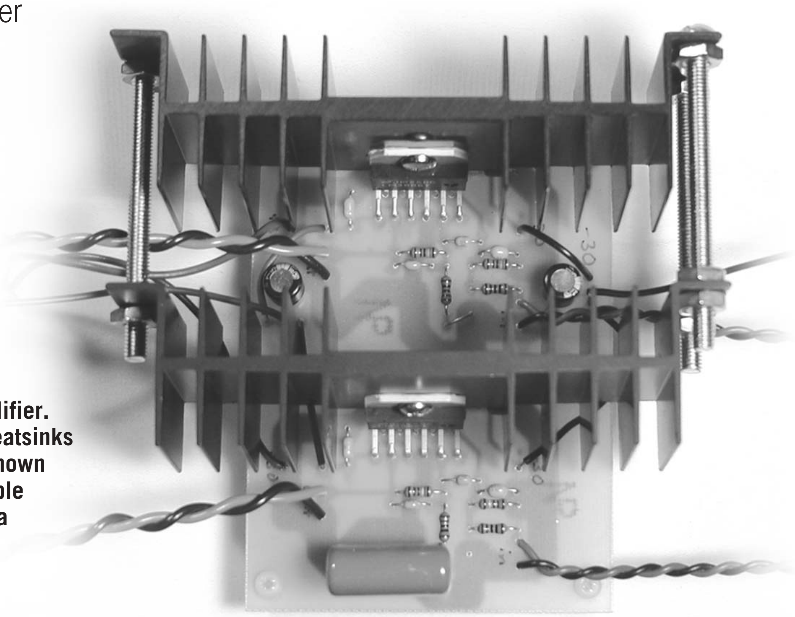

PHOTO 1: Finished amplifier. Bolting the heatsinks together as shown here is a simple way to make a very durable amplifier set.

AMPLIFIER DESIGN

The amplifiers use National Semiconductor's LM3886 ICs, and as such their design amounts to picking external components for these chips. For simplicity, I almost exactly copied the design used by Siegfried Linkwitz for his "Phoenix" loudspeakers[1]. The schematic is shown in *Fig. 1*. Linkwitz makes a provision for an output "padding" resistor to avoid ground loop problems which I omit, since I've never had any of these problems with the boards I've made with this design. The design also closely resembles the "typical application" circuit shown in the LM3886 datasheet[2].

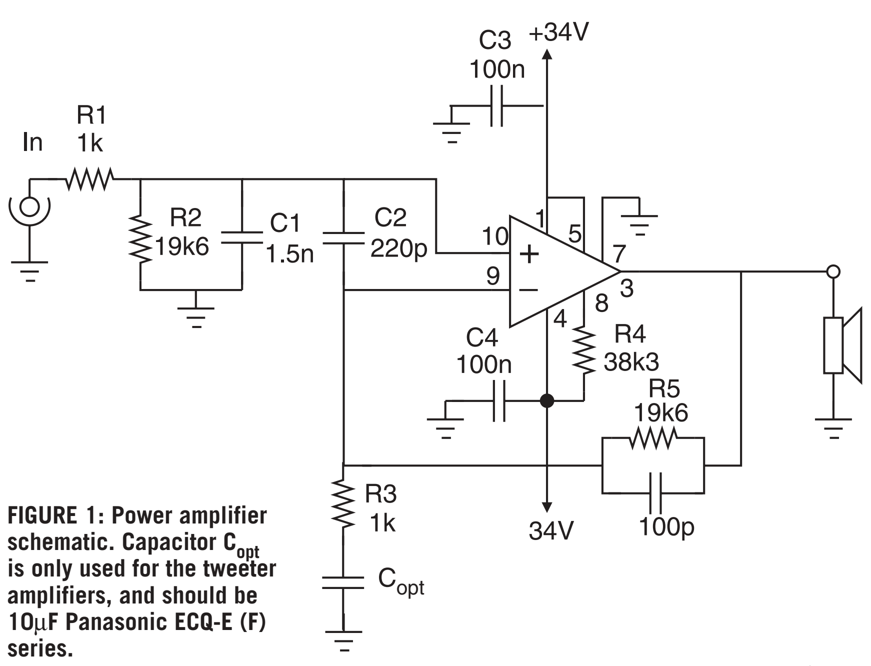

FIGURE 1: Power amplifier schematic. Capacitor C_{opt} is only used for the tweeter amplifiers, and should be 10μF Panasonic ECQ-E (F) series.

ABOUT THE AUTHORS

John R. Peck is a graduate student in physics working in the Electrical and Computer Engineering Department at the University of Wisconsin—Madison. His thesis work involves the use of electrical impedance spectroscopy for the study of electrochemistry. John is an audio hobbyist who likes building loudspeakers but doesn't like passive crossover design.

Daniel J. Cyr is also a graduate student in physics at UW—Madison, but is currently working at Fermi National Accelerator Laboratory in Batavia, Ill. His thesis work involves detection of high energy muons and precision electroweak processes observed at the Tevatron collider. Dan is interested in stand-alone digital signal processing for crossover design.

PRINTED CIRCUIT BOARD

I've found that the 3″ × 4″ pre-sensitized circuit boards available from Parts Express are very easy to expose using an ordinary set of "broad spectrum" fluorescent lights and an overhead projector transparency. The small board size is very handy for someone who doesn't make a business out of circuit board production. The artwork (*Fig. 2*) is designed to match the size of these boards. I determined that two 20W tubes separated from the board by about 3″ will expose the photoresist adequately in 10 minutes. Developing should take under 5 minutes. Ferric chloride etchant is available from Radio Shack.

I use three different drill sizes for the boards. A #56 drill works well for the IC pins, the big 10µF optional capacitor, and all holes where wires will be soldered for inputs, outputs, and the power supply. For all other components (resistors and capacitors), a #68 drill will work nicely. You can drill the four corners for mounting the board for whatever mounts you have.

STUFFING DIAGRAM

Because these are single-sided boards, I ultimately needed to use jumper wires to make some connections. I found it easiest to make the power supply connections to the middle of the board, bypass them with a capacitor, and then run jumper wires to the indi-

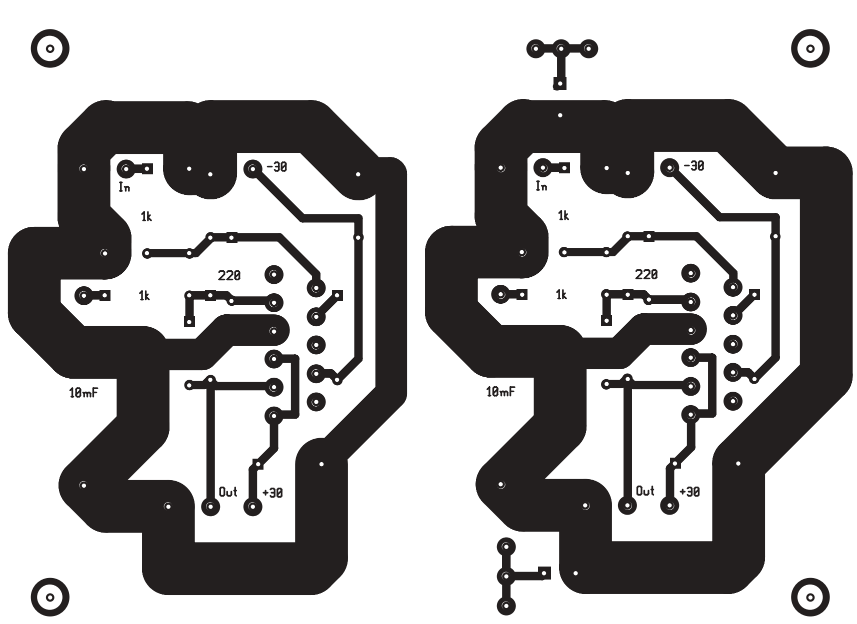

FIGURE 2: Power amplifier artwork (actual size).

A-2277-2

vidual amplifiers. I made all input and output connections with twisted pairs of wires.

Figure 3 shows where to install components on one amplifier on the board. If you want to use one amplifier for a woofer and one for a tweeter, one amplifier should use the capacitor C_{opt} and the other should short one side of resistor R3 to ground with jumper J. It's easiest if you install the big C_{opt} capacitor on the amplifier where it can be closest to the edge of the board.

Heatsinking the amplifier chips is important not only to cool the chips, but to increase the durability of the amplifiers. You'll find the amplifiers much easier to work with if you bolt both heatsinks together as shown in *Photo 1*. I've found some big heatsinks with mounting tabs at the edges to work well for this. I drill and tap the heatsinks for 6-32 bolts to attach the TO-220 amplifier packages, and run 10–32 × 2.5″ bolts between the two sets of mounting tabs to fix their relative positions.

POWER SUPPLY

We built the power supply (*Fig. 4*) to test the finished amplifiers. There's nothing unusual about this supply, but note that the transformer used is quite large (48V CT, 7A secondary). This is overkill for testing a single amplifier, which will overheat before it comes close to being limited by the transformer during continuous testing. Still, a biamped two-way system will use four amplifiers (two boards).

With the components and heatsinking shown here, the amplifiers make about 30W into 8Ω dummy loads as shown in *Fig. 5*. Running four amplifiers at maximum into 8Ω will draw about 3.5A, which is fine for this transformer but not for the alternate listed in *Table 2*.

DISTORTION MEASUREMENTS

We measured distortion with a single amplifier driving an 8Ω dummy load with a 1kHz sine wave. An M-audio DELTA 1010LT sound card[3] used with Windows sound programs produced and recorded the sine waves. MATLAB[4] calculated FFT spectra. As shown in *Fig. 5*, the amplifiers make about 30W into 8Ω with 0.1% distortion.

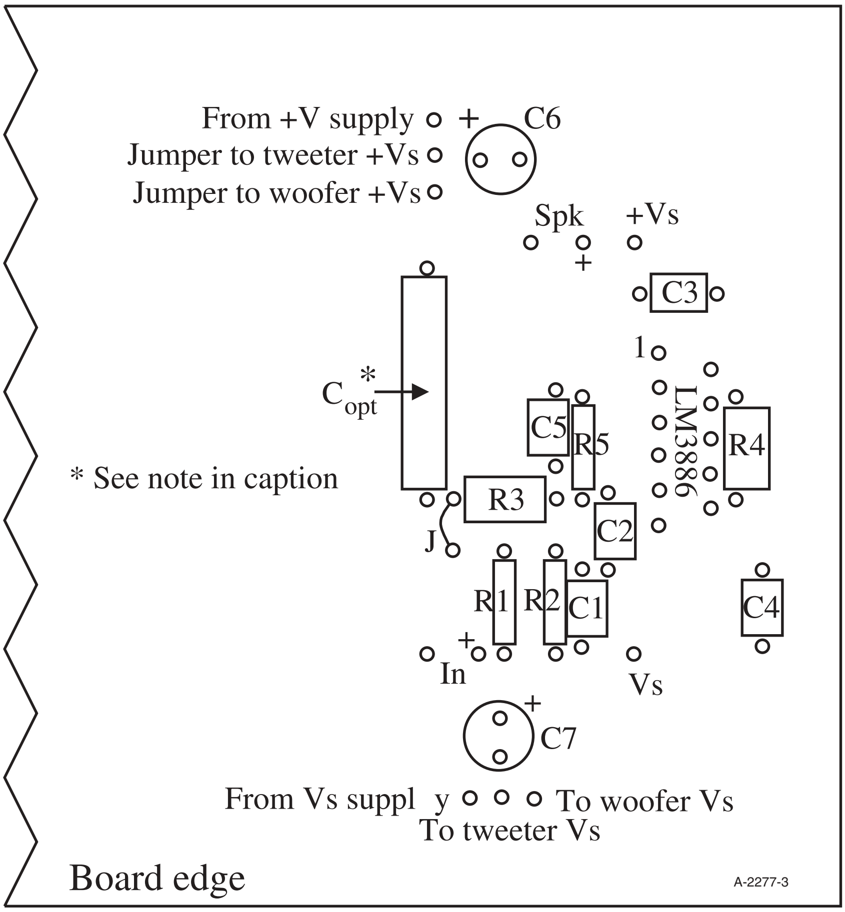

FIGURE 3: Stuffing diagram for one power amplifier viewed from the component side.

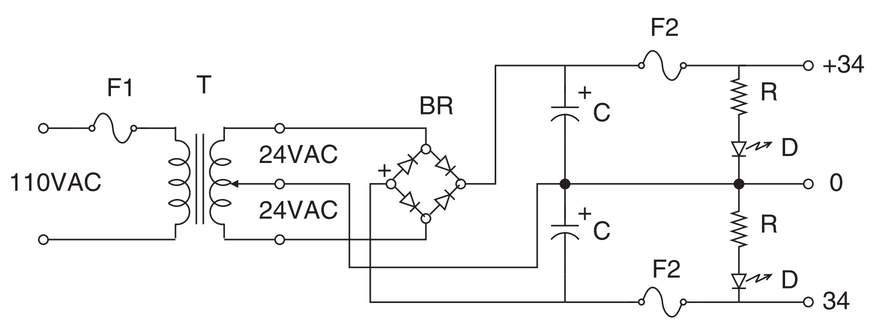

FIGURE 4: ±34V supply schematic.

TABLE 1

PARTS LIST FOR STUFFING DIAGRAM SHOWN IN *FIG. 3*. USE JUMPER WIRE J AND OMIT C_{OPT} FOR THE WOOFER AMPLIFIER, AND VICE-VERSA FOR THE TWEETER AMP.

SYMBOL	VALUE	SOURCE	PART NUMBER
R1, R3	1k 1%, metal film	Digi-Key	1.00KXBK-ND
R2, R5	19k6 1%, metal film	Digi-Key	19.6KXBK-ND
R4	38k3 1%, metal film	Digi-Key	38.3KXBK-ND
C1	1n5 50V 5%, ceramic C0G	Digi-Key	399-1828-1-ND
C2	220p 50V 5%, ceramic C0G	Digi-Key	1004PHCT-ND
C3, C4	100n 50V 10%, ceramic X7R	Digi-Key	1109PHCT-ND
C5	100p 50V 5%, ceramic C0G	Digi-Key	1001PHCT-ND
C6, C7	47µF 35V 20%, Al electrolytic	Digi-Key	P948-ND
C_{opt}	10µF 100V 10%, polyester film	Digi-Key	EF1106-ND
LM3886	LM3886T	Digi-Key	LM3886T-ND
J	Jumper wire		
Pre-sensitized circuit board (3×4″)		Parts Express	055-102
Heatsink		Newark	58F502

TOTAL SYSTEM COST

The cost of these amplifiers depends heavily on the use of surplus outlets. The cost of all parts used for one board excluding the heatsinks is about $25. The cost of the new heatsinks listed in *Table 1* ($9 each) will almost double this.

With all new parts, an amplifier board (two amplifiers) costs about $43. Compare this to the very nice ILP amplifier modules reviewed in *aX* 7/02. The 30W and 60W ILP modules are priced at about $28 and $35, respectively. Of course, calculating the cost of your assembly time is never easy, but, as you know, building your own equipment has a value that can't be measured. ❖

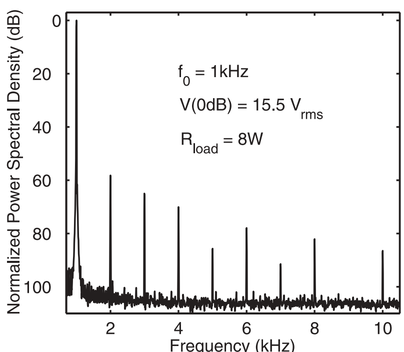

FIGURE 5: Distortion measurement at 30W. A-2277-5

TABLE 2

COMPONENTS USED IN ±34V SUPPLY, FIG. 5. APEX JR. IS A SURPLUS ELECTRONICS STORE, SO ALTERNATE PARTS ARE SUGGESTED.

SYMBOL	VALUE	SOURCE	PART NUMBER
F1	3A slo-blo	Digi-Key	F329-ND
F2	4A fast-blo	Digi-Key	F331-ND
T	48V CT, 7A secondary	Apex Jr.	
T (alternate)	48V CT, 3A secondary	Parts Express	120-225
C	26,000µF 63V DC	Apex Jr.	
C (alternate)	27,000µF 63V DC	Digi-Key	P10641-ND
BR	12A, 100V	Digi-Key	GBPC1201-ND
R	3k ½W	Digi-Key	3.0KH-ND
D	Green LED	Digi-Key	L10065-ND

REFERENCES

1. S. Linkwitz, www.linkwitzlab.com/images/graphics/3886amp.gif (January 2003).

2. National Semiconductor Corporation, LM3886 Overture™ audio power amplifier series high-performance 68W audio power amplifier with mute (August 2000).

3. M-Audio: www.m-audio.com.

4. MATLAB: www.mathworks.com.

5. C. Hansen, "Four ILP Amplifier Modules," *aX* 7/02, p. 58.

SOURCES

Digi-Key Corp
701 Brooks Ave. South
Thief River Falls, MN
56701-0677
1-800-344-4539
www.digikey.com

Parts Express
725 Pleasant Valley Dr.
Springboro, OH 45066-1158
1-800-338-0531
www.partsexpress.com

Apex Jr.
3045 Orange Ave
La Crescenta, CA 91214
1-818-248-0416
www.apexjr.com

Newark InOne
4801 N. Ravenswood
Chicago, IL 60640
1-800-463-9275
www.newark.com

The "MCUTracer," Part 1

This microprocessor-controlled curve tracer allows you to analyze the characteristic curves of tubes and semiconductor devices.

By Jack Walton and Martin Hebel

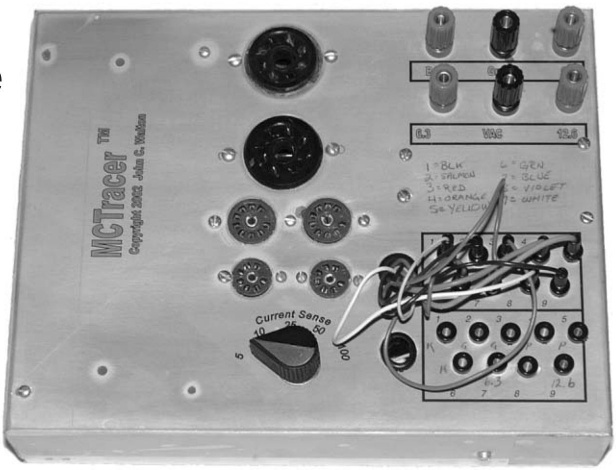

PHOTO 1: MCUTracer.

A number of articles in *Audio Amateur, Glass Audio, Electronics* (nee *Wireless*) *World*, and so on, have described tube and transistor curve tracers that would satisfy one or more of my interests in examining the characteristic curves of vacuum tubes and semiconductor devices. Nevertheless, none of the designs are flexible enough to accommodate the potential uses for which you can use a microcontroller, analog-to-digital converter, and PC.

This article describes a modular, microprocessor-based curve tracer, using the Parallax Inc. Basic Stamp BS2. The design is sufficiently flexible to allow you to plot and display several variables in real time.

Parallax Inc. has a macro for Microsoft Excel[1], StampDAQ, available on their website, which takes the data from the microprocessor and sends it directly to the PC for use in the popular spreadsheet program. With the MCUTracer and Basic Stamp, you can plot data in real time, save the results, or perform statistical analysis. While other microprocessors could be used for this article, we decided to stick with the Basic Stamp as Parallax funded the development of the software used here.

In part 2, we will show how you can POOGE an existing high voltage power supply, the Heathkit IP-17, for control by the MCUTracer, supplying and measuring B+ and C– voltages as well as plate current. In part 2 we use a more robust analog acquisition and plotting tool, StampPlot™, written by co-author Martin Hebel.

You can adapt the MCUTracer basic setup to a variety of purposes, such as semiconductor curve tracer, spectrum analyzer, logic analyzer, and so on (*Photo 1*). As it is configured for part 1, it can test two devices at the same time. Thus you can quickly compare the two triode sections of 12AU7s or a pair of 6Lds (or with a little ingenuity, a quad), and store or print the results. *Photo 2* shows the inside of the unit.

TUBE AND TRANSISTOR TRACERS

Most tube and transistor curve tracers use a ramp generator to provide the voltages for plate and grid, gate and drain, and so forth. As the drive voltage is swept, an oscilloscope display is triggered (or the X-axis is driven) and displays the results. Borbely described a tracer in *Audio Amateur* in 1990[2] and provided a list of references to such designs. Petrowsky offered a clever vacuum curve tracer using the X–Y axes inputs of an oscilloscope[3].

While Petrowsky's Tube Tracer was useful in helping determine the merits of a particular bottle, it would not allow you to store and record data, or to compare data among lots of tubes for matching. Although Joe Carr did not offer a specific tracer design in *Ham Radio*[4], he did lay out the fundamentals for power tube testing and provided the schematic for a *gm* tester for high-power transmitting tubes.

More recently, George Steber wrote an article in *Circuit Cellar*[5] which used a PC soundcard to trace current and voltage. In addition, Maxim has an application note—of course, using Maxim ICs—which shows how you can use a PC printer port as an IV Curve Tracer[6]. While all of these articles provide solutions, using StampDAQ simplifies design implementation, and you don't need to be a Visual Basic or C++ guru to get the device to work. Of course, the problem with a ramp generator is that the measurements lag the movement of the ramp due to the time necessary for analog-to-digital conversion.

The MCUTracer, as shown here, departs from the strictly analog approach by using an inexpensive and easily programmed chip, the Basic Stamp II from Parallax[7], to monitor the output from a Linear Technology 10-bit ADC. While part 2 describes a digitally modified power supply for the MCUTracer, the old "Armstrong Method," i.e., twiddling the knobs on an adjustable high-voltage supply to adjust grid, screen, or plate voltages by hand, will also work. The data collected by the MCUTracer is

ABOUT THE AUTHOR
Jack Walton is an investment consultant and entrepreneur in Short Hills, N.J. His educational background is in chemistry, physics, and finance. He has had a lifelong interest in electronics, ham radio, photography, and the arts.

Martin Hebel is an Assistant Professor in Electronic Systems Technologies at Southern Illinois University, where he teaches process control and microcontroller programming. He is also a partner in SelmaWare Solutions and the developer of StampPlot and StampDAQ software and co-author of BASIC Stamp texts. During the summer he hosts robotics camps.

channeled to your PC using the recently published Microsoft Excel™ Macro, StampDAQ from Parallax Inc., which is available free on the Parallax website (www.parallax.com).

CIRCUIT DESCRIPTION

The circuit consists of six non-inverting op amp front ends. Plate and grid voltages are stepped down and calibrated with multi-turn adjustable potentiometers. No voltage divider is necessary for the plate current measuring channels, since the level is within the acceptable range for the ADC.

The six-channel, 10-bit ADC is decoupled from the op amps with 51Ω ¼W resistors and 100nF capacitors. In application, I found that I could reduce extraneous noise with this method, which, incidentally, is also suggested in the product design file from Linear Tech.

The input resistors for the high voltage measurement circuits should be made up from three ½W 330k resistors to minimize the nonlinearities of carbon resistors at high voltage. Do not use ¼W resistors in this application. Power for operational amplifiers, ADC, and Basic Stamp is derived from the 12.6V tube filament transformer output of the Heath IP-17 power supply (*Fig. 1*). The voltage is rectified, filtered, and regulated to ±5V with a pair of 7805/LM79LO5L regulators.

CONSTRUCTION

I built the MCUTracer on one 3 × 6″ printed circuit board which houses the op amps, ADC, Basic Stamp, and male 0.100″ Molex headers for use in the microprocessor-controlled power supply. The printed circuit board PCB X-ray (with top layer traces as dotted lines) and the reflected image of the bottom (trace layer) are shown in *Figs. 2* and *3*. The stuffing diagram is shown in *Fig. 4*.

Instead of agonizing over a complex switching arrangement for the MCU-Tracer, I decided to place pairs of 7, 9-pin, and octal sockets. Each pin of the tube socket pair connects to a flexible probe fitted with a miniature banana plug. That is, all pin 1s are connected to a probe with—in this case—a black lead. There are nine mini-banana jacks that provide plate, screen, control grid, cathode, and heater voltages. For those enamored of the 300E, 6CW4, or Compactrons, knock yourself out and use the appropriate socket!

TYPICAL SETUP

Plate current is measured by reading the voltage drop from cathode to ground across a resistor. A typical setup, showing an unmodified Heath IP-17 power supply, is in *Figs. 5* and *6.* In the lowest current mode (10mA) the voltage drop across 250Ω, 2.5V, is measured. A 100Ω resistor is used for 25mA and 25Ω for 100mA.

The MCUTracer is designed so that you can test more than one tube. *Figure 7* shows a general outline of how to do this.

TESTING TUBES WITH SCREEN GRIDS

You can connect the screen grid to the plate with a jumper cable. (You can see in *Photo 1* that multiple B+ jacks are available). If you choose to add resistance to the path between the B+ supply and the screen grid, just fashion a jumper cable with the appropriate resistance in the path. Of course, safety considerations are paramount, and any such resistance should be contained in a small box or wrapped in a couple of layers of heat-shrink tubing to avoid the risk of injury.

The code (to see the code, and for instructions on how to use it, visit www.audioXpress.com, then click on the current issue, and follow the link title "articles and addenda.") will operate the MCUTracer with any high voltage power supply. The program uses the full 10-bit range of the LTC1093 ADC to measure positive voltages using the "UNIPOLAR" multiplexer address bit. Negative voltages are measured to 9-bits using the "BIPOLAR" ADC setting. This lower precision is necessary to allow for a "SIGN" bit indicating whether the grid voltage is positive or negative. Choosing the mode is done by deselecting the UNIPOLAR bit when the MUX address is selected. The addressing modes are described on Linear Tech's website. The negative voltage is "Two's Complemented" sent to the spreadsheet. In part 2, we list the code used to control the high voltage power supply and the StampPlot macros.

PHOTO 2: Unit's innards.

SETTING UP THE MCUTRACER

The setup procedure described here makes use of an unmodified high-voltage power supply. If you modify the high-voltage supply as described in part 2, you will use the static, rather than sweep, function.

Each bit of the ADC will measure Vref/1023, or 4.00mV with the LM4040-4.1V reference. Adjust the B+ output of the supply to 100V on the appropriate HV header. Adjust the trimmer pot so that 1.000V is present on the appropriate output pin of the op amp or input pin of the ADC. With this setting, a 100V plate reading will be represented by 250 bits from the LT1093 ADC to the Basic Stamp.

Similarly with the C– supply, apply –12.50V to the grid voltage input header and adjust its trimming potentiometer until the voltage at the corresponding pin of the ADC measures –1.000V. The full-scale reading of the grid-measuring circuit is a little over –50V, and each bit will represent 50mV. If you can only "get close" with the trimming potentiometer, you can always make a correction in the Excel spreadsheet for the actual value.

You make the connections from the curve tracer to the power supply with a

> You should exercise great care in setting up the MCUTracer, because lethal voltages are present on the circuit board. Death or serious injury can occur if you are not careful. If you have used a 3-resistor high-voltage divider, make sure that any exposed nodes are insulated with electrical tape or heat-shrink tubing to avoid personal injury. Use an insulated screwdriver to adjust the trimming potentiometers.

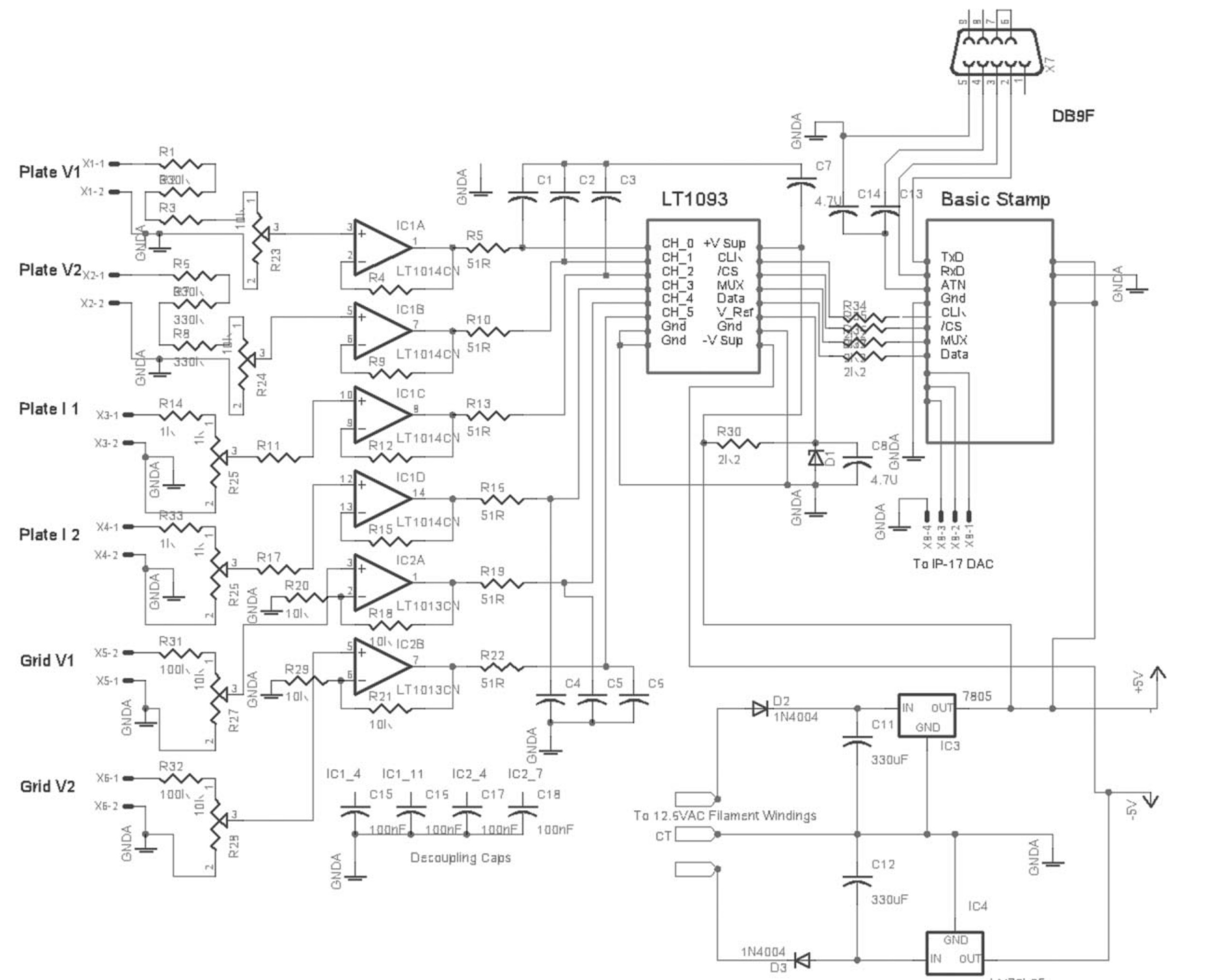

FIGURE 1: MCUTracer circuit.

A-2211-1

cable with banana plugs, which are attached to the MCUTracer (and not the other way around, for safety's sake). Make sure that the B+, C–, heater, and ground connections are correct. Make the connections from the tube pins to the appropriate input of the MCUTracer.

Note: test your tubes for shorts before using the MCUTracer! There's no point blowing up your power supply, so make sure you try this simple test first. Do not remove any of the plugs from their jacks while power is supplied to the MCUTracer!

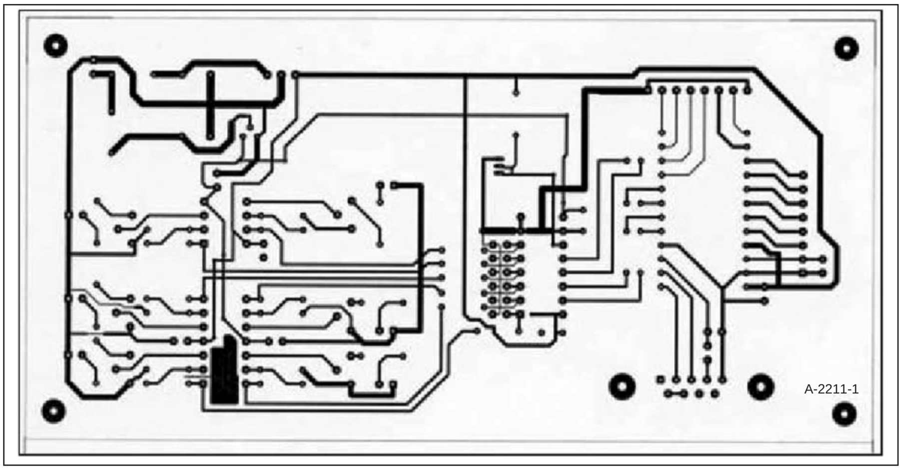

FIGURE 2: Topside PCB pattern 1.

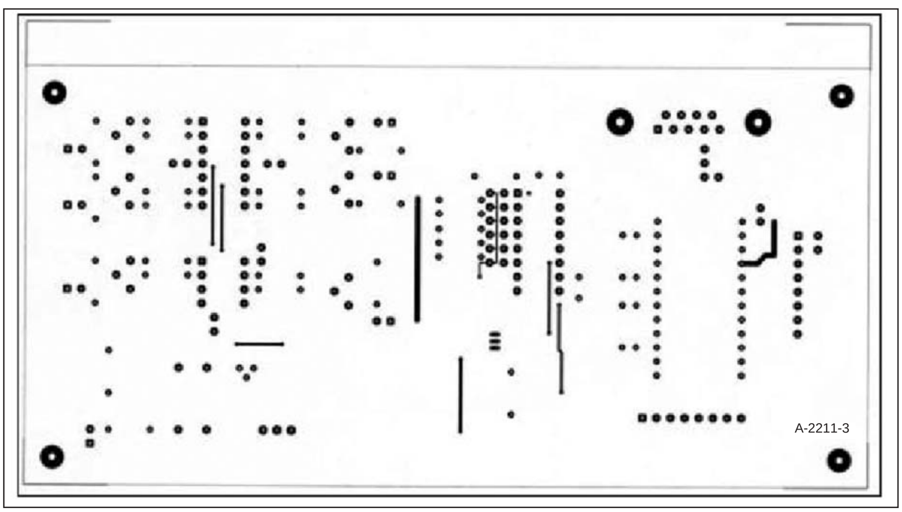

FIGURE 3: Reflected PCB pattern 1.

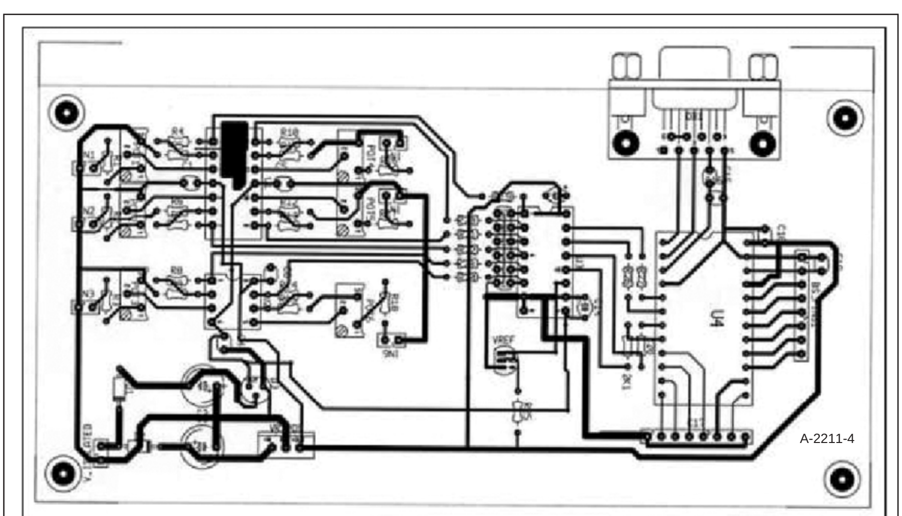

FIGURE 4: Stuffing diagram 1.

RUNNING THE MCUTRACER

After you have checked that the setup is correct, apply power to the MCU-Tracer through the high-voltage power supply. Allow the "device under test" (DUT) a minute or so to warm up. While the warm-up is taking place, start Excel and open the StampDAQ macro within the program. The Stamp-

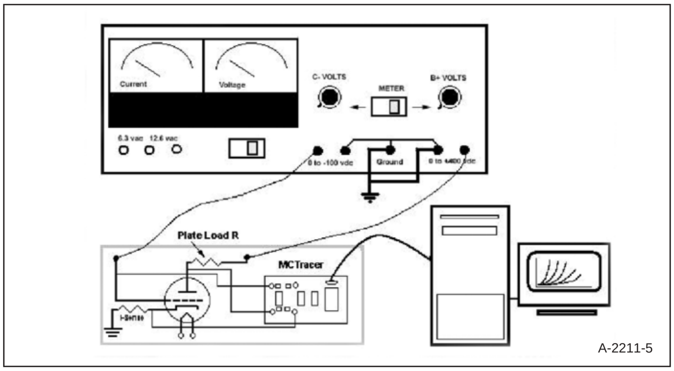

FIGURE 5: System setup.

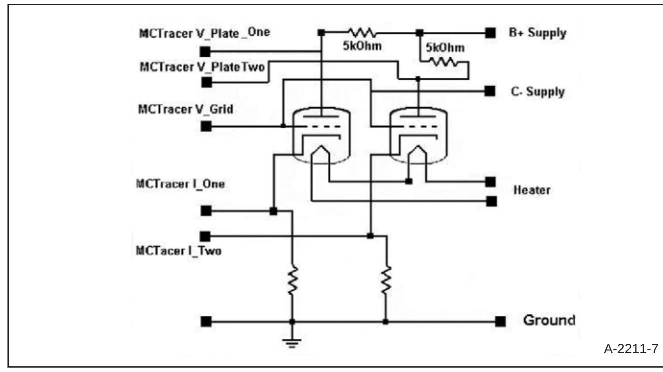

FIGURE 7: Testing two tubes.

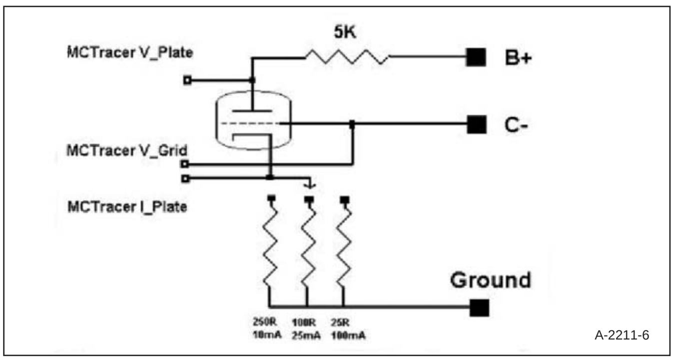

FIGURE 6: Setup schematic.

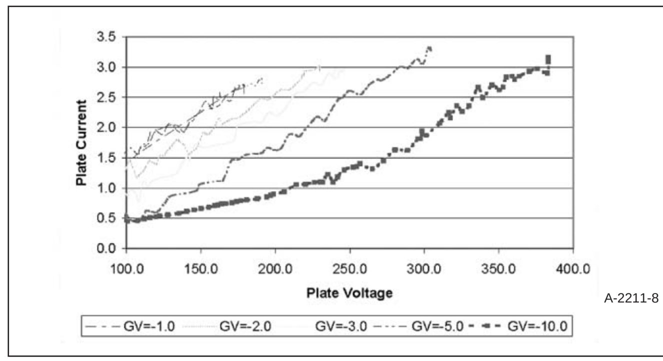

FIGURE 8: ECC86 characterization curve.

DAQ icon will appear on the spreadsheet prompting you to "Connect." Adjust the B+ and C– voltages of the power supply and current measuring range switch to levels that are appropriate for the DUT.

When you press the Connect button, data will be transmitted from the MCUTracer to your PC. Therefore, after connecting, you can sweep the B+ voltage by hand and you will see the B+, C–, and plate current will print in separate rows of the spreadsheet. To go to a different C– voltage, just pause the MCUTracer, adjust the C– voltage, and reconnect.

GRAPHING THE DATA IN EXCEL

To see the results in real time, set up a generic "Scatter Plot" graph with column B, plate voltage, as the X-axis or independent variable. Set up column C, plate current, as the Y-axis. An example of the finished product, a characteristic curve for an ECC86, is shown in *Fig. 8*.

If you set up the graph prior to connecting, the results will be displayed in real time just as though you had a Tektronix 576 or 577 curve tracer! *Figure 9* shows the differences between the two halves of an old 6SN7GT which I found with a box of tubes. Another example, a not so pretty junk box 12AT7 with grid voltages ranging from –2.0 to –10.0V (*Fig. 10*).

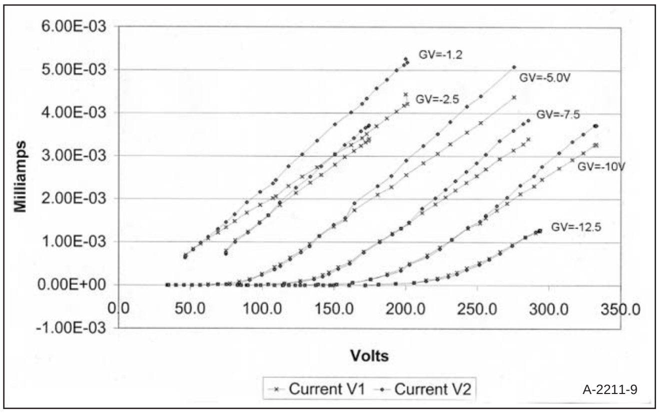

FIGURE 9: Differences between two 6SN7GT halves.

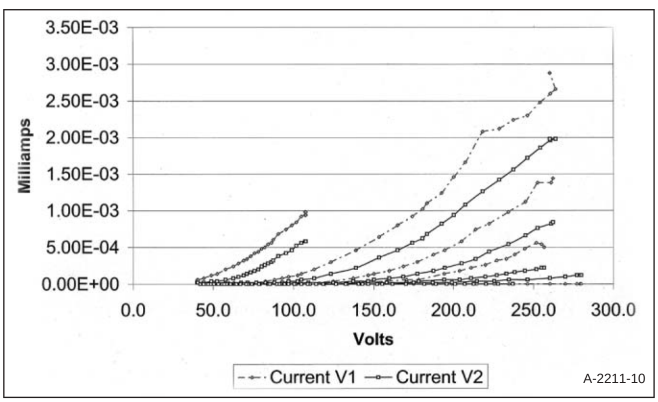

FIGURE 10: A 12AT7.

CONCLUSION

The microprocessor-controlled curve tracer is not a daunting project. For the investment of a few hours, you can have an instrument that will chart voltage and current in real time. While this first application is a vacuum tube characteristic tracer, it is apparent that the modularity of the device lends itself to a wide variety of applications in which one or more variables are plotted against each other.

Part 2 of this series will describe a digital-to-analog converter/voltage amplifier controlled by the Basic Stamp, using a StampPlot Graphical User Interface to automatically provide the voltages necessary to test tube characteristics. ❖

REFERENCES

1. You can find the StampDAQ macro at www.parallax.com/html_pages/downloads/software/software_basic_stamp.asp.

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3. Petrowsky, William J., "From Oscilloscope to Curve Tracer," *Glass Audio*, 2/90, p. 10ff.

4. Carr, Joe, "Testing Power Tubes," *Ham Radio*, April 1978, p. 60ff.

5. Steber, George, "Tracing Current and Voltage," *Circuit Cellar*, January 2004, p. 56 ff.

6. "PC Printer Port Controls IV Curve Tracer," Maxim Semiconductor, www.maxim-ic.com, Application Note 253, July 2001.

7. Parallax, Inc., www.parallaxinc.com. The Basic Stamp II is available from Jameco and many other vendors.

Modifying Dynaco's SCA-35

This article features modifications to Dyna's classic SCA-35 integrated amplifier from power cord to output, with power supply upgrades, new phono section with accurate RIAA equalization based on the RCA manual design, improved tone control section with tone control bypass, and modified power amplifier section. **By James Lin**

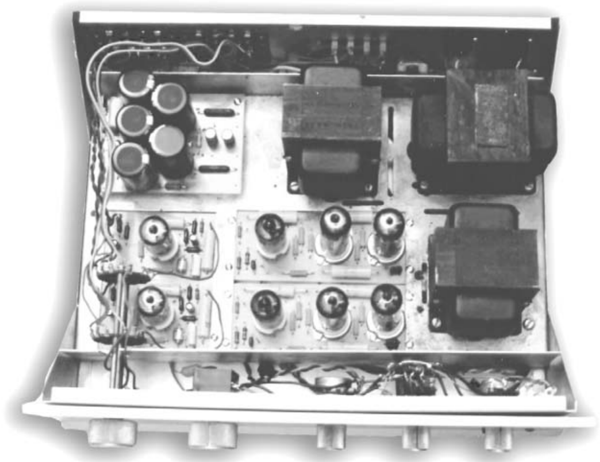

PHOTO 1: Dynaco SCA-35 overall mod.

Last year while I was visiting a friend in Italy, we dug out her father's old hi-fi set. The amplifier, a mass-market transistor design, promptly blew a fuse. Three blown fuses later, the amp was consigned to the Dumpster, and I offered to get her another amplifier. She is a music lover, not an audiophile, so I wanted something simple, reliable, good, and inexpensive.

Since she also has a turntable and records, the amplifier needed a phono section—something that many integrated amps don't have these days. Being a fan of tubes, I thought of the Dynaco SCA-35, which was designed for music lovers with limited space and budget, a desire for simplicity, and modest power demands—it puts out only around 15W per channel. Its low parts count should make it reliable and easy to repair, and new tubes are still being made for it. It was in *Stereophile*'s recommended components list from 1965 to 1971, ranked as high as class C ("far better than average home hi-fi") in 1968, although to be fair, that was the lowest class listed that year.

More recently, the 1999 *Vacuum Tube Valley* catalog said it was "considered by many to be the best sounding integrated tube amp ever made." Considering that there are some very pricey modern integrated tube amps around, I'm not sure I can buy this. Nevertheless, it illustrates the high regard in which it is held even today.

The SCA-35 is pretty common—about 80,000 of them were made between 1962 and 1968[1]. It was discontinued around 1973, so probably over 100,000 were made altogether. They turn up regularly on eBay for $100-$200. Since they are still functional after 30 years or more, they must be pretty reliable!

However, finding one for European power line voltages took several months. In the meantime, I obtained a manual and some old test reports to see what its original performance was. Despite its commonness, relatively little has been published on modifying the SCA-35, perhaps because most of its owners are music lovers, not audiophiles.

MODIFICATIONS

Why modify? If you collect vintage electronics, original condition is important and you shouldn't. On the other hand, if you are interested in using vintage equipment, modifications can improve reliability and/or performance.

Modifications of old electronics generally fall into four categories. First, replacing parts with better, closer tolerance parts of the same nominal value. Second, enhancing the power supply, which was generally the area that was skimped, particularly in budget electronics. Third, circuit modifications to improve performance. Fourth, replacing the original circuits with new de-

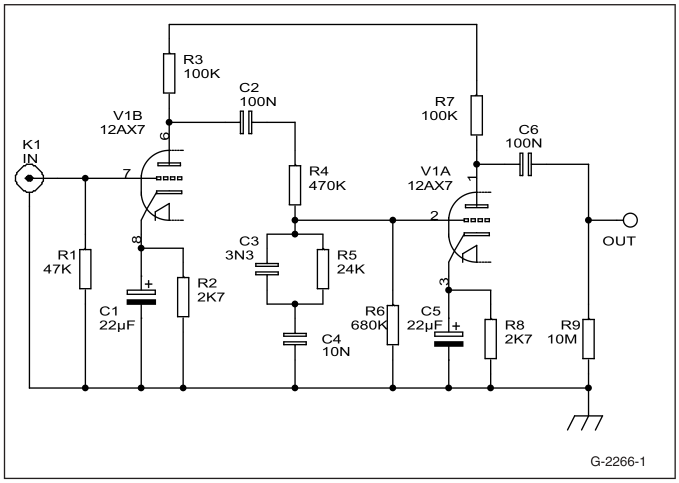

FIGURE 1: Phono preamp schematic.

ABOUT THE AUTHOR
James Lin became interested in building audio equipment from reading *The Audio Amateur.* He has written articles for *The Audio Amateur*, *Speaker Builder*, *Glass Audio*, and *audioXpress.*

signs. In the case of the SCA-35, I did some of each.

Before doing any modifications, it is a good idea to check that the unit is in working order. A manual is another necessity—an Internet search should turn up several sources. A good general reference is the article "Rebuilding Tube Amplifiers" by Staggs and Crawford in *Glass Audio* in 1991[2]. Although it covered Scott integrated amplifiers, there is much good information in it that I won't be covering.

You can divide the SCA-35 into the following sections and consider each individually: 1) the power supply, 2) the phono/tape preamplifier, 3) the line section, and 4) the power amplifier section.

POWER SUPPLY

Like all stereo components of its time, the SCA-35 had a two-wire power cord. I replaced the power cord with a three-wire grounded cord, which allows the case to be securely grounded, eliminating any shock hazard. This particular safety feature is now mandatory in many if not all countries and is a recommended safety modification for any vintage electronics.

The power plugs in Italy are different from US plugs, so it made no sense to wire it with a US plug. Fitting a standard IEC socket would allow me to change power cords easily, but there wasn't space to fit a socket and leave the fuse holder in place. However, a combination IEC socket/fuse holder would cover both the power cord and fuse holder holes—if I fitted it sideways. I used a nibbler to rough out the hole and a mill file for final fitting. A snap-in mount covered up the remaining gaps (*Photo 1*).

In wiring the socket, the center pin is the safety ground and is connected directly to the chassis; the fused pin is the "hot" wire connection, which goes to the power switch; and the other pin is wired to the power transformer. This ensures that if the fuse blows, the power line voltage is disconnected. With this socket/fuse holder you must remove the cord before accessing the fuse, guaranteeing that the power will be disconnected from the unit.

Incidentally, note that the European model does not have the accessory power outlets of the US model. Since these are two-prong outlets, I recommend disconnecting them. This should be no great loss because source components are switched separately anyway.

Next, replace the original power supply diodes with fast recovery or HEXFRED diodes. This will significantly decrease switching noise in the power supply compared with the original silicon rectifiers, which should improve the sound.

Then, replace the old power supply electrolytic caps, which are the parts most likely to fail. Although you can replace the original multi-section electrolytic capacitors with new multi-section caps with similar specifications, a capacitor board using separate caps is cheaper, and can accommodate higher capacitance than the original, for a stiffer, more stable supply with lower hum. Sheldon Stokes has a nice circuit board design on his web site[3], which affords space for the power supply diodes and separate output tube cathode resistors (discussed later).

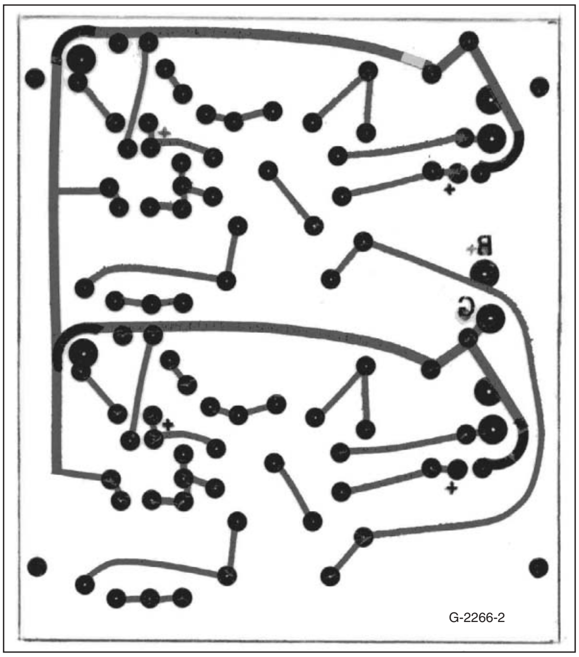

FIGURE 2: New PC-11 phono circuit board.

This board has one ground connection hole, which is not very accessible when the board is in place. So, I soldered a wire to it and used a piece of perforated board across one of the holes for the multi-section capacitors to provide a location for joining all the other ground connections to this power supply ground.

Finally, I added a current surge limiter in line with the primary power supply to soften the jolt to power supply capacitors and tube filaments at turn-on. It decreases the final voltage to the power supply by about a volt.

Incidentally, all the power supply caps should have a minimum 450V rating, because this unit uses solid-state rectifiers, which turn on instantly. Thus, all of the power supply caps will see full voltage immediately—which is close to 450V. Then, as the tubes turn on, the supply voltages will decrease down to their final values. The same holds true for any of the circuit coupling caps that are connected to the high voltage supply via plate resistors.

PHONO/TAPE HEAD PREAMPLIFIER

This section originally consisted of one 12AX7 tube per channel, which functioned as a phono or tape head preamplifier using feedback equalization. The phono preamp could accept low output (moving magnet), high output, and ceramic cartridges at different input sockets by switching between different input loads and equalization feedback networks. The multiplicity of switch contacts compromises sound quality. Since tape head, high output, and ceramic cartridges are long obsolete, you can hard-wire this section for LP record playback, eliminating a couple of switch contacts.

Dynaco did not list any specifications for RIAA equalization accuracy; however, contemporary reviews reported it as "+1, –2.5 dB"[4] or "within better than 2dB"[5]. While this was considered "no significant error"[4] at that time, current standards are considerably tighter.

The specified gain of the low-level phono stage was 48dB at 1kHz, leaving very little excess gain for feedback equalization. In fact, without positive feedback via resistor R3 in the original circuit, there is not enough forward gain to provide any feedback at all below about 50Hz. As Stamler has noted in *audioXpress*[6], with a small feedback margin any variations in gain due to tube aging, tube replacement, or circuit loading will cause significant variations in the equalization curve.

Therefore, I decided to substitute a stable passive equalization design, specifically, the phono preamp in the RCA tube manual. This design uses the same tubes and is even older than the SCA-35. It has a nominal RIAA accuracy within ±0.4dB if you omit the capacitor across the output. Despite its simplicity, it has gained something of a reputation in the "ultra-fi" community[7].

There are four minor changes in my version (*Fig. 1*). First, the input section is not de-coupled from the high voltage supply by its own resistor and power supply capacitor due to lack of space. Second, I substituted a 24k or 24k3 resistor for the original 22kΩ in the RIAA

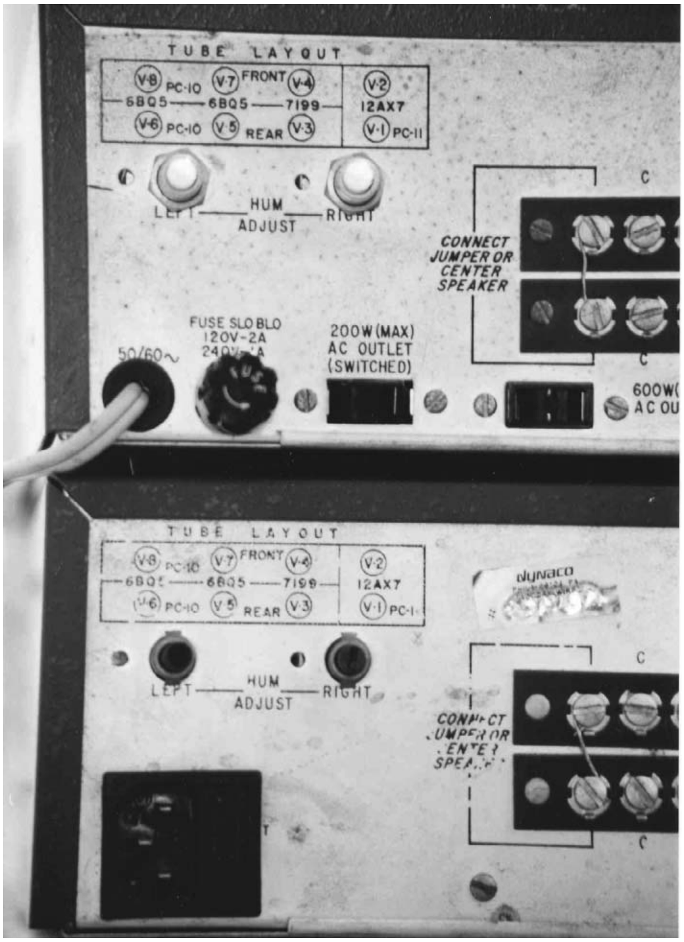

PHOTO 2: IEC socket/fuse holder modification on the bottom, compared with original design on top.

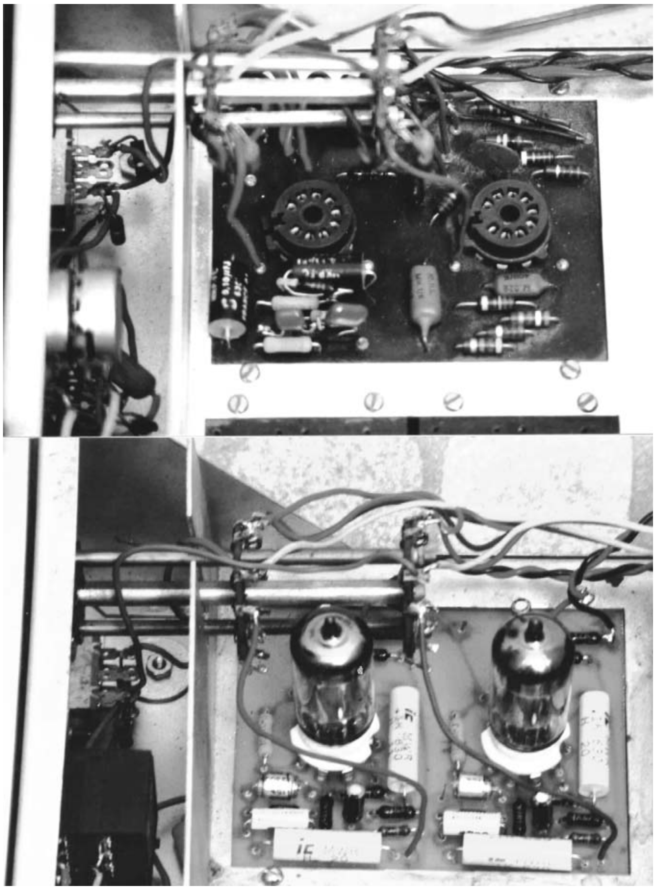

PHOTO 3: Phono preamp modification. Top of photo shows modification partially installed on original phono board. Bottom of photo shows same perspective with new phono board installed. The front of the amplifier is to the left of the photo.

network, which improves accuracy to ±0.2dB. Third, the output capacitor is decreased from 220nF to 100nF to save space. And fourth, I substituted a resistor for the original capacitor across the output to prevent switching transients. Nominal input impedance is 47kΩ in parallel with 110pF.

The major limitation of this circuit is the high output impedance of 32kΩ. This is not a problem in an integrated amplifier where it needs to drive only a few inches of wire; however, any additional capacitive load such as a cable connected to the tape out jacks may result in a high-frequency rolloff. Also, because of its high output impedance, this circuit needs to see a high load impedance—the RCA manual recommends a minimum load of 225kΩ. Thus, for the best sound nothing should be connected to the tape output jacks when playing LPs. The other potential problem, as with all non-feedback circuits, is channel imbalance due to variations in tube gain between channels.

There are two ways to build this circuit. Using the original circuit board, the new C2-R4 series combination goes where the old C2 and C7 were located, and the new C3-C4-R5-R6 network replaces R6 and R18. Making these networks requires a low-wattage soldering iron, use of small heatsinks such as Radio Shack sells to protect the component parts, and some skill with soldering.

The original R5 and R8 cathode resistors are replaced with a combination of 2.7kΩ resistor and cathode bypass cap in parallel. You can do this by soldering the capacitor in place leaving the leads sticking through, then soldering the resistor across the leads on the bottom side of the board. Make sure that the negative side of the capacitor is toward ground.

R4/R16 and C1/C6 are jumpered and R3/R15, R9-11/R21-23, and C4/C9 are deleted. The top half of *Photo 2* shows the process halfway through, with the front channel converted while the other channel remains stock. This was an early version with the cathode bypass caps on the bottom rather than the top as I now recommend.

The other way is to use a new circuit board (*Fig. 2*, component locations in *Fig. 3*) as shown in the bottom half of *Photo 2*. One mounting hole is close to the ground track, and you should use a nylon washer or insulating screw and nut to prevent inadvertent contact between the ground track and chassis.

The phono input jacks are connected directly to the phono board, and the phono outputs are connected to the selector switch where the wires from eyelets 11 and 14 were connected before. Remove all the components soldered to the selector switch, as well as all other wiring from the original PC-11 board to the selector switch. You should also remove the packaged electronic circuits (PECs) soldered to the input jacks.

Now, either tape head or phono will give the phono output. You could substitute a better-quality switch; however, the stock switch has the benefit of disconnecting the tape output when you select tape input, thus preventing a potentially damaging feedback loop where the tape output is fed back to the tape input if the tape machine is set to record.

LINE SECTION

The line/tone control section is a completely passive design, with about 17dB fixed loss in the midrange. In theory, passive tone controls have the potential for the best sound. Unfortunately, these controls are implemented using PECs, which look like large multi-legged ceramic capacitors, and probably sound like it.

In addition, the frequency response of the line section varies somewhat depending on the setting of the volume control. To see this, set the tone controls to flat, and feed a 1kHz square wave into one of the line inputs. Then connect an oscilloscope to the output and watch the square wave change shape as you adjust the volume. In fact, the line section is not as flat as the new phono section, which is a rather dubious distinction.

Simply bypassing the tone controls with a wire and a bypass switch will boost the volume by 17dB. If you flick this switch from tone controls on to bypass while music is playing, you will get an immediate blast of sound. So, with the controls bypassed the volume must be cranked way down, into an area where channel tracking problems are likely to be most prominent, and small changes in the setting produce large changes in volume.

Alternatively, you can hard-wire a bypass directly from the volume control output to the balance control input and skip the tone controls altogether. Then, you can substitute the amp boards from the Dynaco ST-35 power amplifier, which are a drop-in fit, have about 17dB less gain, and reportedly sound better. For the audiophile, this is probably the way to go.

For a music lover, I prefer to retain the tone controls because they can be useful in improving the sound quality of marginal material, but those PECs must go. The schematic of the tone control PEC is shown in *Fig. 4*. I built discrete versions using ¼W metal film resistors and Panasonic 2% polypropylene capacitors, for better channel matching and improved sound quality (*Photo 3*). I'm not sure that the 27pF cap in the original circuit is necessary, so I left it out of my version. I made up the non-standard capacitor values in the tone control network by paralleling caps—120pF plus 180pF for the 300pF value, and 2.2nF plus 2.7nF or 1.8nF plus 3.3nF for a 5nF value.

To install the new circuits, unscrew the front panel from the chassis bottom to get more working room. I did the bottom PEC first, cutting each wire right at the PEC and immediately soldering it to the corresponding wire on the discrete version to prevent mixing up the connections. Do not connect wire number 7 of the tone control modules at this time—you will connect them to the bypass switch later.

For flattest frequency response you must add a tone control bypass switch. A 4PDT slide switch will disconnect the tone control modules at both the input and output—a suitable one is available from Mouser Electronics. Because I believe the loudness switch is useless, I replaced this with the bypass switch. The schematic is shown in *Fig. 5*. Use color-coded wiring to avoid crossing channels.

The 43kΩ and 220kΩ resistors reduce the gain by 17dB in the bypass position. Be sure to remove the two wires connecting the middle terminal of the bass control pots to the balance control pots. These wires are replaced by wires going from wire 4 of the tone control modules to the bypass switch, and from the bypass switch to the balance control.

You can set the tone controls for flattest response in two ways. The simplest is to follow the instructions in the Dynaco manual. The most accurate method is to feed a square wave into one of the line inputs and adjust the tone controls for the best square wave at the power amplifier input. You can do this without turning the amplifier on. I used a splitter to feed the signal into both channels and a dual channel oscilloscope to monitor both channels simultaneously. Since the shape varies with the volume control setting, set the volume control where it is most likely to be used, say around 12 o'clock.

You will probably find that you cannot get a perfect square wave in both channels at the same setting, due to variations between the two sections of the tone control pots. The best I could do was to have the square wave tilted up a bit in one channel and down a bit in the other channel.

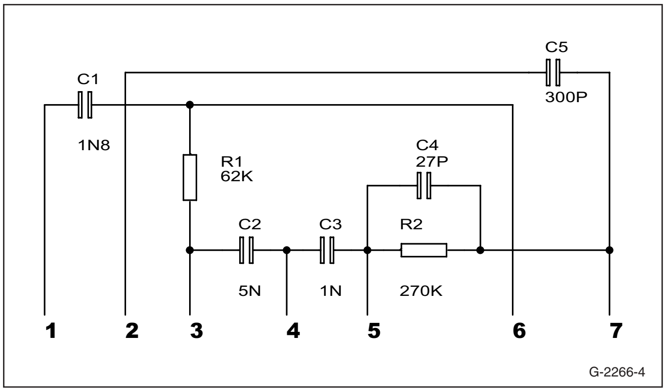

FIGURE 4: Schematic of tone control PEC.

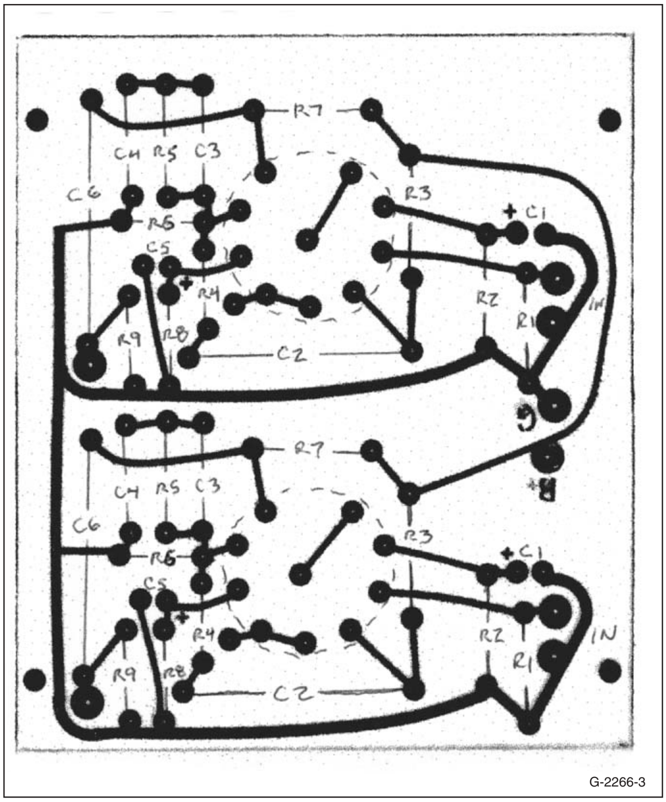

FIGURE 3: Component placement for new PC-11.

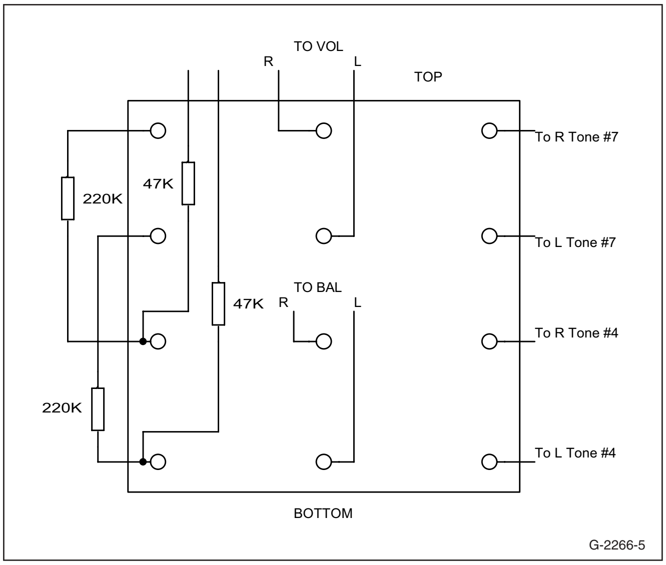

FIGURE 5: Schematic and wiring diagram for tone control bypass switch. The wiring diagram is shown looking at the back side of the 4PDT slide switch.

Next, I modified the filter, which combines a scratch/hiss filter, formed by the resistor capacitor combination of R25 and C13 or R26 and C14, and a rumble filter formed by the combination of C11 or C12 and the volume control. A hiss filter is still useful when playing old recordings, but if your turntable is producing audible rumble, it needs to be repaired. I eliminated the rumble filter by moving the two wires from the right-hand terminals of the filter switch to the center terminals. You can also replace the remaining low-pass filter capacitors with better-quality polystyrene or mica capacitors. The low-pass filter is –3dB at 5kHz with a slope of 6dB/octave.

I also substituted an Alps dual 500kΩ volume control for better channel tracking than the original control. Finally, while you have the controls exposed, it's a good idea to give all the pots and switch contacts a little squirt of Caig DeoxIT® and work them back and forth a few times to clean them off.

POWER AMPLIFIER

The stock power amp boards have a gain of 43dB to compensate for the loss in the passive line section. If you have permanently bypassed the tone controls, your best bet is to substitute the amplifier boards from an ST-35, which have only about 27dB gain. Unfortunately, the ST-35 is something of a collector's item; however, you can find new ST-35 boards on eBay or on the Internet.

The original phenolic board can become brittle with age, and generally has a darkened, burnt look. To be on the safe side, I made replacement boards (*Fig. 6*). Component placement and values are shown in the Dynaco manual. If you prefer not to make your own, I have seen replacement boards on eBay or on the Internet. I also recommend replacing the capacitors and resistors with higher-quality parts.

Unless you have equipment that is DC coupled, you should bypass the input capacitor C17, which is a ceramic cap. If you're not sure, you should at least substitute a film cap.

The original output stage was unusual in that all four EL-84 output tube cathodes were tied together to a common cathode-biasing resistor and bypass capacitor. The Dynaco manual suggests that this improves the output stage, but what this improvement is, other than saving the cost of one bias resistor and capacitor, is hard to see. It is relatively simple to split the two channels apart and provide each one with its own bias resistor and capacitor—a 190-200Ω/3W resistor plus 100µF or larger bypass capacitor will work fine. As mentioned previously, Stoke's power supply board accommodates this modification on the board.

The cathode resistor also provides a positive voltage for the preamp tube heaters, via the hum control pots on the rear of the amplifier. This voltage prevents hum due to heater-cathode leakage. With the shared cathode resistor now divided into two, you can either run a wire from each hum pot separately to each cathode resistor, or take the positive voltage for both off either one of the cathode resistors—since there is no current draw for this bias it shouldn't make any difference.

HARDWARE UPGRADES

I replaced the crummy-looking input and tape out jacks. My sample was factory built, and removing the rivets was a real pain. I used a combination of drilling, grinding with a Dremel tool, and finally a vise-grip to squeeze the inside end small enough to pass through the hole in the chassis. I broke the original jack boards into several pieces and removed them to give enough room for the vise-grip to grab hold of the inside end.

I mounted the new input jacks on a 5¼″ × 2″ piece of photo-sensitive copper-clad PC board, which served as a ground plane. I used the chassis as a template for locating the mounting holes, then drilled holes for five pairs of jacks—one set of phono inputs, three sets of line inputs and the tape out jacks. I removed the photo-protective film only over the phono input holes, exposed it to light, and etched away the copper around the phono jacks, then I removed the remainder of the photo resist and coated the bare copper with liquid tin solution.

I mounted all the jacks with the ground plane facing inside (*Photo 5*). All the jacks were connected to the ground plane except for the phono inputs, which were isolated as per the original design. Note that the phono inputs will need their ground terminal washers since the jack grounds are connected to the phono board.

I soldered all the wires to the board before mounting it with 4-40 ¼″ screws, nuts, and lock washers. Finally, I replaced the original rubber feet with EAR MF-1010 feet, which are made from Isodamp compound for isolation. They are somewhat higher than the original feet, which should help a little with ventilation through the chassis bottom.

PARTS SELECTION

Transformers aside, Dynaco's philosophy of parts selection could be summed up as the cheapest part that will do the job. This makes a lot of sense in terms of bang for the buck—spend the most money where it will do the most good: transformers and circuit design. The fact that Dynaco equipment sounds good in stock form is evidence that their engineering philosophy works.

Still, you may make gains by improving parts quality. I recommend carbon film or metal film resistors, with metal films for the RIAA and amplifier feedback networks for closer tolerance and best stability. Although metal films are technically superior, some prefer the sound of carbon films, reporting a warmer, less analytical sound. I used Illinois metallized polyester capacitors for coupling caps, based on reports by vintage amp specialists that these caps sound good, and are inexpensive.

I replaced small-value ceramic capacitors with similar-value silver mica caps. For RIAA capacitors, either polystyrene caps or Panasonic 50V 2% polypropylene caps from Digi-Key are compact enough to fit the cramped space available on the circuit board. More exotic caps are generally too big to fit. I wouldn't go overboard with expensive exotic parts, since you are still retaining the original switches and other hardware.

If the tubes in your set still provide correct voltages, I would leave them alone—this is probably the best test of proper function. In terms of replacements, I tend to prefer new old stock (NOS) small signal tubes for the best combination of long life, reliability, and sonic quality. For example, Telefunken 12AX7/ECC83s are famous for having a reported life span of over 100,000 hours! Power tubes are a trickier choice because of their shorter life span and the expense of NOS tubes—some of the new manufactured EL-84 tubes are reported to have excellent sound and cost less than NOS.

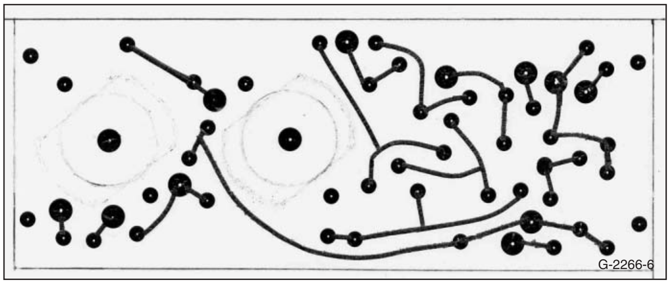

FIGURE 6: PC-10 circuit board.

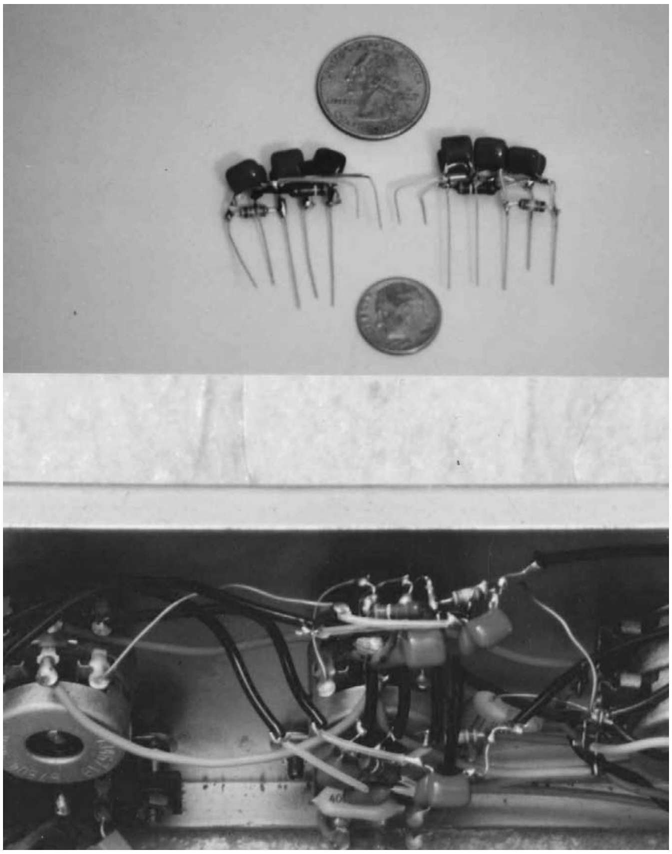

PHOTO 4: Top: Discrete tone control PECs, top and bottom views, dime and quarter shown for size comparison. Bottom: PECs installed.

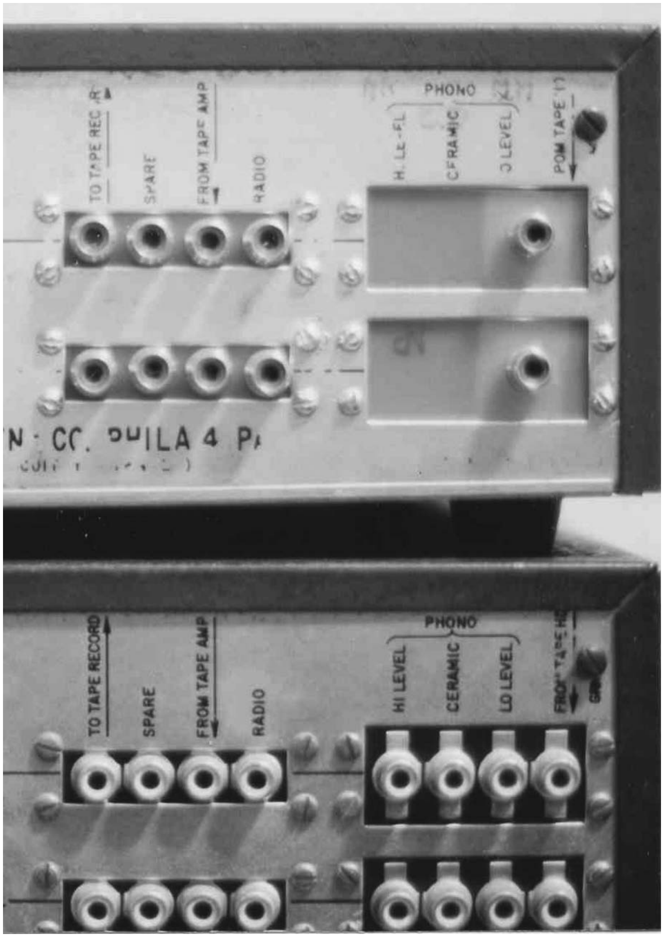

PHOTO 5: New input jack board mounted in place, top, with unmodified set on bottom.

CHECKOUT

I always feel a bit of anxiety when first turning on a piece of home-built or modified equipment. I recommend powering up using a Variac®, watching for smoke, sparks, and burning smells. If you use a Variac, you may also wish to substitute a 1A fast blow fuse for the regular 2A slow blow fuse. The gradually increasing voltage of the Variac prevents the fast blow fuse from blowing. These ratings are for US units with 120V; with the standard European 230V the regular fuse should be 1A slow blow, and ½A fast blow.

Assuming everything looks OK, the next step is to check circuit voltages using a meter with an input resistance of 10MΩ or so. For the power amp boards, use the voltages in the Dynaco manual. For the preamp, the power supply voltage should be around 260V, plate voltages at pins 1 and 6 should be around 195V, and cathode voltages at pins 3 and 8 around 1.3V, give or take 10%.

For safety I recommend the negative test lead of your voltmeter be clipped to the chassis, and keep one hand in your pocket while testing voltages with the positive lead in the other hand. Power down, replace the fast blow fuses with the appropriate slow blow fuse and plug it into your system. Finally, adjust the hum controls as outlined in the Dynaco manual.

MEASUREMENTS

I measured the frequency response of the new phono section using unselected tubes with my Old Colony inverse RIAA kit, modified Heathkit IG-18[8], and digital multimeter. The response was within ±0.2dB between 100Hz and 15kHz, which is close to my instrumentation accuracy limits, rolling off to −0.5dB at 50Hz and −1.2dB at 20Hz. The low-frequency rolloff is due to measuring it with the tone controls bypassed and the volume full on, which loads down the circuit. At normal volume control settings, the low frequencies are flat into the infrasonic region. By comparison, a stock phono section showed a ±2dB variation between 30Hz and 15kHz, with frequencies above 2kHz depressed by about 3–4dB. Input impedance was 47kΩ and input capacitance measured about 130pF using a BAS impedance bridge[9].

Testing the line section only, frequency response was about ±1dB between 20Hz and 20kHz with the tone controls set to flat; however, this varied somewhat with the volume control setting. With the tone controls bypassed, the line section passed 10Hz–5kHz square waves with little or no visible change, indicating flat response within the audible range, and was about 1dB down at 70kHz.

SOUND

I tried the SCA-35 playing into a pair of classic Spendor BC-1 speakers. The phono stage had a bit of audible hum close to the speakers, but this was pretty much inaudible from the listening position. Of course, it doesn't play really loud, but otherwise I thought it sounded remarkably good, especially considering the cost. This combination put out some very smooth, natural sound. Even today, it still fulfills its original design objective as a high-quality low-cost integrated amplifier for the music lover. ❖

TABLE 1

PARTS LIST	
Phono Preamp (1 channel)	
Capacitors	
C1, C5	25µF/25V
C2, C6	100nF/450V
C3	3n3F 2%
C4	10nF 2%
RESISTORS	
R1	47k
R2, R8	2.7k
R3, R7	100k/1W
R4	470k or 475k 1%
R5	24k or 24k3 1%
R6	680k or 681k 1%
LINE SECTION	
Capacitors (2% Panasonic polypropylene)	
300 pF = 120 + 180	
1nF	
1n8F	
5nF = 2n2 + 2n7 or 1n8 + 3n3	
RESISTORS (¼W)	
43k	
62k	
220k	
270k	
BYPASS SWITCH	
4PDT slide switch – CW	
Volume Control	
250k-500k dual audio taper	
Amplifier	
Capacitors	
2 – 12pF	
2 – 100nF/50V or jumper	
6 – 100nF/450V	
2 – 100µF/25V	
RESISTORS	
2 – 190/3W	
2 – 560 1%	
2 – 22k/1W	
2 – 27k/1W	
2 – 82k 1%	
2 – 120k	
2 – 270k/1W	
4 – 470k	
2 – 4.7M	
POWER SUPPLY	
IEC socket/fuse holder	
2A/120Ω current surge limiter	
2- soft recovery diodes, 500mA/1000 PIV or higher	

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Product Review

Selectronic Triphon II Crossover and Quad Amplifier

By Charles Hansen and Richard and Betty Jane Honeycutt

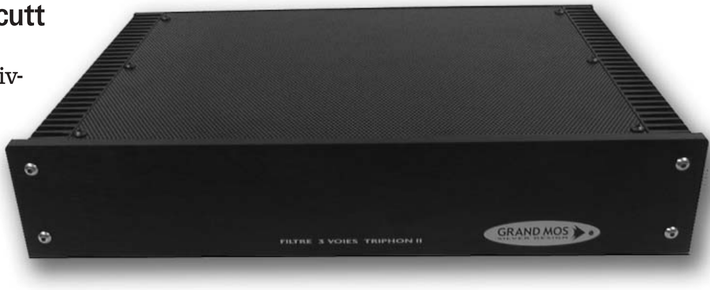

PHOTO 1: Crossover front view.

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84/86 rue de Cambrai
59022 Lille Cedex, France
Phone (33) 328.550.328
Fax (33) 328.550.329
www.selectronic.fr
Dimensions, both units: (W × D × H) 452 × 330 × 95mm (17¾″ × 13″ × 3¾″)
Crossover weight: 10.1kg (22.3 lb)
Quad amplifier weight: 12.3kg (27.1 lb)

The Selectronic Triphon II consists of two stackable units: a three-band active crossover (CO) and a quad power amplifier rated at 16W per channel at 8Ω. The units are designed primarily to drive a speaker system in which the high, mid, and low frequency drivers are directly accessible for tri-wiring. The Quad provides four amplifier channels for the high and mid drivers, and another stereo amplifier provides power to the subwoofers. Selectronic recommends the GRAND MOS for this purpose (see text for details).

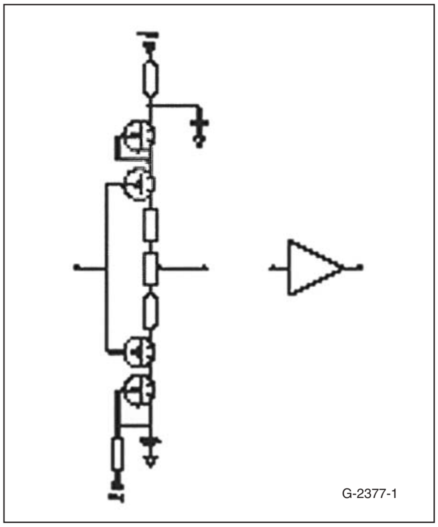


FIGURE 1: Triphon active filter simplified schematic.

Both factory assembled and kit versions of the crossover and amplifier are available. Each kit includes all components, hardware, and a pre-drilled and finished chassis.

The review samples of the Triphon II units were pre-assembled, but the kit assembly guides were included in the documentation, using the same format as the GRAND MOS (reviewed in Sept. '04 *aX*).

Photo 1 shows the front panel of the crossover, which has only a cool blue LED indicator inside the nameplate. The front panel is 9mm thick black anodized aluminum and, except for the lower height of the unit, is similar in construction to the Selectronic GRAND MOS stereo power amplifier (Sept. '04 *aX*).

The back and bottom of the crossover are constructed of 2mm black-painted steel. While not necessary for the low power dissipation in the crossover (no power devices are attached to them), heavy finned aluminum heatsinks are used along each side, and the 1.5mm steel top is perforated to enhance cooling. The heatsink fins have no sharp edges to cut your hands as you move the crossover about. The unit is very stiff and rugged, even with the top removed.

The rear panel (*Photo 2*) has an IEC power receptacle with integral RF line filter. The unit is furnished with a power cord. The power transformer primaries are factory-wired for 120V mains. The third pin of the AC receptacle is connected to the chassis.

Audio input and output signals pass through high-quality silver-plated Teflon™ insulated RCA jacks. Three 12V trigger inputs are located to the right of the IEC receptacle. There is no power switch, so the Triphon is energized whenever the power cord is plugged in.

Photo 3 shows the crossover with the cover removed. The Triphon is a dual-mono input design. Two 30VA Huiran R-type power transformers[1] occupy the

PHOTO 2: Crossover rear view.

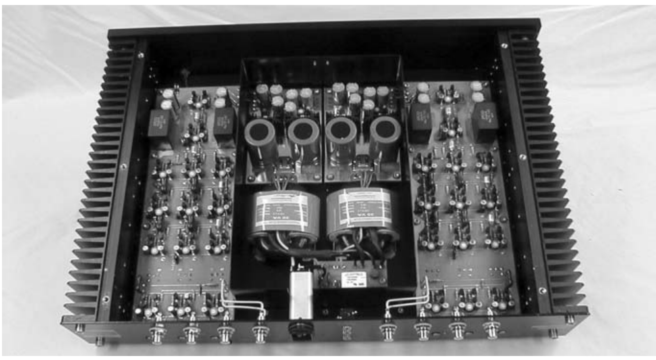


PHOTO 3: Crossover interior view.

center of the chassis. Very little interconnecting wiring is used. The AC line is connected from the line filter output to a small PC board that has MOV surge suppressors, a self-resetting polymer fuse, and a 12V DC relay that supplies AC power to the power transformers. Any one of the three 12V trigger inputs can also operate this relay.

The two identical power supply PC boards each use two 10,000µF reservoir capacitors. The power transformer secondary windings and the DC power to the filter boards are hard-wired. Heatsunk LM317 and LM337 IC regulators provide ±24V DC power supply rails. The power supply transformers use a faraday shield between the primary and secondary windings to reduce the interwinding coupling capacitance.

All resistors are 1% or better metal film. When ordering, you must specify the cutoff frequencies chosen for your system (Selectronic suggests you use the speaker manufacturer's recommended frequencies). They will supply the precision resistors calculated for your chosen frequencies. I assume this applies to the kit version as well.

A level control is provided for each of the three filter outputs, which are soldered to the bottom of the active filter PC boards so they project through the rear panel just below their respective output jacks. The slotted shafts of the pots have no knobs, so adjustment will require a screwdriver.

Connections between the rear panel jacks and the crossover circuit cards are made with Teflon®-covered solid wire. The factory wiring is very neat and should be easy for the constructor to duplicate in the kit version.

TOPOLOGY, CROSSOVER

The simplified schematic in *Fig. 1* was included with the literature, and is available from the Selectronic website. The crossover is a true dual-mono input design, sharing only the AC line input between channels. The active filters are electrically floated from the chassis. The phono jack shells for each channel are connected together at each active filter PC board, but the Triphon does not share a common ground between the two channels.

Each amplifier symbol in the left side of *Fig. 1* is comprised of a discrete Class-A totem pole JFET unity-gain buffer amplifier as shown on the right side of the schematic. Matched constant-current JFETs are used in series with each matched pair of JFET source followers. The input signal is loaded with a 15k resistor and then connected to three input buffer amplifiers. The three filter stages are DC-coupled except for the high-pass filter stages of the HP and BP sections.

The high-pass (HP) filter (sorte aigue) has two active filter stages in order to jumper-select either the 6dB or 12dB performance. If the 12dB slope is selected, it becomes a Linkwitz-Riley filter configuration. The selected filter drives a 10k level control on the rear of the unit (just below the HP output jack) followed by an output buffer stage.

The band-pass (BP) filter (sorte medium) consists of a pair of low-pass (LP) and HP stages, jumper-selectable for 6dB or 12dB performance. If so ordered, it can be configured as just a HP filter

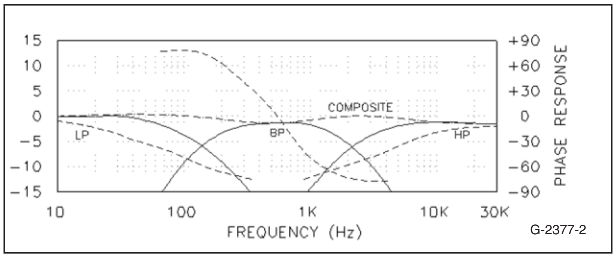


FIGURE 2: CO output and phase response vs. frequency

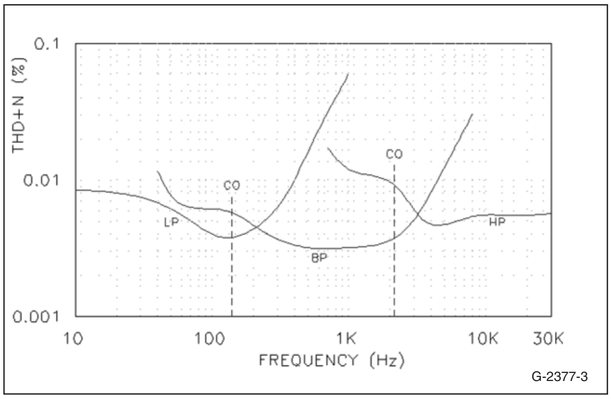


FIGURE 3: CO THD+N vs. frequency.

by interchanging the resistor and capacitor positions of the LP sections of the BP filter. The BP filter drives its own 10k level pot and output buffer stage. The low-pass filter (sorte grave) also has a pair of jumper-selectable 6dB or 12dB filter sections followed by a 10k level pot and a buffer amplifier stage.

MEASUREMENTS, CROSSOVER

I operated the Triphon II Electronic Crossover (CO) with pink noise at 2V RMS and with all six output channels loaded with 15kΩ for 1 hour, after which the unit was cool to the touch. The distortion was essentially the same for each channel before and after the 1-hour run-in. Data for the left channel is presented here and summarized in *Table 3*. The CO was quiet, and there was no noise using the Quad amp during power-up or shutdown.

The Triphon II does not invert polarity. Input impedance was 15.4k for both channels. The output impedance at the peak response frequencies of each filter measured a suitably low 242Ω.

The frequency and phase response for the Triphon crossover as supplied from the factory is shown in *Fig. 2* with a 15k load. The three solid lines are the frequency response for each output (LP, BP, HP). The three dashed lines represent the phase response, and the dash-dot line is the composite gain of the three sections as "perfect" drivers would reproduce them. The combined response results in an acoustic response flat to +0.3/–1.4dB. I truncated the frequency at 30kHz since the HP section response was still not down –1dB (ref 20kHz) even at 500kHz.

The gain at the two crossover points (15k load) with all level controls at maximum was –5.9dB at the conjunction of the LP/BP filters, and –6.0dB at the conjunction of the BP/HP filters. The peak response of each section is –0.4dB at 10Hz (LP), –1.4dB at 550Hz (BP), and –1.2dB at 9.9kHz (HP). This is the result of using passive RC components at the JFET buffer inputs, which have a bit less than unity gain.

Level control tracking was very good at the half-volume position. There was less than 1dB difference between the six CO level controls at this point. When adjusted facing the rear of the CO, the controls were reversed. Counterclockwise (CCW) rotation increased the gain. I suppose this was done to make adjustments from the front of the Triphon II stack intuitive, but it also requires placing the CO on top of the hot-running Quad amplifier. As I mentioned earlier, there are no knobs on the level controls, so finding the screwdriver slot may be difficult, but frequent level adjustments should not be necessary once the CO is set up with your chosen speakers.

The crosstalk performance was first rate. The dual-mono design assures that minimum coupling occurs between channels through the power supply rails (*Table 1*).

THD+N vs. frequency is shown in *Fig. 3* for the three filter sections. During distortion testing, I engaged the test set 80kHz low-pass filter to limit the out-of-band noise. The two vertical dashed lines are the –6dB conjunction frequencies of the LP/BP and BP/HP filters, respectively.

I fixed the input voltage at 2V RMS with a 15k load on each filter. I expected the THD+N to be higher for the HP section since it has a much wider bandwidth over which noise can exert its effect, but all three sections showed excellent performance until the gain fell

PHOTO 4: Amplifier front view.

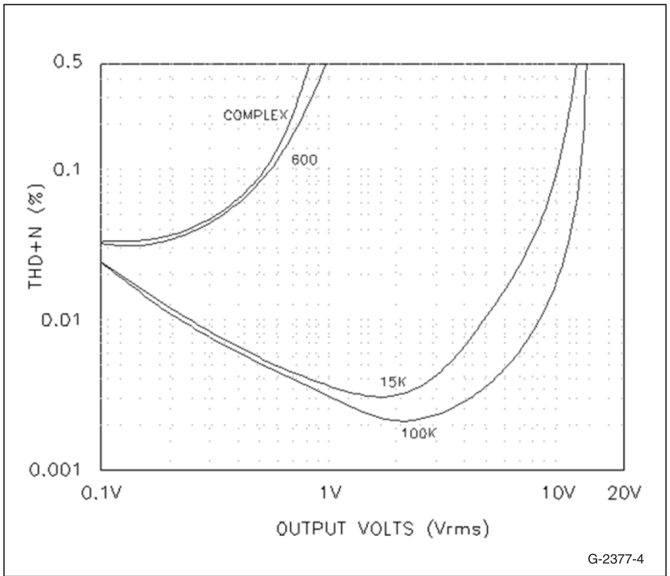


FIGURE 4: CO THD+N vs. output voltage.

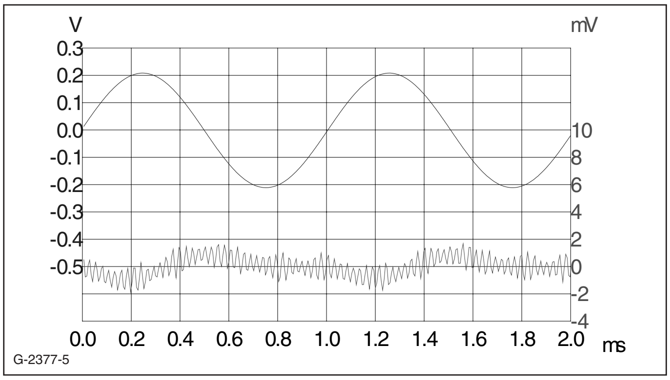


FIGURE 5: Residual 1kHz distortion, BP section.

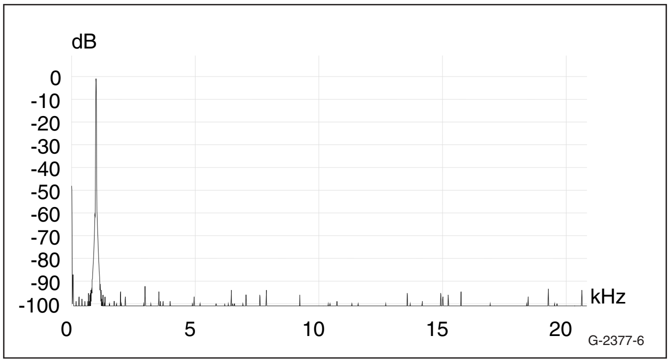


FIGURE 6: CO spectrum of 1kHz sine wave, BP section.

off below –15dB. I did not increase the input voltage to maintain a constant output level, since I believe the results in *Fig. 3* are more representative of the real-world application.

Figure 4 shows THD+N vs. output voltage for the BP filter section at 1kHz, with the loads shown in the graph. I again engaged the test set 80kHz low-pass filter to limit the out-of-band noise. Distortion is relatively flat with frequency, but is very sensitive to the load impedance. The complex load (575Ω paralleled with 1nF), whose capacitive reactance decreases with frequency, only presented a bit more difficulty than the 600Ω resistive load.

Note the distortion minimum with 15k load occurs just about at the optimum 2V RMS point. Since the CO is designed to be used with Selectronic amplifiers, and all have a 15k input impedance, distortion performance of the CO should be excellent.

My signal generators are limited to about 7.5V RMS, and at 15k load the CO clipping exceeds this level. In order to determine the 1% THD+N clipping points, I drove the Triphon CO from the GRAND MOS amplifier's speaker output (using a suitably large series resistor to prevent any possibility of damage) at 1kHz with the same loads as *Fig. 4*. These are summarized in *Table 2*. The clipping waveform in each case was rounded, almost like a tube circuit, with the lower half of the waveform flattening out first (JFETs are transconductance devices like vacuum tubes).

While the lowest load impedance is more than the output impedance of the CO, I suspect the high distortion at low impedances is due to current starving in the constant-current JFETs in series with the buffer JFETs. I do not know the Class-A operating current in this CO design.

The BP section distortion waveform for 2V RMS into 15kΩ at 1kHz is shown in *Fig. 5*. The upper waveform is the amplifier output signal, and the lower waveform is the monitor output (after the THD test set notch filter), not to scale. This distortion residual signal shows only a low level of noise. THD+N at this point is a very low 0.0032%.

The BP section spectrum of a 1kHz sine wave at 2V RMS into 15kΩ is shown in *Fig. 6*, from zero to 20.8kHz. The THD+N is 0.0032%, with the 2nd and 3rd harmonics measuring below –93dB.

Figure 7 shows the output spectrum of the BP filter section reproducing a

TABLE 1
CO CROSSOVER CROSSTALK

FREQUENCY	R TO L	L TO R
LP, 100Hz	–100dB	–100dB
BP, 1kHz	–100dB	–100dB
HP, 10kHz	–98dB	–98dB
HP, 20kHz	–94dB	–94dB

TABLE 2
BP FILTER 1% THD+N OUTPUT VOLTAGE

LOAD	VOUT AT 1% THD+N
1kHz 100k load	14.7V RMS
1kHz 15k load	14.1V RMS
1kHz 600Ω load	1.05V RMS
1kHz complex load	1.0V RMS

PHOTO 5: Amplifier rear view.

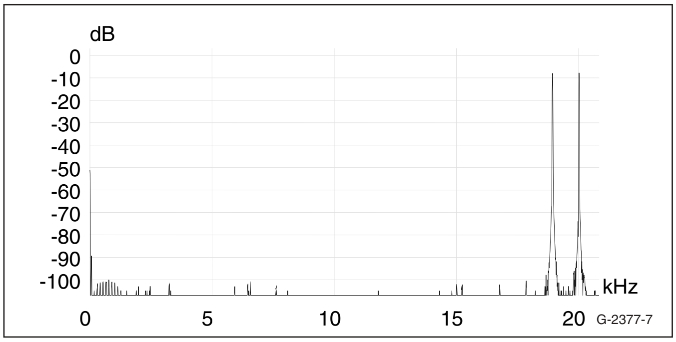


FIGURE 7: CO spectrum of 19kHz + 20kHz intermodulation signal.

combined 19kHz + 20kHz CCIF intermodulation distortion (IMD) signal at 12Vpp into 15k. The 1kHz and 18kHz products at −100dB (0.001%) are distinct, but at about the same level as the residual noise. Repeating the test with a multi-tone IMD signal (9kHz + 10.05kHz + 20kHz, not shown) resulted in a similar outstanding performance. The latter test gives a better indication of the amplifier's nonlinear response, since it is a closer approximation to music than a sine wave.

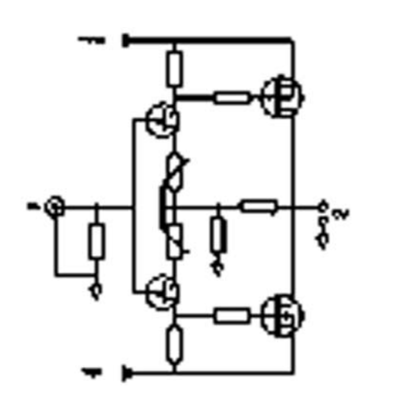


FIGURE 8: Quad amplifier simplified schematic.

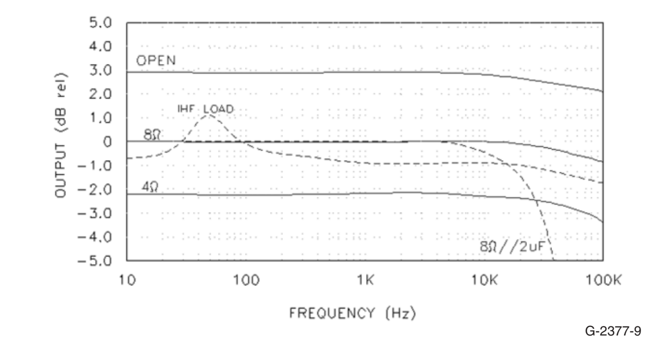


FIGURE 9: Quad amplifier frequency response.

I did not perform square-wave testing on the Triphon II CO, since its job is to selectively alter the frequency response. The CO met or bettered its limited number of specifications, as shown in *Table 3*.

INSIDE THE QUAD AMPLIFIER

The Quad 16W amplifier consists of four channels of pure Class-A amplification in the same enclosure, designed for amplification of the high- and mid-frequency filter outputs of the companion Triphon crossover. *Photo 4* shows the front panel, which has only a cool blue LED indicator inside the nameplate. The chassis construction of the Quad amplifier is identical to that of the Triphon crossover.

The power amplifier output MOSFETs on each of the amplifier PC boards are mounted to the finned aluminum heatsinks used along each side of the amplifier.

The rear panel (*Photo 5*) has an IEC power receptacle with integral RF line filter. The unit is furnished with a power cord. The power transformer primaries are factory-wired for 120V mains. The third pin of the AC receptacle is connected to the chassis. Three 12V trigger inputs are located to the right of the IEC receptacle. There is no power switch, so the Triphon is energized whenever the power cord is plugged in.

TABLE 3
CROSSOVER MANUFACTURER'S RATINGS AND MEASURED RESULTS

PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Frequency Response:	2kHz – 10MHz ±1dB	LP 2Hz – 45Hz ±1dB BP 275Hz – 1.05kHz ± 1dB HP 5.5kHz – >500kHz ±1dB
Crossover Frequencies:	buyer specified	140Hz LP-BP, 2.18kHz BP-HP
Total Harmonic Distortion:	"non measurable"	<0.0085% at response peaks, 2V RMS
IMD – CCIF (19+20kHz):	N/S	0.001% CCIF, 12Vpp
MIM (9+10.05+20kHz):		0.001% MIM, 12Vpp
Insertion Gain:		+0/–1.4dB combined LP, BP, HP
Filter Slopes:	6 or 12dB/Octave (jumper selected)	6dB
Signal to Noise Ratio:	Better than 110dB	113dB A-weighting
Input Impedance:	15k	15.4k
Output Impedance:	N/S	BP, 242Ω @ 1kHz
Input Voltage, Max:	14Vpp, 5V RMS	17.1V RMS in, 11%THD, 15k
Output Voltage, Max:	N/X	14.7V RMS out, 1%THD, 15k

Audio signals are input to four high-quality silver-plated Teflon™ insulated RCA input jacks. Four pairs of high-quality silver-plated Neutrik Speakon binding posts provide the connections for the speakers. These binding posts, in accordance with EU requirements, are not on US 0.75″ spacings, so you cannot use dual banana plugs.

Photo 6 shows the amplifier with the cover removed. Two 120VA Huiran R-type power transformers[1] occupy the front of the chassis. Very little interconnecting wiring is used. The AC line is connected from the line filter output through heavy tracks in each power supply board to a small PC board between the transformers. This board has MOV surge suppressors, a self-resetting polymer fuse, and a 12V DC relay that supplies AC power to the power transformers. Any one of the three 12V trigger inputs can also operate this relay.

The two epoxy power supply PC boards use silver-plated screw connections for the power transformer secondary windings. Eight 10,000µF reservoir caps are used on each board and four discrete BY239 diodes per board rectify the transformer output into ±27V DC power rails.

The four double-sided Teflon® amplifier PC boards are located along the sides of the unit. You can see the two output MOSFETs for each amplifier channel attached to the heatsink with hex socket screws and keratherm insulators.

Connections between the rear panel jacks and the amplifier circuit cards are made with large-gauge Teflon®-covered solid wire. Connections between the power supply boards and the amplifier circuit cards are made with silver-plated screw connections. The factory wiring is very neat and should be easy for the constructor to duplicate in the kit version.

TOPOLOGY, QUAD AMPLIFIER

The simplified schematic in *Fig. 8* was furnished as part of the literature and is available at the Selectronic website. The four amplifier channels are electrically floated from the chassis. The input jack shell and speaker negative post are connected together at each amplifier PC board, and each pair of amplifier channels on the same power supply (A and C, B and D) shares a common ground between channels. There is no common ground connection between these amplifier pairs, however.

The input stage of each of the four amplifiers comprises symmetrical single-ended JFET gain stages. Selectronic uses matched JFETs for best performance. The drains of the JFETs are cascaded to complementary-symmetrical source-follower output MOSFETs (types J162 and K1058), with Class-A quiescent bias current. Individual pots in the source of each JFET allow for adjustment of the bias.

DC feedback is taken from the speaker output to the common side of the bias adjust pots at the input JFETs. There are absolutely no series capacitors in the signal path, so the Quad (like its GRAND MOS big brother) can amplify any DC component in the audio input signal. There is no zobel network or se-

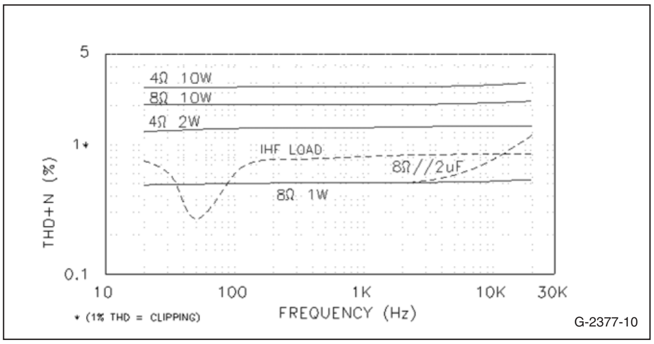


FIGURE 10: Quad THD+N vs. frequency.

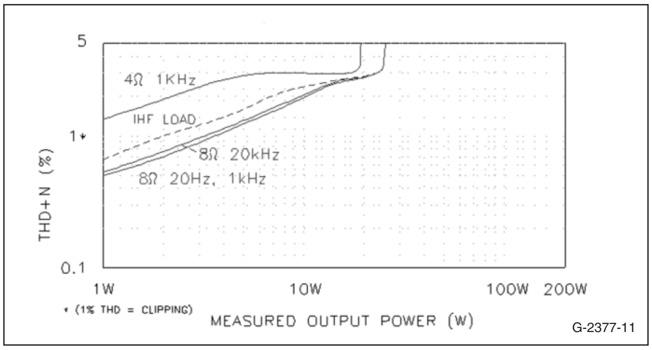


FIGURE 11: Quad THD+N vs. output power.

ries R-L stability network at the output.

The power supply transformers use a faraday shield between the primary and secondary windings to reduce the inter-winding coupling capacitance. Eight low ESR computer-type 10,000µF filter capacitors provide the energy reservoir for each channel pair.

Assuming the schematic is complete, I didn't see any DC servo-control or speaker protection circuitry in the Quad schematic, a characteristic it shares with the GRAND MOS. Since the amplifier is DC-coupled from input to output, this could place DC voltage on your speakers if a small DC offset in the preamp or Triphon crossover output were amplified by the 23dB gain of the amplifier. However, the limited output power of this amplifier may not be enough to permanently damage your speakers.

MEASUREMENTS, QUAD AMPLIFIER

I initially had problems with ultrasonic oscillation (220kHz) with the Quad amplifier when I tried to power any two of the channels using the same supply rails. I needed to isolate the AC line grounds at my signal sources to obtain stable operation. I saw a similar phenomenon with the GRAND MOS amplifier.

Due to a limitation in the number of speaker loads I have available, I operated channels A and C of the Quad amplifier at 2W into 8Ω for 1 hour. The heatsink temperature increased to 55°C, while the heatsink temperature of the two idling channels increased to 57°C. This is not unusual for a Class-A design.

The THD readings at the beginning of this run-in period were 0.65% for channel A and 0.64% for channel C. This decreased to 0.46% and 0.51%, respectively, after the run-in. I repeated the run-in test for channels B and D. The THD readings at the beginning of the run-in period were 0.55% for channel B and 0.43% for channel D. This decreased to 0.32% and 0.42%, respectively, after one hour. Accordingly, the distortion measurements for channel C (the highest) are presented in the data graphs.

There is a very low-level thump when starting up and no noise at all when shutting down the amplifier. Output hum and noise measured –102dBr (A-weighted, input shorted) and was inaudible with my ear against the speaker. I also measured between –10 and –58mV of DC offset across the four channels.

The Quad does not invert polarity. Input impedance measured 15.3k in

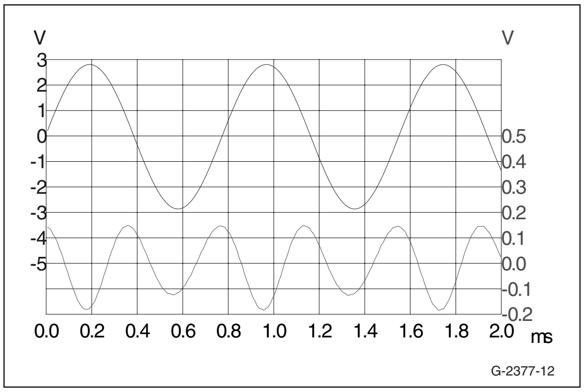


FIGURE 12: Quad residual distortion.

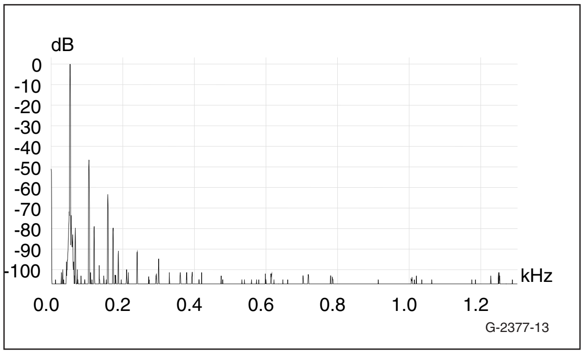


FIGURE 13: Quad spectrum of 50Hz sine wave.

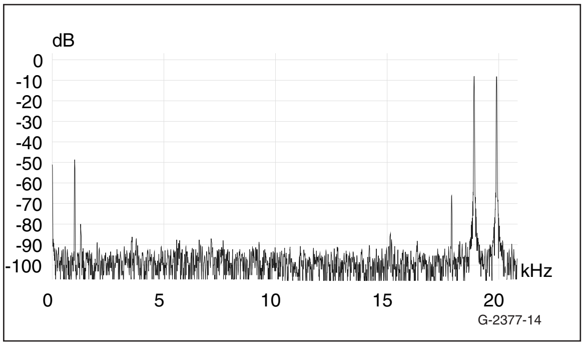


FIGURE 14: Quad spectrum of 19kHz + 20kHz intermodulation signal.

all four channels. The gain at 2.83V RMS output into an 8Ω load ran between 22.63dB and 23.08dB. For some reason, Selectronic designed the GRAND MOS with a much higher 31dB gain. The volume controls in the crossover will allow you to adjust for this gain difference. The output impedance at 1kHz for all four channels was a high 3.11Ω, increasing slightly to 3.22Ω at 20kHz.

The frequency response (*Fig. 9*) was within –1dB from DC to 106kHz, at an output of 2.83V RMS at 1kHz into 8Ω. It wasn't down to –3dB until 223kHz. The output dropped 2.2dB with a 4Ω load, and increased by 2.9dB with an open circuit. The response with a complex load of 8Ω paralleled with a 2µF cap (a test of compatibility with electrostatic speakers) equaled that of the 8Ω load alone to 4kHz, where it began a steep rolloff due to the fairly high output impedance. The IHF speaker load, which has an impedance peak at 50Hz, produced a significant 1.8dB total change in the frequency response. The Quad amplifier will be quite sensitive to any variations in speaker impedance with frequency.

I first measured the crosstalk between channels A and B, since they use two different power supplies. Then I measured the crosstalk between channels A and C, since they use a common power supply. The crosstalk between channels A and B is limited by the noise floor at low frequencies, increasing due to capacitive coupling at higher frequencies. Coupling through the common power supply seems to define the crosstalk performance from channel A to C (*Table 4*). Class-A amplifiers are more susceptible to power-supply-rejection limitations than are conventional Class-B amplifiers.

THD+N vs. frequency is shown in *Fig. 10* for the loads indicated in the graph. During distortion testing, I engaged the test set 80kHz low-pass filter to limit the out-of-band noise. Distortion is relatively flat with frequency, but very sensitive to the load impedance. This is especially noticeable with the IHF load, which has an impedance peak at 50Hz, and the complex load (8Ω paralleled with 2µF), whose capacitive reactance decreases with frequency.

Figure 11 shows THD+N vs. output power for the loads and frequencies shown in the graph. There was absolutely no strain right up to the point of maximum power. The amplifier, with channels A and C driven into 8Ω loads, reached its 1% clipping point at only 3.5W, well below its 16W rating.

While the Quad is not explicitly rated for 4Ω loads, I never saw less than 1.35% THD at any power output into 4Ω. The negative half-cycles clipped before the positive half-cycles. The heatsinks reached a maximum temperature of 58°C at 10.7W into 8Ω (2/3 power rating), while the two idling channels had a 57°C heatsink temperature.

The distortion residual waveform for 1W into 8Ω at 1kHz is shown in *Fig. 12*. The upper waveform is the amplifier output signal, and the lower waveform is the monitor output (after the THD test set notch filter), not to scale. This distortion residual signal shows mainly the 2nd harmonic. THD+N at this test point is 0.51%.

The spectrum of a 50Hz sine wave at 1W into 8Ω is shown in *Fig. 13*, from zero to 1.3kHz. The THD+N here measures 0.50%. The 2nd, 3rd, 4th, and 5th harmonics measure –47dB, –63dB, –106dB, and –92dB, respectively. The second harmonic alone is 0.45% of the total distortion, and undoubtedly will define the listening characteristics of the Quad. The fairly high levels of distortion measured in these tests contrast sharply with manufacturer's statement of "non measurable" distortion.

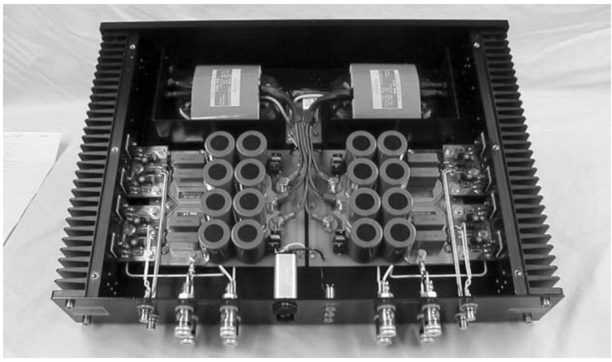


PHOTO 6: Amplifier interior view.

Low-level power-supply artifacts are also present at 60Hz, 120Hz, 180Hz, and 240Hz. The spectrum of a 1kHz sine wave (not shown) had a nearly identical distribution of harmonics.

Figure 14 shows the amplifier output spectrum reproducing a combined 19kHz + 20kHz CCIF intermodulation distortion (IMD) signal at 12Vpp into 8Ω. The 1kHz IMD product is –49dB (0.35%), and the 18kHz product is –65dB. Repeating the test with a multi-tone IMD signal (9kHz + 10.05kHz + 20kHz, shown in *Fig. 15*) resulted in a 950Hz product of –68dB and a 1050Hz product of –55dB. The nonlinearities that produce high levels of the 2nd harmonic also result in high levels of intermodulation distortion, as the two figures show.

A 2.5Vpp square wave at 40Hz into 8Ω produced a slight LF tilt that did not reveal itself in the frequency response test of this DC-coupled design. The 1kHz and 10kHz square waves were just about perfect. When I connected 2μF in parallel with the 8Ω load, there was absolutely no ringing and only a slight bit of additional rolloff at the leading edge of the 10kHz square wave.

TABLE 4
QUAD AMPLIFIER CROSSTALK

Frequency	A to B	A to C
100Hz	–102dB	–74dB
1kHz	–100dB	–68dB
10kHz	–92dB	–63dB
20kHz	–89dB	–60dB

The manufacturer recommended Triphon II setup, including the GRAND MOS amplifier, is shown in *Fig. 16*. There is no volume control (except for the level controls of the CO) in this setup. Volume needs to be adjusted at the source component or by means of an added preamp or passive line stage.

A comparison of the measured results and the manufacturer's ratings is shown in *Table 5*.

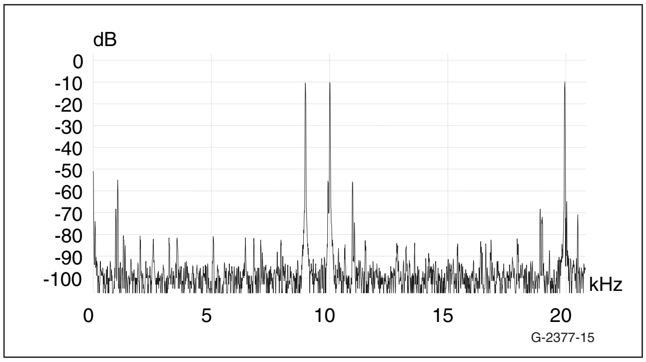


FIGURE 15: Quad spectrum of 9kHz + 10.05kHz + 20kHz intermodulation signal.

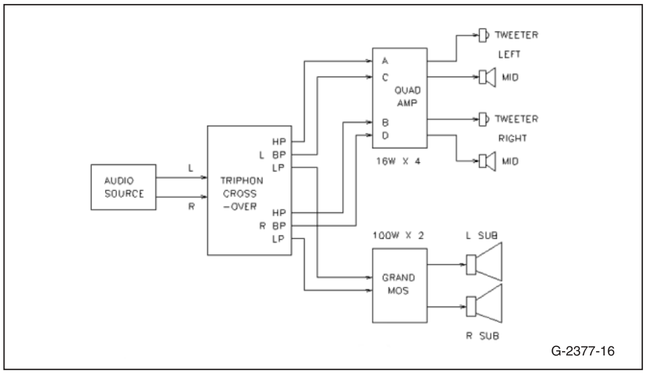


FIGURE 16: Recommended Triphon II setup.

LISTENING TESTS OF TRIPHON ELECTRONIC CROSSOVER

By Richard and Betty Jane Honeycutt

The Triphon electronic crossover is a sturdily built piece of equipment with a no-nonsense (some might say "industrial") appearance. The unit we tested was supplied with two power amplifiers built by Selectronic: a MOS 4, 4-channel, 16W/channel MOSFET unit, and a Grand MOS, 2-channel 100W/channel MOSFET model. The construction of all three items was impressively sturdy, and especially so the terminal posts. Those who have chosen the various manufacturers or "Beast" cables as their preferred charities will have no trouble inserting their cable ends into the Triphon terminal posts!

The Grand MOS amplifier has rack-mount-style handles on the rear! Its power switch is also on the rear. The MOS 4 has no power switch. However, all three pieces of equipment have blue power LEDs, which will at least make them stand apart from most other equipment you may have.

You do not expect many flashy controls on an active crossover—especially

TABLE 5
QUAD AMPLIFIER MANUFACTURER'S RATINGS AND MEASURED RESULTS

PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Power Output:	16W RMS (sic) 8Ω	3.5W 8Ω, 1% THD 23.4W 8Ω, 3% THD
Frequency Response, 1W, 8Ω	DC – 300kHz power bandwidth	DC – 247kHz ±3dB
Total Harmonic Distortion:	"non measureable"	2.5%, 16W 8Ω, 1kHz
IMD – CCIF (19+20kHz):	N/S	0.35% CCIF
MIM (9+10.05+20kHz):		0.18% MIM
Input Impedance:	15kΩ	15.3k
Signal to Noise Ratio:	N/S	–102dBr, A-weighting
Gain:	1V for 16W RMS (sic)	22.8dB
Output Impedance:	N/S	3.11Ω 1kHz

one that has fixed crossover points and slopes, as this one does. All you need in this case is a level control for each of the six outputs (it's a three-way crossover). Those on the Triphon are wisely located on the rear, presumably to discourage tampering by the inquisitive.

For some reason unknown to me, these controls operate in reverse: counterclockwise is louder. This is not really a problem, since you will normally set up the controls only once when you install the unit, unless you change speakers from time to time.

TEST PHILOSOPHY AND METHODOLOGY

Over the years, numerous very dedicated audiophile/researchers (many having ears high in gold content) have studied the problems of correlating the sound and the measurements of audio equipment. As part of this effort, there is near-universal agreement that listening tests that examine only one piece of equipment at a time are useful only in showing up fairly obvious problems. Meaningful subjective tests of good-to-excellent equipment require that the tests be conducted within some fairly strict protocols.

The first of these is not more than one piece of equipment be evaluated at a time. Otherwise, there is absolutely no way of knowing whether what the listener heard is an artifact of the equipment under test; of another part of the system; or of the listener's health, recent exposure to noise, or ingestion of ototoxic drugs such as aspirin, or—shall we say—others.

Second, switching between different pieces of equipment must be done in a way that does not allow the listener to know which piece of equipment (s)he is hearing. Tests have been conducted in which the brand markings on speaker systems were switched between sets of listeners, and the listeners consistently "preferred" the speakers bearing the well-known brand, even though they may have been made by a different company. It is also of paramount importance that the sound levels of the various signal chains under test be matched to within 0.1dB. Otherwise, listeners will very reliably prefer the louder of the two systems, even though not conscious of the actual level difference.

The reliability of the listening test increases with the number of listeners. Finally, researchers have consistently found that providing a specific list of descriptors with a well-defined rating scale is essential to meaningful results. Otherwise, you get responses from five different reviewers couched in what amount to five different personal languages. These reasons are why there are not many useful listening tests performed: meeting all these requirements is expensive.

Knowing all this, and still being restricted in the time and money that we could devote to this project, we devised a test setup using relays to switch between two systems, with the listener not knowing which system (s)he was hearing at any given time. We further set up the equipment beforehand using a graphic equalizer to

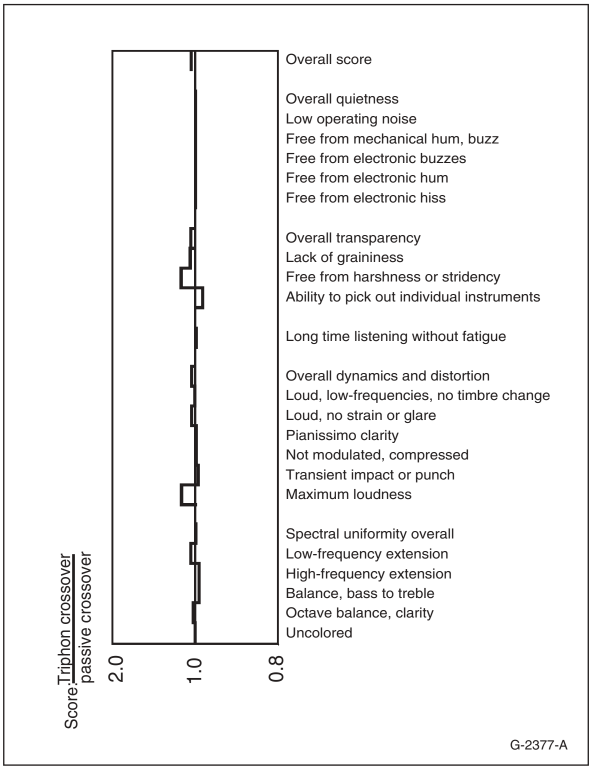


FIGURE A: Expanded vertical scale.

match the final response of the systems as closely as possible. Thus not only overall levels, but spectral balance (in octave bands) were matched.

The speakers were professional types, using a subwoofer flat from about 40Hz to 300Hz, a cone midrange speaker reasonably flat from 100 to 2500Hz (some 1 to 2dB ripples above 800Hz), and a horn/compression driver flat from 1200Hz to about 16kHz. The crossover points used were 150 and 2kHz.

SELECTRONIC KIT-BUILDING INSTRUCTIONS

The kit-building instructions for each of the Selectronic items were included in the documentation. I took a look at each one from the kit builder's point of view.

The GRAND MOS kit assembly guide, while only 11 pages long, is complete and includes color photos and drawings of the entire assembly process. You must build two power supply and two amplifier PC boards.

Each assembly step has a checkbox, and the components you need are listed on each page with the required assembly picture. Solder is not provided, but the manual recommends lead-free 96% tin-4% silver solder. All you need are a few metric tools (all hardware is metric) and a good digital multimeter to set the output stage bias. Temporary 1Ω fuse/shunts are included for this purpose.

The Selectronic Triphon II Quad 16W amplifier is a bit more involved, with 19 pages of drawings and color photos. There are seven PC boards to build (four amplifiers, two power supplies, and the mains relay board). The solid wire links have detailed bending dimension drawings, so wire routing is accounted for ahead of the final assembly stage.

As with the GRAND MOS, temporary 1Ω fuse/shunts are included in order to adjust the 200mA Class-A quiescent current. For remote control (what we call 12V trigger) operation, you must cut a link on the mains relay board so that any of the three external jacks can control the mains relay.

The Selectronic Triphon II Crossover Unit assembly guide has 16 pages and follows the same photo/drawing format of the others. There are five PC boards to build (two filters, two power supplies, and the mains relay board). The stranded wire lengths needed for the final assembly stage are specified in the instructions. Capacitor values for the three filter bands are fixed. The instructions give you the formulas to calculate the required resistor values for the cutoff frequencies that you select for your particular speakers.

If you want to add the recommended RF input filter, you need to cut a track on each filter board and install another RC circuit (I believe this should be standard, with drilled PC board tracks provided). As with the Quad amp, you must cut a link on the mains relay board for remote control of the mains relay. After assembly, you must adjust the four power supply board plots for ±24V DC and the DC offset voltage for each buffer (14 more pots).

The instructions show you how to configure the crossover sections for 6dB or 12dB/octave and how to make a two-way (HP/LP) or three-way (HP/BP/LP) configuration. This is done by means of small gold-plated computer-style jumpers. Finally, one page of instructions contains suggestions for adjusting the level of each crossover using either a sound-level meter (preferred) or FM tuner interstation white noise.

I recommend that you have prior kit-building experience for all these kits. This is not to say that building these kits is difficult or the instructions lacking. But there is none of the novice handholding (soldering instructions and so on) that was the hallmark of the Heathkit assembly manuals. If you have any difficulty, help is available from "your local dealer" and the Selectronic on-line support service.—CH

For comparison, we chose to use a passive crossover having the same slope (6dB/octave) and crossover points. We chose passive crossover for two reasons. First, we guess that most *aX* readers are more likely to be considering the purchase of an active crossover as an upgrade from a passive than as a replacement of another active unit. Second, if we had chosen to compare the Triphon to another active crossover, which one would we have used as the other unit? Such a comparison would be fair only in the case of a "shootout" among several different crossovers, conducted with many listeners and strict protocols.

In this listening test, we used the following selections from the *Hi-Fi News and Record Review* Test Disc III:

"Jerusalem"—Parry
"Henry V Extract"—Doyle
"Trumpet Concerto in C" —Vivaldi
"Welcome, Welcome"—Purcell
"Rio Napo RSS Demo"

Each listener was provided a separate scoring sheet for the tests; scores from 1 (worst) to 5 (best) were assigned for each descriptor, and results for each musical selection were placed in one of the five numbered columns. For each descriptor, there were thus five scores for each listener (one per musical selection), and since there were two listeners, that made ten scores for each descriptor.

The individual scores are not printed in this report, because we believed they might be misleading. For example, on selection 5, both listeners gave both crossovers very poor scores on "graininess," making it extremely likely that the musical selection, not the crossovers, were being jumped. Averaging the scores removes most of such artifacts.

Figure A shows the results of our tests, with the average score of both listeners charted above the relevant descriptor. The score is stated as a ratio of the average active-crossover score over the passive-crossover score for the two listeners, for all five musical selections. "No perceived difference" corresponds to a ratio of 1. A value greater than 1 indicates superiority of the passive crossover. Note that in order to show the small variations in preference, the graph's vertical axis only extends from 0.8 to 2.0, even though the preference scale extended from 0.1 to 5.0.

You will notice that under these controlled test conditions, there was very little perceived difference between the active and passive crossover. In fact, the overall average preference ratio was 1.03, which is not significantly different from "no preference." This is not surprising, because a well-designed crossover should be audibly transparent. Seemingly, both the active and the passive units performed quite well.

There are, of course, other reasons to prefer an active crossover besides the sound quality. Perhaps the most important is that since the *Q* of inductors in a passive crossover is always pretty low (seldom above 5), the maximum stopband rejection is less than optimal. This can affect distortion if you listen at very loud levels (which we did not, valuing our ears!) as well as the lifetime of tweeters, which can easily be damaged by low-frequency energy. Also, if you build your own speakers, an active crossover allows very easy and precise matching of levels among woofer, mid, and tweeter. As another example, the use of electrostatic mid/high speakers with electromagnetic woofers is made much easier by the use of an active crossover.

On the other hand, substituting an active crossover for a manufacturer's passive one may have unintended consequences, since the manufacturer's crossover may include notch filters or response-tilting filters optimally matched to flatten the response of the drivers.

At any rate, if you decide to use an active crossover and do not need the flexibility of user-adjustable crossover frequencies, the Triphon unit is an excellent one that you should seriously consider. ❖

REFERENCE

1. The oval laminated core of an R-type transformer is continuous like a toroid, but has a circular cross-section made up of varying widths of grain-oriented silicon steel laminations. This gives the minimum iron weight for a given core area. The winding coil forms are also circular in cross-section. This allows a shorter wire length per turn and less overall coil resistance.

Xpress Mail

ELECTRONIC SENTIMENTALISTS

I have subscribed to your various magazines for the past 20 years, including *Speaker Builder*, *Audio Amateur*, and your current effort, *audioXpress*.

My professional audio experience goes back to the early '50s, and I am still very active in the field. In the past I have enjoyed and profited from the thoughtful design and insight which was evident in almost every issue.

My subscription is due for renewal, and I find myself questioning whether there is any future in your current direction. Some of the projects are undoubtedly beautiful pieces of craftsmanship, and perhaps art. The designer's emotions are obviously very involved.

However, the lack of serious objective data, the ardor expressed for a "lovely, liquid sounding" 2W single-ended power amplifier, and designers who don't believe in measurements, has pushed any pretense of science over the edge. The pages appear to be overrun by Luddites and electronic sentimentalists entranced with 2nd order harmonic distortion.

Please let me know when you come back down to the ground and I may be happy for your return to reality.

Richard K. Fullmer, P. E.
Salt Lake City, Utah

I just *love* your magazine! I think that the critics ought to realize that there is probably no other mag that attends at all to any tube enthusiasts!

I'm building Joseph Norwood Still's 60W Triode/100W UL Control Amp (*aX* 6/03), Pete Millet's LO-MU Preamp (*aX* 2/04), and now I have something to replace the failing solid-state amp in the office with Rick Spencer's Mini SE Amp (*aX* 4/04). My speakers have a 98dB sensitivity figure, so it'll work fine! Looks as though I'll have plenty to keep me busy for quite some time. I'm a semi-retired organist (pipes, classical, and the mighty WurliTzer!), so I can appreciate good audio construction articles. "Toobers" bring me back to the dear days of my teen-age childhood.

So, don't let the "complainers" get you down. As I tell my other organist colleagues who continually put down an instrument for what it doesn't have, they should be grateful for what it does have, enjoy it and make it the best!

Martin Boehling
Deemartrns@aol.com

HEGEMAN SUB

Cornelius Morton's "A Hegeman Subwoofer" was a thought-provoking article (Dec. '03), and I am contemplating building a similar subwoofer. However, I had some additional questions for Mr. Morton, and I would appreciate his insights.

1. The article states that the stub frequencies should be centered about the driver parameter Fs. Does this imply that I can simply pick a driver, and adjust the length and volume of the stubs to bracket this parameter?
2. Is there any guidance to the spacing of the stub resonant frequencies? How about Q?

Rob Weinstock
rweinstock@archstonconsulting.com

Cornelius Morton responds:

Thank you for your interest in my article. I hope that the following will be beneficial.

1. *The design as presented has been tested with drivers with an Fs of 28Hz to 35Hz and should operate over a range of 26Hz to 37Hz. Beyond that you may need to adjust the tuned frequency of the tunnels. The length of a tunnel may be found as L in inches equals 3360/frequency. Note that I try to keep the driver Fs within the bandpass of the second tunnel, the 30Hz tunnel in this design.*
2. *The area of the tunnel opening is dependent upon the effective piston area of the driver cone. The tunnels shown—all four—have a total area of $266cm^2$, or about 80% of a $330cm^2$ piston area. Piston area for similar drivers may vary by 10% depending on the surround width and so forth. The $266cm^2$ total tunnel area will be good for any 10″ driver.*
3. *The tunnel frequencies are spaced as multiples of the square root of two. Starting with the lowest frequency tunnel, the second tunnel is tuned 1.414 times higher and so on.*
4. *The unloaded Q of a tunnel is determined by the circumference and area of the tunnel opening and the tunnel length. For this design, the longest tunnel should have an unloaded Q of about 12–15 and the shortest about six. Loading the tunnels with open cell foam will set the Qs to between two and four as the system is tuned for best response.*

See the "Notes on Drivers" box at the end of the article for driver requirements and a few recommended units. Due to the very low distortion and extra long Xmax, my favorite is the Adire Brahma 10, which I am running in my sub. I would think that any on the list or others of similar specs would make an excellent-sounding subwoofer. The 12.5mm figure for Xmax should be an absolute minimum, as bottoming may occur on very loud low notes.

This box is unique in one area. The effective box volume is determined by the volume of the four pipes and the volume of the plenum less the volume of the driver. In a typical cabinet the value of Fs is determined by the square root of (1 + Vas/cabinet volume). This is due to the trapped air acting as a spring or compliance. In this cabinet the trapped air acts as a resistance and has little effect on Fs except for reducing the resonant peak.

OPTICAL READ-HEAD CLEANING

The literature for neither my CD player nor DVD player contains any instructions for maintaining the cleanliness of the read-head optics. But since there is a slow, continual accumulation of contaminants, I had a look at the WWW for a cure.

About all I found on the web were sites offering the same fix—a disk with brush hairs fastened to it which are intended to clean the optical system surface. I wanted a more thorough cleaning job than that. The factory literature for disk players always shows how to remove the housings, so it was a simple matter to expose the optical lenses for examination and cleaning.

Equipped with a flashlight and magnifying glass, I studied the lens surfaces. There was evidence of particulate dust and a foggy coating of condensed outgassed material from the surrounding plastic. Rubbing alcohol and a cotton swab took care of the contaminants, followed by inspection with the flashlight and magnifier to verify all was well, with no cotton filaments clinging to anything. I also cleaned everything within the disk environment.

In the case of the DVD player, a frequent failure to initialize certain disks stopped occurring, and the imagery sharpened up with detectable improvements in color rendering. The replay of CDs in this player also experienced an improvement in low-level detail and accuracy of high frequencies.

Darcy E. Staggs
Orange, Calif.

SHEET-METAL PUNCH

I just read the article "Tube Audio Construction Tips, Pt 3: Metalwork" (*aX* 6/04). It's always been difficult to do the metalwork for a project, but I came up with what I think is a revolutionary solution. I built a computer-controlled punching machine that quickly, easily, and accurately makes chassis, panels, brackets, shields, and other flat-metal parts (*Photo 1*). I am offering this product for sale. Readers can visit http://aircastle.biz for more info.

John C. Osborne
Aircastle Enterprises
aircastlebiz@yahoo.com

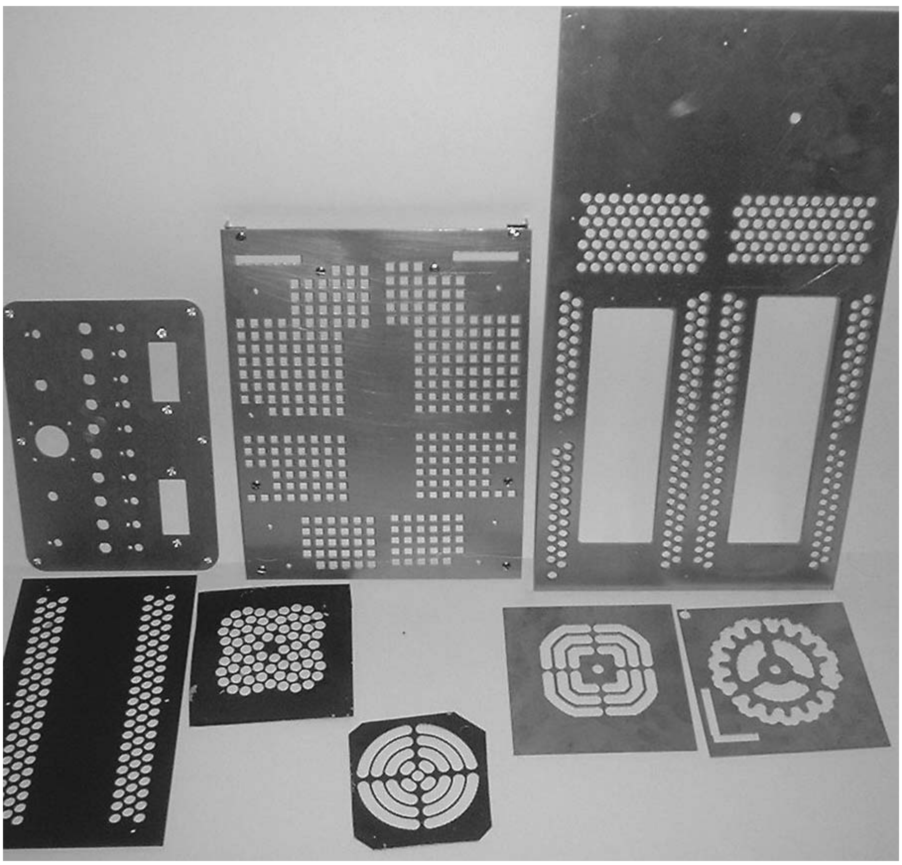


PHOTO 1: Metalwork using the Aircastle computer-controlled punching machine.

AMP VARIATIONS

The article by Joseph Norwood Still in the June 2004 issue of *audioXpress* ("35W Triode and 60W Ultralinear Control Amp," p. 26) is the latest in a long series of construction articles by Mr. Still that have appeared in *Glass Audio* and *audioXpress* over the last six years. The first article by Mr. Still appeared in 2/98 *Glass Audio*. (This article was a revised version of an article originally published in 1959.) The article presented a fixed bias version of the Mullard style amplifier using 6550 output tubes.

Since the initial article, additional articles by Mr. Still have covered numerous variants of this same basic amplifier incorporating alternative tube lineups and offering different control functions. Why do you continue to publish essentially the same article by Mr. Still over and over and over again? Please use the limited editorial space for material that is new and stimulating for audio hobbyists.

Dean Hiebert
dhieber@mchsi.com

Articles by Mr. Still are a good example of variations on a theme. His articles have been among the most popular with more evidence of his projects being built by readers than almost any other. —E.T.D.

PROJECT HELP

Dick Crawford's article, "A 1PPM Distortion Analyzer" (aX March '04, p. 8), was a challenging construction project, noted to be so by the author. Reader Frank Glabach contacted the author for help and has recorded his adventure for other constructors. The extensive list is posted on our website for reference. Those without access to the web may send a #10, stamped envelope to "Distortion Analyzer" at the magazine's address.—ETD

USHER REVIEW (CONT.)

Being the designer of the Usher CP8871, my comments may appear self-serving, but here goes! In his response to Ross Herbert's letter (June '04 *aX*, p. 65) James Moriyasu states that his comment on resistor position referred to the tweeter crossover. In particular, he says, "My comments on the resistor placement refer to the one in series with the tweeter." Clearly this is not true. There is no resistor in the tweeter circuit. Furthermore, on page 63 of his CP8871 review in the 2/2004 issue, he states, "The placement of R2 ahead of the capacitor and inductor in the mid/bass crossover compromises power handling, because R2 takes the full bandwidth of the signal."

This is nonsense. Mr. Herbert is correct. All input current passing into the midrange crossover must pass equally through the series-connected R2, C2, and L4. Thus the power dissipated in R2, which is proportional to the current squared, is the same at every frequency regardless of the order in which R2, C2, and L4 appear, and the order in which these parts are placed does not change the circuit response.

I'll take this opportunity to comment further on the Moriyasu review. I have no argument with the reviewer's measurements. His data closely parallels our own data, produced during the development of the CP8871. However, his interpretation on the sonic impact of the data is not backed up with any critical listening tests. In fact, in the 10+ pages of his review, there are only two sentences devoted to the speaker's sound, and this is qualified with his admission to lacking a "golden ear."

Mr. Moriyasu makes many negative comments on parts selection, parts placement, port tubes, and construction with only conjecture on how these points may impact sonic performance. I won't respond to each one of his comments, but as an example he takes issue with coil placement and possible mutual coupling between them. Yet his own plot of crossover voltage response (his Fig. 12) shows the out-of-band response of all the crossovers is down 35dB with slopes that clearly show response is continuing to fall below the level of the graph. Our own data shows the effect of mutual coupling in and between crossovers is over 50dB down in the crossover voltage responses.

His experiment with the open circuit voltage response of coupled coils does not take into account how that coupling holds up in an actual circuit.

With regard to his comment on the use of fast-on quick connects, contrary to soldered connections, these connectors are actually more reliable in production. Every high-end manufacturer I am familiar with uses them. Regarding sandcast resistors, our extensive listening tests with more exotic resistors showed no discernible sonic benefit.

In all cases, parts selection was carefully weighed against cost. Unlike *audioXpress* readers and other DIY enthusiasts, manufacturers of loudspeakers for the retail market must work with cost constraints if they are to meet a price point. They must also contend with the dealer markup, typically 100% to 110%. Mr. Moriyasu considers the CP8871 an expensive loudspeaker, but a US- or European-made speaker of comparable quality would cost two to three times as much. This explains the success of the CP8871 and other Usher speakers in the US.

Joe D'Appolito
audioltd@metrocast.net

James Moriyasu responds:

Sorry, my mistake; while I clearly identified R2 as the resistor ahead of the midrange crossover in the review of the Usher CP8871, I erred in my response to Mr. Herbert's letter. Clearly, there is no resistor in the tweeter crossover.

According to Watt's law, power can be calculated three different ways: $P = I \times I \times R$, $P = V \times I$, or $P = (V \times V)/R$, where P is power, I is current, V is voltage, and R is resistance. So it seems that voltage comes into play when looking at how a crossover affects power, since a crossover attenuates voltage at different frequencies.

Since I'm not very adept at analyzing circuits on a theoretical basis, I decided to build the circuit in Fig. 1 *and see whether the resistor R1 would experience different levels of power and thus be hotter or cooler than R3. R2 is an 8Ω resistor, which functions as a dummy load, representing a loudspeaker driver. The capacitors and inductors form the familiar second-order network. By carefully choosing values, the transfer functions of the networks are virtually identical. See* Fig. 2, *which is the voltage transfer function measured across R2 for both cir-*

to page 72

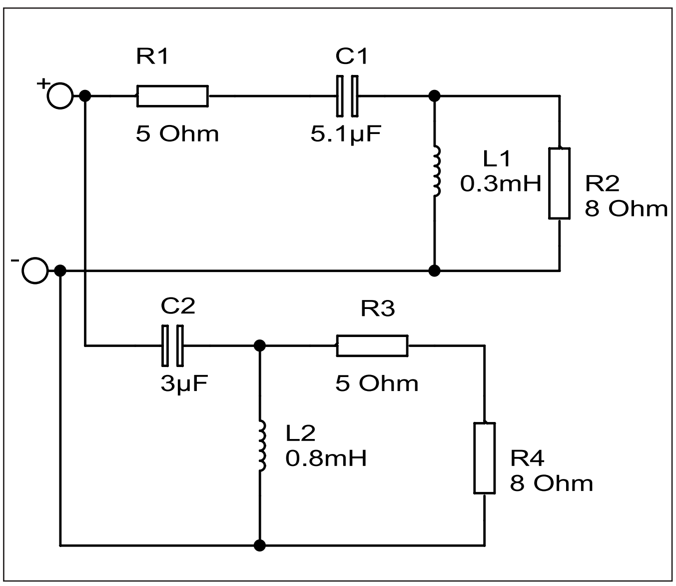


FIGURE 1: Circuit testing effect on resistors.

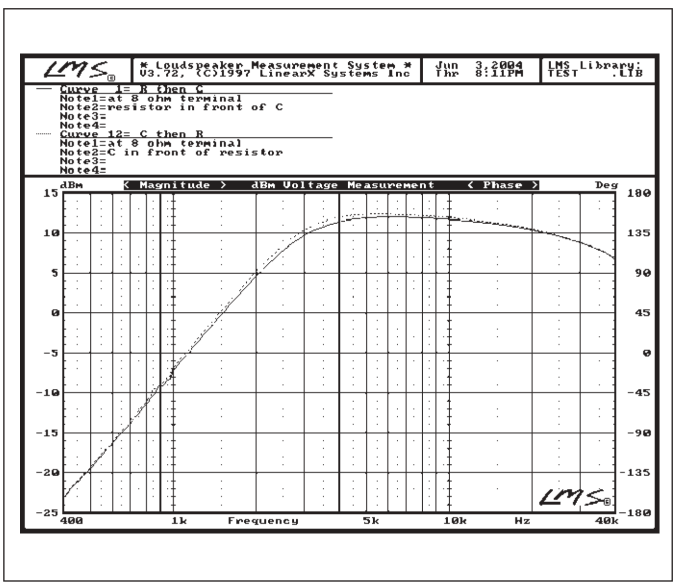
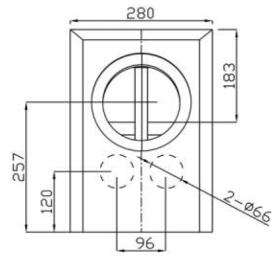


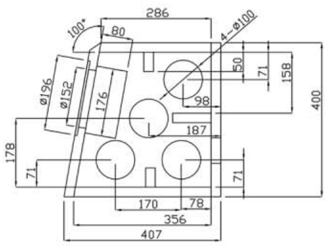
FIGURE 2: Voltage transfer function measured across R2.

Swans Tempus Kit

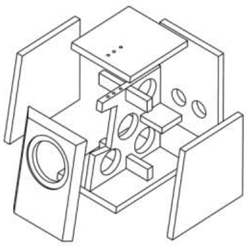
Swans proudly presents their first European-style, independently developed loudspeaker kit. Due to exceptionally high standards for a kit speaker, TEMPUS is a top-level, world-class performer in all regards.
German sound magazine KLANG+TON featured the Tempus in the June 2002 issue, commending its excellent acoustic response.

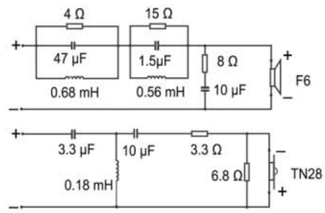


Front View

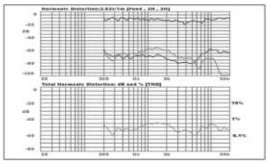


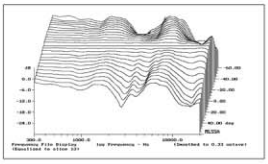
Section Side View

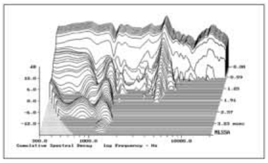

Explosion View

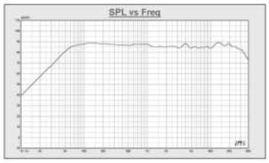


Filter Circuit

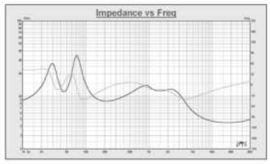

Tempus Harmonic Distortion


Tempus Horizontal response +60deg ~ - 60deg


Tempus Cumulative Spectral Decay



Tempus Frequency Response



Tempus Impedance

The Tempus project was initially conceived as a private work of the acoustic arts, to be executed independent of all professional affiliations. Despite a lack of traditional marketing and initial editorial commentary, the design has since received wide acclaim in the independent press and has gone on to tremendous commercial success. Tempus' reputation is therefore solely the result of unexpectedly high fidelity from a speaker of its modest origin and cost.

For Tempus' development, Swans agreed to supply premium components for what evolved into a rigorous electro-acoustics design program, executed to the highest European standards. The resulting design's performance was subjected to stringent laboratory confirmation and completely documented in the audio press, including revealing measurements not typical for any loudspeaker much less one of this type. Tempus has since gone on to claim top performance awards from acclaimed KLANG+TON magazine of Germany, during which time Swans cancelled all advertisements and further commentary.

Tempus is now one of Europe's most favored loudspeakers, leaving listeners admiring its naturally musical sound and excellent value. The design is already a classic, destined to sweep the audio world in a fashion similar to its current success.

The new Swans Tempus set includes:

- 2 x 1.1"/28mm Swans TN28 soft-dome tweeters with open-air housing, Neodymium motor, ferrofluid cooling, decompression chamber, integrated grille and German-made silk dome;
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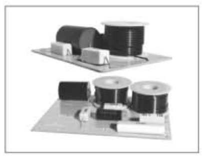
- 2 x completed crossover networks with premium-grade components;
- Two sets heavy input terminals;
- Two pairs internal cable sets;
- Complete manufacturer's instructions.



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One pair of THOR crossovers purchased from Zalytron, mounted on plywood; all wired up, tested, ready to go, $75 for the pair. ACI Jaguar (J2) built from kit; all drivers and assembled crossovers, $500 (purchased in 1999, cabinets are available but not included). Joanleswinter@aol.com or (908) 832-0347.

Speaker Lab 1208R 12″ a.s. lf (2), $60; E-V T35 hf (3), $50; E-V 1823-M horn drivers (2), $75; Linaeum TLS 1.25 dipole hf, 2.5–25kHz (2), $70; Mission 70 compact speakers, new film caps, Vifa hf (2), $80. Patrick Hughes, 7095 Baxter Grade, Auburn, CA 95603.

Marantz Model 18 receiver, any offers. Two Dyna Mark III amps; one sounds fine, other has blown tube and no sound from unknown cause. Both amps have *Audio Amateur* modifications (James E. Boak, 1/78). Call (212) 866-1877.

Counterpoint electronics tube/MOSFET power amp, $200; Crown DC-300 power amp, $150; Pioneer Spec 2, $200; Altec 1594B (two available), $125 each. Reed Hurley, 110 Country Place Drive, Stockbridge, GA 30281, (770) 474-6594, j-rhurley@webtv.net.

Edgar Monolith enclosures free to good home—must pick up. Parts, tubes, speakers, and more; e-mail mcottrell@comcast.net for complete list.

AD INDEX

Xpress mail
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cuits. What the circuit does is provide a high-pass crossover at around 3kHz.

Using Linear X's LMS, I applied a 32W signal to both circuits simultaneously. At 1kHz, R1, which is in front of the capacitor, was definitely hot to the touch, within a minute or two. R3, which is after the capacitor, remained cool to the touch. At 500Hz, R1 was warm while R3 remained cool. At 5kHz, both resistors were very hot to the touch; I could touch them for 2–3 seconds before having to pull my fingers away.

I think this simple experiment proves that there is more power being dissipated in R1 than in R3 at frequencies that are being attenuated by the crossover. This leads to the conclusion that putting a resistor ahead of a capacitor, in a tweeter or midrange crossover, exposes that resistor to more power than if it were after the capacitor.

Since audioXpress *is a magazine aimed at the hobbyist, I thought it would be appropriate to conduct a loudspeaker review that would be more in keeping with what the average reader would be interested in. I didn't think I was qualified to produce the type of review found in "high end" magazines, since I don't regularly audition expensive loudspeakers. I do think that I am reasonably qualified to conduct a "forensic" type of review where the innards of a loudspeaker are dissected, however, since I regularly design and build loudspeaker systems.*

Thus, my criticism on parts selection, parts placement, port tubes, and construction was made with the speaker builder in mind. In past conversations with Vance Dickason, I have become well aware of how important it is to the manufacturer to keep the cost of a loudspeaker as low as possible. But that isn't necessarily important to many speaker builders who are willing to buy the best parts and make the extra effort to build loudspeakers that rival commercial ones.

So while I went into the review without any pretenses or biases, in the end, the review proved that hobbyists can not only save thousands of dollars, but they can also have better quality than a loudspeaker with an $8000 price tag.

But is this fair criticism of the Usher CP8871? I think so. If I were shelling out relatively large numbers of dollars for a product, I think I might be appalled to find out that the manufacturer chose to use the cheapest parts and cut corners. Maybe there should be more "forensic" reviews that dissect consumer products and analyze them for quality.

ADDENDUM

The power supply schematic for Satoru Kobayashi's article "KT88 Hybrid PP Stereo Power Amp" (6/04, pp. 34–43) was inadvertently left out of the magazine. It can be found at our website (www.audioXpress.com/magsdirx/aX/addenda/index.htm).

HELP WANTED

I would like to know whether the Universal Stereo Preamp kit number KV-K2572 could be used to increase a DVD's line-level output without being overloaded. I sometimes copy DVDs to EP videotape for future playback. I find that the output level of DVD is far below the normal output level of prerecorded VHS tapes, so I must turn up the volume almost to its full extent.

Please be advised that I do this only for my own personal use and do not sell or give copies to other people. Sometimes I need to boost the volume level just in order to hear dialogue that would be misinterpreted without the aid of closed captioning. I seldom ever use the center-channel mode.

I am currently running the audio signal through a cassette recorder in its record mode in order to attain this higher level—a rather awkward procedure. The last time I made a copy of a DVD was over six months ago, and I find that I can usually find many economically priced DVDs at Wallyworld (Wal-Mart) or Best Buy.

Robert D. Vance
374 NW Heather St.
Port St. Lucie, FL 34983

I am searching for a schematic diagram for the Mark Levinson Preamp ML6A, especially L3A and L3 phono modules. If you have some information about this, please let me know.

Raymond
DRAY177@aol.com

Readers with information on these topics are encouraged to respond directly to the letter writers at the addresses provided.—Eds. ❖