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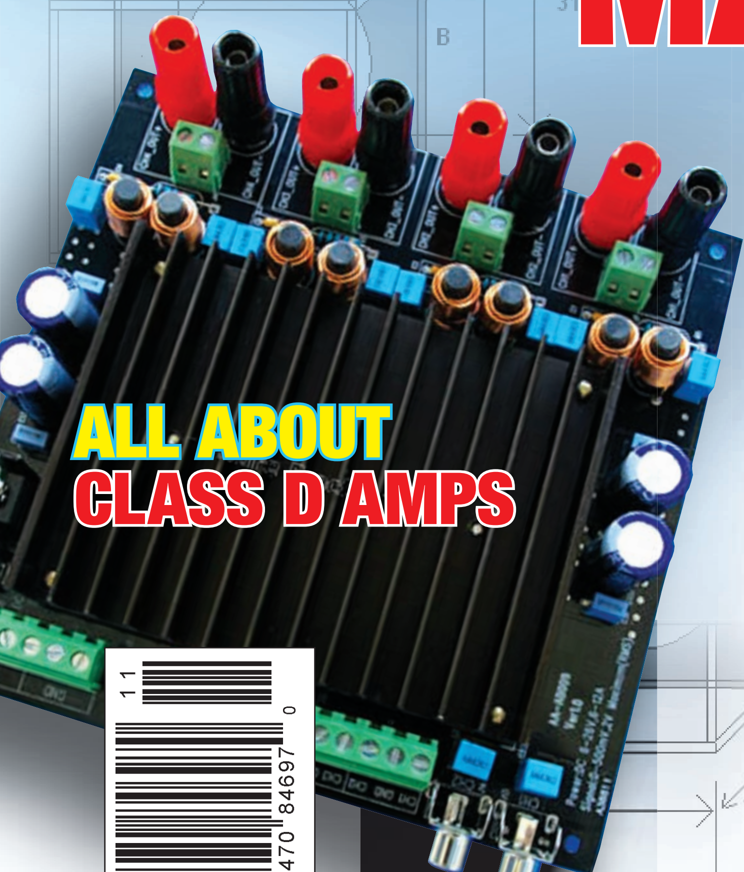

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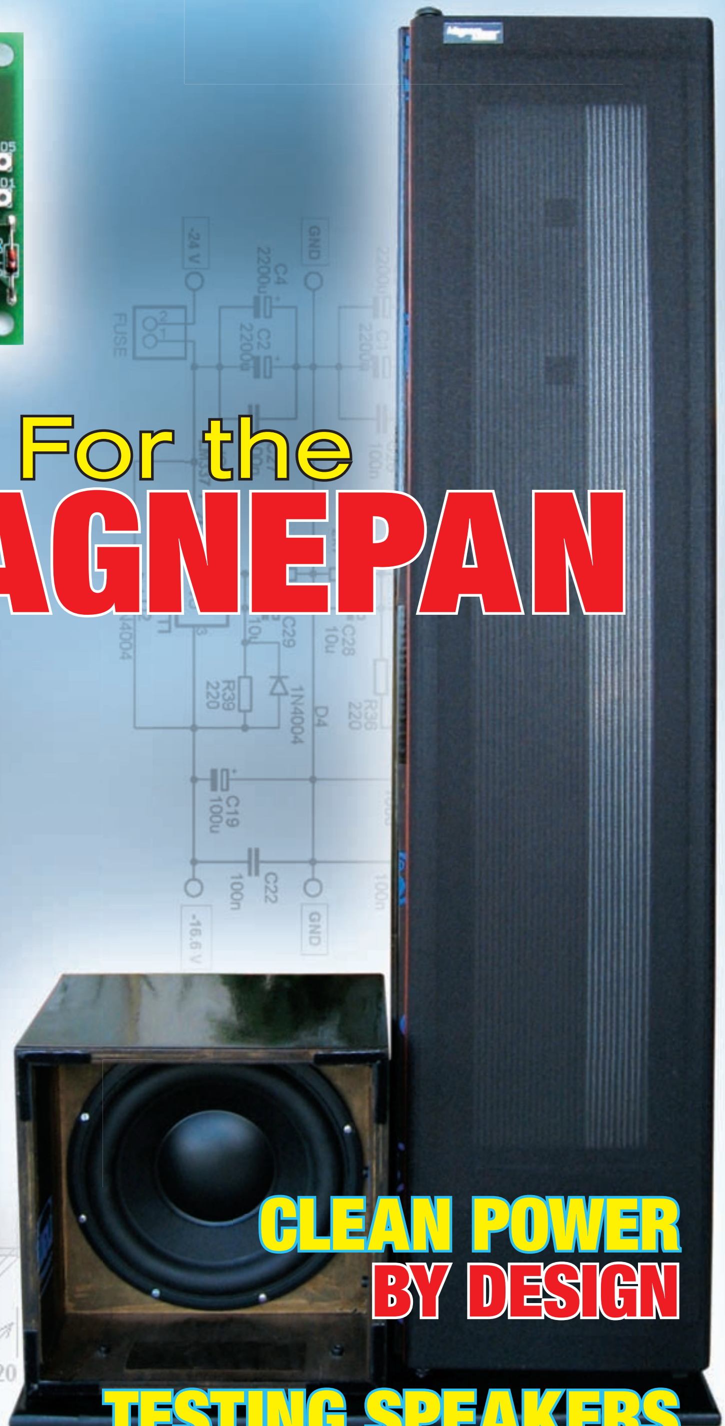

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Open Sourcing

There's good news for speaker builders and also for anyone who does chassis layout. I have been increasingly intrigued of late with Open Source software. Many of you have doubtless already discovered the joys of Open Source. If not, you have a treat in store. The Open Source movement is a well-organized cooperative effort by thousands of interested people all over the globe. This includes some large companies as well such as IBM, and Sun Microsystems.

The philosophy behind this movement is basically that software ought to be free. It might seem strange that when you go to a website to download such software, you must agree to a set of usage rules. All true Open Source software is copyright protected. The agreements generally require you to post the copyright notice with your downloaded copy. You are free to modify it for your own use if you document the changes clearly within the code.

Open Source groups have just won a highly important case in federal appeals court in Washington. A commercial software company had appropriated, without notice, pieces of an Open Source program for control of model railroads and sold it as part of their own product. This decision essentially validates the license almost all Open Source software providers use. If you don't honor the license, you lose the use of the software. And pay a penalty for infringing the copyright protection.

I have been enjoying one of the prime examples of Open Source work in the form of Open Office from Sun Microsystems. This is a six-module office suite which includes a word processor (Writer), a database manager (Base), a spreadsheet, a presentation program, a math unit, and a drawing program. I consider the word processor the best I have ever used and I have tried most of them. It has every bell and whistle anyone could want and is quite capable of producing a professional scholarly manuscript with all the features a student could need to achieve one. I see no reason why an author could not produce the text, footnotes, references, index, and formatting for a scholarly book. At the same time it is simple to write a one-page memo with it.

The drawing program is a big surprise. You may draw objects in two or three dimensions, and scale them absolutely accurately. The program allows you to draw lines of any weight, and allows you to locate them exactly from any starting point you choose. If you draw a line, the program keeps track of how long it is and where it is. If you choose to add a visible measurement, just draw the measuring line with arrow-ends and the distance will be supplied for you midway in the object. And all this is editable. Objects may be moved to a chosen location, or they may be moved by using the arrow keys. Like any new program this drawing facility takes a bit of time to learn. The help files are very clear and extensive.

I have yet to find the time to use the other four modules. However, I have converted Excel files with Open Office's spreadsheet, which OO does perfectly. It also converts the other pieces of Microsoft's Office for use in OO. Writer allows you to switch to any of the other five programs with a mouse click. Font selection is quite good and the color management for Writer and Draw are extensive and wide ranging.

These programs are constantly being upgraded by thousands of volunteer programmers—who receive no pay. Their motivation is a passion for quality and reliability—and the freedom from any domination of the software market. Open Source also offers operating systems, based on the Unix and Linux operating systems. I have had great fun with Ubuntu, which originated and is supported by an open source firm in South Africa. It has all the functionality of Windows without the glitches and sudden failures. It also has a beautiful presentation with a crisp, minimalist desktop which is pleasing. It can also reside alongside Windows on the same computer, offering dual booting which takes a bit of extra attention during boot up.

If you decide to download Open Source software, you will find you may do so with no immediate obligation. It is nice to be able to try the product first. However, nothing in this world is free. I make it a practice to forward modest payments once a year to those who provide the products I use. This includes Mozilla, Thunderbird, and Sun Microsystems. I believe strongly in freedom of choice for the consumer and certainly it is only right to support such efforts by putting your money where your mouth is. I plead guilty to preferring the underdog and always put AMD processors in machines I build. Underdogs always seem more interesting, somehow.—**E.T.D.**

By George Danavaras

The Lamda Loudspeaker

A subwoofer for the Magnepan MC1.

Lamda is a dipole loudspeaker consisting of a Magnepan MC1 loudspeaker and an open baffle cabinet with the Peerless 830452 XLS woofer (**Photos 1** and **2**). Each unit is driven through the Lamda active crossover by its own power amplifier. The Lamda active crossover contains a fourth-order Linkwitz-Riley crossover at 240Hz and several equalizers for the loudspeaker units. It covers the spectrum down to about 50Hz. You should use a subwoofer below this frequency.

THE MAGNEPAN MC1 LOUDSPEAKER

The Magnepan MC1 loudspeaker is a dipole speaker consisting of two vertically oriented drivers—one planar magnetostatic driver for the low frequencies

In this article, we've let stand the author's unconventional (perhaps phonetic) spelling of the eleventh letter of the Greek alphabet referring to his speaker system. —Eds.

PHOTO 1: The LAMDA loudspeaker.

PHOTO 2: Another view of the LAMDA loudspeaker.

and a quasi-ribbon driver for the high frequencies. The speaker is rather small considering its dipole nature, and is not capable of reproducing the full spectrum.

According to the manufacturer, the frequency response of the MC1 when attached on the sidewall of the listening room is from 80Hz to 20kHz, ±3dB. Of course, the response is quite different when you remove the MC1 from the sidewall. Although in the Lamda loudspeaker the MC1 operates only at frequencies above 240Hz, electronic equalization is applied by the Lamda active crossover to compensate the dipole loss to the frequencies below 1kHz.

CABINET DESIGN

The MC1 is a dipole source, and its ideal companion for the lower frequencies would also be a dipole source such as an open baffle woofer. When considering a woofer for an open baffle, the most important parameters are the large effective piston area, the large linear excursion, and the low distortion.

My choice was the Peerless 830452 XLS woofer. According to the manufacturer, it has an effective piston area of 352cm^2 and a 12.5mm peak excursion. Additionally, it has low distortion and high power handling capability. The woofer will cover the lower part of the spectrum down to the frequency of 51Hz. Below this frequency you should use a subwoofer.

You'll find a very good analysis for the design of an open baffle speaker at the site of S. Linkwitz (www.linkwitzlab.com). The main factor for the design of the open baffle cabinet is the distance that separates the positive and the negative sides of the woofer radiation. There are several different shapes that you can use as open baffle speakers.

The purpose of all of them is to increase the distance between the two opposite sides of the woofer radiation in such a way that the total size of the cabinet remains as compact as possible. One of the simplest shapes to perform this task is the H-frame.

Place the woofer on a baffle that extends its sides both in the front and in the back. The result is that the distance between the positive and negative sides of the woofer radiation becomes larger, while the total size of the loudspeaker remains acceptable. Assuming that the near field response of the dipole loudspeaker is absolutely flat, the far field frequency response of a dipole loudspeaker in the free field is shown in **Fig. 1**. The frequency Fpk where the first peak occurs is given by:

$$Fpk = (0.5{*}c)/D$$

where c is the sound velocity and D is the distance between the two dipole sources.

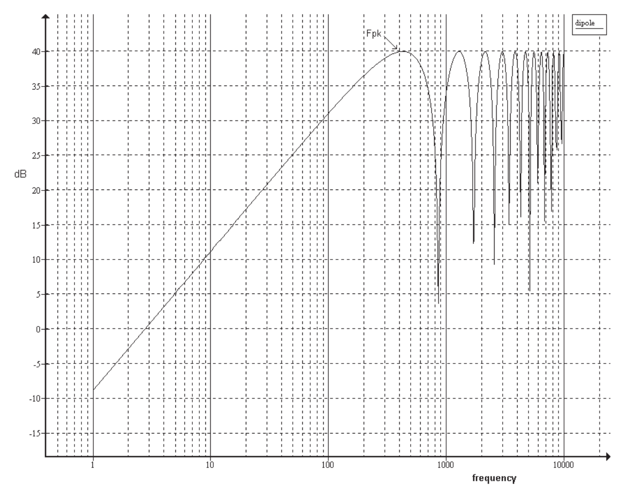


FIGURE 1: The far field frequency response of a dipole loudspeaker in the free field.

This peak in frequency response results when the negative polarity source after the 180° of phase shift due to the distance D adds to the positive source, and the resulting pressure is twice that of a single source (+6dB). Above the peak frequency there are also many other peaks and notches for the same reason.

Below the peak frequency Fpk, the sound pressure falls with 6dB/octave. This is due to the front-back cancellation of the sound. At the low frequencies the distance D is small compared to the length of the sound, so it cannot provide any isolation between the positive and negative sources. The distance D for my cabinet including the front and back covering is about 270mm, thus the Fpk will be about 640Hz.

The frequency at which the open baffle speaker has the same level output with a single similar monopole (for example, a closed box woofer) is given by: Feq = (0.17*c)/D = 220Hz.

Above this frequency the open baffle speaker has more output than a single monopole, and below this frequency it has less output that falls with -6dB/octave, where the single monopole has a flat response.

There is another peak in the frequency response of the open baffle cabinet, due to the cavity resonance which exists in the front and back of the woofer. This peak should be equalized with a notch filter so that the near field response of the baffle will be flat. Additionally, the far field low-frequency rolloff will be equalized with a circuit that boosts the low frequencies with -6dB/octave in order to flatten the frequency response.

The directivity of the open baffle subwoofer is maintained at the lowest frequencies. The response at all frequencies drops –3dB when the listening point is at an angle of 45° with the dipole axis and –6dB when it is at 60°.

The dimensions of the open baffle cabinet are shown in **Fig. 2**. The front and rear views of the cabinet are shown in **Photos 3** and **4**. I used plywood for the construction of the sides of the cabinet and MDF for the internal baffle that holds the Peerless woofer.

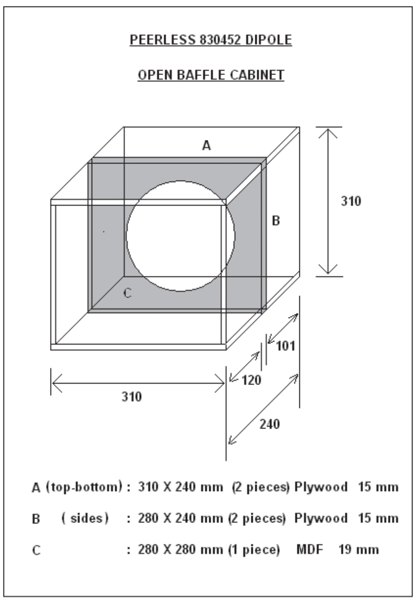


FIGURE 2: The dimensions of the open baffle cabinet.

PHOTO 3: Front view of the Peerless woofer.

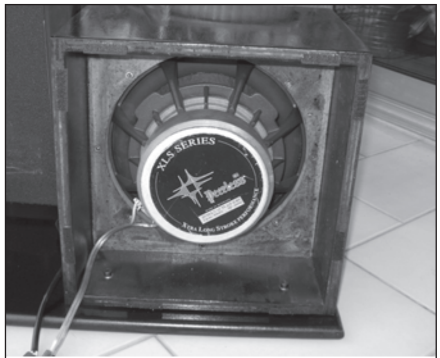


PHOTO 4: Rear view of the Peerless woofer.

The open baffle cabinet is also used to support the MC1. A wooden rod is screwed on the side of the open baffle. On the rod, the two supporting hinged brackets that are provided by the Magnepan are installed and support the MC1.

The MC1 is not vertical to the floor and has a slope of about 5° (**Photo 5**). This helps the center of the axis of the MC1 speaker to point directly to the listening position, which is assumed to be ideally at a distance of 2.6m and at a height of 1.1m above the floor.

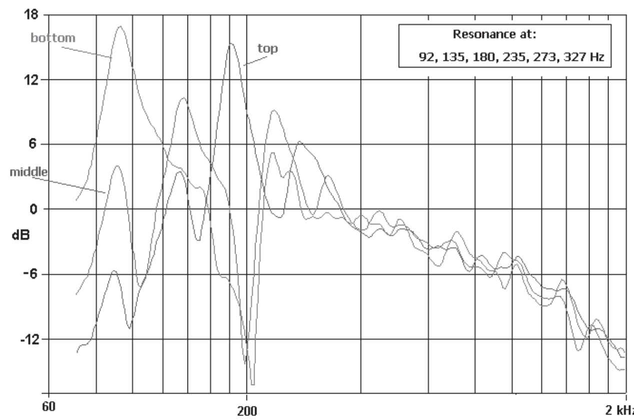


FIGURE 3: Near field responses taken at the bottom, at the middle, and at the top of the MC1 midrange panel.

CROSSOVER DESIGN

The Lamda active crossover includes the filters and the equalizers for the MC1 and the Peerless dipole. The selection of the crossover frequency is one of the most important decisions for the design of a loudspeaker so you should choose the frequency very carefully. It cannot be too high because of the woofer open baffle resonance and not too low because the MC1 will be overdriven.

My decision was made easier when I measured the near field response of the low-frequency driver of the MC1. The MC1 is a panel speaker with a very light diaphragm and boosts the low frequencies by using the resonance of the diaphragm. Actually, the diaphragm is divided into several areas and each one resonates at a different frequency. To evaluate this, I put the microphone very close to the diaphragm to measure three near field responses—at the bottom, at the middle, and at the top of the MC1 low-frequencies panel. The results are shown in **Fig. 3**.

The fundamental resonance of the bottom part of the panel is at 92Hz, while the fundamental resonance of the top part of the panel is at 180Hz. Two other resonances exist at 135Hz and 235Hz.

It seems that the MC1 should be ideally operated above 400Hz. I chose not to operate the Peerless woofer at such a high frequency, so a good compromise for the crossover frequency was 240Hz. The crossover topology is a fourth-order Linkwitz-Riley. The full electronic diagram of the lamda active crossover is shown in **Fig. 4**.

The input signal passes through a simple first-order low-pass and high-pass filter (R1, R2, C1, and C21) before it goes to the IC1A, which buffers the signal and applies equalization with a gain of about +2dB for the frequencies above 1kHz. At the output of the IC1A, the crossover is divided into three main parts:

a) The Magnepan MC1 part consisting of two second-order high-pass filters in series to form a fourth-order Linkwitz-Riley crossover at 240Hz, an equalizer of the MC1 response, and two stages of delay circuits.
b) The Peerless part consisting of two second-order low-pass filters in series to form a fourth-order Linkwitz-Riley crossover at 240Hz, a second-order high-pass filter at 51Hz, a buffer-inverter, a Linkwitz transform circuit to equalize the woofer low Q value, a notch filter to equalize the λ/4 resonance of the cavity, a low-frequency equalizer, and, finally, an equalizer for the compensation of the dipole loss.
c) The subwoofer part consisting of two second-order low-pass filters in series to form a fourth-order Linkwitz-Riley crossover at 51Hz.

THE MAGNEPAN MC1 PART

The first step in the design of the crossover is the measurement of the MC1 frequency response. This frequency response is taken as the reference response for the computation of the crossover and the equalizer. It is obvious that you should take this measurement very carefully.

I took all the measurements in my listening room, given some limitations. I placed the MC1 on the floor and at least 0.5m away from the sidewalls. The microphone was always at a distance of about 2.6m from the speaker, and the height of the microphone was about 1.1m above the floor. In all measurements the microphone was pointed directly to the center of the MC1.

I took several measurements by moving the speaker and the microphone in different places to eliminate room influence as much as possible. This proved to be a very complex and time-consuming procedure.

The average frequency response is shown in **Fig. 5**. It is not much different from the response that was given in my previous article for the MC1 ("Analog Bass Control for Magnepans," *audioXpress* 8/07), although the measurement conditions were not the same.

The average response is very flat (within ±1.5dB) from about 1kHz up to 10kHz. Below 800Hz the response is falling due to the front to back cancellation, and above 10kHz there is a broad peak of more than +6dB at about 12 to 14kHz. For the design of the crossover, I used the Orcad 15.7 demo version. This is a free version of an excellent simulation program for electronic circuits, offering many capabilities.

I entered the topology of the circuit on the Orcad Capture Demo editor, and I also entered the measured frequency response of the MC1 using the special device EFREQ. I computed the initial values of the components using the relevant theory for the design of the Linkwitz-Riley filters and of the equalizers.

Then I modified the values of the components until the total response of the fourth-order filter and the equalizer was as close as possible to the reference output which was produced by a theoretical fourth-order filter using the special device GLAPLACE. The final values of the components for the MC1 Linkwitz-Riley crossover are shown in **Fig. 4** around the IC2A and IC3A. As expected, they are different from the nominal values.

The components around IC4A equalize the MC1 response by amplifying the frequencies below 1kHz for several dBs. The circuits around IC5A and IC6A form delay circuits that compensate for the difference in the phase response between the Peerless woofer and the MC1 at the crossover frequency. This is due to the different physical distances for the MC1 and the Peerless, to the drivers and to the filters.

I performed the compensation as

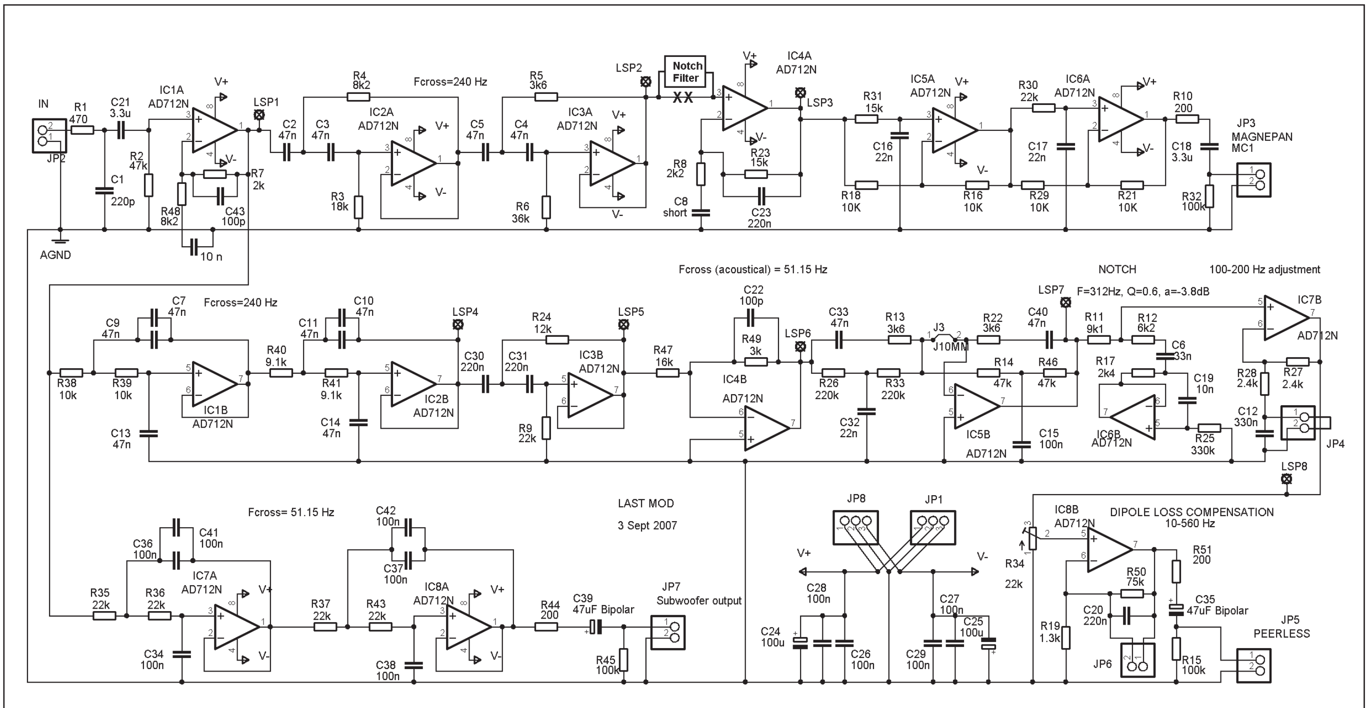


FIGURE 4: The electronic diagram of the active crossover.

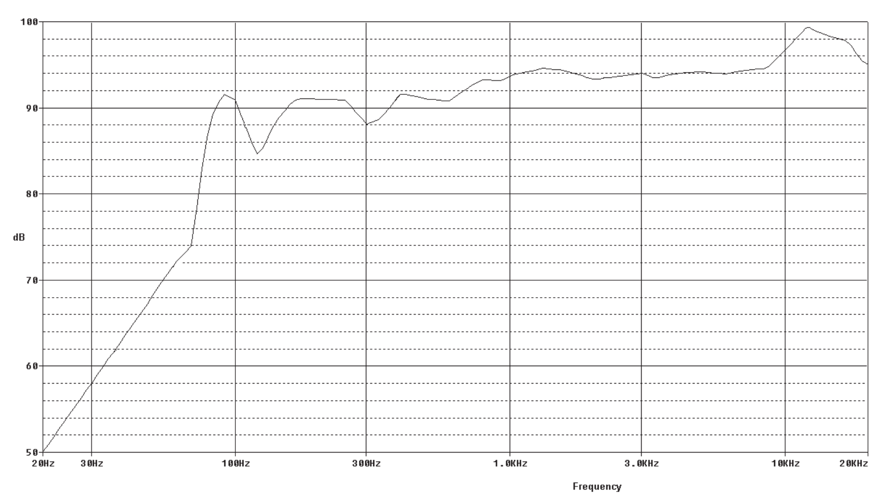


FIGURE 5: The average frequency response of the MC1.

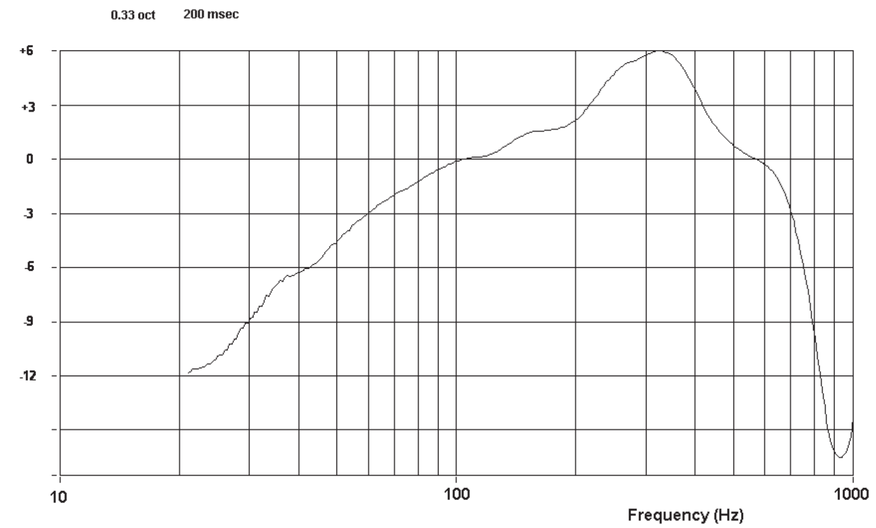


FIGURE 6: The near field response of the open baffle loudspeaker.

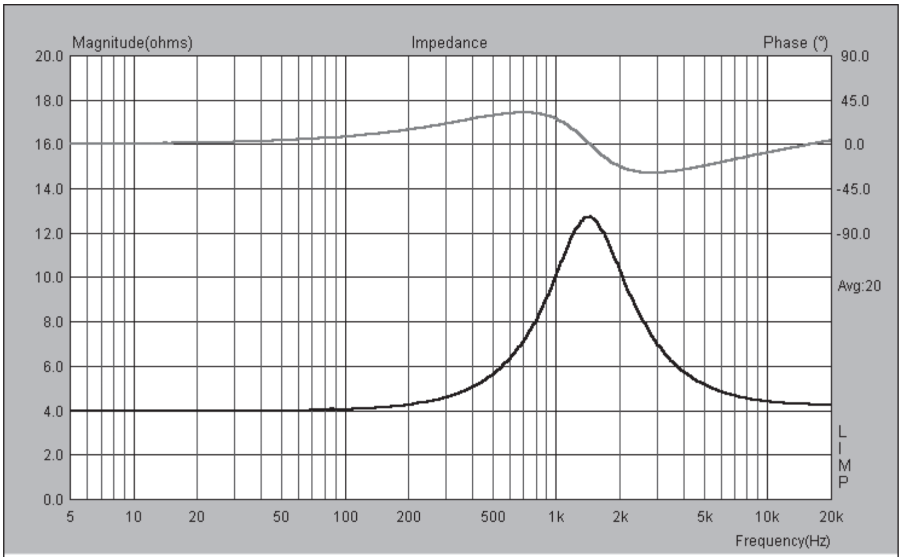


FIGURE 7: The impedance of the Magnepan MC1.

follows: I connected the MC1 and the Peerless to their power amplifiers through their active crossovers, but connected the MC1 out-of-phase in reference to the Peerless woofer. Then I changed the values of R30, R31, C16, and C17 and successively took frequency response measurements until the notch at the crossover frequency of 240Hz became as sharp and deep as possible. This means that the Peerless and the MC1 are in phase at the crossover frequency.

Figure 7 shows the impedance of the MC1 as measured with the demo version of another excellent program that can be used for loudspeaker measurement, the ARTA software (www.fesb.hr/~mateljan/arta/). The minimum value of the impedance is 4Ω, while the maximum value is about 13Ω at about 1500Hz. The phase has a minimum value of –40° and a maximum value of +40°.

THE PEERLESS 830452 PART

The Peerless crossover part is much more complicated than the MC1 part.

The components around IC1B form a second-order low-pass filter at 240Hz, while the components around IC2B form a second-order low-pass filter at 263Hz. Both filters with the acoustic response of the Peerless form an acoustic fourth-order low-pass filter at 240Hz.

The components around IC3B form the first part of the fourth-order high-pass filter at 51.5Hz. The IC4B is an inverter which is needed because the next circuit also inverts the signal. The circuit around IC5B is a Linkwitz transform circuit, whose operation I will explain later.

First, look at the near field frequency response of the open baffle with the microphone placed in the front opening as shown in **Fig. 6**. The peak in the response at 312Hz is caused by the λ/4 resonance of the cavity which is produced in front (and similarly behind) of the woofer. In the near field measurement, the negative side radiation of the woofer does not come into play. The loss in the frequencies below this peak is due to the low Qts value of the loudspeaker.

The response (excluding the peak) can be approximated by a second-order high-pass filter with Fo = 22Hz and Q = 0.34. This response is corrected by a Linkwitz transform circuit. The circuit removes the above pole and introduces a new pole at Fo = 51.5Hz with Q = 0.7 to form the second part of the fourth-order high-pass filter at 51.5Hz.

The components R11, R12, R17, R25, C6, and C19 around IC6B form a notch filter which removes the λ/4 resonance peak of the cavity of the open cabinet. The notch has its center frequency at 312Hz with Q = 0.6 and an attenuation of -3.8dB.

The next circuit around IC7B is an optional equalizer for the 100-200Hz range and is not used in the Lamda loudspeaker. The jumper JP4 is short circuit. The trimmer R34 is used to adjust the level of the Peerless woofer relative to the MC1.

Finally, the circuit around IC8B is the dipole loss correction circuit. It starts to boost the low frequencies with a 6dB/octave slope from the frequency of 560Hz down to the frequency of 10Hz. Resistor R51 buffers the output of the crossover and capacitor blocks any DC component from the signal.

In **Fig. 8** the impedance of the Peerless open baffle woofer is shown. I measured the impedance with the ARTA software. The minimum value is 5Ω/-7.7° at 127Hz, while the maximum value is about 56.7Ω/0.7° at 19.3Hz. The phase has a minimum value of –58° and a maximum value of +58°.

THE SUBWOOFER PART

As I mentioned before, the Lamda loudspeaker is used down to the frequency of 51Hz. Below this frequency you should use a subwoofer. For this reason, I included a standard low-pass fourth-order Linkwitz-Riley filter in the PCB. The filter is composed of two second-order low-pass filters around op amps IC7A and IC8A.

THE PCB OF THE LAMDA CROSSOVER

The construction of the Lamda active crossover is very complicated. For this reason, I designed a PCB using the demo version of the Eagle Layout editor. You can download the program for free from Cadsoft (www.cadsoftusa.com). The demo version is fully operational, except for a limitation on the maximum dimensions of the PCB. As a result, the dimensions of the PCB are 90mm × 120mm. The placement of the components on the PCB is shown in **Fig. 9**.

I used pin headers for the connection of the input and output cables. I placed the trimmer that is used for the level adjustment on the PCB, but you can use a potentiometer of similar value if you need to have this adjustment externally. I placed all the op amps on gold-plated sockets.

As with every filter construction, I strongly recommend measuring the values of the resistors with an ohmmeter and the values of the capacitors with a capacitance meter, at least for the components that are used for the filters and the equalizer. A tolerance of less than 1% for the capacitors will guarantee that the response of the filter will be very close to the theoretical response, which is very important. I usually buy a larger quantity of the capacitors than I need. Then I measure the value of each capacitor and choose the components with

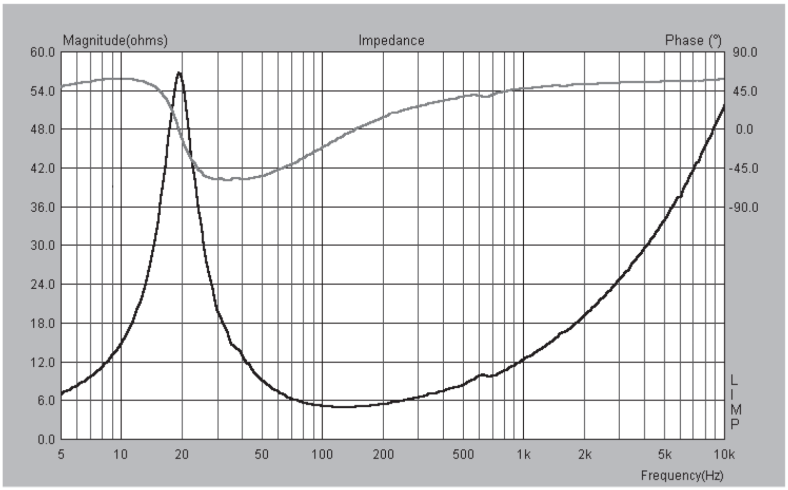


FIGURE 8: The impedance of the Peerless open baffle.

Notes:
Minimum impedance value 5 Ω / -7.5 deg at 127 Hz
Maximum impedance value 56.7 Ω / 0.7 deg at 19.3 Hz
Maximum phase value 58 deg / 14.8 Ω at 10 Hz

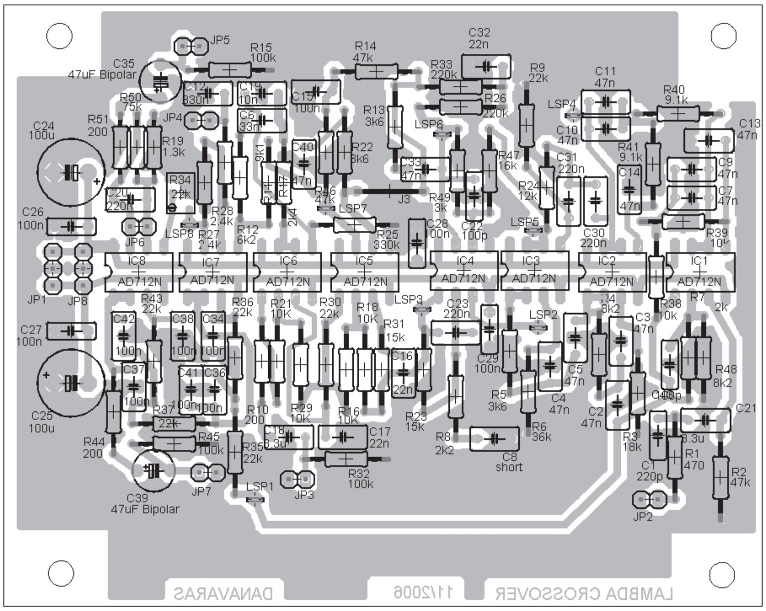


FIGURE 9: The placement of the components on the PCB.

the closest value to their nominal value. This is a very good method to have high accuracy filters with minimum cost. The assembled PCB of the Lamda crossover is shown in **Photo 6**.

I used a regulated power supply based on the LM317T and LM337T regulators for the supply of the circuit (**Fig. 14**). An output voltage of ±16V at about 200mA is suitable to drive the crossover. You should mount the regulators on small heatsinks.

Place the power supply board on the same box very close to the PCB of the crossover. Note that two PCBs are needed, one for the left and another for the right channel. All the boards and the power supply are on the same box (**Photo 7**).

NOTCH FILTER

Initially I had not intended to equalize the high frequencies peak of the MC1. I thought that no benefit would come from this equalization. But when I looked more carefully at the impulse response of the MC1, I changed my opinion.

The response is shown in **Fig. 10**. After the initial pulse there is a long oscillation which lasts for approximately 0.3-0.4 msec. I decided to design a notch filter to correct the response peak and suppress this oscillation.

After several tries, the best result was with a notch filter with the following parameters:
Notch frequency fs = 16.6kHz
Quality factor Q = 3

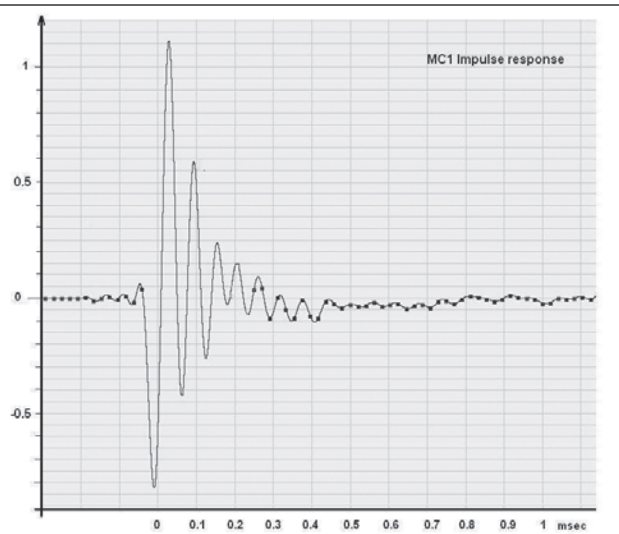


FIGURE 10: The impulse response of the MC1.

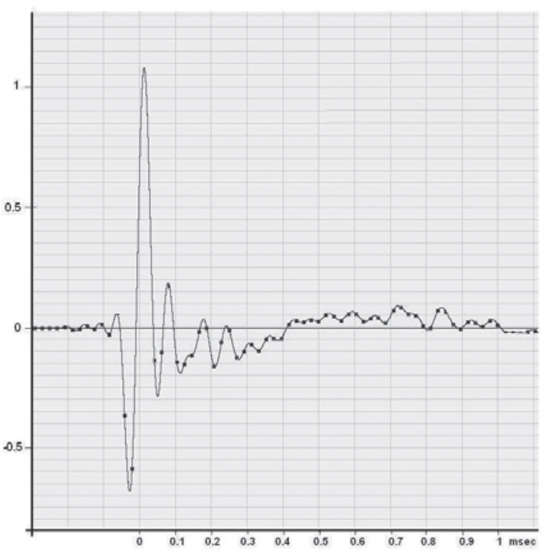


FIGURE 11: The electronic diagram of the notch filter for the MC1.

Attenuation at the notch frequency a = -8.5dB

The electronic diagram of the notch filter is shown in **Fig. 11**. I designed the notch filter with the method described by S. Linkwitz (www.linkwitzlab.com). The components R5, R4, and C2 around the op amp form an electronic inductor, which resonates with the capacitor C1 to form the notch filter.

Because the circuit was designed in a later stage, it is not included on the PCB. I assembled it in a small general-purpose PCB that I inserted between the output of the IC3A and the input of IC4A as indicated in **Fig. 4**. The track on the PCB that connects these two points is cut before the circuit is connected. The small general-purpose PCB with the circuit is shown in **Photo 6**.

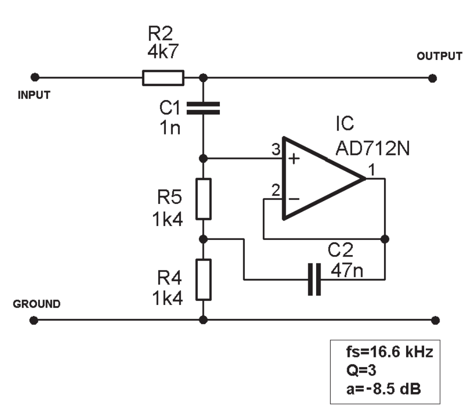


FIGURE 12: The impulse response of the MC1 driven through the Lamda crossover.

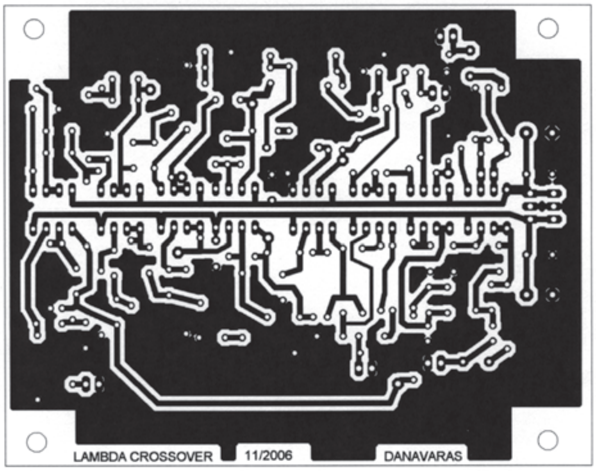


FIGURE 13: The PCB for the Lamda crossover.

The impulse response of the MC1 driven through the Lamda crossover with the notch filter is shown in **Fig. 12**. The oscillation is now well suppressed and the impulse response is much better than without the notch filter.

TESTING THE ACTIVE CROSSOVER

After assembling all the components on the PCB, you should test the crossover to verify its proper operation. For the tests, I used some basic equipment such as a low-distortion audio frequency oscillator, an oscilloscope, and an AC voltage meter.

I attached an oscillator at a frequency of about 240Hz and a level of 1V RMS to the input of the crossover. Then, with the oscilloscope, I checked that the output at the JP3 Magnepan MC1 and at the JP5 Peerless was clean


PHOTO 5: Side view of the Peerless woofer.

without noise or high-frequency oscillations.

Then I changed the frequency to about 55Hz and connected the oscilloscope at the JP5 Peerless and at the JP7 subwoofer outputs. I again checked that the output signal was clean without noise or high-frequency oscillations.

The next step is to verify that the filters and the equalizer have the correct frequency response. You can do this by measuring the frequency response at specific frequencies as given in **Table 1**. With the oscillator connected to the input of the crossover, measure the values indicated in **Table 1** at the JP3 Magnepan MC1 connector. The 0dB reference is the output signal at 3kHz.

You should follow a similar procedure for the Peerless part of the crossover. Measure the values (**Table 2**) in dB at the JP5 Peerless Out connector with an oscillator connected at the input. The 0dB reference is the output signal at 20Hz. Finally, check the frequency response of the subwoofer channel.

Measure the following values in dB (**Table 3**) at the JP7 subwoofer output connector with the input signal connected at the input of the crossover. The 0dB reference is the output signal

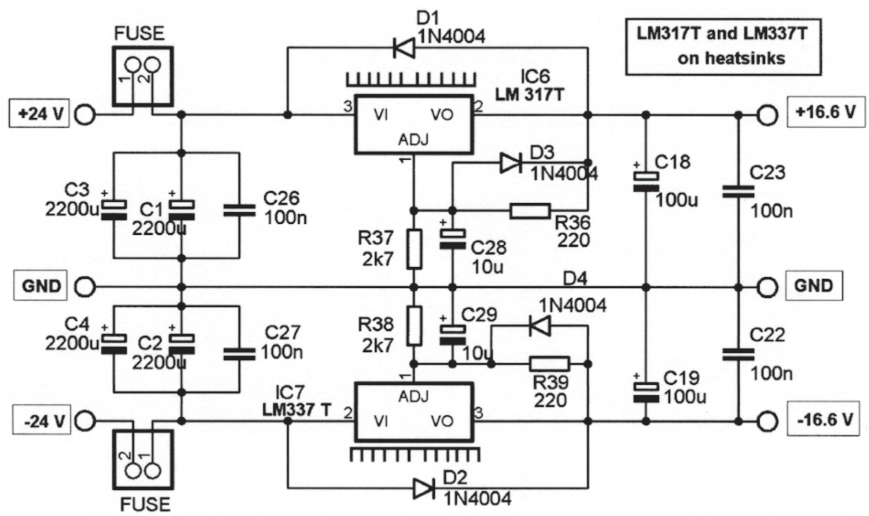


FIGURE 14: The regulated Power Supply for the Lamda crossover.

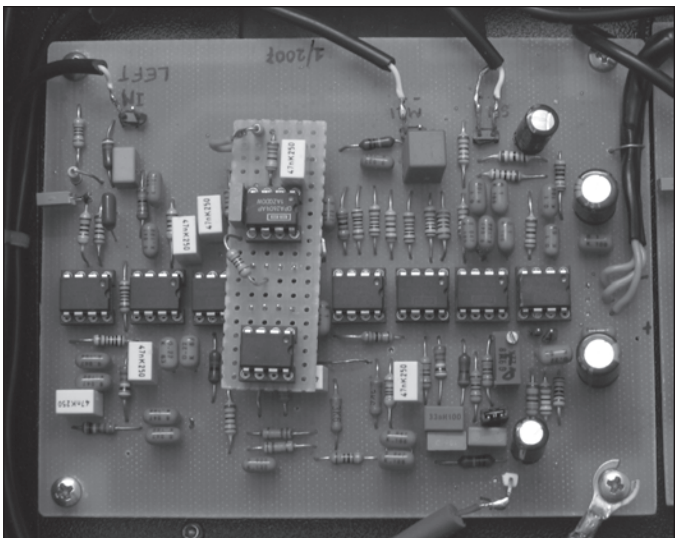


PHOTO 6: The PCB of the Lamda crossover.

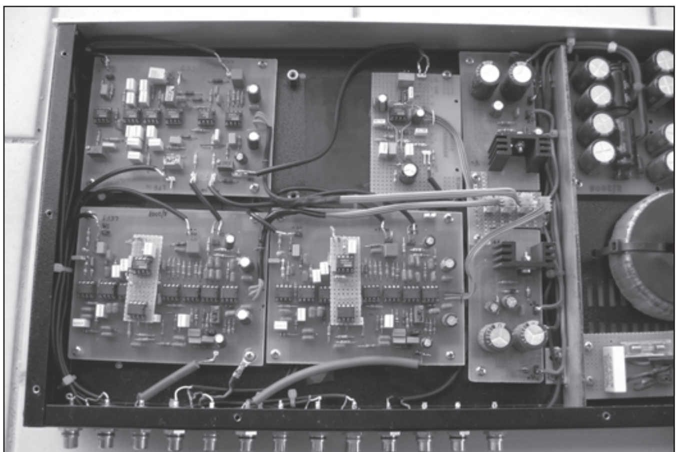


PHOTO 7: Top view of the active crossover box containing two Lamda crossover cards for the left and right channel, one subwoofer management card, one subwoofer equalizer card, and the power supply.

at the JP7 subwoofer output connector at the frequency of 20Hz. If everything is according to these specifications, the testing of the crossover is completed.

ADJUSTMENT

The adjustment procedure for the woofer level of the Lamda loudspeaker is as follows: Connect a generator with a signal of 240Hz to the input of the crossover. Adjust the level of the input signal until the voltage level at the input of the MC1 measures 1.5V. Then adjust the trimmer R34 until the voltage level at the input of the Peerless woofer is 0.21V (-17dB in reference with the MC1 input).

Note that the setting of the woofer level in reference to the MC1 is not an easy decision. This is because the in-room measurement of the frequency response below 300Hz is more a measurement of the room response than a measurement of the loudspeaker response.

The above level works fine for my room. If I move the loudspeakers to a different room—and since the operating range of the Peerless woofer is within the range that has the bigger influence from the room—it is possible that different adjustments of ±2dB or more on the above level will be needed. It is also possible that the level adjustment could be different for the left and right channel.

CONCLUSION

Lamda is a well-balanced loudspeaker. It maintains the huge stereophonic image of the MC1 and combines its transparency at the middle and high frequencies with the robustness of the Peerless woofer in the middle low and low frequencies. *aX*

TABLE 1

Frequency(Hz)	MC1 out (dB)
105	-25
140	-16.3
180	-8.5
240	0.15
350	5.1
420	4.1
850	0.5
1k	0.2
3k	0
10k	-1.2
12k	-2.6
14k	-5.2
16.6k	-8.5
18k	-7.1
20k	-4.8
30k	-0.8

TABLE 2

Freq (Hz)	Peerless out (dB)
20	0
35	8.5
58	11.2
82	9.8
105	8
140	5.1
180	1.4
240	-5.6
350	-17
420	-21.7
850	-44.7

TABLE 3

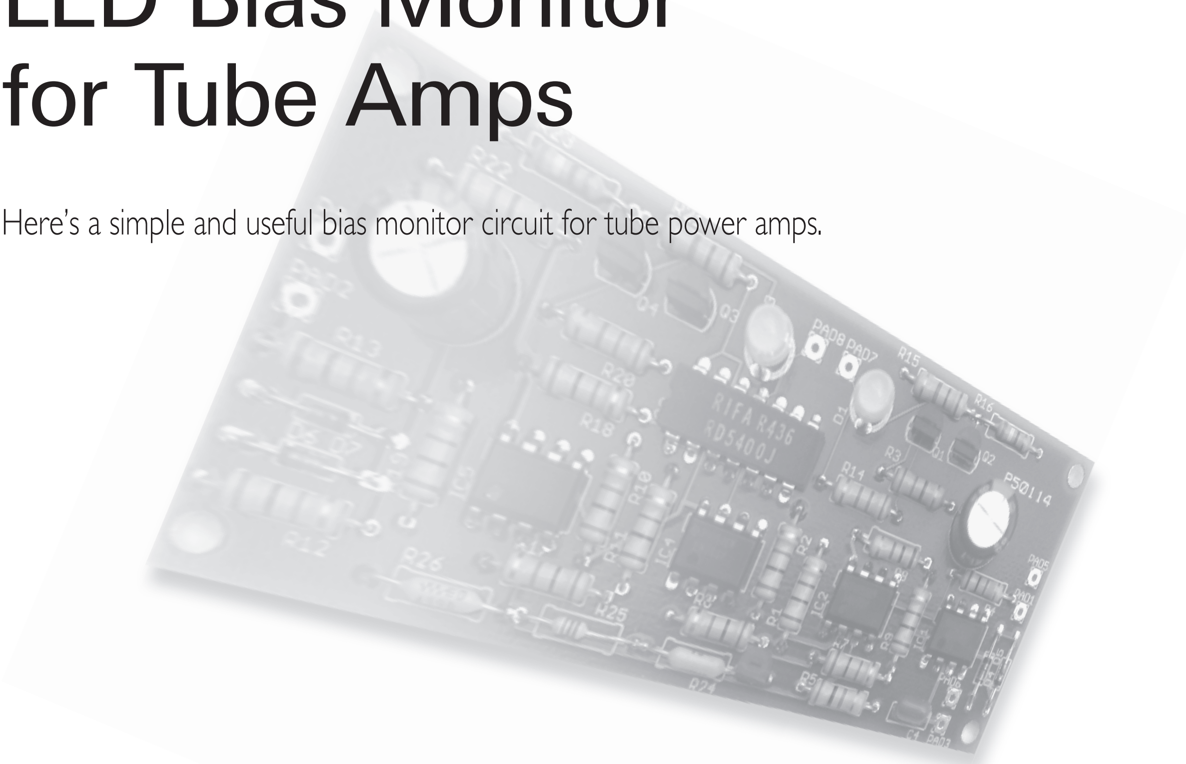
Freq (Hz)	Subwoofer out (dB)
20	0
35	-1.5
58	-8.3
82	-17.4
105	-25.3
140	-34.9

Lamda Crossover Parts List

Part	Value
C1	220p
C2-5, 7, 9-11, 13, 14, 33, 40	47n
C6	33n
C8	short circuit
C12	330n
C15, 22-9, 34, 36-8, 41, 42	100n
C16, 17, 32	22n
C18, 21	3.3µ, MKT
C19	10n
C20, 23, 30, 31	220n
C22, 43	100p
C24, 25	100µ, 25V
C35, 39	47µF, bipolar
IC1-8	AD712N
J3	Jumper 10mm
JP1, 8	Pinhead, 1 × 3
JP2-7	Pinhead, 1 × 2
All resistors 1% 0.25W	
R1	470
R2, 14, 46	47k
R3	18k
R4, 48	8k2
R5, 13, 22	3k6
R6	36k
R7	2k
R8	2k2
R9, 30, 35-7, 43	22k
R10, 44, 51	200
R11, 40, 41	9k1
R12	6k2
R15, 32, 45	100k
R16, 18, 21, 29, 38, 39	10k
R17, 27, 28	2k4
R19	1.3k
R23, 31	15k
R24	12k
R25	330k
R26, 33	220k
R34	22k, trimmer
R47	16k
R49	3k
R50	75k

LED Bias Monitor for Tube Amps

Here's a simple and useful bias monitor circuit for tube power amps.



This small circuit, which may be of interest to *audioXpress* readers, is a bias monitor for tube power amplifiers, consisting of a window comparator, some simple logic gates, and a bi-color LED. It indicates visually whether the tube bias current is within some user-determined range. A number of such circuits have been published over the years in *audioXpress*, *Glass Audio*, and *Audio Amateur*, but this one is versatile enough so you can add it to almost any new or existing tube amp.

Tube power amps that use fixed bias require occasional readjustment due to tube and component aging and line voltage variation. A pronounced drift in the required bias voltage is often the first sign that a tube needs replacement. The most common way of monitoring the bias current is by a test point that you can measure with a handheld multimeter. But I find this is often inconvenient, especially when the test points are inaccessible or the amp is located where I don't keep a meter. It's much more convenient if the amp can monitor itself.

MONITOR CIRCUIT

The circuit in **Fig. 1** is designed to do just that. Comparators IC1 and IC2 form a window comparator that can monitor the voltage drop across a current-sensing resistor in the tube cathode circuit, as shown in **Fig. 2**. For one of my amps, the tube is a KT88, which I bias at 50mA cathode current, and which equates to a 0.5V drop across the 10Ω current-sensing resistor. I decided that I wanted an indication when the current fell below 45mA or rose above 55mA.

In **Fig. 1**, the window edges are set by a voltage divider consisting of R26, R25, and R24, which divide the output of a 1.23V precision reference diode D2. If the input voltage is between 0.45 and 0.55V, the output of IC2 is high and the output of IC1 is low, which makes the output of logic gate IC3B low, and the LED D1 glows green. If the voltage is outside that range, the output of IC3B is high and D1 is yellow. You can set the lower and upper window edges *VL* and *VH* between 0 and 1.23V by changing the voltage divider resistors R24-R26, according to the formulas

$$V_L = 1.23\left(\frac{R_{26}}{R_{24} + R_{25} + R_{26}}\right) \quad \text{and}$$

$$V_H = 1.23\left(\frac{R_{25} + R_{26}}{R_{24} + R_{25} + R_{26}}\right)$$

You can also set them up to 2.5V by using the 2.5V version of reference diode D2.

Resistor R4 and capacitor C1 form a low-pass filter so that fast signal voltages across the cathode resistor don't affect the bias monitor. But sustained large signals will raise the average voltage across the cathode resistor, so the bias monitor is only accurate under no-signal or low-signal conditions. In practice, music signals don't seem to affect the indication.

Diodes D4 and D5 prevent a large transient above 5V or below ground from damaging the comparators. Feedback resistors R7 and R9 (and R6 and R1) add a few mV of hysteresis to ensure positive switching action at the window edges. The TTL output of IC3B may be useful elsewhere in your system; in one of my amps, a timer shuts down the power if the bias is out of range for a prolonged period.

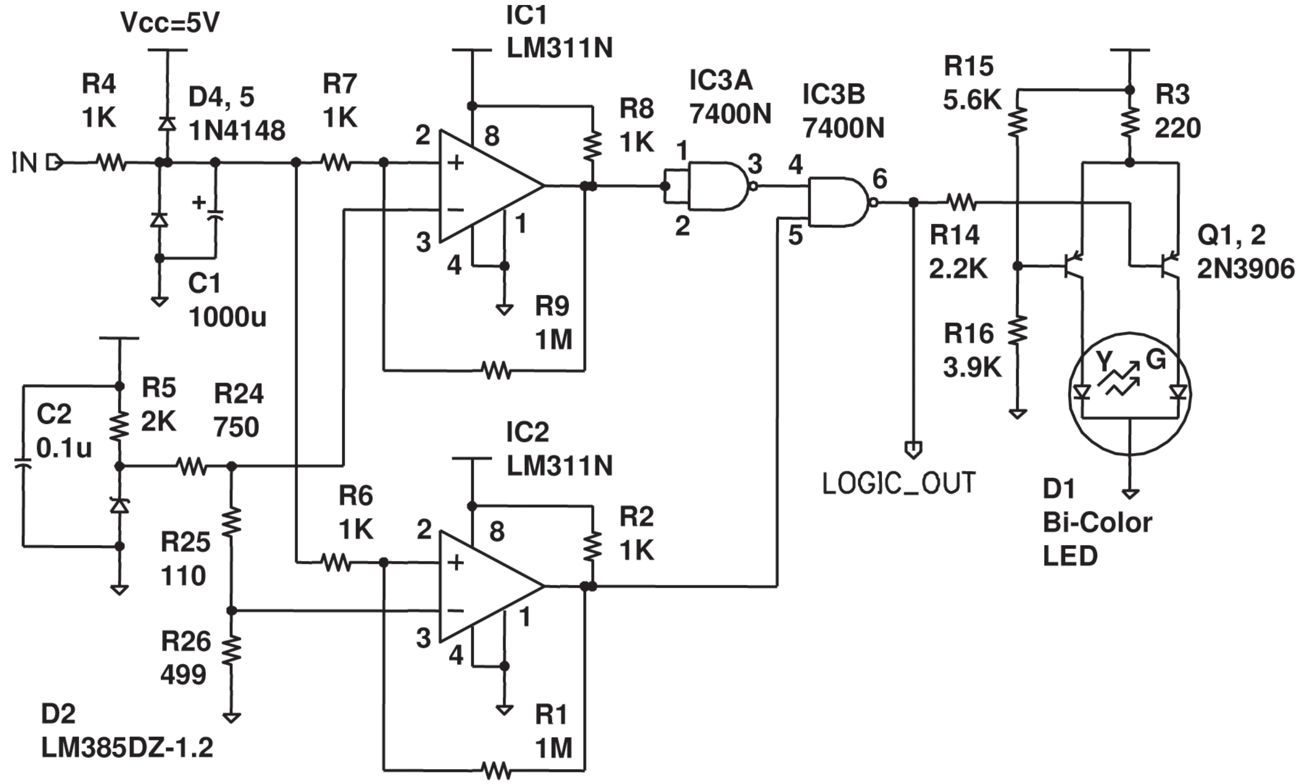


FIGURE 1: Bias monitor schematic. The PCB shown in *Photo 1* contains two bias monitor circuits, but R5, C2, D2, R24, R25, and R26 are common to both.

SIMPLIFIED VERSION

My original prototype was more complicated, with multiple comparator windows and a four-LED display for each tube. While that was useful for designing and testing a new amplifier, it was much fancier than it needed to be and had too many LEDs. This design has just one bi-color LED per tube. Although it is intended mainly for monitoring, you can even use it to set bias. Turn down the bias current until the LED turns yellow, noting the position of the bias potentiometer. As the bias is turned up, the LED turns green and then yellow again; you can set the bias pot halfway between the two trip points.

The circuit requires only a filtered and regulated 5V ground-referenced supply, at about 15mA per monitor. My homemade audio equipment almost always has a 5V supply for operating control circuits and timers. If not, you can usually derive a 5V supply from one of the tube heater supplies.

The wide variation in tube amp heater arrangements makes it impossible for me to give specific advice on deriving 5V, but I can give one general caveat. If the supply is obtained by rectifying a 6.3V AC heater winding as shown in **Fig. 3**, then the AC side of the rectifier bridge can't have a ground connection or a DC bias connection. In many cases, for instance, the center tap of the heater winding is grounded, which won't work in **Fig. 3**. But if the winding is left floating, the action of the bridge rectifier in **Fig. 3** will cause the center tap of the winding to be at about 4V, which is usually OK. You can then ground the center tap for AC through a small electrolytic capacitor

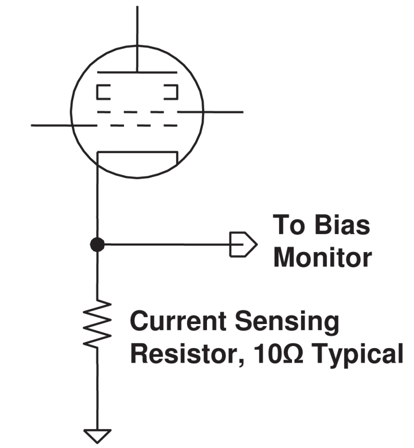


FIGURE 2: Output tube with current-sensing resistor.

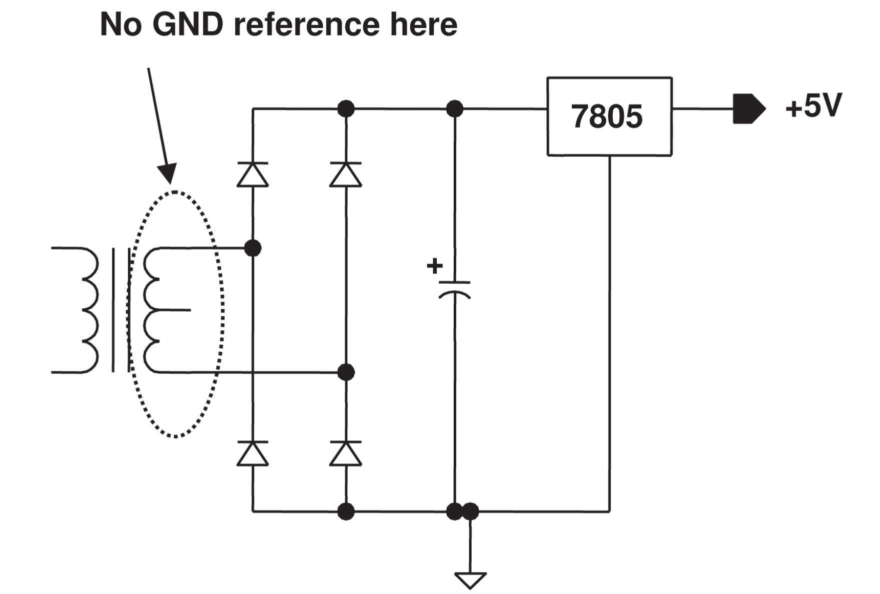


FIGURE 3: Possible power supply arrangement for bias monitor.

TABLE 1 Parts List

Component Description	Digi-Key #	Part #Reference
1000µF elec., 10V or more (2)		C1 (C2)
0.1µF film, 10V or more		C4
G/Y Bi-color LED (2)	160-1714-ND	D1 (D3)
LM385Z-1.2 precision reference diode	LM385Z-1.2NS-ND	D2
1N4148 Si diode (4)	1N4148FS-ND	D4-5 (D6-7)
LM311N comparator (4)	LM311NFS-ND	IC1-2 (IC4-5)
7400N	296-14641-5-ND	IC3
2N3906 (4)	2N3906FS-ND	Q1-2 (Q3-4)
1MΩ (4)		R1, R9 (R11, R19)
220Ω (2)		R3 (R22)
1kΩ (10)		R2, R4, R6-8 (R10, R12-13, R17-18)
2kΩ		R5
2.2kΩ (2)		R14 (R20)
3.9kΩ (2)		R16 (R23)
5.6kΩ (2)		R15 (R21)
750Ω, ¼ or ½W, 1%		R24
110Ω, ¼ or ½W, 1%		R25
499Ω, ¼ or ½W, 1%		R26

Part numbers in parenthesis are for the second monitor. All resistors are ¼ or ½W, 5%, carbon film, except as noted. Digi-Key part numbers are given for the semiconductors, but these are very common parts, and you can use components from any vendor.

to reduce hum and noise. If you are in doubt, it is best to add a small transformer for the 5V supply. Digi-Key and Mouser stock some very tiny ones these days.

CIRCUIT BOARDS

The bias monitor is straightforward to build on a piece of perforated board or a general-purpose prototyping board. The parts list is in **Table 1**. But to make the monitor compact and easy to add to my amps, I designed a 3.9″ × 1.7″ double-sided PCB (**Photo 1**), which contains two of the monitor circuits shown in **Fig. 1**, for the two tubes in a push-pull amp. I usually solder the two LEDs on the back of the board and mount it on the front panel of the amplifier chassis, so that the LEDs can protrude directly through LED mounting clips on the panel.

The PCB is a silk-screened board with solder-mask, plated-through holes, and fine lines, so it is not practical for most amateurs to make it at home, but I made a zip file with the Gerber files, an exact board schematic, and a parts placement diagram. I'm happy to send it to any amateur for their own personal (non-commercial) use. Although it can be expensive to have only a few boards made, PCB fabrication is a competitive industry, and sometimes the PCB fabrication houses on the Internet offer specials that make it quite reasonable. It pays to shop around. I also have a few boards left that I'm willing to sell to other readers. You can contact me by e-mail at scott@tavishdesign.com.

With this bias monitor, I no longer need to look at my power amps and wonder whether the bias is still in range. Nor do I need to go to the trouble of finding a meter, pulling the amp out of my equipment rack, and checking the bias, only to find that it is still OK. I've found that the monitor is simple and compact enough to be added unobtrusively to almost any amp.

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

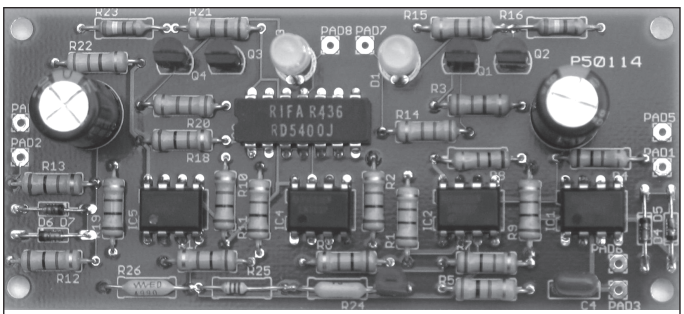


PHOTO 1: Printed circuit board (3.9″ × 1.7″) containing two bias monitors and the common voltage reference circuit. Although the two LEDs are shown here on the top (component) side of the board, I usually mount them on the back (see text).

 By Bruce W. Brown RPh.

Class D Amplifiers

Where these interesting switching amps rank in terms of efficiency and distortion reduction.

Before we get started, let me confess that I am not an engineer! I read a letter in a past issue of *audioXpress* that led me to investigate Class D amplifiers, so I am sharing my research and experiments with readers. I have included references with this article, should you wish for more information on the theory of operation and design. One of the most comprehensive is International Rectifier's website www.irf.com.

There is a misconception concerning Class D amplifiers; the D is not an abbreviation for "Digital." Digital and analog input processor designs exist for Class D, but the D stands for nothing more than the time and place in design—Class A, then B, then AB, then C, and now D. Class D amplifiers were originally designed in the late 50s, but it is only with the need for high power, compact, efficient amplifiers for car audio use that they have expanded into home audio equipment.

CLASS DIFFERENCES

Class A amplifiers: output devices are continuously conducting for the entire sine wave cycle. This design produces the least amount of distortion, but the efficiency is normally around 20%.

Class B amplifiers: output devices conduct only half of the sinusoidal cycle. One device conducts during the positive side of the wave and the other conducts during the negative part. If there is no signal there is no current flow through the devices. Efficiency is about 50%, but there can be an issue with crossover distortion due to the time it takes for the devices to switch on and off.

Class AB amplifiers: both devices are allowed to conduct, but just a small amount at the crossover point. Each device is conducting for more than half cycle but less than a full cycle, so the nonlinearity is overcome without the inefficiency of Class A. Efficiency is about 50%.

Class D amplifiers: also called switching amplifiers. Because the output devices switch on and off very rapidly, efficiencies of 90-95% are possible. The devices are either totally on or off, so there is a dramatic decrease in power utilization for a given output. This leads to fewer heatsink mass requirements and a decreased need for heavy-duty power supplies. If you have ever had any experience with a MOSFET Class A amplifier, such as some Pass designs, you clearly know the size and mass of heatsinks needed and the size of the power transformer to drive these babies! (I do love the sound of Pass Class A amps.) The high switching frequency in Class D amps has the potential of rf interference, so special design and shielding is necessary.

The basic function of these amps is to modulate a triangle or square wave with a sine wave audio signal. The pulse wave modulated (PWM) carrier signal drives the devices (almost always fast MOSFETs). The final stages of the amp contain a low-pass filter to remove the PWM carrier frequency. This is a simplification of a complex process, but it gives you an idea of the way they work. Gain is proportional to the bus voltage, unlike linear amplifiers, so some Class D amps use feedback to compensate for bus voltage variations.

There are two types of signal processing used in Class D amps. The first is analog processing, which is the type being used for audio amplification. Digital signal processing is also possible, but the cost increases substantially and may be prohibitive for general audio use. Similar to Class AB amplifiers, Class D can also be categorized as half-bridge and full-bridge configuration (**Figs. 1** and **2**). Each of these has pros and cons:

Half bridge is potentially simpler in design with two MOSFETs per output channel, but requires a bipolar power supply and may need substantial feedback and DC offset adjustment. It does have the advantage that the stereo outputs can share a ground or two channels can be bridged, doubling the output of the amp. It also potentially suffers from "power supply pumping," in which, as the output devices switch, there is substantial energy being pumped back from the amplifier to the power supply causing bus voltage fluctuations. These can cause distortion.

The full-bridge design uses four MOSFETs and two drivers, and the complementary switching legs tend to consume energy from the other side of

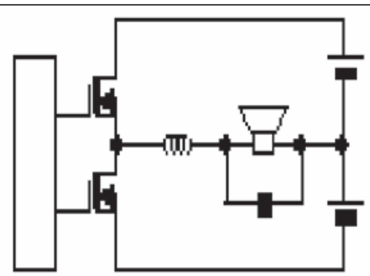


FIGURE 1: Half bridge.

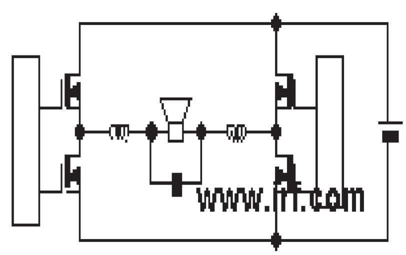


FIGURE 2: Full bridge (courtesy of IRF.com).

the leg, so no energy is pumped back to the power supply, eliminating distortion caused by the power supply. The power production is greatly increased in the full-bridge arrangement; however, because the speakers are being driven in a bridge configuration, the channels share no common ground. Theoretically, a Class D amplifier has no distortion and can provide 100% efficiency; however, this doesn't occur for a number of reasons related to timing jitter, switching device characteristics, ringing, and power supply fluctuations. Quality Class D amplifiers need to handle frequencies in the 100kHz to 1MHz range, requiring very high speed power and signal devices.

Because of extremely high switching speeds, a compact layout is essential, and surface-mount devices are required to get the performance needed. Stray capacitance and inductance of conventional throughhole components makes home-built units almost impossible to stabilize. The design of the PCB is also extremely critical. Well-designed Class D amps will have a high-order filter or special carrier suppression sections to avoid problems with EMI.

APPLICATIONS

Until recently, these amps were used primarily as subwoofer amps and in industrial applications, but there seem to be a few audiophile versions surfacing. In addition, this technology is being used extensively in mass-market surround-sound receivers. This allows very compact receivers weighing less than 25 lbs. to have seven channels of amplification (100W/channel RMS). But there are always compromises made in manufacturing. Not only do these amps provide a very compact footprint, but they also needed very little for heatsinking, and because of their efficiency, power transformers can be substantially smaller.

Some audiophile sites have claimed "tube-like sound"; others, harsh, cold, commanding bass and such. Many of these claims are based on economic gain, and some are just plain bunk. Because what I have read on Class D construction leads me to believe the design is well above my level, I decided to search the web and see what was out there. My eBay searches for "Class D amplifiers" resulted in mostly car subwoofer amplifiers. I tried "D class" and stumbled upon a huge assortment of pre-built D class modules from Shure Electronics in China; I really had to give some of these a try.

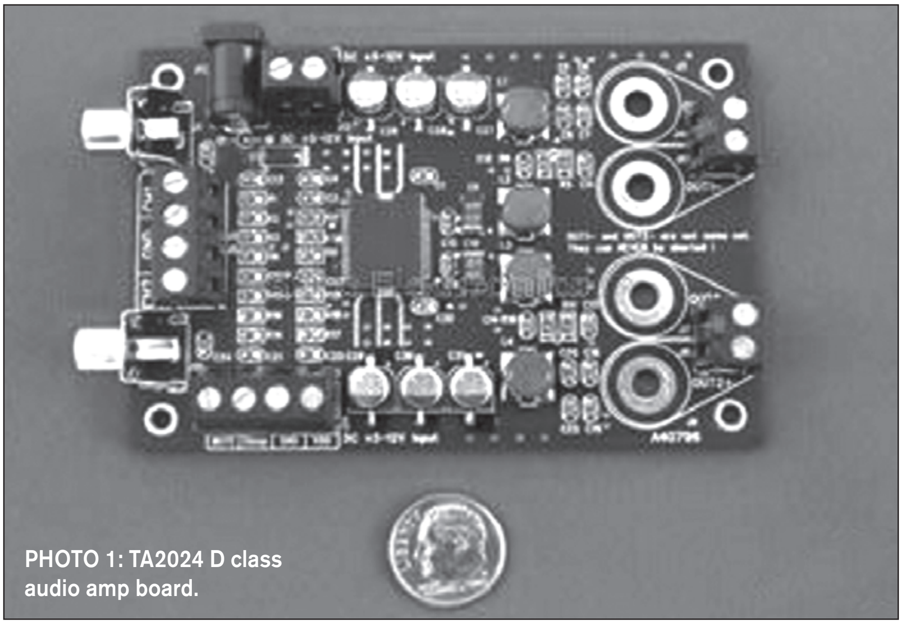


PHOTO 1: TA2024 D class audio amp board.

I ordered a 2 × 15W TA2024 D class Audio Amplifier board (**Photo 1**). Also a friend requested that I build him an outboard amplifier for surround-sound use. I had planned to build a gain clone amp with four channels supplying about 75W per channel in a compact unit. Shure's website showed a 4 × 100W 4Ω amplifier module available for a very reasonable price (read really cheap). They also sold a 24V 12A switching power supply. I combined the order and shipping and waited for the items to arrive

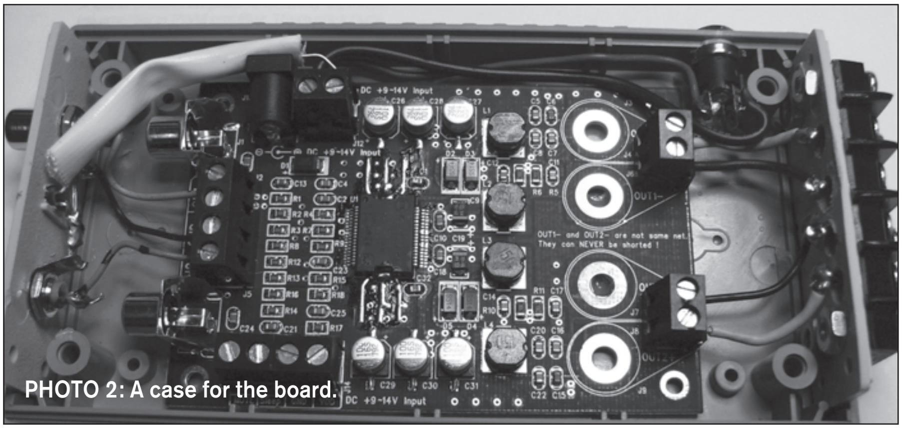


PHOTO 2: A case for the board.

to begin my experiments.

The specs of the 2 × 15W unit are:

10% THD at 15W RMS output into 4Ω
0.1% THD at 10W into RMS 4Ω
0.04% THD at 9W RMS into 4Ω
0.18% IHF-IM at 1W RMS into 4Ω
Efficiency >88% at full power
No heatsink required up to 15W per channel
Turn on and off pop suppression
Short circuit and over temp protection
Operation on 12V for full power output
Weight of complete module is 3.6 oz
Size 96 × 60mm

I sometimes need a small amp to power speakers outdoors at family gatherings, so I built this unit in a very compact plastic case with input and output connections mounted on the case. This module comes with pre-mounted input jacks, as well as a terminal strip for input and output connections, in addition to binding posts (**Photo 2**). I mounted a power jack in the side of the case to allow me to plug the amp into a cigarette lighter for portable use.

I also had a computer 12V 3A switching power supply laying around for use as a 110 supply (**Photo 3**). I connected it to a set of JBL L 100s in my office fed with a Luxman tube preamp, with a Philips CD player and a Luxman FM tuner. The sound was tight with substantial bass using a variety of sources and music. Even turning the volume up substantially resulted in a very loud signal, before clipping set in (I am not really sure it was clipping or just the 10%). I reduced the volume to a comfortable listening level and proceeded to spend a few hours listening while I worked on other projects.

I also evaluated it outdoors driven with an iPod as a music source, and a small set of mini monitors. It turns out the outdoor use was better for this amp. I experienced the same thing when I listen to commercial solid-state amplifiers—listening fatigue. I wouldn't say it sounded bad; it just didn't light me up.

ALTERNATIVE

The next experiment was with the 4 × 100W module

- 4CH D Class amp
- 75W 4CH (THD 1% 4Ω)
- 150W 4CH C 2Ω
- Full bridge output
- 1 Input power
- Free Voltage (6V ~ 26V)
- Protection (thermal, overload)
- Analog signal input
- High Power efficiency up to 90% (2CH OUT at 30W)
- No On/Off Pop Noise (Sleep, Mute)
- Operating Environment Temperature Range: 0~+55 C
- Storage Environment Temperature Range:-25~+85 C

Outline Dimension 5.7 × 5.7″
Item Net Weight 400g/14.3oz

The size of this unit is pretty impressive, only 5.7″ × 5.7″, yet it will provide four channels of 100W (**Photo 4**). The switching power supply I purchased is also pretty compact yet will provide 12A of current at 24V. I played around with several ideas for a case for this amp and decided to make one out of some scrap wood and plexiglass. I have made a number of chassis with this technique and they typically receive a very positive response from the eventual owners. I buy scrap pieces of plexi from a large hardware-type surplus store. **Photo 5** shows the final cabinet.

IMPRESSIONS

So how did it sound? I hooked it up to the same system as before and started working it out. I was struck by how quiet the amp was. There is no perceived sound coming from the speakers. Apparently, the switching power supply noise was effectively filtered out by the output filtering network. I was pretty impressed by the available power. This amp produced very solid bass and seemed very clear in the midrange and treble regions. I listened to the amp for several hours and found the same type of listening fatigue I experienced with the smaller unit.

Not wanting to be completely prejudiced, I asked my buddy Larry to listen

PHOTO 3: Computer switching supply.

to it for a while. As I have mentioned in previous articles, Larry is a McIntosh fanatic and believes their vintage tube equipment is the best sounding equipment you can own. He hooked up the amp to his reference system in place of his MC 60s. I didn't reveal my impressions and let him listen to the unit for a couple of weeks. One day he brought the amp over and told me he didn't like it. Usually Larry is pretty diplomatic about my projects, so I was a little surprised by his assessment.

He thought it had pretty harsh midrange, and while the treble was airy, it wasn't natural. He said the bass was impressive, with several instances of the house vibrating from the music. He listens to many different kinds of music, and tried to use a variety with this amp. He said the one that was the most telling was Alison Krauss, whose vocal range and ability is pretty impressive.

Overall, our impressions were similar. But I work in the medical research area and am skeptical of any trial that isn't double blind and randomized, so a trial of two, with no blinding, is not very impressive.

We both thought that the sound was clean but sterile, the bass solid and tight with mild midrange. Even though each of these modules provides turn-on suppression, they both exhibited some turn-on noise and mild thump. Once powered up, they were extremely quiet, even though they were being powered by switching power supplies.

I am going to attempt to listen to a Bel Canto Evo power amp to see how a very expensive Class D amp really sounds. Most of the sources of information I accessed for this article agree that there can be very impressive Class D amps built with advances in manufacturing and technical development. *aX*

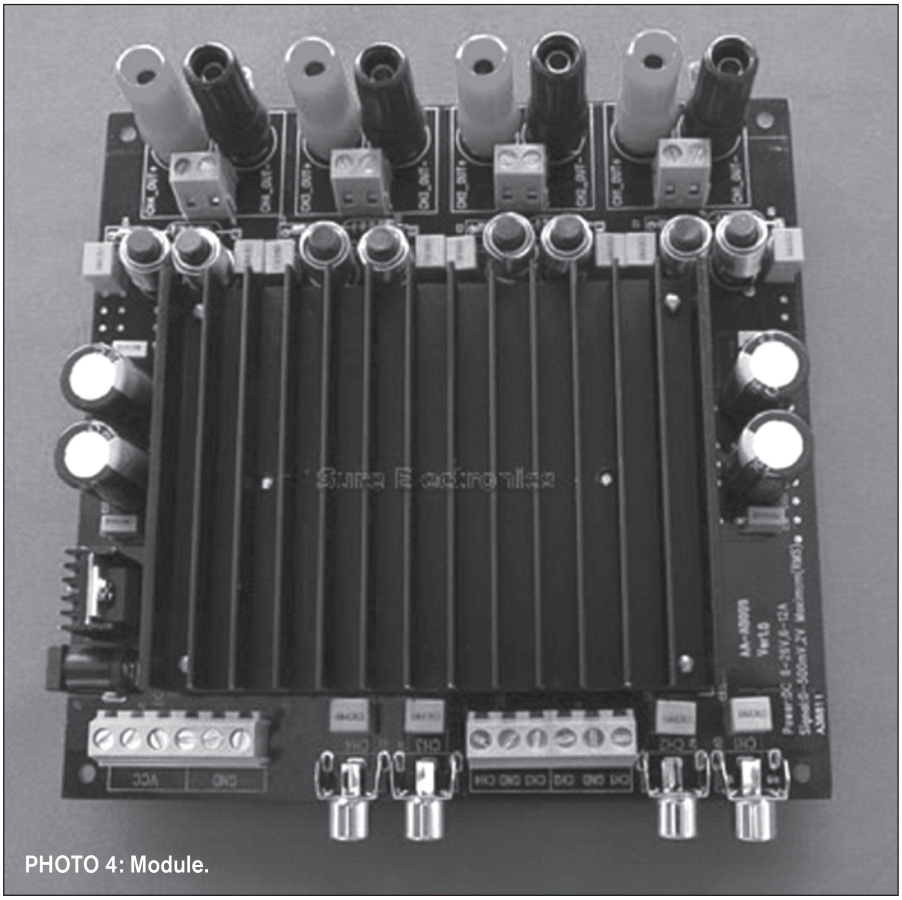


PHOTO 4: Module.

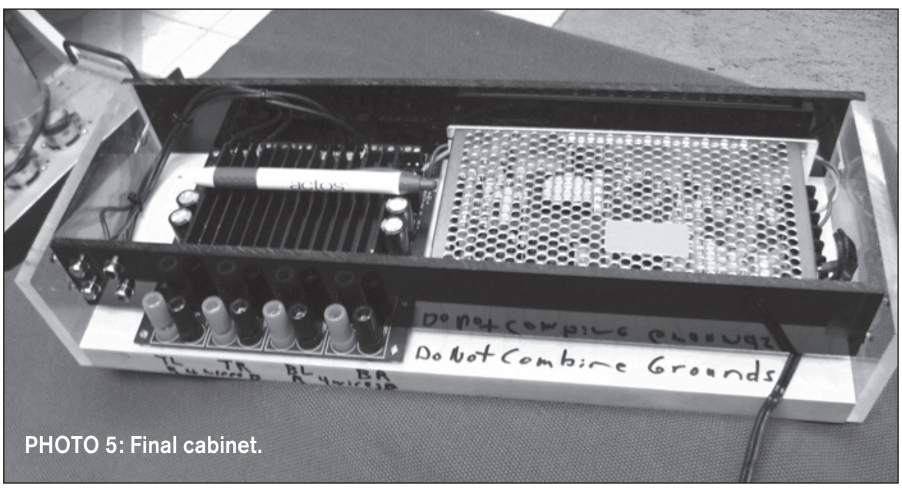


PHOTO 5: Final cabinet.

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By Ed Simon

250W of **Clean AC Power**

Clean power to the people, part 4.

In the first three parts of this series, I looked at why you need to clean up the AC line, three passive ways to do it, and one method reusing a larger power amplifier. In part 4 I look at building smaller amps and using them to produce clean AC power.

I start with a design value of 250W of AC power for a simpler version of an AC 120V supply. I picked this because it is enough to power all of my sources and preamplifier. In addition, transformers of this size are readily available.

For this design you will need an amp that can produce a lot of current because as the voltage is stepped up, the current is stepped down. The amp should be efficient to minimize wasted power.

HEATSINK APPROACH

The first issue that jumps to my mind with any high power amp is heatsinking. I can find power transistors rated for 200W if the actual transistor die is kept at 25° C. The power handling goes to zero watts at a temperature of 150° C.

A quick catalog search shows I can buy a Wakefield heatsink 423K for about $15 to $20. This has a temperature rise of 47° while dissipating 50W. I know from experience that I like to limit my heatsink temperature to a 35° rise. That means that a well-designed material-efficient heatsink will allow a free-air dissipation of 37W.

In the worst-case short circuit I expect my power transistors to dissipate the entire power supply, which can be as high as 600W. For normal use I cannot calculate the exact value because I do not control the reactance of the load. I estimate it to be 200W. If I design my heatsink to handle 200W normal load and add a fan to kick in when the temperature rises, I can get the best of both. A system that is quiet in normal use and is less likely to fail when I am being stupid.

Six of the 423K heatsinks will do. The cost is almost reasonable. I am now about to reveal one of the great secrets of saving money on heatsinks. The loose of pocketbook should read no further.

If you take your shopping cart from the heatsink aisle to the structural aluminum aisle, you will find some great bargains. I can buy 8′ of 4″ × 2″ U channel for under $50! If you choose to add fins, add two 1/8″ thick angles (1 × 1.5 and 2 × 1.5) and keep the bill under $100. All of this gives you 8′ of heatsink! Even better is when you find some used scrap! I got a giant pile of 5″ channel for the asking.

You will need to cut the aluminum, clean it, paint it, and bolt it all together. To cut the aluminum you can use a hacksaw, a saber saw with a metal cutting bit, a band saw, a chop saw, or, my favorite, a powered miter box! I use a standard DeWalt saw with a 12″ blade made for cutting aluminum. I bought them both at Home Depot. (If you do not wear gloves, eye and ear protection while cutting the aluminum, please consider another hobby.)

To prep for paint you can sand it all or use a cleaning solution that contains lye. You used to be able to buy lye in small quantities, but it is usually not available that way anymore. You might read the labels of other cleaners or just use a bit of Drano crystals. Wear gloves and eye protection, using only enough to quickly wet the aluminum. You will see it almost instantly become bright and shiny. Rinse it off immediately and dry it. You have about 15 minutes from when you clean it until paint will not stick to it. As soon as it is dry, paint it with a thin coat of black paint.

If you drill a few holes, you can bolt the angles to the channel for additional radiating surface area. This heatsink is not as efficient per volume as a commercial unit, but on a watt per dollar basis it is unbeatable.

I attached resistors to my structural aluminum heatsink and measured the temperature rise. With an input of 110W a 14″ length of 5″ channel without any extra fins rose by 65° for a rating of 0.6W per degree! Adding a few fins gives me a heatsink for this amp at the bargain price of less than $15.

If you wish to save even more, you might try a visit to an aluminum distributor, which often has a room with scrap for sale to hobbyists; sometimes you may even find actual heatsink leftovers. My local distributor charges me 85 cents per pound for his leftovers. At those prices this becomes a $4.04 heatsink!

HEAT LOADS

The next issue is to actually get the heat into the sink. For small heat loads I like to use the mica washers made for the purpose with a very small amount of heatsink grease. There is a loss of about one-half of a degree per watt for using the insulating washer.

In this case I mount my output transistors directly to an aluminum heat spreader and reduce my insulator heat losses. A small bit of grease helps here also. The spreaders get the most heat possible out of the transistor and then spread this over a larger area in order to reduce the loss through the insulator. This is the same effect as paralleling resistors. Note: The heat spreaders will be at the same voltage as the transistors' cases.

To insulate the spreaders you can buy thermally conducting rubber sheets made for this purpose. See the black sheets below the heat spreaders in **Photo 1**. Digi-key's BER232 is fine for this use. If you wish to save a bit you can even try greased paper!

The rubber sheets are fragile, so you must be sure you have no sharp burrs on your heatsink or spreader that will punch through. I find sanding with 400 grit sandpaper cleans up the burrs. I then wash the material with dishwashing soap and water.

I use nylon standoffs ¼″ in diameter by 1/8″ long in the mounting holes to keep the steel mounting screws insulated. I found the mounting holes in the power transistors were too small for the standard 6-32 machine screws, so I reamed the holes to fit. You may wish to use 4-40 hardware or go metric. After I had mounted everything, I let it sit overnight and then checked for shorts with an ohmmeter (**Photo 2**).

You certainly can try another approach to heatsinking. The nice thing is that if you are conservative in your electrical design, a small increase in the temperature of the heatsink causes a greater increase in its efficiency. Painting the heatsink black almost doubles the power dissipation for a convection-cooled unit. Fan cooling is even better.

TRANSISTOR SELECTION

The next design issue is how many output transistors are required. I like a safety margin on the ratings of my transistors, so at a heatsink temperature of 60° my 200W transistors are good for 150W, and allowing a 1.67 margin still leaves 90W per transistor. That would give die to heatsink losses of another 40 to 50°, which would reduce the power dissipation to 80W (not a bad first guess).

At a worst-case dissipation of 600W, seven and a half output transistors would be fine. I could use four per amplifier module but I will round it up to six, five not really being a choice. A handy doublecheck on power dissipation is the rule of thumb 30W per TO220 case and 60W for T03 or TO264 cases.

I had a number of leftover Hammond 182T12 transformers with two 12V 12.5A secondary windings. If I use these to make a power supply for my high current amp, I can get a target voltage of 16.5V. This could range from 14.75 to 18.1V (diode drops included). At 14.75V, 300W out would require 20A, so I can use two of these transformers either together or as two separate supplies.

I would like to use the cheapest commodity transistors I can for the output stage. Unfortunately, when my associate

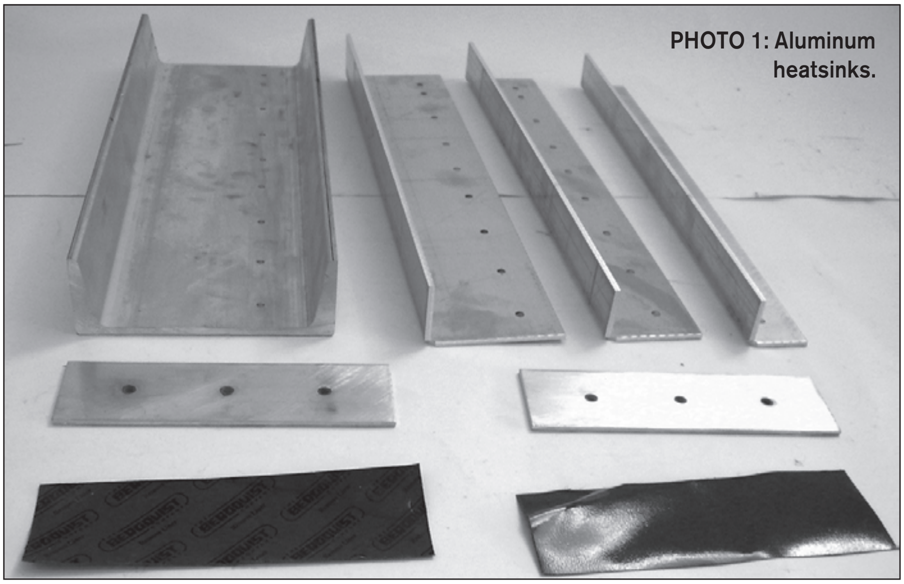
PHOTO 1: Aluminum heatsinks.

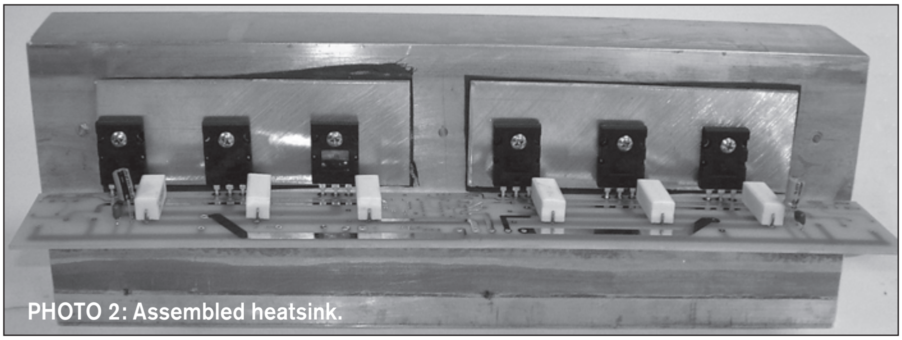
PHOTO 2: Assembled heatsink.

Billy did that he found out that batch-to-batch the transistors had quite different parameters. After many blownup amps, he went to a better grade. So you can learn from the mistakes of others.

About the best bipolar audio power amplifier output transistors are the ON Semiconductor MJL4281A/4302As. These are a bit pricey, so I drop a step to the MJL1302/3281, which costs one-third less at one-eighth less power. They have enough bandwidth to minimize oscillations. The betas are well matched between devices and are reasonably high, allowing an efficient design. I have found it is still best to beta-match sets for the output. For some reason I find the PNP transistors are pickier.

A simple beta test is to use a 12V battery, an ammeter, and a 330Ω ½W resistor. Clamp the transistor to a heatsink and measure the current drawn when you touch the resistor to the base. The transistor will heat up and increase the beta so you will need to get a feel for when to read the meter. The faster you get your reading the better. Expect to see currents from 1 to 5A. I line up the transistors from low to high and then use three in a row for my output stages.

POWER SUPPLY

To keep the design efficient I use one set of supply rails for the output stage and a second higher voltage set for the drivers and gain stages. The higher voltage rails need not have the same current capacity as the output rails.

To get a second higher voltage rail, I could use another transformer. Instead, I wound a few extra turns on my transformer. When I powered up the transformer I found that I get 0.33V from a single turn on the output. I wound two pairs of 12 turns using 18-gauge magnet wire. This gave me two extra secondary windings that I then placed in series with the original secondary windings to get the higher rail voltages (**Fig. 1**).

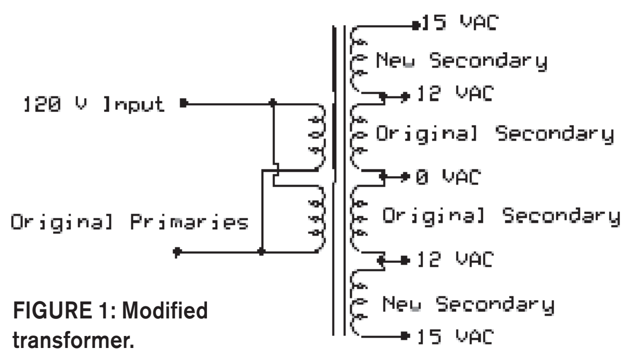

FIGURE 1: Modified transformer.

The rest of the power supply is plain vanilla. I used high-current low-loss Schottky diodes for rectifiers and big capacitors to keep the ripple down. Multiple capacitors on the high current rail are an effective way to get enough filtering.

I selected the values for 5% ripple at full load. It turns out that 39000µF at 25V was the biggest low-cost capacitor, and because the audio does not go through these capacitors I went cheap. I used 22000µF at 35V for the higher rail voltages. I stayed with the same diodes even though the current requirements were much less (**Fig. 2**). I built this supply on a folded piece of 1/8″ aluminum and used the chassis for the heatsink (**Photo 3**).

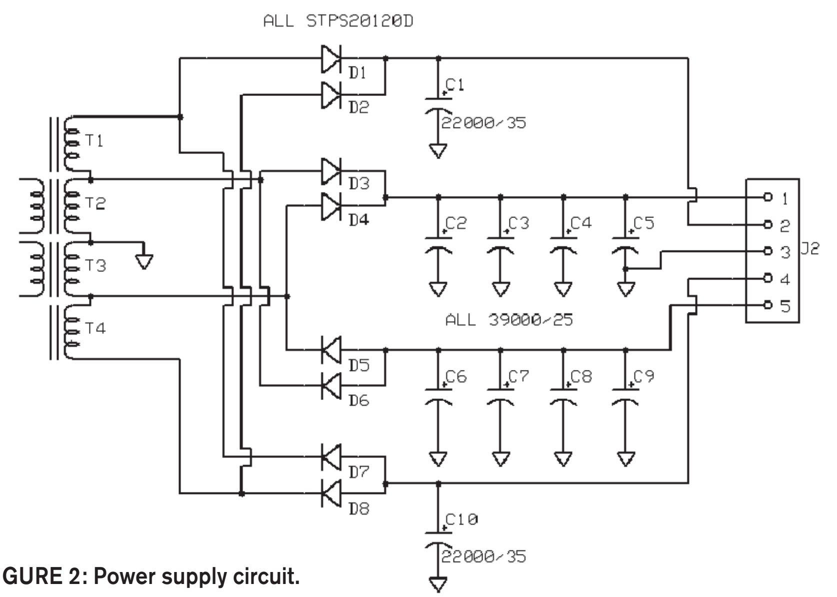

FIGURE 2: Power supply circuit.

There is still some ripple on the rails even with lots of capacitance. This ripple may show up as a slowly varying change in the output voltage. This is due to the minor difference in the AC line frequency and the oscillator. For example, if the oscillator is actually 60.05Hz, then you would see the effect of the power supply ripple over a 20-second cycle in the output voltage.

I could use an electronic regulator to fix this. That may reduce the efficiency, so you must allow for a 10% low line condition in addition to the ripple when you calculate the available voltage from the supply. So 12V AC would become 17V DC less 10% for low line, less ripple of 1.2V and a diode drop. Thus I can expect a rail voltage of 13.6V. You will have some additional losses from the transistors and the emitter resistors. These should be about 1.2V. An output voltage of 12.4V peak would allow an AC RMS voltage of 8.8V. Thus you could expect efficiency to be no greater than 54%!

A bridged amplifier configuration helps some of these problems. It makes it easy to use two power supplies. Each supply now needs to provide only one half of the current because you now have twice the voltage available. This also lowers the ripple. You now have twice the output transistors, so fewer heat concerns. You can use a 12 or 15V to 120V transformer, which is a bit more common. Your line voltage and desire for efficiency should shape your choice.

Keep in mind these transformers are designed for stepdown use, not step-up. A cheaply made or well-designed transformer (depends on your point of view) may have some characteristics that become a problem when you reverse it. The typical issue is at startup if the transformer saturates and how this affects the rest of the circuitry. To prevent

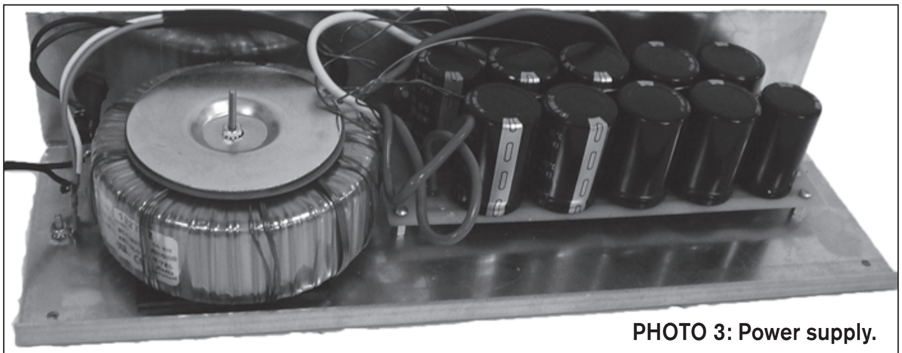

PHOTO 3: Power supply.

this, place a 2Ω resistor in series with the output; after startup this is switched out.

DESIGN TWEAKS

Now that you have the power stages, you should look at driving them. The two power rail system allows a higher voltage for the stages that drive the output stage. That way there is a minimum of excess voltage across the output transistors. When you produce 250W at 15V to drive the transformer, you require 16.7A RMS. That is a peak current of 24A. The output transistors have a specified minimum beta of 45 at 10A.

Using a safety factor, I will design for a gain of 30. Therefore, you will need to provide 800mA of current to drive the output. The driver transistor will need to dissipate a worst-case heat load of 25V high rail voltage at 800mA or 20W. This, of course, would only be if an output failed. Under normal use it would be closer to 2W. You already have a giant heatsink, so mounting this transistor to the heatsink should allow it to survive even under failure conditions.

This driver transistor would require a current of about 15mA to drive it. This is a bit more than I would like to see, so I will use a Darlington configuration here.

You could use a standard Darlington driver for this design, but at only 24V of supply rail a double diode drop loses 5% of the voltage. That is why I chose to use a PNP/NPN conjugate Darlington. There is a cost to using so much amplification, though. That is the risk of oscillation. Adding emitter resistors limits the gain of each stage reducing the risk of oscillation. Good layout is still important.

In theory this amp could run as a class B design. In a class B there is not steady current in the output transistors when the output is zero volts. For audio use these designs are biased so there is always some current flowing from the positive rail to the negative rail through the outputs. This class AB design lowers distortion.

Here you would have a dead zone of four base emitter drops if you did not use a bias circuit. I have found that when the voltage amp tries to compensate for this, oscillation results, so I have included a bias circuit. This is set for just enough voltage across it to start some current flowing through the outputs. I have used the same transistor as the driver type to try for some thermal matching.

Even with the input to the Darlington running cold there is still significant mistracking. This means the bias must be adjusted after the circuit is warmed up and at operating temperature. You can adjust the bias by monitoring the input current and turning it up until you see the current rise and then backing it off a hair to stay as close to class B as you can. Remember there are two amplifiers and two bias adjustments.

The driver transistors still need a voltage source. I decided to use an op amp for the front end. It is a simple way to get a lot of gain in a few parts. To keep it stable I use an external compensation network at the op amp. The particular op amp can swing more positive than negative, which is why I can connect it above the center of the bias point. I picked an LM201 because it can run from 22V rails giving the headroom you need to drive the other stages (**Fig. 3**).

The oscillator to drive these amplifiers can be any good stable 60Hz oscillator. I used the transversal filter oscillator from part 3.

There is one other small circuit I used to build this version of the clean AC power supply—the fan controller. I used

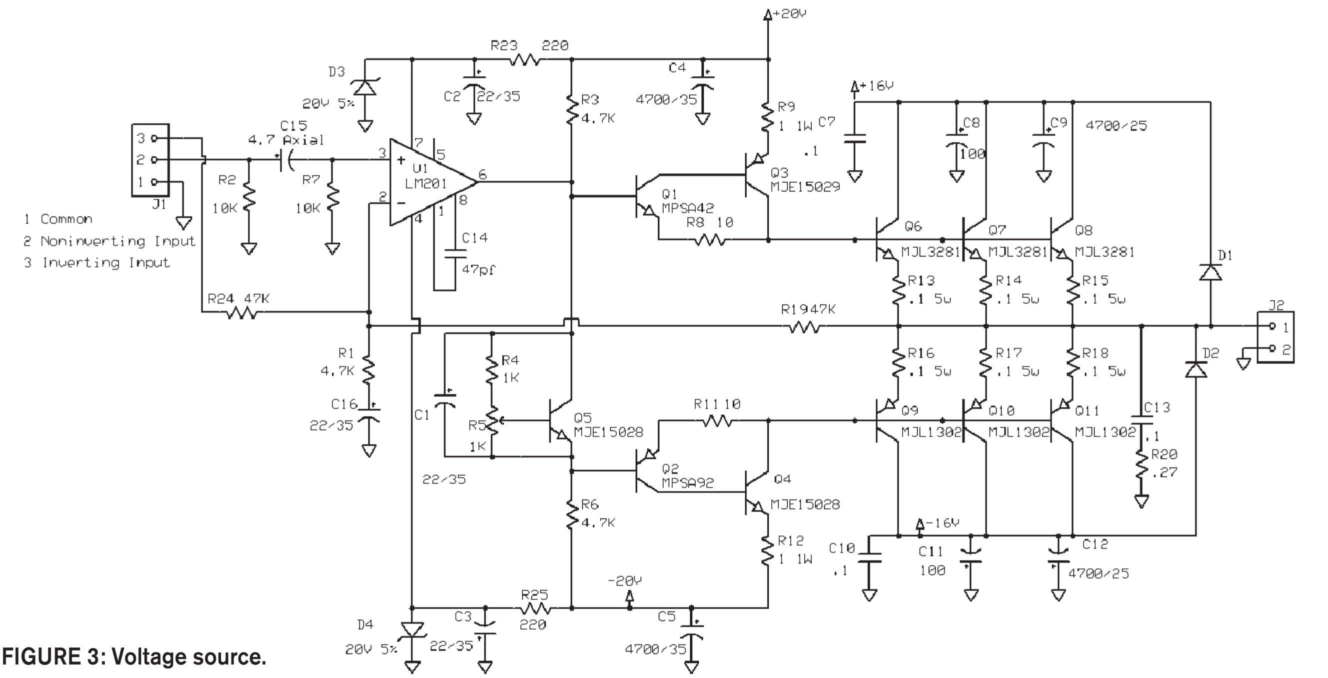

FIGURE 3: Voltage source.

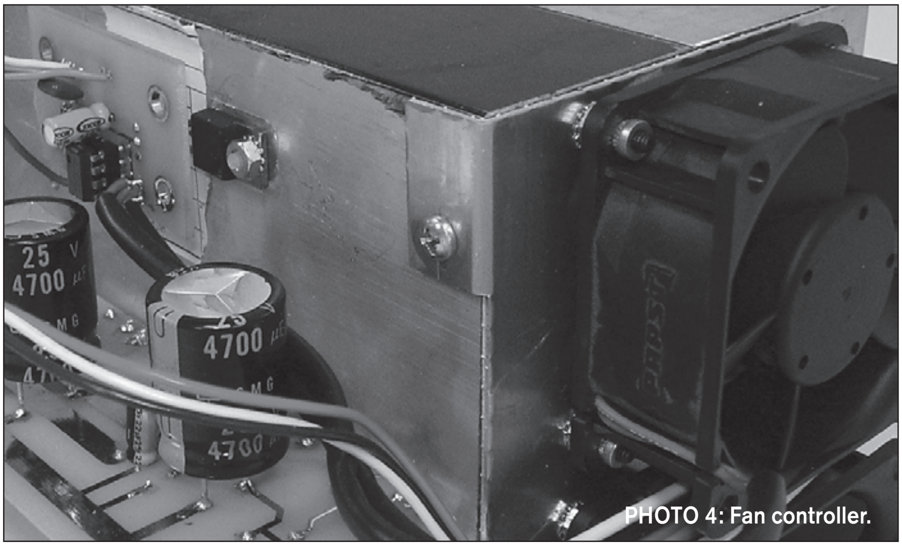
PHOTO 4: Fan controller.

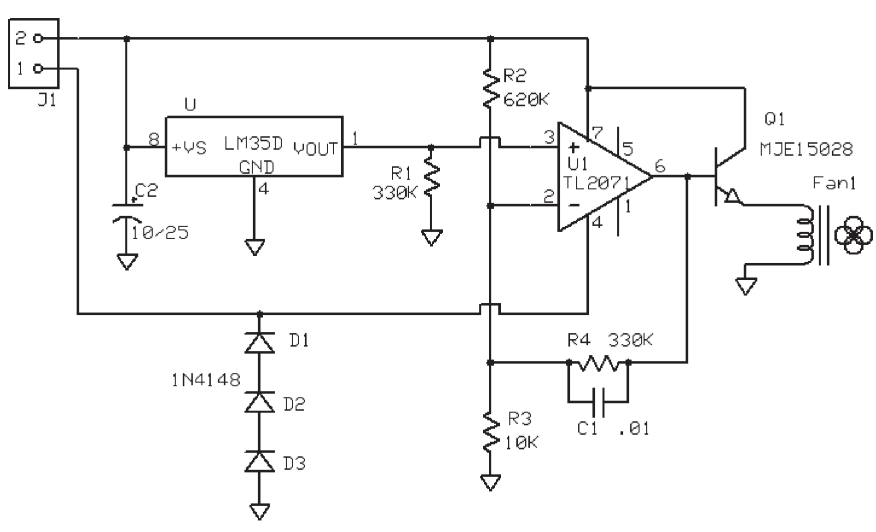

FIGURE 4: Fan controller circuit.

two small 12V fans with this circuit. The LM35 temperature sensor gives a very precise output voltage related to temperature. This is compared to a reference voltage provided by R2 from the power supply.

As the temperature rises, the op amp will turn on the fans. For my fans the transistor did not require a heatsink. The diodes allow the op amp to work close to the ground rail. If you use a true rail-to-rail op amp you can omit them (**Photo 4**).

You should mount the LM35 in a good place to sense the temperature. I super-glued it directly to an output transistor case. I then mounted the two fans between the two heatsinks, giving me a nice wind tunnel. The fans barely come on during normal use.

TESTING

The complete 120V AC supply is almost done. Two DC power supplies, one oscillator, two amps, a fan controller, and one stepup transformer. The oscillator drives the first amp, and the output of this amp drives the inverting input of the second amp. Both amps now drive the output transformer. The return path for each amplifier output is through the other, so a good ground connection between the two power supplies is quite important. If the power supplies are matched, you can also strap the output supply rails to each other (**Fig. 5**).

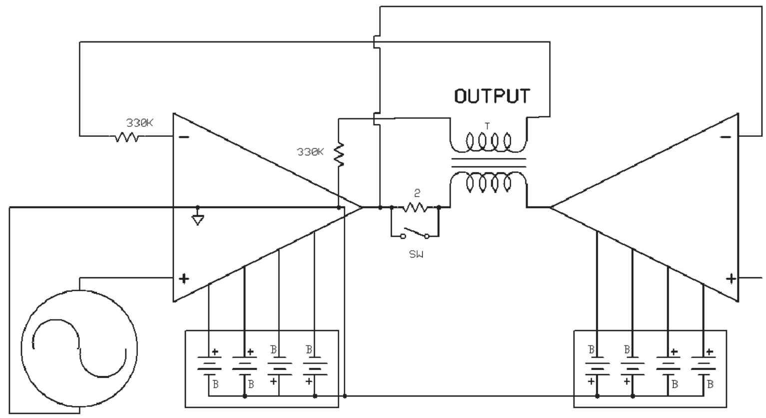

FIGURE 5: Supply layout.

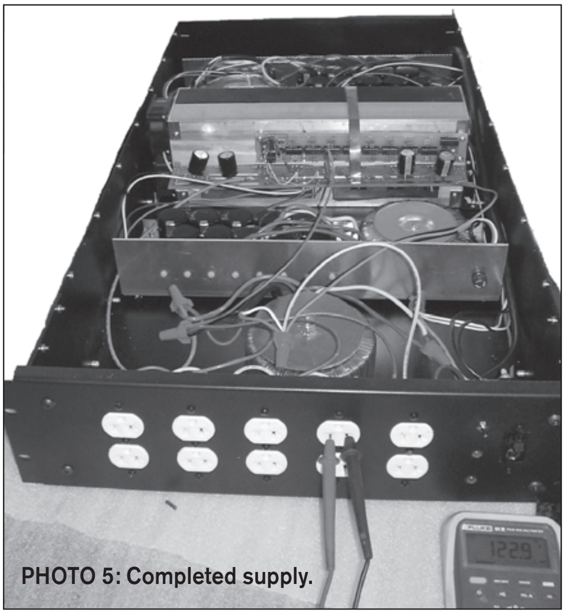

PHOTO 5: Completed supply.

You should test the circuits without a load through a current-limiting lightbulb box until you are sure it works. You will find the 2Ω resistor is required at startup for most transformers. Check the oscillator for a good signal and then see whether both amps have a clean output without distortion, oscillation, or ripple.

Once the AC supply works during testing without feedback, connect the feedback. If it goes crazy, you most likely have the transformer providing 180° of phase shift. Just switch the leads to the feedback resistors. Now the output voltage will drop and you can reset the oscillator level for 120V.

Once everything checks out, try the circuit on something simple such as a few light bulbs. When you are confident that all works well, you can connect it to your system. There are two power switches: one turns on the input power to the circuit, the other bypasses the startup resistor. Turn the input switch on first, then the bypass switch. You may wish to have your equipment turned on before you engage the bypass switch to minimize current surges while your amplifier power supply capacitors charge. Another possibility is to use a relay to bypass the resistor.

I did not determine the output impedance of the finished product. If you want to increase the gain of the amps set by R19, decrease the 330K feedback resistors or both; you should be able to get a stiffer supply. This was not important for my use. If you decide to build a supply such as this, you will have a less than 50% efficient method of turning 120V 60 cycles into 120V 60 cycles (**Photo 5**).

A better way may be to use a smaller amp in series with a moderately cleaned up AC line. This approach avoids reinventing the wheel. You use less power and just fix what is wrong. It is more efficient, costs less, and, although actually quite complicated, more of a "you can do this better than the commercial product approach."

You can use the same amps as used in the smaller supply, but add just a few more parts and get a regulated clean AC supply of 20A! I will look at that in part 5. *aX* product review | By G.R. Koonce and R.O. Wright, Jr.

Dayton Audio
WT3 Woofer Tester

The Dayton Audio WT3 Woofer Tester is available from Parts Express, 725 Pleasant Valley Drive, Springboro, OH 45066-1158. The catalog price is $99.88[1]. See daytonaudio.com and parts-express.com. No serial number was found on the units.

INTRODUCTION (BY GRK)

When I agreed to do an equipment review on the WT3, I recalled that many times coauthor Bob Wright had purchased a WT3, so we agreed to do the review together. Bob has considerable experience in measuring V_{as} via the added-mass method and will concentrate on that topic. This gives results from two WT3 units and two users—a more balanced review because I tend to be a nit-picker.

HISTORY

The "Woofer Tester" name has been applied to several earlier and current devices. In 1994 Dickason[2] reviewed one sold by C&S Audio Labs (Model DSP-126), and in 1997 Weems[3] reviewed a revised version (Version 4.4) sold by Parts Express. These AC powered devices both used DOS-based software. Currently a variety of devices with this name, having additional options and higher cost, are available.

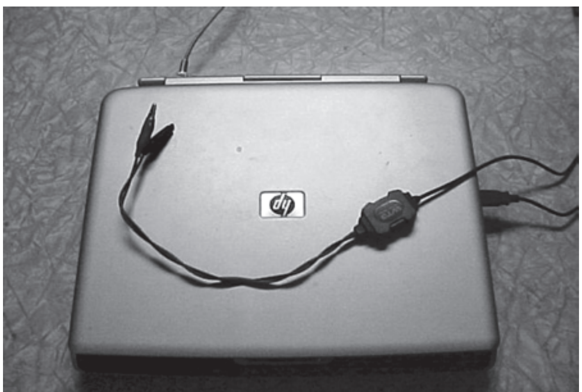

PHOTO 1: WT3 unit setting on a laptop computer.

WHAT YOU GET (BY GRK)

The WT3 arrives in a box about the size used for handheld DVMs. Inside are the WT3 tester, a calibration resistor, the software CD, and an instruction sheet covering how to get started. The WT3 itself is a rather small unit with built-in cables (**Photo 1**). The computer side has a 4′ cable with a USB connector, while the test side is a short pair of red and black wires with alligator clips. The WT3 is self-powered via the USB cable with a blue LED indicating power on.

There were four files on the CD:

1. License.Txt—The licensing agreement stating what you can and cannot do with the software.
2. Readme.Txt—Tells where to get updates and current information. An item I found confusing is this document lists the operating system requirement as Windows 98SE, while other places list Windows 98. If running early Windows 98, you should check with Parts Express before ordering.
3. WT3 Quick Start.Pdf—A copy of the two pages on "Quick Start for the WT3 Woofer Tester" that you can view with Adobe software.
4. WT3_setup.Exe—The program that contains and installs the WT3 software.

SYSTEM REQUIREMENTS

To operate the WT3 you need an IBM PC or compatible computer with the following characteristics:

1. 64MB memory RAM or more.
2. 500MHz Pentium III processor or higher.
3. A USB port, USB 2.0/1.1 compliant.
4. MS Windows 98(SE), Me, NT, 2000, XP, or Vista.

GRK operated the WT3 from an HP Pavilion zv6000 laptop computer, 1.99GHz processor, 512MB of RAM, and MS Windows XP Home Edition version 2002. ROW's computer is a custom-built 64-bit CAD/graphics computer using MS Windows XP as an operating system.

WHAT WILL IT DO?

While you may use the WT3 to measure impedance of any passive electrical circuit, we are interested in its application to speaker work. The WT3 directly measures the input impedance of a driver or a speaker system. With the WT3 you may do the following:

1. Directly measure the f_s and Qs of a driver.
2. Directly measure the driver V_{as} by any of three techniques.
3. Directly measure inductors of the size used in crossover work.
4. Develop driver zobels and other impedance correction networks for crossover design.
5. Help tune a vented-box design by locating the impedance minimum frequency which approximates the box tuned frequency.
6. With the proper modeling or previous-test results, you can verify the proper wiring of crossover networks.
7. With additional software you can calculate how severely a speaker system stresses an amplifier.
8. With simple software you can predict the bass response curve for a closed-box speaker system.
9. With additional software you can design all the standard bass enclosure formats.

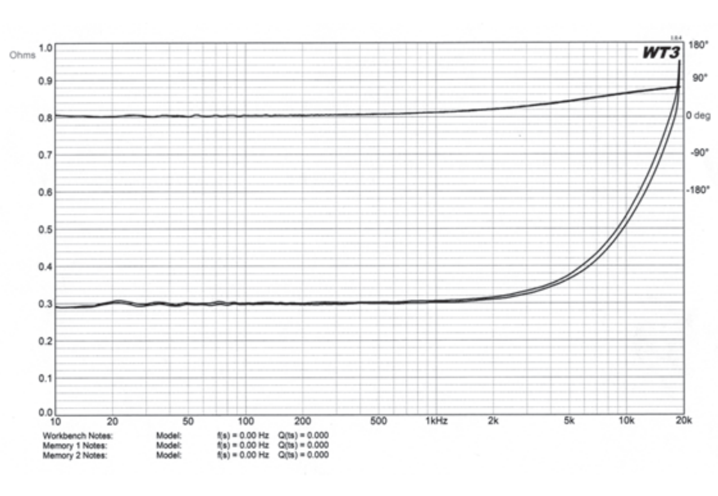

FIGURE 1: Test Lead Calibration with and without leads twisted.

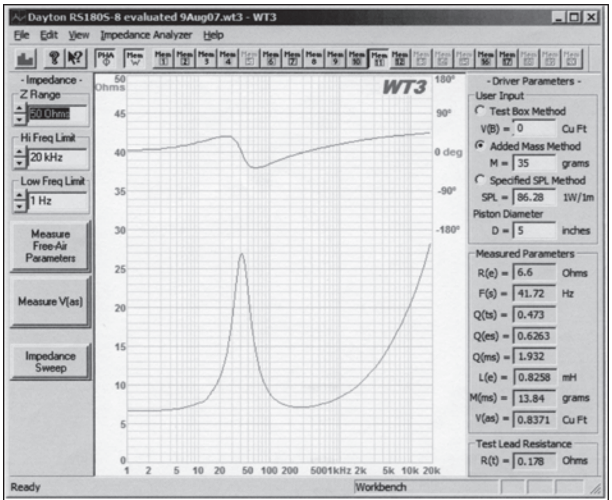

FIGURE 2: WT3 main screen.

GETTING STARTED (BY GRK)

The "Quick Start..." instructions tell you how to install the WT3 program and get the unit running. For me installation went without a problem. Once installation is accomplished, you do two calibration procedures, as noted in the instructions. This calibrates the WT3 with a precision 1kΩ resistor and establishes the resistance of the test leads.

When running the test lead calibration, I noted that the plotted lead "resistance" showed a rising magnitude above 2kHz. The rising phase shift indicates this is due to inductance, which is related to loop area, so I repeated the calibration with the two leads mildly twisted together. **Figure 1** shows that this helps slightly, and for the following testing I kept the leads twisted.

OPTIONS (BY GRK)

When I first started the WT3, I did not like the dark background on the plots (a nit-pick). It made the curves hard to see. After viewing the information in the help section, I found I could change the plot background color. The program has many options of what you can display and print, and it is recommended you study the help files once you have finished your initial playing! The program will print the help information, so I did this for off-line study.

Figure 2 shows the very busy main screen for the WT3 software. It is in color, which helps a lot. You need some study of the help information to really understand program operation. Once you have learned how, the operation is simple and straightforward.

The program has multiple file options as follows:

1. It will save and recall project files that save what you have committed to the 20 available memories. This allows retrieval of information from saved testing should some question arise.
2. It will save and recall impedance curves to/from export files. It offers the option of files with the extension of .TXT or .ZMA. These are both text files and look identical. Unfortunately, the .ZMA files do not match the original format developed by Liberty Instruments for the IMP tester. While certain software may properly read the WT3 .ZMA files, the post-processing programs I had written would not. They objected to the long text header before the actual data. If you have this

problem, you can use a text editor to put the WT3 .ZMA files into a format your software can read. The quickest solution was for me to write a filter program that takes in the WT3 .ZMA file and writes it out in the classic format of a Liberty Instruments' .ZMA file.

3. It will save and recall driver files (extension .DVR) that are common with the format used by the WinSpeakerz program.

The one missing file option seems to be the ability to output the graphs to files as pictures. This would be useful when using the WT3 to prepare material for publication, such as this review.

The program also offers several options relative to the printed output. You should check the options under both the File -> Print Setup... and the Edit -> Preferences menus.

REPEATABILITY (BY GRK)

One of my first applications for the WT3 was to measure the impedance of a baffle mounted woofer several times. The results for ten tries are shown in **Table 1**.

I think this repeatability is very good. I no longer beat the box design out to several decimal places because the parameters will vary with temperature and humidity, and so on, and the box response changes slowly with minor parameter changes. The WT3 is so quick you should run the free-air test on a driver a few times to verify that the parameters you are going to use for the V_{as} extraction are a typical set.

COMPARISON WITH LIBERTY INSTRUMENTS' AUDIOSUITE (BY GRK)

In an attempt to determine the accuracy of the WT3, I free-air tested the three drivers in **Table 2** with both the WT3 and Audiosuite. **Figure 3** shows the comparison results for the woofer mounted on a small baffle. The two curves match extremely well; only near the resonance and around 20kHz can you identify that there are two curves on the plot. The same comparison is shown for the mid/bass driver in **Fig. 4** with, again, a near perfect match. **Figure 5** is the comparison for the tweeter also showing very good agreement. The slight differences seen at high frequency may be due in part to the WT3's lead inductance as discussed earlier.

TABLE 1: MEASUREMENT OF REPEATABILITY OF THE WT3

Try	R_e	f_s	Q_{ms}	Q_{es}	Q_{ts}	L_e
1	5.95	49.8	1.66	0.62	0.45	0.59
2	5.96	49.8	1.66	0.62	0.45	0.59
3	5.98	49.8	1.66	0.63	0.46	0.59
4	5.93	49.8	1.66	0.62	0.45	0.59
5	5.95	49.8	1.66	0.62	0.45	0.59
6	5.98	49.8	1.66	0.62	0.45	0.59
7	5.96	49.1	1.63	0.62	0.45	0.59
8	5.97	49.1	1.64	0.62	0.45	0.59
9	5.96	49.8	1.65	0.62	0.45	0.59
10	5.95	49.8	1.65	0.62	0.45	0.59

TABLE 2: THREE TEST DRIVERS USED BY GRK (ALL NEW WITH NO BREAK-IN)

Woofer	Vifa* P17WJ00-08 6½″
Mid/Bass	Vifa* MG10SD09-08 4″
Tweeter	Vifa* XT25TG30-04 1″ dual concentric ring

*Note: Tymphany Peerless now calls these V-Line drivers

I had both units compute the driver T/S parameters. Only the woofer had V_{as} extracted via the closed-box method. **Table 3** compares the test results obtained with the catalog[4] data.

The two measurement systems show reasonable agreement with each other but not always with the catalog data. With two different test instruments some variation is to be expected. For the woofer, either measured set of parameters should result in a reasonable enclosure design. It might not be the same size as one based on the catalog data.

For the mid/bass and tweeter the measured data is in better agreement with the catalog data. These drivers were all new and had no break-in. My experience is modern drivers don't change as much with break-in as we saw in the past.

The WT3 allows you to extract the driver V_{as} the "easy way," by specifying the driver SPL in dB/W/M. This technique must be used very carefully. First, some manufacturers will specify the SPL in dB/2.83V/M. If your driver is truly 8Ω, then the two specifications are the same, because 2.83V RMS will put 1W into an 8Ω load. If the driver is other than 8Ω, you must correct the dB/2.83V/M rating according to the following:

$$dB/W/M = dB/2.83V/M + 10 * \mathrm{Log}_{10}(R_n/8)$$—where R_n is the nominal driver impedance

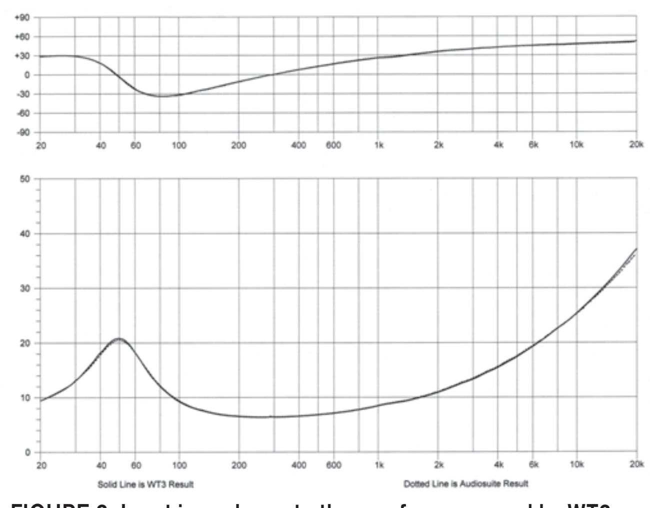

FIGURE 3: Input impedance to the woofer measured by WT3 and Audiosuite.

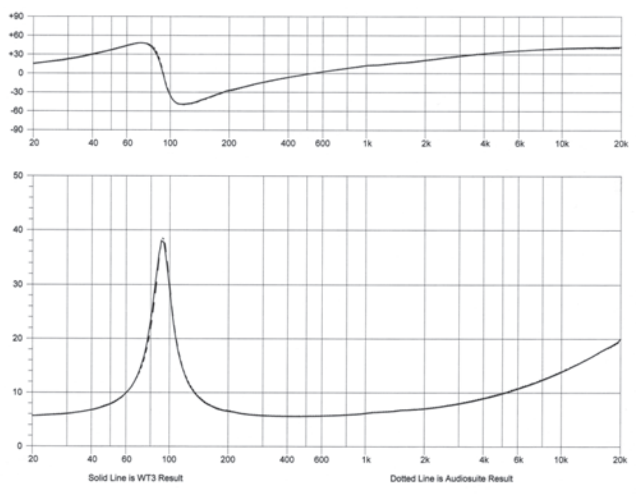

FIGURE 4: Input impedance to the Mid/Bass measured by WT3 and Audiosuite.

TABLE 3: COMPARISON OF WT3 AND AUDIOSUITE RESULTS

	6½″ Woofer			4″ Mid/Bass			Tweeter			
	WT3	AuSw	Cat.	WT3	AuSw	Cat.	WT3	AuSw	Cat.	Units
R_e	6.01	5.97	5.8	5.42	5.23	5.5	2.93	3.08	3.0	Ohms
f_s	49.8	48.6	37	91.5	92.0	81	513	513	500	Hz
Q_{ms}	1.56	1.40	1.55	5.34	5.22	3.40	2.84	2.74	2.5	
Q_{es}	0.61	0.57	0.45	0.88	0.82	0.73	0.62	0.63	0.71	
Q_{ts}	0.45	0.41	0.35	0.76	0.71	0.60	0.51	0.51	0.55	
V_{as}	0.45	0.43	1.23			0.09				Cu. Ft.
L_e	0.59		0.55	0.22		0.33				mH
SPL	85.8	85.7	88			85			88.5	dB/W/M

For example, a 4Ω driver with a 91dB/2.83V/M SPL would add 10 * Log_{10}(0.5) or -3dB yielding SPL = 88dB/W/M. This makes sense because 2.83V RMS would deliver 2W to 4Ω, which is 3dB higher than 1W.

I had the WT3 extract the V_{as} in cubic feet for the woofer and mid/bass drivers based on the catalog SPL (**Table 4**). The value for the woofer does not agree well with the closed-box measured value of 0.43-0.45 or the catalog value of 1.23. The mid/bass result of 0.079 is much closer to the catalog value of 0.09ft^3. If the free-air f_s and Qs reported by WT3 agree well with the catalog values, then this technique is probably safe. If those values do not agree with catalog data, then I would not use the SPL approach to predict V_{as}. If the catalog f_s and Qs agree with your driver, then you might just as well use the catalog V_{as} value.

TABLE 4: V_{AS} EXTRACTION BASED ON SPL

Driver	SPL	V_{as} - Cu. Ft.
Woofer	88	0.74
Mid/Bass	85	0.079

MEASUREMENT OF DRIVER V_{AS} VIA THE ADDED-MASS TECHNIQUE (BY ROW)

In the WT3 program under "Driver Parameters," the second of three methods provided for in the program for determining the driver characteristics is named the "Added Mass Method." This method uses two physical parameters to generate the basic outputs of the program. These are (1) the effective piston diameter designated "D" in the program input and (2) the added mass designated "M" in the program input. The speakers used in this test were not "broken in." It has been my experience that the T/S values do not change significantly under these conditions.

It is very important to get a very accurate effective or active cone diameter and/or equivalent cone diameter. It is generally accepted that the mean piston diameter includes one-half of the speaker surround. This measurement must be done as accurately as possible, because the area of the driver cone is a "power" function in the WT3 program. Any inaccurate input to the program will make a more than proportionate difference in the results generated. **Figures 6-8** give real and equivalent mean diameters of the most common driver cone shapes found on the market today.

The mass to be added to the speaker cone must be weighed very accurately to obtain good results. It, too, is a "power" function in the WT3 program equation and must be done quite accurately.

When I was searching for a suitable added mass for the driver, I found that "poster caulk" obtainable at any office supply store turned out to be an ideal choice. It is low-cost, easily obtainable, does not stick to the driver cone, and stores well in a sealed container. The required added mass is almost always 200 grams or less (1 gram = 0.035274 ounces).

To accurately weigh the added mass, I found in my research that there are scales on the open market referred to as "pocket scales" that have a weighing capacity of 0 to approximately 200 or 400 grams and are accurate to within 0.1 grams (0.1 gram = 0.00022046 pounds). This accuracy is excellent for driver measurement work. The cost of these scales is approximately $25.

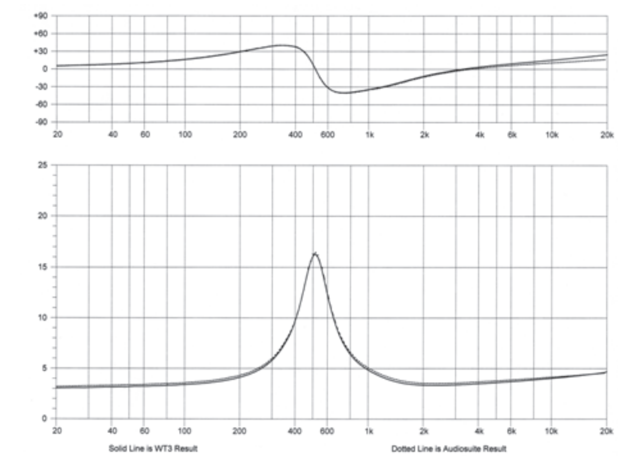

FIGURE 5: Input impedance to the tweeter measured by WT3 and Audiosuite.

When executing this method shape the poster caulk into a "donut" and apply it to the driver cone. This allows the added mass weight to be evenly spread over the surface of the cone, emulating a balanced cone. The amount of weight to be added is determined by the WT3 program itself. However, there are some suggestions in the help file of the WT3 program. The program will analyze the amount of weight added to the cone and determine whether it is adequate to properly calculate the driver parameters and, when necessary, instruct the user as to what remedial action should be taken. When all of the input parameters have been satisfied, the program will produce a driver parameter datasheet.

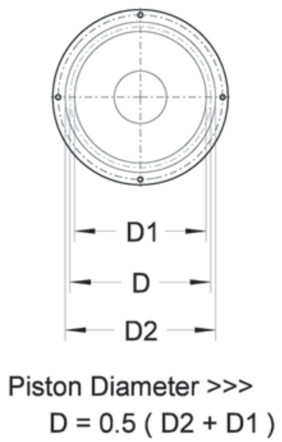

FIGURE 6: Effective piston diameter calculation for a circular driver.

The drivers selected for testing were two 8″ woofers and two 6½″ woofers. Based on a recommendation by George Augspurger, we tested each woofer with three different values of test weight, then averaged the V_{as} values. The results of these tests are listed in **Table 5**.

In the 6½″ unit 1 of the woofer testing, the V_{as} variations from three different test weight values were 0.958 (44.4g), 0.901 (59.5g), and 0.808 (74.7g), with the average = 0.889 the others are not listed.

ZOBEL AND OTHER COMPENSATION NETWORK DEVELOPMENT (BY GRK)

The WT3 is ideal for developing zobels

TABLE 5: RESULTS FOR DAYTON DC160S-8 6½" WOOFER AND DAYTON DC200-8" WOOFER

	6½″		Catalog	8″		Catalog	
	Unit 1	Unit 2	Data	Unit 1	Unit 2	Data	Units
f_s	31.6	31.6	33	31.6	30.3	29	Hz
Q_{ms}	2.04	2.28	3.89	2.23	1.93	3.23	
Q_{es}	0.34	0.33	0.34	0.53	0.47	0.44	
Q_{ts}	0.29	0.29	0.32	0.43	0.38	0.38	
V_{as}	0.89	0.92	0.78	1.28	1.31	2.04	Cu. Ft.
L_e	1.54	1.49	2.40	2.08	1.86	1.60	mH
SPL	86.0	86.0	87	84.9	84.9	88	dB/W/M

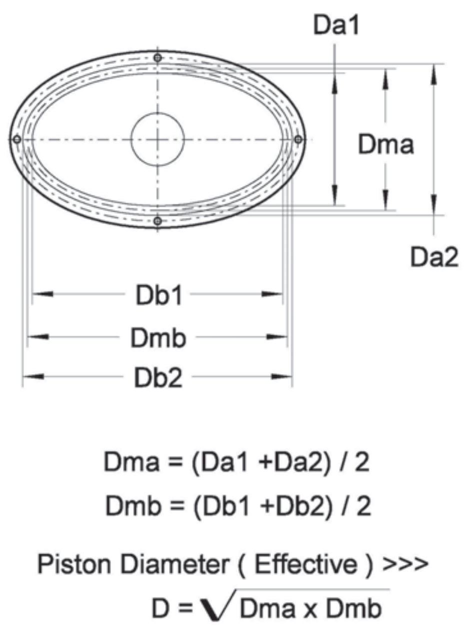

FIGURE 7: Effective piston diameter calculation for ellipsoidal driver.

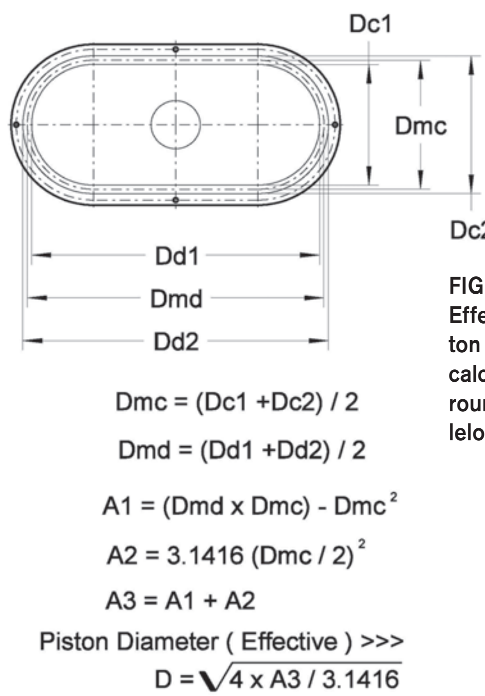

FIGURE 8: Effective piston diameter calculation for rounded parallelogram driver.

and other impedance compensating networks. **Figure 9** shows the free-air impedance into the test woofer with and without a simple R-C zobel as printed by the WT3. It is easy to play with the zobel values and have the WT3 retest because of its speed.

Last summer I was developing a system using a first-order crossover with a tweeter having a very high impedance peak at resonance. This was causing trouble with the low slope of the first-order high-pass network. **Figure 10** shows the bare tweeter impedance and the impedance with a zobel and series L-R-C network across the tweeter to tame the resonance peak. Note that not only do these networks improve the magnitude, but they also improve the phase angle.

My experience is that correcting such effects by networks shunted across the driver results in a better-sounding system than doing it with networks placed in series with the driver. My rule for crossover design is to keep the number of components in series with the driver to a minimum. The WT3 is ideal for the development of such shunt networks.

A corollary application of the WT3 is in verifying that you have connected the components in a crossover correctly. If you know, from previously testing or modeling, what the impedance into the crossover should be, then the WT3 can quickly verify you are still showing that impedance.

Last summer I was simultaneously working with three pairs of breadboard three-way crossovers. This involved making changes in them and trying them on speaker systems. The WT3 would have been ideal in verifying I had the pairs wired the same and that I had not mistakenly connected the woofers where the tweeter should have gone. Using the WT3 would have been a lot safer than firing up the system to verify everything was properly connected.

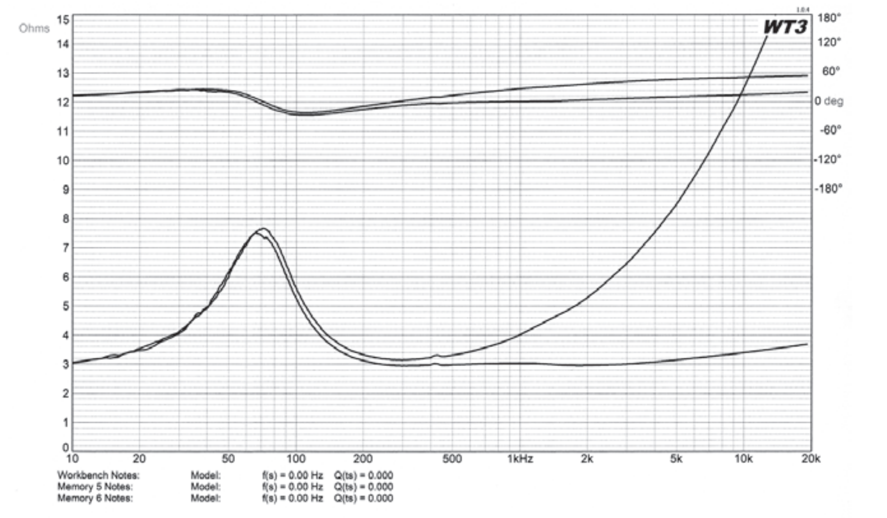

FIGURE 9: Input impedance to woofer with and without zobel.

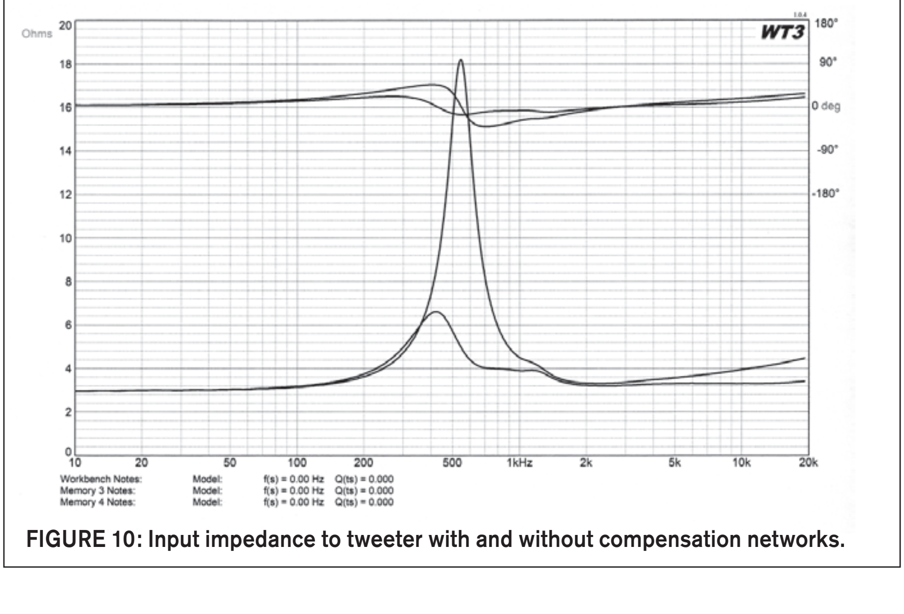

FIGURE 10: Input impedance to tweeter with and without compensation networks.

AMPLIFIER PEAK STRESS LOADING (BY GRK)

Howard[5] and others have investigated how severe a speaker system is in terms of the peak dissipation it causes inside the amplifier at each frequency. If you have WT3 measure and export the input impedance to a speaker system, you can investigate this effect. **Figure 11** shows the input impedance into a three-way, five-driver system as measured by the WT3.

Using the techniques of Howard, **Fig. 12** shows the Peak Amplifier Stress Ratio and the EPDR for this system. The Peak Amplifier Stress Ratio shows the peak amplifier dissipation versus frequency relative to driving a resistor equal to the system's nominal impedance, which is 4Ω in this case. Around 2kHz the peak stress

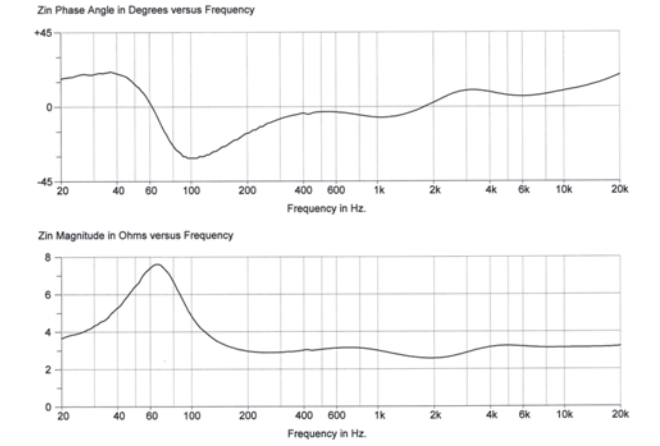

FIGURE 11: Input impedance to five-driver three-way speaker system.

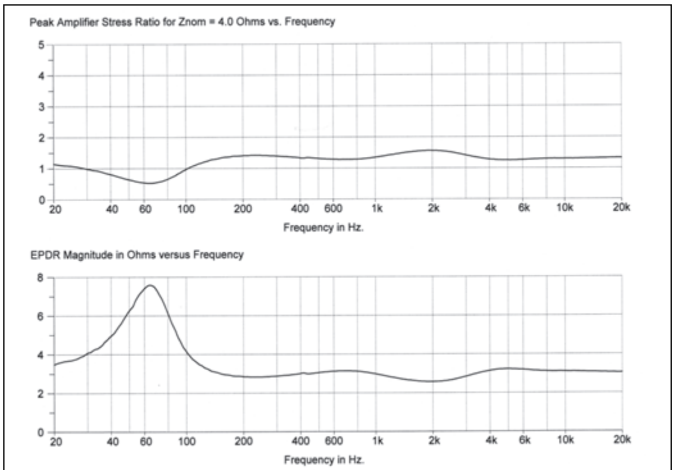

FIGURE 12: Peak amplifier stress ratio and EPDR for five-driver three-way system.

is 1.5 times what the amplifier would see driving a 4Ω resistor. The EPDR is the Equivalent Peak Dissipation Resistance and is the value of a resistive load that would produce the same peak amplifier dissipation at each frequency as does your speaker system.

What these plots show is that this system is really a 3Ω load and is relatively benign in terms of amplifier peak dissipation. Systems that have low impedance magnitudes at the same time they have large phase angles can be a severe load on an amplifier causing it to have a peak stress much higher than you would normally expect. This is one more area that the WT3 allows you to investigate.

PREDICTING CLOSED-BOX BASS RESPONSE (BY GRK)

The WT3 will allow you to predict the bass response of an existing closed-box system. It is best if you can get directly at the bare terminals of the woofer, but you can try it through the crossover network if that is your only option. You simply have the WT3 measure the woofer in box impedance via the "Measure Free-Air Parameters" button. The value it reports for f_s is the closed-box f_c, and the reported Q_{ts} is really Q_{tc}. Now if the closed-box system is nearly lossless, it will have the response of a second-order high-pass with f_c and Q_{tc}. A lossless closed-box is one with no stuffing and no built-in air leaks (aperiodic vent, and so on).

We used the WT3 to measure the 6½″ woofer in a 0.299ft^3 closed box, reporting f_c = 74.69 and Q_{tc} = 0.7388, and then fed these values to a program that plots the second-order high-pass response (**Fig. 13**). This response predicts an f_3 of about 72Hz. Audiosuite measured the response of the box at low frequency via the near-field technique. The result in **Fig. 14** predicts f_3 is about 74Hz. This is a useful technique, but keep in mind it is only an approximation. It can lose accuracy if the crossover is involved or the box is stuffed. If you are interested in the effect of the coil resistance in series with the woofer, you should substitute a matching resistance rather than test through the crossover.

While this technique is relatively safe to use on woofers, I do not recommend it for subwoofer drivers in a closed box. I have found some subwoofer drivers have a peaking built into their response which this technique will not predict.

MEASURING INDUCTANCE (BY GRK)

If you connect an inductor to the WT3, it will report the inductance in mH in the L(e) box. I decided to give this a try with some inductors measured initially with a B&K-Precision 878 LCR meter. **Table 6** shows the results in mH for measurement at 1kHz. The technique shows reasonable results for the larger inductors, but should be viewed with some suspicion for smaller inductances. If you have no other inductance measurement technique, it will at least put you in the ballpark.

TABLE 6: INDUCTANCE MEASUREMENT VALUE IN MH

Inductor	B&K	WT3
Air Core	0.181	0.163
Ferrite bobbin core–small	0.341	0.320
Ferrite bobbin core–large	1.054	1.031
Laminated iron core–large	1.016	0.994

NIT-PICKS (BY GRK)

I noted a couple of items in the software (version 1.0.4) and help information that I think should be changed in any future revision.

1. When I clicked the Help button on the Driver Editing Screen, I received an error message stating that help did not

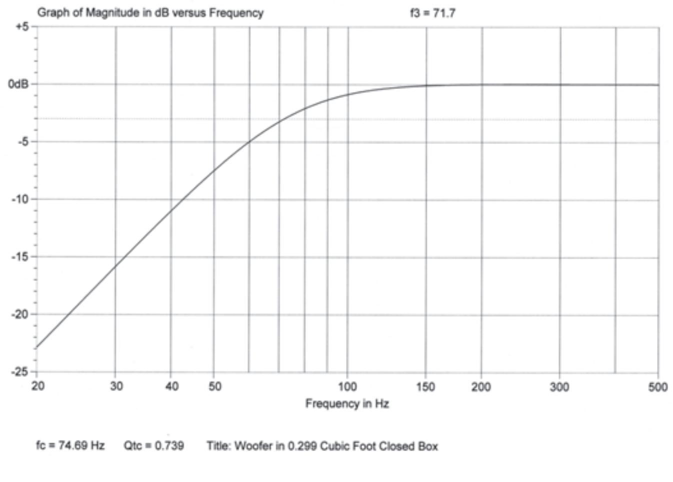

FIGURE 13: Predicted response of woofer in closed box.

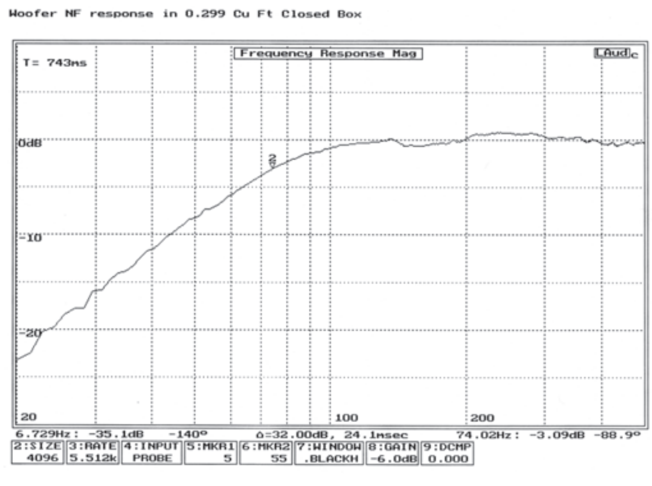

FIGURE 14: Measured response of woofer in closed box.

load, instead of giving me the help I really needed at that time.

2. When you elect to print in B&W, as you do for publication figures, I noted that on the Print Preview and on the print "Ohms" and "0 Deg" were still in color.

3. The help information indicates you can attach individual text notes to each of the 20 memories, but I could never discover how to do this. Probably a misunderstanding on my part.

4. I am at a loss to understand why a passive network impedance measuring devices plots phase over ±180°. Any passive network that goes past 90° is displaying a negative resistance. Plot limits of ±90° would improve readability.

PROTECTION OF THE WT3 (BY GRK)

The instructions warn that you must never connect the WT3 to a driver that is connected to anything else. I don't know how much protection is built into the WT3, but just a bare woofer can be dangerous. You should be careful not to hit or push the cone of a large driver when the WT3 is connected because such drivers can put out a reasonable energy spike if the cone is moved rapidly. For safety I adapted the habit of disconnecting the WT3 as soon as measurements were done or when mounting a baffle mounted driver onto the test box. A bad case might be slamming the trunk lid on a woofer using the trunk volume with the WT3 connected.

SUMMARY (BY GRK)

The WT3 is a very fast and capable impedance measurement instrument which has a lot of flexibility that is very handy in speaker work. It showed good accuracy and repeatability on drivers, but a little less accuracy on inductors. To me a main advantage is its small size offering great portability when used with a laptop computer. For my work it is a must have. Now I must contact Parts Express to see whom I pay to keep the evaluation sample!

SUMMARY (BY ROW)

I found the software easy to install and easy to use. The "Help" file was adequate and informative and gave the tester many excellent procedures in speaker testing. One attribute that would be a nice addition would be the ability to export data to one of the common graphic file formats such as a PDF, PNG, JPEG, TIFF, and/or DOC files. All in all, I believe the WT3 is an excellent buy for the amateur speaker builder who wishes to build his/her own speakers. *aX*

REFERENCES

1. Parts Express 2008 catalog.

2. Dickason, Vance, "The Woofer Tester," *Speaker Builder*, 5/94, p. 56.

3. Weems, David, "The Woofer Tester," *Speaker Builder*, 5/97, p. 44.

4. Madisound catalog May 2006.

5. Howard, Keith, "Heavy Load," *Stereophile*, Vol. 30 No. 7, July 2007, p. 51.

Manufacturer's Response:

Thank you to *audioXpress* and to authors Koonce and Wright for a detailed review of the WT3 Woofer Tester. Evaluating a highly technical product such as this is no small task. Fortunately the authors were up to the job and produced an informative review of the WT3 without getting bogged down in technical minutia.

First, I would like to note that the software version reviewed here is V 1.0.4. As of 12 May '08, version 1.1.0 began shipping and this upgrade has already addressed two of the issues raised in the review. In particular, inductance measurement accuracy has been improved and the Notes field has been duplicated at the main screen to facilitate notes entry and visibility. The software upgrade is available to all WT3 users as a software patch file which can be downloaded at www.daytonaudio.com/wt3updates.html.

In the section on measuring inductance the reviewer raises a concern about the accuracy of inductance measurements at low values. Indeed using WT3 with V1.0.4 software the reviewer measured .163mH for a small air core inductor where the reference system measured .181mH—a difference of 10%. Fortunately, accuracy for small inductor measurements was specifically addressed and improved with the V1.1.0 software release. Users should notice significantly improved accuracy on small inductor measurements with the new software release. On a recheck using the new V1.1.0 software the reviewers would likely find that the WT3 measures the small air core inductor more accurately.

In his nit-picks section GRK reports that the Help button at the Driver Editor window failed to open the Help system. This problem is acknowledged and will be addressed in a future software update. Likewise the appearance of certain labels in color when printing in B&W will be addressed in a future release. Thanks to GRK for identifying these issues.

The reviewer reports that he could not find a way to attach notes to each of the memories. While V1.0.4 did allow you to keep notes with each memory you had to enter your notes at the Driver Editor General Information page which was somewhat out of the way. In V1.1.0 a duplicate Notes field has been added immediately below the Plot window to make it easier to write notes on each measurement. Notes from older projects that were previously entered at the Driver Editor now appear both below the plot window and at the Driver Editor.

The plot limits of ±180° may be wider than strictly necessary but an enlarged view of the phase response is available by clicking the Phase button on the Toolbar. This rotates the phase display among three different modes. First is the reduced display mode at the top of the screen as seen in the WT3 screen shot in **Fig. 2**. Clicking the Phase button once shifts the phase display to center screen and enlarges it. Clicking the Phase button again hides the phase display. Another click and the small phase response plot is restored at the top of the screen. Based on GRK's comments I may evaluate a change to 90° limits.

I wish the reviewers had also compared the two measurement systems with a few precision resistors as this would have revealed the very high accuracy of the WT3 using readily available 1% and even 0.1% tolerance resistors. Although the unit ships with a 1% 1kΩ resistor for calibration I find that if I calibrate my own units with a 0.1% 1kΩ resistor they will typically measure a wide range of 0.1% resistors accurately to within the resistor's 0.1% tolerance. Other audio impedance measurement systems show their limitations clearly on this simple resistor test. Remember, the accuracy of driver parameters is only as good as the impedance data from which the parameters are calculated—and this is where WT3 excels!

John L. Murphy
Physicist/Audio Engineer
john.murphy@trueaudio.com

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Ad Index

CORRECTION

In the "$450 Triode/Ultralinear" article in Sept. '08, the parts list for the chassis box is in error. The chassis boxes are 13.5 × 5 × 2″ and not 17 × 10 × 3″. The correct part number is P-H1444-18.

Joseph Norwood Still
Towson, Md.

1958 MEMORIES

A hearty "thanks" and round of applause for Barry Fox's *Touring Recording's Memory Lane: 1958*, which should be required reading by anyone with access to the August issue.

It was the philosopher George Santayana who said, "Those who cannot remember the past are condemned to repeat it." Sure enough, those of us who grew up during the time that great strides were made in audio reproduction technology do remember (and appreciate!) all the fits, starts, and (thankfully) progress in this field. But even in this age of off-the-shelf, all-done-for-us, increasingly-digital audio reproduction solutions, the current generation cannot help but profit from a study and understanding of the roots and foundation of our interest, hobby, or livelihood. . . the science (and sometimes black magic!) of audio.

Thank you, *audioXpress,* for including Barry's piece in your August issue. I hope to see more articles with reference to historical significance in the future. We owe so much to our audio pioneers, and can learn so much from their legacy.

Jim Wood
INOVONICS, INC.
jim@inovon.com

I enjoyed reading the Barry Fox reprint in *audioXpress* because I don't seem to be receiving *Hi-Fi News* anymore. I had a beef with some of the facts in the previous reprint and expressed them in an editorial (www.audaud.com/article.php?ArticleID=4141).

I also must disagree with a couple things in his column in the July/August *MMM* issue. The Bert Whyte tape recordings were not stereo—they were all binaural. So the "stereo image" is not really correct for speaker reproduction because they were created for headphone playback only. I had tried to get Bert Whyte and Les Paul together since Whyte no longer had a staggered head deck and Paul did; but I was unsuccessful.

In writing about the "pre-RIAA stereo" from the three-track optical recorders in Hollywood, Fox doesn't seem to be aware of the fascinating series of CDs issued in 1997 by Rhino Movie Music, which were taken from various music tracks recorded for MGM movies now owned by Turner Classic Movies. One was titled "Alive and Kickin': Big Band Sounds at MGM." About half the tracks are in excellent stereo from 1943-44. There is also a 1945 Beethoven piano concerto recorded by the Germans in stereo; it was reissued on a Varese-Sarabande CD. I was surprised no mention was made of the two 8-minute 1932 Ellington stereo medleys made for RCA's ill-fated long play record project of the time. These discoveries by Brad Kay were issued on CD some years ago, and even with the losses in the poor transfer they sound very impressive.

Finally, the question of "Is it really stereo?" on these so-called early "accidental stereos" wouldn't even be suggested, as Fox does, if one simply listens to the results and compares it to mono. Executives at both EMI and RCA suggested the whole thing was a hoax (and still do) without ever hearing the examples! Kay believes there are many more such backup masters in the vaults which were made with a second mike at a different location, for even such performers as Toscanini!

John Sunier
Portland, Oreg.
info@audaud.com

Barry Fox responds:

It is increasingly difficult to get hard facts from people with first-hand knowledge so I—and I am sure readers—will be very grateful for this extra information. I have an LP of the Brad Kay work, and wrote and broadcast with enthusiam about it in the UK many years ago. For my pains I was pilloried and ridiculed by "experts" at EMI in London.

I lost touch with Brad Kay but was reminded of him recently when I read the recent book about the late singer Susannah McCorkle which talks about his friendship with what I assume to be the same Brad Kay.

I never knew the Ellington material was available on CD. I would love to get hold of a copy. At the moment I am looking into some similar work done on Stokowski recordings and hope to be writing about it later this year.

ONLINE AUCTIONS

The article on buying and selling on the web by Bruce Brown (Aug. '08, p. 6) was well written and very enjoyable. May I please add the following: Rather than use eSnipe, you can use eBay's "Bid Assistant," which lets you place a single bid for multiple similar items (Check eBay for details). If you want to sell, first check the eBay listing and see how your particular item is selling. If you see 12 listings with an occasional bid, then leave it in a box in the attic! A pristine piece of equipment should sell, but keep the starting bid price reasonable (you could lower it if no bids).

Vincent Mogavero
VMogavero@aol.com

First, I want to thank Bruce Brown for a very useful and comprehensive article about online auctions. It is long overdue. There are one or two things that I would like to add (I can never resist).

1. eBay has now changed their policy so that sellers can no longer submit feedback about buyers. They claim this prevents sellers from doing the "feedback ransom" thing. Sellers can

still file complaints against delinquent buyers, but not ordinary feedback. The system is the same as before for buyers submitting feedback for sellers.

2. eSnipe has another interesting feature worth mentioning: an option to bid on "groups" of items. This allows you to set a price and bid limits on multiple similar items—so, if several of the items you want are being auctioned, you can instruct them to bid to a certain amount on each until you win *one* of them—then stop bidding so you get only one. This can be handy if there are multiples of the item you want, but all of them are closing between 4 AM and 5 AM. (If you were to place separate bids on multiple items, you might win more than one.) New merchandise often appears in such groups. I should also mention that eSnipe is a pay service—so you pay them a separate (reasonable) fee to bid for you.
3. eBay also now offers a service similar to the one I described for eSnipe (the one for bidding on items in groups). Theirs is free and is called, "Bid Assistant."
4. eBay uses a system for most auctions called a "proxy bid." Many people seem somewhat confused by the way this works. You enter the *maximum* amount that you are willing to bid for an item. eBay will then increase your bid *up to* that amount if necessary so that you are the high bidder. This means that you will not end up paying more than your proxy bid, but you might well pay less. It also makes the bids appear to jump oddly at times.

   And, yes, you are "trusting" eBay not to show the seller or other bidders what your maximum proxy bid is. (This works exactly as though you were participating in a "real" auction and had authorized someone at the auction house to proxy on your behalf because you couldn't be there or on the phone.) The way to avoid "getting sniped" is to place a proxy bid that is really as high as you are willing to pay for the item—no fast reflexes will allow anyone (or anything) to win an item away from you for less than the highest proxy bid you have entered.
5. I was unaware of the "dislike" of shippers for reused packing—in terms of insurance—and that is very good to know. From my experience, oddly, it is much easier to get a shipper to "total out" an item and pay off the entire value than it is to get them to pay for partial damage. For partial damage they will often ask you to provide an estimate for the repair, and ship the item to them for inspection—which is problematic if the item is so badly damaged that re-shipping it will render it irreparable. Oddly, however, I've had several items entirely paid off as "destroyed," and was never asked to provide the item for inspection (they did ask for pictures). In fact, they allowed me to keep the items (so, at least, they served for parts).

   As an idea of what is expected from packing, UPS used to suggest that an item should be packed such that it would survive three drops of 2 or 3′ onto concrete, with any side up, to be considered "well packed." Also remember that some items, notably speakers, can be damaged by acceleration—so the packing must be able to absorb impact (which is why bubble wrap or foam is good). You could pack an item tightly in very strong cardboard, for example, but with too little padding, and still have internal parts wrenched loose if it were dropped. I received a pair of speakers packed this way once—the box was perfect, as was the interior packing, but all of the drivers and the crossover panels had been ripped off the front panels due to the box being dropped quite hard, and the drivers were beat up beyond repair.
6. Last but not least, PayPal now offers a hardware security token for accessing your account. This small plastic item looks like a small pocketwatch. When you press the button it returns a six-digit number—which changes every minute or two. You log in by entering your password and the current six-digit number—so someone with your password but not the token cannot access your account.

   The token costs $5 (including shipping) and is free thereafter to use. This is the way security *should* be done, and is a very good idea—

because it prevents anyone who steals your password from using it, and the current number won't work for them two minutes from now even if they see it. Nothing else changes in your account except the need to append the six-digit token code to your password.

You can get more details by typing "security key" at the search prompt on the PayPal site. (They promoted it for a while then stopped talking about it—now you must look to find it.) The token is similar to the SecurID token. . . for those who are familiar with security. You *can* still log in without the key, which weakens it a bit, but it requires more information than just the password. At a minimum, it is far more secure than only a password.

Keith Levkoff
kLevkoff@panix.com

Thanks to both Bruce Brown and reader Keith Levkoff for their helpful thoughts on eBay. The company now has new management, which I hope will fix its problems. The cancellation of complaints about vendors is unfortunate, but it has been largely ineffective, in any case. Several articles which I bought are still not delivered, and the complaint process is medieval and more complicated than dealing with Internal Revenue. —ETD.

AURORA

I built Patrick J. Amer's Aurora preamp (3/87 *TAA*) back in 1989 and use it on Westrex power amplifiers and the Western Electric 91 power amplifier. I have used many brands of loudspeakers on the Westrex amps and I use Lowther on the Western Electric 91. The Aurora sounds great on both amps and speakers. I have also A-B tested with other preamps, and my friends and I prefer the Aurora. The Aurora has given me years of music pleasure, and I would like to thank the author for sharing his circuits.

Peter Hawthorne
United Kingdom

AD CLAIMS

I read Ed Dell's response to the letter from Mr. Richard Campbell in the July issue and believe that I must add my comments to the discussion. Accord-

CONTRIBUTORS

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

George Danavaras ("The Lamda Loudspeaker," p. 7) graduated from National Technical University of Athens, Greece in 1986 with a degree in Electronic Engineering. He currently works in the R & D division for a Greek telecommunications company. His hobbies include design and manufacturing of audio crossovers, amplifiers, and loudspeakers.

Scott Reynolds ("LED Bias Monitor for Tube Amps," p. 18) works for a communications company in New York.

Bruce Brown ("Class D Amplifiers," p. 22) is a registered pharmacist who works in the medical research area for a major pharmaceutical company. He has been experimenting with electronics for over 35 years, remaining actively interested in electronics, building kits, and "home brew" audio. Online auctions have stimulated new interests. He currently uses many restored and homebuilt audio amplifiers and welcomes communication about vintage equipment and restorations. He can be reached at Tuninfork@aol.com.

Ed Simon ("250W of Clean AC Power, Pt. 4") received his B.S.E.E. at Carnegie-Mellon University. He has installed over 500 sound systems at venues including Jacob's Field, Cleveland, Ohio; MCI Center, Washington D.C.; Museum of Modern Art Restaurants, New York; The Coliseum, Nashville, Tenn.; The Forum, Los Angeles; Fisher Cats Stadium, Manchester, N.H.

G.R. Koonce (Product Review: Dayton Audio WT3 Woofer Tester, p. 32) is an electrical engineer who has enjoyed the hobby of designing and building audio equipment, test gear, and speaker systems. Professionally, he worked for over 30 years producing audio frequency equipment for military use. He is a member of AES and has been a contributing writer to *Speaker Builder* since 1981.

Robert O. Wright, Jr. (Product Review: Dayton Audio WT3 Woofer Tester, p. 32) is a graduate of the University of Tennessee. He is a retired mechanical engineer from the Tennessee Valley Authority and is now the owner of a custom design-engineering firm in Knoxville, Tenn. It specializes in machine design, pollution control equipment design, sound design, and speaker design. He is also a member of the AES.

John Sunier ("Super Fidelity," p. 46) is a CD reviewer for *Australian HiFi and Home Theatre Technology*. His website is www.audaud.com.

ing to Webster, hyperbole is defined as "exaggeration for effect, not meant to be taken literally." It appears that you are confusing hyperbole with actual claims.

If an ad mentions that RFI is reduced and distortion lowered, that is different than saying that you will feel better about owning the product. Those are claims that can be verified as compared to the creation of a feeling of pride of ownership. There is no doubt that having a product with excellent form and workmanship can result in higher total satisfaction than a product that may perform equally as well but look and feel clunky. However, that higher level of craftsmanship should not be presented as providing higher technical competence, unless that is actually true.

Your statement about ads that center on how you feel when driving an expensive car does not relate to an ad that would state that the car gives you better mileage or has faster acceleration. The former might be hyperbole, but the latter would be a misstatement of facts unless true. The problem with many claims in audio is that the advertisers do, in fact, wish you to take them literally and buy the product based on their claims. It is also interesting that the degree of this "exaggeration," especially with claims for cable performance, is much worse in the audio industry than many other consumer businesses. Even if the problem were simply a matter of the use of hyperbole in advertising, just because other people are doing it still does not make it desirable.

The danger in this excessive use of "hyperbole" is that consumers often discover the truth and then feel "ripped off." This is bad for long-term prospects for the industry. I admit that I am personally disturbed with the excessive claims of many, if not most, of the cable ads, which tend to displace the resources of the consumers to areas that at best might have a minor effect on sound compared to major players such as speakers and room treatments. I'm not talking about ads that claim that gold-plated #10 gauge wire will be more impressive to your girlfriend, but rather those that claim this wire will eliminate distortion, provide fantastic transient response, and so on.

I believe it is important for you, as a publisher, to maintain balance in the presentation of ideas, but I also believe that passing off performance claims as hyperbole does not fall into that category. I am strongly in the camp of Mr. Campbell in his assessment of what I believe is a significant problem in the industry.

Tom Perazella
tom.perazella@verizon.net

ATTRIBUTION

Jim Dungan's "HD-2: A Single-Ended Class-A Headphone Amp" in the June issue is a rip-off of Scott Dorsey's "Kludge 4 Single-Ended OTL Headphone Amp Project," which appeared in *Vacuum Tube Valley* # 11, Spring 1999. Give credit where it's due.

Tom Tutay
Transition Audio Design
Fort Walton Beach, Fla.

aX

Super Fidelity

By John Sunier

www.audaud.com

Brazil|Los Angeles Guitar Quartet|Telarc Multichannel SACD-60686
The Los Angeles Guitar Quartet has been exploring a wide variety of music for nearly 30 years. This new album explores the cultural melting pot of music that is Brazil. The quartet admitted one of its most prominent guitar influences was Brazil, but that they had avoided recording Brazilian music, feeling that other great composers and arrangers who were Brazilian were doing it better. Brazilian jazz vocalist Luciana Souza talked them into it, encouraging the LAGQ to just follow their instincts in performing it. They brought in a percussionist for *A Furiosa*, by their friend Paulo Bellinati, and also performed *Carlo's Dance* by the same guitarist-composer, together with flautist Katisse Buckingham. All 15 tracks are a delight, and the very specific placement of the four guitarists across the front of the soundstage adds much interest to the music, so if you're a two-channel diehard, you're missing out!

Fantaisie Triomphale|French Symphonic Organ Works|Chandos Multichannel SACD CHSA 5048
It was Franck and others of the French Organ School who expanded the capabilities of the instrument and also created a new style of composition—the organ symphony. The combination of the new French concert grand organs with a symphony orchestra made for an impressive panoply of sound—a far cry from the modest little concertos of Mozart and Handel. They literally pulled out all the stops, and this disc samples some of the exciting repertory. All were written for the combination of pipe organ and orchestra except the opening Gigout work and all abound in the unique showy French style of triumphant music, the expression in sound of the concept of "La Gloire." There's lots of brass, thrilling climaxes of sound, and a strong spatial element. Recorded in Liverpool Cathedral—which has a giant organ of the French School—the six or seven second reverberation results in a rather rumbly organ sound that even the five-channel pickup seems to have some trouble reproducing.

Don Gillis|Symphony X; Tulsa, Symphony No. 3|Sinfonia Varsovia/Ian Hobson|Albany SACD TROY933
Gillis, who died in 1978, was quite a character—a cowboy-country Neil Simon of the concert hall. He represented Middle America and played up nostalgia and patriotism in his music. Tulsa was one of his first big hits—thought to be the first time a major financial institution commissioned an orchestral work. The composer's early Symphony for Free Men was a serious attempt to imply American ideals through music. The 1968 Symphony X is a tribute to Dallas, with three bouncy, pop-concertish movements portraying "All American City," "Conventioneer," and "Cotton Bowl" — plus an elegiac fourth movement—"Requiem for a Hero"—that references the assassination of JFK. For this release, the original stereo master has been processed by the Zarex HD Studio to derive a 5.1 surround sound signal. The left and right front channels still retain the main part of the signal: I had to raise the surround level to achieve a pleasing ambience in multichannel playback after which it sounded quite convincing.

Stone Rose|Ola Gjeilo, piano|2L Multichannel SACD 2L48
This is a contemplative and melodic collection of mostly piano solos by Norwegian pianist Ola Gjeilo, who has performed around the world and is especially taken with New York City, as shown by the titles of several of the tracks here. On this CD he's enlisted the aid of guest artists Tom Barber (flugelhorn), David Coucheron (violin), and Johannes Martens (cello). The recording was made in a church in Oslo and the piano tone is very natural, as is the placement of the various guest performers. Tom Barber's solo on *North Country II* is striking and a fine demo of the advantages of well-recorded surround sound even for a simple ensemble such as this. In his notes the pianist mentions that both Keith Jarrett and Glenn Gould have been uppermost as influences. Sorry, but I couldn't hear the slightest connection with either one, but that doesn't detract from a well-performed and recorded album of classically influenced new age music.

The Invitation|Music for Saxophone Quartet|Tetraphonics Sax Quartet|Cybele Records Multichannel SACD 261001
As with most sax quartets, Germany's Tetraphonics play many diverse genres of music. On this disc they play works by Philip Glass (Concerto for Saxophone Quartet), Frank Reinshagen (The Invitation), Barbara Thompson (Saxophone Quartet No. 2), Zdenek Lukas (Rondo for Four Saxophones), and a transcription of a Bach Fugue. Glass' minimalist patterns work beautifully in this setting and especially with the spatial enhancement of surround sound playback. Thompson is a British saxophonist and composer who collaborated with Andrew Lloyd Webber on his musicals. The last movement has the other three players switching to soprano sax for an effect similar to female voices. The sounds that issue from the spatially separated soprano, alto, tenor, and baritone saxes are captured with great transparency and presence by Cybele's 5.1 channel SACD surround.

Pepe Romero, guitar|Flamenco|with Chano Lobato, Maria Magdalena, Paco Romero|First Impressions K2HD LIM022
This amazing disc is the first complete album I have heard that uses FIM's K2HD process. Previously I'd heard only a sampler, reviewed last column (Sept.). FIM's Winston Ma advised me that I erred in calling the K2HD discs "a variety of xrcd." He says they're not. Whatever, the K2HD process is not a gimmick—it overshadows the improvements of most xrcds. Ma also said that in his opinion, this disc is "*the best flamenco recording on the planet*." I would tend to agree. Pepe Romero is a member of the famed Romeros—The Royal Family of the Guitar. Paco provides the explosive zapateado sounds. Audiophiles will go nuts over the sounds on this disc, which is possibly the ultimate audio demo. The rich sound of Romero's guitar is matched by the sharp percussive sounds of the castanets, but the greatest excitement is generated by the super-explosive sounds of the dancers' zapateado. The presence and immediacy must be heard to be believed. Wow!