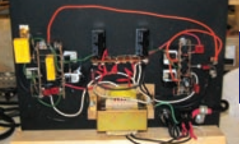


A Solid-State Single-Ended Design

APRIL 2006
US $7.00 • Canada $10.00

audioXPRESS

The Audio Technology Authority

Back

A Superhero Tube Amp

Speaker Test Report: Interference

Build a Simple Booster

Simple Speaker Stands

www.audioXpress.com

PLUS: Ed Frias Speaker Kit Tested

Driver Interference and Display Techniques, Pt. 1

The first part of an extensive study on several design considerations affecting driver performance.

By G.R. Koonce

You know that the presence of each driver on an enclosure front panel affects the performance of the other drivers. You also know that the cabinet edges and any grille structure tend to modify the response of all the drivers. In truth, any structure near the cone of a driver—either on the front or back—can modify the response of that driver. Symmetrical structures can be particularly destructive.

Numerous publications, including *Speaker Builder* and *audioXpress*, have published articles that address these interference problems[1–6]. Generally the published graphs show comparison responses for the test driver with and without the interference present. This clearly illustrates the change in the driver's response, but does not provide any sense for exactly how the interference causes the measured response change.

The power cepstrum plot, which is sometimes also displayed in interference studies, tells you at what times there are delayed echoes of the original signal contained within the signal with interference. Even these leave you with little idea of what the "interference signal" composed of reflections and/or diffractions really looks like. This article will show you.

There are other ways to modify the response of a driver, such as changing the acoustic loading via a horn or waveguide structure. The structures studied here are believed to not change the acoustic loading, but to modify the driver's response via reflection and diffraction.

In this article I will investigate the interference of several structures on the performance of drivers. I will show the change in the driver's response and present plots of the interference signal that is causing the response change. I will also illustrate how, with testing capability, you can develop these interference signal plots.

HISTORY

This all started when I was again measuring damping materials for potential use as front-panel-covering materials to reduce the destructive effects of grille structures (I had earlier co-authored work on this topic[1, 2]). For this testing I decided on a round grille frame mounted concentrically with the tweeter. My thinking was that with such a frame all "echo" effects would occur at the same time on-axis, thus making them more easily detected. The term "echo" is used here to refer to an interfering signal that may be any combination of reflection and diffraction effects. The round grille frame decision was terrible for testing damping materials, but good in that it led to the interference signal display technique covered in this work.

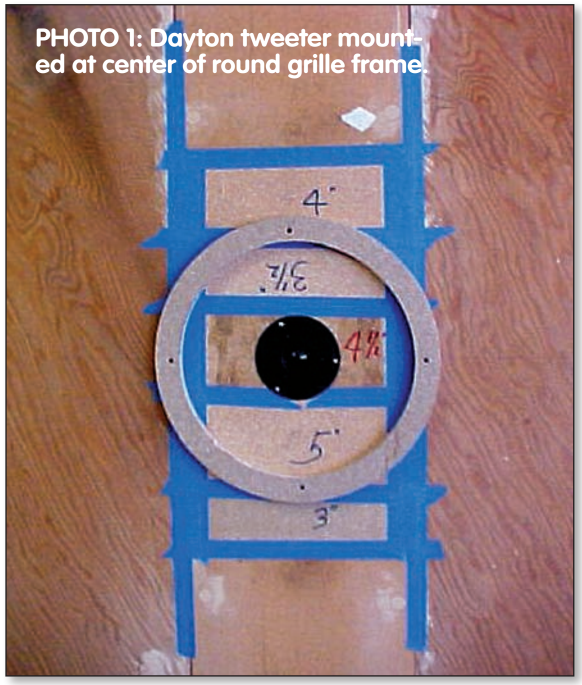


PHOTO 1: Dayton tweeter mounted at center of round grille frame.

The round, concentric grille frame certainly did concentrate the echoes. The test tweeter's response was so badly destroyed that no damping material applied to the front panel had a chance of restoring a useful tweeter response. I sent a copy of the results to George Augspurger (GLA), who computed the energy in the interference signal. Even though an interference may be composed of multiple signals that are diverse in frequency and time, at each frequency you can resolve these signals to a single resultant sine wave. Based on GLA's work, I found that I could compute and plot this resultant interference signal, which I call the composite reflection signal. For some cases, at certain frequencies, the magnitude of this composite reflection signal will exceed the magnitude of the original driver on-axis signal. Under these conditions the interference may be delivering a higher SPL signal to your ears than the tweeter!

THE ROUND GRILLE FRAME

I used the round, concentrically mounted grille frame to demonstrate the plotting of the composite reflection signal. It also clearly demonstrates that you should not choose to use a round grille frame, or any other symmetrical grille structure, for that matter. I first examine the results as they have generally been presented in the past and compare them with the effects of the composite reflection signal.

Photo 1 shows a dome tweeter mounted in the center of the round grille frame on my test baffle. Initial testing starts before installing the grille frame. The test tweeter is a 1-⅛″ silk dome (Dayton #275-070). **Figure 1** shows the measured 1m on-axis response of this tweeter surface-mounted with and without a diffraction ring. The diffraction ring (DR) is a simple device for spreading out the diffraction caused by the faceplate of a surface-mounted tweeter[3]. The DR does smooth the tweeter response somewhat, but in this case not nearly as much as flush-mounting might do.

Figure 2 shows the power cepstrum (PC) plot for the surface-mounted tweeter without a DR, displaying a major echo group around 0.15ms. This corresponds to a delay distance of about 2″, which is close to the 2.15″ radius of the tweeter faceplate. **Figure 3** shows the PC plot with the DR added, which has clearly spread the diffraction echoes out in time with a corresponding magnitude reduction. This plot led me to expect more improvement in the frequency response than in **Fig. 1**. I also expected the major problems from the round grille frame to start at about 0.47ms, so I decided to test without the DR, which left the time period after ½ms more free of echoes.

The round grille frame is ¾″ thick (height above test baffle) with a 12″ ID

FIGURE 1: Responses of Dayton tweeter with and without diffraction ring.

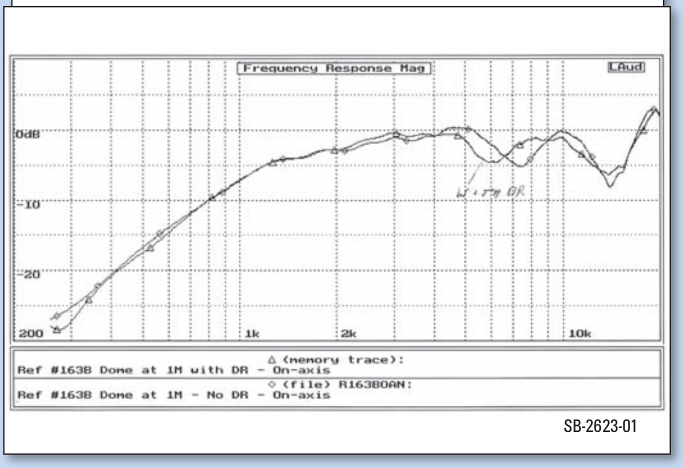
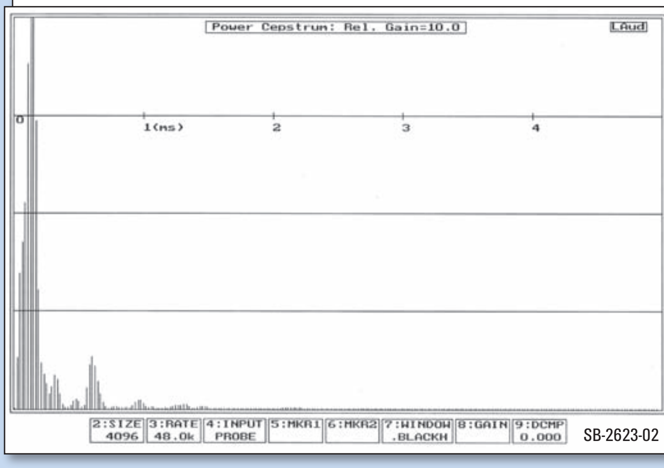


FIGURE 2: Power cepstrum plot for Dayton tweeter without diffraction ring.



and 14½″ OD. The test microphone is back to 44″for a valid far-field test range including all the output from the grille frame. **Figure 4** shows the tweeter response (at 44″) with and without the grille frame (GF) mounted. The GF modifies the original tweeter response at some frequencies by many dB.

The PC plot with the GF (**Fig. 5**) shows, as expected, large echoes spread out over greater than 1ms. In the past I have tried to relate these echo groups to the physical dimensions of the interfering structure without learning much.

In an attempt to "cure" the destruction of this GF, I covered the front panel area inside the frame with an increasing number of thin (0.042″) felt layers. I made the bottom layer oversized to cover the inside of the frame (**Photo 2**). From the measured responses with the various felt layers (**Fig. 6**), it is clear that even this much felt does not restore the tweeter response. There is, however, a bit of a mystery in **Fig. 6**: Why does the dip at 3.5kHz become deeper with more layers of felt? Something is going on here that can't be determined from the frequency response or power cepstrum plots.

THE COMPOSITE REFLECTION SIGNAL

How to implement the testing and computation to produce these plots of the composite reflection signal is covered in Appendix A. For now, I will examine the test results and the composite reflection curve plots; i.e., the resultant reflection signal relative to the bare tweeter response. The basic approach is to test the frequency response of the bare tweeter and export that result as a data file. Next you add the interference object—here the round GF—and measure just the change in response, both magnitude and phase shift.

Figure 7 shows such an error plot, titled echo signal. As you can see, the amplitude variations are about 20dB peak-to-peak (PTP), and phase shift variations are up to 180°. These curves

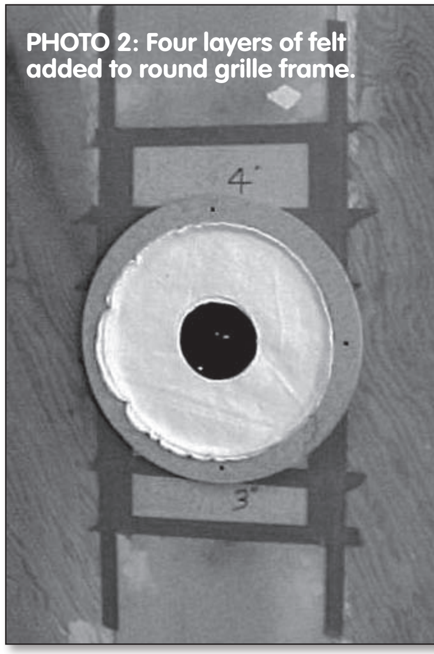


PHOTO 2: Four layers of felt added to round grille frame.

FIGURE 3: Power cepstrum plot for Dayton tweeter with diffraction ring.

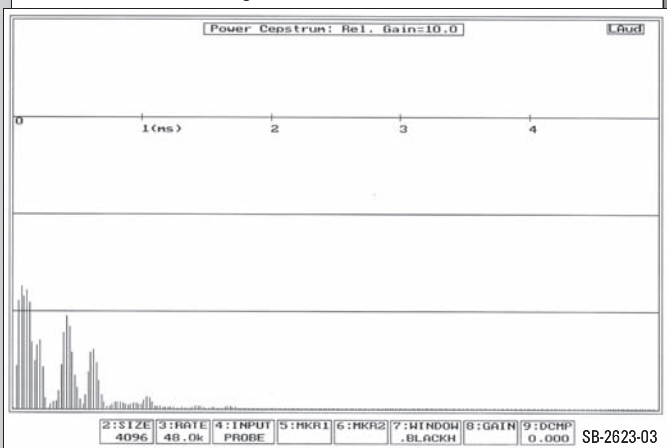


FIGURE 5: Power cepstrum plot of Dayton tweeter with round grille frame.

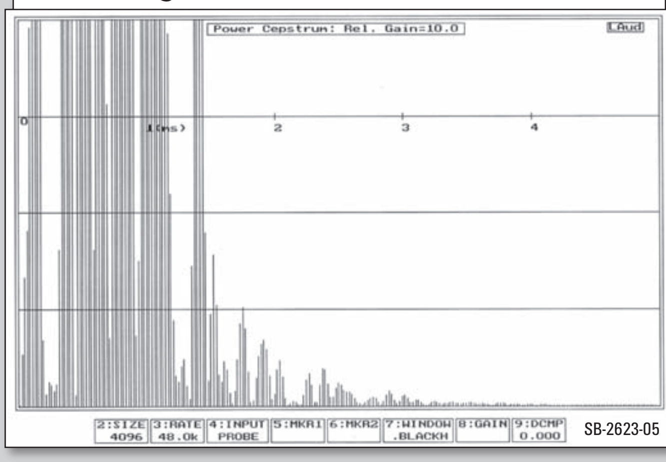


FIGURE 4: Responses of Dayton tweeter with and without round grille frame.

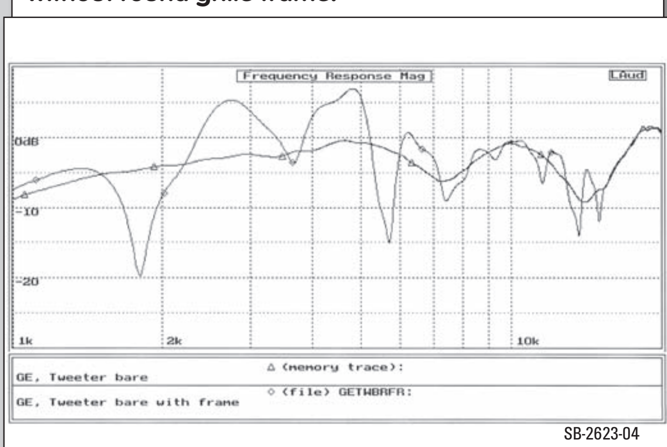
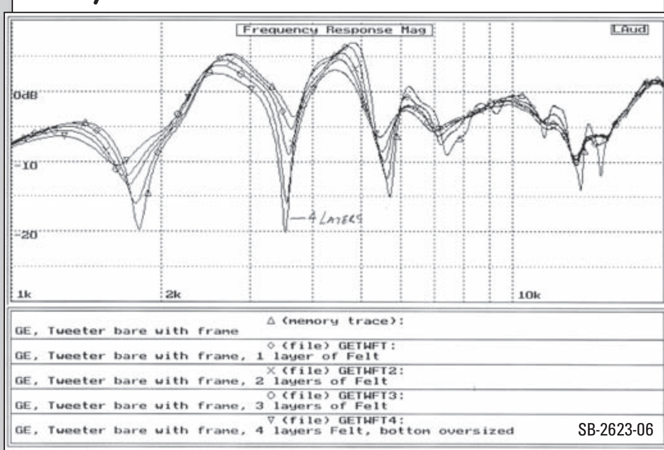


FIGURE 6: Tweeter responses with various numbers of felt layers added.

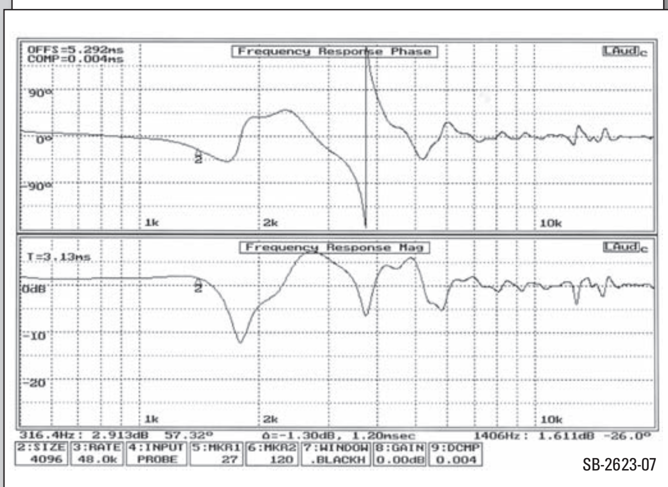


are also exported as a data file for computer processing.

The curves in **Fig. 7** depict the changes in magnitude and phase shift that the GF would make on an "ideal" tweeter; that is, a tweeter with a perfectly flat magnitude response at 0dB and a phase shift of zero degrees. Based on these curves, your computer program can calculate the magnitude (and phase if desired) of the resultant interference signal that caused the response changes to your "ideal" tweeter. I computed only the magnitude function, calling it the composite reflection curve (CRC). You can then re-normalize the results to match your actual tweeter.

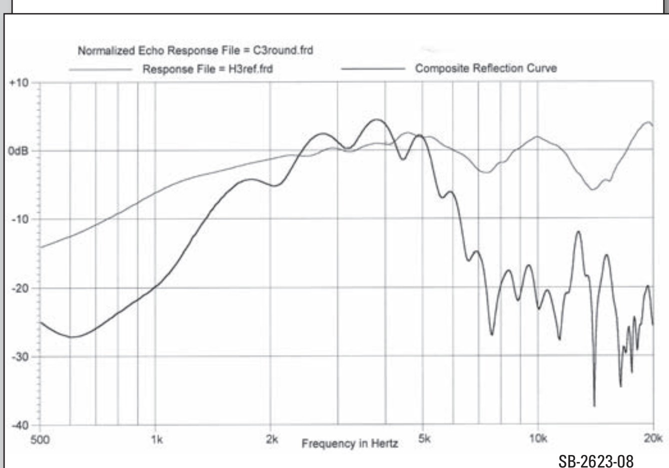
Figure 8 shows the composite reflection curve (the bold plot) for the round GF versus the bare tweeter response. The CRC is the magnitude curve of the interference that causes the measured changes in the actual tweeter's response. It is clear that the CRC exceeds the magnitude of the tweeter response and thus, depending on phase, can cause major peaks and dips. Note that the CRC has a bandpass shape for the major interference, producing less destruction at low and high frequency. Keep in mind that while I called the CRC a reflection, its origins may be any combination of reflections and diffractions caused by the

FIGURE 7: Error curves for tweeter with round grille frame.



SB-2623-07

FIGURE 8: CRC plot for tweeter with round grille frame.



SB-2623-08

interfering object and arriving at the on-axis microphone during the FFT time window used.

Figure 9 shows the error signal magnitude and phase shift for the "ideal" tweeter with four layers of felt added to the GF. Both the magnitude and phase shift variations are reduced, as you would expect. The CRC plot in **Fig. 10** shows that the interfering signal still nearly matches the tweeter response magnitude in the 2kHz to 4.5kHz range. It is clear that depending on the phase relationship the GF is capable of producing the deep dip that was noted with four layers of felt.

I believe that the CRC plot adds new insight into what happens when some object interferes with a driver's response. You see in what frequency range the major destruction will occur and how large the magnitude variations might be. I cover only on-axis testing in this article, but you can apply the same technique to off-axis testing.

A MORE RATIONAL GRILLE FRAME

I tested the same surface-mounted tweeter with a more realistic GF constructed of ½″ × ½″ bars. This half-inch-tall rectangular structure was 10″ wide by 14½″ high on the inside and positioned as on a small two-way system. The tweeter was offset from the centerline by ¾″ to destroy symmetry (**Photo 3**).

To try and learn more about the mystery of GF diffraction, I did not install the frame all at once, but one bar at a time. I measured the effect of each additional bar on the tweeter's response. The microphone was back 57″ to provide a valid far-field response for the full GF.

The first bars installed were the top of the frame and then the left side. These two bars are nearest the tweeter, so I expected them to cause the most trouble.

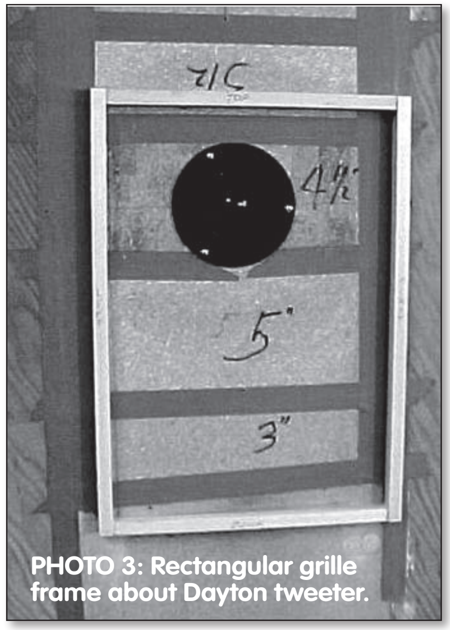


PHOTO 3: Rectangular grille frame about Dayton tweeter.

Figure 11 shows the tweeter response bare, with only the top bar and with the top and left bars. There is very little destruction of the tweeter's response.

FIGURE 9: Error curves for tweeter with round grille frame and four layers of felt.

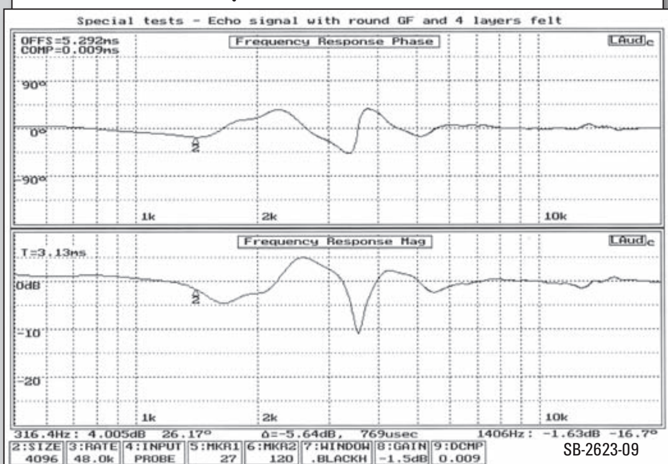


FIGURE 11: Responses for bare tweeter, top GF bar, and top and left GF bars.

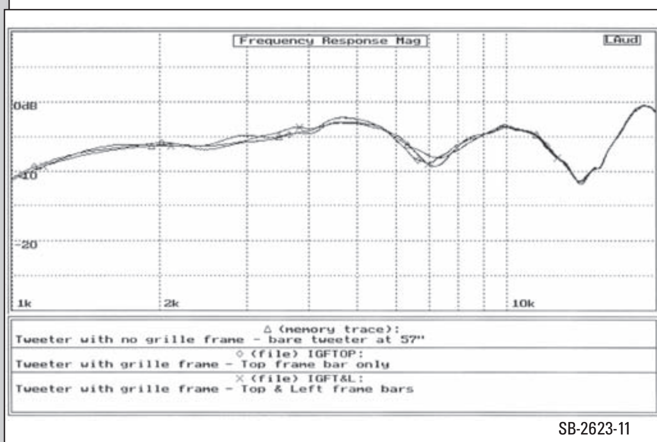


FIGURE 10: CRC plot for tweeter with round grille frame and four layers of felt.

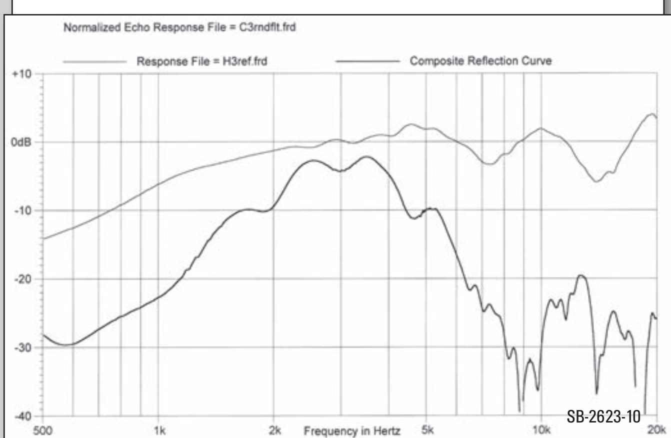


FIGURE 12: Power cepstrum plot for tweeter with top and left GF bars.

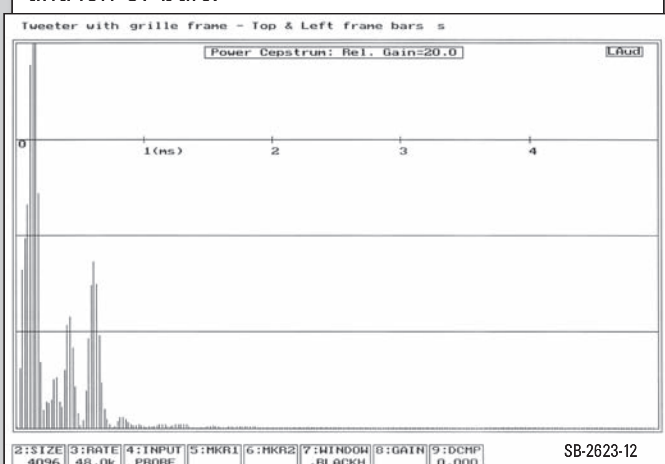


Figure 12 is the PC plot for the top and left bars with echoes out to about ¾ms indicated. **Figure 13** shows the error curves for the case of the top and left bars installed, again showing a minimum of response change. The CRC plot in **Fig. 14** shows an interference signal running about 20dB below the tweeter response from about 2kHz to 9kHz. This indicates a wide but gentle destruction of the bare tweeter response.

Next I added the bottom (farthest) and finally the right bars. **Figure 15** shows the responses for the bare tweeter, then the top, left, and bottom bars, and finally all four bars. To my surprise, the most distant bars cause more response destruction than the closest bars.

The PC plot for the four bars (**Fig. 16**) contains echoes out past 1ms. The error curves for these four bars (**Fig. 17**) indicate more error than for just the two closest bars. The CRC plot for the four bars (**Fig. 18**) shows that the interference signal magnitude has risen to the -10 to -13dB level over the range 2kHz to 6kHz, resulting in the increased bare-

FIGURE 13: Error curves for tweeter with top and left GF bars.

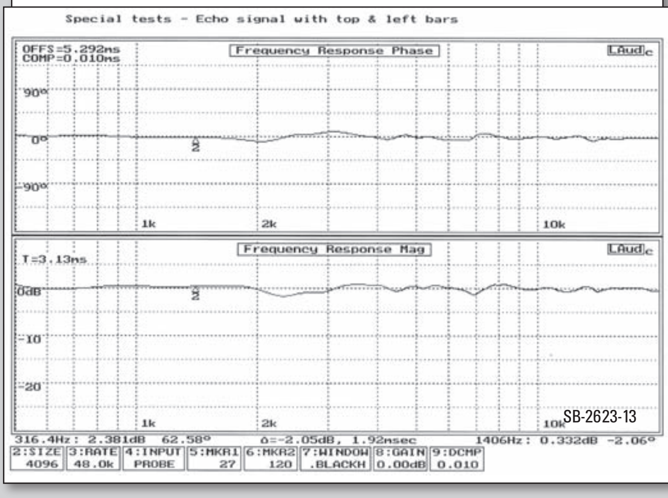
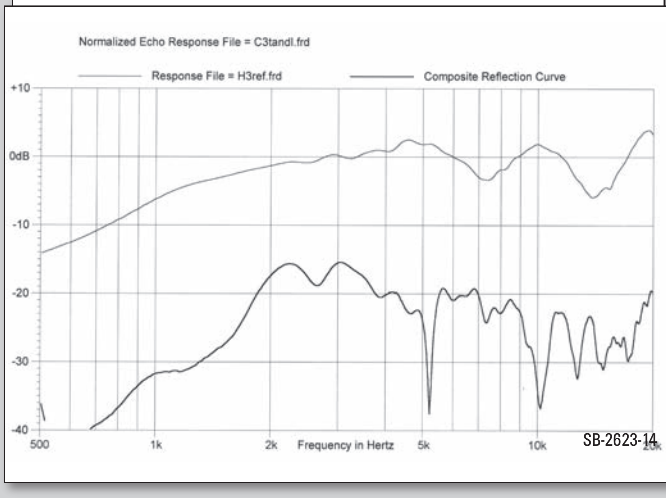


FIGURE 14: CRC plot for tweeter with top and left GF bars.



tweeter response destruction.

You might argue that the reason for the increase is because four bars are worse than any two. While not shown here, I tested this option by removing the two closest bars and verifying that the two furthest bars alone produce most of the destruction. I don't really know why this is so. It may be that the faceplate of the surface-mounted tweeter masks the sound energy to the closest bars or that the longer time delay of the farthest bars causes more trouble.

Remember, I am measuring only the on-axis response destruction. It may well be that the closest bars cause more trouble off-axis. The point to remember is that all of the grille frame—not just the portions close to the tweeter—should be considered as a potential problem.

MIDRANGE'S EFFECT ON TWEETER'S RESPONSE

It is well known that when you add drivers to the front panel, you modify the response of each driver from what it was alone. I decided to investigate the changes in the response of a tweeter when a midrange driver was added. To minimize the interference, I flush-mounted the drivers, which included the Terra CS-1.1 tweeter, an inverted 1.1″ aluminum dome, and the 5¼″ Terra CS-4ABSE open-back midrange, which is actually a mid-bass driver, again with aluminum cone. **Photo 4** shows the two drivers in the test baffle mounted at 6″ center-to-center (CTC). Due to the reduced span of the two drivers, the microphone is back only 1m for a valid far-field test.

Figure 19 plots the bare tweeter response (midrange replaced with a solid panel) and the tweeter response with the midrange mounted. I was thinking "two-way system" when taking this data, so the figure mentions a "woofer," but it is actually the 5¼ Terra Mid-Bass noted above. Response changes are noted from low frequency right to 10kHz, but seem worse in the 2–3kHz and 5–7kHz ranges.

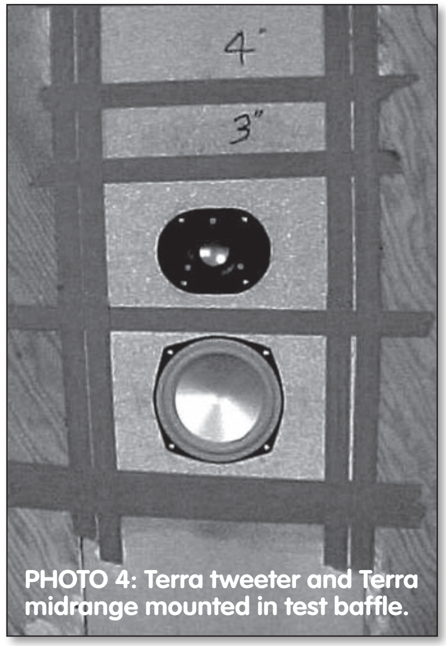


PHOTO 4: Terra tweeter and Terra midrange mounted in test baffle.

The error curves (**Fig. 20**) display

FIGURE 15: Responses for bare tweeter; top, left, and bottom GF bars; and all four GF bars.

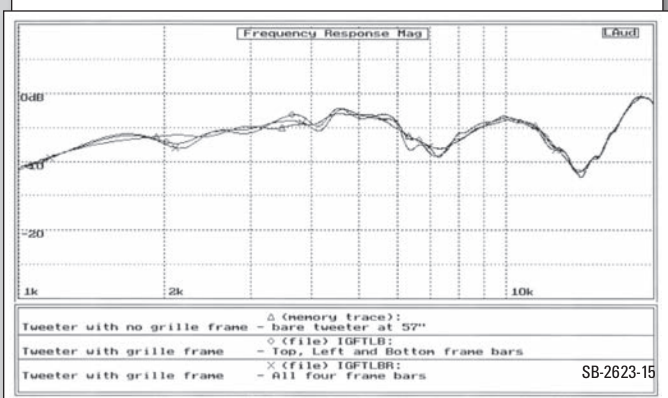


FIGURE 16: Power cepstrum plot for tweeter with four GF bars.

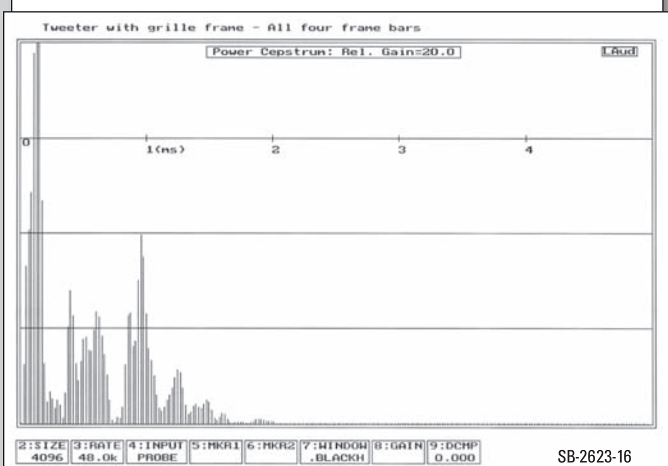


FIGURE 17: Error curves for tweeter with four GF bars.

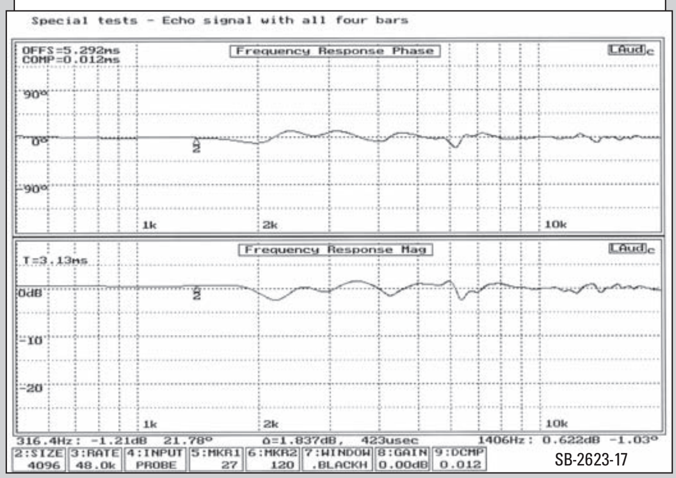
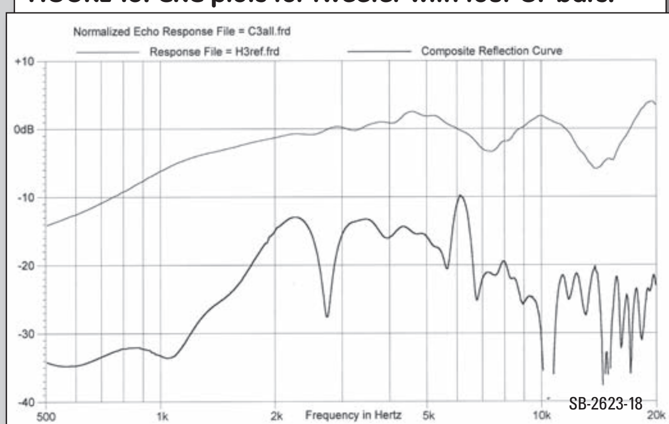


FIGURE 18: CRC plots for tweeter with four GF bars.



slight response errors in the "ideal" tweeter response. The CRC plot (**Fig. 21**) shows the interference signal is down only 18–20dB in some ranges and demonstrates why the bigger response changes do occur 2–3kHz and 5–7kHz. So even with both drivers flush-mounted and fairly widely separated, the midrange can have a measurable effect on the tweeter's on-axis response.

TWEETER'S EFFECT ON MIDRANGE'S RESPONSE

While the presence of the midrange added about ±2dB of response variation to the tweeter, would the tweeter have any effect on the midrange response? I performed this test using the same two drivers as above tested at 1m. **Figure 22** presents the response of the midrange with and without the tweeter. Note that while this midrange shows the typical break-up ripple of an aluminum cone driver above 4kHz, it is actually capable of a smoother response below 4kHz than is shown here (to be discussed later). It appears the presence of the midrange has little effect on the tweeter response, as the error curves of **Fig. 23** show. In the CRC plot (**Fig. 24**) the interference signal is down about 30dB, near the test capability

FIGURE 19: Terra tweeter response with and without midrange mounted.

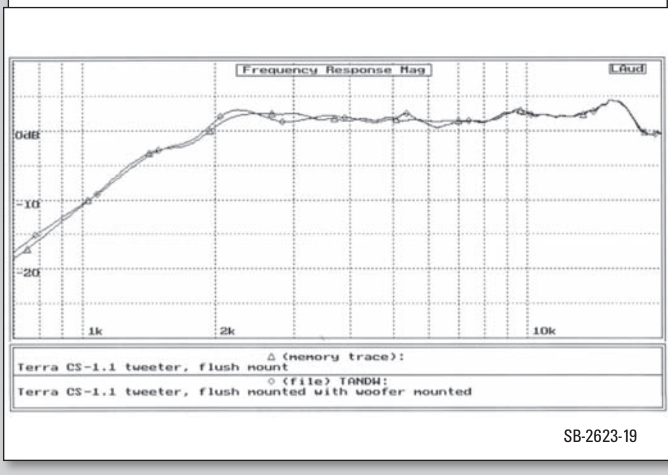
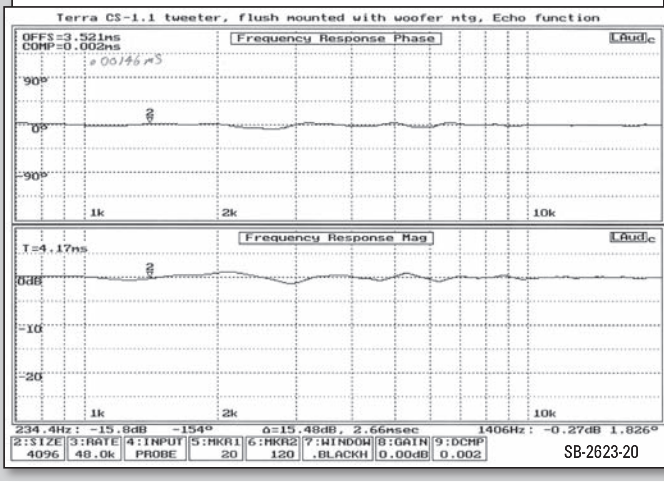


FIGURE 20: Error curves for Terra tweeter with midrange mounted.



limit discussed in Appendix A.

DAMPING STRIPS BETWEEN DRIVERS

In a personal communication, GLA wondered whether or not placing a strip of dense-fibrous damping material between the midrange and tweeter would reduce the response interaction. I decided to test this concept with three different densities of Owens-Corning fiberglass material—#701, #703, and the high-density #705. Mr. Ralph[4] showed good performance of felt strips in similar applications. Would the high-density fiberglass (FG) show similar results?

There are two parts to this testing. First, just adding the FG strip with no

FIGURE 21: CRC plot for Terra tweeter with midrange mounted.

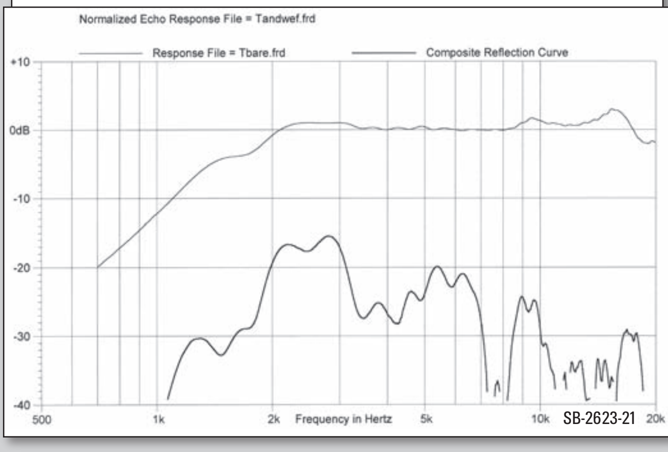


FIGURE 24: CRC plot for Terra midrange with tweeter mounted.

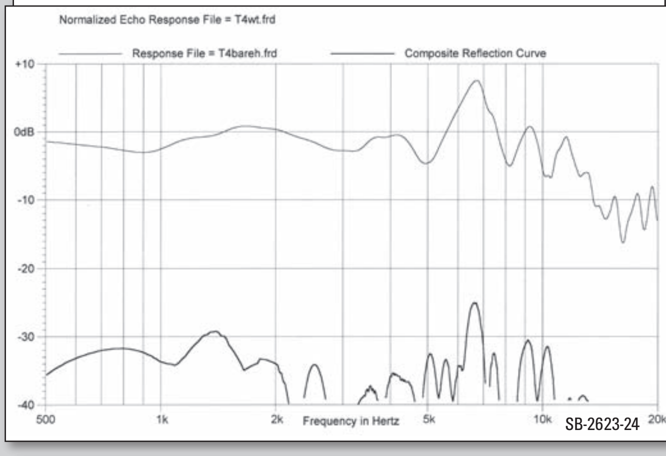


FIGURE 22: Terra midrange response with and without tweeter mounted.

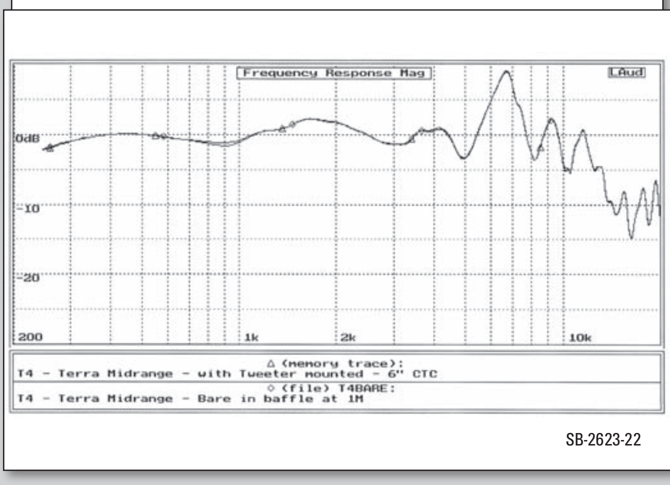


FIGURE 25: Error curves for #701 strip below bare tweeter.

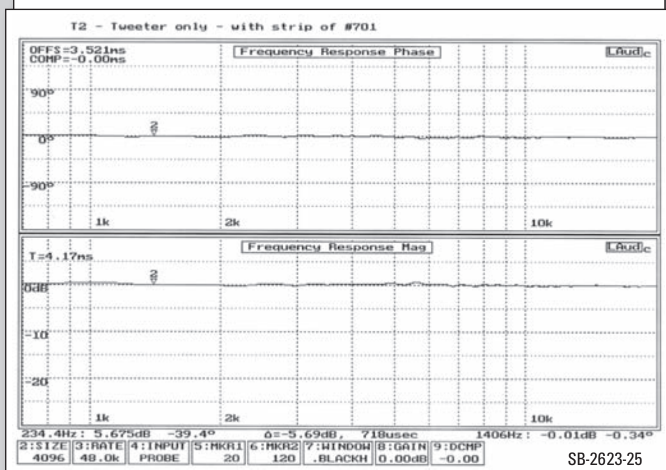


FIGURE 23: Error curves for Terra midrange with tweeter mounted.

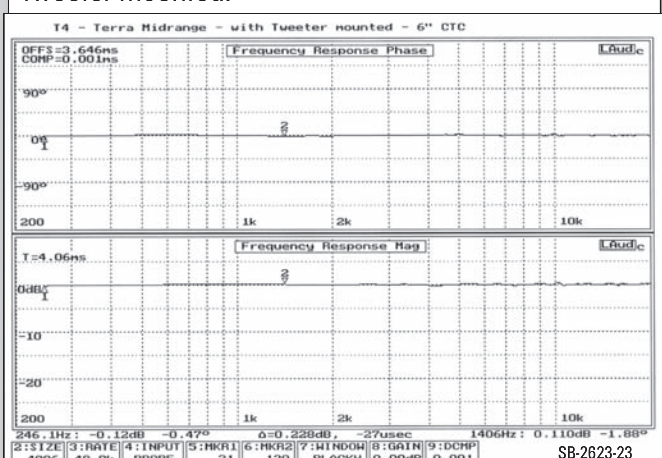
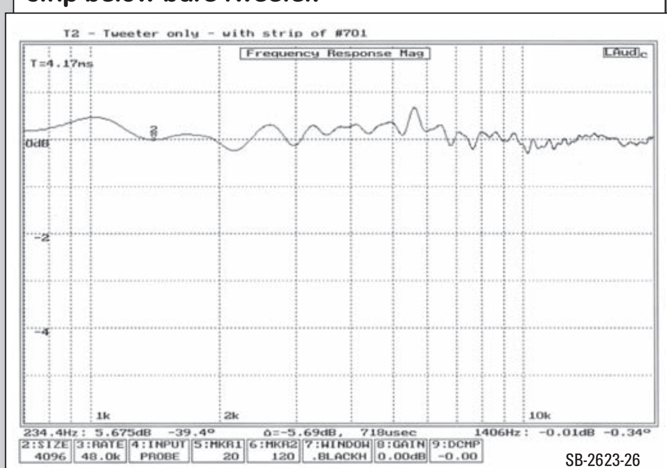


FIGURE 26: Expanded error magnitude curve for #701 strip below bare tweeter.



midrange present might cause tweeter response problems, because these materials do themselves reflect sound. Second, do the strips reduce the effect of the midrange on the tweeter's response?

I prepared strips of the three materials that were 6″ long, 1″ tall, and 1½″ wide. With the 1½″ face placed against the baffle, the strip nearly fills the gap between the midrange and tweeter and covers the full width of the two drivers. I performed the testing with the microphone back 1m.

I replaced the midrange with a solid insert and installed the #701 strip below the tweeter. **Figure 25** plots the error curves for the "ideal" tweeter when the #701 strip is installed. An expanded view of the error magnitude (**Fig. 26**) shows some small magnitude ripple. The CRC plot (**Fig. 27**) shows that the reflection from the #701 is down 20–30dB relative to the bare tweeter response and covers most of the frequency band.

Figure 28 presents the CRC curve with the #703 strip installed. Compared to #701, the reflection starts to show the common bandpass shape. The low- and high-frequency levels are reduced, but the reflection level runs around -25dB from 2kHz to 7kHz.

Finally, **Fig. 29** displays the CRC curve with the #705 strip installed. Compared to the #703 reflection, the high frequency

FIGURE 27: CRC plot for #701 strip below bare tweeter.

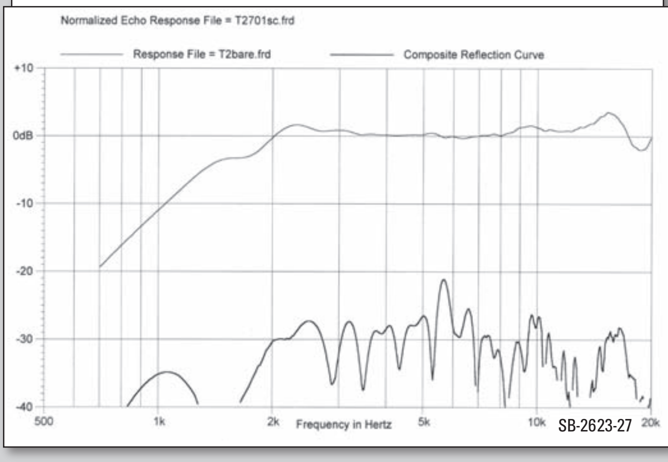
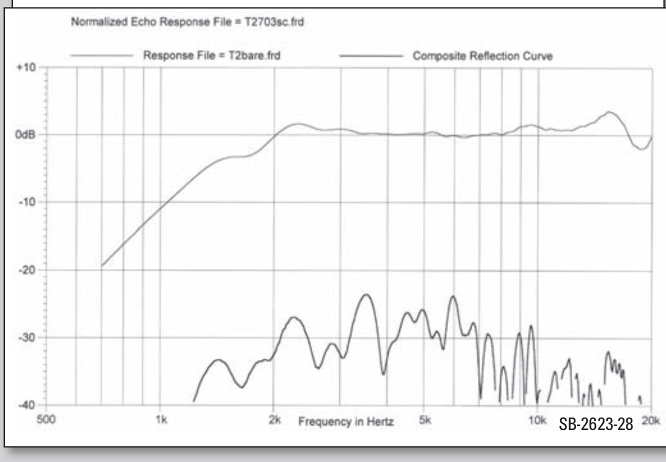


FIGURE 28: CRC plot for #703 strip below bare tweeter.



is about the same, 2–7kHz a bit worse, and low frequencies improved. The #705 material seems great at absorbing lower frequencies, but shows a high reflection level in the midrange frequencies. When the damping strips are used with the midrange installed, it will be interesting to see whether the improvement in driver interaction is enough to offset the reflections from the strips themselves.

DRIVER INTERACTION WITH STRIPS

I then installed the midrange in the test baffle with no damping strip. Because this was a new test setup, it was necessary to repeat the test of midrange interference on the tweeter's response. The CRC curve for this setup is shown in **Fig. 30**. It shows the same bandpass shape as in **Fig. 21** with the 3–5kHz wiggles replaced

FIGURE 29: CRC plot for #705 strip below bare tweeter.

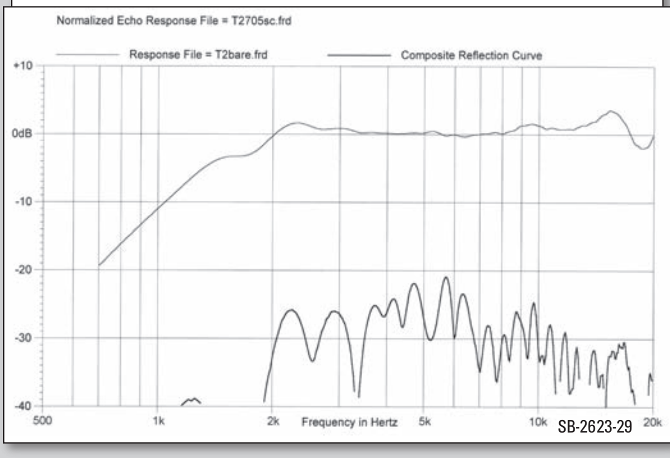


FIGURE 30: CRC plot for tweeter with midrange mounted—no strip.

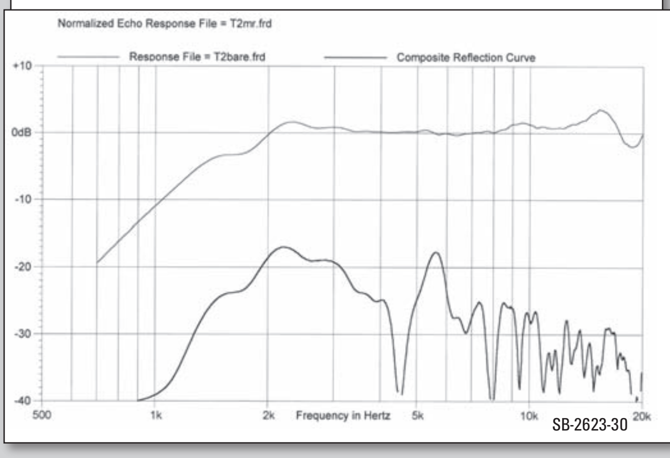


FIGURE 31: Error curves for #701 strip between tweeter and midrange.

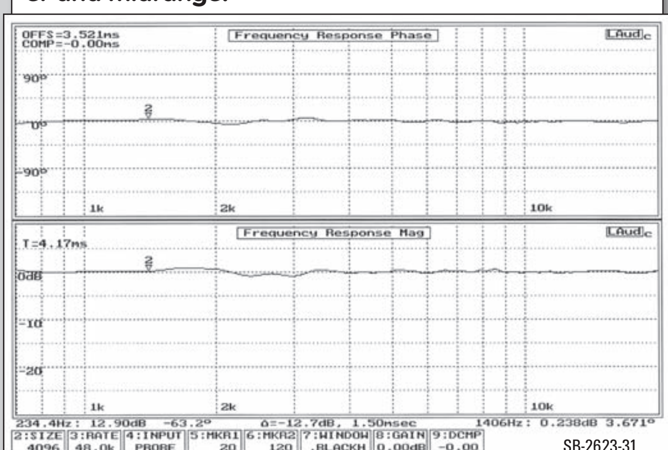


FIGURE 32: Expanded magnitude error curve for #701 strip between tweeter and midrange.

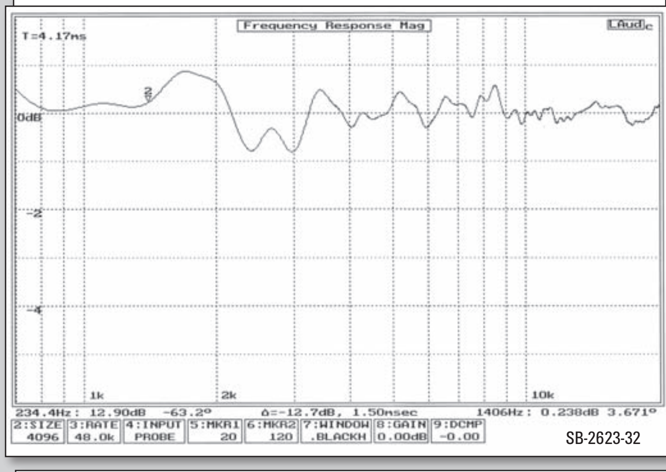


FIGURE 33: CRC plot for #701 strip between tweeter and midrange.

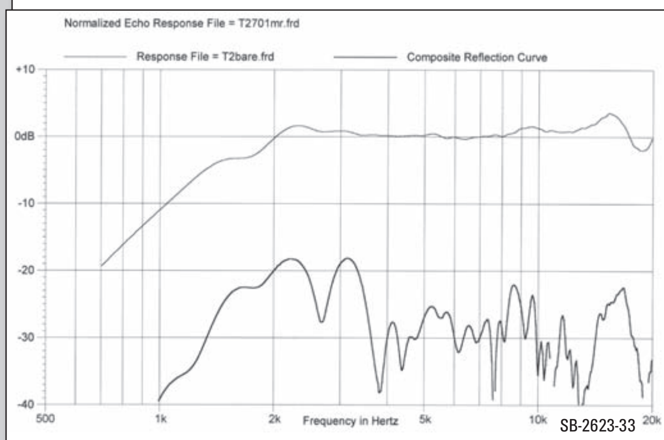
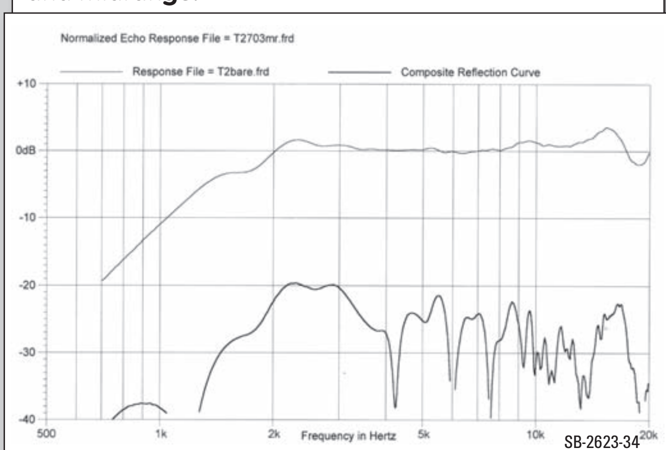


FIGURE 34: CRC plot for #703 strip between tweeter and midrange.

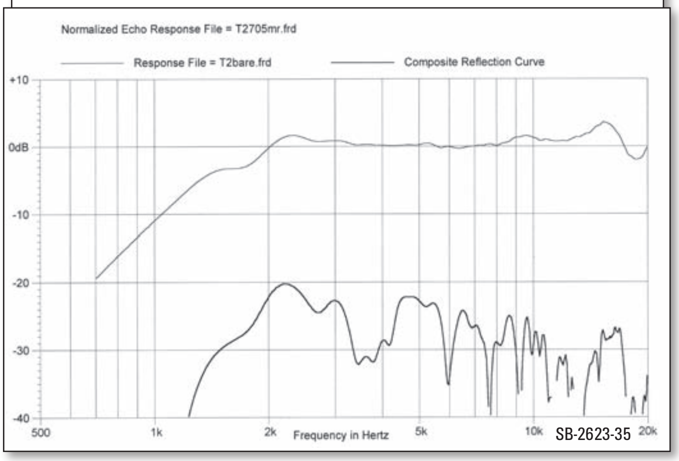


with a single dip. **Figure 30** is the plot to which improvements made by the damping strips should be compared.

I installed the #701 strip (see the error curves in **Fig. 31** and an expanded view of the magnitude errors in **Fig. 32**). The resulting CRC plot (**Fig. 33**) displays only modest improvement. The interference level is not improved in the 2–4kHz range, somewhat improved in the 4–8kHz range, and worse at high frequency due to the #701 strip's own reflection. Because the tweeter would probably be used only above 3–4kHz, it appears the #701 strip would do more harm than good.

Figure 34 presents the CRC plot with the #703 strip installed. This strip shows some low-frequency improvement, but is worse above 5kHz where the tweeter would be used. The CRC plot with the #705 strip installed (**Fig. 35**) improves in the low-frequency area, but again is about the same or worse in the high-frequency range where the tweeter would be used.

FIGURE 35: CRC plot for #705 strip between tweeter and midrange.



In general the strips of high-density FG did not improve performance with a midrange driver mounted near the tweeter. Perhaps the felt used by Mr. Ralph and some varieties of foam materials would do better and should be tested. My past testing indicates felt is very good at high frequencies, while not very effective at low frequencies.

Part 2 will continue our look at interference problems affecting driver response. *aX*

REFERENCES

1. G.R. Koonce and R.O. Wright, "A Modest-Cost Three-Way Speaker System, Part 3," *Speaker Builder* 8/96, p. 34.

2. G.R. Koonce with Kim Girardin, "Testing Front-Panel Damping Materials," *Speaker Builder* 5/98, p. 32.

3. David B. Weems and G.R. Koonce, *Great Sound Stereo Speaker Manual—Second Edition*, McGraw-Hill, 2000, pp. 172–179.

4. David Ralph, "Diffraction Doesn't Have to Be a Problem," *audioXpress* June 2005, p. 55.

5. Jim Moriyasu, "A Study of Midrange Enclosures," *Speaker Builder* 7/2000 and 8/2000.

6. Joe D'Appolito, "Testing the Parts Express MTM Kit," *audioXpress* 4/05, Fig. 12, p. 61.

APPENDIX A: HOW TO TEST FOR AND COMPUTE THE COMPOSITE REFLECTION CURVE

DEVELOPING THE CRC MAGNITUDE CURVE

This work uses Liberty Instruments' Audiosuite measurement system, but the approach should be applicable to any test set that allows you to define the curve used for calibration (Cal). It is important to keep the test driver and microphone positions fixed throughout the test sequence. You must also keep the test signal level and all gains constant.

Remember that you will be performing computations on very small magnitude differences, so you need to maintain all the accuracy possible. I recommend using the MLS signal while averaging several tests. Also try to keep the noise in the test area to a minimum.

It is important for accurate results that you properly select the microphone distance and FFT window time. You need to make a valid far-field measurement, meaning the microphone should be back at least three times the maximum extent of the object under test. For multiple drivers the maximum extent is the largest of the total height, width, or diagonal of the multiple drivers; for a round GF it is the OD, and for rectangular GFs it is the outside diagonal. Also, the FFT window used must be long enough to include all the major "echoes" within the maximum extent of the test configuration and must be kept constant throughout the testing.

The first step is to test the bare driver (without the interfering object) in the normal way, allowing Audiosuite to generate a normal Cal correction. This produces the bare response for the driver under test to use as a reference. If you plan to compute the phase shift for the CRC plot, then the procedure from here on will change as discussed later. For CRC magnitude only computation, you would now export the bare driver response as an ASCII file.

You now again measure the bare driver response, but in single-channel mode with no calibrations. Save this curve and make it the special Cal curve for further testing. This curve contains the raw driver response along with any errors in the amplifier, wiring, and microphone and additionally contains the severe phase shift caused by the transit time from driver to microphone. When it is used as the Cal curve, it will subtract all these items from whatever you test next.

To verify the test approach is working properly, test the bare driver a third time using the special Cal curve just mentioned. This generates the error curve for no change in the test system. If everything is perfect, you are subtracting two identical curves, producing the results of straight lines at 0dB and zero degrees phase shift.

Figure 1A displays such results for the bare Terra tweeter. **Figure 2A** is an expanded view of the error magnitude which really

FIGURE 1A: Error curves for repeat test of bare Terra tweeter.

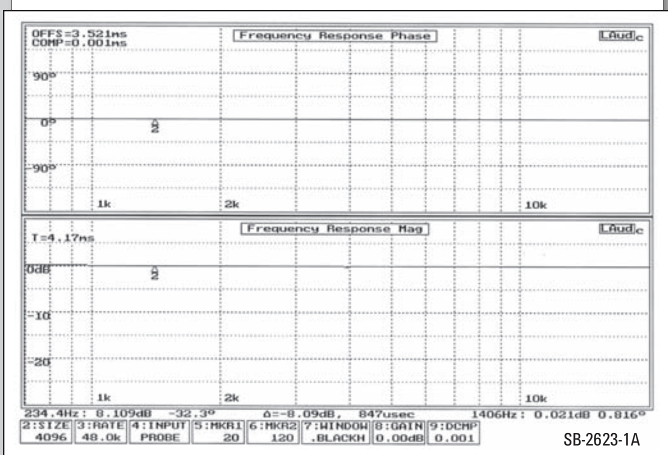


FIGURE 3A: CRC plot for repeat test on bare Terra tweeter.

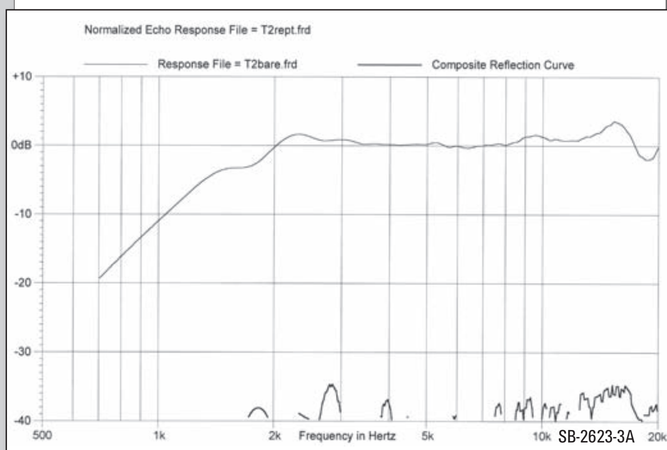


FIGURE 2A: Expanded plot of magnitude error in repeat test.

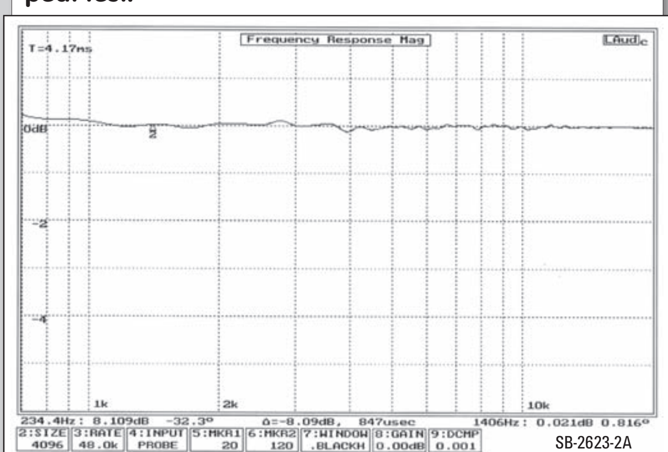
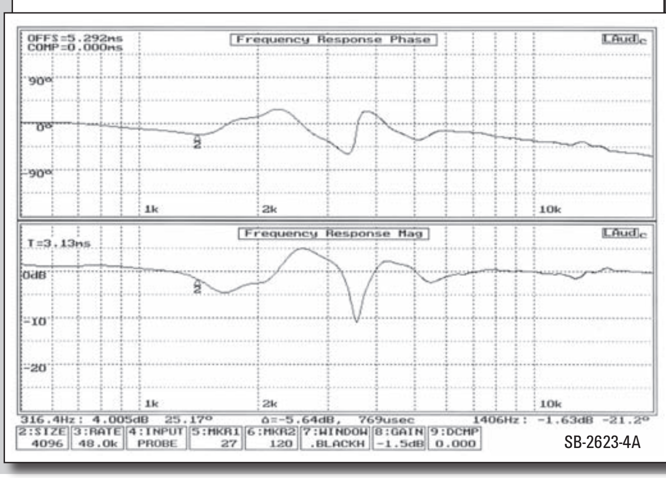


FIGURE 4A: Error curves for round GF with felt, no time correction.



moves about a fraction of a dB. The error curves were exported as an ASCII file and processed by the CRC program. **Figure 3A** gives the results indicating the test approach has a background interference level of -35dB or so. You need to have your procedure "clean" enough that you can produce this kind of background level.

At this point you make the change that represents the interference you choose to compare to the bare driver. Be careful not to move the test driver or the microphone. Now measure the response of the driver with interference using the special Cal curve.

Figure 4A shows the results when this was done for the round grille frame with four layers of fiberglass. The magnitude curve is fine but the phase shift curve has some error, as it should go to zero at high frequency. This error is due to something being physically moved or a change in the acoustic origin[α] of the driver.

I found that interfering objects tend to slightly change the acoustic origin position, causing a phase error. You must use Audiosuite's Compensate to insert a time correction to move the phase shift to zero at high frequency. The phase curve in **Fig. 4A** took a time correction of 9μs to yield the phase curve back in **Fig. 9** of the text.

The time corrections are usually small, but if you do not make them the CRC magnitude will show a rising amplitude at high frequency which is not valid. If you move the driver's acoustic origin, such as by comparing a surface-mounted versus flush-mounted driver, the time correction will be much larger and it is critical that it be made. Once you have the error curves the way you want them, output them as an ASCII file.

While you could pick data out of the ASCII files to make the computations by hand at a few desired frequencies, the better approach is to write a short computer program to take in the ASCII files and then compute and plot the CRC magnitude curve. The program reads in the bare driver ASCII file and uses only the magnitude values, unless you choose to plot phase shift as covered later. The program reads the error curves ASCII file where it uses both the magnitude and phase shift information.

MAKING THE CALCULATION

The data shown in the error file is the vector sum of two items. The first is the "ideal" driver response at 0dB and zero degrees phase shift at all frequencies. This vector is thus always magnitude = 1, angle = 0. The second item is the CRC, which you want to recover. Thus the CRC is a vector equal to the error vector minus the "ideal" driver vector. At each frequency:

CRC vector = ((Em * Cos(Ep)) - 1) + j
(Em * Sin(Ep)) in polar notation

where Em is the error signal magnitude (as a number, not in dB), and Ep is the error signal phase (in the format your computer program requires). The * symbol means "times."

The real part of the CRC vector =
Re = ((Em * Cos(Ep)) - 1)

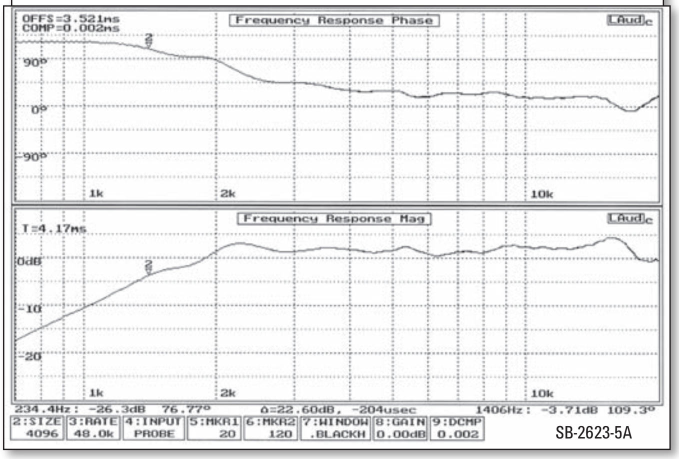
The imaginary part is = Im =
(Em * Sin(Ep))

The CRC magnitude = |CRC| =
√((Re * Re) + (Im * Im))

The CRC magnitude in dB would
be = 20 * Log(|CRC|)

Now this gives the value of the CRC normalized to the "ideal" driver response.

FIGURE 5A: Hilbert Phase for Terra tweeter with midrange mounted.



To re-normalize it to your actual driver, add the dB value of the bare driver response at each frequency. For a plot that is easy to understand, I generally move the bare driver response so that it sets on 0dB. You can do this by adding the needed constant dB value to both curves (bare driver and CRC) at each frequency.

THE CRC PHASE SHIFT CURVE

The CRC phase shift curve is more difficult. The computation is not difficult, it is the re-normalization that causes the problem as you need the bare driver true phase shift data. The phase data normally exported with the bare driver ASCII file includes the phase shift caused by the sound transit time from the driver to the microphone. So for CRC phase shift plots, you must start right with the testing to obtain the proper bare driver phase shift curve.

There is a simple, but not always accurate, approach. Audiosuite will compute the Hilbert phase shift, which is the driver phase shift based on its frequency magnitude response assuming the driver is minimum-phase throughout the test range. So one approximate approach is to have your test set substitute the Hilbert phase shift for the bare driver response before you export the ASCII file.

The better approach is to try to establish the actual driver phase shift response by adding the right time compensation to remove the transit-time phase shift. This can become a bit of trial-and-error. Keep in mind that Audiosuite's preamp contains a sometimes confusing phase inversion. I solve this by connecting the driver under test with inverted polarity.

After saving the measured bare driver response to a file, I have Audiosuite do the Hilbert transform and print the response with magnitude and phase shift. This gives me an idea of what the phase should look like when I recall the original bare driver response and play with compensation times. **Figure 5A** shows the Hilbert phase shift for the Terra tweeter with the midrange present. **Figure 6A** shows the measured phase shift generated by adding the proper time compensation. The differences are slight except at the frequency extremes.

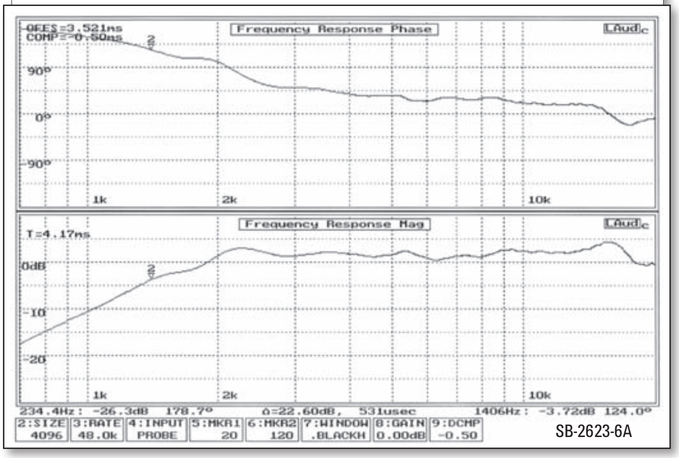
Once you have exported the bare driver response with the proper phase shift, testing and computation, proceed as above for no phase shift curve. Once you compute the real (Re) and imaginary (Im) values for the CRC vector, you must then compute the phase shift.

CRC phase = Arctan(Im/Re)

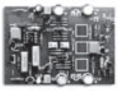
Make sure that you compute the phase in the proper quadrant. This gives you the phase shift for the CRC relative to the "ideal" driver. To re-normalize this to the actual driver, add these phase shift values to the bare driver phase shift values. You could then plot the CRC phase shift relative to the source signal.

I did not believe that the phase shift curve added enough information to be worth the trouble. The magnitude of the CRC curve tells you how much trouble you are in. Phase shift determines direction—whether you get a response dip or peak—and weight of the CRC magnitude applied against the reference response. If you modify your construction so that the CRC magnitude is reduced sufficiently then you don't care what the CRC phase shift is. ■

FIGURE 6A: Measured phase for Terra tweeter with midrange mounted.



α: *Most of you were probably expecting the term "acoustic center" for the location in space from which a driver appears to radiate. In a personal communication, GLA pointed out to me that because acoustic center had two definitions in the audio field, the preferred term for the driver radiation location is now "acoustic origin."*



Back

A Solid-State Single-Ended Power Amp

Discover the enjoyment of listening to a single-ended amp . . . in a solid-state design!

By Ed Simon

William of Occam (born Occam, Surrey, England ca. 1285, died Munich, Germany 1349) is frequently credited with being the engineer's philosopher. What is frequently called Occam's razor is a medieval principle of economy that can be translated as "plurality should not be assumed without necessity."[1] This is frequently reduced to "the simplest possible explanation is best" or "keep it simple, stupid."

DESIGN GOAL

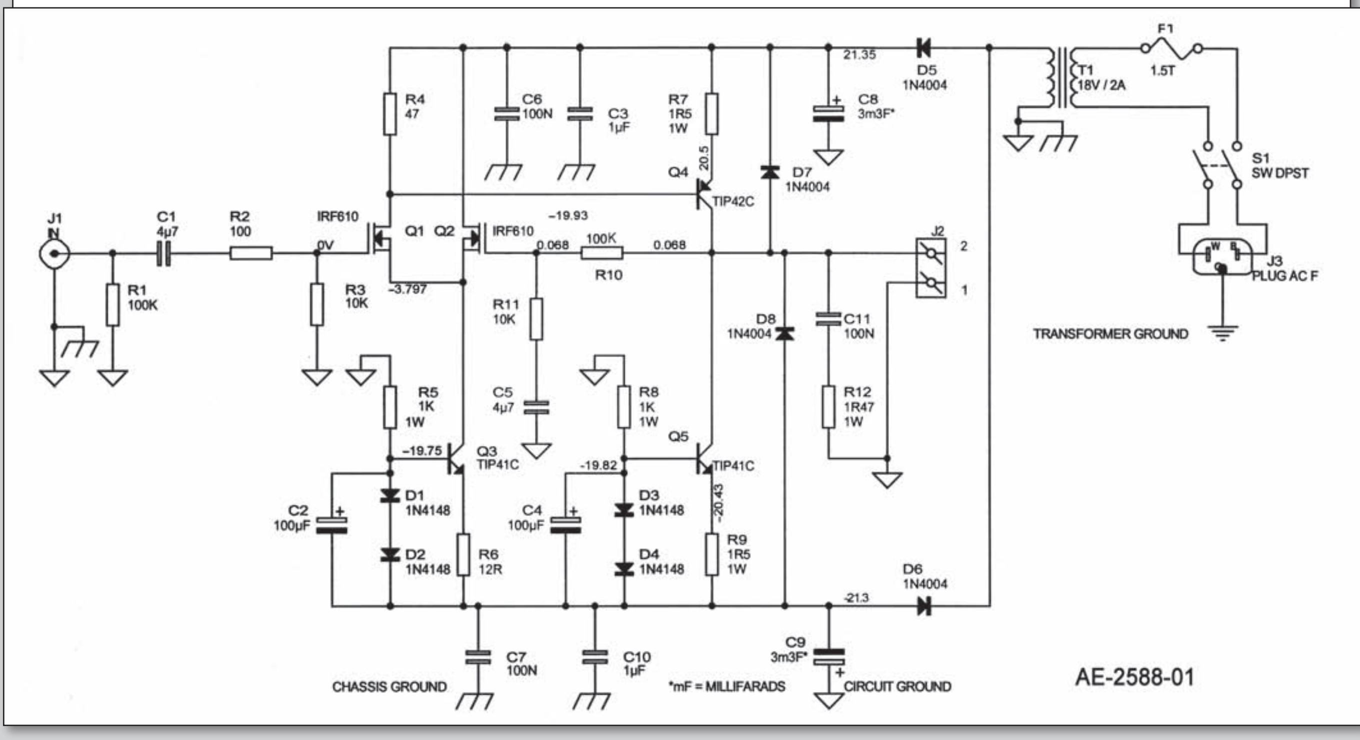
When I design using this philosophy, I must first decide what the goal is and then use the simplest possible method to obtain it. My simplest goal would be to reproduce music.

I then consider music as an emotional expression transmitted in three dimensions through a pressure modulation in the air. It comes to me encoded as a time-varying voltage of one or more channels. I wish to return this to an emotional expression, but I do not have all of the original information; by the very nature of the recording process, information is lost. I therefore do not feel guilty about dramatizing what I have to work with, to increase the emotional content in keeping with my interpretation of the original performance.

The engineering interpretation of this goal in building an amplifier involves, first, the easy stuff; that is, the bandwidth, starting with high frequency response. In the past, using a digitally synthesized signal to produce waveforms with equal peak amplitude but different time delays between tones, I established a design goal for my listening threshold as less than 5° of phase shift at 20kHz.

I like good low-frequency response. Even though very few rooms can allow

FIGURE 1: Amplifier circuit.



a good loudspeaker to fully express itself, it is simple and inexpensive to make sure that the amplifier is not limiting the rest of the system. I have not determined a rule of thumb about the exact low-end design values, and my current rules are subject to modification as better speakers appear.

The power for an amplifier is a bit trickier. I have observed from previous designs that each amplifier has a sweet spot for the level at which it works best and the loads that it likes to see. Some amplifiers are very fussy and others are not.

I have a wide assortment of loudspeakers. Some are very inefficient, others can do quite a bit with 1W. This particular amplifier design is built to be scaled. The best-sounding amplifier I ever built had an output of only one-fourth of a watt. Unfortunately, there does not seem to be a loudspeaker that would use this to produce a practical listening level.

One of the other fun tidbits to be considered is that amplifiers put out voltage, but most loudspeakers use current. The smallest version of this amplifier can produce about 15V RMS and can safely drive very low impedances. However, the current output is not symmetric. It can push several amps into the loudspeaker, but can pull back only a fraction of that. It is not what many would consider a good design, unless, of course, they actually listen to it.

The assumption that a loudspeaker requires symmetric current sources is a "plurality" that "should not be assumed without necessity." When you apply a positive current to a loudspeaker, its cone should move inward, increasing the pressure in the box. When you release the current, the cone will move out on its own. This is, of course, a gross simplification of what is going on—just blame any problems on the speaker designer.

Distortion is another concern of amplifier design. As you can tell from the previous paragraph I am not particularly worried about harmonic distortion at high levels. The amplifier should be able to deliver low distortion of the components of the music that are important to transmit the emotion. To me this means making sure the distortion inherent in all amplifiers does not get in the way, but not to chase one measurement of distortion just because it is possible.

Settling on a circuit topology is where my biases show up. I like Class-A amplifiers. There is also floating around the audio world a superstition that amplifiers using a single output device sound better than ones that use many. The best amplifier I ever built followed this principle, but was not practical. So the basic first design cut is to scale up that amplifier.

POWER REQUIREMENTS

Look at the output transistor first. What is the most power a single device can yield? Power transistors are rated at 200W or more. The fine print, however, says 200W at 25° C.

As you actually put power through the transistor, it heats up. According to the data sheet, you derate it 1.14W for every degree of temperature rise. Without a heatsink, that 200W transistor is actually good for about 5W. At this experimental stage, I will not use the biggest transistor, but I will start with cheap ones so that mistakes won't hurt as much.

The TO-220 package is the most popular case for moderate power transistors, with a wide selection of devices under a dollar. A typical transistor in that package is rated for 65W, derates at 0.52W per degree, and has a thermal resistance of 1.67° per watt. A very optimistic assumption would be an additional thermal resistance of 0.5W per degree each, for both the heatsink and mounting method.

Using a bit of algebra shows the transistor can safely operate at 27W without any safety margin. Interestingly, power transformers are rated for 115V input, and the AC line can run as high as 132V. So you should allow for a bit more voltage increasing the power dissipation.

It would be nice to assume the amplifier will run at 25° C. If placed in a cabinet, expect a temperature of 60° C. So 27W is way too high. I will use 15W as a first-try design value.

A 15W amplifier would be about 11V RMS into an 8Ω load. That would require a supply voltage of 15.5V and a supply current of 1.94A. If, for some reason, the worst-case stress were placed on the amplifier, the transistor would need to dissipate about 30W. The actual load placed on the output transistor cannot be calculated because the exact load is an unknown combination of resistance, inductance, capacitance, and back EMF. The load also has a habit of changing.

In small power transformers the three most common voltages are 12, 18, and 24V. I will use an 18V transformer, which should give a peak voltage of:

$$18 \times 1.414 \times 130 / 115 \times 1.2 = 34.5\text{V}.$$

That is the minimum voltage the filter capacitors should be rated. The voltage to really expect depends on the transformer. I am using one rated at 18V 2A.

I have learned that when you run a loudspeaker at high power its voice coil heats up and it draws less current. In order to get more voltage from the 15W thermal limit, I will allow more voltage and less current. The current draw for my first target will average only about ½A.

The current will only be drawn when the peak AC voltage is greater than the DC voltage stored on the filter capacitors. Allowing for 5% ripple at 18V AC (25.45 peak), current will only be drawn when the AC peak voltage exceeds 24.18V. The inverse cosine of 24.18/25.45 is 18.19°. So there are only 18° of the 180 during the positive half cycle to charge the filter capacitor. My ½A through the output transistors is now a 5A load on the transformer.

The data sheet for the transformer states that at no load it produces 20.6V and 18V at 2A load. (That is why there is a 1.2 in the maximum voltage equation.) This would be modeled by a 1.3Ω internal resistance. At 5A it should lose 6.5V. So expect:

$$20.6 \times 1.414 - 6.5 - .8 = 21.8\text{V}.$$

The .8 comes from the diode drop. It is larger than the .7 most folks assume because of the higher current. So for my ½A the power from the transformer should be 11W. It's not.

There is some other bad news for the transformer. The amplifier does not have symmetrical outputs, so the load on the transformer will not be symmetric. This means when I play loud music there will be a DC bias magnetizing the core of the transformer and reducing its capacity. I could use a bigger transformer, but I will stick with my 18V 2A unit for now, using a much bigger one for a larger version of the amp.

This power supply leaves about 22V for the output transistor. In a Class-A amplifier the current always flows through the transistor. If the current is set for 600mA, the transistor will dissipate about 13W, which is close enough to allow for component and temperature variations.

I have mentioned the main filter capacitors before, but at first pass sizing them is simple, using electric field theory, not circuit theory as has been used up until now. Field theory is rarely used to design systems, because it is way too cumbersome although more accurate.

The definition of capacitance is $C = Q/V$. (Q is charge, or the number of electrons.) With algebra you rearrange this to be $CV = Q$. You then use calculus to take the first derivative:

$$C\,dV/dT + V\,dC/dT = dQ/dT.$$

Current (I) is defined as dQ/dT. Most folks assume that dC/dT is zero. Drop the term and end up with:

$$C\,dV/dT = I.$$

This is not a good assumption for using capacitors in audio. dC (or the change in the value of the capacitance) can occur from a change in the voltage across the capacitor, a change in the pressure surrounding the capacitor, or even movement of the capacitor. This value does not appear on manufacturers' data sheets. Keep this in mind when a capacitor sees a large voltage swing.

Rearranging the simple form gives me:

$$C = I \times dT/dV.$$

For a first try at picking a filter capacitor value, I will allow 10% ripple because there is now a greater current draw:

$$C = .6 \times (1/60)/2.2 = 5000\mu\text{F}.$$

4700µF is the closest standard value, and I can use either 35 or 50V. A quick check of the data sheet shows that a capacitor of this size can handle ripple current of up to 1.9A. A 50V capacitor will allow me to try a higher voltage transformer, and in some suppliers' stocks of capacitors I can only get that up to 3300µF.

CIRCUIT TOPOLOGY

It's time to figure out the actual amplifier (**Fig. 1**). Starting with the output, I can use either bipolar or MOSFET for inexpensive output devices. (If you were expecting a vacuum tube, there are lots of those designs out there.)

There are three possible ways to hook up the transistors. Common base (or gate) gives voltage gain but no current gain. Common collector (drain) gives current gain without voltage gain. There is both current and voltage gain from common emitter (source), which is a good starting place. A common-emitter amplifier with a fixed current source is the most popular gain stage in solid-state amplifiers. It is reasonably well understood and has a number of fun quirks, such as oscillation, internal parameter changes with applied power, and so on. Always an old friend.

Using a MOSFET as the output device requires driving it with a voltage source that can supply enough current to maintain the design goal bandwidth. A typical MOSFET (IRF610) has in the range of 200pF of input capacitance, and there is also Miller effect capacitance to worry about. Driving this to 100kHz requires a current of less than a few hundred microamps. The output will have a loss of less than 0.5V and meet the power goals. These are pretty good numbers.

The downside is that my selection of MOSFETs is quite limited. They are designed for industrial power handling, and devices designed for audio are rare and harder to obtain (yes, they do exist).

A bipolar transistor will require a drive current of 30mA or so. It will have an output loss of less than 0.4V, and I have a wide selection. If I wish to scale the amplifier up in size, there is a clear path of easily available larger bipolar transistors, some of which have nicely linear gain characteristics. I can add an emitter resistor to obtain some local feedback. The problem is getting enough drive current.

Using a bipolar transistor to drive a bipolar transistor output stage will require an input current of almost a milliamp. This would give me a low input impedance and loss of low frequencies, unless there are more than two stages. A MOSFET has a higher input impedance and is a voltage to current device, so providing the drive is not a problem.

If I use a differential input stage with matched MOSFETs, my higher distortion device will be in a circuit with limited gain minimizing the problem. The match between the two devices will also allow me to reduce distortion. It also makes it easy to apply global feedback.

With just two stages, one device must be of N construction and the other P. The bad news is P constructed devices don't work as well as Ns for audio applications. The difference is worse with MOSFETs than with bipolars. To minimize the damage it makes sense to use an N channel MOSFET as the input and a PNP bipolar as the output.

A single-ended Class-A amplifier needs some current sources. It is easy to make a simple current source, such as a high voltage supply and just a resistor, or become slightly more complicated and use a transformer and high voltage supply (done that—different story), or use a solid-state current regulator. Although the regulator has more parts, it costs the least. So sometimes "plurality has a necessity."

The usual design is a transistor, fed from a constant voltage source, dumping into a resistor. I chose to use bipolar transistors for this because their base voltage is more uniform than MOSFETs. I did a small experiment testing an LED, a power diode, and a signal diode to see how uniform their forward voltage drop was with changes in current. The signal diode had the flattest slope, and its sweet spot was 15 to 30mA. The exact voltage across the diode varies with temperature.

When using a diode voltage source for the output current source, be aware that as the output transistor heats up, its base voltage drop (Vbe) goes down and it passes more current. I will therefore assume 0.9V across the emitter resistor rather than 0.6 or 0.7V in the output current source.

CONSIDERATIONS

The voltage regulators formed by the series connected diodes are also bypassed by a capacitor to reduce any noise and power supply ripple. Any noise in the voltage source is amplified by the output current source.

I now have most of the basic topology of the circuit. It is pretty standard stuff to use a modest amount of global feedback at AC and full feedback at DC. I have elected not to use additional parts until I listen to the amplifier and see what each change or new part will add.

Another really important goal is to be able to actually build the amplifier. In its simplest form this amplifier can be a beginner's project. I built this amplifier using only a drill (with bits), a soldering iron, a few screwdrivers, a pair of pliers, a small diagonal cutter, wire cutter, hot air gun, and voltmeter. I did not lay out the piece with a square and scribe; instead I used my CNC turret punch, which is a bit expensive, so you may choose to stick with the square.

If you make a mistake in building this basic amplifier, it's probably because I forgot to tell you some important detail or you read the words I wrote, not the ones I meant. Long ago I once asked a mentor about Boltzmann's constant. He told me it was "basic." After about two weeks I apologized for not getting it and asked him for more information. He began by saying, "I said it's basic, not easy."

This is a "basic" amplifier. If you have built a few kits and choose to try a scratch build, this may be for you. It certainly helps if you have a friend who can help you with the rough spots.

I have listed all of the parts along with the Mouser Electronics part number. Note that they do not sell some of the parts such as resistors individually. You may also wish to buy a few extra MOSFETs so you can match pairs for the input stage. I expect anyone can scrounge up some of the parts, even if it's only a power cord from the weekly trash. If you can scrounge up the aluminum, the power cord, and some scrap wire, you should be able to build two of the amplifiers for around $100.

TABLE 1: SMALL AMP PARTS LIST.

C1, C5	4.7µF/100V	1430-1475	1.83	3.66
C2, C4	100µF/6.3V	647-UVZ0J101MDD	.22	.44
C3, C10	1µF/100V	1430-1105	.69	1.68
C6, C7, C11	100nF/100V	1430-1104	.25	.75
C8, C9	3300µF/50V	647-UVZ1H332MHD	2.00	4.00
D1-D4	1N4148		.05	.20
D5-D8	1N4004		.04	.16
F1	1.5A 3AG fuse			
	Time delay		.60	.60
1	Fuse holder	441-R314A	1.40	1.40
J1	RCA	161-1004	.69	.69
J2	Binding posts	164-19B2	1.85	1.85
J3	AC Cord	17237 8 B1	2.82	2.82
Q1, Q2	IRF610, A, B		.46	.92
Q3, Q5	TIP41B or C		.58	1.16
Q4	TIP42B or C		.62	.62
R1, R10	100K ¼W	291-100K	.70	.70
R2	100Ω ¼W	291-100	.70	.70
R3, R11	10K ¼W	291-10K	.70	.70
R4	47Ω ¼W	291-47	.70	.70
R5, R8	1K0 1W	294-1.0K	1.40	1.40
R6	12Ω	291-12	.70	.70
R7, R9	1.5Ω 1W	294-1.5	1.40	1.40
R12	.5Ω 1W	294-.5	1.40	1.40
SW1	DPDT toggle	10TE008		2.22
T1	18V 2A	41FJ020	9.36	9.36
5	Transistor kit	532-4880M		3.85
7	6-32 nylon	561-P635-.5		.21
5	Tie strips	158-1008		3.48
2	Tie strips	158-1002		.50
1	12 × 12 × 1/8″			
	Aluminum plate			
5 optional	sockets	571-874995		1.77
15 optional	pins	571-860163		3.15
1 box	6-32 machine screws			
	Box/100	5721-632-3/8		2.40
1 box	Lockwashers	5721-LWI-6		2.46
1 box	Nuts	5721-632		2.40
1	Heatsink grease	567-120-SA		4.40
1	Heat shrink tubing assortment			
1	Hook up wire			

You can build the entire basic amplifier on one 12″ × 12″ piece of aluminum. I used 1/8″ thick alloy 5052 with a temper of H32. The thickness is important to allow the heat to spread out; if you must, you can use two thinner pieces.

If you are shopping for aluminum, another choice is 6062 with just about any temper except T0. You should be able to find a piece of scrap anywhere from a junkyard to a metal shop, and, of course, you can always buy a piece on the web. There are also many other alloys of aluminum which will all work, just don't buy already anodized aluminum.

I used terminal strips screwed to the plate to hold all of the components. That way you can easily make changes. For example, I used a TIP42C, which isn't a great audio transistor, but it's cheap.

You can play with others. Try an MJL4302A. You can adjust the amplifier's sweet spot by increasing the value of R9. You can also try a higher voltage transformer. (Be sure to increase R9 also.) You can add gain stages, try all MOSFETs or bipolars.

I did not use transistor sockets, but I list them in **Table 1** for those interested. They aren't intended as sockets but will work just fine.

CONSTRUCTION

To begin, lay out your piece of metal. You should use a 12″ adjustable square. Set the ruler to one of the dimensions shown on the drawing (**Fig. 2**), place the edge of the square firmly against the metal just above where you want your finished scribed mark. Then place the scriber (which in many models comes stored in the handle) in the notch in the end of the ruler, and pull the tool to you, while keeping the scribe in the notch. This will allow you to place a line within .005″ with just a bit of practice. When you extend the ruler more than 6″, start from the opposite edge except when doing a matching hole (for example, you should measure the two holes that mount the transformer from the same edge).

Scribe all of your lines before using a punch to leave a starting dimple for your drill. If you are careful and have a sharp punch, you can feel the intersection of the scribed lines and precisely punch an indent exactly where you want it.

Machinists actually have three sharp tools. The scribe, which is used only to make scratches, has a very fine tip that would be damaged if struck. A prick punch is almost the same as a center punch except the tip is at a sharper angle. It is used with a light blow to make a small indent. The center punch is then used to make an indent big enough that the drill bit will not slip out.

When done this way your hole will

be exactly where you want it. Of course, for normal folks nails work well, but you will need to change often as they become blunted.

To make things simple I consider the side with the lines scribed on it to be the face, the other side is the back. The transformer mounts on the bottom.

Put on a pair of gloves and safety glasses, then drill all of the holes with a 5⁄32″ drill bit. If you have trouble staying in your center punches, use a smaller bit. If you use too small a bit, the aluminum will gum it up and break it. It is a good idea to dip the drill bit in oil or even soapy water before you drill each hole.

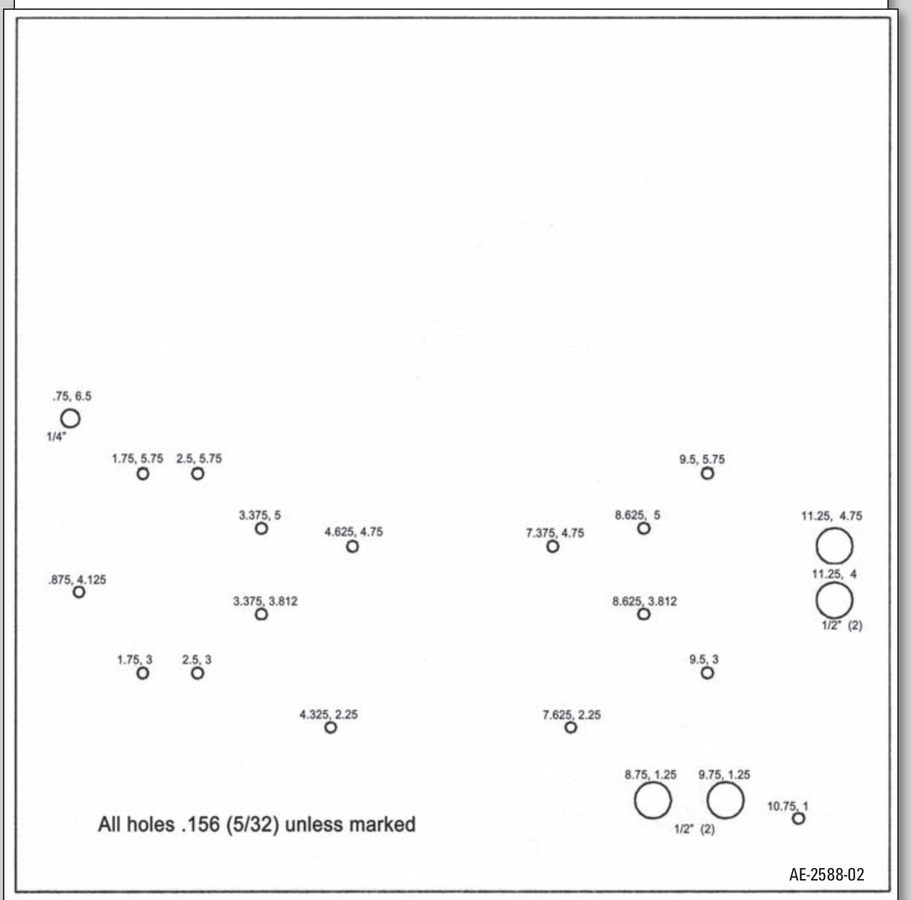
After drilling all of the holes, you can simply go back with a larger bit to enlarge the holes. When you do this, step up gradually to the final size, checking for fit as you go. You can go from 5⁄32″ to ¼″ to ⅜″ to ½″. Check the holes for the switch, fuse holder, binding posts, and input jack as you drill. Bigger is easy, smaller is not. You may need to wiggle the drill bit in the hole to get a precise fit for some of the parts.

After all the holes are their final size, sand the aluminum before painting it. I use 120-grit sandpaper in a small electric palm sander. You can do it by hand; it doesn't take very long. The idea is to remove most of the layer of oxide that has formed on the aluminum so the paint will stick to it. After you have removed the dull gray aluminum to reveal mostly bright aluminum, you can wipe it with a dry paper towel or one lightly dampened with solvent. Do not use water.

Quickly mask all of the mounting holes on both sides of the plates. I use masking tape cut into ⅜″ squares. On the face side (the way it looks in **Fig. 2**) use ¾″ pieces to mask the five transistor mounting holes. On both sides mask the ¼″ hole for the input jack. It is not necessary to mask the holes for the binding posts, the fuse holder, or the switch.

The process from sanding the metal to painting it should take less than 15 minutes. I prefer the cheapest black spray paint, which is usually thin and dries fast. I spray one very light coat from all four edges on the back side, wait two minutes,

FIGURE 2: Small amp drill guide.



flip it over on three small blocks, and paint the other (face) side. It takes me three times around all four sides with a really, really light coat to get the aluminum uniformly black. I do not want a thick coat on the face.

At this point there is the finish coat on the face and a protection coat on the back. After the face is almost dry (the paint I use takes 10 minutes), remove the masking tape. When the face is dry (about 1 hour), flip it over onto the blocks and put just enough paint on the back to make sure it is uniformly black. After a few minutes remove the rest of the masking tape. By taking the tape off while the paint is not completely dry, the edge of paint moves into the masked area very slightly and leaves a smooth transition. Then let the panel dry overnight.

For my "case" I used just a 12″ piece of 2 × 4 with a ⅛″ slot cut in it off center. The plate just rests in the base and stands vertically (**Photo 6**).

ASSEMBLY

After the panel is absolutely, completely, and really dry, you can begin assembly. Start by mounting the transformer on the face side with a piece of thin cardboard underneath it, using nylon 6-32 machine screws. Mount a two-terminal strip to the right screw. Be sure the fuse is not in the fuse holder and mount it so that the terminals come up on the face side. Only the nut should be on the face side; the rubber washer should be on the bottom.

Snugly tighten the nut. If you overtighten it, you will break the fuse holder. If it is too loose the fuse holder will spin when you try to change the fuse. If you want to cheat, use a drop of super glue.

Next mount the switch, again with terminals to the face side. If you use two nuts on the switch, tighten it from the face side to get a trim appearance on the back. The power cord should be clamped firmly in place.

I leave about 4″ of the three conductors exposed from the outer jacket of the power cord. The green wire is connected to the grounded terminal attached to the power transformer. No heat-shrink tubing is needed here. This makes a safety ground without introducing extra noise into the audio heatsink-ground plane. You can use the other isolated terminal to store the white center tap lead from the transformer.

Place a piece of heat-shrink tubing over the black (or brown) and white (or blue) wires of the power cord. Then hold the tubing away from the ends as you solder the leads to the switch terminals closest to the edge of the switch, one wire to each side. Black or brown—depending on the power cord you have—goes closest to the transformer. From the center terminal of

PHOTO 1: AC supply.

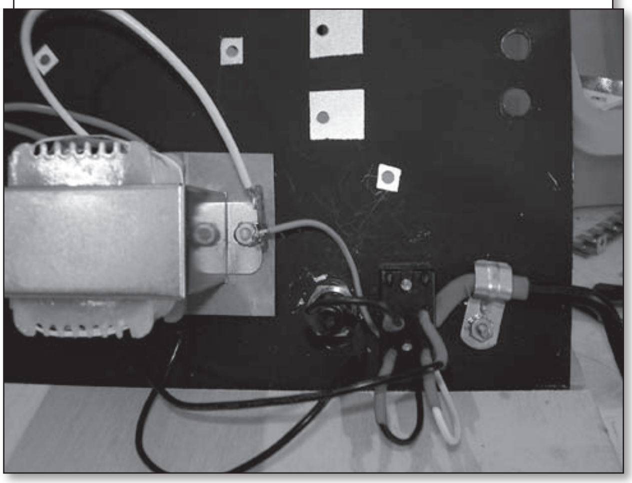
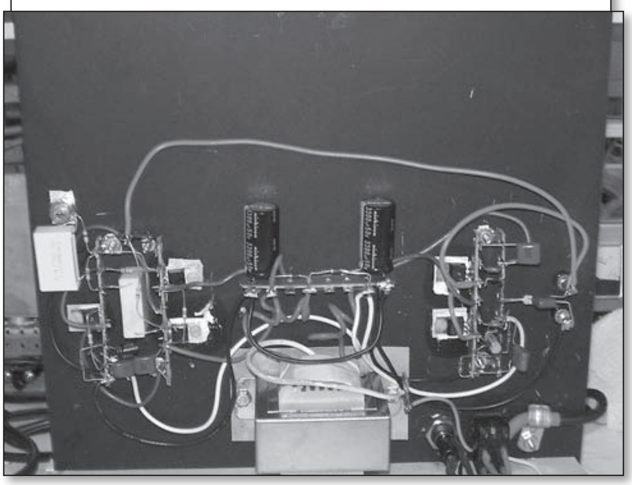


PHOTO 2: Complete amp.



the switch on the side that the black wire is connected to, attach a wire that also goes to the rear terminal of the fuse holder. Do not forget the heat-shrink tubing!

The two black transformer wires are now ready to be connected, again with the tubing on the wires—one goes to the open terminal on the switch and the other goes to the fuse holder's terminal closest to the panel. Check to be sure your solder joints are shiny and firmly melted to both the wire and its terminal before you slide the heat-shrink tubing over the exposed metal. You can use a hot-air gun, a hair dryer, a match, or a lighter to heat up the tubing until it shrinks firmly in place. Now wrap everything with electrical tape so that you are sure no metal is visible. This is the only place you have enough voltage to hurt you (**Photo 1**).

You can now put in the fuse. This is a good time to check your work.

TESTING

If you have the parts, you can build an AC test jig, which is simply an AC line cord, an outlet, and a 60W light bulb. Wire the ground and neutral (green and white) from the cord to the outlet. The hot (black) from the cord goes to the light bulb and the other light bulb lead goes to the outlet. Be sure to build this device using standard electrical box parts and strain reliefs. To test this device, plug a lamp into it and the device into the wall. When you turn on the lamp, it should light as well as your test jig's 60W light bulb.

If you are brave, you can just plug your amp straight in; either way you should measure about 21V AC on the two red wires. If you don't, try turning on the power switch. Now be sure to unplug it before proceeding.

The next step is to mount the terminal strip above the transformer that holds the power-supply capacitors. You will now find out whether you put the holes in the right place. Push the steel machine screws in from the bottom. They are a tight fit going through the mounting feet of the terminal strip. Use a lock washer and nut on the face side. These screws should be fairly tight.

I mount the capacitors to the hole in the strip where it is riveted to the insulator. This is not the normal method, but it gives me a good fit and keeps the capacitor close to the ground plane. Connect the left capacitor so that its stripe showing the negative lead is connected to the grounded terminal. The capacitor on the right has the positive lead grounded.

Connect the diodes using one of the center terminals for both and the other end of each going to the capacitors. Both

PHOTO 3: Input section.

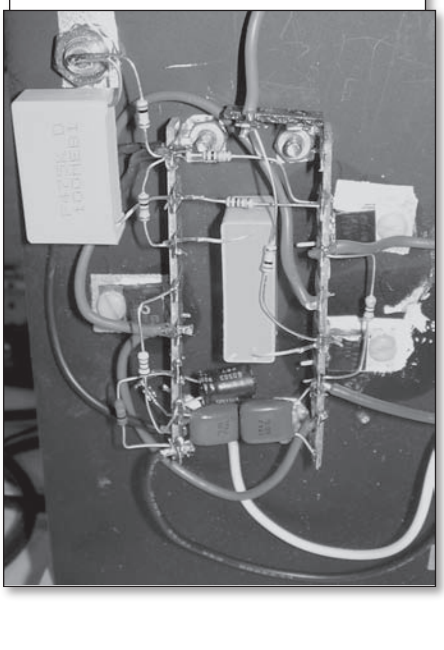
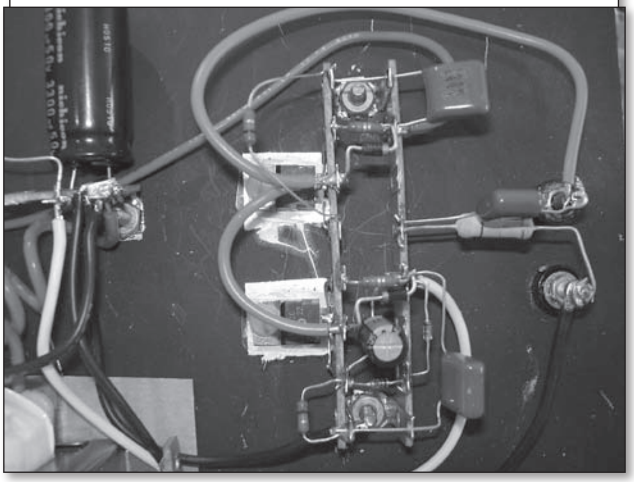


PHOTO 4: Output section.



diode bands should be on the left side. Connect the transformer leads and a jumper between the two grounded terminals. I don't want to just trust the mechanical connection to the ground plane. After you have this hooked up, you should test your work.

Plug the amp back in. If you are using a test box, you may notice a brief bit of light as the capacitors charge. I didn't. Now quickly unplug the amp. Because the capacitors have no load, they will hold a charge for a while.

You have a good chance now to be sure you didn't wire anything backward without risking an explosion. (Capacitors wired backward can explode!) You should measure about 28V on each capacitor. Be sure that positive is actually positive on both. If you have reached this point, you should have made at least one mistake by now and have figured out a way to hide it or fix it!

The next step is an option to make a temporary circuit to sort the MOSFETs. I connected one of the 1K0 1W resistors to the power-supply terminal strip and connected this to a .1″ spaced triple socket. Pins 1 and 2 went to the resistor, which I then connected to the positive supply. Pin 3 went to ground. I then measured the voltage between 1 and 3.

Without a MOSFET in the socket it was about 28V. Inserting a MOSFET dropped it to 2.78V for four of them and 2.79V for five of them. One of the ten I bought measured 2.77V. This gave me four matched sets and reassured me that the MOSFETs were well-made.

Next, assemble the output stage. Mount two terminal strips with a gap between them using two screws as before. Both strips mount in the same holes. I bent the leads of the TIP41C and TIP42C transistors quite severely to get them to mate to the rivets in their terminal strip. If you try to bend the leads too quickly or too severely, you may break them off. I use the old-fashioned method of mounting the transistors (**Photo 5**).

I put a very small amount of heatsink grease on both sides of a mica washer and then place this between the transistor and the heatsink. I put a nylon 6-32 machine screw through the assembly and tighten the nut from the back. Some folks prefer a steel screw and an insulated bushing; others use a thermal pad that does not require grease. The screw should be snug now and tightened later once the amplifier has become warm. Mount all of the parts to the terminal strips first and, then when you are sure you have it right, solder them. Use bits of wire to interconnect all of the ground terminals.

Insert the output binding posts at this stage and attach the output filter (C11 and R12) now. I used old green posts I had around and just painted them red and black. When you mount your posts be sure that the wire hole faces to the side, otherwise it will be difficult to connect your speakers later.

Once you have installed R7, R8, R9, R12, C4, C8, C10, C11, D3, D4, D9, and D10, you can fire up the amplifier again. The output voltage should be about -26V. Next turn the amp off and temporarily connect a resistor (1K 1W) between the base of Q4 and ground. Turn the amp back on.

The output voltage now should be more positive. It can be anywhere from -10 to +15V. If you are using a light bulb box, the bulb should glow. The output transistors will begin to warm up. The measurement from the base of Q5 to the negative supply should be between 1.3 and 1.4V. There should be about 0.7V across R9.

If it all works, turn it off and take out the extra resistor. If it didn't work, start by checking to make sure you have your transistors connected correctly and the bands on the diodes are right. If you have C4 in backward, it won't make much difference because the voltage is so low across it. However, it will eventually fail. If a resistor smokes, Q4 or Q5 may be shorted.

Once the output stage is working, start on the current source for the input. Install both terminal strips and mount Q3. Install R5, R6, C2, C6, C7, D1, and D2. Once these parts are connected, temporarily connect a 100Ω ¼W resistor from the positive supply to the collector of Q3. Expect to measure about 5 to 10V across this resistor. Be happy! The current source is working. Also, it's helpful to check from the negative supply to the base of Q3; the reading should be between 1.3 and 1.9V.

Next, install the input jack and all of the passive components you can without making it hard to mount Q1 and Q2 (MOSFETs). I have never blown any of these up myself, but I am told that if you try to assemble this amplifier in the winter on a wool carpet sitting in a wool-covered easy chair while wearing silk undies, you certainly will. Handle the MOSFETs as little as possible and then

PHOTO 5: Power supply with test socket.

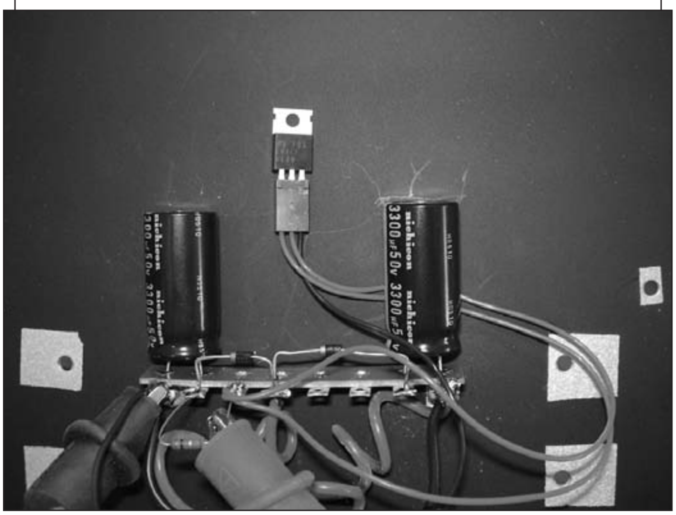
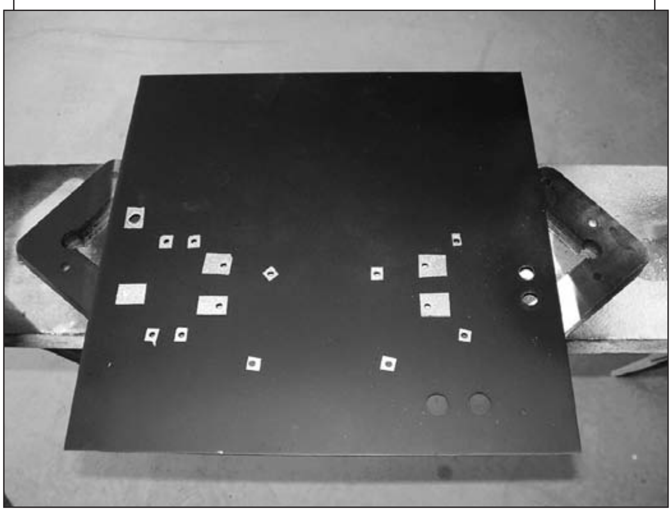


PHOTO 6: Painted chassis.



hold them by the big metal tab (drain), not by the gate. The input MOSFETs do not require heatsinking but do it to make sure they are both at the same temperature. This is important to minimize DC offset at the output.

When you assemble the input, it is important that you mount R2, R3, R10, and R11 close to their transistors. If you don't, oscillations may occur. Once you have assembled everything, take a break.

When you get back, recheck your wiring, making sure you have soldered all of the terminals (I never make this mysstake). Plug the amplifier in without a load. Measure the output voltage. If it is less than .080V, that's great. Measure the positive and negative supplies, which should be 21V, give or take a volt or two.

If all checks out and you have access to test equipment, look at the output with an oscilloscope or computer-based equivalent. The output should be a smooth line without bursts of ringing. Sweep the amplifier with a sine wave; you might see some distortion in the sine wave up past 50kHz. If you see bursts of ringing, be sure C11 and R12 are connected (**Photo 3**).

If that is not it, you can poke your finger around until you see what changes what. An extra capacitor to ground is OK at any power point. I expect that if you do see ringing the most likely problem is either an incorrect hook-up or shorting.

If you don't have test equipment and your meter reads little DC and less than 50mV of AC, put a 1A fast blow fuse in series with an old loudspeaker and hook it up for a second or so. If all's quiet, give the amp an input and plug the speaker back in. You should hear music. If it sounds good, it will sound really good at this point. If it doesn't make music, remove the speaker and check all the voltages and compare them to the ones on the schematic to see whether you can find where you went wrong.

The amp will become hot to the touch, but that's OK. If you put your finger directly over the output transistors and don't scream much, they are fine. The power transformer will also become warm. Any capacitor heating up is a bad sign.

As you listen to your amp, you may not notice why people prefer the sound of single-ended Class-A amplifiers at first. After you listen for a couple of hours, try your old amp again. All of a sudden you will find it sounds funny! In one of those lovely psychological effects, you may not notice when something is better, but you are much quicker at noticing when something is worse.

You now have an amplifier you can both listen to and play with.

If it's occurred to you that this amplifier can be built with a PC card, or scaled up and with better parts, you are right. But then that is a different amplifier. *aX*

REFERENCE

1. Dave Beckett, University of Kent at Canterbury, UK.

Ed Simon *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, New Hampshire.*

Back

Mighty Mouse: A PP 25L6 Amplifier

Don't be misled by this unit's power output figure.

Like its cartoon namesake, it packs a wallop.

By Bob McIntyre

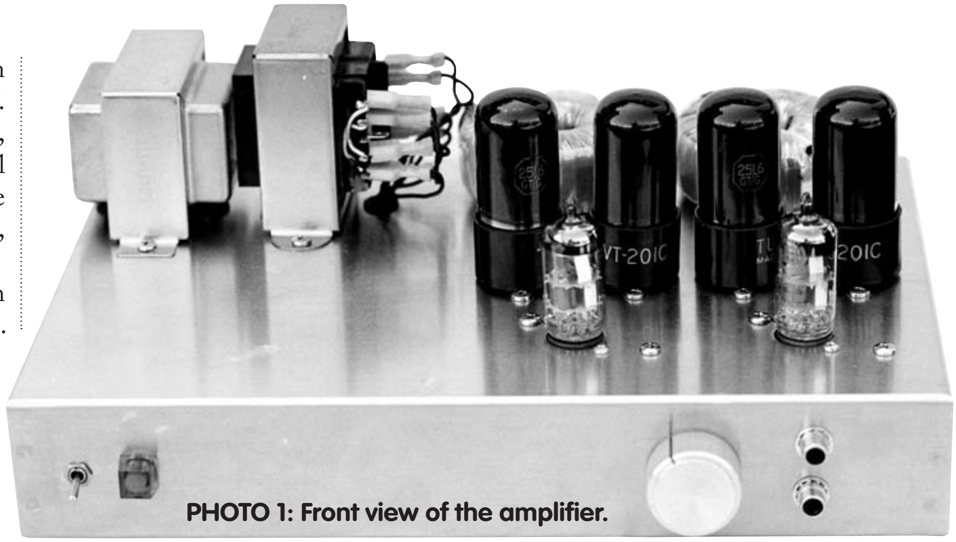


PHOTO 1: Front view of the amplifier.

For years, I've wanted to build an amplifier based on the 25L6GT. If you haven't heard of this tube, it's not surprising, as its principal use was in table radios and portable phonographs. Don't let that fool you, because the 25L6 is a remarkable tube.

To begin with, the tube has high transconductance, so it's easy to drive. It has low plate resistance, so it doesn't require a high plate voltage. Finally, because it was designed for low-cost applications, it was produced by the millions. NOS tubes are relatively cheap and easy to find.

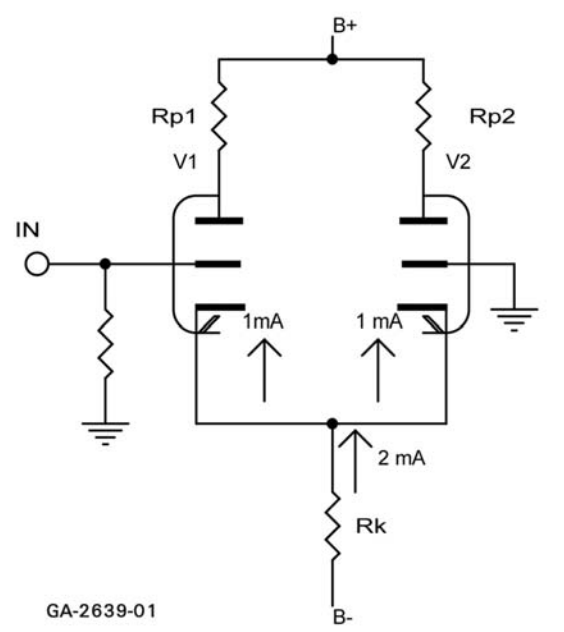
What stopped me was the lack of a suitable output transformer. That low plate resistance meant that the transformer needed to have low impedance and high current-handling capability. All the available transformers that would handle the current were ridiculously large.

Then I ran across Steve Bench's website (http://members.aol.com/sbench101/). Steve discovered that ordinary toroidal power transformers make surprisingly good output transformers. Finally, I had the transformer I had been looking for!

What I ended up with is a miniature amplifier with a big sound. It will make a good monitor amp, a great first project, or a suitable introduction to "tube sound." Power output is a microscopic 3W, but don't let that fool you. This amp has great bass response, and with efficient speakers, is quite capable of room-filling volume.

Besides using cheap, off-the-shelf parts, the Mouse is easy to build and modify. It uses a minimum of feedback—6dB—so it is extremely stable. Frequency response is dead flat from 20 to 30kHz. It even features dual independent power supplies (**Photo 1**).

FIGURE 1: The long-tailed pair phase inverter.



THE CIRCUIT

The output stage is not unusual, except for the zener diodes, which develop a 15V bias in the cathode circuit. Because the voltage cannot change, it acts like a fixed bias. On the other hand, this voltage develops as a consequence of each tube's output current, and it comes from the plate supply. These are properties of cathode bias.

The result is thus a combination of the best properties of both types of bias. More important, the diodes will not permit any AC to develop across themselves, so there is no degenerative feedback in the cathode circuit. This contributes to the Mouse's excellent bass properties.

I agonized for some time over the driver circuit. I would have liked to have used an interstage transformer, but cost, size, and performance factors ruled it out. Finally, I settled on the long-tailed pair.

Figure 1 is a simplified view of this circuit. Notice that there are two identical branches, and that each branch draws an identical current. This current must be supplied through R_k from the negative supply. In a practical circuit, the internal resistance of each vacuum tube is smaller than its plate resistor, and much smaller than R_k, so the action of either tube has negligible effect on the current flowing through R_k. Yet you know that current must vary through R_{p1} in order for an output signal to develop. Where does this current variation come from?

The answer is that it comes from V_2. Every time the current through V_1 increases, the current through V_2 must decrease by an identical amount. Because this decreasing current flows through R_{p2}, an equal but opposite signal develops across R_{p2}, and you have phase inversion.

How does V_2 "know" what V_1 is doing? Easy. Suppose a signal were applied to V_1 such that its current increased. The voltage across R_k must increase, if only by a tiny amount. This voltage must also be impressed on the cathode of V_2. This represents an input signal to V_2, and because its grid is grounded, it is the only signal V_2 can see. V_2 therefore adjusts its conduction accordingly.

Notice that the signal is impressed on V_1's grid, and on V_2's cathode. V_1 is acting like a common-cathode circuit, and V_2 acts like a common-grid. The gains of these two circuits are different, so the plate resistor of V_2 is often made larger to compensate.

The whole circuit depends on the degree that the current through R_k can be made constant. Normally, this requires a very large resistor, and a high B- voltage, but you now have the advantage of many new solid-state devices.

One of my favorites is the LM334 current source. This little IC provides an extremely constant current, and it needs only a couple of diode drops to do it! The only extra component required is a resistor to program the magnitude of the current.

Figure 2 shows how this IC is employed. Potentiometer P2 sets the operating current. As long as the bias voltage of V_1 is larger than 1.4V, the IC is happy, and as long as the voltage across the IC equals the bias voltage, V_1 is happy. I did not use high-voltage protection on the LM334 because it never sees high voltage, even when the tube is cold. If this worries you, bypass the IC with a 12V zener.

CONSTRUCTION

I built the Mouse on an aluminum chassis box, 12 × 8 × 2 (Hammond 1444-22). I found that it was easier to reverse one output tube socket so that both coupling capacitors were on the outside. Other than that, the layout is quite straightforward. **Photo 2** shows the overall layout of my amp.

Grounding is relatively uncritical. In the final version, I ran a bus line to pin one of the output tube sockets, then back to the power supply, but in the prototype, I relaxed the rules a little. Both amplifiers are dead quiet, with no trace of hum.

You may find it easier to wire up the tube sockets with their associated diodes and resistors before installing them. **Photo 3** shows how they are wired. The LM334 gets its own terminal strip; this seemed to be the neatest way to mount it.

PARTS LIST

Component	Part No.	Description	Source	Comments
C1, C2	C-PD1-400	100nF, 400V	AN	
C3	C-SD0015-600	1.5nF, 600V	AN	
C4, C6	140-XAL250V47	47µF, 250V	MO	
C5	140-XAL250V100	100µF, 250V	MO	
R1, R2		47k, ¼W	RS	Part of Assortment
R3, R9, R10		1k, ¼W		271-312a
R4, R5		470k, ¼W		
R6		3.9k, ¼W		
R7	262-1000	100, 2W	MO	
R8	271-1118	1k, ½W	RS	
BR1	583-RS204L	1A bridge	MO	
D1, D2, D3, D4	625-1N4737A	7.5V 1W zener	MO	two diodes in series, 15V, 2W, total each tube
D5, D6	625-1N4758A	56V 1W zener	MO	
F1 *	270-1005	1A fuse	RS	
IC1	511-LM334Z	IC	MO	
P1 *	313-2420-100k	potentiometer, dual	MO	
P2	531-PT15B-100	100Ω trimpot	MO	
PL *		pilot light	RS	
S1 *	275-634	power switch	RS	
T1	553-VPS230190	transformer, plate	MO	TRIAD
T2	273-1512	transformer, heater	RS	
T3	Y236102	transformer, output	AL	2 × 120V pri, 2 × 12V sec, 30V-A
V1	6DJ8	vacuum tube	AN	
V2, V3	25L6GT	vacuum tube	AN	

• Misc. hardware, chassis, RCA jacks, output terminals, terminal strips, tube sockets, fuseholder, and so on.
• Note: Duplicate parts for stereo amplifier, except * do not duplicate. You will need eight 7.5V, 1W zener diodes for a stereo amp.
• AL–Avel-Lindberg, AN–Antique, MO–Mouser, RS–Radio Shack

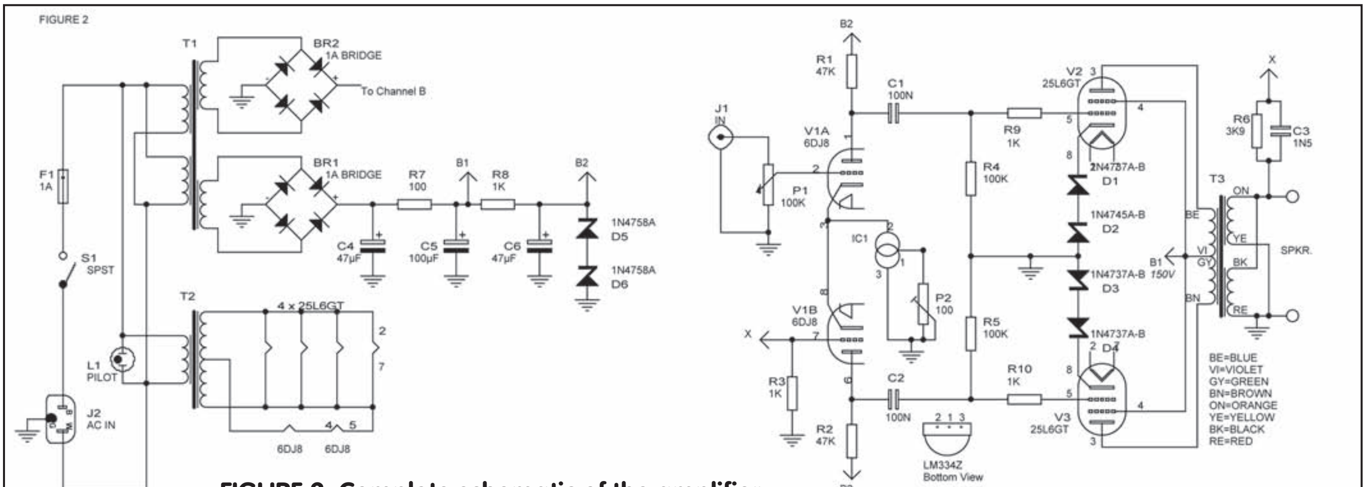


FIGURE 2: Complete schematic of the amplifier.

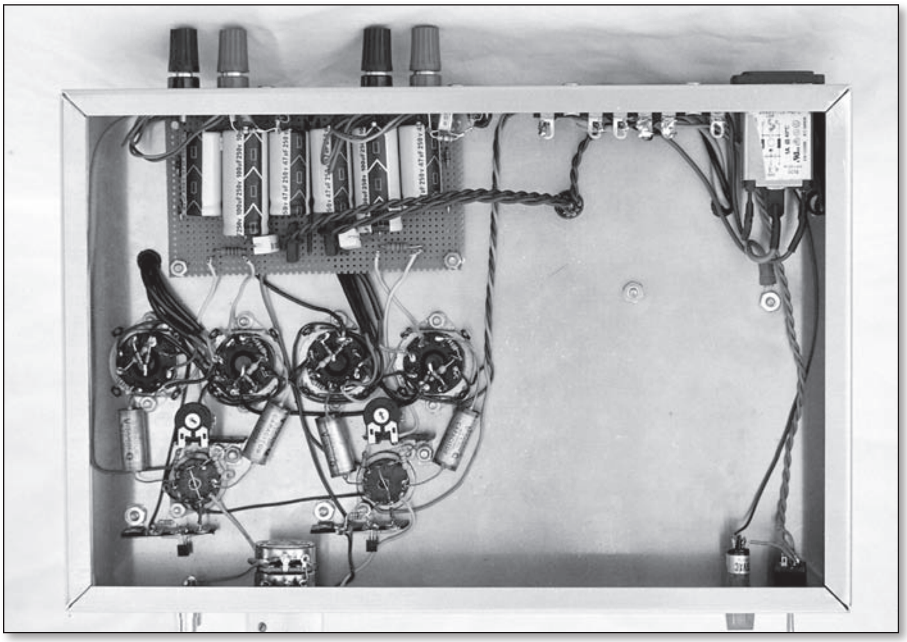


PHOTO 2: Parts layout. The white squares on the power supply board are cement resistors.

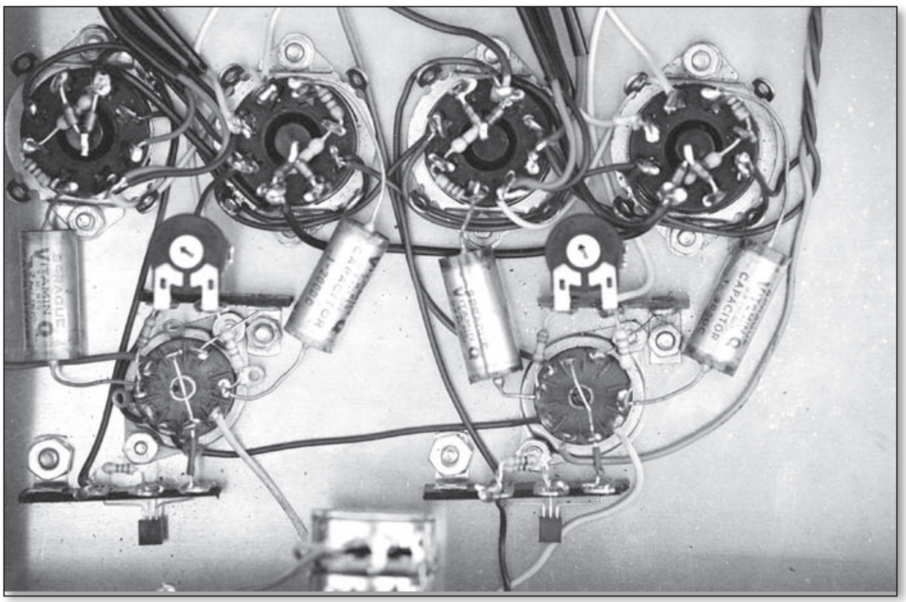


PHOTO 3: Close-up of tube socket wiring. The potentiometers were deleted from the final amplifier.

You need to allow a couple of extra terminals to mount P2. I did not use a potentiometer in my amp; I tacked a pot in temporarily, then replaced it with the proper size resistor. It takes more steps, but it seemed easier than trying to permanently mount a potentiometer.

Start the chassis wiring with the heater circuit, then the ground wires, signal wires, and finally, the resistors and capacitors. Do not mount the transformers until most of the wiring is done. Once the transformers are mounted, the chassis will not lie flat, and will be much harder to work with. After mounting the transformers, you can do the power wiring.

The power supply is assembled on a piece of perfboard, approximately 2½ × 4¾. If I had it to do over again, I would allow more room for connections to the bridge rectifier. I had originally intended to use wire-wrap pins, which my local electronic store no longer carries. As a result, I had to wire directly to the board. You should be using stranded wire for the input, so you will need to pre-tin the wires before trying to connect the bridge up to them. I mounted the board with 6-32 × ¾ screws and ¼″ spacers and nuts.

The plate transformer presents something of a problem, because it does not have wire leads. In the interest of safety, you *must* use *fully* insulated quick-connect terminals (Mouser 517-3250)! You will need to place two wires into each of the primary terminals in order to wire the primary in parallel, so be sure to allow for this. You may need to order some of the terminals large enough to accept two wires. Mouser 517-7250 terminals will accept two 18-gauge wires.

When installing the primary terminals, start with one side of the power line, then install the terminals for the other side. Be very careful not to wire the primary out of phase. It takes only a few seconds for the transformer to burn up. (Don't ask me how I know!)

Remember, it's terminals 1 and 4, then terminals 2 and 5. There are no such problems on the secondary side. I have added a fuse to the schematic; it's very cheap insurance, and may well save you the expense and bother of replacing a burned-out plate transformer.

When you've finished the assembly, install the 6DJ8 tubes and check that they light normally. Next, measure the 6DJ8 plate voltages and adjust P2 until the voltage is around 75V. This adjustment is not very critical. Finally, install the output tubes one by one, checking that the voltage on pin 8 of each tube is around 15V.

Connect a set of speakers to the unit, and turn it on. If it makes a howling sound, turn it off and reverse either the primary or the secondary leads of the output transformer. Now you are ready to hook up an input and listen to music! aX

Bob McIntyre has retired after spending 40 years as an audio technician. He was reintroduced to tubes in the late 70s while looking for an inexpensive way to build a 50W amplifier; he liked the sound, and has stayed with tubes ever since. Bob's other hobbies are music, photography, and building computers.

Sources

Antique Electronic Supply
6221 S. Maple Ave.
Tempe, AZ 85283
www.tubesandmore.com

Avel Lindberg Inc.
47 South End Plaza
New Milford, CT 06776
860 355-4711
www.avellindberg.com

Mouser Electronics
958 N. Main
Mansfield, TX 76063
800 346-6873
www.mouser.com

Back

Stands for Tiny Speakers

Follow the plans in this simple DIY project and get your speakers into the action.

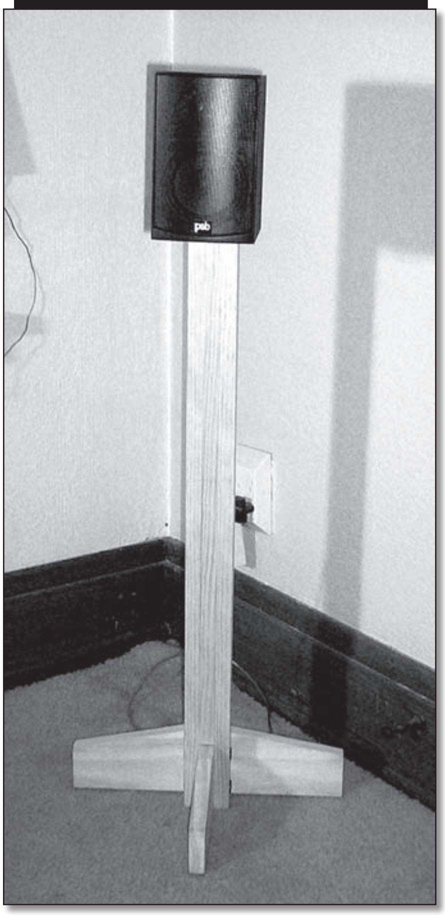
By Jerry Stump

I needed some stands for a pair of PSB Alpha Intro LR speakers to flank the entertainment center of our house we moved into recently. Some projects don't work or look funky and end up in what an early mentor referred to as "the graveyard of broken dreams." This project turned out differently.

It went together quickly, works, and looks pretty good. To top it off, it would make a good beginner's project because construction and assembly require just a few simple tools that any beginning speaker builder living in an apartment has.

The PSB Alpha Intros measure a diminutive 4.25″ wide, 6.75″ tall, and 5.75″ deep, and the stands place the center of the enclosure about 33.5″ above the floor. You can adapt the basic design to a fair range of speaker sizes by simply changing the dimensions of some of the components. Wider speakers may require a wider base; deeper speakers may need a longer foreleg on the base to keep them from taking a nosedive. And, some of you may find a more elegant approach to cable management.

PHOTO 1: Speaker stand complete in corner.

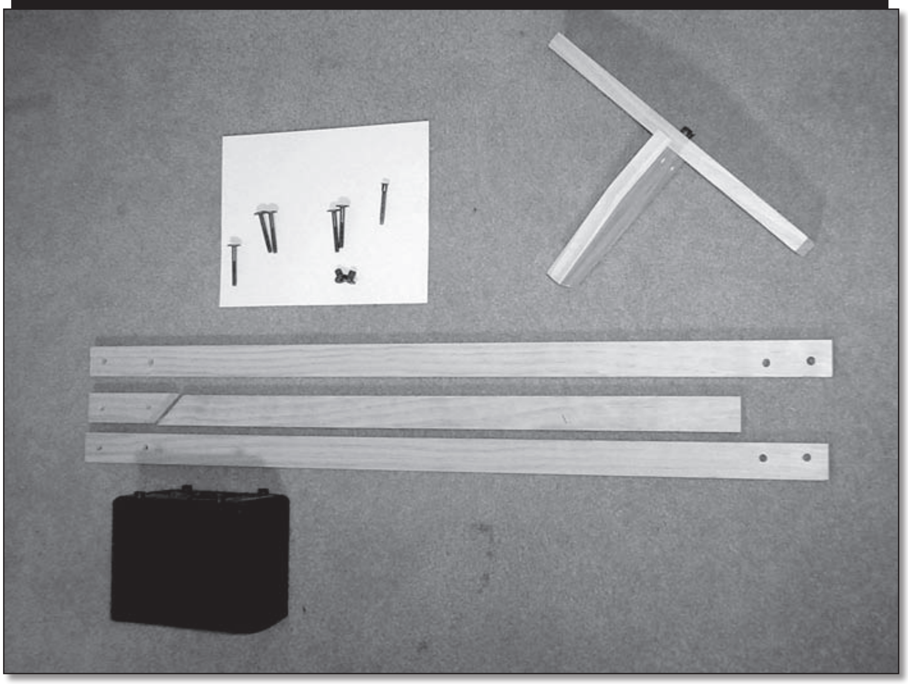


PARTS

I purchased my materials at Home Depot, but you could also try any home center/lumberyard with a good selection of hardware. Some folks may be able to pull most of the materials out of the scrap bin. Here is a list of materials to build one unit:

- 1 – 10′ 1 × 2. I used select pine, but you could substitute any species.
- 1 – 4′ 1 × 4. Match species you used for the 1 × 2 if matching matters.
- 2 – 1.5 to 2″ confirmat screws (Rockler's, Home Depot).
- 4 – 50mm cabinet connector bolts with nuts—5⁄32″ hex socket by ¼ × 20 thread (Rockler's, Home Depot).
- 1 – 2″ cabinet connector bolt. A ¼ × 20 thread pitch will match many sockets on speaker backs, but you can change this to fit the application. McFeely's threaded inserts, which you can install on home-built units, will take machine-threaded attachments like the commer-

PHOTO 2: Millwork laid out with tools.



cial units.

• 3 screw-on feet of your choice, depending on whether the units will stand on carpet or bare hardwood, as mine do.

Here are the tools you'll need:

• Saw (handsaw will do)
• Electric drill/driver
• 5/16″ drill bit and pilot drill to match confirmat screws and feet screws
• 5/32″ hex key
• Sandpaper

ASSEMBLY

1. Millwork.

You can get a full service lumber house to cut these for you.

• 2 – 36″ 1 × 2s
• 1 – 32.5″ 1 × 2
• 1 – 14″ 1 × 4 and 1 – 7.5″ 1 × 4
• Cut taper in 1 × 4s starting at width of 2 to 2¼″, leaving 2¼″ of the length of each the original width (**Fig. 1**). That portion will be in the center of the long 1 × 4 and at one end of the short 1 × 4.

2. Drilling

Drill pilot holes for the two confirmat* screws attaching the front leg to the back leg. Screw the front leg to the rear legs. Then attach three feet to the bottom of the legs near their ends. I used screw-on rubber feet for placing the units on my hardwood floors. You could use t-nuts and screw in spikes for placement on carpet.

PHOTO 3: Stand with speaker on back.

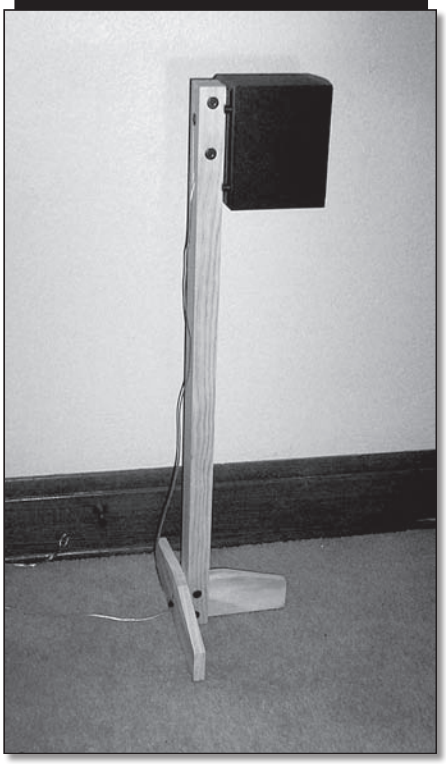
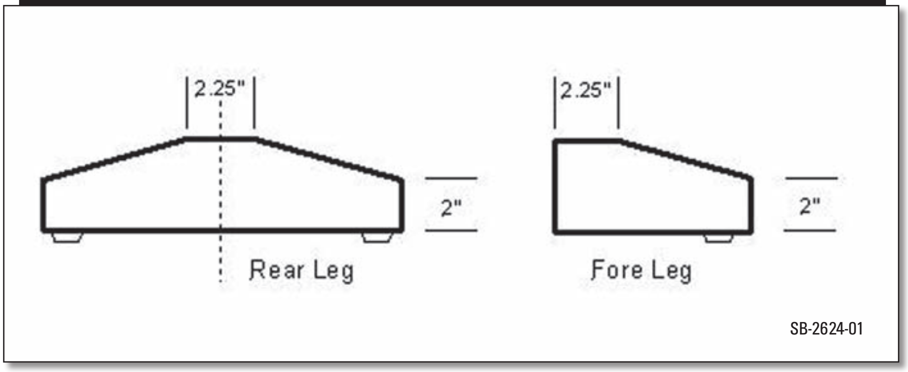


FIGURE 1: Legs and feet for stands.

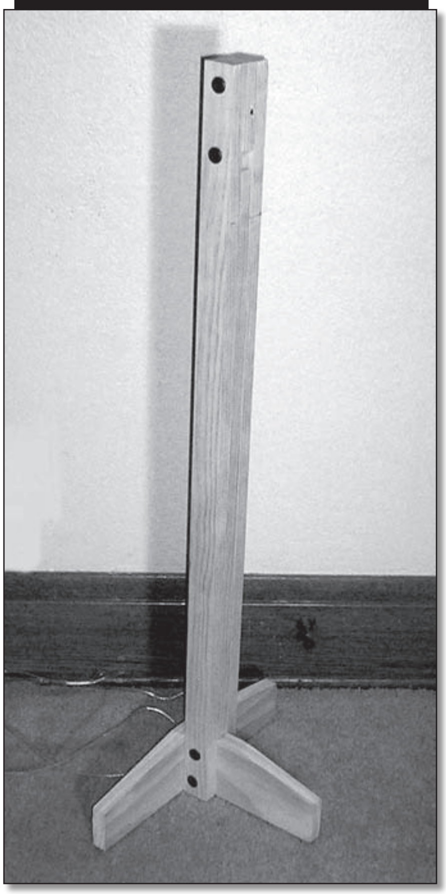


Sandwich three 1 × 2s together straddling the front leg and drill four 5/16″ pilot holes for the cabinet coupling bolts and nuts.

3. Assembly

Insert the cabinet coupling bolts and tighten with a 5/32″ hex key (Allen wrench). Stand the unit up and drill two 5/16″ holes. One should be lined up with the socket in your speaker to attach it to the stand. The second should be lined up for the passage of the speaker cable to the terminals.

PHOTO 4: Stand with speaker removed.



You will notice in the pictures that I actually used two pieces for the center part of the upright, leaving a gap for the speaker cable. If you use a hole, this second hole should be large enough for the size of speaker cable used.

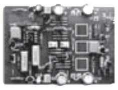
Thread the speaker cable through the stand, attach cable ends to the speaker terminals, and insert the 50mm cabinet coupling bolt to attach the speaker to the stand. I used a cable staple to secure the cable to the backside of the upright.

4. Boogie on

Attach the other end of the speaker cable to the amp and fire up the system. You can lightly sand the stand, leave it natural, paint it, or stain and clear-coat it to match your preference. *aX*

*You can substitute regular wood screws. Confirmat screws work especially well in MDF, and are available from McFeely's and Rockler. (*www.mcfeelys.com/subcat.asp?subcat=20.1.1.1; www.rockler.com*)

Jerry Stump *is a custom training instructor for the Dunwoody College of Technology in Minneapolis, Minn. He started building Heathkits, Eico Kits, and Dynakits in high school and has built dozens of speakers and a few church sound system installations over the years. He holds an MA in Speech Communication and has several years' experience in plant engineering in the fuel ethanol industry.*



Switchable Gain Booster Amplifier

This simple audio booster amplifier raises the amplitude of a signal up to 15×, and, in a stereo configuration, allows individual adjustment when R and L channels are not equal in level.

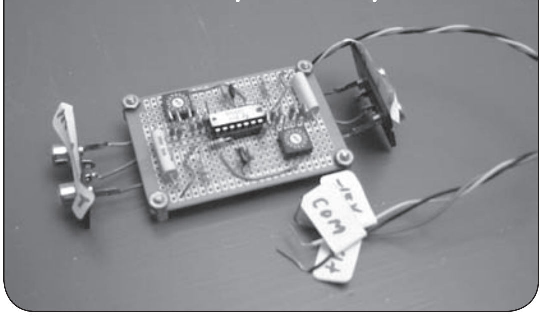
By Carl R. Casper

Consider this scenario: A user had a signal source connected to the AUX input on an integrated amplifier that had a consistently lower level than the other sources. This made it necessary for the user to advance the loudness control higher to compensate and then experienced "blasting" when setting the input selector to another source. It wasn't feasible to attenuate the louder sources, so a simple preamplifier was deemed necessary.

A physically small preamplifier that would fit the back of the equipment rack was needed. It required that the gain be adjustable from 1 to 15 (22dB). You may wish to plug a different signal source in and adjust the gain accordingly. When the user reconnected the original source, he wanted a way to repeatedly restore the original gain setting.

It is easy enough to design a simple amplifier given the present state of technology. But what is the best way to adjust the gain? Memories of pots with pointer knobs and scales and impreciseness come to mind. A 10- or 15-turn trimmer pot would give resolution but no repeatability.

PHOTO 1: The amp with rotary switches.



SOLUTION

In dealing with op-amp theory, the inverting configuration in **Fig. 1**, the expression for the stage gain, directly relates to Rf/Ri, which is the value of the feedback resistor divided by the value of the input resistor. This is true because the inverting input is held at 0V by the action of the op amp and feedback resistor.

FIGURE 1: Inverting op amp amplifier. Gain factor = $\frac{R_f}{R_i}$.

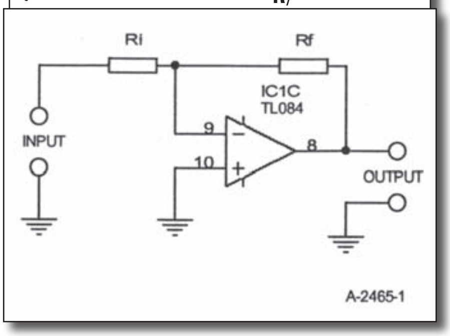


FIGURE 2: Four different switched gain resistors weighted 8-4-2-1 .

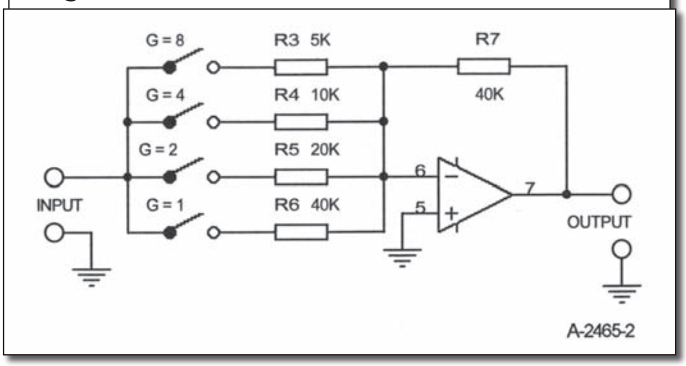
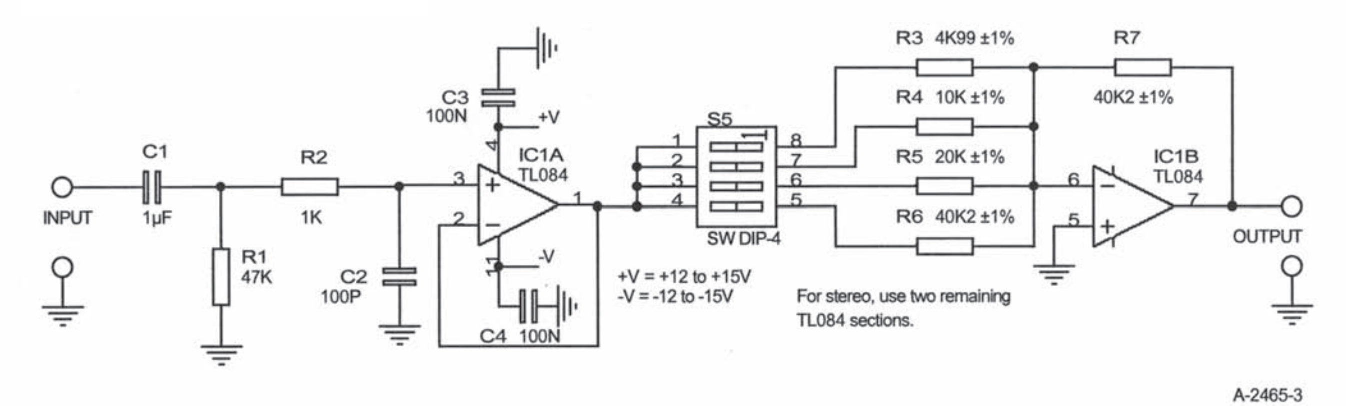


FIGURE 3: Final amp schematic.



In other words, you can change the gain either by holding the input resistor constant and changing the value of the feedback resistor or holding the feedback resistor constant and changing the value of the input resistor.

In **Fig. 2**, it follows that four different gain selections are possible using four switches and four different input resistors with a single feedback resistor. With the inverting input at 0V, the gain setting resistors R3 to R6 can work without interaction.

If you select more than one feedback resistor, the overall gain is equal to the contribution of each resistor; i.e., for all switches closed:

$$\text{Gain} = R7/R3 + R7/R4 + R7/R5 + R7/R6$$

If R6 is 40k (the same value as the feedback resistor), gain = 1, R5 = 20k, gain = 2; R4 = 10k, gain = 4; R3 = 5k, gain = 8.

With this scaling, it is possible to select gain factors from 1 to 15 by setting various switches to the On position (**Table 1**). Gain selection can be made a step easier by substituting a hexadecimal-coded rotary DIP switch (Omron A6R-162RF). Codes 10 to 15 are labeled abcdef. This switch will select the proper input resistors for each gain setting.

The final schematic is presented in **Fig. 3**. The first stage IC-1a is a buffer used to isolate the loads presented by R3 through R6 from the input. R2 and

TABLE 1: GAIN SETUP.

Gain Factor X	Switch S1 D C B A	Hex #
Off	0 0 0 0	0
1	0 0 0 1	1
2	0 0 1 0	2
3	0 0 1 1	3
4	0 1 0 0	4
5	0 1 0 1	5
6	0 1 1 0	6
7	0 1 1 1	7
8	1 0 0 0	8
9	1 0 0 1	9
10	1 0 1 0	a
11	1 0 1 1	b
12	1 1 0 0	c
13	1 1 0 1	d
14	1 1 1 0	e
15	1 1 1 1	f

C2 form a simple RFI filter to prevent a local AM radio station from getting through. C1 removes any DC bias on the input signal. You may add IC-1C after IC-1b for a non-inverting signal output.

Regulated ±12 to ±15V power supplies are required, 10mA maximum; in this situation, the power was tapped off another piece of equipment.

COMPONENT SELECTION

I chose a TL084 quad op amp based on the need for a medium quality amplifier for use with a mid-level signal (0.1 to 1V RMS). You can use low noise or low offset op amps, depending upon your requirements.

The gain setting resistors are 1%, although you can use 0.5% or 0.1% units to give more precise gain steps. The amplifier is inverting; you may prefer to add A1-C (**Fig. 4**) to re-invert the signal.

CONSTRUCTION

I used standard construction techniques, building a two-channel inverting output unit using two sections each of a TL084 DIP. You may construct version 1 using DIP switches or version 2 using rotary switches (**Photo 1**). Version 2 requires a different layout of the gain select resistors (**Fig. 5**) to match the pin layout of the rotary switch.

I built my prototype using a Radio Shack 270–150 multipurpose board, mounting resistors R3-R7 vertically to save space and to minimize the area that IC–1B input includes. You can install the completed circuit board into a small metal enclosure to minimize interference pickup. You can make 1/8″ holes to facilitate adjustment of the switches.

USAGE

With the amplifier installed and power applied, start with gain of one and increase to the desired level on each channel. The frequency response is 10Hz to greater than 20kHz. Distortion measurements were not available. *aX*

FIGURE 4: Adding former non-inverting output.

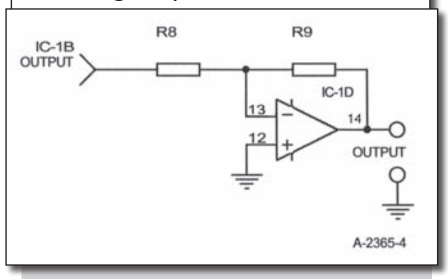
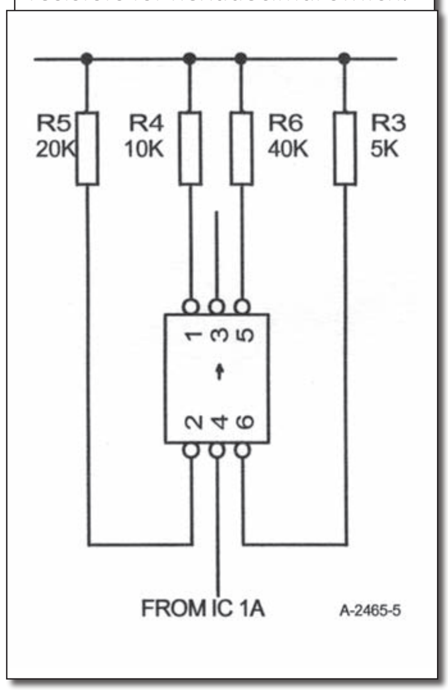


FIGURE 5: Layout of gain select resistors for hexadecimal switch.



__Carl Casper__ is an engineer with experience in electronic circuits and instrument calibration. In his free time, he designs and builds audio and test equipment.

PARTS LIST (DUPLICATE FOR STEREO).

R1	47kΩ 5%
R2	1kΩ 5%
R3	4.99kΩ 1%
R4, R8, R9	10.0kΩ 1%
R5	20.0kΩ 1%
R6, R7	40.2kΩ 1%
C1	1.0μF 50V polystyrene
C2	100pF 50V ceramic
C3, C4	100nF 50V ceramic
IC-1	TL084 T1 or equivalent
S-1	Quad DIP switch-Grayhill 76B504 or equivalent OR hexadecimal rotary switch. Omron A6R-16ZRF

Build on Radio Shack 276-150 multipurpose PC board or perforated board.

Sources for Components

Local electronics store.
Newark Electronics, 800-263-9275, Fax: 800-718-1998.
Mouser Electronics, 800-346-6873, www.mouser.com.

Back

Testing the Ed Frias EFE Loudspeaker Kit

By Joe D'Appolito

I ran a series of impedance, frequency response, polar response, and distortion tests on the Ed Frias EFE loudspeaker kit supplied by Speaker City and built by the MacArthurs (March '06 *aX*, p. 27).

IMPEDANCE

It is my habit in testing these kits to arbitrarily pick one unit of a pair for all tests and then check the second unit simply to see how it matches against the first. **Figure 1** is a plot of system impedance magnitude (A) and phase (B). At low frequencies the plot displays the double peaked curve typical of vented systems. The impedance minimum between the peaks at 50Hz indicates the vented-box tuning frequency (F_b).

The impedance at this point is 8.4Ω. There is an absolute impedance minimum of 7.4Ω at 200Hz. A maximum phase angle of –40° occurs at 108Hz. The impedance magnitude at this frequency is 14.5Ω. I would rate this an 8Ω speaker.

The impedance minimum 8.4Ω at F_b is rather high compared to the driver voice-coil resistance of 6.1Ω. This is characteristic of a low box Q and an overdamped vented box response. I'll say more on this later.

FREQUENCY RESPONSE

Figure 2 shows the full-range frequency response of one of the pair of speakers.

FIGURE 1: EFE kit impedance magnitude (A) and phase (B).

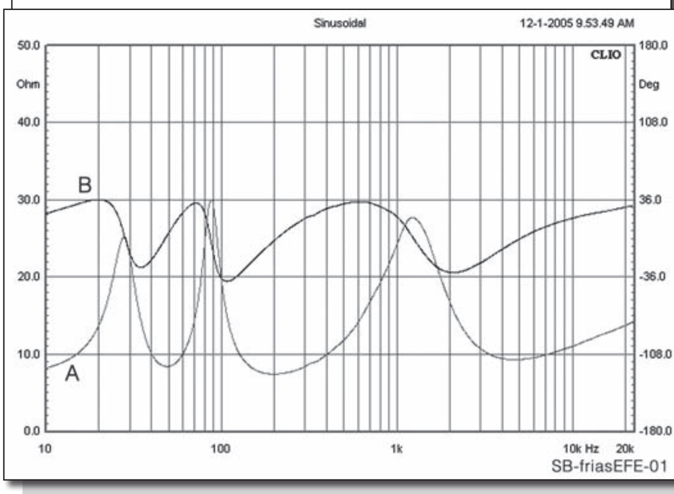


FIGURE 3: EFE kit step response.

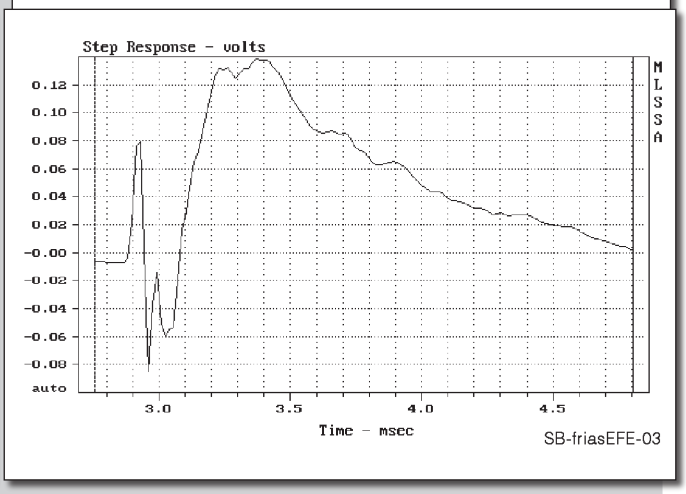


FIGURE 2: EFE kit full-range response.

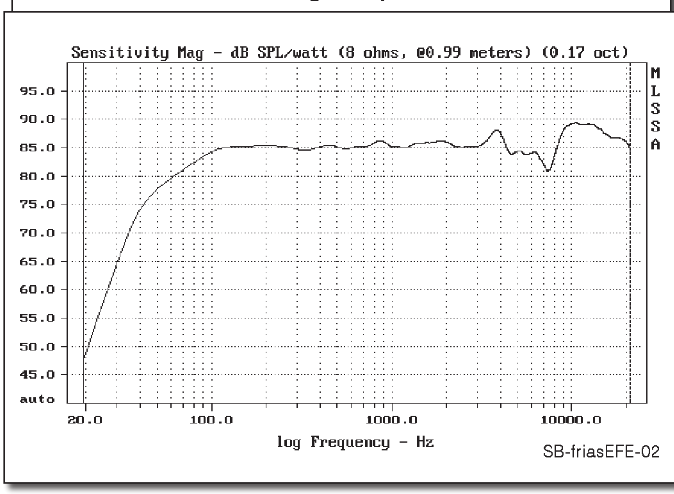
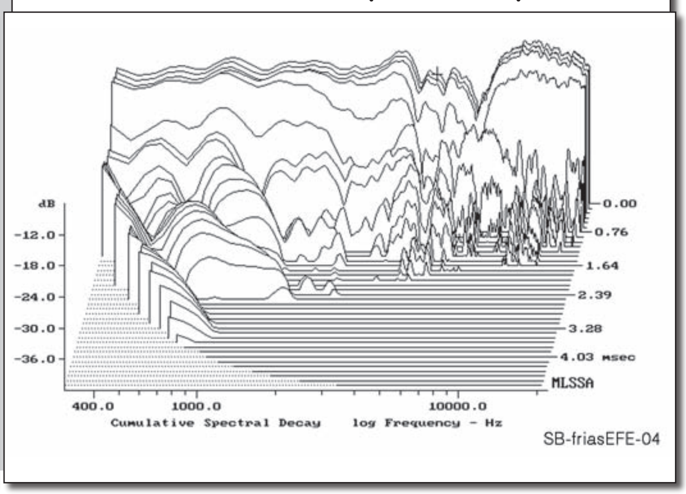


FIGURE 4: EFE kit cumulative spectral decay.



This response is obtained as a combination of the far-field quasi-anechoic response in the freestanding position and a set of appropriately combined near-field woofer and port measurements[1]. I placed the microphone along the tweeter axis at a distance of 0.99m to produce the far-field freestanding response, and spliced near-field and far-field responses together at 200Hz to produce the full-range response[1]. The response data is ⅙ octave smoothed. Sensitivity averages 86dB SPL/2.83V/1m in the two octaves between 500Hz and 2kHz.

Relative to this level, response is down 3dB at 80Hz. This is a rather high F3 point for this driver in a vented box. I'll comment on this in more detail at the end of my report.

Between 120Hz and 3kHz response is extremely flat, varying by only ±0.8dB. This is the active frequency range of the woofer. From 3kHz to 20kHz, however, tweeter response shows a peak-to-peak variation of 8.4dB! This is surprisingly poor. The Frias EFE is not configured for bi-wiring, so I could not determine responses for the individual drivers, but it is clear that the tweeter is operating above 3kHz.

WOOFER/TWEETER TIMING

The step response is plotted in **Fig. 3**, which shows two separate arrivals of acoustic energy. The initial sharper positive spike is the tweeter arrival, followed by the woofer arrival, also positive. The woofer peak is delayed relative to the tweeter peak by approximately 300µs. This is a rough estimate of the tweeter/woofer delay.

Although both drivers are connected with positive polarity, the system is not time-coherent. The double negative peaks approximately 60µs apart are characteristic of a reflection in the tweeter back volume, which may explain the poor tweeter response.

CUMULATIVE SPECTRAL DECAY

The cumulative spectral decay (CSD) response is presented in **Fig. 4**, a waterfall plot that shows the frequency content of the system response following a sharp impulsive input at time zero. On the CSD plot, frequency increases from left to right and time moves forward from the rear. Each slice represents a 0.12ms increment of time. The total vertical scale covers a 30dB dynamic range.

Ideally, the response should decay to zero instantaneously. Real speakers, however, involve inertia and stored energy that take a finite amount of time to die away. A prominent ridge parallel to the time axis indicates the presence of a strong system resonance.

The first time slice in **Fig. 4** (0.00ms) represents the system frequency response. There are no distinct ridges in the CSD over its valid time interval, although there is a lot of "hash" in the tweeter's decay response.

HORIZONTAL POLAR RESPONSE

Horizontal polar response is examined in **Figs. 5**, **6**, and **7**. **Figure 5** is a waterfall plot of horizontal polar response in 10° increments from 60° right (+60°) to 60° left (–60°) when facing the speaker. All off-axis plots are referenced to the on-axis response, which appears as a straight line at 0.00°. For this reason, the plotted curves show the *change* in response as you move off-axis. For good stereo imaging the off-axis curves should be smooth, evenly spaced replicas of the on-axis response with the possible exception of some tweeter roll-off at higher frequencies and larger off-axis angles.

Below 3kHz and within ±30° off-axis, the curves are indeed fairly smooth replicas of the on-axis response. Above 3kHz, however, there appears to be substantial off-axis flare in the tweeter response. This

FIGURE 5: EFE kit horizontal polar response.

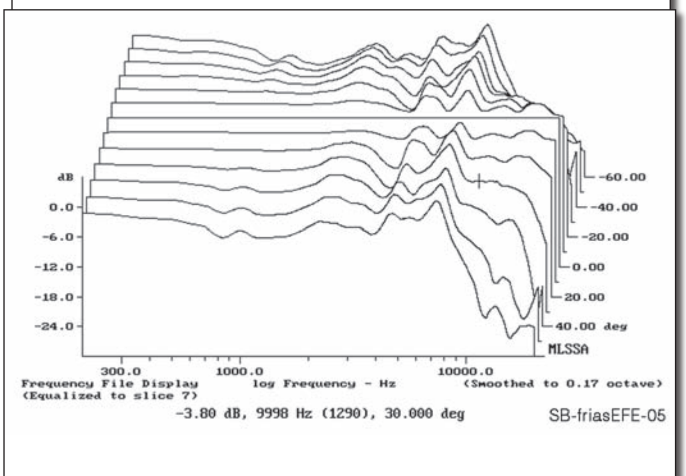


FIGURE 6: Comparing on-axis (A) and 30° off-axis (B) responses.

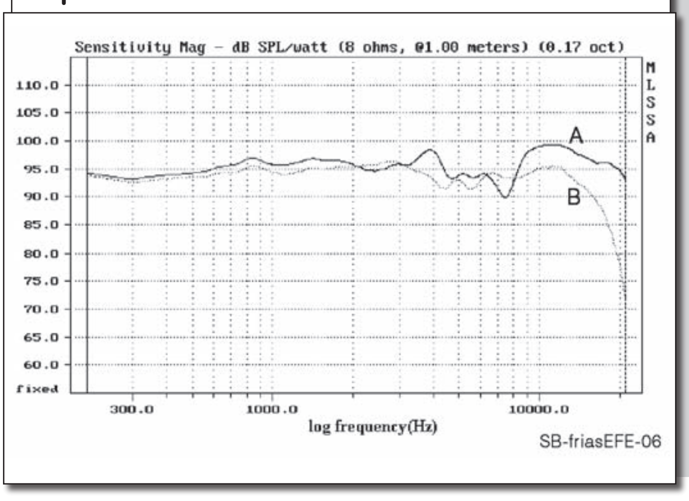
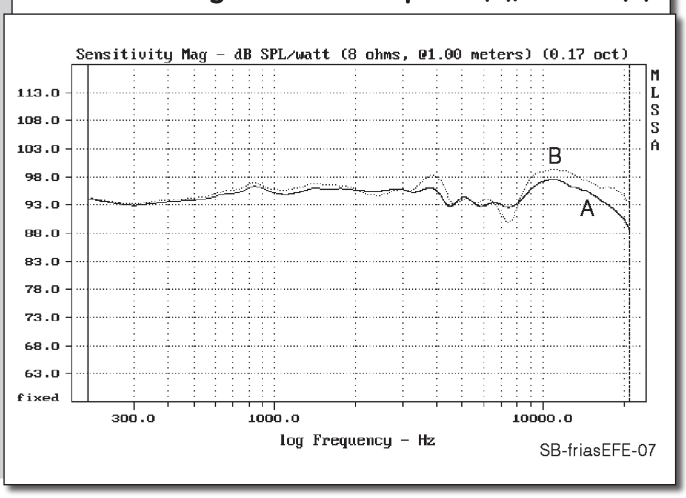


FIGURE 7: Average horizontal response (A), on-axis (B).



is an artifact of the waterfall plot. When subtracted from the on-axis response, the flaring tweeter response appears.

As **Fig. 6** shows, the tweeter is actually smoother off-axis. In this plot the on-axis response (A) is compared with the response 30° off-axis (B). Even though response falls rapidly beyond 12kHz, this suggests that you may prefer to listen to this speaker somewhat off-axis. The –3dB coverage at 15kHz is ±20°. At 10kHz the -3dB coverage extends to ±28°. This is somewhat poorer than typical 25mm dome tweeters.

The average response over a 60° horizontal window (±30°) in the forward direction is a good approximation of the way a speaker will sound in a typical listening environment. This response is plotted in **Fig. 7** (A) and compared with the on-axis response (B). Here you see that much of the tweeter response variation has smoothed out. This further suggests that off-axis listening may be preferred.

VERTICAL POLAR RESPONSE

Figure 8 is the waterfall plot of vertical polar response. Responses are shown in 5° increments from 25° below (–25°) the tweeter axis to 25° above it. Moving below the tweeter (negative angles), the response changes little from the on-axis curve out to –15° for frequencies below 3kHz.

As you move above the tweeter, a deep response notch develops rapidly centered on 3kHz. Moving off-axis vertically, you see the same tweeter response flare that you saw on the horizontal waterfall plot. This further suggests that you may prefer to listen to this speaker off-axis and below the tweeter axis.

INTERMODULATION DISTORTION

I believe that intermodulation distortion (IMD) tests are more revealing of loudspeaker performance than harmonic distortion tests. In IMD tests two frequencies are input to the speaker. Intermodulation distortion produces output frequencies that are not harmonically related to the input. These frequencies are much more audible and annoying than harmonic distortion and a better measure of driver nonlinearities.

Let the symbols f_1 and f_2 represent the two frequencies used in the test. Then a 2nd-order nonlinearity will produce intermods at frequencies of $f_1 \pm f_2$. A 3rd-order nonlinearity generates intermods at $2f_1 \pm f_2$ and $f_1 \pm 2f_2$.

First, I examined woofer intermods by inputting 400Hz and 550Hz signals at equal levels. These frequencies should

FIGURE 8: EFE kit vertical polar response.

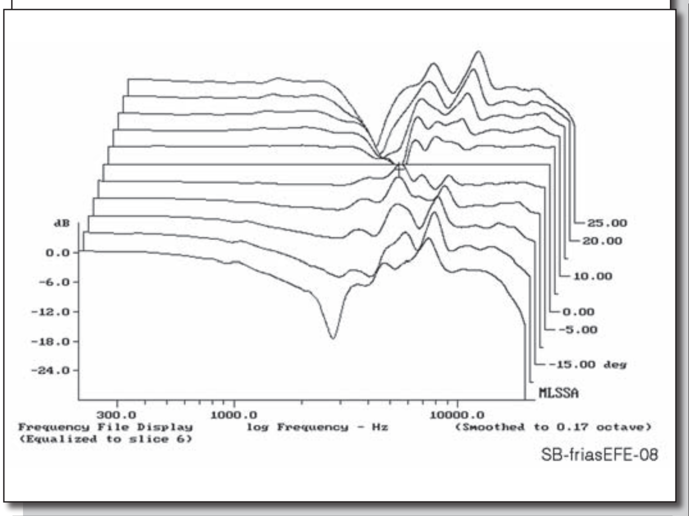


FIGURE 9: EFE woofer IMD.

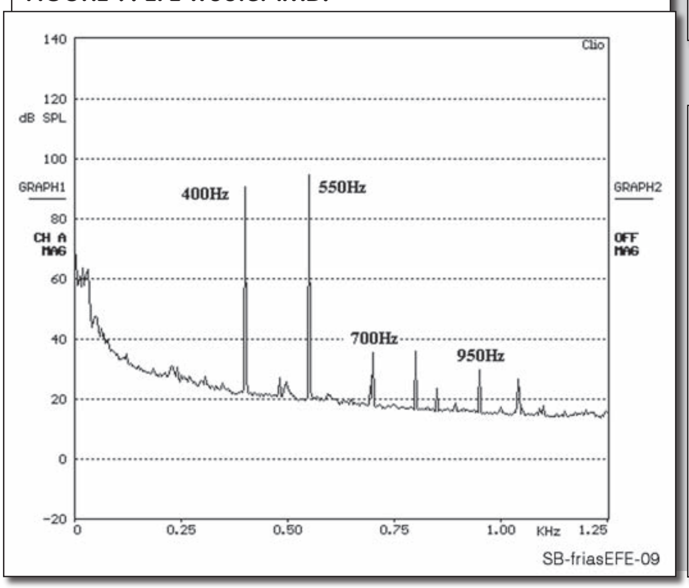


FIGURE 10: EFE tweeter IMD.

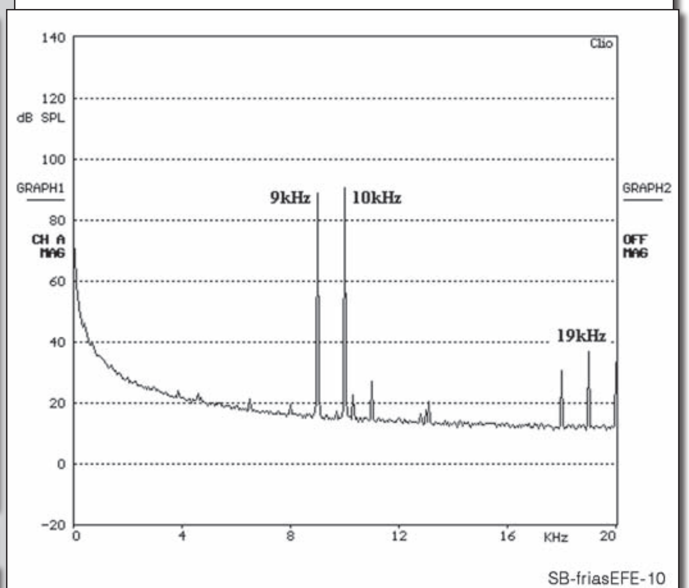
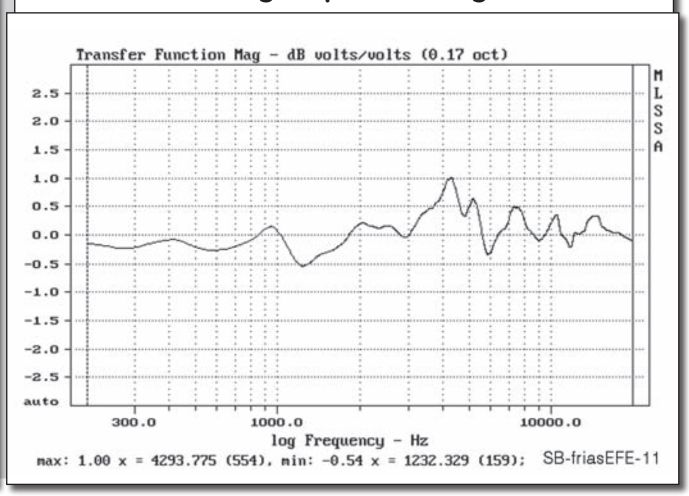


FIGURE 11: Illustrating EFE pair matching.



appear predominantly in the woofer output. Total SPL with the two signals was adjusted to 90dB at 1m. **Figure 9** is a plot of the resulting output spectrum. The principal output frequencies of 400 and 550Hz are highlighted on the graph.

In addition to these frequencies, you will see many other vertical lines representing both IMD products and harmonics of the 400 and 550Hz signals. Counting only the IMD products, there are significant lines at 700 and 950Hz. The overall level of 0.12% is typical of many good 6.5″ woofers.

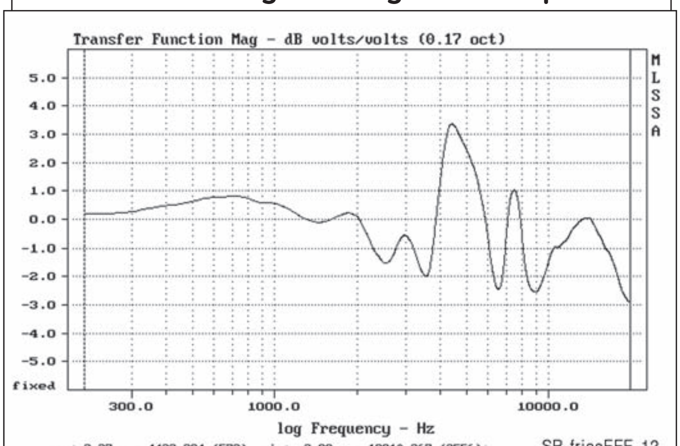
Next, I measured tweeter intermods with a 9kHz and 10kHz input pair adjusted to produce an 87dB SPL at 1m. Results are plotted in **Fig. 10**. The major IM products occurred at 11 and 19kHz. Total IMD was 0.16%, a result somewhat on the high end of the many tweeters I have tested for *audioXpress*, but still acceptable.

The last IMD test examines cross-intermodulation distortion between the woofer and tweeter using frequencies of 550Hz and 10kHz, again at a 90dB level. Ideally, the crossover should prevent high-frequency energy from entering the woofer and low-frequency energy from entering the tweeter. A single tweeter sideband appeared at 9450Hz (not shown) at an overall level of 0.05%. This is a very good result and indicates good inter-driver isolation by the crossover.

SPEAKER MATCHING

All of the tests results reported so far were obtained from a single sample.

FIGURE 12: Illustrating effect of grille on EFE speaker.



Now look at how well the two speakers match in frequency response. A frequency response plot of the *difference* between the two samples is shown in **Fig. 11**. The two samples match within +0.1 and –0.5dB below 3kHz. Thus woofer samples match well. The tweeters match less well, but at +1dB and –0.4dB the matching is still quite good. Sonically, the pair will sound little different.

GRILLE EFFECTS

I conducted all of the above tests with the grille off. **Figure 12** shows the EFE Frias response with the grille on, but referenced to the response with the grille off; that is, it plots the *change* in response under the two conditions. The grille has a fairly substantial effect on response at all frequencies above 2kHz. Overall response variations of +3.4dB to –2.9dB are seen. As usual, the grille is there only for cosmetic effect and to protect the drivers from prying fingers.

FINAL COMMENT

The low-frequency –3dB point of 80Hz seemed very high for this Peerless driver. Just to check the near-field approach I used to get the low-frequency response I tested the speaker again in a much larger test facility at Boston Acoustics. Here a full-range response measurement is possible. The result was the same.

Given the rather high Q_{TS} of the Peerless woofer, the SC4/C4 alignment is appropriate and should produce an F3 of 33Hz in a 34 ltr box tuned to 32Hz. The EFE Frias box of about 12 ltr is tuned to 50Hz and lightly filled with dacron polyfil. This produces a very low box Q. As mentioned earlier, the first indication of this was the rather high impedance of 8.4Ω at F_b relative to the voice-coil resistance of 6.1Ω. Typical box vented box Qs are in the range of 7 to 10. Using Eq. 3.30 of reference 1, the calculated box Q is 2.

The net effect of the very low Q, the small box, and the high F_b is to cause the woofer to act as though it were in a closed box. Indeed, in the first octave below F3 the rolloff is just 12dB rather than the 24dB you would expect from a vented enclosure. It would be interesting to know why the designer chose this alignment. *aX*

REFERENCE

1. J. D'Appolito, *Testing Loudspeakers*, Audio Amateur Inc., Peterborough, NH, 1998.

A note on testing: The EFE Frias speaker was tested in the laboratories of Audio and Acoustics, Ltd. using the MLSSA and CLIO PC-based acoustic data acquisition and analysis systems. Acoustic data were measured with an ACO 7016 ¼" laboratory grade condenser microphone and a custom designed wide-band, low-noise preamp. Polar response tests were performed with a computer controlled OUTLINE turntable on loan from the Old Colony Division of Audio Amateur Inc.

WORTHWHILE PETITE PROJECT

I am a retired (on disability) music lover who doesn't have bucketloads of disposable income to spend on my passion. So after DIYing a pair of speakers with Fostex FE166E speakers, I was most intrigued with Tim Smith's PP Pentode Petite amplifier project in June 2005 *audioXpress*. It seemed that this type of amp would be a superb match for my speakers.

Almost right off the bat I realized that building an amp from scratch, even though Tim's article claimed it would be suitable for a novice, would be quite some undertaking. Nevertheless I started assembling parts and almost from the get-go e-mailing Tim with questions—some that now seem a bit absurd. Tim was very forthright in answering all my questions and offering suggestions and never once "talked down" to me. He even apologized for taking a week to answer one of my questions when he was out of the country on business!

Well, having now finished the amp, I can say that the project was worth all the effort. It mates synergistically with my full-range speakers and is more musical and satisfying than the SET "spud" amp I was using previously. It has also added some very wonderful "authority" that I never had before.

I had to learn metal work, woodwork for the base, and a whole lot of electrical knowledge. *audioXpress* was a factor in my quest for knowledge with all the many DIY articles over the last months. I have found a new passion. Building tube electronics! I look forward to more articles of interest in your magazine.

Ron Steed
Nanaimo, BC
Canada

REGULATORS REDUX

I have a question regarding the "High Performance Regulators" featured in the Jan. 1995 issue of *Audio Amateur* by Gary Galo, Walt Jung, and Jan Didden. I built a pair of the "POOGE" 5.515 regulators for my preamp and they have been rock-solid performers for years, but now I am starting a new line stage that requires 30V.

My question to Mr. Jung is, can the 5.515 regulator be easily changed for a 30V output? Thanks so much both for an answer and for those regulators.

Larry Campbell
Englewood, Fla.

Walt Jung replies:

Regarding Fig. 3 of "High Performance Regulators, Part 1," very few changes are necessary to reach a 30V output. The transistors are rated for 50V or more, save Q855. This part should be changed to a 2N5210, as per the original diagram. That should do it for the active parts.

To program the output voltage for 30V, the R861 value should increase to a value such that the ratio of R861/R863 is close to 3, inasmuch as the reference voltage seen at the base of Q855 is about 7.5V. I suggest an R861 3.01k value as a starting point. Note that the tolerance of the 6.9V LM329 could mess up this nice relationship, so a check of Q855's base voltage can verify if it deviates from the 7.5V. Again, trim the R861/R863 ratio, if necessary.

This describes changes relative to the positive regulator. To modify a negative output as per Fig. 4, substitute 2N5210s for the 2N5089s, and adjust R862 as above, to yield a -30V output.

Of course, it goes without saying that any caps on the higher voltage output line(s) should be rated for appreciably higher than 30V, for example 50V. It is good to hear that these circuits are working well for you after all these years!

ADCOM 565 MOD

I finally received the circuit boards to use as substitutes for all the perfboards Gary Galo used (Nov. '03–Feb. '04), and I am in the midst of making the Adcom 565 Mods. I note that removal of J7 and J9 disconnects D911 and D912. I went through the articles and Xpress Mail and could not find any reference to these diodes. I can see no reason for disconnecting these diodes and am wondering if it was intentional?

Chet Staley
cstaley12@comcast.net

Gary Galo responds:

D911 and D912 are discussed in Part 2, page 49 of the Dec. 2003 issue. As I noted under "Wiring the Regulator," I have never actually seen these diodes in a GFP-565 preamp, even though they are shown in the schematic and the PC layout. J7 and J9 were removed to eliminate extraneous power supply traces in the preamp. If your preamp does have these diodes, simply move them to any convenient location along the main power supply bus, as noted in the last bullet on page 49.

I would like to take this opportunity to correct an error in Part 3. On page 44 (Jan. 2004) I wrote "The AD711KN has a unique feature: the output can be taken from compensation pin 5..." AD711KN is incorrect. This should be the AD744KN noted in the previous sentence. I thank Chuck Hansen for bringing this error to my attention some time ago.

COMPUTER-AIDED DESIGN

Did Bill Fitzmaurice ("David: The Giant Killer," Feb. '06, p. 10) use computer drawing software, and if so which one? How about an article on CAD products available for the (hobby) speaker builder? It would be great to get some advice on this topic—if you have any leads and/or suggestions, please let me know.

Thanks,
Andy Janes
tajanes@comcast.net

Bill Fitzmaurice responds:

I now do my drawings with photo software (MS Picture It); you'd be amazed what you can do once you figure out how, and it's much more intuitive than any CAD programs that I've come across. The 3D views are done with SketchUp software (www.Last Software\SketchUp), but not by myself; I have a friend who does them for me. I'm comfortable with sawdust and glue, but not so much with computers. I only just put aside the pencil and paper for good a few months ago.

SPEAKER KIT?

I am a long-time subscriber and am very interested in the David horn speaker (by Bill Fitzmaurice) in the February 2006 issue of *audioXpress* (p. 10). However, I do not have the necessary equipment or woodworking expertise to build such a project. Will this speaker be built as a kit in the future?

Harold Cooper
Albuquerque, NM

Bill Fitzmaurice responds:

I don't have any plans to offer a kit myself. Your best bet is to contact someone nearby you willing to build them for you. I'd try the Madisound and Parts Express forums or my own forum at www.audioroundtable.com/BillFitzmaurice/.

CLASS D FAN

I have been a subscriber for some years and have a request for new product coverage in your magazine. The Hypex UcD Amplifier modules (i.e., UcD180 and its new brother the UcD 400) seem to be taking the DIY world by storm. I would like to see some coverage on these class D amps.

The only set I have heard were quite excellent-sounding, so much so that I was shocked because I thought class D would sound bad. Currently, I am saving to purchase the parts to build a UcD 400-based stereo amp, but it will be some time on my currently limited budget.

Class D, and from what I can tell the Hypex modules (one of a few modules available to the DIY person), is changing the way we do music and would be of interest as an article in your magazine.

Greg Starkey
New Baltimore, MI

AUDIO NEWS

The VTV Audiophile & Music Expo is being held from May 6-7, at the Embassy Suites Hotel in Piscataway, N.J. Featuring exhibitor and vendor rates, the EXPO is a sales venue for small to medium-sized manufacturers and dealers of audiophile equipment, musical instrument amps, guitars, and more, as well as those selling quality vintage and used gear. For more information on this opportunity to meet your customers in person, contact Charlie Kittleson at cathode99X@vacuumtube.com.

New from **D2Audio** is a real-time, all digital feedback amplifier that automatically corrects the audio amplifier signal, using a new breakthrough in closed-loop terminology. These second-generation DAE-2 products integrate full-bandwidth digital feedback and an advanced feedforward for digital power correction capability. The family of products, including the D2-91413-LR and D2-91433-LR, feature embedded digital signal processing, scalable power and performance solutions, adaptive sample rate conversion, and more. To learn more, visit www.d2audio.com.

The MC-884 in-wall speaker, from **RBH Sound,** features two 8″ aluminum woofers, a 2½″ aluminum dome midrange, and a 1″ aluminum dome tweeter. Handling up to 200W of power, the MC-884 measures 10¼″ W × 19″ H × 3¾″ D. The tweeter is designed to swivel and includes protection to avoid being over-driven. Also available is the A-414 in-wall center channel speaker, with an off-the-floor, center channel/LCR solution and a 1″ fabric dome tweeter and two 4″ polypropylene woofers. Using a woofer-tweeter-woofer configuration, the A-414 increases efficiency and power handling up to 130W, which produces a more controlled dispersion pattern. info@rbhsound.com.

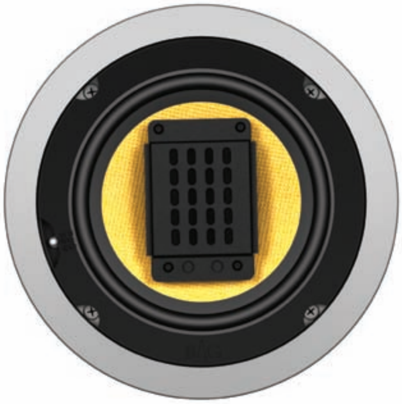
Featuring nearly zero popcorn noise, high input impedance, and monolithic dual construction, the LSK389 series, from **Linear Integrated Systems,** obtains tighter IDSS matching, and better thermal tracking, than matching individual JFETs. The LSK389 incorporates three grades of IDSS: 2.6-

6.5mA, 6-12mA, 10-20mA, with an IDSS match of 10%, a gate threshold offset of 20mV, a voltage noise of 1nV/Hz at 1kHz, 2nV/Hz at 10Hz typical, with a gain of 22ms, and 22pF of capacitance. LIS also offers custom screening for users who require different ranges of IDSS, VGS, or other DC parameters than found in the datasheet. For more information, e-mail sales@linearsystems.com, or visit www.linearsystems.com.

Morel Acoustics USA announced a mutual agreement with Israeli-based **Morel**. Morel Acoustics USA decided to discontinue using the Morel name and will in the future operate under the **Renaissance Audio Group** name only. Renaissance Audio Group, a division of Morel Acoustics USA, has been in existence for over 16 years and will now become the new business name for the company. All previous Morel Acoustics USA products will now be labeled Renaissance Audio Group. For more, visit the new Renaissance Audio Group website at www.renaudio.com, or address correspondence directly to info@renaudio.com.

Mouser Electronics entered into a global distribution deal with interconnect systems manufacturer **Harwin, Inc.** Mouser will feature Harwin's professional range Datamate connectors, which carry 3.0A per contact, with both latching and jackscrew options for high-vibration environments. Designed for wire-to-wire, wire-to-board, and board-to-board applications, the Datamate series have applications for both PC tail and SMT options. Also signing a deal with Mouser was **Digi International**, a manufacturer of ARM-based embedded modules. Digi products include device, terminal and secure console management servers; multiport serial cards; USB devices, and more. To learn more, visit www.mouser.com.

Scantek's ROGA MI-17 and ROGA RG-50 ¼″ measuring microphones are easily attached to existing data acquisition systems, sound cards, and so on. Powered by constant-current sources, the MI-17 is a type 1 compatible mike, while the RG-50 is Type 2. Call (800) 224-3813 for more information, or visit www.scantekinc.com.



The first dual-channel custom loudspeaker that correctly reproduces information from two channel stereo sources, the PD-6CS, is now available from **Bohlender Graebener**. Accurately integrating signals from both stereo channels into one, the PD-6CS can deliver true high fidelity from a single loudspeaker, and features an exclusive patented planar ribbon tweeter as well as a dual voice coil Kevlar cone woofer with rubber surround. BG also unveiled five new models in its Radia Series of in-wall loudspeakers, which feature a variety of new driver technologies and design configurations. For more, visit www.bgcorp.com. *aX*

Back

VENDORS

Pro audio kits: mic pre, mixers, buffers, crossovers, power amps.
Also custom designs.
www.borbelyaudio.com
Selected Borbely Audio kits in Japan:
http://homepage3.nifty.com/sk-audio

Amplifier building is easy with ezChassis®. Pre-punched holes and slots.
www.designbuildlisten.com

Revised Audio Restoration, new Audio Help, new products models 465 and 912.
www.tdl-tech.com (505-382-3173)

Tube amplifier kits starting at $109.
Low cost custom transformers.
www.engineeringvista.com

WANTED

stereo gear old/new/quality, speakers, amps, preamps, turntables, reel to reels, vacuum tubes, tube testers, guitars, parts, etc. **850-314-0321**
Email: **sonnysound@aol.com**

ALL ELECTRONICS CORPORATION

Electronic and Electro-mechanical Devices, Parts and Supplies. Many unique items.

www.allelectronics.com

Free 96 page catalog
1-800-826-5432

meniscus®

Real wood cabinets

- 5 sizes, made in USA
- Oak & Cherry
- Custom Cut outs
- Ready to finish
- Full kits available

616-534-9121
info@meniscusaudio.com
www.meniscusaudio.com

SUBSCRIBE AND SAVE!

SAVE NEARLY $50 OFF THE COVER PRICE!

Call 888-924-9465
or subscribe on-line at
www.audioXpress.com.

SPEAKERBITS

Foam, cloth and rubber surrounds, spiders, cones, dustcaps, horn diaphragms, world's best voice coils and brad wire.

Genuine foam surrounds for Dynaudio, Tannoy, Scan-Speak and more.

Trade welcome-email for your user ID.

www.speakerbits.com

Soundstring
CABLE TECHNOLOGIES™

FOR THE PUREST

See our color ad on the front inside cover.
For more information on our Patents and Technologies visit www.soundstringcable.com

www.AUDIOCLASSICS.com

AUDIO • HOME THEATER • VIDEO • GUITARS
3501 Old Vestal Road, Vestal, NY
New • Preowned • Bought • Sold • Traded • Repaired

607-766-3501 OR 800-321-2834 HOURS: M-F 8-5:30, SAT. 11-4

Ad Index

Yard Sale

FOR SALE

Tube amplifier kit, PCL82.
Visit http://audio.engineeringvista.com.

Headphones (all in original boxes). Stax Electret, SR-34 with SRD-4 Energizing Adapter. Like new. $90. (Ref. *Stereophile* 12/87).
Sennheiser HD-540 Reference. Excellent condition. $60. (Ref. *Stereophile* 12/87, Audio "Auricle" 6/92.)
Sennheiser HD-424. Like new. $30.
Sony MDR-V6. Like new. $60.
Speaker Builder & audioXpress. 120+ issues, 1984 on.
(904) 641-5830 (EST—10 AM to midnight)
JohnKlaus@Bellsouth.net.

THETA DS Pro GEN. Va Digital Processor, with balanced option. List $5,595 excellent condition, original box and owner's manual. Asking $2,250. Please contact Clay Jordan at (334) 272-4090.

Goodmans, English 15″ woofer.
Tannoy Enclosure for above.
Bass reflex.
Rusyl Hillstrom
(406) 487-2023
PO Box 325
Scobey, MT 59263

Will swap 2W for 1V and 2A. Will also consider 2V and 1A. allen@hahax@verizon.net.

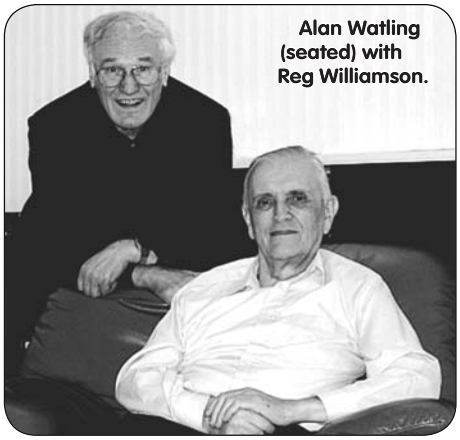
"Yard Sale" is published in each issue of *aX*. For guidelines on how subscribers can publish their free ad, see our website.

Alan Watling—A Tribute

By Reg Williamson

My extra career as an audio designer and technical journalist began in 1953, but it wasn't until the transistor started to dominate all innovative audio design work that it became really interesting. Most of my early work was published in the British magazine *HiFi News*, which, at that time, encouraged its readers with frequent construction projects. Not any more, of course.

I began to get regular letters from Alan Watling, who clearly shared much of my thinking and invariably had some valuable ideas of his own. It was only later that we discovered we both worked for the same company, the British telephone utility that eventually became British Telecom. Alan was a senior management executive in our Regional Headquarters; I (at that time) a junior area field engineer.



Alan Watling (seated) with Reg Williamson.

It became obvious Alan had a range of unique graphic and construction skills and was the only enthusiast I knew who could draw a printed circuit by hand. Give Alan a schematic and before long he could produce a beautifully constructed prototype. In this respect I was relatively inexpert so before long it became obvious that between us was the makings of an exceptional partnership.

It also became a friendship that extended over 50 years; but what makes it unusual is that our personalities were so disparate. I am sure that at times, he thought my love of good food and fine wine somewhat hedonistic. Similarly, my tastes in music as being questionable. But, as is so often the case between friends, such differences could be accepted with humor and mutual respect. They never got in the way.

I soon recognized that if I spun off an idea on Alan, and he thought it viable, then I only had to design the circuit and within a week or so I would have back a prototype for testing. I acknowledge without hesitation that the majority of my work published in *Audio Amateur* might not have seen fruition without Alan's participation. He, in turn, used to make occasional contributions with articles of often dry humor and wit.

He was also a talented painter, specializing in highly detailed pictures of flowers, which he called his "botanicals," and the occasional landscape. Some of these grace my own home. Alan was also a keen speaker designer, pursuing the ultimate in sound all the time.

In 1997 he asked me whether I would design a high and low pass filter for an active system he wanted to make. I had long been familiar with the application of gyrators in filter design, culminating in our last major project ("A New Control Preamplifier," *TAA* 1/92) so I was able to meet his request using the most advanced audio analog circuitry. Not only did I get back a prototype for testing within a couple of weeks, but to my amazement, he'd decided to make it by hand with surface-mounting components, thus pushing his construction skills to the ultimate limit. I can only speculate on the time and patience involved, using a magnifying glass and miniature soldering iron.

Sadly, it was to be his last construction venture since a crippling stroke was a few months away, paralyzing most of his right side. For a dedicated painter, that could have meant the end of a passion. Not so for Alan. With impressive fortitude and characteristic determination, he began painting with his left hand until ultimately, it became very difficult to tell the difference.

He was 82 when he passed away after a short stay in hospital. *aX*