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JOHN STUART MILL

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About This Issue

Aided by the expanded capabilities of Liberty Audiosuite, veteran loudspeaker tester **G.R. Koonce** examines driver response in two-way speaker systems. He concludes that a simple repositioning of your drivers can make a big difference in a pair's sound performance, provided they have the proper crossover configuration ("Side Saddle: A New Two-Way," p. 10).

Like life, speaker building is a journey. For **Gregory Smith** ("A Shielded Trio," p. 22), his path to complete a home-theater system covers three years. His crowning achievement is a (large) center-channel speaker. Along the way, he added a pair of speakers to his computer system and wrestled with video-display interference problems.

In the aptly titled "Measuring Loudspeaker Resonance" (p. 30), **Mark Williamsen** shows how you can measure loudspeaker resonant frequency using several different circuit configurations.

The audio industry's annual rite of winter—CES in Las Vegas—is usually an accurate barometer of the status of the industry. This year was no exception, as **C. Victor Campos** discovered several interesting new technologies ("Covering the Winter CES," p. 34).

Publisher Ed Dell's crack staff of reviewers worked overtime to present their assessment of some new loudspeaker products: a versatile spectrum analyzer for speaker analysis (p. 40), a protective device for your loudspeaker (p. 42), and a software analyzer for audio testing (p. 45).

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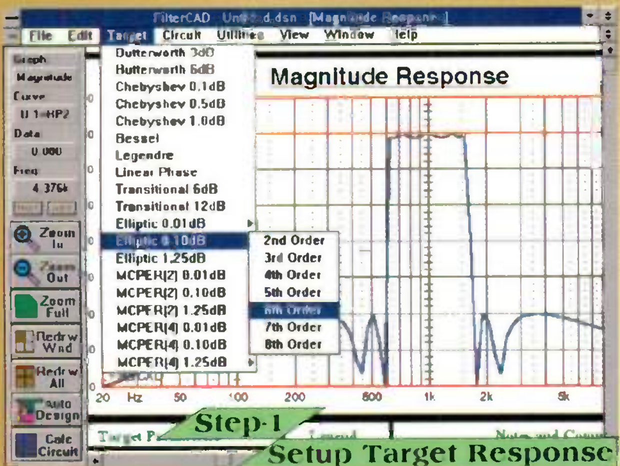

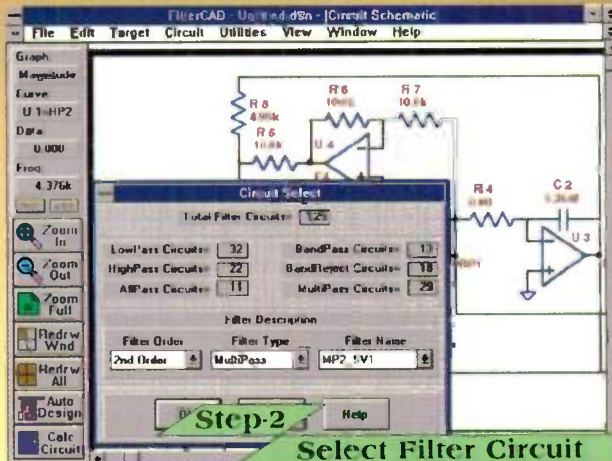

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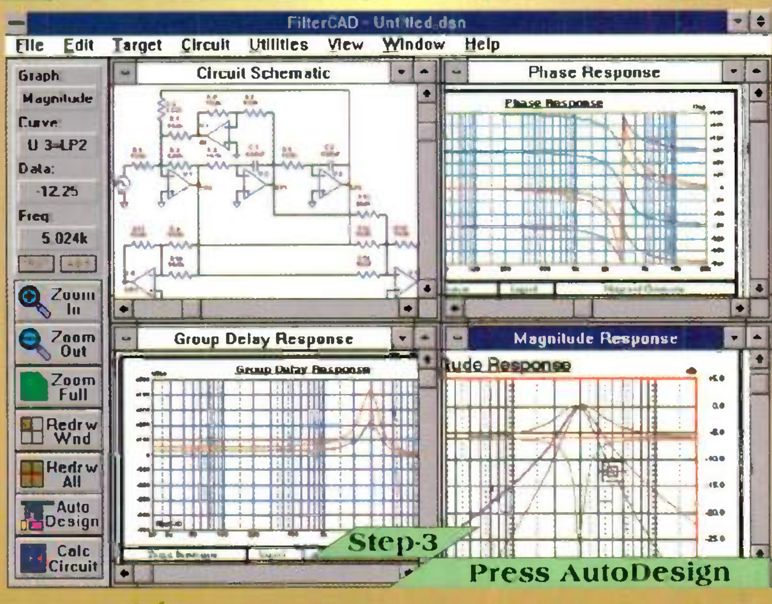

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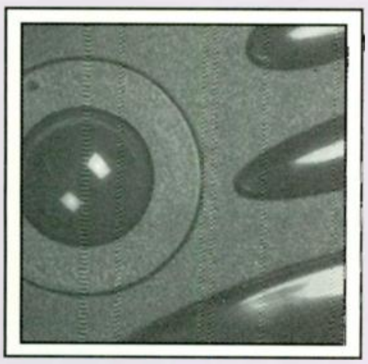

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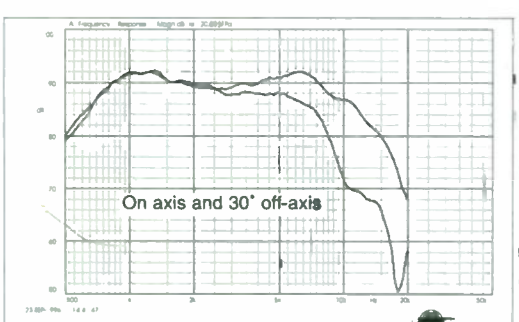

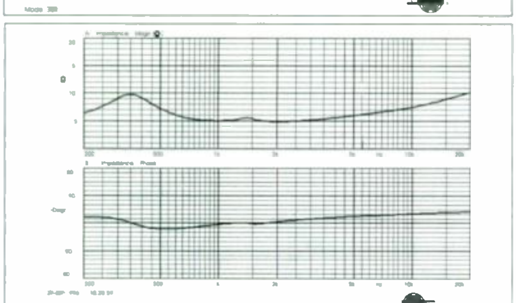

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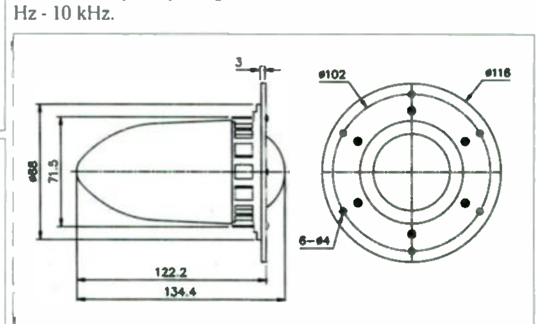

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SIDE SADDLE: A NEW TWO-WAY

By G.R. Koonce

For many years I have built two-way systems with a vertical alignment of the tweeter over the woofer. Now, with the testing capability of Liberty Instruments' Audiosuite, I believe other configurations may offer advantages. This article describes my investigation, including the results of experimenting with time-adjusting a small two-way system.

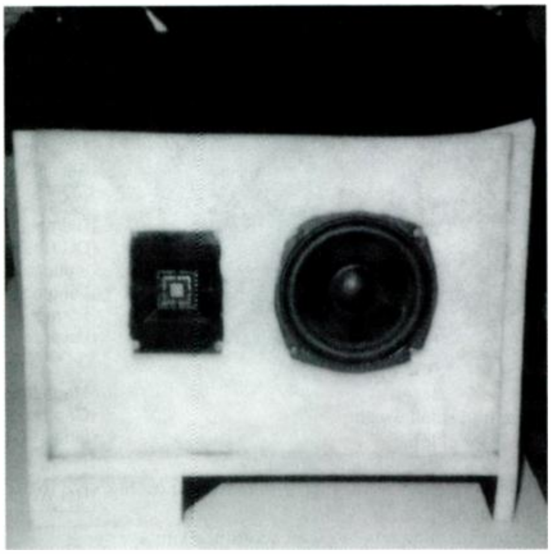

PHOTO 1: Finished two-way 5¼″ woofer system—front view.

TESTING

Liberty Audiosuite allows you to make anechoic measurements in a real room by gating the period of the acoustic time response that generates the frequency response. This enables frequency-response measurements on the individual drivers, the drivers with their crossover (CO), and the summed response of the driver set with the CO at desired angles off-axis. Most of the data presented in this article was obtained with the drivers mounted in a round, 6′-diameter test baffle offering results good down to at least 400Hz. All Audiosuite testing was via MLS test signal.

The systems covered in this work were not available to me at the time I prepared this article, so I had to use test results I had saved as files. Thus, sometimes there are slight differences in the CO network between data sets, etc. This is not an attempt to use "smoke and mirrors" on readers; when the differences are important to the discussion, I will explain them.

While anechoic testing allows you to see what the drivers are doing, it does not show what will happen in a real room. Both of these areas are important and must be considered. I will present some data taken via pink noise and 1/3 octave real-time analyzer (RTA). Note that the newest versions of Audiosuite can perform this test, but the data I present here are via hardware RTA.

INITIAL DEVELOPMENT

During the development of a three-way system with R. O. Wright ("A Modest-Cost Three-Way Speaker System," *SB* 6–8/96), we passed through the examination of a two-way system that used an 8″ woofer and a low-resonance tweeter. It was during work on that system that I developed the conclusions presented here. Keep in mind that all I say here is valid only for a two-way system using a third-order Butterworth (BW) CO with the woofer and tweeter connected with the same polarity.

In addition, the driver-response limits were not sufficiently far from the CO frequency to make the acoustic CO shapes true third-order BW, a condition that is likely with any two-way system. Keep these limitations in mind when considering application of the approach presented here to any two-way design of your own.

The 1″ dome tweeter and 8″ woofer were mounted vertically aligned, with the tweeter at the top, in the 6′ test baffle. *Figures 1* and *2* show the anechoic response of the system

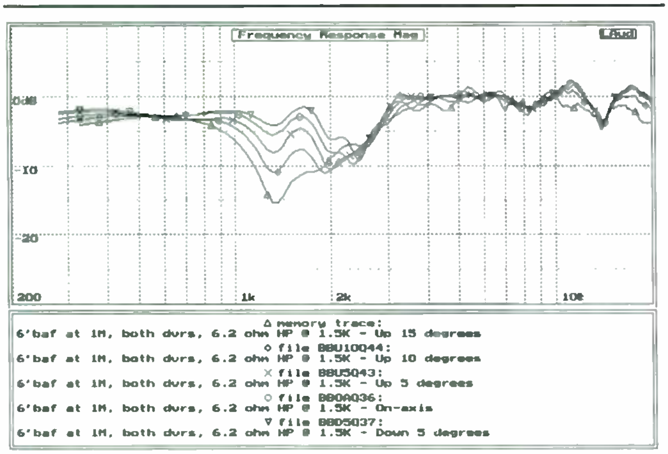

FIGURE 1: Acoustic summation response of 8″ two-way system at angles from up 15° to down 5°.

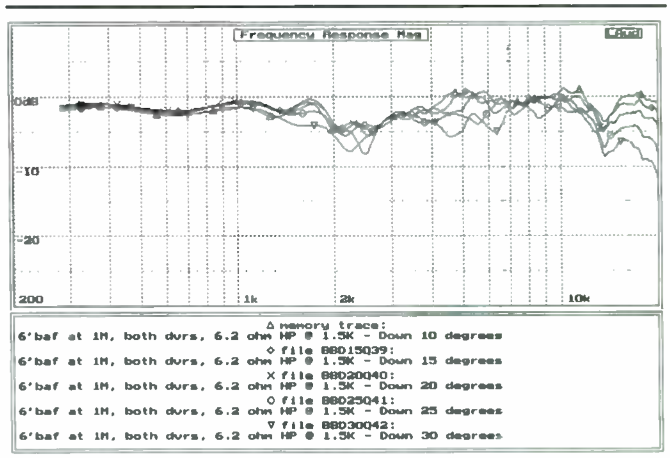

FIGURE 2: Acoustic summation response of 8″ two-way system at angles from down 10° to down 30°.

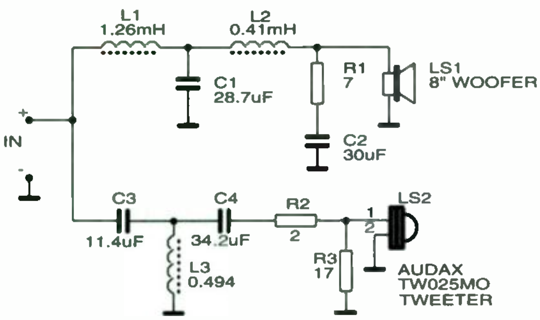

FIGURE 3: Crossover schematic for two-way system.

measured at vertical angles from up 15° (above tweeter) to down 30° (below tweeter and toward woofer). It is clear that for angles above about down 10°, a "hole" is present in the 1–3kHz region.

Figure 3 shows the CO I used for these tests; I modified the high-pass (HP) from the theoretical design values to "push" the tweeter response for a smoother summed response. Further modification of the HP CO can help to fill in the 2–3kHz dip shown in *Figs. 1* and *2*. The results of the modification appear in *Fig. 4* for various test angles, and *Fig. 5* is the HP schematic.

The penalty paid for this response improvement is shown by the system input impedance (*Fig. 6*), where the impedance dips to 2.3Ω. This is typically the problem you encounter when trying to "push" the driver responses to produce a good two-way system, and it is basically why we had modified our project to a three-way approach.

SMOOTH LISTENING

It is clear that you will get the smoothest response by listening to this system on an axis of about down 15–20°. That is, you must position the system front panel so that when you're seated, your ears are on a line about 15–20° below the tweeter and toward the woofer. If the enclosures are set on speaker stands, this is achieved by slightly tipping the front panel, but for a system standing on the floor, it can be difficult.

I normally design to fit my normal listening condition of having my seated ear height at about 42" and back about 10' from the front of the speakers. *Figure 7* shows various ways of implementing this listening angle with a floor-standing two-way system. With the normal tweeter-over-woofer configuration (*7a*), you need to tip the front panel about 30°—an excessive tip angle that can lead to an enclosure that will fall over backwards. I breadboarded this configuration, and it did sound good, but it looks strange and has a sound quality that varies with listener height, i.e., standing versus seated.

Inverting the configuration by putting the woofer above the tweeter (*Fig. 7b*) yields a nearly vertical front panel, which won't tip over and looks reasonable. But breadboard tests showed the bass response was hurt badly by having the woofer so high, and it also suffers from the problem of sound-quality variation with listener height.

The basis of this article is shown in *Fig. 7c*. It may look a bit strange, but the woofer and tweeter are mounted side-by-side at the same vertical height. The front panel has been tipped about 15° so that both drivers are nearly on-axis toward the listener in a vertical angle. The result is that the vertical response is now nearly independent of listener height, since it is composed mainly of the directivity of the individual drivers.

VERTICAL-PROBLEM SOLUTION

So the configuration in *Fig. 7c* solves the vertical-response problem very nicely, but what happens in the horizontal plane? *Figure 8* shows the typical listening setup in my garage. If the enclosures face straight out, rather than being toed in, it is clear the listener is on an angle of about 20° to the front panel. So if the boxes are built in mirror-image pairs, with the tweeters always outboard, the listener will be on about a "down" 20° angle, which is the most desirable.

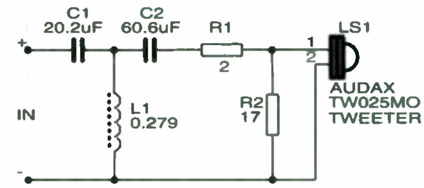

FIGURE 5: Schematic for modified high-pass crossover.

I reconfigured the breadboard system to match *Fig. 7c* and judged the sound in monaural to be quite good, consistent for both standing and seated positions. This still leaves questions about the stereo imaging, but I felt this approach deserved further investigation. I tested the breadboard two-way system discussed above with pink noise and hardware RTA. For these tests, the front panel was vertical, with the normal tweeter-above-woofer alignment. For most tests, the microphone was back 54" and at the tweeter height, so vertically the measurements are on the tweeter axis. As noted above, these tests include reflected energy, more indicative of the system power output than the summed response on any axis.

The CO and tweeter-padding values I used are not always the same as in the baffle tests above, since work was going on throughout the testing period to smooth the system response. *Figure 9* shows the results

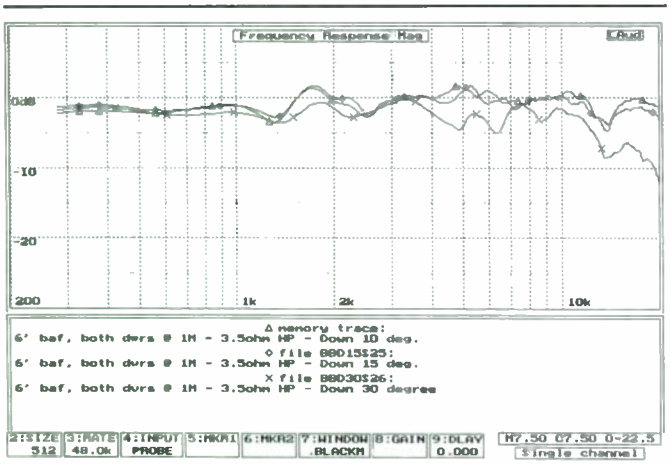

FIGURE 4: Acoustic summation response of 8" two-way system with modified high-pass crossover.

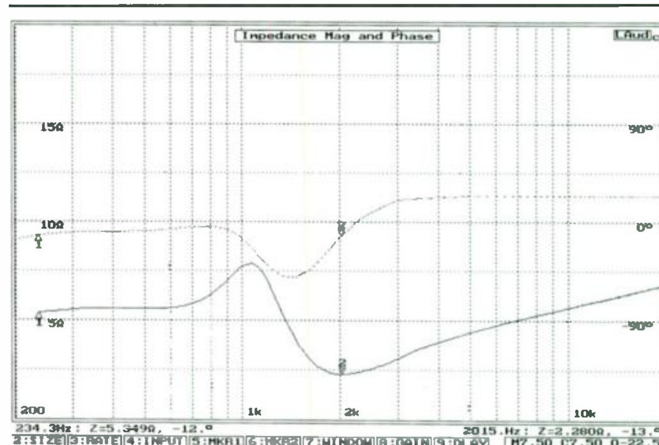

FIGURE 6: System input impedance with modified high-pass crossover.

for three tweeters. The Audax TW025V2 is a nominal 4Ω tweeter that requires a different HP CO network, but the Audax TW025MO and SEAS 25TAC are nominally the same impedance and use the same HP network.

Note that in these RTA tests, the plotted points are the RTA 1/3 octave band centers and represent the average of more than one test. When I tested with noise, the data moved around a bit, so do not try to read too much into small variations. It's interesting that an indicated excess energy occurs in the region of the 1.5kHz CO frequency, whereas the on-axis anechoic response would lead you to expect a "hole" there.

Figure 10 shows measurement results with the Audax TW025MO tweeter for on-axis and horizontally off-axis at 15° and 30°; this is moving to the side of the vertically aligned woofer and tweeter. Note that the on-axis curve does not match this tweeter in *Fig. 9*—typical of noise-measurement results. *Figure 11* shows the same horizontal directivity results with the SEAS 25TAC tweeter, again with some differences in the on-axis response versus *Fig. 9*.

The basic result of these off-axis tests is that the response changes smoothly, with only a slight loss of high-frequency response. If you tip the system over 90°, resulting in the side-by-side woofer and tweeter, these curves represent the vertical response, showing how insensitive this approach should be to listener height.

THE BASIC ASSUMPTION

My basic assumption is that the best way to build a two-way system (using a third-order BW CO with both drivers wired with the same polarity) is to rotate it 90° from the typical tweeter-over-woofer configuration. The advantages are that the now-vertical response changes very gradually with listener height, and the normal stereo positioning for speakers and listener (*Fig. 8*) tends to place the listener on the best horizontal listening axis.

A slight toe in of the boxes can fix the horizontal listening axis for unusual speaker or listener locations. All you need do is build the boxes with the front panel tipped back properly to put both drivers on-axis with the seated listener's ears, and then build mirror-image boxes with the tweeter always outboard of the listener.

In trying this approach, I had on hand a pair of rather unusual 5.25″ woofers by Dayton (Catalog #295-200 from Parts Express) and some surplus planar tweeters. The woofers were unusual in that they would go to almost 40Hz in a vented box of about 0.55 ft^3 (see Small model results in *Fig. 12*). The 2–3dB peaking is intentional; it not only keeps the box smaller, but I believe it results in a better-sounding system when you're building with small woofers.

BENSON MODEL RESULTS

Figure 13 shows the Benson model results for one of the woofers. Note that below about $1.2 \times f_s$ (about 70Hz) the port is making the main contribution to the system output—which means these things have a chance of working well for moderate listening levels. I decided to try them in a two-way system with the bargain planar tweeters.

Figure 14 shows the on-axis frequency response for these woofers as measured in the 6′ baffle. To get a high enough frequency to work with the intended tweeters, it's necessary to push the woofer up to about 3.5kHz. This means you must tweak the low-pass (LP) CO to account for the woofers' rising response above 1kHz. This is not something I want to do with systems that others would try to duplicate, but it is reasonable in "one off" pairs, since driver repeatability is not a concern.

Figure 15 shows the on-axis acoustic responses of the two drivers with a third-order BW CO designed for an electrical CO at 3.5kHz. The tweeter response is quite good; note that these bargain planar tweeters

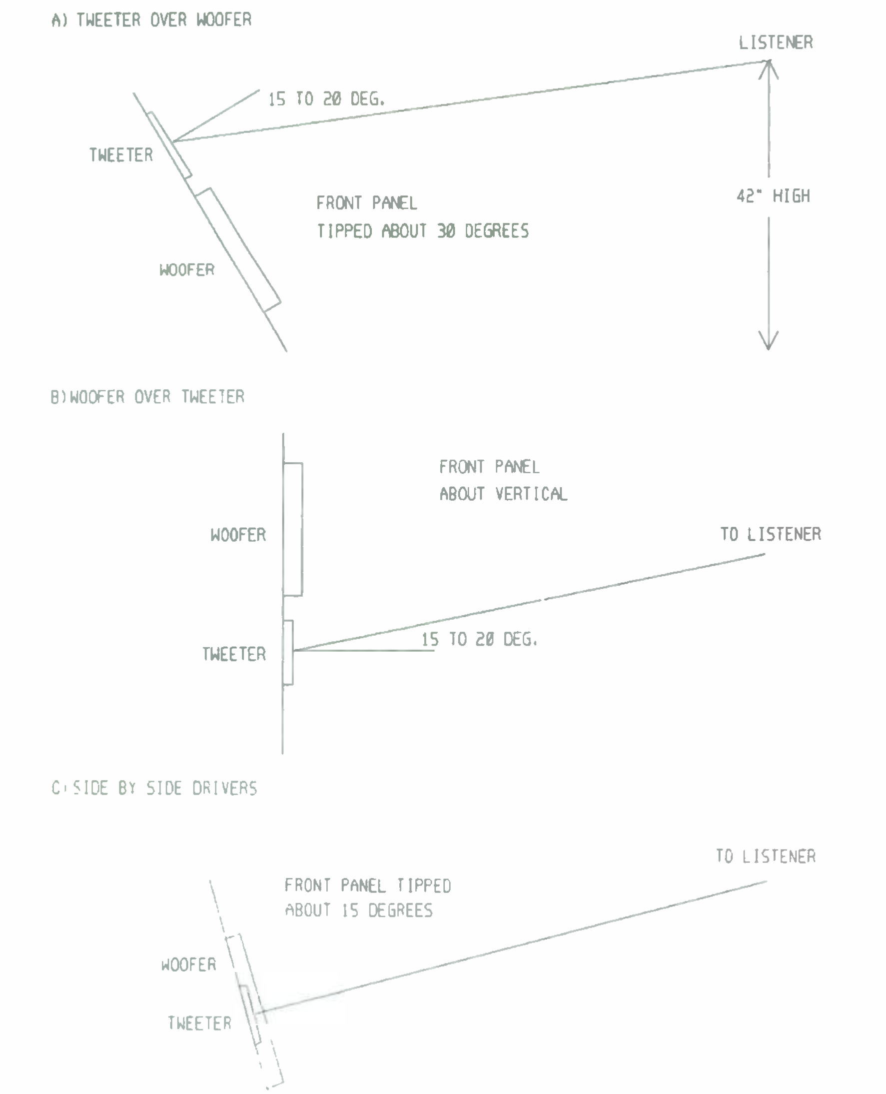

FIGURE 7: Front panel layouts for the proper listening angle.

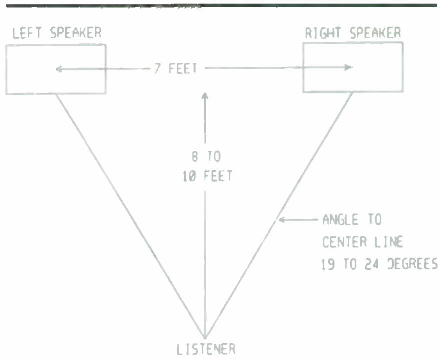

FIGURE 8: My typical stereo listening geometry.

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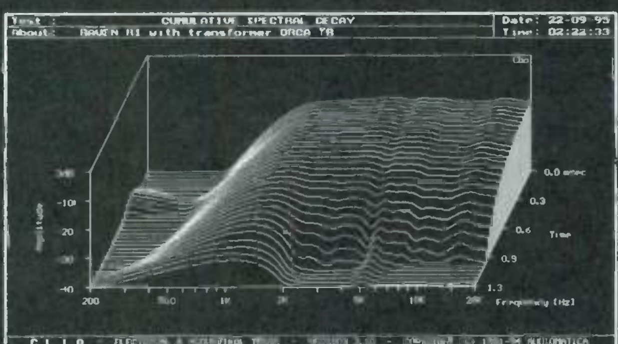

RAVEN R1
KGS 1.14
LBS 2.5
92 x 80 mm
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Moving mass:
0.0061 g
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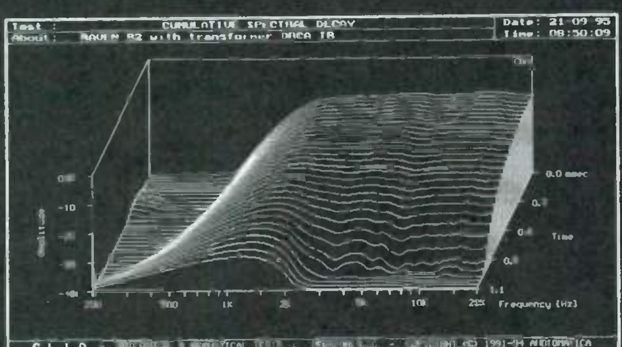
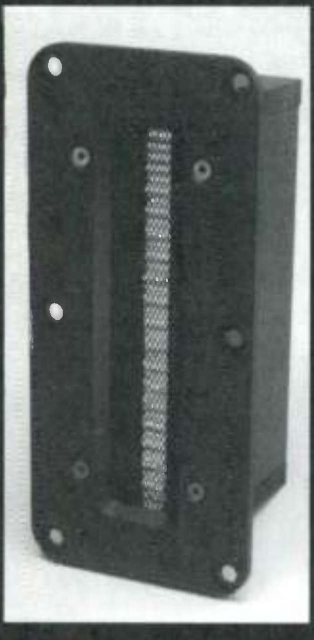
dB/W/m 95
2 KHz to 40 KHz



RAVEN R2
KGS 2.22
LBS 4.9
166 x 76 mm
6.54 x 2.99 in.
Moving mass:
0.013 g
0.0005 oz.
dB/W/m 98
2 KHz to 40 KHz



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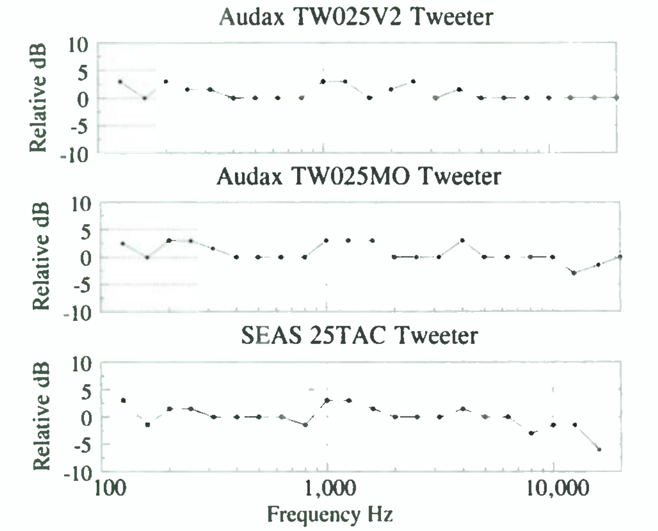

FIGURE 9: Live room on-axis response of system with three tweeter types.

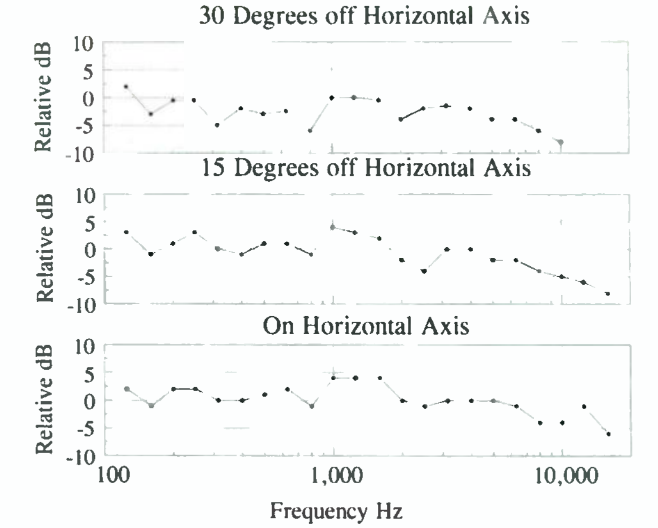

FIGURE 11: Live room system response at horizontal angles—SEAS tweeter.

go to 20kHz without the dip common to 1″ dome tweeters.

As expected from *Fig. 14*, the woofer response is not an acoustic third-order BW LP, and you must play with it. *Figure 16* shows the results after reducing the LP design to 3.0kHz for the total system response at angles of down 5° to down 20°. These look quite good for listening on a down 15° to 20° angle to a two-way system.

I breadboarded this system and listened to it on an axis of about down 15°, and it sounded quite good. The remaining question was how a pair of them would image. You have to have a pair to answer this.

A TIME-ADJUSTED SYSTEM

One of the capabilities of the later Audiosuite software is the ability to plot the "step" response of a driver or system. This is the response of the test item to a single input edge, such as a step from 0V to 1V and then staying there. Since speakers will not generate DC acoustic output, what you ideally get is a sharp rise (or fall) and then an exponential decay back to zero acoustic output.

Real drivers alone or in groups will not produce this ideal response. However, if the multidriver system is in proper alignment on the test axis, you will get a single rise and decay toward zero. If they are not aligned, you will see individual rises and decays for each driver.

I placed the 5¼″ woofer in a vented test box and positioned the tweeter on top so I could move it back and forth. *Figure 17* shows the step response for the system with no offset in tweeter position relative to the woofer. The tweeter step is clearly visible ahead of the woofer step, at about 2.5ms in the plot. Somewhat surprisingly, the tweeter step appears to be of opposite polarity to the woofer step, although both are connected with the same polarity. Is this a mistake, or an effect of the CO I used (still third-order BW)?

Figure 18 shows the impulse time response of the woofer. The bottom trace shows the electrical impulse into the woofer with 3.0kHz LP CO at the extreme left edge. The top trace is the acoustic impulse out of the woofer for this input. The inversion is normal in my test setup.

Figure 19 is the same result for the tweeter wired with inverted polarity and without its HP CO; as you would expect, it is clearly out of phase with the woofer. *Figure 20* is the same except with the 3.5kHz HP added, and now the main tweet-

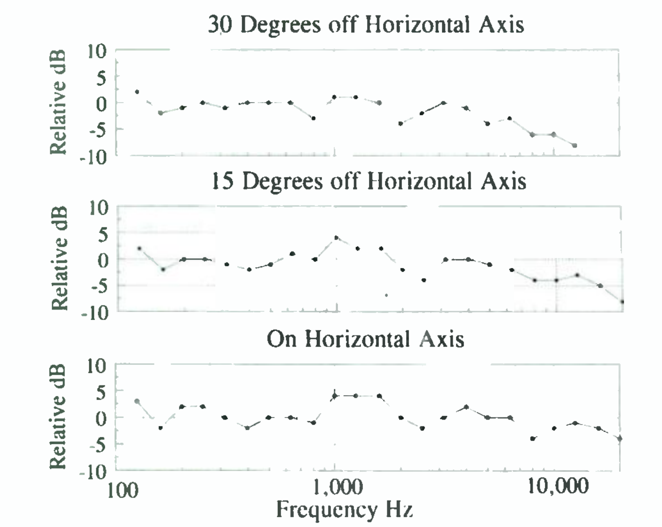

FIGURE 10: Live room system response at horizontal angles—Audax tweeter.

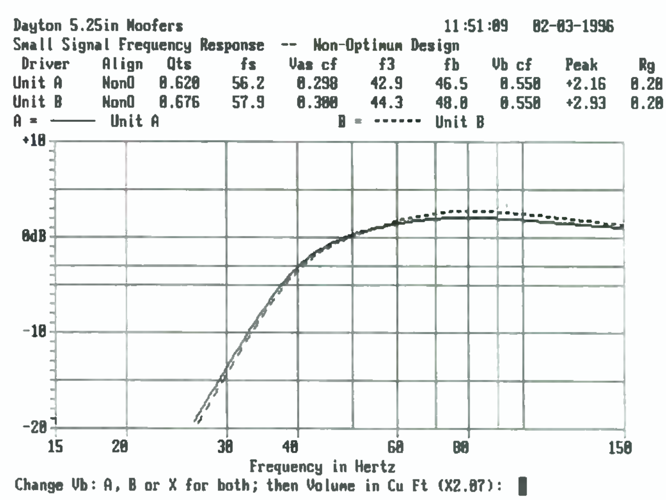

FIGURE 12: Small model predicted bass response for 5.25″ woofers.

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270	**Roederstein(ERO) 1.0 mfd Mylar** cap., PC mount, 250V, 31mm x 25mm x 14mm, 6mm leads	10/$4.00
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9,000	**ERO 2.2 mfd Mylar** cap., axial, 10%, 250V, 12mm x 30mm long, 43mm long leads, Green	10/$4.00
1,000	**ERO 2.2 mfd Mylar** cap., PC mount, 63V, 18mm x 13mm x 7mm , 6mm leads, Green	10//$2.00
1,200	**Panasonic 2.75 mfd Mylar** cap., 10%, 250V, 24 mm x 8 mm x 16mm, Dipped, 32mm leads	10/$3.00
30,000	**T.I. 3 mfd Mylar** cap., axial, 10%, 100V, 32mm x 14mm diameter, 90mm leads, Yellow	10/$4.00
1,200	**ICC 3.9 mfd Mylar** cap., axial, 10%, 100V, 30mm, 17mm x 7mm, 42mm leads	10/$6.00
1,400	**Unknown Mfg 4.7 mfd Mylar** cap., PC mount, 400V, 26mm x 17mm x 8mm, white	10/$4.00
18,000	**Elpac 5 mfd Mylar** cap., axial, 10%, 50V, 10mm T x 7mm W x 31mm L, 55mm leads, Yellow	10/$4.00
1,000	**Marcap 6.8 mfd Mylar** cap., 10%, 100V, 28mm x 15mm x 26mm, Dipped, 31mm leads	$0.80
11,000	**Mitsubishi 10 mfd Mylar** cap., 10%, 100V, Dipped, 14mm x 24mm x 30mm, 34mm leads	$1.00
34	**Steel Bobbin 10.0mH 16awg inductor**, 0.95Ω DCR, 64mm diameter, 44mm tall	$6.00
600	**Audax DW50M 14mm poly dome tweeter**, 4Ω, 94dB, Fs 2050, 45 watt, direct drive poly dome with diffuser, fluid cooled, 50mm round flange(no mounting holes)	$5.50
200	**Philips AD163 1" textile dome tweeter**, 8Ω, Fs 1300Hz, 50 watt, 92dB, 92mm x 80mm flange(3.6" x 3.2"), frequency response from 3,000Hz to 20kHz, a classic	$10.00
140	**MB Quart MCD-25S 1" Titanium dome tweeter**, 8Ω, F3 1000Hz, 90dB, 100W, 3.75" square, similar to MCD-25M tweeter but with a single magnet, smooth response to 20kHz	$18.00
200	**Audax TW60Ti 10mm titanium deposited dome tweeter**, 4Ω, 60mm square flange, comes with autosound hinged wedge mount and 3mfd mylar capacitor for 6dB crossover	$15.00 a pair
100	**Peerless 821214 2" dome midrange**, textile dome, chambered back, Fs 298 Hz, 89dB, 200W@1kHz, 134mm flange(5.27"), 106mm cutout(4.17"), smooth response to 5kHz, useful for frequencies down to 800Hz with a 12dB crossover, depth is 74mm (3")	$28.00
36	**Dynaudio 19W38 6.5" poly cone woofer**, 4Ω, rubber surround, vented pole piece, 1.5" voice coil, sandwich cone construction, 89dB, 70 watt nominal, 3mm X-max, Fs 46 Hz, Vas 17.6 liters, Qms 2.4, Qes .44, Qts .37, Re 2.9Ω, very smooth response to 5kHz, F3 of 87Hz in .2 ft^3 sealed or F3 of 50Hz in .4ft^3 vented with 2" diameter vent by 6.7" long	$40.00
90	**Eminence 8253 8" poly cone woofer**, foam surround, 40 oz magnet, 2" diameter voice coil, 4mm X-max, Fs 34Hz, Vas 58.5 liters, Qms 5.99, Qes .34, Qts .32, good response to 2.5kHz, 91dB, F3 of 75Hz in .5ft^3 sealed or F3 of 47Hz in .9ft3 vented with 2" Ø vent x 4" long	$29.00
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28	**Audax AMTW74A 10mm dome tweeter**, 4Ω, 3" flange, poly dome, shielded magnet, 48mm cutout with 22mm depth, Fs 2900Hz, 94dB, 40 watts @ 5kHz, freguency range 5kHz to 25kHz	$5.50
93	**Vifa D25SF-04-06 1" poly dome tweeter**, 4" truncated flange(round with top and bottom squared off), shielded magnet, 6Ω, 92dB, Fs 1000Hz, 50 watt, frequency range 3-24kHz	$15.00
200	**Stanford Acoustics MSF77B20U, 3" Full Range** speaker, 4 ohm, shielded magnet, paper cone, foam surround, 20 watt, 89dB@2.83V, Fs 83Hz, Vas 2.38 liters, Qms 2.75, Qes .55, Qts .46, F3 of 130Hz in 1 liter sealed enclosure, full range response to 10kHz	$5.50
250	**Stanford Acoustics MSF100B20U, 4" woofer**, 4Ω, shielded magnet, paper cone, foam surround, 20 watt, 89dB, Fs 68.5 Hz, Vas 4 ltrs, Qms 1.86, Qes .43, Qts .35, smooth response to 3kHz, F3 of 140Hz in 1.3 liter sealed or F3 of 85Hz in 2 liter vented, 1" Ø vent x 4" long	$7.50
700	**Vifa C13SG-15-09, 5" woofer**, 9Ω, semi-shielded with bucking magnet, doped paper cone, stamped frame, rubber surround, 89dB, Fs 49 Hz, Vas 16 liters, Qms 2.1, Qes .44, Qts .36, 35watt, frequency range of 95Hz to 9kHz in .2 ft^3 sealed enclosure	$12.00
200	**Stanford Acoustics MSF165B30R, 6.5" woofer**, 8Ω, shielded magnet, paper cone, foam surround, 30 watt, 90dB, Fs 33.7Hz, Vas 49.3 liters, Qms 3.17, Qes .36, Qts .32, good response to 4kHz, F3 of 73Hz in .4ft^3 sealed box or F3 of 48Hz in .7ft^3 w/ 2" Ø vent x 5" long	$12.00
200	**Seas P17RCD 6.5" woofer**, Same as P17RC, but semi-shielded with bucking magnet, cast frame, poly cone, rubber surround, Fs 35Hz, Qts .32, Vas 40.8 ltrs, 60 watt, 90dB, 3mm x-max, F3 of 78Hz in .4cf^3 sealed or F3 of 50Hz in .6ft^3 vented, 2" Ø vent x 6" long, smooth to 4kHz	$28.00

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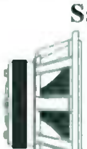

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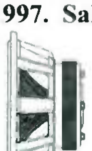

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MADISOUND MADISOUND MADISOUND MADISOUND MADISOUND MADISOUND er impulse has been inverted. It was clear that to get a well-aligned system, I would have to use the tweeter polarity inverted, still a valid configuration for good theoretical on-axis summation with a third-order BW CO.

PADDING FOR TESTING

After some playing with the tweeter offset, I obtained the step shown in *Fig. 21* with 3.1″offset; this was about the best I could do. Note that this amount of tweeter setback corresponds to about 0.23ms time delay. Interestingly, the testing in this breadboard configuration required much less tweeter padding than the baffle tests. I was using a different tweeter sample, but the change was much greater than this would require.

For the baffle tests, I padded the tweeter 5dB to match the woofer (a 5¼″ woofer that goes to 40Hz is not too efficient), but on the alignment breadboard, the padding required to match the woofer was only 2.5dB. The tweeter is being used in these tests without any baffle, but I have never been able to fully account for this difference.

Figure 22 shows the measured response of this system from up 10° to down 10°. These responses are not as nice as those obtained for down-angle listening in the baffle with normal tweeter polarity, but are certainly good enough to justify listening to see what precise alignment offered.

I set up the breadboard system so that it was on-axis toward my seated listening position, with the tweeter moved to the correct 3.1″ offset. About 30 seconds of the first CD told the story: the sound was terrible. The tweeter level was way too high. I removed the 2.5dB pad and replaced it with the 5dB pad from the baffle tests. I thought this was too much padding, estimating that about 3.5dB would be best, but I made no more padding changes.

I changed the tweeter polarity to normal (destroying the alignment), which greatly improved the sound, but it was still not nearly as good as listening at the down 15° approach I developed earlier. It was clear that the alignment I had arrived at did not work well at all, a condition that is typical with high-order COs.

This shows why breadboard listening is an important part of system development, since more than a nearly flat on-axis response is needed to produce a good system. I'm sure time adjustment has merit, but not at the expense of other important requirements for a system. I decided to proceed with the side-by-side system.

THE SIDE-BY-SIDE

I built the side-by-side system with a 0.55 ft^3 vented box and diffuser port (*Photo 1*). About one half of the bottom area is consumed by the port diffuser while the CO is packaged in the remaining space. The two boxes were constructed as mirror images and used with the tweeters outboard.

I was extremely pleased with the sonic performance of these boxes within their limited power capability. If I build again with small drivers going to 40Hz, I will increase the box size and reduce the design response peak to perhaps 1.5dB or less. (I have since built with a 6½″ woofer going down to 40Hz with slightly under 1dB peaking and now recommend a very low peaking, even with

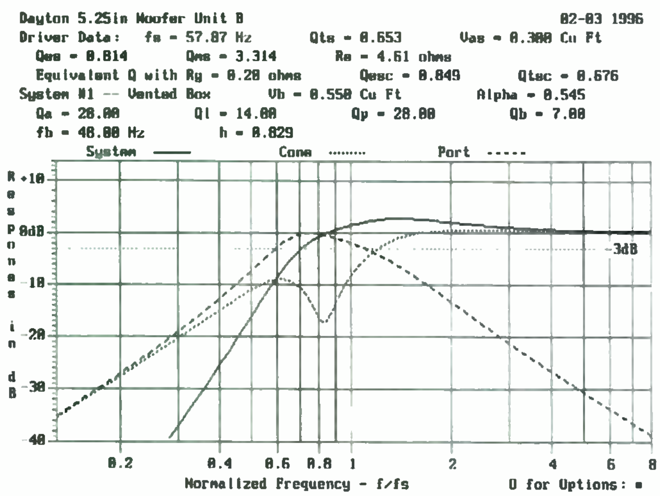


FIGURE 13: Benson model predicted bass response for woofer B.

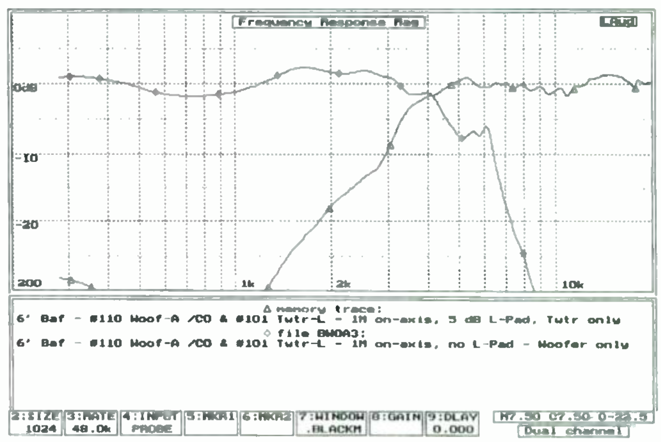


FIGURE 15: Acoustic high-pass and low-pass responses with 3.5kHz crossover.

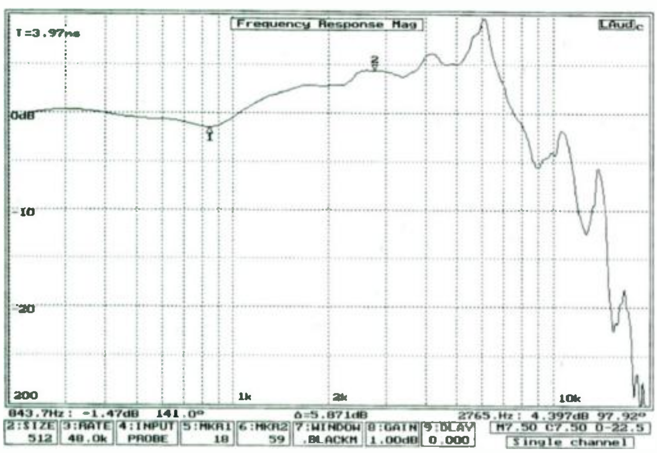


FIGURE 14: Measured acoustic response of 5.25″ woofer.

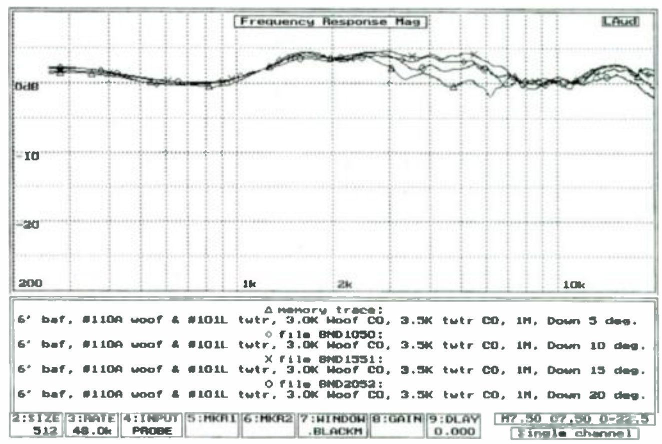
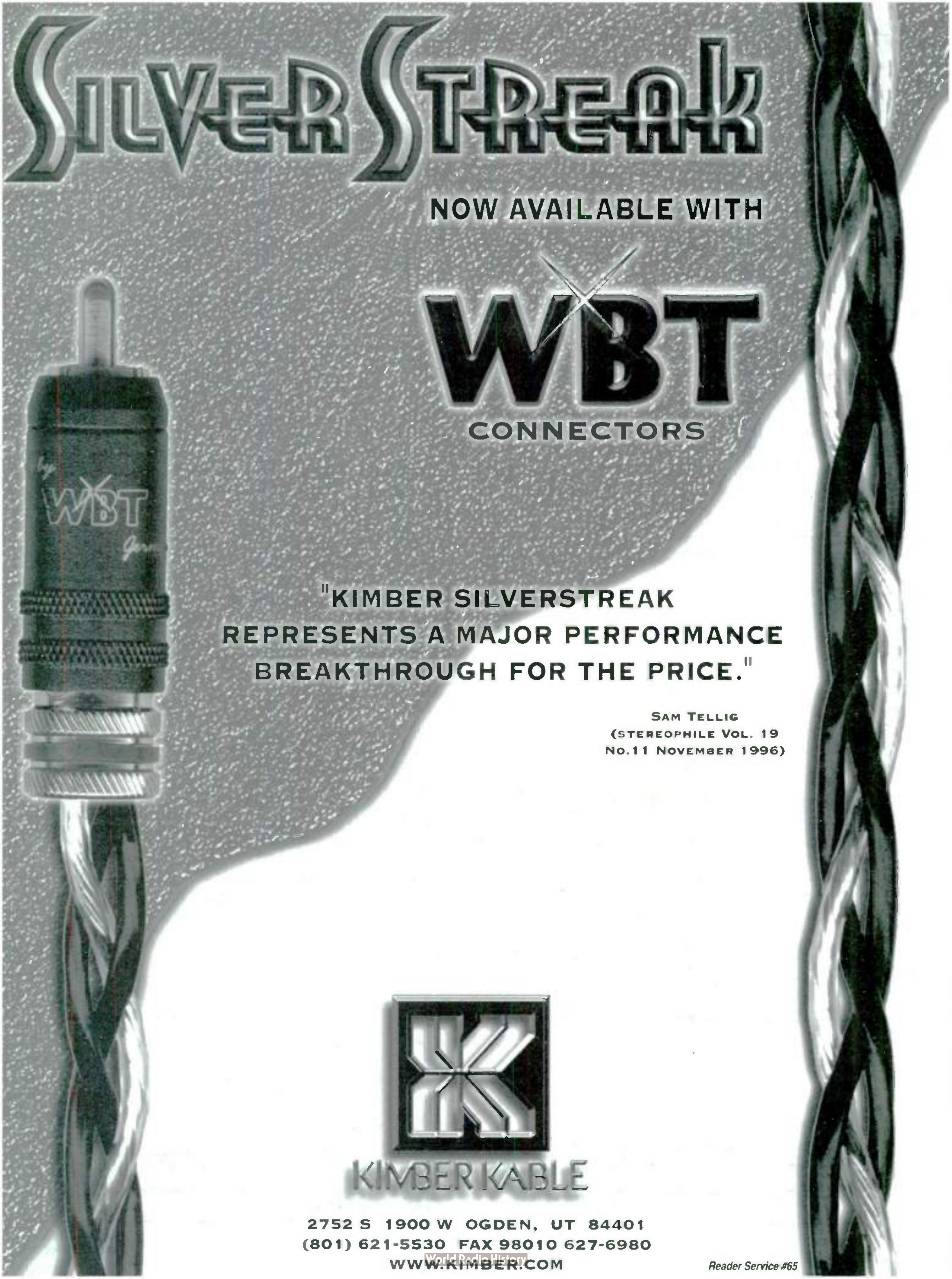


FIGURE 16: Acoustic summation response for 5.25″ two-way system at angles from down 5° to down 20°.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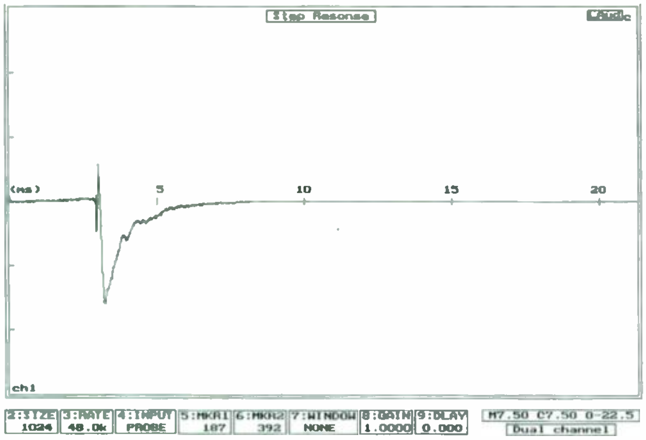

FIGURE 17: System step response with normal tweeter polarity and no tweeter offset.

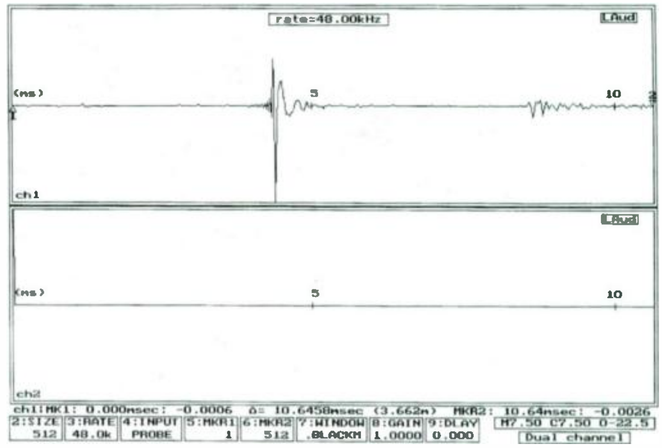

FIGURE 20: Impulse response for inverted tweeter polarity with crossover.

small drivers if they go low in frequency.) I measured the bass response of one box via near-field test, and the summed system response is shown in *Fig. 23*. This indicates the planned 3dB peak is there and the –3dB cutoff is about 42Hz.

You should take these as approximations, for to generate this curve, you must get the near-field driver and port responses set to the proper relative level, which is difficult with the diffuser port. Without arguing about 1 or 2Hz, it is clear the box measures very close to what the models predicted.

My main question about using the side-by-side approach was how well they would image. The answer is: very well. At the sweet spot, I could hear nothing to indicate the configuration was not "typical." The sound quality was consistent whether I stood or sat.

The real surprise was when I moved off the sweet spot. Normally I have found that the third-order BW CO yields a rather narrow sweet spot, with the image decaying very quickly as you move off center. These boxes formed an image as long as you were anywhere within the bounds of the two boxes; clearly, the image pulled off center, but it was still there. I plan to try this configuration more in the future.

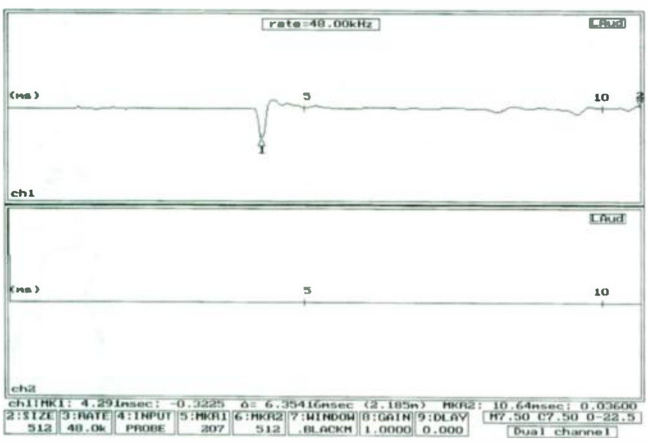

FIGURE 18: Impulse response for normal woofer polarity with crossover.

CONCLUSION

While not covered in this work, data show that if the tweeter is wired in inverted polarity to the woofer, then the boxes should be used with the tweeters inboard of the

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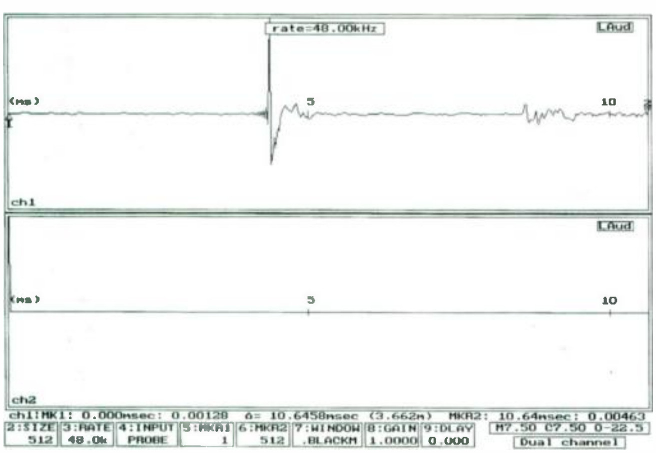

FIGURE 19: Impulse response for inverted tweeter polarity without crossover.

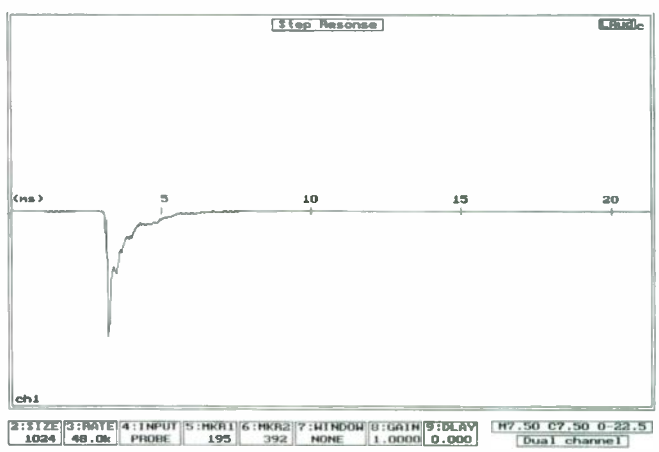

FIGURE 21: System step response with inverted tweeter polarity and 3.1" tweeter offset.

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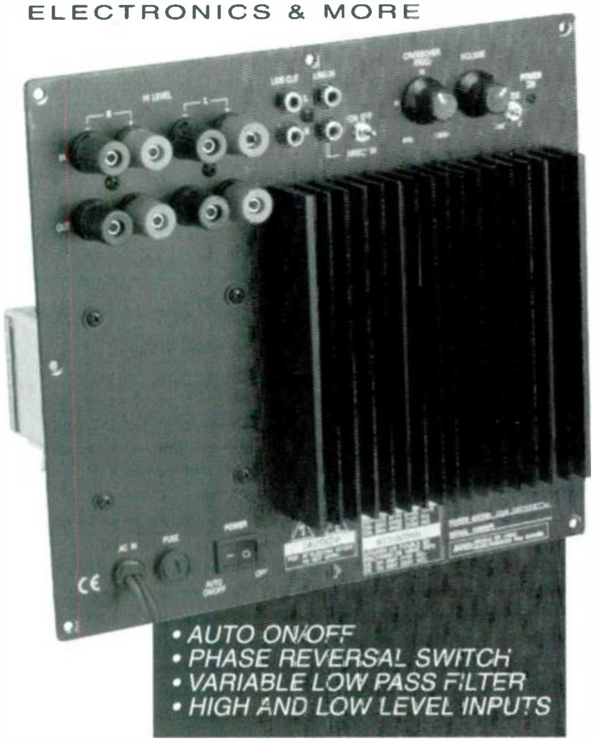

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woofers. For the two systems tested, the measured performance in this configuration was not as good as for the tweeter hooked in normal polarity.

One advantage is that the sound quality of the system is relatively insensitive to listener height when the drivers are mounted at the same height. The normal geometry used for two-channel stereo listening also positions the listener on a good horizontal axis for a flat response. The one pair of speakers built with this approach produced an image over a wide horizontal angle, making them ideal for general listening while still having a realistic and normal image at the sweet spot.

If you own a pair of two-way systems with the proper CO configuration, it is easy to try this concept. Tip them over on their sides with the tweeters outboard for normal tweeter polarity, or the tweeters inboard for inverted polarity, and put something under the front of each box to tip them so the listener is on-axis in the vertical plane. The results could surprise you.

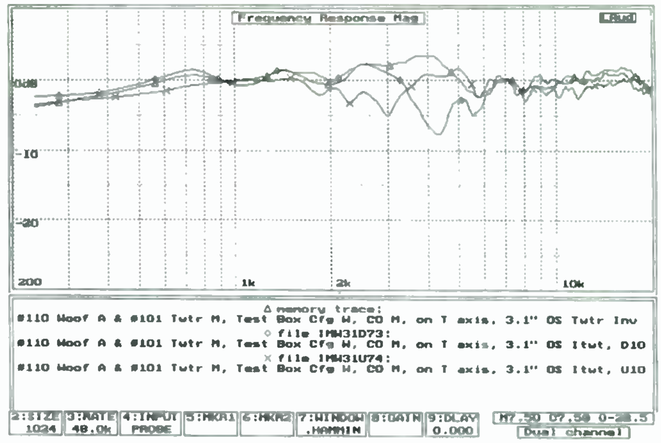

FIGURE 22: Acoustic summation response with inverted tweeter polarity and 3.1″ offset for angles from up 10° to down 10°.

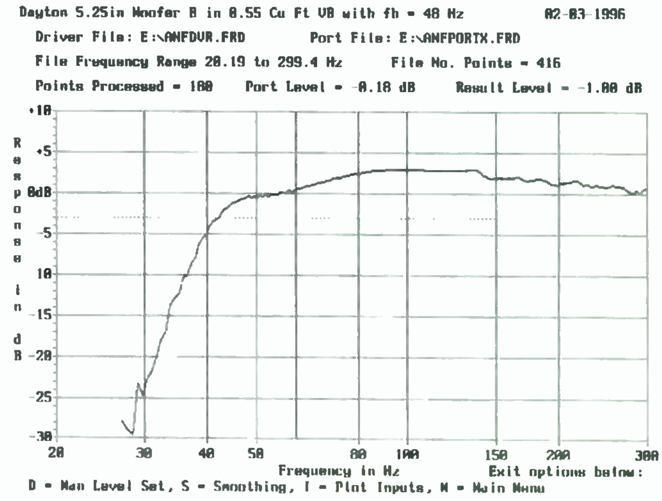

FIGURE 23: Measured bass response via near-field for box with woofer B.

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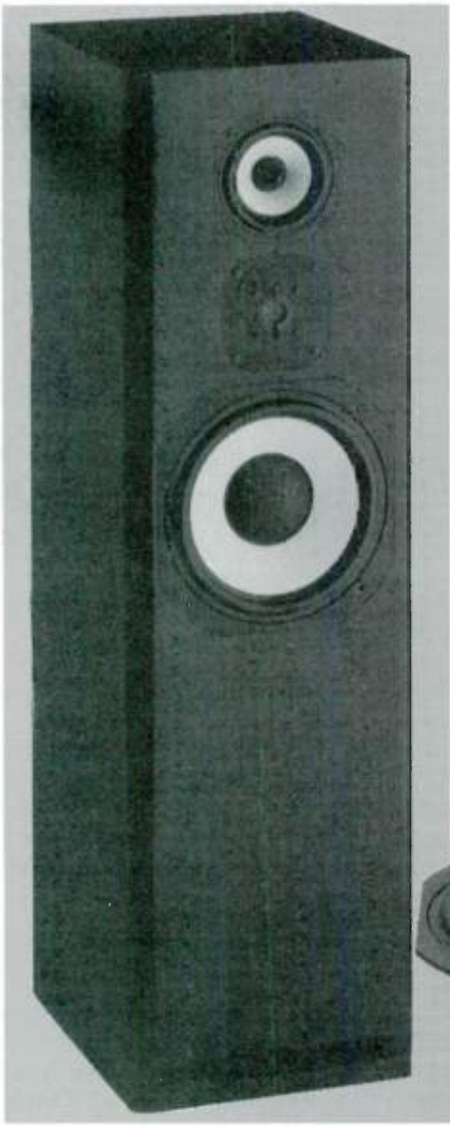

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A SHIELDED TRIO

By Gregory Smith

When I started building speakers, I already had many, in stereo pairs. Only when I started putting together a home theater and a computer-based music system did I need more—specifically, the magnetically shielded designs that both these applications require to preclude interference with their nearby video displays.

My original intention was to build a center-channel speaker and a pair of speakers for the computer system. That's not what I ended up with, but what I learned in achieving my goal is much of the story.

GETTING CENTERED

Usually the centerpiece of a home-theater system is a Dolby Pro-Logic surround receiver. In this configuration, you have at least two front speakers, and a mono surround channel is usually carried by two rear speakers that are fed the same signal. (My previous *SB* article[1] details construction of my inexpensive rear channels.)

The main advantage of Pro-Logic over the earlier Dolby Surround is that it can utilize a center-channel speaker (although it's possible to operate with a "phantom" center image created by the left and right speakers). At first, I never really missed having a dedicated center channel in my home theater, but when my audiences grew, it became difficult to get everyone seated in the sweet spot where there is a center image. So I decided to build a center-channel speaker.

Usually such speakers are designed to sit on top of a television, but the top of my TV was already crowded, so I decided to make mine large enough so that the TV could sit on top of *it*.

PROCESSOR OPTIONS

Depending on your surround processor, you may include an option for your center speaker to run either in "wide" mode, where it is fed the entire frequency spectrum, or in "normal" mode, where the deeper bass (below 100Hz) is distributed equally to the left and right speakers.

Some processors use other techniques; for example, the Audiosource processor I've been working with normally runs all the front channels full-range, but if you utilize the subwoofer output, it gets the low-pass output from an 80Hz crossover, while all the front channels get the high-pass output. If you don't need to reproduce all the lowest frequencies, you can have a much smaller center channel.

If you wish to experiment with a center-channel speaker without using a surround processor, it's easy to derive a center signal from a stereo source. Simply wire the center signal between the grounds of the left and right channels. You can do this at the preamp level and drive three amplifier channels.

A sample of this configuration by Paul Klipsch appeared in *SB*,[2] and an identical circuit drives a center speaker in my stock Chrysler/Infinity car stereo, so it's fairly popular. However, be careful in experimenting with this sort of thing, for unless your equipment has a common ground between channels, it can cause big problems; check your amplifier's manual to see if this is allowed.

In a center-channel-equipped home theater, the dialog in the sound track comes from the center speaker. Therefore, in order for playback to be as realistic as possible, you should locate the center speaker as near the display as possible. Having the sound seem to come from somewhere other than the screen is disconcerting. In a movie theater, speakers are located behind the screen, which is perforated with enough holes to let the sound through (the treble on film sound tracks is boosted to compensate for the loss incurred by beaming the sound through the screen). Since home systems with such screens are quite rare, the best way to synchronize the sound and picture is to put the center speaker right next to the display.

PHOTO 1: Crossover in the open back of the enclosure.

COMPUTER SOUND

Recent improvements in my computer had finally made its audio good enough that I thought it deserved something better than the mediocre speakers I had been using. Hooking up the SoundBlaster 16 card to my big stereo to test things out, I learned there was some pretty impressive sound to be had in some modern computer titles.

It seemed wise to have real full-range

ABOUT THE AUTHOR

Gregory Smith pays for his extravagant audio hobbies by working as a computer programmer. His writing on a variety of topics, including music and equipment, can be found on his World Wide Web home page at http://www.westnet.com/~gsmith.

speakers, for the bass on some sound effects and music added considerably to the effect. However, there was no room near my computer for a decent-sized subwoofer—only enough space for a stereo pair. I decided that in addition to the center speaker, I should build two more speakers for my computer, and that one of my design goals should be to achieve as deep a bass response as possible in the size I had to work with.

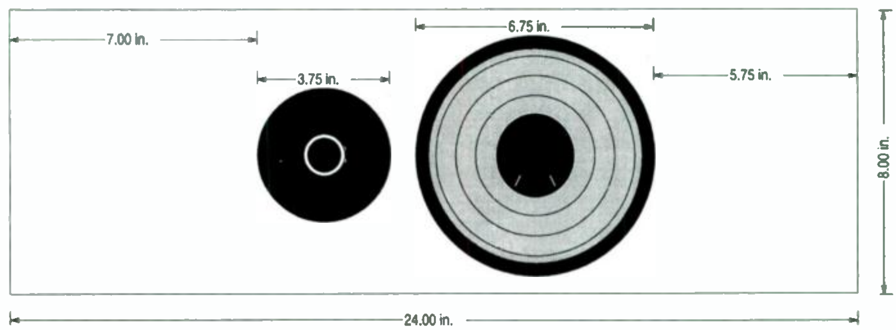

FIGURE 1: Center speaker front layout.

FIELD OF GREENS

Video displays using a cathode ray tube work by shooting a beam of electrons toward the screen. If you place a magnet near the tube, the magnetic field will exert enough pull to change the path of the electrons. The result is a smeared picture, usually a big green spot.

Traditional speaker drivers (even tweeters) contain magnets large enough to cause this problem. The simplest way to solve it is by moving your speakers farther away from the display to reduce the influence of the magnetic field on the CRT.

This may be an acceptable solution in many setups, but if you create your center-channel speaker close enough to the display so that the dialog appears to originate from it, it's probably near enough to warp the color. The real solution in this case is to use speakers that are magnetically shielded.

BUCKING THE TREND

A common myth is that you can shield drivers by using big sheets of metal. Lining the inside of your speakers with steel is not only expensive and heavy, but it doesn't work reliably. Some hold that you can get slightly better results by wrapping the magnet itself in a steel cup, but even this isn't very effective.

When creating shielded drivers, most manufacturers put a second magnet, known as a bucking magnet, behind the first to cancel out the stray rear part of the field that causes problems. The definitive article on this topic is Richard Pierce's "Tour de Magnetic Force," [3] a commentary on issues brought up in a Brian Smith article titled "Adjusting Woofers for High Performance."[4]

The focus of that discussion is that you can take another magnet, smaller than the one on the woofer, and attach it to the existing magnet with its polarity reversed. If properly done, this can weaken the rear field while altering the front and side fields, changing Q_{ES} in the process. But as Mr. Pierce has pointed out, "this is not a task that can be easily and successfully accomplished as a weekend no-brainer project."[5]

Obtaining the proper size of magnet is the biggest problem. If you have drivers you don't mind disassembling, it is possible to separate the magnets from the rest of the assembly. You can remove magnets from their pole plates by heating them enough to cause the adhesive to weaken.

Matthew Honnert recommends removing the easily melted plastic parts from the main magnet assembly, then heating the drive in an oven at 350°F for 20–25 minutes.[6] After removing the magnet and voice coil, the remaining portion could be recycled as a passive radiator.[7] However, my reading on the topic of adding a bucking magnet indicates that this is a very haphazard method, with the final outcome unpredictable.

DRIVER LESSONS

Having given up on shielding existing drivers because of all the discouraging information available, I set about finding good-quality drivers that were already shielded. I decided to purchase Vifa drivers from Madisound. At the time, Vifa had available the \$13 D19TD-03-08 ¾″ dome tweeter, designed to be used from 4kHz up (with an f_S of 1.4kHz).

Of the two midrange woofers available, I decided to use the \$32 6½″ M17SG-09-08, which claimed to be usable up to 5kHz. I was looking for a full-range system, and the 6½″ woofer was rated to have an f_3 point of 69Hz in a sealed enclosure and 45Hz in a vented one. The catalogs gave me the impression that Vifa had designed these drivers to be used together, but after finishing my system, I found their integration somewhat less than perfect.

What *was* excellent was the shielding on the drivers. I got only slight interference between the drivers and my TV and monitor, and that was only when the magnets were very close and pointing toward the screen. With the drivers enclosed, and their fronts parallel to the screen, no hint of magnetic-field distortion appeared in the picture.

Since I wasn't going to use the center speaker full-range, I had no need to make a vented design to extend its low-frequency output; a smaller closed design would be sufficient and would obviate sensitivity to low-frequency rumble. On the other hand, the computer speakers would benefit from the deeper bass of a vented alignment.

CENTER ENCLOSURE

Since I wished my center speaker to be a base for my 25″-wide TV and two feet is a convenient length for plywood pieces, I decided to make the center enclosure 24″ wide. To accommodate the 6½″ and yet keep the enclosure of ¾″ plywood as short as possible, I designed it to be 8″ high. Vifa's spec sheet for this woofer states an internal volume of about 0.6 ft^3 for a $Q = 0.7$ closed alignment. With internal dimensions of 22½″ × 6½″, this would mean a depth of 7″, which is not nearly deep enough to rest the TV securely on top.

I decided then to put an internal chamber in the enclosure for the woofer to unload into, making the external depth a full 13½″. As a bonus, I could leave part of the back open (nobody can see it behind the TV anyway), making it easy to remove and modify the crossover (*Photo 1*). *Figure 1* shows how I positioned the drivers on the front, but that's not what I recommend now.

When you're crossing over a woofer as high as this one, most of the frequency spectrum comes from it. Accordingly, when you place the woofer off-center, all the sound from the speaker seems also to be coming from off-center. If I were designing another speaker like this, I'd put the woofer right in the middle, and put the tweeter as close as possible to it, probably just to its upper left or right.

CROSS CROSSOVER

Checking the driver data, it seemed that the best crossover point was at 5kHz. I picked a 12dB/octave, second-order design (*Fig. 2*), which appeared to be the best compromise between tweeter protection and design-implementation cost. Since there was a moderate sensitivity difference between the woofer and tweeter, I inserted some L-pads to drop the level on the tweeter.

As an experiment, I initially left out the impedance-compensation resistor and capacitor across the woofer portion of the circuit; everything I read highly recommended this, and I was interested to hear the difference.

After installing the drivers and hooking up the removable crossover to the rear binding posts and L-pad, I did some listening. At first, I was quite pleased. Then, while adjusting the L-pad, I noticed a very strange phenomenon. As I was shifting the volume around, it seemed as though the crossover point was moving with it. Looking at the impedance graph for the woofer dispelled my surprise; it rapidly rises in this area, so that by 5kHz it's up to 16Ω (instead of the approximate 8Ω it has below that and above resonance).

Maybe you can make other designs work without impedance compensation, but with L-pads involved, I couldn't get anything close to a correct match between woofer and tweeter. After installing the compensation on the crossover board, things worked much better. Approximately 2dB of reduction matches the levels quite nicely, and you could just as easily use fixed resistors to attenuate this amount.

Overall, the frequency response seemed fairly flat, with a peak in the upper tweeter region. Vifa's graphs showed a fairly broad on-axis peak of about 3dB between 10 and 20kHz, which seemed to match with what I was hearing. The bass quality was excellent for a driver of this size, leading me to conclude that the driver parameters weren't far off from the specs.

The center speaker seemed done enough for the moment, so I put it into permanent position under the TV for extended evaluation (*Photo 2*). Building the enclosures for the computer pair was the next step.

BAFFLING CHOICES

I like narrow baffles, which not only look nice, but help reduce diffraction. An 8″ external width was as small as I could go. The monitor is about 17″ deep, so that was a good depth for the matching speakers. Doing the math to meet the specified ported alignment volume of almost 1 ft^3 made the external dimension for the height also 17″.

It's not exactly a golden-ratio-based design, but space concerns in this case must dominate over sound-quality issues. You can cut each of the speakers out of a 2′ × 4′ piece of plywood without too much trouble. Assembling everything, cutting holes for the drivers and installing them, and installing the port (2″ diameter, 5″ long) completes the enclosure itself (*Fig. 3* and *Photo 3*). I wired two duplicates of the crossover that I had worked out for the center speaker and attached them to binding posts in the back.

DISPERSION

Now that I had a stereo pair to work with, I did some serious listening to evaluate the speaker quality. The ported enclosures considerably improved low bass quality and quantity. There's always a price to pay, though; the modulation from lower bass notes made worse an already problematic crossover. Deep bass notes smeared and distorted the entire upper midrange.

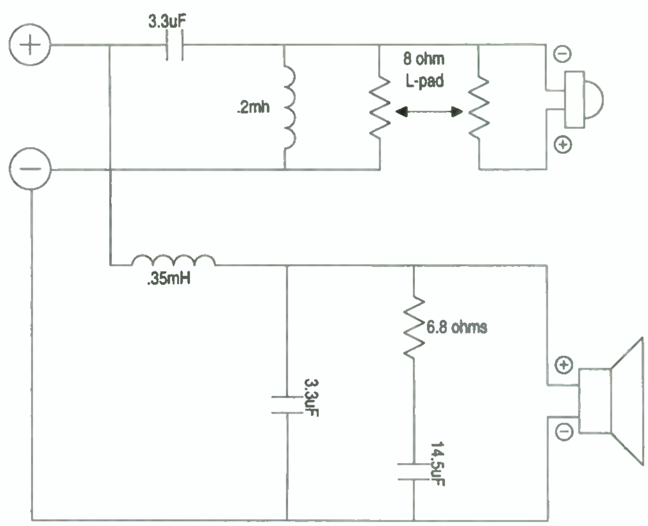

FIGURE 2: Crossover.

Imaging was weird. Looking at the tweeter-response graphs again, I saw that the response is flat if you move 30° off-axis. When I tried decreasing the speakers' toe-in to get enough off-axis to flatten out the tweeter problems, the centered vocals dropped off, and the center image collapsed.

This all makes sense when you think about it. The 6½″ woofer starts dropping its response fairly quickly above 2kHz when it's off-axis. Decreasing the toe-in had put a dip in the response from 2–4kHz, right where most vocals are. The lesson here is never to select drivers without considering the off-axis response interaction along with the simple on-axis one; if the manufacturer's data doesn't show it, be concerned.

There's a reason that certification programs for sound systems such as THX have very stringent requirements for dispersion. One way to get the narrow vertical and sufficiently broad horizontal coverage required by THX is to use a D'Appolito-inspired vertical MTM configuration. A horizontal MTM setup is not a good choice for a center-channel speaker. This factored in to my own design, although I certainly won't claim that controlled dispersion was one of my original goals. It was only after hearing some of the problems I got from ignoring this facet of the design that its importance really sank in.

PHOTO 2: The TV on the speaker enclosure.

ONE TO THREE

In my home theater, I had been using two original Advent speakers, rebuilt with new woofers, for my left and right channels. It became very apparent that even my non-optimal design for the center channel was far outclassing the old Advents.

Now that I'd finished the other two shielded speakers, it seemed reasonable to try out all three together as a matched set of front speakers. This worked so much better that I found another home for the Advents. I

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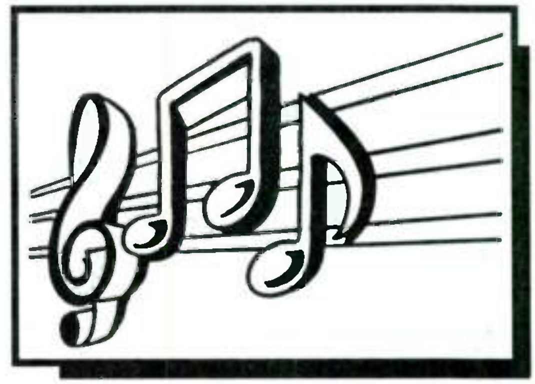

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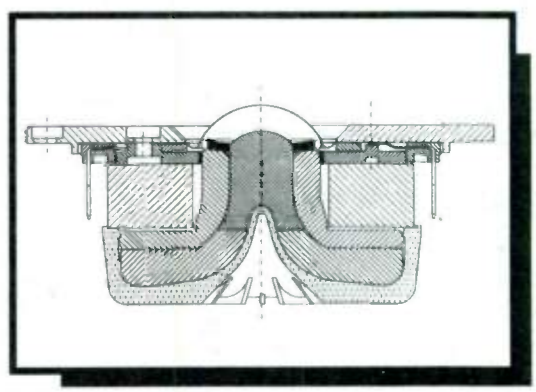

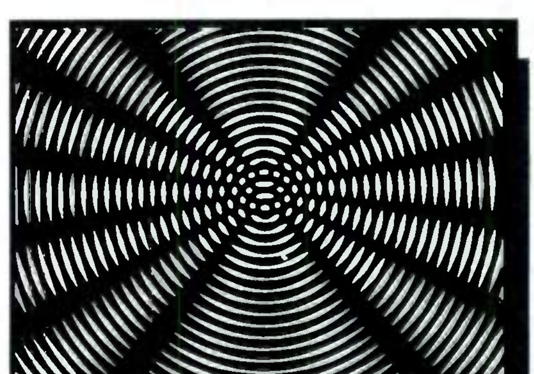

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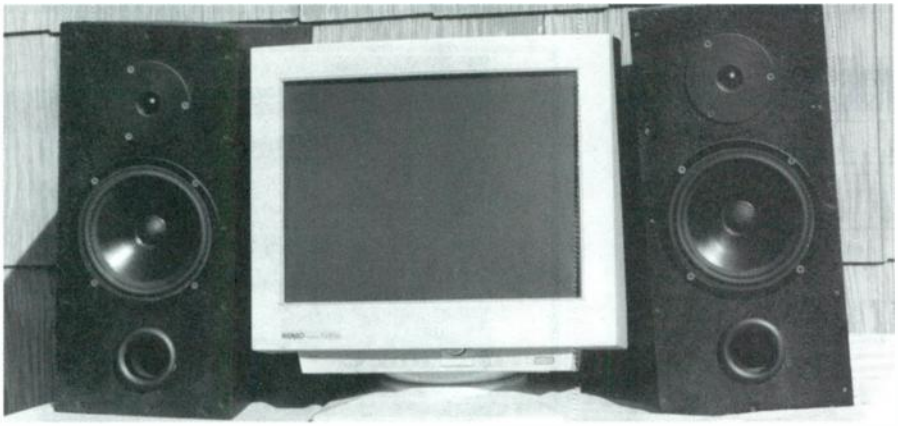

PHOTO 3: The vertical computer speakers.

built some simple stands and left the trio hooked up together. I made other arrangements for the computer system, since the matched set of three worked too well on the home theater to break them up.

There was another bonus to this arrangement. The treble of most movie sound tracks is unnecessarily boosted for my ear, producing a screeching I've always tried to correct with treble controls. The dip in off-axis response that I was trying to fix corrects this harshness perfectly. I hate to leave a flaw in something I build, but this one matched the movie-sound problem too well to correct. Therefore, I decided to call this one a wrap.

BOTTOM FEEDING

All these front speakers go down respectably into the bottom end, but they can't keep up with the deep material found on today's sound tracks. Because of this, I ended up recycling my first speaker project—a subwoofer originally designed for my car's trunk—for the home theater. My original design goals for that project were reasonable size (it had to fit in the trunk) and response down to below 30Hz.

To accomplish these aims, I used a simple design based on the popular Madisound

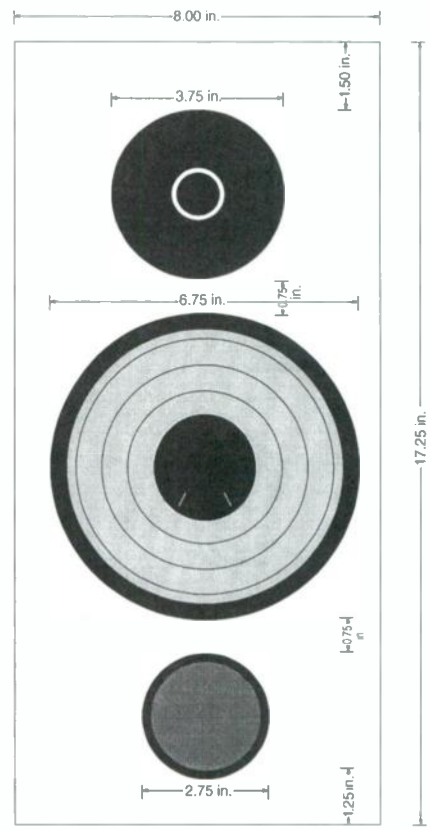

FIGURE 3: Left and right front layout for computer speakers.

1252DVC, a dual-voice coil 12″ model. Each coil is 8Ω and can handle about 50W for a decent output volume. The driver is well constructed, especially for its $42 cost.

Most sealed-box designs have an f_3 of approximately twice their resonant frequency, and this one is no exception. The unusually low resonant frequency of 15Hz lets the 1252DVC go down pretty deep, even without venting. Measuring the woofer specifications, I found it very close to the design chart, which recommended a sealed-box size of 85–130 ltr. This corresponds to a system Q_{TC} of 0.96–0.8 and a corresponding f_3 of 32–30Hz. Normally, a Q_{TC} this high would produce "boomy" bass, but since the resonant frequency of the design is in the 40Hz range, it's not objectionable.

The box I built was 80 ltr., rather on the small side. To make it work as well as possible, I bought two Scan-Speak flow-resistive vents, which are plastic hole covers with fiberglass-like filling, similar to the Dynaudio variovent. Placing these in the cabinet lowered the resonant frequency a few hertz. I calculated that the box size necessary to get that response without the vent was 90 ltr. The catalog description that these vents "let the woofer behave as if it's in a larger enclosure" seems accurate.

If you're putting something together with this woofer for home audio, you should certainly increase the size as much as possible while still keeping a good box tuning. My box required most of a 4 × 8 piece of plywood to make, and dealing with that is what convinced me to stick to smaller sheets in the future. The subwoofer cost about $125.

I used the electronic crossover in the surround processor. Since this woofer has considerable output all the way up to 2kHz, it will really screw up the rest of your system if you don't cut out higher frequencies. I don't recommend passive crossovers for subwoofers. Having an active crossover and amplifier dedicated to the sub lets it rumble away without affecting the other channels.

This dual voice-coil woofer lets you run it as a stereo 8Ω design or in parallel for a mono 4Ω load. Hook it up to 100W or so of power, and it sounds quite nice. Adding the subwoofer to the five channels of sound I already had finished the setup, but I wasn't happy with how it looked.

PAINT PAIN

My usual construction method utilizes lots of glue to keep a good seal, combined with an array of drywall screws holding everything tight. While this makes cabinets of fairly good quality, they certainly aren't good looking. For this project, I decided to try Abilene's Texturelac®, a spray finish that gives a textured look and feel to wood while covering all those nasty blemishes far better than regular paint. I got two gallons of black for approximately $90.

I employed a painting expert to take care of spraying the paint, which was not a simple process. The thickness of the Texturelac made it trickier to apply than regular paint, and my expert used an air-compressor and paint gun suited to this application. My painter told me that similar textured paint more suitable to amateur application is sold in auto-supply stores, with the intended application being things such as rocker panels.

We put two coats of paint over five average-sized speakers and two stands, using about ¼ of the two gallons. I recommend more than just two coats now that I've seen the final result. The value is excellent if you have a number of speakers to paint, and the drying time was right on Abilene's specification of 15 minutes.

Overall, I was rather pleased with the product, but don't believe Abilene's extravagant boast that it will fill in all the little holes in your wood. The screws in my boxes were all set slightly below the surface of the wood, but the Texturelac really didn't do well in filling in those small spots. You might investigate automotive body filler to fill in the holes cleanly before you spray.

I can't fully recommend Texturelac to everyone, as it really appears to be intended for industrial consumption. If you don't have good painting facilities available, along with the appropriate expertise, this isn't a good product to start with.

HINDSIGHT

In this project, my greatest regret concerns the tweeter. It was inexpensive and easily worth its cost, but was not really up to my inflated expectations from working with more pricey components. Since I blew out two of them under the high demands of the center speaker, I really can't recommend this model for that application.

On the other hand, Vifa's woofer is one of the nicest parts I've ever worked with, with the one caveat that you really shouldn't use it up to a full 5kHz because of its poor off-axis response. I believe a system with this woofer and a larger tweeter suitable for use with a 18dB/octave crossover at 3–3.5kHz could perform really well.

Overall, I'm satisfied with how this system turned out. My entire five-channel-plus-subwoofer home-theater speaker package cost around $500, and I think the result compares favorably with similar packages the big manufacturers have been releasing lately—usually for about $1000.

If you include the value of the time I spent building everything, it may not seem like much of a bargain, but that's not actually the case—building speakers should be fun, and you can't put a simple price on what you learn from building them yourself.

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SOURCES

Madisound Speaker Components
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MEASURING LOUDSPEAKER RESONANCE

By Mark Williamsen

One day recently it occurred to me that measuring speaker parameters is a lot like measuring other circuit components. Of course, I had read this many times, but I understood it anew when I suddenly realized that I could use a loudspeaker as the frequency-controlling element in an oscillator, and that the frequency of oscillation would exactly equal the loudspeaker's resonant frequency.

RESONANCE DEFINED

I knew this would be the case because resonance is usually defined as the frequency at which a loudspeaker, driven by a constant-current source, exhibits maximum cone motion. You measure this by looking for a voltage peak at the speaker's terminals, on the assumption that back electromotive force (EMF) will peak when cone motion is greatest.

The circuit of *Fig. 1* is typically used for this measurement.[1,2] R1 is an arbitrarily high value—between 100 and 1,000Ω—that approximates constant-current drive to the speaker. The sine-wave generator is assumed to be a voltage source that is flat with respect to frequency. This circuit also lets you measure impedance, as it varies with frequency, by reading the voltages across R1 and the speaker at each frequency of interest.

Note that of the techniques shown in this article, this is the only one that allows you to develop the impedance-curve data required for Thiele/Small design methods. Voltages in the circuit are determined at each frequency by the ratio $V_{spkr}/Z_{spkr} = V_{r1}/R1$. Solved for impedance, this becomes: $Z_{spkr} = R1 \times V_{spkr}/V_{r1}$.

RESONANCE REDEFINED

When making this measurement, you'll find that for some speakers the impedance peak is rather broad, making it hard to find the peak's center. You achieve a more precise measurement of resonant frequency by seeking the frequency at which voltage and current in the voice coil are in phase.[3]

Figure 2 shows an X-Y oscilloscope connected to present a Lissajous pattern[4,5] measuring the phase angle between voice-coil current (X input, which actually measures voltage across a resistor in series with the voice coil) and voice-coil voltage (Y input, connected across the speaker terminals).

You find the resonant frequency by adjusting the sine-wave generator's frequency until the oval figure on the scope closes to a diagonal line. For speakers in open air and in sealed cabinets, this will occur at one frequency. For reflex-type cabinets, you typically will find three frequencies that exhibit zero phase: two peaks and a minimum between them.

You can estimate the magnitude of impedance at resonance by reading the X and Y voltage amplitude of the scope trace.

TABLE 1

PARTS LIST

R1	100Ω ½W resistor	Radio Shack #271-1108
VR1	5kΩ linear potentiometer	Radio Shack #271-1714
IC1	UA741 operational amplifier	Radio Shack #276-007
D1, D2	1N914/1N4148 silicon diode	Radio Shack #276-1122
C1	0.0047µF disc capacitor	Radio Shack #272-130

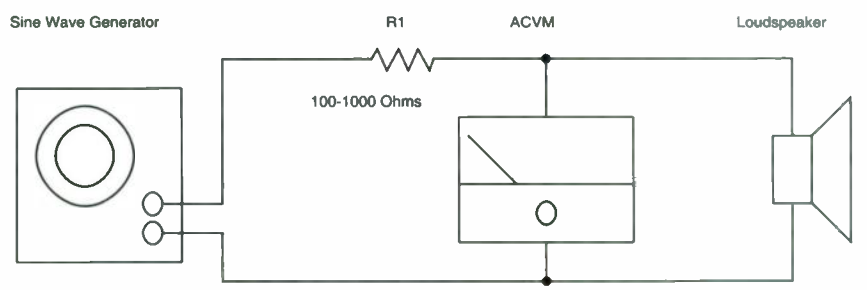

FIGURE 1: Standard setup for measuring loudspeaker resonance and impedance.

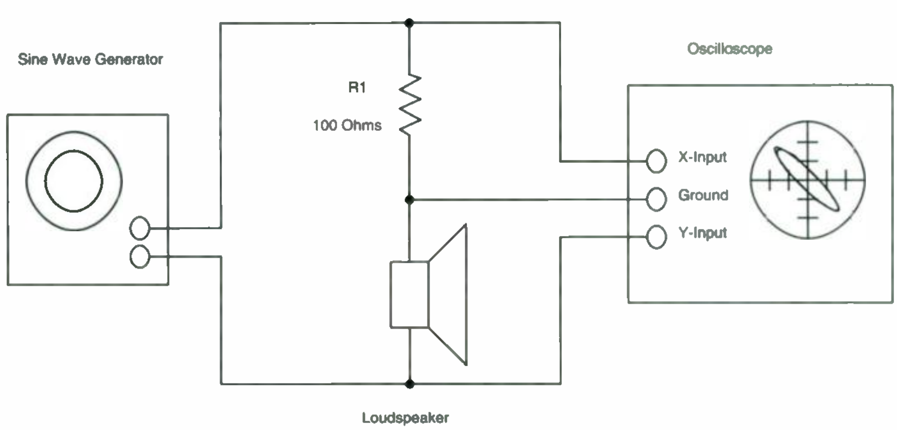

FIGURE 2: Oscilloscope connected to find zero-phase frequency.

ABOUT THE AUTHOR

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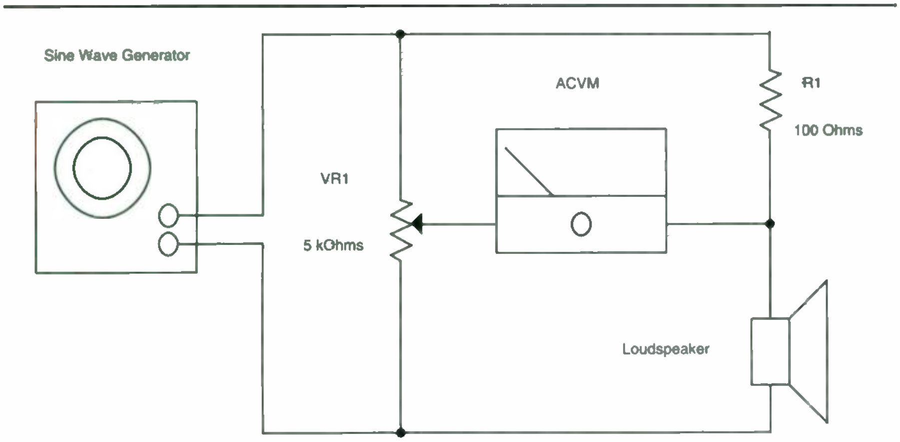

FIGURE 3: Bridge circuit for finding zero-phase frequency.

The scope's ground is shown connected to the junction between R1 and the speaker, since most scopes have a common ground between X and Y inputs. This means, of course, that the sine-wave generator's output must float with respect to the scope's ground. If you can't make the generator output float, use a 600Ω line transformer between the generator and the measurement circuit to obtain the necessary isolation.

For this measurement, I have been using a general-purpose scope that has an input bandwidth extending beyond 50MHz. In my urban situation, I see a lot of radio-frequency interference (RFI) superimposed on the scope trace. For a clear picture, you may need to experiment with shielding, grounding, or low-pass filtering of the scope inputs.

I have had good results by just inserting a 470kΩ resistor in series with each input to the oscilloscope. This combines with the scope's internal 30pF input capacitance to form a simple R/C low-pass filter that wipes out most of the RFI. You can also select a sine-wave generator with more voltage output in order to achieve a greater signal-to-noise ratio.

The X input is shown connected across R1, rather than across the speaker, thus resulting in greater amplitude, since many scopes have less gain on their X inputs. Select DC coupling, if available, for both inputs. If one or both inputs must be AC coupled, make sure that phase shift in the scope won't be a problem by connecting both X and Y inputs to the sine-wave generator and verifying that a diagonal line, not an oval, is maintained down to the lowest frequency of interest, typically 20Hz.

RESONANCE REQUIRES MOTION

The concept of resonance is meaningful only if the loudspeaker's voice coil is moving. Anything that interferes with that motion will alter or inhibit resonance. This could range from a chip or rub between the voice coil and pole piece, to placing the loudspeaker face down on a table while trying to measure it.

To see exactly how the voice coil moves in relation to the pole piece, buy a cheap loudspeaker and take it apart. Since the process is not easily reversible, I suggest spending not more than $10. First cut out the dust cap, the dome in the center of the cone. Inside, you will see the top of the pole piece, which is stationary while the voice coil moves. Push the cone up and down by hand to see this motion.

Next, with a knife or scissors, cut completely around the cone's surround, the outermost edge that connects the cone to the basket. This may be made of cloth, foam, rubber, or the cone material itself. Finally, cut completely around the spider, the corrugated cloth ring that holds the narrow end of the cone to the magnet structure. The cone and voice coil should then come straight out as one assembly.

With the cone's suspension removed, if you connect a 1.5V battery to the voice-coil leads with the correct polarity, it will jump right out of the basket. Trying to push the voice coil back into the motor with the battery still connected will give you an idea of the amount of force that is involved in making sounds.

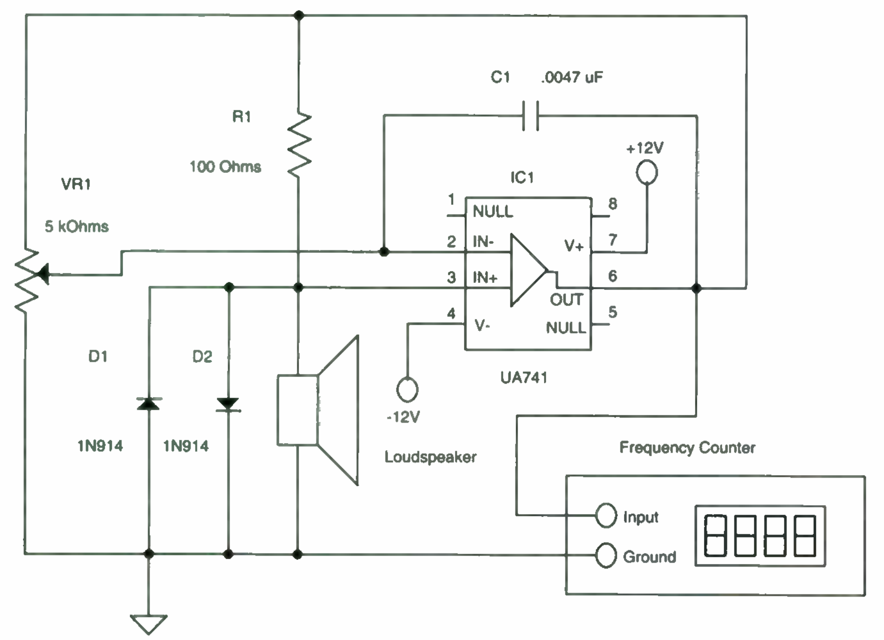

FIGURE 4: Bridge circuit connected to oscillate at resonance frequency.

	A	B	C	D
1	1	0		
2	=A1+B1/10	=B1-A1/10		
3	=A2+B2/10	=B2-A2/10		
4	=A3+B3/10	=B3-A3/10		
5	=A4+B4/10	=B4-A4/10		
6	=A5+B5/10	=B5-A5/10		
7	=A6+B6/10	=B6-A6/10		
8	=A7+B7/10	=B7-A7/10		
9	=A8+B8/10	=B8-A8/10		
10	=A9+B9/10	=B9-A9/10		
11	=A10+B10/10	=B10-A10/10		
12	=A11+B11/10	=B11-A11/10		
13	=A12+B12/10	=B12-A12/10		
14	=A13+B13/10	=B13-A13/10		
15	=A14+B14/10	=B14-A14/10		
16	=A15+B15/10	=B15-A15/10		
17	=A16+B16/10	=B16-A16/10		
18	=A17+B17/10	=B17-A17/10		
19	=A18+B18/10	=B18-A18/10		
20	=A19+B19/10	=B19-A19/10		
21	=A20+B20/10	=B20-A20/10		
22	=A21+B21/10	=B21-A21/10		

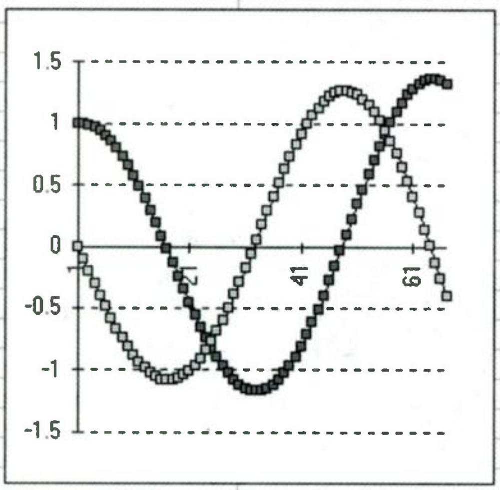

FIGURE 5: Spreadsheet showing how sine and cosine values emerge from simple mathematics.

A BRIDGE TO THE PAST

What if you don't have an oscilloscope to check for zero phase? *Figure 3* shows how to add a potentiometer to the circuit of *Fig. 1*, making it into a bridge circuit capable of precisely detecting zero-phase frequencies. Instead of looking for peaks, iteratively adjust input frequency and potentiometer VR1 to find a sharp null on the AC voltmeter (ACVM).

When the bridge is in balance, you can remove the potentiometer from the circuit and measure the resistance from each end to the wiper. The ratio of these values will be the same as the ratio between R1 and the loudspeaker's impedance at the balance frequency: R_{vr1}-hi/R_{vr1}-lo = R1/Z_{spkr}. Solving for Z_{spkr}, you have Z_{spkr} = R_{vr1}-lo × R1/R_{vr1}-hi.

The potentiometer is shown as 5k, but the value is not critical. I chose it to be low with respect to the ACVM's input resistance, and to be greater than the sine-wave generator's output impedance. You can use a multiturn potentiometer for finer resolution.

A BRIDGE TO THE FUTURE

But what about the loudspeaker as oscillator? The Nyquist criterion for sustained oscillation is that the loop gain must be greater than or equal to 1, and the feedback signal must be in phase with the input signal at the frequency of oscillation.[6] *Figure 4* shows how to add an operational amplifier to the bridge circuit of *Fig. 3* to make it self-oscillating at the speaker's resonant frequency.[7]

I added C1 to suppress ultrasonic oscillation, and D1 and D2 to protect the op-amp input. With this circuit, you can measure loudspeaker resonant frequency without using an ACVM *or* a sine-wave generator. All you need is a frequency counter, and many digital voltmeters (DVMs) now have a counter built in.

Adjust the potentiometer VR1 so that oscillation is sustained at a reasonable level, below clipping. At this point the bridge is in

REFERENCES

1. Alexis Badmaieff and Don Davis, *How to Build Speaker Enclosures*, Howard W. Sams & Co., 1970.

2. Ray Alden, *Advanced Speaker Systems*, Master Publishing, 1995.

3. Richard H. Small, "Vented-Box Loudspeaker Systems, Part II: Large-Signal Analysis," *JAES*, July/August, 1973.

4. Hermann Helmholtz, *On the Sensations of Tone*, Dover Publications, 1954.

5. W.J.J. Hoge, "Measuring Loudspeaker Driver Parameters," *dB Magazine*, August, 1981.

6. Joseph J. Carr, *Secrets of RF Circuit Design*, McGraw-Hill, 1991.

7. Alexander Belousov, "Negatron Yields Real Natural Frequency," *EDN Magazine*, August 19, 1993.

balance, and, as before, you can remove the potentiometer from the circuit and measure to calculate the speaker's impedance at resonance. If you aren't interested in measuring impedance, replace the potentiometer with an automatic-gain-control (AGC) circuit to stabilize output level.

WHY A SINE WAVE?

Now why should an oscillator produce a sine wave, instead of some simpler waveform, such as a sawtooth or square wave? This is a complex question. Answers begin with the Pythagorean Theorem and continue through to the basic nature of time and our physical universe. I put my mind at ease by playing with the spreadsheet of *Fig. 5*, in which sine and cosine results emerge from simple linear math operations.

Assuming you have a spreadsheet similar to Works or Excel, put the value 1 in cell A1. In cell B1, put the value 0. These are the initial conditions. In cell A2, put the formula = A1 + B1/10. In cell B2 put the formula = B1 – A1/10. Now fill down cells A2 and B2 for about a hundred rows. Specify relative coordinates if your spreadsheet requires this.

Calculated values in the two columns will vary from around –1 to +1. Plot columns A and B on a line graph, and you will see sine and cosine curves, as in *Fig. 5*. Columns A and B represent displacement and velocity of the loudspeaker cone, which vary in quadrature relationship (that is, with a 90° phase difference) to each other over time (represented by the horizontal axis of the chart).

Each state of A and B, or displacement and velocity, depends only on their previous state. If your spreadsheet can plot X against Y, you will see a circle—or actually a spiral, unless you apply some AGC. I leave this as an exercise for the interested reader.

You may be wondering, as I did, what would happen if you connected a reflex-type cabinet to the circuit of *Fig. 4*. I tried it and found that I could force it to oscillate at either of the two maximums, but that it much preferred the one with higher impedance.

COVERING WINTER CES

By C. Victor Campos

PHOTO 1: NAD Electronics' 118 digital preamplifier.

For a CES, the 1997 Winter Consumer Electronics Show was somewhat unusual. Realizing that business in our area of the industry decreased by more than 30% in 1996, exhibitors and attendees were perhaps fearful and, at the same time, expectant. Despite some exceptions—and contrary to promises made during the 1996 Winter CES—the market over all has suffered substantial decreases in revenue.

In some respects, dealers must accept part of the blame for this business climate. Many large dealers, or those who purchase through buying combines, have pretty much demanded from manufacturers the type of products they want and dictated the direction which the market will take. Although these demands might be feasible for big companies, they present, in some instances, an impossible set of timing and circumstances for the smaller specialty manufacturers who are the bread and butter of profit for the smaller specialty dealers.

WAIT TILL NEXT YEAR

All in all, the 1997 WCES failed to unveil any ground-breaking technologies or radical improvements to existing products. The entire tone at the show could be summed up by the first major press conference on the first day. Robert Minkhorst, Philips Consumer Electronics Company President and CEO, announced new Philips headquarters in Texas, and manufacturing and design engineering facilities in Fremont, CA. The mandate for these new facilities is to design and manufacture (in the US) products that are targeted and desirable to the American public: flat TV screens (presumably plasma, although this was not discussed directly), HDTV, Internet TV for Web use, LCD projectors (with "cutting-edge" technology), and DVD, which, according to Minkhorst, is the most important development since the compact disc.

All these products are scheduled for the end of 1997 or beginning of 1998, creating the impression that these announcements are essentially meant to preclude sales of similar devices this year and, perhaps, to "save" the market for these forthcoming Philips products. In addition, plans call for both digital and analog cordless phones, digital answering machines with and without caller ID, and many other products.

Delivery of DVD is slated for sometime in the spring, which was the same promise made last year. When a reporter from *Variety* asked Minkhorst what software would be available for DVD, Minkhorst needed to consult with one of his colleagues. He replied that about seven or eight titles—with unspecified contents—would be available from PolyGram, which is one of the companies Philips controls. It was also unclear whether any major movies or cinema presentations would be available this year.

Perhaps underlining the experience at Philips was Sony's press conference, featuring a "Presentation on Sony's Total Bold Approach to DVD." Sony could have used last year's CES presentation. Not much more information than that reported in 1996 was given except that the price of DVD has now increased to around $1,000, instead of the original projected price of $600.

By the way, Sony's display, which was located in the North Hall of the Las Vegas Convention Center, must have set back the company by close to $1 million. A neighboring exhibitor commented that the display lighting alone must have cost around $100,000.

HOW BIG?

I should note that the 1997 WCES was so spread out that it would be impossible for any individual, or individuals, short of a group, to cover all the exhibits in a four-day period. The show occupied the entire facilities of the Las Vegas Convention Center and the Hilton Ballroom and peripheral spaces, the majority of the Alexis Park Resort Hotel, and a very large portion of the Sands Convention Center. Although the final attendance figure was around 96,000, it was difficult to visually gauge attendance. Also, no matter how accurate the official tally, it can't control duplicate registrations from press people, public relations personnel, and similar attendees, who represent more than one company.

CES is essentially an opportunity for dealers, buyers, and foreign distributors to preview upcoming products. Generally speaking, foreign distributors are very much in attendance at CES, but it is unclear how many dealers and buyers are really present at these shows. On the other hand, many exhibitors seemed well satisfied with the attendance and the effect their displays had on their business.

FLAT LOUDSPEAKERS

Actually, most of the show did not reveal any new products meriting particular attention or comment (press releases notwithstanding). Most products, particularly those

PHOTO 2: An array of Acoustic Research loudspeakers.

in the Specialty Audio area, were really old bodies in new clothes.

Nonetheless, a few items merit noting, either because of their promise as a product, their potential for new product technology, or a significant step forward in the technology. Among these was a new flat-panel loudspeaker technology from NXT, a part of the Verity Group PLC. The Verity Group developed this technology, which was derived from work by the British Defense Ministry. These flat loudspeakers can be fabricated in almost any size and shape, with thicknesses as small as 1/8", and can be enlarged to video-screen size.

The secret of this technology is the use of a rigid panel driven by one or more tiny exciters (NXT's word), which are equivalent to transducers. The size of the panel determines the number of exciter elements required. According to NXT, this technology differs from conventional flat-panel speakers in that its transducer has complex vibration modes (mode distribution) over its entire surface instead of the usual single-point source, which travels in waves to the edges of the panel. NXT claims that these panels are responsive between frequencies of 120Hz and 20kHz or more.

The high-fidelity performance of these panels failed to impress, but the technique showed definite promise for commercial applications in either public-address systems in large enclosures or in subway stations, bus terminals, bus stops, and the like. The use of those devices in high-fidelity, home-theater, and multimedia and computer applications left the listener somewhat less than satisfied. Nevertheless, the format has potential for much further development. The attractive point of this technology is that it does not need an enclosure as we know it, but can be freestanding. Since it operates as a bipole, its positioning in the room—against a wall or in a wall—is not particularly detrimental to the function of the device.

NEW FROM NAD

NAD Electronics Ltd. really stepped forward with two digital products that are bound, at the very least, to pave the way for future products. The first one, the NAD 2.2, is a digital room-correction system that seems to complete the work begun by Robert Berkovitz at Acoustic Research in 1979 and continued by Kevin Voecks at Snell Acoustics in the early '90s. The resulting product has almost all the elusive parameters defined.

In a closed demonstration, in what might kindly be called a horrible room at the Hilton, Peter Lyngdorf (majority owner of NAD) and Dr. Radomer Bozovic (the final designer of the hardware and software for the 2.2) kindly demonstrated the device to me. The NAD 2.2 did an outstanding job (second to none in my experience) of minimizing the room resonance modes in the bottom end and restoring a surprising naturalness to the midrange. Also, localization, depth, and width of the stage were enhanced considerably.

The improvement the NAD 2.2 effected was greater than the improvement you would gain from replacing your loudspeakers with others costing ten to twenty times as much; even then, the original room problems would not be addressed. Although this item is supposedly in production, it remains to be seen if the problems of manufacturing such a complex and sensitive device can be resolved for speedy delivery to the marketplace. The 2.2 is aimed at the professional market, for use in studios and mixing rooms.

NAD also showed the 118 digital preamplifier (*Photo 1*), which is a concrete step toward answering all the problems which have plagued analog preamplification in the past. Based on patents held by Mike Gerzon (the late, and famous, British mathematician and researcher), this preamp was executed by the well-known Dr. Geoffrey Barton in conjunction with Bjørn-Erik Edvardsen of NAD 3020 fame. This preamp not only includes digital-to-analog conversion of CDs, DATs, and mini-disks, but also has full 18-bit analog-to-digital systems to convert analog input signals to digital format so they can be processed within the circuitry of the 118.

Perhaps its most salient feature is the capability of processing signals, including boosts and cuts of bass, midrange and treble, as well as filtering, without the inevitable phase-shift problems and group-delay consequences of such processing. Along with its compression and expansion modes, it synthesizes FM stereo from mono signals and reduces noise and distortion from FM stereo signals (always the result of weak RF signals). Its performance appears to be a clear step forward in the digital development of audio.

AR IS BACK

While walking through the North Hall of the Las Vegas Convention Center, I was pleasantly surprised by the Acoustic Research display. AR, which for the past couple of decades seemed to have lost its way, has made a resurgence that can only be described as remarkable. Embracing, once again, some of the original philosophies of the company, its new products (*Photo 2*) seem to have the commitment to quality and performance that has been absent for so long.

This is not to say that all the philosophies have reemerged. For example, the all-acoustic-suspension design has been supplanted in some products by other methods such as vented woofers. But the quality of execution in manufacture is, at least from the products shown at CES, impeccable. It is quite evident that much thought and research went into the design of these products.

The same can be said about the electronics that power its home-theater systems and subwoofers: quality of execution is impeccable. The company has obviously made a very serious attempt to marry the old philosophy of AR quality and research with modern production methods, materials, and quality control.

The company plans to expand its manufacturing and R&D facilities in Northern California this spring from its present location in Chatsworth. Although it is difficult to make final judgments from this initial showing, AR's progress bears close observation. Cary Christie, who is the new CEO, and ostensibly one of its principal designers, is one of the original founders (along with Arnie Nudell and John Ulrich) of Infinity Loudspeakers. As a result, he brings to the position a vast amount of experience in the design and manufacture of loudspeaker systems. Also, a not inconsequential by-product of its new ownership by Recoton Corporation, is the infusion of capital necessary to the fulfillment of AR's goals.

HAVING A BALL

At the Alexis Park, ADCOM demonstrated a remote-control system (*Photo 3*) that simplifies and integrates operation for even extremely complex home entertainment systems, including two-channel, home-theater, and audio-video systems. Its hand-held remote control, operating in the 900MHz band, consists simply of four controls: volume up, volume down, enter, and a mouse-like ball (which provides the name for the system known as "The Ball") to move an indicator on the video screen. The system is extremely easy to operate and the user,

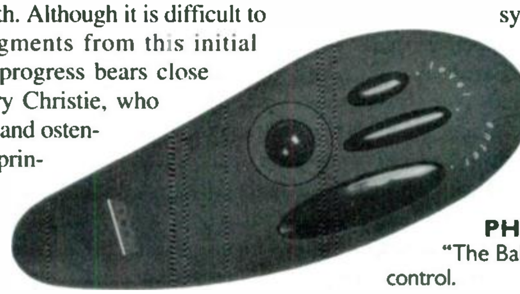

PHOTO 3: ADCOM's "The Ball" hand-held remote control.

the dealer, or the custom installer can configure it to almost any system, since it includes facilities for cable, DSS, broadcast signals, and the gamut of common source equipment.

The mouse-like ball moves a cursor over the custom-configured master control panel, which is shown on the system's main screen. Centering the pointer over an icon on the screen and simply pushing the "enter button" activates the proper response in the system. The ball actually controls all the functions in any system, no matter how complicated, and permits addressing all remotely controlled components without compromising, or in any way changing, their individual remote-control transmitters or functions. The Ball even controls the lighting in a room, can raise and lower projection screens and projectors, and even has a similarly controlled second zone. Unfortunately, the system has so many features that it is nearly impossible to describe its complete operation here.

I had perhaps the most fun with the minute, easily-carried-in-a-shirt-pocket Motorola StarTAC cellular telephone. This appears to be the smallest and most easily usable cellular telephone, with a full complement of features. Through the courtesy of Motorola's Cellular Subscriber Group, I used one, which served me well indeed, for three days during the show. While it includes many features and conveniences, unfortunately, the price is daunting. However, I trust that, like all new technologies, the price will drop substantially in the future.

OUTBOARD SHOW

I encountered a rather unexpected event as I checked into the Debbie Reynolds Hotel on Convention Center Drive: an unpublicized outboard show, the first annual High End Show (HES), sponsored and managed by Maverick Audio Systems. The exhibitors, who were not official participants of the Winter Consumer Electronics Show, were part of an association known as the International High End Society, based in San Diego, CA, and were obviously manufacturers who had been dissatisfied in one way or another with CES's performance in the past.

Because the show had not been widely or properly publicized, the attendance was quite small, consisting mainly of foreign distributors who may have been advised of the Debbie Reynolds' exhibits by word of mouth. Most of the exhibitors expressed satisfaction with the business they transacted at the High End Show, and I heard very few disgruntled comments. However, as with the main exhibits at the Alexis Park and CES, there were no really earth-shaking discoveries or demonstrations.

In total, it seems as if the Winter CES is evolving into a completely different kind of show. Whereas those of the 1980s more than adequately served the audio industry, the '90's shows seem to have shifted the focus to technologies other than audio, surround sound, and home theater. I suppose the reins have been yielded to other technologies and industries.

RESOURCES

Philips Electronics
2099 Gateway Pl., Ste. 100
San Jose, CA 95110

Sony
1 Sony Dr.
Park Ridge, NJ 07656

Verity Group PLC
Stone Hill, Huntingdon
PE 18, 6ED, England

Acoustic Research
9424 Eton Ave., Ste. J
Chatsworth, CA 91311

ADCOM
11 Elkins Rd.
E. Brunswick, NJ 08816

Motorola
1303 E. Algonquin Rd.
Schaumburg, IL 60196

NAD Electronics Ltd.
89 Doug Brown Way
Holliston, MA 01746

seas

Audiophile Price List April 1997

MADISOUND SPEAKER COMPONENTS
8608 UNIVERSITY GREEN
P.O. BOX 44283
MADISON, WI 53744-4283 U.S.A.
TEL: 608-831-3433 FAX: 608-831-3771
e-mail: madisound@itis.com
Web Page: http://www.itis.com/madisound

Model	Size Description	Ω	Fs Hz	Qts	Vas Liters	X-max mm Peak	Power Watts	Sens. 1W/1M db	Cost Per Unit
Tweeters - *We have replacement voice coils for most Seas tweeters.*									
25TAFN/QG (H623)*	25mm alum. dome tweeter, neodymium magnet, grill	6	1800	*Audax wedge mounts fit this tweeter. Shielded.			100	89	$22.40
19TFF (H586)	19mm textile dome tweeter	8	1700				80	87	$18.50
20TFF (H830)	20mm coated textile dome tweeter	8	1500				80	89	$17.80
25TFFC (H519)	25mm textile dome tweeter w/chamber	6	1200				80	90	$22.80
25TAF/G (H398)	25mm aluminum dome tweeter with grill	6	1400				100	90	$21.90
25TAC/G (H400)	25mm alum. dome tweeter with chamber & grill	6	660				55	91	$25.20
27TFF (H831)	27mm coated textile dome tweeter	6	1200				90	92	$21.00
27TFFC (H881)	27mm coated textile dome tweeter w/chamber	6	900				80	91	$24.20
27TAF/G (H882)	27mm alum. alloy dome tweeter with grill	6	1200				100	91	$23.10
27TAFC/G (H883)	27mm alum. alloy dome tweeter w/chamber, grill	6	900				90	90	$27.50
T25 001 *Excel* (E006)	25mm Sonotex fabric dome tweeter with chamber	6	750				90	90	$64.00
Midranges									
MCA11RC (H143)	4.5" treated paper cone midrange	8	140	.72	1.3	.9	110	89	$35.70
MP12VC (H453)	5" poly cone, PVC surround, glass fibre plastic frame	8	90	.44	2.9	.9	90	89.5	$31.50
MP14RCY (H422)	5" poly cone midrange	8	42	.20	17.2	3	110	90	$38.60
MP14RCY/P (H522)	5" poly cone with phase plug	8	43	.18	16.6	3	120	90	$38.20
K2852 and K2851	Chamber for 11cm drivers: $2.00	Chamber for 14cm drivers: $3.00							
Woofers - *All woofers feature injection moulded magnesium frames and rubber surrounds.*									
CA11RCY (H149)	4.5" treated paper cone woofer	8	58	.24	5.4	3	60	86	$39.00
L11RC/P (H759)	4.5" aluminum cone woofer	8	53	.30	4.6	3	70	85	$45.70
P11RC (H454)	4.5" poly cone woofer	8	55	.34	5.3	3	60	84.5	$32.90
P11 RCY (H455)	4.5" poly cone woofer, large magnet	8	55	.23	5.3	3	60	86	$37.10
W11CY 001 *Excel* (E001)	4.5" magnesium cone woofer with copper phase plug	8	65	.34	3.7	3	75	86	$111.00
P14RC (H395)	5" poly cone woofer	8	40	.28	18.9	3	60	89	$33.40
P14RC/TV (H626)	Shielded Magnet 5" poly cone woofer	8	40	.21	18.9	3	60	89.5	$42.20
L14RC/P (H761)	5" aluminum cone woofer, phase plug	8	39	.31	14	4	80	85.5	$50.90
W14CY 001 *Excel* (E008)	5" magnesium cone woofer with copper phase plug	8	43	.36	10.6	4	70	87	$122.00
P17REX (H416)	6.5" poly cone woofer with large voice coil	8	34	.24	30.5	3	80	89	$46.50
P17RE (H419)	6.5" poly cone woofer	8	34	.33	30.5	3	80	87.5	$42.70
P17RE/TV (H690)	Shielded Magnet 6.5" poly cone woofer	8	34	.27	30.5	3	80	88.5	$52.50
P17RC (H353)	6.5" poly cone woofer for sealed enclosures.	8	35	.32	40.8	3	60	89	$35.20
L17RCY/P (H763)	6.5" aluminum cone woofer with phase plug	8	32	.27	36	4	80	87.5	$58.80
T17RE (H823)	6.5" Clear PP/TPX cone woofer	8	34	.32	28	3	80	87	$49.40
W17E 002 *Excel* (E009)	6.5" magnesium cone woofer with copper phase plug	8	34	.32	29.5	4	100	87	$138.00
CA21REX (H333)	8" Paper cone woofer (used in Seas NJORD kit)	8	31	.34	81.3	3	80	93	$52.70
P21REX (H282)	8" poly cone woofer with 1.5" VC	8	33	.37	68.9	3	80	91	$53.30
P21RF/P (H511)	As above with phase plug and 2" diameter voice coil	8	34	.34	48.3	4	125	88	$57.30
P21RE4X/DC (H442)	Dual VC 8" woofer with poly cone	8/8	31	.30	66.4	3	90	90	$61.50
P25REX (H283)	10" poly cone woofer	8	27	.44	156.8	3	80	93	$58.90
25F-EW (H085)	10" paper cone **Repl. for Dynaco A25 Speaker**	8	26	.35	175	4	70	89	$47.70
CA25RE4X/DC (H372)	10" treated paper cone woofer	8/8	25	.31	187.9	4	90	91	$61.75
Coincidental Coaxial Driver - *The tweeter is mounted at the base of the woofer cone.*									
T17REX COAX/F (H723)	6.5" coaxial T17RE woofer / 25TFFN/G tweeter in a coincidental configuration. (Clear cone woofer)	T6 W8	T1.8K W38	W .29	W 20.8	W 3	T90 W80	T89 W87.5	$76.70
T17RE COAX/TVF (H825)	Same as above, but with a bucking magnet and shielding cup for use near CRT's	T6 W8	T1.8K W38	W .31	W 20.8	W 3	T90 W90	T89 W87	$86.40
P17REX COAX/F (H489)	6.5" coaxial P17REX woofer / 25TFFN/G tweeter in a coincidental configuration.	T6 W8	T1.8K W35	W .25	W 26.9	W 3	T90 W100	T89 W89	$71.20
P17RE COAX/TVF (Shielded) (H653)	Same as above, but with a bucking magnet and shielding cup for use near CRT's	T6 W8	T1.8K W35	W .31	W 26.9	W 3	T90 W100	T89 W89	$80.30

Product Review

AUDIOCONTROL INDUSTRIAL SA-3050A SPECTRUM ANALYZER

Reviewed by Publio Morera

AudioControl Industrial SA-3050A Spectrum Analyzer, *$995–$1300, 22410 70th Ave. W, Mountlake Terrace, WA 98043, (206) 775-8461, FAX (206) 778-3166.*

PHOTO 1: AudioControl's Model SA-3050A third-octave real-time spectrum analyzer.

Speaker builders can be divided into two basic varieties: those who yearn to measure and those who don't. When economics or other reasons force the measuring-inclined into the latter group, individuals subscribing to the "quantifying musicality" measuring ideology can become very frustrated. My reasons for pledging allegiance to the second active group were many, including gnawing questions that eventually launched me on a search for an affordable measurement system.

How could I easily measure what I heard and liked and that which I didn't like? Even more worrisome, how could I measure what I could not hear and did not know whether I was supposed to like? (Non-measuring types need not understand this powerful drive.) I was also very worried that the relative goodness of my speaker creations could be unmasked by some highly sensitive apparatus and exposed as "not so good."

In any case, I will not discuss my neurosis any further. Once I moth-balled my old Radio Shack sound-level meter and sine-wave generator, I was on my way. But where?

AUDIOCONTROL TO THE RESCUE

The model SA-3050A one-third-octave real-time spectrum analyzer is enclosed in a sturdy gray-and-black metal case with a carrying handle (*Photo 1*). It includes an internally calibrated microphone and a battery-pack option for portability. The instrument features a series of bandpass frequency-dependent filters and associated displays, with a resident pink-noise generator.

The relative energy in each one-third-octave frequency band is displayed on 30 vertical rows of LEDs on the front panel. The number of illuminated LEDs increases vertically in each row in proportion to the energy received by the microphone for each band of frequencies. The LED display forms a frequency-versus-amplitude curve, which constantly updates itself in relation to changes in the device being tested or the acoustical environment (in real time). You can conveniently adjust the analyzer's gain and display decay time (fast/slow).

Built-in capabilities include six available memories for response-curve storage, sound-level-meter function, peak hold mode, and averaging for stored curves. Obtain several readings at various angles to emulate a speaker's composite response into the listening room, somewhat comparable to the power response. A print option is available at extra cost for those requiring a hard copy of their measurements.

NO MORE GRAPHS

Judging from the material contained in the comprehensive owner's manual, the 3050A appears to be primarily intended to help in setting up sound-reinforcement and stage sound systems and general evaluation of acoustical environments. I use this device mostly for home-audio and loudspeaker-measurement applications. When making the transition from an AC voltmeter and a microphone to a real-time measurement system such as the AudioControl analyzer, the biggest adjustment you'll face is the variety of often confusing results that you can obtain.

A slight movement of the microphone can cause radical shifts in the displayed frequency spectrum of the device under test. As you might expect, some discipline is required to obtain repeatable and meaningful results during acoustical measurements. I recommend a microphone holder (not available with the 3050A) to maintain proper microphone placement during loudspeaker measurement sessions. The pink-noise stimulus generated by the RTA alleviates some of the measurement error caused by reflections, for example, but judicious use of absorbent material or outdoor measurements can improve the measurement accuracy.

The memory/storage feature is very helpful for comparing curves, as when you evaluate various speaker or listening locations. Tweeter horizontal dispersion assessment becomes easy by simply rotating the speaker in increments on a stand, while watching the display with the microphone located a few feet away. In this way you can also fairly assess a speaker's polar pattern or detect undesirable lobes.

POOR MAN'S FFT ANALYZER

To anyone who has seen or heard the apparently unstable swishing output of pink-noise sources, the RTA display monitoring such noise may seem like a wild dance of frantic red lights. However, with a little practice and patience, you can obtain response curves in a fraction of the time it would take to manually graph the readings from a microphone, voltmeter, and warble/sine-wave generator

ABOUT THE AUTHOR

Publio Morera is a "quasi"-speaker builder; he gleefully drags home former store-bought speakers purchased at garage sales, for example. He then dissects these neglected and under-damped orphans in a highly secret ceremony akin to Frankenstein's. He fashions the speakers into creations of new and perhaps even somewhat improved sonic character, often using only the original enclosure.

(*Fig. 1*). Although the 3050A lacks the fine resolution and the ability to show waterfall plots, impulse response curves, and the data manipulation capability of PC measurement setups, it does a good job of pointing out response anomalies that are most responsible for loudspeaker sound colorations.

If you're not inclined to spend time calculating crossover component values, you may choose an easier and more entertaining path for crossover development: first check the acoustical response of a driver mounted on the baffle or enclosure and then set up a circuit on a breadboard to determine the proper component values by trial and error. You should first test individual drivers without crossovers, then with crossovers installed, followed by all drivers connected to the crossover.

The attenuation slopes and curves will reflect the combination of acoustical driver characteristics and electrical crossover component behavior. Any peaks, dips, or other frequency/amplitude anomalies, caused by driver location or by reversing connections, will also become apparent on the display. The analyzer permits fast training by allowing you to relate the sound of a particular driver/component combination with the frequency-amplitude curve shown on the display.

You may even be disappointed to find that the 3050A will show variations in frequency energy contents that are apparently inaudible. You must remember that acoustical phenomena, such as masking, are present, and that one-third-octave filtering is not quite the equivalent of our own hearing apparatus. The 3050A will probably not detect problems caused by obstructive grille frames or "acoustically bizarre" grille cloth.

It can't tell you which is the better stuffing material or crossover capacitor to use, or directly measure enclosure boominess, for example. It can, however, allow for a rapid evaluation of room treatments, equalizer settings, and basic loudspeaker measurements. Regardless of your speaker-design preferences and level of knowledge—or to which of the above groups you subscribe—it is very important to remember that, "If it does not sound good, it probably *really* doesn't sound good." It then becomes irrelevant how it measures, doesn't it?

Manufacturer's response:

Thanks for the opportunity to examine Mr. Morera's review of the SA-3050A.

It is reviewed using a microphone for speaker analysis. While it may not apply here, one of the overlooked uses of the product is to measure line-level signals as well. That is, you can measure directly the output of an electronic crossover, CD player, etc., as well as acoustic signals. Measuring line-level signals is very useful for troubleshooting a system's components, as well as measuring frequency response. Also, you can measure the output of a passive crossover as shown in the manual.

Tom Walker
AudioControl

PRODUCT FEATURES AT A GLANCE

- 30 one-third-octave bandwidth filters.
- Fourth-order filters conform to ANSI S1.11-1971 Class II, type E standards.
- Internal pink-noise source.
- 9 × 30 large-format LED display with 1dB resolution.
- Six internal memories with battery backup.
- Peak hold function.
- SPL scale with bargraph and full-screen digital readout.
- 92dB display range.
- Signal input from balanced microphone input, balanced phone jack or unbalanced BNC connector.
- Battery operation (optional).
- Printer port (optional).
- A and C weighting (optional).
- Cost is between $995–1,300.

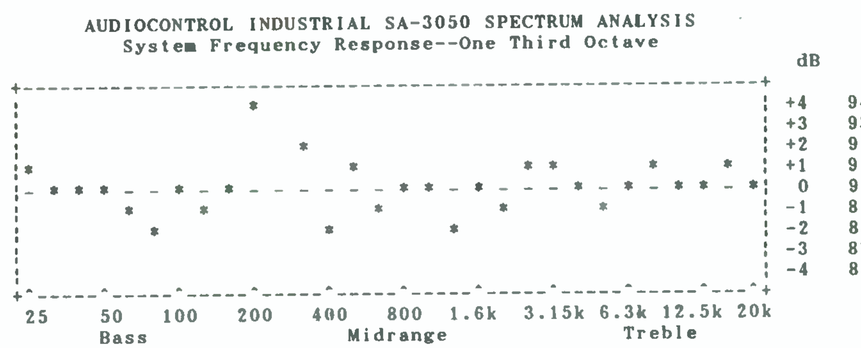

FIGURE 1: Sample printout from the SA-3050A.

Product Review

SPEAKER-KNIFE

Reviewed by Fernando Garcia

***Speaker-Knife:** electronic speaker-protection module; Dynastar Electronics, 1531 E. Starpass Ave., Fresno, CA 93720; MSRP $49.99.*

Anyone who is deeply involved with sound reinforcement knows the hazards of the trade: blown amplifiers, blown speakers, or, worse, a blown amplifier damaging a speaker. Wiring mishaps occur with surprising regularity, reminding us that Murphy's Law is alive and kicking. I became interested in this problem primarily to protect the power amplifier, and wrote an article (*TAA* 4/90) describing a protective device for power amplifiers. Although amplifiers have become very reliable, whenever they fail they usually pump a large amount of DC current (in the case of transistor amplifiers) or hum (in the case of tube amps), which can fry your speakers in fractions of a second.

Amplifier failures are by no means the only way to destroy your speaker and no doubt many audiophiles have experienced such sad occurrences. Even if you discount the expense of replacing a driver, the sound balance usually suffers. For lack of a better comparison, it is analogous to losing a single shoe and attempting to complete the pair with a replacement. You'll always feel the difference between the new and the worn shoe. The pair simply doesn't match or feel right!

And if you own vintage or discontinued speakers, blowing up a driver could mean discarding the speaker pair—a nightmare audiophiles may choose to avoid.

PROTECTION SELECTION

The Speaker-Knife seeks to protect the receiving end of the business, the speaker itself. It basically removes the drive from the speaker whenever the voltage exceeds a certain level. It comes in a small (2¼″ × 1 7/8″ × 1 3/8″) epoxy-filled plastic cube, with three color-coded wires protruding from it.

Protection is accomplished in two stages, clamping and removal. Whenever the voltage being fed to the speakers exceeds a certain threshold, a solid-state clamp (thyristor) quenches the voltage by clamping it hard to ground. If the overload condition remains longer than a few milliseconds, then the normally closed relay kicks in, opening the amplifier path. The clamping action is quite fast, less than a half microsecond, and can draw substantial current levels. Needless to say, your amplifier must have an output or DC-bus fuse.

Before you rush out to buy a Speaker-Knife, you must first specify the device's rating. This is not a straightforward matter; unfortunately, the enclosed literature, in an effort to be all-encompassing, makes the selection a little more confusing. The *SKs* are rated in peak power, so you must first convert the speaker rating to peak-power rating. The literature contains a conversion chart to do this, and it lists many ratings, some of which I was unfamiliar with: music power, program power, continuous RMS power, continuous program power, long-term power, continuous sine power, RMS power, sine power, watts, peak power, pink noise, short-term power, and continuous power.

Then you must figure the impedance of the speaker and select an *SK* model from a table of 4, 8, and 16Ω impedances. The trouble is, with so many speakers employing different impedances, you may be confused as to which impedance to select. In my case, should I choose a 4Ω or an 8Ω device for my 6Ω speakers?

Again, this is not the *SK* manufacturer's fault, as the speaker vendors are quite inconsistent in rating impedances and power capabilities. This is especially true in the automotive audio business, which makes all sorts of ludicrous claims. When selecting a device, remember that the Speaker Knife triggers on a voltage threshold. The power rating will therefore be a function of the speaker impedance.

INSTALLATION

Once you have overcome the selection procedure, installation is a snap. The black wire goes to the speaker and amplifier's common, the red wire splices to the wire from the amp, and the orange wire connects into the speaker itself. Mounting and wiring hardware is included, although the supplied splice was too small for the speaker-wire gauge I employ for my setup (#16AWG). Audiophile speaker cables would be hard to splice.

The instruction sheet suggests that you mount the device on the speaker cabinet and place the sensitive relay inside, but warns you to use the supplied rubber grommets to minimize nuisance tripping due to vibration. This piqued my curiosity, so I mounted the device without grommets on the subwoofer cabinet in an attempt to make it trip. After the device failed to do so, I then tapped it lightly and then much harder, again failing to make it trip.

Although my review sample was quite "shock-resistant," I would heed the advice and use the grommets. In addition, the manual warns against mounting the unit too close to the speaker magnets, as this could impair the internal relay's response speed.

My review sample had a small power rating (30W into 8Ω), allowing me to test it without worrying about exceeding my speaker's ratings. When the threshold cutout was exceeded, the protective relay kicked in. During loud music, a cutout would occur during heavy percussive lines. Likewise, any crescendos would be cut out, and the music would be restored a fraction of a second later.

Clearly, if you use a Speaker-Knife, it should have a rating higher than your system's peak listening levels. Otherwise, the peaks will be muted out, which is certainly an annoying effect.

The insertion loss of the device was negligible; my review sample had a resistance of 0.016Ω. I performed blind testing to figure out whether I could detect the presence of any coloring or artifacts during normal operation. I heard none. While I don't claim to have golden ears, the audible effect should be negligible for a reasonably powered amplifier, which can sustain the loading imposed by the quiescent current the device requires.

The current draw from the amplifier was around 50mA during normal (non-tripped) mode, surging to about 200mA in the tripped mode, indicating that the relay has been activated. Response speed was less than half a microsecond for the clamp to trip, and about 7mS for the relay to activate. I tested this by dumping the charge of a 50V, 4700µF capacitor into the module and monitoring the results with a Tektronix TDS420 storage scope.

CONCLUSION

Is the Speaker-Knife for you? It depends on your application. For example, low-power vacuum-tube amps that will never produce a DC offset should not require it. On the other hand, higher-powered automotive equipment could benefit from it. Sound reinforcement, public address, and other mishap-prone systems would also welcome the unit.

However, I doubt that any serious audiophile who has spent a substantial amount of money on speaker cables would consider degrading the performance by inserting the Speaker-Knife in the signal path. And while most amplifiers should be immune to the momentary shorting of the internal clamping device, others may become unstable.

One aspect of the Speaker-Knife concerned me. The spec sheet states, "Do not attempt to send more than 46V continuous sine wave through any module. It will damage the module and may cause it to explode at extremely higher levels." This is an extremely large and unlikely overload.

However, as a power-supply designer, I understand safety aspects and the need to protect the user from hazards during use. As a matter of fact, regulatory bodies such as UL, CSA, and VDE assume that fools are actually very clever when it comes to creating hazards. Thus, the Speaker-Knife should have had an internal fuse, fusible link, or trace to protect the user.

Manufacturer's response:

I would like to thank Mr. Garcia for taking the time to review the Speaker-Knife. However, there are a few statements that need to be corrected, without intending offense to Mr. Garcia.

The review states that the SK clamps the audio voltage "hard to ground." This is not the case. The SK clamps the audio through a low resistance, quickly reducing the gain to the driver and buffering the power amp from damage.

As quoted, "your amplifier must have an output or DC-bus fuse" is simply not the case; this is one of the main reasons the Speaker-Knife exists. Many amplifiers do not have this, except for the primary AC lines fuse. We have had no "fuse blown" complaints from the field.

Dynastar recommends that SK be mounted on the inside of a speaker cabinet as opposed to the outside, for aesthetic and physical protection reasons only. It's the installer's choice.

The main concern with Mr. Garcia's review is the idle current in which he claims measuring 50mA. While we're not sure how he arrived at that figure, our lab tests demonstrate that in the worst case (capacitance and all) at 20kHz RMS sine, the module idles at about 800µA or less just before triggering. Another example shows about 0.001% of power taken from a 100W amplifier: less power loss than most speaker cables!

When triggered, the SK can quash transients with energy levels of up to 22 joules as fast as 300ηS. Our lab tests demonstrate this by charging a 4200µF computer-grade capacitor to 102V, and instantly dumping it through the module. (Don't try this with a driver, or else!) I question how Mr. Garcia derived his 200mA figure.

The Speaker-Knife uses materials that meet UL-94V fire-retardant requirements, making it safe for permanent building installations, plus internal emergency bypass circuitry—protecting itself and equipment—should an extreme problem occur. Of course, all products have a limit. Again, thank you for the review.

Aram Tokatian
President
Dynastar Electronics

Product Review

SPECTRA PLUS PROFESSIONAL EDITION VERSION 3.0

Reviewed by Mark Zachmann

Spectra Plus Professional Edition Version 3.0, $395 software only, $629 software plus pro quality sound card. System requirements: 386/486/Pentium and Windows 3.1, Windows-compatible sound card, 4MB RAM minimum, mouse and math coprocessor, and 256 color VGA (for spectrogram). Pioneer Hill Software, 24460 Mason Rd. NW, Poulsbo, WA 98370, (800) 401-3472, FAX (360) 697-7730.

OVERVIEW

Spectra Plus is one of a new breed of programs that uses a PC sound board as an I/O device for audio testing. This program turns your sound board into a spectrum analyzer equivalent to a stand-alone commercial audio-band analyzer, with immensely more memory.

Overall, I was very impressed with the program's user interface and documentation. It has the feel of a professional software product and, generally, works like one. There may be better products aimed specifically at analyzing and designing speakers, but Spectra Plus performs well as a general spectrum analyzer, and you'll be happy with the quality of the results. I completed all of my tests using version 3.0a on a 120MHz Pentium with 32MB RAM running Win95 at a resolution of 1024 × 768 in 256 colors.

FEATURES

Spectra Plus includes two main modules—a signal generator and a spectrum analyzer. The signal generator uses your sound board to generate digitized signals: noise, sweep, tones (multiple sine waves), sawtooth, square, or pulse. Each choice has a fairly full option set (*Table 1*).

Although the signal generator and spectrum analyzer are separate modules, they require a sound board that can perform simultaneous input and output in order to work together (most sound boards do not). My sound board (AudioTrix) does support simultaneous I/O, but I was never able to use the signal generator with the analyzer. Instead, it produced error messages. I ran Media Player at the same time as the spectrum analyzer, and I managed to analyze some MIDI clips as well as WAV files.

Pioneer Hill recommends that you install two sound boards if you need simultaneous I/O, which, in my opinion, is a terrible choice. Using one board has to be easier on Windows proper.

The spectrum analyzer works in one of three modes—real-time, recorder, or post-process. Real-time mode is primarily used for setting up scaling and ensuring connections are right—it runs in display priority. The analysis occurs in real-time, but no data is stored.

In recorder mode, the data is shown in real-time, but it is also logged to RAM or disk (with limits to storage implying limits to record time). It also runs in data priority. You can analyze an existing time series or load and analyze a WAV file in post-process mode. In recorder mode, the display may lag the data reception (especially if you have multiple analyzer windows open).

In all cases, FFT sizes are limited to, at most, 16k points (fine enough for 4Hz resolution on a 20–20k test). In post-process mode the FFT analysis can be done in an overlapped manner (which can increase time resolution).

VIEWS

Spectra Plus "analyzes" electrical data by providing five graphical views: 1.) the time series data itself; 2.) the spectrum of the time series (as a common frequency response chart); 3.) the phase of the data; 4.) a 3D view that looks like a waterfall chart

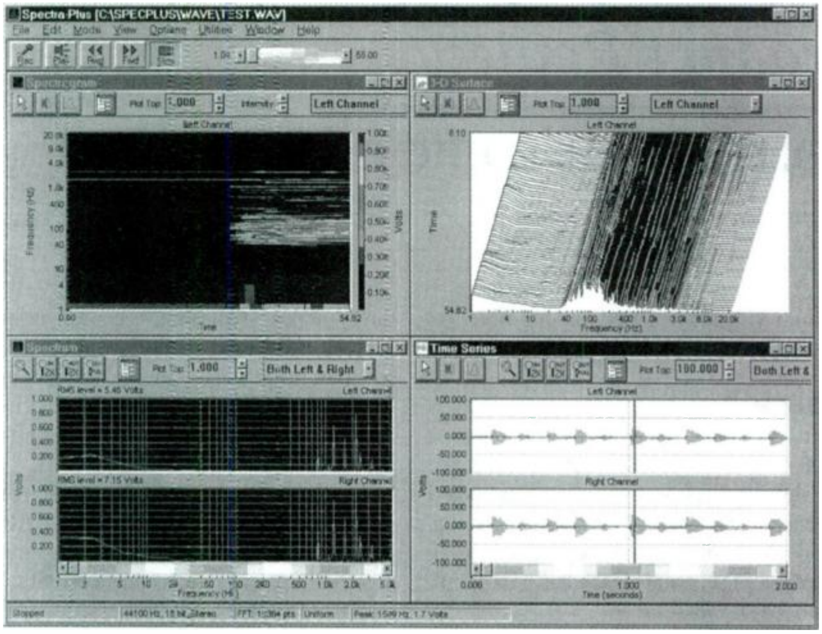

FIGURE 1: Sample Spectra Plus screen display.

TABLE 1	
SIGNAL CHOICES	
STYLE	**OPTIONS**
Noise	white or pink
Pulse	pulse width, repetition rate
Sawtooth	frequency, positive, or negative slope
Square	frequency, duty cycle
Sweep	start and stop frequency, linear or logarithmic sweep, sweep time (in ms)
Tones	frequency, amplitude, of up to ten simultaneous sine waves

but shows snapshot frequency responses; 5.) spectrum data as a spectrogram—colored information with one columnar colored bar per FFT (*Fig. 1*).

The spectrum and the phase together provide a "frequency response chart." The spectrogram is good at detecting when a particular event occurred and also determining general frequency spectral trends. The 3D view is another way to spot spectral trends. The combination of all these views provides a very good graphical understanding of the input sound data.

The view windows each have a toolbar with, among other features, a button to change the chart display, which you can window and zoom in a number of ways to view the data more easily. X and Y have set-table range and origin; they can be linear or log display, and you can select colors for data lines, bars, and grids. You can also choose an amplitude or energy display for time series.

In the spectrum and phase views, you cannot manipulate the data. Other buttons include:

- Zoom in by 2
- Zoom out by 2
- Zoom in range
- Display left/right or both

The time series, spectrogram, and 3D views (as time-based displays) also include buttons to let you select, hear, and filter a time range. The data in a range can be frequency filtered (high, low, band, notch, with selectable frequency, and Q), and copied to or pasted from the clipboard. By double-clicking on the chart, you can also do a "local" FFT for comparison with the entire data set.

All views can export their data sets to ASCII data files, which you can import into a spreadsheet. You can print any of the views, and the printout is a replica of the on-screen window display, but it fills a page of paper, and you can add a title and a subtitle.

In each graphical view you can access data by just poking with the left-mouse button in a chart. You get a full-window crosshair and a display of (X,Y) for the cursor tip. Use the right-mouse button to drag a line to get differential data. Unfortunately, there is no way to link this to the data in the chart, so the display precision is limited to monitor resolution and your ability to hit a spot with the mouse. Selecting the 3D view was particularly chancy.

In addition, Spectra Plus features a distortion analyzer (which I did not try to use). This finds the first big peak and then calculates 2–7 harmonic distortion as well as total harmonic distortion. Other goodies include microphone compensation (you create an ASCII file describing the microphone response), frequency markers, and triggering.

The frequency markers were visually of very limited value. Triggering seems like a very nice feature (I didn't try it) during speaker pulse testing. The program lets you trigger a predetermined time before or after an event—for example, you can program Spectra Plus to begin recording just as the pulse hits the microphone and avoid confusing calculations and leading white noise eating up RAM. I used the program to measure noise and examine an existing WAV file.

NOISE MEASUREMENT

I questioned the amount of noise I seemed to be recording on my sound board. Since my sound board has lots of inputs and outputs, I thought perhaps there was leakage between some channels. I ran Spectra Plus and switched to real-time mode. Then I brought up the spectrum and the time-series windows (because SP lets you save configurations, this could be much simpler if I did it more often).

Finally, I connected my speaker out to my line in and ran Media Player, my mixer, and Spectra Plus. First, I ran Media Player with Canyon.mid to set zero-level position (push the Record button and lower the volume until the time series is within bounds). Then, I turned off Media Player, and as I watched the values swish around, I tried moving sliders up and down to minimize the noise value on the time-series window. Next, I switched to record mode to look at the data more accurately.

This let me lower the range of the data chart to a point where the noise was clearly visible. The spectral chart showed that the left channel was clearly more noisy, with the additional noise having a very non-flat spectral signature. This suggests a likely production problem with the board.

As you record, SP keeps track of the maximum value and displays it in the status bar. This nice feature sure makes setting zero a painless exercise. Of course, the program also has an automatic calibration facility, but this requires a constant input signal and I didn't have one handy (*Fig. 2*).

WAV FILE

Analyzing some voice-data files was pretty easy. You simply switch to post-process mode, open the file, then select the "Run" button so the program acts as if the data were input rather than output (all four non-time-series graphical views update with data based on the current time series). Select a time range and "hear" it to determine which phrase you're analyzing. Then try filtering out some oddities to improve the S/N ratio.

EASE OF USE

With some exceptions, I found Spectra Plus very easy to use. The installation program, while primitive, works well and reliably. There is no uninstall program. The hard-copy documentation is a 63-page 8½ × 11 well-illustrated document with a table of contents,

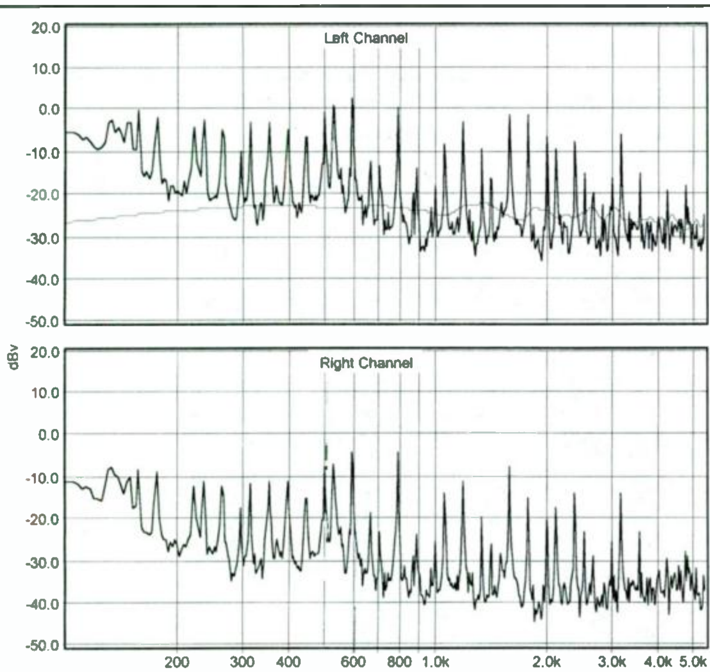

FIGURE 2: Sample Spectra Plus printout.

but no index. It was well-organized and fairly complete. The on-line help was nearly sufficient (and very well organized), but I needed to read through the documentation after having trouble with a few options. I would have benefitted from some pictures of dialogs in the on-line help section.

Some moderately useful information about sound boards—essentially an endorsement of Turtle Beach—was also included with my package. There was also a useful FAQ (frequently asked questions) sheet.

Using Spectra Plus is simple—almost all controls are easy to access, the icons are straightforward, and the text is often clear. Just select the mode, each of which has a toolbar that is labeled, as well as menu options to select the appropriate function. The verb label is not *always* self-explanatory and I needed to refer to on-line help to actually use the product successfully. After the first time, however, it was easy to collect and analyze data using the program. And Spectra Plus struck me as a very professional piece of work.

I experienced some user-interface glitches along the way. For example, the only way to produce an FFT of a spot on a loaded WAV file is by double-clicking at a spot in the time-series window, which is not something you would immediately think of when trying to perform an analysis.

Another glitch is that very often options are grayed out, with no indication why. I tried hard to switch from 11kHz mono to 44kHz stereo and was stymied (and almost wrote a review believing that the program didn't really support two channels or high resolution!). Because recording performs an append, the program only lets you change the data format with a blank dataset. In other words, you must use the New command to change the sound format. Since I started with a low-res signal in memory, I spent some time trying to "ungray" a menu on the status bar.

STUFF

This product is a good fit for its intended audience—people looking for an audio-band spectrum analyzer. For the speaker builder audience, I would have preferred Pulse and MLS testing of speakers and more speaker-specific functionality (e.g., Thiele/Small parameter estimation and impedance calculations). As an engineer, I would have liked more functionality for data comparison and storage. Furthermore, the post-process control was primitive (just add and subtract right/left, which is nowhere near the functionality of SPICE's calculation language, for example), restricting the usefulness for speaker testing.

Some obvious improvements might include charting combined amplitude and phase, and just overall better charting control (changing bounds, for example, is slow and painful). The program contains overlays for viewing more than one signal at a time, but accessing and using that feature was tedious and not all that useful.

There was also the occasional bug. I found a way to crash the program repeatedly. For example, a certain paste insert ran for three minutes on my P120, filled the hard disk (with an easy-to-find file), and then crashed. I was also unable to play/record simultaneously. Cutting a portion from the Spectrogram did not really work right, and took a long time. In fact, the Spectrogram and the 3D view seemed to be later program additions that weren't completely worked out. The 3D-view filter icon sometimes did not work, and the selection mechanism seemed strange and awkward.

I ran the program under debug Windows to check for programmer errors and found only a few places where the debug kernel indicated a fatal error—an average plus result. In retail Windows, these errors generally cause no harm, but they do indicate places where the programmer isn't achieving the desired result (for example, Win Word would score average on this test). Finally, the File/Import and File/File Options... features were in "not yet implemented" developer's heaven.

OVERALL REACTION

I really liked this program, as much for the presentation as for the functionality. But overall it is too much like a real spectrum analyzer, without the additional functionality a keyboard and CPU with lots of main memory should give you.

Would I buy this product? Probably not. I don't really need a spectrum analyzer, and my IMP does a pretty good job of speaker testing. On the other hand, a good PC sound board will have much higher resolution (both S/N and frequency resolution for the FFT) than the IMP. Certainly, for more general applications, I think Spectra Plus is a darned good offering.

I am pleased with the quality of the product and the professional results this medium-priced sound card offers. If Spectra Plus is indeed representative of a new breed of sound-board-based PC software, then speaker builders should experience some interesting years in the future.

Ask *SB*

FLARING CONCERNS

I have some questions about determining port dimensions for bass reflex, specifically, bandpass speakers. In the general equation to determine port length as taken from *SB* 2/81, p. 20, how were the numerical values for the k factor determined? It appears that the easier it is for air to move in and out of the port, the shorter the port can be.

$$L_v = 1.463 \times 10^7 R^2/(f_B^2 V_B) - kR$$

where:
L_v = length of vent (in.)
R = radius of vent (in.)
f_B = desired tuning frequency system (Hz)
V_B = volume of box (ft^3)
k = 0.83 for each flanged end and 0.613 for each unflanged end

My fluid-flow textbook gives loss factors for various pipe entrances and exits. Curiously, the loss factor is given the variable K (capitalized). However, I can't see how the k factors relate (if at all) to the K factors. For example, consider the following entrances of a pipe and the respective K factors:

pipe protruding from wall: K = 0.78
end of pipe flush with wall, sharp edge: K = 0.45
end of pipe flush with wall, slightly rounded edge: K = 0.2
end of pipe flush with wall, well rounded (with radius of edge = 0.2 × diameter of pipe): K = 0.05
for all exit shapes: K = 1.0

These K values are used in the equation

$$hm = (K \times V^2)/2 \times g$$

where: hm = minor head loss (in.)
K = loss factor
V = velocity of fluid flow (in/sec)
g = acceleration due to gravity (386.1 in/sec^2), which is then used in Bernoulli's equation.

Note that k and K appear to be inversely related. In other words, as it becomes easier for air to enter a pipe, k increases in value, while K decreases.

The textbook also gives K values for various pipe roughnesses and bends, which are helpful if the port tube has a bend (assuming the K values can be incorporated into the port equation). Were the K values used to determine the k values? If so, can the K values for a pipe with rounded edges be "converted" into an appropriate k value, so such a port can be used in a speaker enclosure?

I'd like to know if adding a considerable flare (such as for the bell of a trumpet or other instrument) to both ends of a port would permit a shorter port with possibly a smaller diameter than one without flares. Can I calculate, at least somewhat closely, what the port dimensions would be if the tube had "well-rounded" edges at both ends? (I plan on making flares out of wood and then joining them to the tube.)

I have a picture of Bose's three-chambered Acoustimass subwoofer (the walls are made of a clear material), and the outermost port has flares (as I described) on both ends of this tube. (Amusingly, Bose refers to this port as an "elliptical torrid conduit," for those who are under the impression it is just a port tube.) The ad states that this port "provides for laminar flow. Benefit: No audible noise caused by turbulence, even at high levels." This is what I'd like to have for my speaker.

J. David Keener
Lancaster, PA

G.R. Koonce responds:

The K (uppercase) factor used in computing loss in fluid flow is different and independent of the k (lowercase) used in the referenced duct-length equation. This is not to say that loss in a port is unimportant; it is needed to determine the anticipated system response and the amount of noise generated by the port. But the duct-length computation has nothing to do with loss.

Let's look at the tuning of a vented enclosure in a simplistic way to understand this difference. When a duct (port) is added to an enclosed volume containing a driver, it forms an additional tuned system (in addition to that of the driver itself) composed of the compliance of the air in the volume acting with the mass of the air in the duct. If you think of the air volume as a capacitance, then the duct air mass represents an inductance. To tune the fixed volume enclosure to a desired frequency, vary the "inductor" by changing the length of the port duct.

The duct has the problem of "end effect"; that is, the air mass that forms the inductor consists of more than just the air in the duct. The amount of air at each end of the duct, based on the duct's radius, will move with the air in the duct, making the duct appear to be longer than its physical length, resulting in the end effect. The amount of air that moves external to the duct depends on how the duct is terminated—standing bare in space or mounted in a baffle, for instance.

The k in the duct-length equation tries to correct for this termination to allow you to establish the duct's proper physical length. The k values are established from data on the effects of various duct end terminations on the effective duct length.

If loss is introduced into the port duct, it affects the damping applied to the woofer at the box tuned frequency and may affect the shape of the system response curve and the system efficiency. Near-field test results showing how grille cloth over the port affects driver damping are shown in SB 3/95, p. 14 (be sure to see the correct Figure 7 shown in SB 4/95, p. 54). It is clear that you need to maintain low losses in the port unless you are intentionally factoring such loss into the system design.

To the basic question, "Is there any reason to flare or otherwise treat the duct ends?" the answer is definitely yes. When the air velocity becomes high in the vent, due to high power output or too small a vent, the problems of audible noise and other adverse effects can occur. Additionally, trying to flow too much air through the duct will cause loss, affecting the system response and efficiency. It is generally noted that the high-velocity air flow exiting the port (at either end) into the low-velocity air external to the port produces turbulence that causes noise and increased loss.

Many years ago I placed a diffuser at the port output (see SB 2/81, p. 16, and 2/91, p. 45). You can think of this diffuser as a rather crude flare at the room end of the duct. This approach allows a higher level of and better-sounding bass than a conventional port duct for exclusive use on vented systems.

I have no experience with bandpass systems, as I have never built one. The diffuser port approach precludes the computation of the required port-duct length, but this is no problem, since the duct-length equation produced very inaccurate results in my experience with conventional ducts. With the diffuser port, the duct portion will be shorter than the equation predicts for a conventional duct, allowing you to use a larger diameter than would be possible with a conventional duct.

The best published work on using special ends on the duct is by Matthew Polk ("More Bass in Less Space: A New Approach to Subwoofer Design," Audio, May 1996, p. 28). In developing a high-power subwoofer bandpass system of a new type requiring a duct of 6″ diameter and 30″ length, he found that the duct severely limited the output power capability of the system. A duct of 9″ diameter and 60″ length provided better performance, but was totally impractical from a size standpoint.

He then changed the system to a "Power Port," featuring a flared plug at one or both ends. The end of the duct is slightly rounded, and the flared plug is installed on a rod through the port, resulting in a structure looking like a trumpet with a mute, except the outside diameter of the plug greatly exceeds the duct diameter. According to the designer, the Power Port plug converts the high-velocity air flow in the port duct to low-velocity flow around the circumference of the plug without generating turbulence.

Polk developed a 14″-long Power Port consisting of a 5.5″ diameter duct (with a rod running through it) for the above system, which performed better than with the 9″-diameter duct and was 1.5dB more efficient than with the 6″-diameter duct. The maximum diameter for the flared plug with this design was 12″. Polk provides no information on computation of the dimensions of a Power Port, but states that additional information is available for a small fee by contacting Polk Audio (5601 Metro Dr., Baltimore, MD 21215, 410-358-3600, FAX 410-764-5266). Mr. Keener and others wishing to produce quiet, efficient ports at high-power levels should review this work.

In summary, no direct relation exists between the K in Mr. Keener's flow-loss equation and the k based on duct-end termination used in the duct-length equation to correct for end effect. Loss in the port system of a vented structure is important to both the system response shape and efficiency. High-velocity air flow through a duct produces turbulence when meeting low-velocity air at the ends of the duct, which produces loss and noise.

Duct-termination devices, ranging from the simple diffuser through port flaring to Polk's Power Port, can be effective in producing quiet, efficient port ducts. Use of port flaring or the Power Port approach lets you use a smaller diameter (and thus shorter) duct, as maximum air velocity in the port is less critical. I know of no equations for computation of the dimensions of these special-shaped ports, but believe you must individually tune any port, since I have never found the classical port-length equation to produce accurate results.

DRIVERS FOR HORNS

I have been following your horn articles in SB since the start. Prior to this, I built horn cabinets that were mostly unsatisfactory. At the time of your midrange horn article (SB 1/86), I was using JBL 2440s on their particleboard studio horn. The lower end was filled by standard reflex cabinets loaded with JBL LE15As.

Now I run JBL slot tweeters (driven by 60W at 6k), 2440s on your mid horn, with a precision one-pencil-thick air space (driven by 150W at 500–6k), twin 2220s one side (as with the Show Horn), and one side front and rear loaded (driven by 200W at 35–500k). I find the front and rear loaded arrangement gives a "sharper," but certainly not linear, impact.

My problem is that I have built a subwoofer and cannot find a suitable driver(s). The cabinet is a 25Hz corner horn with one 90° bend. I am fortunate enough to have a 60-year-old barn for a shop. The horn is 7′ at the mouth and about 25′ long. Using a JBL 2215 gave a massive peak at 70–80Hz, and even when driven with my 500W amp it was a poor match for the upper bass. Using a JBL 2220 gives a more linear response but at levels as low as 1W, I can hear it bottoming out. I am currently looking for an 18″ driver with the 2220 type of structure. Am I on the right track?

Mike Edwards
MEdwa2515@aol.com

Bruce Edgar responds:

I'm not surprised that the JBL 2215 gave you problems. With an f_S of 20Hz, the suspension is very loose and would not like working into a horn throat. I can't tell from your description of the bass horn what the mouth and throat areas are, but that information is needed to perform any analysis on your horn.

The resonance you describe may be caused by a "bell mode" resonance around the mouth. You can detect the resonance by feeling around the mouth when the driver is driven hard around the resonant frequency by an oscillator/amp combination. Cross bracing at the mouth and sand weights are the usual prescribed cures. If the mouth is not the location of the resonance, feel along the horn to find the problem point. Again, try cross bracing and sand weights.

Most of the 18″ drivers have mass rolloffs of 150–200Hz. You will not find any 18″ drivers with higher mass-rolloff frequencies comparable to the JBL 2220 or EVM15L 15″ drivers. I recommend using two 18″ drivers on a 25Hz horn. The use of two drivers will give better low-end response and shorten the physical length of the horn. Good driver candidates include the EVM18B and the PAS ER-1880. I've used both. The recommended throat areas are in the range of 150 to 180 in^2.

HORN DESIGN

Can you recommend the best, most comprehensive issue and/or book containing math formulae for the design of mid and bass horns? "The Show Horn" (*SB* 2/90) and "The Monolith Horn" (*SB* 6/93) both look interesting, but which is better? Is there another source to consider?

Harry Siddeley
Dracut, MA

Bruce Edgar responds:

Over the years I have amassed almost every article on horns that I could find and, unfortunately, I have not discovered an article that gives an amateur speaker builder a good comprehensive way of designing horns. Most authors avoid treating horns in a serious manner.

In my survey, I have also found that many articles just repeat erroneous notions about horn design without verifying their validity. The best examples are Dinsdale's horn papers in Wireless World *(1980: pp. 18–24, March; pp. 133–139, May; pp. 186–190, June 1974). He started using a flare frequency modified by a factor of 1.2. So, if you computed the flare contour for 100Hz with Dinsdale's method, you would use a frequency of 84Hz.*

The problem with this notion is that you will end up with a horn much larger than you actually need. For midrange horns, the size increase is not too bad; but, for bass horns the size increase will result in a gargantuan horn. The second problem is that the resultant design will have a mouth size too small for the actual flare rate. A mouth too small results in a peaky response. Unfortunately, several software programs use this erroneous method for computing horn contours.

I have tried to write my horn papers so that the amateur can follow each step in the horn-design process. My Show Horn article details how to design a bass horn. The Monolith Horn article shows the pitfalls encountered when you try to fold a bass horn with too many 180° bends. Since I wrote those articles, I have reached some more observations about horn design.

For a forward-firing bass horn, a quarter-sized mouth always gives the best low-end-response results. The Show Horn uses a 1/8 size, but a ¼ size mouth for another 50Hz horn with the same driver and folding geometry had better low-end response. The forward-firing horn design gives better imaging, but your listening position has to be far away (15′, for example) in a large room for the best integrated sound.

The Monolith horn uses a 1/8 sized mouth, but the mouth is positioned right at the wall/floor interface for the best low-end response. The advantage of this configuration is that it works well in a smaller room. I have used it effectively as a center-channel subwoofer. The disadvantage is that the bottom-firing horn spreads the sound out in such a way that the imaging suffers.

Also, the configuration sounds best with a concrete floor. For a raised wooden floor, the bass goes right through the floor unless you isolate the base with a concrete or marble slab or a sand base.

SB Mailbox

PHASE LINEARIZATION

The subject of linear phase in loudspeaker design has received much attention over the last 30 years. Apparently, it is difficult to design a complete loudspeaker system that has linear phase, but still preserves other qualities of interest to the designer, such as side frequency range, good directional properties, and good power handling, in addition to the obvious requirement of flat amplitude response.

Dennis P. Colin ("Waveform Phase Distortion," *SB* 1/97, p. 18) states that the all-pass response cannot be equalized by filtering so that it becomes a linear-phase response. I'm sure he doesn't mean this, because one of the subjects in advanced filter textbooks is phase equalization, which can be accomplished by passive, active, and digital means. Preprints from the AES concerning filter design for loudspeaker dividing "networks" in the digital domain usually state that they have been rendered phase linear by suitable digital filtering.

Phase linearization in the analog domain, especially with passive networks, is difficult, and you don't see it much in speaker designs. As Mr. Colin says, linear-phase loudspeakers usually arise from careful dividing network choice and physical driver time offset. But as he may have seen, some speaker manufacturers make digital equalization boxes (reviewed in *Audio* and *Stereophile*) tuned for specific models, and one of their functions is phase linearization.

I agree that phase linearization in the mid and low frequencies is desirable and improves transient response. In fact, I try to push linear phase down to as low a frequency as I can, because I don't believe that waveform distortion is more tolerable at low frequencies. I made my own active filter low-frequency phase equalization box for the KLH 6 speakers, and tested them in both a listening room and in the anechoic chamber at the Altec Lansing Corp. in Anaheim. I also published the circuit for this box, and the supporting theory, in the *JAES*.

It was easy to tell by listening whether the low-frequency phase equalization filter was "in" or "out" in the anechoic chamber when the test signal was a single square-wave pulse (one full cycle at 60Hz), repeated about once per second. At home, the test signal was music. The box made "Also Sprach Zarathustra" sound terrific. My own home-brew full-range speaker designs have been either rigorously linear phase or approximately so up to as high a frequency as I can manage, usually just above the tweeter crossover frequency.

Victor Staggs, Ph. D
Orange, CA

Dennis Colin responds:

Victor Staggs is right; it is possible to compensate the phase of an all-pass, at least over a finite bandwidth. (The idealized cases I showed, with zero-rise-time pulses and response to DC, would require an infinite bandwidth high-frequency delay and an endlessly growing exponential step response, respectively.)

In practice, over our 20Hz–20kHz bandwidth, compensation is very difficult to do in analog, because many cycles of the highest frequencies must be delayed to become aligned with the lowest ones. Digitally, however, it is not difficult, and I'm glad to hear of the work to which Victor Staggs referred.

I would also like to correct two other not strictly accurate statements I made.

1. Some frequency-response-altering functions do have a constant zero phase shift. For example, the Laplace functions $1-s^2$ and $1-1/s^2$ have a flat region which curves upward to 12dB/oct. boost, above or below s (normalized frequency) = 1, and zero phase everywhere. However, these functions (polynomials with only even powers of s) have limitless boosts, obviously not useful in crossovers.

2. Crossovers higher than first order can be realized with perfect combined phase linearity, but these have two disadvantages:

a) At least one driver signal must have a response peak near the crossover frequency, increasing power loading, and

b) The driver signals in the crossover region differ in phase by perhaps 120° or more. This makes the correct combined response occur only over a narrow angle relative to the axis of separated drivers.

I thank Victor Staggs for pointing out the feasibility (and desirability) of phase compensation. It may be difficult, but to my ears it's well worth it. Regarding transient reproduction, you could say that phase linearity provides a "degree" of bringing us one "step" closer!

SUBWOOFER EXPERIMENT

Bill Fitzmaurice's well-prepared article regarding the use of a "series-vented" design was of great interest to me ("An Eight-Inch Subwoofer Test Box," *SB* 8/96, p. 28). It was easy to understand and provided a possible answer to my PA low-frequency needs. He referred to the Speak-Easy computer program, which models this type of alignment, but noted his lack of a computer on which to run it.

I use a computer program called Bass Box 5.1 (upgraded to 5.18), which models the series-tuned alignment. I input the data from the article regarding chamber volumes, vent sizes and lengths, and driver, and the result was a similar low-end, but very uneven, overall response in the passband of ±9Hz. This program allows for various amounts of stuffing, and I selected "minimal," given the amount of eggcrate foam used. Inputting other amounts resulted in little change, other than to alter the peak values.

Fred I. Mullins, Jr.
Charleston, WV

Bill Fitzmaurice responds:

I appreciate your interest in my subwoofer experiment. While your computer-generated response curves do not exactly match the actual test results of the test-box, this is not surprising. A computer program can produce a result only as accurate as the database on which it depends. Other computer plots by G. L. Augspurger give results much closer to the actual measured response.

I suggest you try a simulation using a pro-driver, such as an EVM-15B, to see if *you can get wideband response without losing too much efficiency. My experiments using actual cabinets have not been successful in doing this with the series-port enclosure without requiring enclosure sizes so large that a folded horn is still a better alternative. I'd be interested to hear about your results.*

FREELINE FACELIFT

I was interested in Bing Yang's questions ("Ask *SB*," 1/97, p. 39) relating to John Cockroft's "Freeline" design (*SB* 5/95). After 40 years of diddling and fiddling, I have concluded that loudspeaker specifications don't really mean a whole lot, and Mr. Cockroft's allusion to "spiritual and emotional involvement...with the music" is infinitely more important than actual (or computer-modeled) response flatness, for example.

Intrigued by the simplicity of the "Freeline" design, I built a pair from pine shelving rather than the specified particleboard (these are not nearly as dense, of course, but TLs aren't supposed to generate much internal pressure). Besides, a hand-rubbed oil finish looks great on real wood.

To say that my "Freeline" results were impressive is an understatement. I wasn't prepared for such a big, open sound from such small enclosures. Nonetheless, by accommodating the one unusual factor of this design, you can achieve a dramatic performance improvement.

The "Freeline" design places a fixed resistor in series with the woofer to yield a bit more "Q," hence output, near the (weighted) woofer resonance. This is a neat trick, but with two shortcomings.

First, the speaker appears to take more power for an expected SPL—not in terms of watts, however; my 25W test amplifier "went to the rails" before it truly ran out of steam. I proved this by inserting a 1:2 step-up transformer. The ultimate solution was a home-brewed amp of a "bridging" configuration which could deliver nearly 100V peak-to-peak at modest current.

The series resistor also spoils the amplifier's damping effect. This is evidenced by excessive woofer excursion at obviously subaudible frequencies. By incorporating a third-order Butterworth active low-pass filter (−3dB at 30Hz) into the input circuitry of the new amplifier, I easily doubled the system's apparent power-handling capability. (It is surprising how much subaudible "noise," that is, stuff unrelated to the music, can be found in many recordings. Take Eric Clapton's 1992 "unplugged" *Layla*, for example. What *is* that thumping?)

The "Freeline" won't blow doors off their hinges. A 4″ cone can move only so much air. If you're after contrived movie effects and other sounds that can't possibly exist in the real world, you'll need a similarly artificial reproducer. But for music, the "Freeline" will get you as spiritually and emotionally involved as you're likely to want to be.

John Wood
Felton, CA

John Cockroft responds:

Your use of a step-up transformer (and later your home-brew high-voltage amplifier) is a unique solution (at least I've never heard of the process) to what I also thought were shortcomings in the Freeline's performance. I eventually made quite a few modifications to the Freeline in several areas, including removing the 8Ω series resistor circuit. Just a couple of weeks ago I did an entire makeover of the crossover.

The Freeline now has a different woofer, the 4½″ Pioneer full-range A11EC80-02F (Parts Express #290-010) costing $10.50. This is slightly larger than the original Radio Shack woofer, so you'll need to enlarge the mounting hole a bit. It's best to remove the woofer, tweeter, and most of the stuffing in the woofer hole to give the sabre saw a bit of elbow room.

At this point it is a good idea to spot-glue some ¼″ felt pads to the inside back and side panels, centered on the centerline of the woofer. They should be about 6″ long. I forgot to mention these in my original article, but in view of your kind description of the way your Freelines sound, perhaps the pads aren't all that important.

Apply two coats of white glue to the woofer's cone and dust cap, in the manner of the original article. Also add eight BB-size split lead-shot fishing sinkers (as opposed to nine on the original Radio Shack woofer). The Pioneer's dust cap is slightly smaller, but its resonant frequency is about 15Hz lower to start with, so it all works out). Split the shot as explained in the article and mount them around the dust cap. They fit better on the Pioneer if the little polywog tails point inward towards the dust cap.

The current Freeline has higher stuffing densities than the original. I raised the lower (longer) line density from 0.9 lb/ft³ to 1.375 lb/ft³ by adding 2¾ oz of stuffing. The Freeline isn't stuffy. You merely need to push up the original stuffing to accommodate new stuffing. The upper line density increased from 1.2 to 1.8 lb/ft³, by adding 1¾ oz stuffing.

As a result of a recent discussion with Vince Bruzzese of Totem Acoustique and Scott Frankland of Wavestream Kinetics (and the Analog Room), I decided to redesign the Freeline's crossover (again).

The treble section now uses a 0.22mH coil (Parts Express #266-658, $6.30) shunted across the tweeter terminals, and a 2.5µF capacitor shunted across the woofer terminals. I used a 1.5µF (#027-528, $2.15) and a 1.0µF (#027-520, $2.15) Solen capacitor in parallel. I discarded the original 3.3µF cap and the 8Ω and 10Ω resistors, while retaining the 8Ω variable L-pad (Fig. 1)*. The Freeline sounds best in my room with the L-pad wiper turned slightly less than ¼-turn from the off position.*

This current setup is about 6dB more sensitive than the original Freeline. It is very noticeably cleaner and clearer and more naturally alive. I prefer the Freeline out away from the wall farther than before—about 2½ to 3ft.

To augment this, I devised a simple passive line-level filter to partially compensate for diffraction loss that occurs when the lower frequencies wrap around the enclosure and are depressed on the speaker axis. It is a simple first-order low-pass filter with an additional resistor slipped in between the ground leg of the capacitor and ground. I built the two-channel filter on the back of a four-jack RCA jack board from Radio Shack and mounted it in a small plastic project case from the same source. It goes between the preamp and the amplifier (Fig. 2)*.*

ATTENTION TO APPEARANCE

I would like to suggest that the use of the term "Wife Acceptance Factor" in speaker building is a result of not fully analyzing a series of aesthetic, marketing, and engineering problems ("*SB* Mailbox," *SB* 1/97, p. 42). Most of the speaker systems I build are for friends and neighbors, and one is in a church sanctuary. I designed each to meet specific sonic and aesthetic needs. I am lucky to earn a dollar per hour for my efforts, so I consider this a hobby. My paid work is in courtroom sound reinforcement and recording.

Many outsiders critique my work, unlike

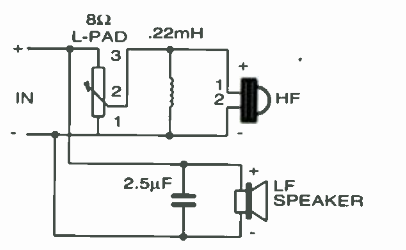

FIGURE 1: Current Freeline crossover unit.

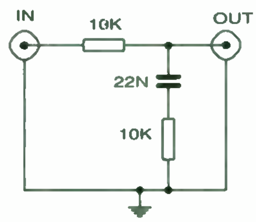

FIGURE 2: Partial diffraction compensation filter. Resistors are ¼W, such as Radio Shack #271-1335. Use a polyester film capacitor, such as Radio Shack #272-1066.

that of the pure hobbyist. Some know me well, some do not, some are women, some are men. I have not found women to be any more or less difficult or critical than men. Success is related to how well I can integrate my system into the existing living room, church, or courtroom.

Most people who build speaker systems are male. They build for themselves and hope that the rest of the family will at least be tolerant. Due to circumstances only, the first critic will almost always be the spouse, that is, a female.

To increase the acceptance factor, first consider sending the back wave from the woofer out of the living space. Minimizing the space your speaker system occupies increases acceptance. Most of a speaker system is empty space that serves only one purpose: to contain or phase-shift the back wave so it does not cancel the front wave.

Second, take your wife to antique shops and furniture stores to determine what is acceptable. You could transform many different hutches and corner cabinets into speaker or subwoofer cabinets. Sign up for a woodworking class to learn how to produce nice furniture. Ask your wife to clip pictures of furniture out of magazines and talk about them. Modify the design of a classic piece of furniture and build it.

Keep your woofer/midrange crossover point as low as possible by using a very wide response midrange. This will make your speaker system less sensitive to driver placement, thus increasing the styling options. Achieving good time adjustment has produced some very ugly systems. Crossing over below 350Hz permits a flat vertical baffle board. Crossing over below 90Hz with a fourth-order network has been reported to allow many feet of separation between the woofer and a bass midrange.

Install a prototype pair of speakers in the room for a limited time to test a design. Then transplant the drivers and crossovers to a finished pair of cabinets.

Consider removable doors for your speakers. If these have inset grilles, you may be able to get fair sound with the doors closed, provided care is used in driver placement. For serious listening, remove the doors.

Use nontechnical examples for why speakers need to be big. Explain that the major difference between a calliope and a fine pipe organ is that the calliope uses small-bore pipes and high-pressure steam to produce a sound full of harmonically unrelated noise and weak fundamental tone. Large pipe organs use very low-pressure air and large bore pipes. Fundamental tones are strong, harmonics are pure, and there is little noise.

I encourage *Speaker Builder* and its readers to explore ways to improve the cosmetic aspect of speaker systems. The magazine has done an excellent job of presenting the technical aspect of speakers.

Jesse W. Knight
Woburn, MA

UPDATE

NETCALC, reviewed in *SB* 1/97 ("Software Review," p. 40), is now distributed only by Audio Components, PO Box 554, NL 5340 AN Oss, The Netherlands; (+31) 412-626610, FAX (+31) 412-633017, E-mail audiocom@worldonline.nl. In addition, NETCALC's price has dropped from the original $149 to $85. Readers with technical questions or needing other assistance may contact the designer directly: Mr. Rien den Boer, Ariba, Barnoniehoeven 32, NL 5244 HZ Rosmalen, The Netherlands, FAX (+31) 73 52 15841, E-mail mdenboer@worldonline.nl.

HELP WANTED

I have been searching for a suitable crossover schematic. I wish to replace my extremely noisy Mirage LFX-2 with something I build myself. My system consists of B/W 803s and a set of subs powered with STASIS amps. I would like to vary the high-pass/low-pass points from 50–80 cycles with 24dB/octave slopes. I'm looking for the best design around.

Pete Perciavalle
perciavalle.p.r@wec.com

I own a venerable pair of Dahlquist DQ 10s. I have frequently been told that I should replace the super tweeter, but with what? Does anyone have a recommendation, or know someone who could help me?

Geoffrey Binder
GeoffreyBinder@vut.edu.au

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Hovland Company	*MusiCap Film & Foil Capacitors*	1-209-966-4632
Kimber Kable	*Speaker cables, interconnects, etc.*	1-801-627-6980
Klein Tech Systems	*Acoustic Interface Pads*	1-561-845-9844
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Markertek Video Supply	*Unique & Hard-To-Find Accessories*	1-914-246-1757
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Solo Electronics	*CFAC Inductor*	1-510-887-1657
Sound Clearing House	*Speaker Drivers*	1-541-343-3112
The Parts Connection	*Crossover Parts*	1-905-829-5388

AD INDEX

Speaker Builder 1980–88, 94–96, $120; LinearX Leap and LMS, $1,075. All plus shipping. Dennis Mohs, (612) 937-0741.

Peerless 1727 10″ woofers, never used, $40 each; Audax TSX1125JSN, new, $19 each; AW02551, new, $19 each; CFAC 0.47mH coils, 12 gauge, $19 pair; MC preamp kit, battery operated, $25; power transformer 20V CT at 1A, $5. Dave, (412) 274-8149.

Dynaco PAT-5 updated boards, defeated stereo-mono and low-filter switches, blend pot added, $50; Integrex Dolby-B decoder, new, $48; SSS EQ1 12-band equalizer kit with extra parts, $30; Hafler DH-330 tuner, $125; Pioneer TVX-9500, $20. Collectable hi-fi books, SASE for list. Andy Keller, 1455 Twinsisters, Longmont, CO 80501, (303) 772-4801.

Pair ScanSpeak Kevlar 18W8546 7″ mid-bass, $210; pair D2905/9300 1″ soft-dome tweeters, $110. New matched pairs. Ed, (206) 562-9848, E-mail clrsoft@wolfenet.com.

Pair JBL 1″ horn drivers, Alnico with aluminum diaphragms; pair Altec 291A large Alnico horn drivers; pair 511, 811, and 421 woofers; Pioneer Spec I, II, and IV preamp and amps; two Nakamichi 6702X rackmount cassette decks; Community horns. David, (914) 688-5024.

Pair Advent I tweeters, $20; E-V RE85 Dynamic Lavalier mike, $35; E-V 649B Dynamic Lavalier mike, $25; Audio Technica AT816 unidirectional Dynamic mike, $20; one University N2B crossover, $10. Also have Advent I woofers (needing surrounds) and Advent I crossovers. Jim, (408) 425-6719.

Krell PAM-3 preamp, $1,100; Lexicon CPI V2, MSB modified to accept AC3 and DTS, $1,200; TDM three-way stereo electronic tuner, $200; speaker selectors by Nilen SPS-1 and HDS-1; Technics SU9070 preamp, $100; Technics SH9010 stereo EQ, $100; Dynaudio 21WS4, D76, D28, and D21; Audio Concepts AC5; Vifa D25; JVC HS 1101 ribbon tweeter. Bert Lopez, (914) 362-0927.

BOSE Acoustimass bass modules: AM3, AM4, AM5, AM7, no cubes, $75 to $125 each; Bozak woofers, mids, tweeters, crossovers; pair repro E-V "Patrician 800s" with original drivers, $4,500; pair E-V 30″ woofers, factory rebuilt in 1996, $1,000 pair. Bud, (912) 432-9885 after 6 p.m. (EST).

HP 200CDR wide-range oscillator, 5Hz–600kHz, rack mount, balanced/unbalanced output, low distortion, $125; HP 302A wave analyzer 0–5kHz tunable voltmeter for distortion product measurements, $200; TEK 561A scope, three vertical and two horizontal plug-ins, rack mount, spare CRT, $100. Manuals included, plus shipping. Offers. Ron, phone/FAX (408) 246-2874.

B&W 805 in walnut finish, $775. Harold Goldman, (212) 724-9843 (NY).

Altec 511B, 604E, 803B; Apogee; Bryston 10B, Bryston speaker selector; Cambridge SW-1; Entec SW-1; E-V 30″ subwoofer; HP334A, 339A, 8903B; HP301 HV 0–1kV 500mA; JBL B380/X63A; KEF 139B; Magnapan bass panels; M&K speaker selector; ESR-6; Sound Lab A1/B1; Tannoy monitor gold; Vandersteen 4A; VMPS tower. Best offer. John, (408) 737-2980, FAX (408) 735-1426.

Sonic Frontiers SFD-1 digital converter, Telefunken 6DJ8, soft recovery diodes, choke-loaded PS, asking $700; Pioneer PD-65 transport only, analog section removed, separate reg. PS for master oscillator, highly damped enclosure, 75Ω BNC, asking $350; UltraAnalog AES-21 AES/EBU receiver, $50. Ashby, (954) 723-5225.

Kenwood Dolby 3 processor, 30W center-channel amplifier, rear channels need repair for 15W/channel, rear-channel delay, heater, hall, and wideband modes. Rear-channel and center-channel line output. Dr. L.H. Steinberg, (215) 592-9765.

One Dyna PAS, silver face, cosmetically and sonically good shape, $125; ST-70 with Van Alstine pcb, $450; one KLH Model 18 FM tuner, just re-aligned, $75; Revox A77 Model 1224 built-in speakers, everything works to specification, very nice shape, including heads, $500. All items shipped UPS COD. Lloyd, (617) 232-2887.

Dyna PAS-2, $75; Dyna FM-3, new faceplate, $75; Dyna ST-150, Boak-Jung mod, $125; pair KLH 2s; Janszen 130 tweeters, woofer and tweeter in one cabinet, $400. All plus packing and shipping. Paul Leo, (718) 459-5443.

Vifa tweeters: D27TG-35 and D27TG-05 1″ silk domes, D25-AG37 aluminum domes, $12 each; used replacement domes, $2 each; matched pair Focal 8K516 Kevlar woofers, new, $150; Sony WM-D6C micro portable stereo cassette, $225. New equipment in original cartons (with warranty): Technics SL-1200MKII professional turntable, $390; Audio Research LS3 preamp (*Stereophile*-rated Class A), $1,250; Adcom GFA5400 HEXFET power amp, $525. Fred, (610) 693-6167.

Pair AR 4X speakers, fair, $70; Bogen PA amplifier, 100W mono, all transistor, mutiple output taps, good condition, $100; Brainard stereo integrated amp, all tube, 15W/channel, 8Ω and 16Ω taps, 6V6 outputs, works, needs TLC, $65. Ralph, (810) 264-6784.

WANTED

Information on theory of operation of "Karlson" enclosure, a.k.a Distributed Constants Exponential Coupler. Will pay costs. I have the building plans published by Mullard circa 1960. Mark Kelly, PO Box 71, Wahgunyah 3687, Australia, phone/FAX (011) 61-3-57261615.

Vitavox S-2, 220S/431, 220E/321; Lansing 415, 801, 901, 808, N-400, N-800, 284, 285, 287; Jensen V180C, L-180C; Q tweeter, F-SA, F-7, PS-1, 2, 3, 4; W.E. 12025, 12027, 31, 32, 15, 24, 25, 26, 555, 518, 6368, 713, 754, N03, 555W, 594, 595, 596A, 597, 4151, 4181, 4194, 4171, 91, 92, 94, 86, 124, 142, 20A/B, 87. John, (408) 737-2980, FAX (408) 735-1426.

Cash paid for working or damaged Carver ribbons: 30″, 48″, 60″—does not matter. Hal, (540) 459-5571, FAX (540) 459-3809.

Knowledgeable speaker designer/builder experienced in electronics and quality assurance seeks position or association with audio company in New York or Florida. K.J.M. Audio, (516) 868-8863.

PRIME NUMBERS FOR BOX DIMENSIONS

When you calculate V_B, you should address the question of the relationship of width to length to height. Many would try a combination that produces the maximum number of panels from the minimum amount of plywood. There is nothing wrong with saving money, but how correct is this approach with regard to the sound quality?

Some authors have suggested various dimensions for rectangular enclosures designed to prevent the coincidence of standing-wave modes that could produce severe colorations.[1] Standing-wave modes are certainly unavoidable, but there are ways to control them. One is to use absorbent linings inside the box. Vance Dickason writes of other proposed relations and describes interesting results when you use different arrangements of linings.[2] Here we will see that prime numbers can also be a valuable tool.

I have asked many people for whom standing-wave modes are not a clear phenomenon to listen to a fixed low-frequency tone (e.g., 50Hz) in a room with only one speaker, and to stroll through the room. It is then easy for them to find points in the room where cancellations or reinforcements of the tone occur. This effect also happens inside the box, but at different frequencies, because of the smaller dimensions of the box and the involved wavelengths that affect the sound quality.

```
10 CLS
20 PRINT:PRINT:PRINT
30 PRINT"        Please enter the dimensions of the box:"
40 PRINT
50 INPUT"              large (m)          Lx = ",LX
60 INPUT"              wide (m)           Ly = ",LY
70 INPUT"              height (m)                   Lz = ",LZ
80 PRINT
90 PRINT"       enter the highest frequency of interest:"
100 INPUT"                                         Fmax = ",FM
110 V = 344.5
120 PRINT
130 PRINT"                               Fswm               wavelength"
140 PRINT"            Mode               (Hz)               (m)"
150 FOR I = 0 TO 10
160 FOR J = 0 TO 10
170 FOR K = 0 TO 10
180 A = (I / LX)^2
190 B = (J / LY)^2
200 C = (K / LZ)^2
210 D = SQR(A + B + C)
220 IF D = Ø THEN 29Ø
230 F=V . D / 2
240 IF F > FM THEN 270
250 WL = D/2
260 PRINT"           (";I;",";J;",";K;"")",F,WL
270 NEXT K
280 IF F > FM THEN 290
290 NEXT J
300 IF F > FM THEN 310
310 NEXT I
```

FIGURE 1: BASIC program for calculating the standing-wave frequencies in a rectangular box.

PRIME NUMBERS

Trying different sets of three prime numbers, you can find a suitable relation that accords with good performance and your own aesthetic tastes.

Recall that a prime number is an integer that is evenly divisible only by itself or 1. For example: 1, 2, 3, 5, 7, 11, 13, 17, 19, 23,.... The use of prime numbers may prevent the coincidence of standing-wave modes because no prime is a multiple of any other integer.

First, here are some sets of three prime numbers, excluding the number 1 for obvious reasons:

TABLE 1

FUNDAMENTAL FREQUENCIES OF THE TONES OF THE MUSICAL SCALE

A	27.50	55.00	110.00	220.00	440.00	880.00	1760.00
Bb	29.14	58.27	116.54	233.08	466.16	932.33	1864.65
B	30.87	61.74	123.47	246.94	493.88	987.77	1975.53
C	32.70	65.41	130.81	261.63	523.25	1046.50	2093.00
Db	34.65	69.30	138.59	277.18	554.37	1108.73	2217.46
D	36.71	73,42	146.83	293.66	587.33	1174.66	2349.32
Eb	38.89	77.78	155.56	311.13	622.25	1244.51	2489.02
E	41.20	82.41	164.81	329.63	659.26	1318.51	2637.02
F	43.65	87.31	174.61	349.23	698.46	1396.91	2793.83
Gb	46.25	92.50	185.00	369.99	739.99	1479.98	2959.96
G	49.00	98.00	196.00	392.00	783.99	1567.98	3135.96
Ab	51.91	103.83	207.65	415.30	830.61	1661.22	3322.44

TABLE 2

STANDING-WAVE MODES, BOX 1

ORDER	FREQUENCY	DEVIATION FROM CLOSEST TONE
(0, 0, 1)	171.05Hz	2.08%
(0, 0, 2)	342.11Hz	2.08%
(0, 0, 3)	513.16Hz	1.97%
(0, 0, 4)	684.21Hz	2.08%
(0, 1, 0)	398.73Hz	1.72%
(0, 1, 1)	433.87Hz	1.41%
(0, 1, 2)	525.38Hz	0.41%
(0, 1, 3)	649.86Hz	1.45%
(1, 0, 0)	598.09Hz	1.83%
(1, 0, 1)	622.07Hz	0.03%
(1, 0, 2)	689.02Hz	1.37%

2, 3, 5	3, 5, 7 (*)	5, 7, 11
2, 3, 7	3, 5, 11	5, 7, 13
2, 3, 11	3, 5, 13	5, 7, 17

7, 11, 13	11, 13, 17
7, 11, 17	11, 13, 19
7, 11, 19	11, 13, 23

(The relation marked with an asterisk (*) is very close to one proposed by Colloms.)

Denoting a set of three primes as n_1, n_2, and n_3, and their product as P,

$$P = n_1 \cdot n_2 \cdot n_3$$

In order to use these numbers as weighting factors for the dimensions that define V_B, calculate the inner dimensions of the box (i.e., l_x, l_y, and l_z) as follows:

$$l_x = n_1 \sqrt[3]{V_B/P}$$

$$l_y = n_2 \sqrt[3]{V_B/P}$$

$$l_z = n_3 \sqrt[3]{V_B/P}$$

Here it is important to remember that V_B is the net internal volume, after adding the displaced volumes of drivers, braces, and ports.

Now, applying the given equations to some examples, suppose that V_B is 125 liters. First select the primes 2, 3, and 7. Then:

$$P = 2 \cdot 3 \cdot 7 = 42$$

and:

$$l_x = 2 \sqrt[3]{125/42} = 2.887\text{dm} = 28.8\text{cm}$$

$$l_y = 3 \sqrt[3]{125/42} = 4.315\text{dm} = 43.2\text{cm}$$

$$l_z = 7 \sqrt[3]{125/42} = 10.069\text{dm} = 100.7\text{cm}$$

To verify: $V_B = l_x \cdot l_y \cdot l_z = 125{,}287\text{cm}^3$, which is close enough to 125 liters. Call this box 1.

Now try 7, 11, and 19 for the same volume (box 2):

$$P = 7 \cdot 11 \cdot 19 = 1463$$

$$l_x = 7 \sqrt[3]{125/1463} = 3.083\text{dm} \sim 30.8\text{cm}$$

$$l_y = 11 \sqrt[3]{125/1463} = 4.885\text{dm} \sim 48.8\text{cm}$$

$$l_z = 19 \sqrt[3]{125/1463} = 8.368\text{dm} \sim 83.7\text{cm}$$

$V_B = l_x \cdot l_y \cdot l_z = 125{,}773\text{cm}^3$, which is still close to 125 liters.

JUDGING THE RESULTS

In both cases, the resulting volumes do not deviate more than 0.25% from the original volume, and the dimensions fit all right with standard 10"-diameter woofers. But which one is best?

At this moment, it looks as though we are again at the beginning of the problem, but this is not so. Here I suggest a criterion for the assessment of any W × L × H relation. This concerns the fundamental frequencies of the tones in the musical scale. You could select a box for which the standing-wave

ABOUT THE AUTHOR

Hugo Rodriguez graduated in 1976 as an electronics engineer from the National Polytechnic Institute of Mexico. He has worked for about 16 years in the loudspeaker industry for four of the six major companies in Mexico, and at Oaktron Manufacturing in Monroe, WI. He is currently a metrologist in the Pyrometry Lab of the National Center of Metrology of Querétaro, Mexico, and is a charter member of the Mexican Society of Acoustics.

modes are less coincident with the tones of music. *Table 1* gives the values of the fundamental frequencies of these tones.

From *Table 1*, the result of calculating the change of pitch between two adjacent semitones is about 5.95%. The term "semitone" is used to denote this change of pitch. A change of pitch of approximately 12.25% denotes a full "tone"—not twice 5.95%, because music is built on a logarithmic scale.

Then, if any standing-wave mode results from two semitones having an optimal deviation of 2.93% ($\sqrt{1.0595} = 1.0293$), our box will perform at its best. Indeed, this is a very difficult condition to achieve. Those deviations closest to 2.93% must offer better results.

DETERMINING SWM FREQUENCIES

Now we need to determine the frequencies of the standing waves that will occur in our boxes and compare them with the frequencies of *Table 1*. As you can find in some acoustics texts[3] or in reference 1, these frequencies are calculated with the following equation:

$$f_{swm} = \frac{c}{2}\sqrt{(i/l_x)^2 + (j/l_y)^2 + (k/l_z)^2}$$

where:

c = 344.5 m/s (the velocity of sound at room temperature),

l_x, l_y, and l_z are the inner-box dimensions we have calculated,

i, j, and k are the mode orders (0, 1, 2, 3, 4, 5,....) excluding the 0, 0, 0 case.

(Note: This equation is valid not only for calculating the standing-wave modes of boxes, but also for rectangular rooms.)

A BETTER WAY

However, it is tedious to do all these calculations. I wrote a BASIC program to get the first 10 order modes, limiting them to a suitable maximum frequency, because the higher frequencies are absorbed by the inner lining of the box and are also attenuated by the crossover network. The program works with any relation of box dimensions, not only those based on prime numbers. *Fig. 1* is the listing of this program.

In the examples, I limited the calculations to a maximum frequency of 700Hz. With their results, I calculated the deviations of the modes to the closest musical tone. See *Tables 2* and *3*.

TABLE 3

STANDING-WAVE MODES, BOX 2

ORDER	FREQUENCY	DEVIATION FROM CLOSEST TONE
(0, 0, 1)	205.79Hz	0.90%
(0, 0, 2)	411.59Hz	0.90%
(0, 0, 3)	617.38Hz	0.79%
(0, 1, 0)	352.97Hz	1.07%
(0, 1, 1)	408.58Hz	1.64%
(0, 1, 2)	542.21Hz	2.24%
(1, 0, 0)	559.25Hz	0.88%
(1, 0, 1)	595.92Hz	1.46%
(1, 0, 2)	694.38Hz	0.59%
(1, 1, 0)	661.33Hz	0.31%
(1, 1, 1)	692.61Hz	0.84%

TABLE 4

STANDING-WAVE MODES, BOX 3

ORDER	FREQUENCY	DEVIATION FROM CLOSEST TONE
(0, 0, 1)	232.17Hz	0.38%
(0, 0, 2)	464.35Hz	0.39%
(0, 0, 3)	696.52Hz	0.28%
(0, 1, 0)	325.04Hz	1.41%
(0, 1, 1)	399.45Hz	1.90%
(0, 1, 2)	566.81Hz	2.24%
(0, 2, 0)	650.08Hz	1.41%
(0, 2, 1)	690.30Hz	1.18%
(1, 0, 0)	541.74Hz	2.33%
(1, 0, 1)	589.39Hz	0.35%
(1, 1, 0)	631.77Hz	1.53%
(1, 1, 1)	673.08Hz	2.10%

As shown in *Table 2*, box 1 has frequencies of 525.38 and 622.07Hz, which produce standing waves that are very close to musical tones. The latter is essentially Eb, but, as I stated above, this frequency will be better absorbed by the lining in the box than the lower ones. Also, it will be attenuated by the crossover network by 8dB, assuming a third-order crossover network and 400Hz crossover frequency.

Frequencies of modes (0,0,1) and (0,0,2) in box 2 are closer to musical tones than those in box 1. The mode (1,0,1) that is coincident with the Eb tone in box 1 does not occur in the same way in box 2, since it deviates by 1.46% from the D tone.

I tried a third option, using the 3, 5, and 7 set, with the results shown in *Table 4*, where you see a poorer performance. This is because their first modes are closer to musical tones that are in the range of the woofer operation. I do not think that this is a bad relation, but the best one will depend on the size of the box's volume.

For the reason stated above, if I built a 125-liter box, I would not use the dimensions of box 3. The standing-wave mode at 525.38Hz of box 1 makes it difficult for me to select it instead of box 2, but the dimensions of box 1 appeal more to my aesthetic taste.

Hugo Rodriguez
Querétaro, Mexico

REFERENCES:

1. Martin Colloms, *High Performance Loudspeakers*, 4th ed., (Pentech Press Ltd., London, 1991).

2. Vance Dickason, *The Loudspeaker Design Cookbook*, 5th ed., (Audio Amateur Press, Peterborough NH, 1995).

3. Leo Beranek, *Acoustics*, (McGraw-Hill, New York, 1954).

(All books are available from Old Colony Sound Lab, PO Box 243, Peterborogh, NH 03458, 603-924-6526/6371, FAX 603-924-9467.)

PRINTED Circuit Boards

PC BOARDS SHIPPING WT. 1 LB.

Part No.	Description	Size & Reference	Price
PCBC-4	Electronic Crossover DG13R	2″ x 3″ from *Audio Amateur* 2/72	$10.00
PCBD-2	Mitey Mike Test Mocrophone	1-5/16″ × 4-1/8″	$10.00
PCBD-3A/B	Didden Super Regulators (Each PCB consists of one positive and one negative regulator)	1-7/8″ × 4-7/8″ from *Audio Amateur* 3/95	$24.95
PCBF-6	Jung Filter Crossover WJ3	3″ × 3″ from *Audio Amateur* 4/75	$10.00
PCBGK-B1A	Curcio St-70 Power Supply	5″ × 9″ from *Glass Audio* 1/89	$27.00
PCBGK-B1B	Curcio St-70 Driver Board	3-1/4″ × 7″ from *Glass Audio* 1/89	$17.00
PCBJ-5	Pass A-40 Power Amp	3″ × 3″ from *Audio Amateur* 4/78	$6.00
PCBK-13A	Valkyrie Preamp Main Board	4-1/2″ × 5-1/2″ from *Audio Amateur* 1/94	$17.65
PCBK-13B	Valkyrie Preamp Power Supply Board	3″ × 5-7/8″ from *Audio Amateur* 1/94	$10.25
PCBK-6	Waldron Tube Crossover	2″ × 4-1/2″ from *Audio Amateur* 3/79	$12.00
PCBP-10A	Pass/Thagard A75 Power Amplifier	4-1/2″ × 4-3/4″ *Audio Amateur* 4/92-1/93	$14.95
PCBP-10B	Pass/Thagard A75 Power Amplifier Power Supply	3-3/8″ × 5″ *Audio Amateur* 4/92-1/93	$8.95
PCBP-11	Zen Amplifier	5″ × 5″ from *Audio Amateur* 2/94	$12.95
PCBP-3	Borbely 60W MOSFET Amplifier	3-3/8″ × 6-1/8″ from *Audio Amateur* 2/82	$11.75
PCBS-1	Borbely Servo 100 Power Amplifier	4-1/8″ × 6-1/2″ from *Audio Amateur* 1/84	$16.00
PCBS-3	Borbely DC100 Power Amplifier	6-1/2″ × 4-1/8″ from *Audio Amateur* 2/84	$16.00
PCBS-6A	Curcio Tube Pre-preamp Master Power Supply	4-3/4″ × 3-3/4″ from *Audio Amateur* 5/84	$10.35
PCBSB-E4	Muller Pink Noise Generator	4-1/8″ × 2-3/16″	$9.40
PCBT-1	Curcio Tube Preamp Regulator	10-11/16″ × 2-11/16″ from *Audio Amatuer* 2/85	$27.00
PCBT-1A	Curcio Tube Preamp Master Power Supply	3-3/16″ × 2-11/16″ from *Audio Amateur* 2/85	$8.00
PCBV-2	Lang 20W MOSFET Power Amp	5-11/16″ × 6-1/16″ from *Audio Amateur* 2/86	$21.40
PCBV-3A	Curcio 'Daniel' Auto Mute	1-1/2″ × 2″	$8.00
PCBW-3	Borbely Improved Power Supply	4-1/4″ × 5-1/2″ from *Audio Amateur* 1/87	$16.00
PCBW-4	Imp Audio Analizer	4 × 6-3/8″	$39.95
PCBW-4MLS	Imp Audio Analizer Maximum Length Sequence (MLS) Add-on Board	1-1/2″ × 2-1/16″	$13.95
PCBW-4MLS/PR	Imp mLs pcb plus stmls imp prom		$38.95
PCBW-5	Lib Instr mic/probe preamp pc board		$14.95
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PCBY-3	Youtsey Anti-Jitter Board	3-1/2″ × 4-3/4″ from *Speaker Builder* 3/96	$15.95
PCBK-3	Crawford Warbler	3-1/4″ × 3-3/8″	$11.20

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3	$ 4.75	$ 7.50	$20.25	$ 8.25
4	$ 5.00	$ 9.00	$26.25	$10.00
5	$ 5.25	$10.50	$33.75	$14.75

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Vintage Design

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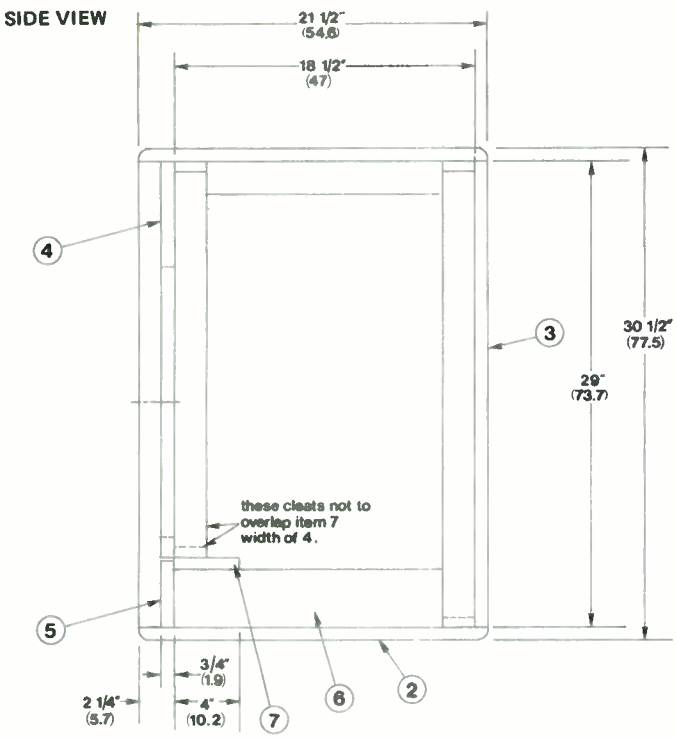
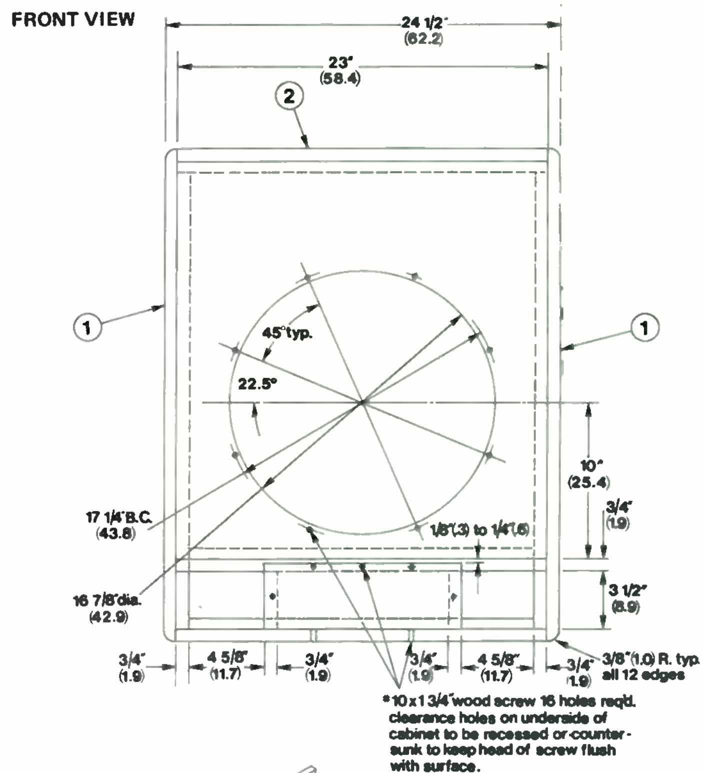
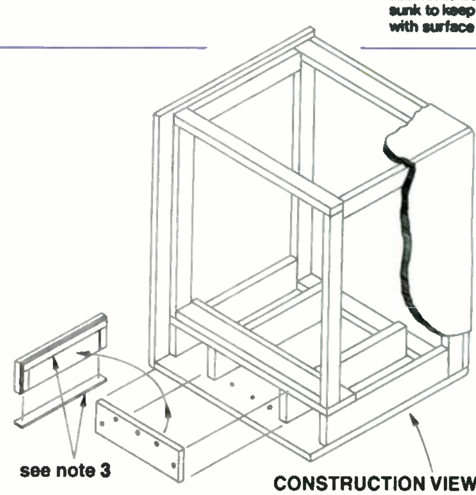
Vented-box (bass-reflex) design with gross internal volume of 7.1ft^3 for use with the Electro-Voice EVM18B 18″ driver. You can select two different low-frequency response curves by the optional use of a removable port cover. Dimensions are in inches; centimeter dimensions are in parentheses





CONSTRUCTION NOTES:
1. All joints should be securely glued and nailed (or stapled).
2. All joints must be airtight. Seal questionable joints with silicon based caulking compound.
3. Use 1/2" wide weather stripping tape around port cover (item 5) for airtight seal.
4. Speaker must be mounted from front of cabinet.
5. Handles, trunk corners, and furniture glides or casters may be added at builders option.
6. Input connector should be selected by builder and mounted on rear.
7. Grille not shown.
8. Parts listed and dimensioned in chart below must conform to dimensions on drawing for proper cabinet tuning.
9. The builder may select material and dimensional fit for parts not listed in chart.
10. The cleats (items 6) to be flush with rear of baffle (item 4) and run completely back to rear cleats.
11. Line top, both sides, and back with 3" fiberglass insulation. Insulation must not block port opening on inside of cabinet.



PARTS LIST – TL505

ITEM	MATERIAL	SIZE	QTY.	REMARKS
1	3/4" (1.9) Plywood or particle board	21-1/2" (54.6) x 30-1/2" (77.5)	2	Sides
2		23" (58.4) x 21-1/2" (54.6)	2	Top and bottom
3		23" (58.4) x 29" (73.7)	1	Back
4		23" (58.4) x 24-3/4" (62.9)	1	Baffle
5		4-1/16" (10.3) x 12-1/4" (31.1)	1	Port cover
6	3/4" (1.9) x 3-1/2" (8.9) Fir	See Note 10	4	Cleats
7	3/4" (1.9) x 4" (10.2) Fir	23" (58.4)	1	Port top

Reader Service #19 →

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Parts Express Speaker Kits

Each kit includes the loudspeaker drivers, unfinished cabinet plus grill frame and 1 yard black grill cloth, grill guide kit, pre-routed baffle board, pre-cut internal brace, port tube, acoustic foam, premium air core inductors, Solen polypropylene capacitors, crossover PC board, internal wiring, terminal cup, F-disconnects, black oxide screws, and cabinet/crossover assembly instructions. Each unfinished cabinet features 1-3/4" thick medium density fiberboard front baffle(!), 3/4" thick MDF shell, oak laminate veneer finish, and solid oak rounded corners. To complete the project requires the following steps: ◆Assemble and solder crossover components to the printed circuit board ◆Cut hole for terminal cup ◆Glue in internal brace ◆Mount crossover PC board to cabinet wall ◆Install internal wiring ◆Cut acoustic foam and put in place ◆Glue in baffle board ◆Finish cabinets to your preference ◆Install port tube and terminal cup ◆Wire and mount the drivers ◆And that's it; you're now ready to enjoy your new speaker system! *Note: Basic woodworking and soldering skills are recommended.*

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2 Way, 5" Morel System

Perfect for smaller rooms or for use in a home theatre system, this loudspeaker is housed in a diminutive .22 cubic ft. enclosure. It offers the soundstaging and superb imaging reminiscent of the classic British mini-monitors. Combines the impressive Morel MW 142, 5" woofer and MDT 20, 1" soft dome tweeter. The MW 142 features a huge 3" voice coil for very low distortion and exceptional control. The port tube is mounted on the rear panel (port hole is *not* pre-cut). The crossover features 6 components; one 16 gauge CFAC air core inductor, one 14 gauge air core inductor, two Solen polypropylene capacitors, and two wirewound resistors. Frequency response: 63-20,000 Hz (+/-3 dB). SPL: 85 dB w/2.83V @ 1 meter. Crossover frequency: 2,400 Hz. Impedance: 8 ohms. Power handling: 150 watts RMS. Dimensions: 12" H x 8" W x 8" D. Net weight: 15 lbs.

#SB-300-750 .... $184.50 EACH

2 Way, Dual 5" Vifa System

Dual 5" woofers mounted in the popular D'Appolito configuration allows this system to produce a superior vertical frequency response as well as lowered distortion in the bass frequencies. It uses the Vifa D27TG-05-06, 1" silk dome tweeter for extended highs and the P13WG-00-08, 5" woofers for neutral midrange reproduction and good bass definition. The six component crossover network includes one 16 gauge CFAC air core inductor, one 14 gauge air core inductor, two Solen polypropylene capacitors, and two wirewound resistors. The system is contained in a .70 cubic ft. enclosure. Frequency response: 55-22,000 Hz (+/- 3 dB). SPL: 89 dB w/2.83V @ 1m. Crossover frequency: 3,000 Hz. Impedance: 4 ohms. Power handling: 60 watts RMS. Dimensions: 23" H x 9" W x 10" D. Net weight: 28 lbs.

#SB-300-760 .... $199.90 EACH

Dayton Loudspeaker Co.® *Cabinets*

Why Our Cabinets Are Better

- *1-3/4" MDF Baffle Board: Since the front baffle receives the highest amount of mechanical energy, reducing this energy by utilizing 1-3/4" thick medium density fiberboard material is critical to ensuring low levels of coloration.*
- *3/4" MDF Shell Medium density fiberboard outperforms conventional particleboard in strength and vibration damping ability.*
- *Bracing To further reduce panel resonance's, all of our cabinets include a 3/4" MDF "shelf" brace.*
- *Superior Joinery Dado and slot joinery are used throughout to provide incredible strength.*
- *Oak Laminate Veneer High quality, real oak laminate veneer can be stained and finished to your liking.*
- *Solid Oak Corners Solid oak corner quarter rounds compliment the oak laminate veneer for beautiful appearance and added strength.*

Each speaker cabinet is shipped unfinished and is suitable for staining and/or painting. Includes grill frame (fabric not included). Pre-cut internal brace and front baffle board must be glued in using carpenters glue, not included. Basic woodworking and finishing skills are highly recommended.

Part #	Description	Price
SB-300-700	.22 Cubic ft. bookshelf cabinet	$40.50
SB-300-705	.46 Cubic ft. bookshelf cabinet	45.50
SB-300-710	.70 Cubic ft. cabinet	65.50
SB-300-715	.96 Cubic ft. cabinet	71.50
SB-300-720	1.55 Cubic ft. cabinet	75.50
SB-300-725	2.70 Cubic ft. subwoofer cabinet	91.50
SB-300-730	3.04 Cubic ft. tower cabinet	132.00
SB-300-735	3.29 Cubic ft cabinet	139.95

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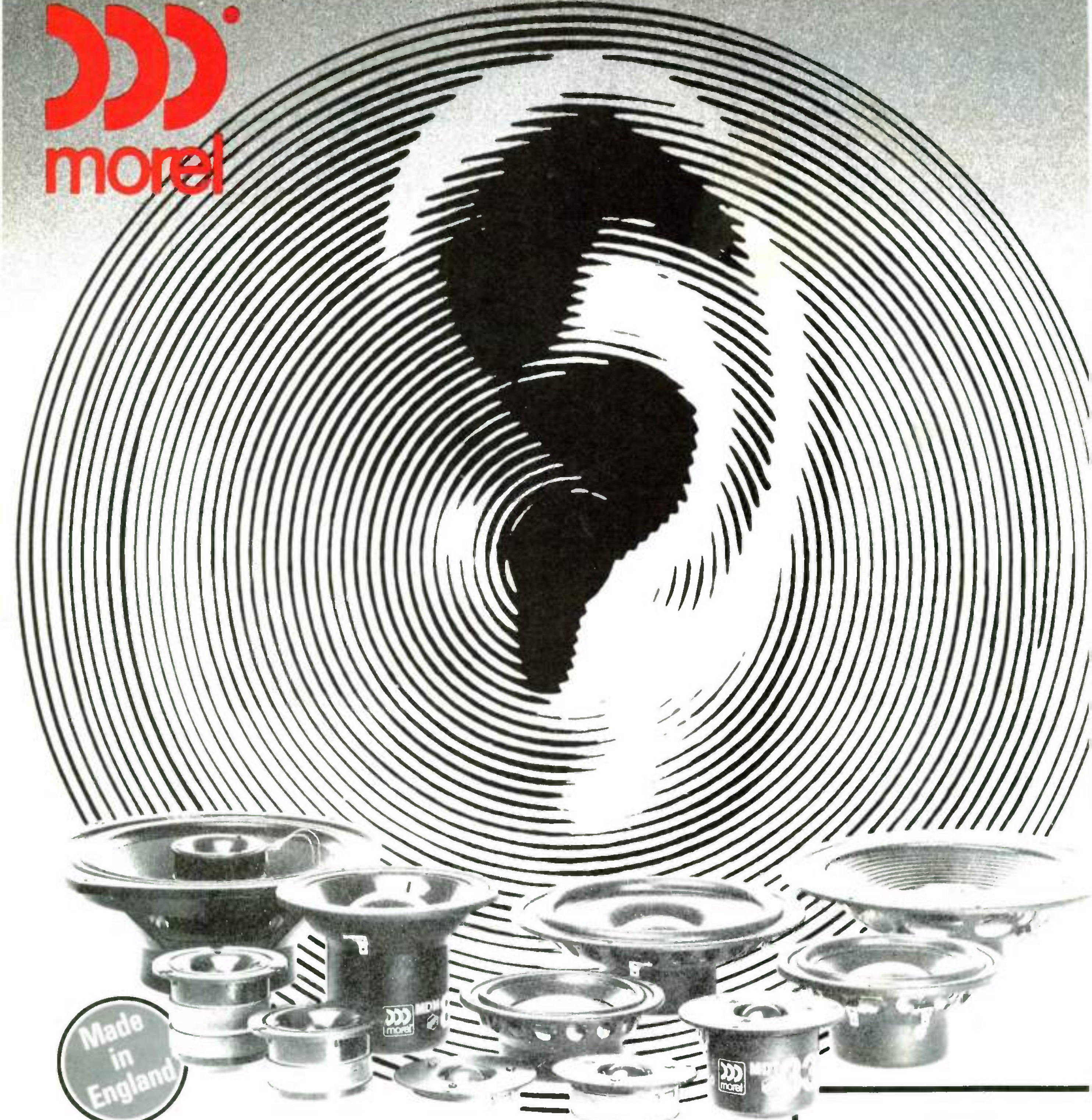

Typical Double Magnet Woofer Cross Section