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2001 aX Index

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Editorial

Audio Evangelism

How did you discover that there was a better way to listen to music than a boombox on your shoulder, or from a table radio?

Think back to your first experience. For me it was a neighbor who had a better-than-average system—which he'd built himself, fortunately. I say fortunately because I was a poor student at the time, and doing it myself—at least partially—was the only viable path to improved sound. I suspect that missionary work, intended or otherwise, is the most viable method for spreading the good word on good sound. Which is probably why so many U.S. homes are still without anything better than an average-quality reproduction.

One of the early industry efforts to spread the word about good sound was the early years of the Audio Engineering Society annual meetings in New York, where consumer goods which ordinary mortals could afford were showcased by all manner of manufacturers. That began to change when the money began to flow toward the pro interests. This affected many companies who discovered that the big concerts of groups such as the Beatles required audio on a different scale altogether. Whereas companies such as JBL, Electro-Voice, Altec-Lansing, and James Lansing, which had been offering kits for home builders in the early '60s, were suddenly much more interested in the big money being made in the pro field.

By the mid-70s the AES shows were becoming more and more dedicated to professionals, with shrinking participation by consumer companies. Recent shows have had renewed consumer representation for the home theater market, but that may be an anomaly.

Better-quality sound was presented to attendees at shows in the '50s and '60s by English manufacturers such as Wharfedale and Acoustical Manufacturing (Quad). In this country the simple technique of putting a sound room on the mezzanine at Grand Central station in New York gave many their first taste of "hi-fi." The shows by *Stereophile* magazine have migrated to "high end" participants, representing very small niche manufacturers, at very high prices. The big annual Consumer Electronics shows in Las Vegas certainly have representation from all types of audio manufacturers, but attendance from a real consumer point of view is relatively small, since most are professionals looking at the wares of other professionals.

Thus, for over 20 years good audio reproduction for the average homeowner has had no national showcase—or regional ones—which might interest middle America in having a good sound system in the home. The closest we have come to such a reach toward the consumer is the mall, and its use by some manufacturers. Cambridge Soundworks®, still largely represented in the northeast of the country, does offer better than average listening environments and affordable product lines. I am sure there are many dealers scattered throughout the country which offer similar service.

What would happen, however, if the audio industry got together to put 50 listening rooms in the largest metropolitan centers of the U.S.? Not to sell goods, but just to sensitize listeners to how good sound can be, with a list of internet addresses of the participating companies. The effort could well overcome many of the prejudices and misconceptions that have arisen that hi-fi is only for the rich, or people with superior ears. The most convincing and compelling advertisement for good audio is extremely well-reproduced sound. Music is a powerful and potent attraction to all humans.

While awaiting any such utopian dream, each of us might do a little bit toward spreading the gospel of good sound to our friends. It may not be much, but the effort doesn't require sermons, or argument—just the sound itself.—**E.T.D.** ❖

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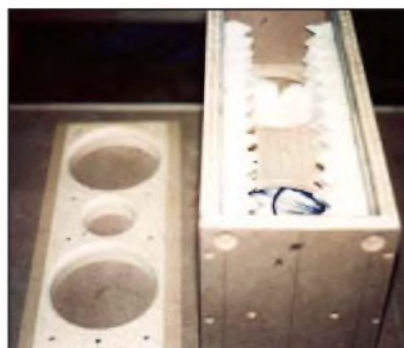

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.

JOHN STUART MILL

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Passive Phono Preamps—A Universal Network

Here's information you need to design phono preamp equalization to enjoy listening to your 78s and old LPs—RIAA and otherwise.

By Paul J. Stamler

Twenty years after the introduction of compact discs, audiophiles and music lovers still play a remarkable number of phonograph records. Phono preamp designs pop up—in *audioXpress* and elsewhere—and many of them use the venerable passive EQ circuit to compensate for the RIAA pre-emphasis included on most LPs.

Unfortunately, many of the EQ networks published are inaccurate, seemingly the result of cut-and-try methods. This isn't necessary; formulas for accurately designing passive EQ networks have been available since 1979, when Stanley P. Lipshitz published his classic article, "On RIAA Equalization Networks"[1].

Equally unfortunate, no one has published network values for non-RIAA equalization curves, although Lipshitz's equations make it simple to do so. Those of us who play 78 rpm records know the bewildering assortment of curves used in their cutting, and fans of pre-1955 non-RIAA LPs are in similar straits.

Old LPs and 78s are fun. Since computers make calculating networks fairly simple, I decided to produce a table of networks for non-RIAA curves and a good RIAA network while I was at it. Before that, however, I need to address a simple question: why?

WHY PASSIVE?

There are two types of EQ commonly found in phono preamps: active or feedback equalization, and passive equalization. (There are also a few designs that use a combination of active and passive elements.)

In an active circuit (*Fig. 1*), the EQ network is placed in the feedback loop of an amplifier; by varying the amount of feedback at different frequencies, the network produces a response that follows the desired curve. The amplifier can be differential, as shown here, or it can be single-ended, with R_{in} doubling as a cathode or emitter resistor for the first stage. (Examples of the latter design include the Dynaco PAS preamps and the Audio Research SP-3 and SP-6 series.)

The active circuit has several problems. The first is obvious from inspection of the schematic: At high frequencies the impedance seen by the output of the amplifier is very low.

Consider a preamp with a gain of 34dB at 1kHz, not unusual for preamps designed to be used with moving-magnet cartridges. At 20kHz, the impedance of the feedback network (R_{in} plus the reactance of the capacitors) will be 4.7 × R_{in}. (This doesn't take into account loading from subsequent stages.)

Since R_{in} is typically low (to minimize noise), the total impedance will also be low; for R_{in} = 1k (Dynaco PAS), the impedance at 20kHz will be 4k7, which is a very low impedance for a 12AX7 to drive. The impedance drops as frequency increases; modern cartridges go out to 40kHz and beyond, and there seems to be significant audio out there on many recordings, so the tube will be working quite hard. Solid-state circuits have it easier, if they're generously biased, but it's still a dubious idea for an amplifier to be driving what is essentially a capacitative load at high frequencies.

OPEN AND CLOSED

In a feedback circuit whose amplifier has infinite open-loop gain, the closed-loop gain will be:

$$\text{Gain} = (Z_{\text{feedback}}/Z_{\text{in}}) + 1 \qquad [1]$$

Real circuits, however, don't have infinite gain, and if the feedback factor is small, the actual gain will be somewhat less than this equation predicts.

IC op amps have a gain that falls off above a few hundred hertz, usually rolling off at 6dB/octave in the audio region. Happily, this is similar to the frequency response of a phono preamp, so the amount of feedback remains roughly the same across the frequency spectrum, except for the shelved region at mid-frequencies. If the IC's gain at audio frequencies is high enough, the circuit will behave properly, and the closed-loop gain will be predictable and stable.

Tubes are another story. Most tubed phono circuits have flat open-loop gain across the audio spectrum—for the PAS design, about 63dB. Since midband gain is 40dB, gain below 50Hz is about 60dB. At bass frequencies, therefore, the preamp is operating with only about 3dB of feedback.

David Hafler took this into account when designing the feedback network, so the frequency response is more or less correct, but the low feedback factor can produce higher distortion in the bass, leading to the stereotypical "mushy bass" associated with the unmodified PAS. Furthermore, the open-loop gain is not stable over time. As tubes age, their gain decreases, and the frequency response of the preamp changes—the bass becomes thinner, but mushier. Not good.

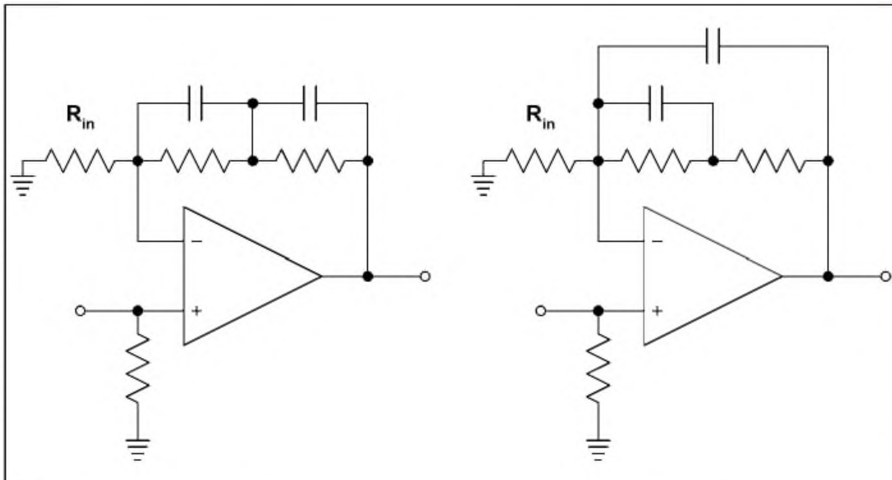

FIGURE 1: Actively equalized phono preamp circuit—two popular configurations.

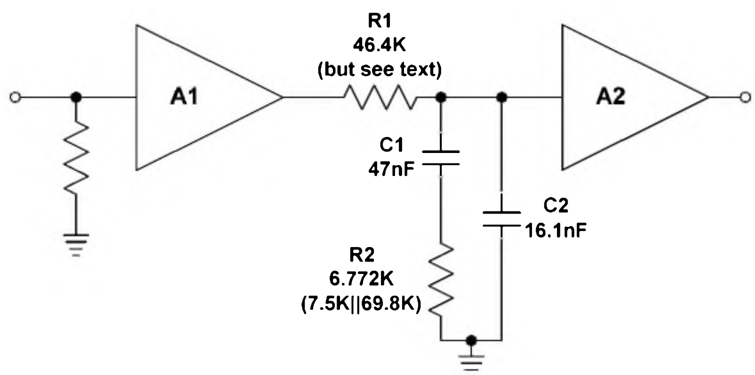

FIGURE 2: Passively equalized phono preamp circuit; with the values shown, this yields the RIAA curve. For typical moving-magnet cartridges, the two amplifiers should have a total gain of about 60dB. See text for discussion of R1's value.

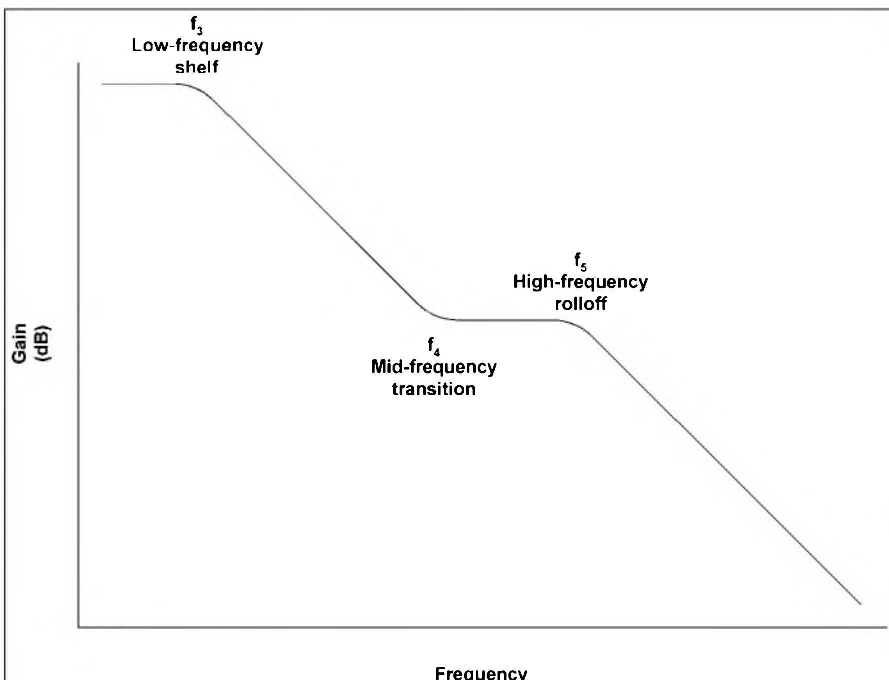

FIGURE 3: Generalized phono equalization curve.

THE PLATEAU

It's clear from Equation 1 that non-inverting feedback amplifiers have a minimum gain of one (at least in the audio regions). This causes an error in phono preamps. Instead of continuing downward forever, the frequency response flattens out somewhere in the 50–100kHz region unless corrected by an external network.

By increasing the gain of a feedback phono preamp circuit, the plateau is pushed farther from the audio band. Unfortunately, increasing the gain decreases the feedback at bass frequencies (see previous).

INPUT PROBLEMS

Consider the Miller effect. The inherent capacitance of an input device—be it tube, FET, or bipolar transistor—is effectively multiplied by its gain. In a feedback amplifier, the closed-loop gain multiplies the input capacitance.

Unfortunately, if the closed-loop gain varies at different frequencies, Miller-effect multiplication also varies. This can play hob with the response of the cartridge, which wants to be terminated by a constant resistance and capacitance. In effect, the feedback loop becomes part of the cartridge's load impedance, sometimes with spectacularly flaky results.

ENTER PASSIVE

How does the passive preamp meet these challenges? Consider *Fig. 2*, drawn with an RIAA compensating network. Cursory inspection will show that, even at extremely high frequencies where the capacitors are effectively short circuits, the minimum impedance will be set by R1, which is nice and high.

The amplifiers, if they employ feedback, don't get anywhere near open-loop operation at audio frequencies. In the case of tubes and discrete solid-state circuits, feedback is approximately constant across the audio band.

What about long-term stability? As devices age in the two amplifiers of *Fig. 2*, the overall gain may change a bit. Frequency response, however, should remain constant, as it's determined almost entirely by the passive network. ("Almost?" Yes. See the discussion under "Subtleties," which follows.)

The passive circuit's response is essentially dictated entirely by the network itself, not the amplifier's characteristics, so it's easier to calculate nonstandard responses. Because the passive network's response is not limited by a non-inverting amplifier's gain equation, the high frequencies keep on rolling down as they should, way past the audio region.

If the input stage employs a low-impedance output, Miller-effect multiplication should be constant across the audio spectrum. The cartridge will see what it wishes to see. Even without a low-impedance output (if, say, A1 is a simple triode stage), the effect on cartridge loading will be minimal.

Are there drawbacks to passive preamp designs? Yes, there are: noise and input overload. With active equalization, a phono preamplifier has an effective noise bandwidth of about 3.3kHz (RIAA curve). A passive pre-

TABLE 1
PARALLEL COMPONENTS FOR PASSIVE PHONO EQUALIZER

HIGH-FREQUENCY ROLLOFF

FREQUENCY SETTING (HZ)	COMMENTS	C2 COMPOSITE (NF)	C2A PARALLEL CAPS (NF)
1580*	U.S. Decca/Folkways/Elektra, early Col. LP	32.9	22 + 9.1
1580	U.S. Columbia, King, late Brunswick	25	22 + 1.2
2122	RIAA	17.2	15 + ~ 400pF
2500	Capitol, Mercury, MGM	14.2	10 + 2.4
3180		10.7	5.6 + 3.3
3180**	RCA, Victor, Bluebird	26.3	22 + 1.0 + 1.5
3750	Odeon, OKeh, Parlophone, Polydor	9.0	3.6 + 3.6
3750***	Early Brunswick	See 1580	
4000		8.3	4.7 +1.8
5600	Eng. Decca, London ffrr, Eur. Columbia	6.0	2.0 + 2.2
7500	Acoustic	4.5	2.7
10,000	Acoustic	3.4	1.6
17,000	Acoustic, early Eur. 78s	1.8	None

LOW-FREQUENCY TURNOVERS

FREQUENCY SETTING (HZ)	COMMENTS	C1 COMPOSITE (NF)	C1A PARALLEL CAPS (NF)	R2 COMPOSITE (KOhms)	R2A PARALLEL RESISTOR (KOhms)
Flat	Acoustic	None	Jumper	5.302	32.4
20/200		151	100 + 16 + 2.0	5.302	32.4
25/250	Eng. Decca, London ffrr, early Eur. 78s	117	51 + 33	5.494	41.2
30/300	Columbia 78, Disc, late Brunswick, most Eur. 78s, Velvet Tone	95	47 + 15	5.593	47.5
40/400	Capitol, Mercury	66	33	5.979	105
50/500	RIAA, King	49.9	10 + 3.3 + 3.6	6.34	None
45/630	Folkways/Elektra	45	12	5.494	41.2
50/750	U. S. Decca, early Columbia LP	34.5	1.5	6.199	280
50/800	RCA, Victor, Bluebird, HMV	74.2	33 + 8.2	2.679	4.64
50/1000	Early Brunswick	64.3	27 + 4.3	2.4601	4.02

FOR THE ULTRA-FINICKY

In the main body of this article I've talked about the effects of amplifier output impedance and input capacitance on passive RIAA networks. Here's how to compensate for them.

First, begin by using clip-leads to connect up the amplifiers as shown in *Fig. 6*: no EQ network, and the output of A1 directly connected to the input of A2. That output is also loaded by two parallel 20k resistors, running to ground. If you're using an AC coupling network (*Fig. 5*), include the resistor Rc as well as the two 20k resistors.

Set your signal generator for 2kHz (or use the 2kHz band of a test CD), and adjust its level until 1V appears at the output of A2. (You may need to use an attenuator at the input of A1.) A digital multimeter will work fine to measure the output voltage V_1 (you can also use it to measure the exact value of the paralleled resistors, R_{par}). Compute i_1:

$$i_1 = V_1/R_{par} \quad [2]$$

So if V_1 is exactly 1V and R_{par} is exactly 10k, i_1 = 100mA.

Now unclip one of the 20k resistors and measure the output voltage (V_2) again across the single resistor R_{sing}; it will have increased slightly. Compute i_2:

$$i_2 = V_2/R_{sing} \quad [3]$$

Let's say V_2 is 1.05V and R_{sing} is exactly 20k; i_2 will then be 52.5mA. From these numbers you can compute the output impedance of A_1:

$$Z_{a1} = (V_2 - V_1)/(i_1 - i_2) \quad [4]$$

In the example, this comes out to about 1053Ω. In an amplifier using our multi-frequency EQ network, this would suggest using 43k147 for R1, rather than 44k2. This is right handy, as 43k2 is an E24-series preferred value.

Now for A2's capacitance. I'll describe the procedure for the multi-frequency network. You'll need a signal generator that you can set to 200Hz or 2kHz, or a test CD that has those bands. You'll also need an analog audio voltmeter (or a DAT recorder with a digital headroom readout); most digital multimeters have very poor frequency response.

Connect as shown in *Fig. 7*. Don't use clip leads this time. Instead, solder the network board to the amplifiers the way you will actually use it. Install R1—corrected for the output impedance of A_1 if necessary—and C_o/R_c if you're using them, but don't install C2 or the rest of the network.

Set the signal generator on 200Hz and adjust its output (or your attenuator if you're using a CD player) so that A2 is producing about −10dBu. (On a DAT machine, set the level control at about 12:00 and adjust the generator so the headroom display reads −5.0dB.) Switch the generator to 2kHz and measure the output again. If it has changed, record the difference in decibels.

This is the fudge factor attributable to your test setup—generator imperfections, meter irregularities, or whatever. (If the amplifiers are designed properly, the error won't be coming from them.) Let's say, for purposes of discussion, that the setup produces a −0.3dB change at 2kHz, relative to 200Hz.

Switch the generator back to 200Hz and temporarily wire in C2, 1.8nF. (It helps to have a couple of pins on the network board to wrap capacitor leads around.) Measure the frequency response, which should be down 3dB at 2kHz (well, 2.97dB if you really want to nitpick), plus the fudge factor. So, in our f'rinstance, the response should be down 3.3dB when switching from 200Hz to 2kHz.

If there's appreciable capacitance at the input of A2, however, the 2kHz response will be down too far. In the example, with 100pF of input and stray capacitance, the response will be down 3.5dB rather than 3.3dB. You need a smaller cap for C2. Change to something smaller (say, 1.5nF) and begin adding parallel capacitors until you get the proper response drop.

Repeat for the other channel (the strays may be slightly different). Now your basic C2 is correct, so go ahead and install the rest of the network, switching, and so on.

What if you're doing only an RIAA network? In that case, you can probably live without compensating for A2's input capacitance. Even if it's 100pF, that should introduce an error of only about .05dB in the treble. Do correct R1 for A1's output impedance, though.—**PS**

amp has two amplifier stages. The first, A1, has a noise bandwidth similar to that of an active preamp; its noise is rolled off by the passive network. A2, however, runs flat across the audio band—it can contribute significant noise to the preamp if not carefully designed.

In a passive preamp, A1 amplifies the cartridge's output with a flat frequency response. Since the cartridge is playing back a recording with pre-emphasized highs, it's putting out a significant amount of high-frequency energy. Then there are scratches: fast spikes with surprisingly high peak levels, all at high frequencies. The first stage of any passive preamp must be carefully designed to avoid overload or high-frequency intermodulation distortion.

With care, however, noise and overload problems can be minimized. Given the major drawbacks inherent in active phono preamps, I think passive is the way to go for most applications—especially if you want to play with non-RIAA curves.

TWEAKING THE CURVES

If you intend to play only RIAA-equalized records (essentially everything after 1955), the network in *Fig. 2* will work fine, as long as the output impedance of A1 is low (see "Subtleties"). A simple pair of low-noise IC op amps will do, or you can use discrete op amps à la Nelson Pass or Erno Borbely, or tube stages, simple or elaborate. The network shouldn't care.

If, on the other hand, you need switchable curves, take a look at *Figs. 3* and *4*. *Figure 3* shows an idealized frequency-response curve. I conform to Lipshitz's conventions by labeling the breakpoints f_3 (low-frequency shelf), f_4 (mid-frequency turnover), and f_5 (high-frequency rolloff). You can adjust these by tweaking the values of C1, C2, and R2, which I do by adding components in parallel with them. R1 remains constant at 44k2, a bit lower than I used in the RIAA-only network.

Unfortunately, the various bits of the network interact with one another; Lipshitz's equations show that twiddling one frequency affects the others. When I set up a spreadsheet into which I fed frequencies while the computer cranked out network values, I discovered it didn't work very well. The interactions made designing a switchable, buildable network difficult.

Instead, I produced a spreadsheet into which I plugged network values and the computer figured out frequency breakpoints using the equations in the "R3 = 0" columns of Lipshitz's Table 1(c). I discovered that by tweaking R2 along with C1 I could make the curves come out right, within 0.1dB of their ideal shapes. *Figure 4* shows the switchable network; *Table 1* lists component values to connect in parallel with the basic network. All components are standard E24 (capacitors) or E96 (resistors) values.

Please understand that the settings suggested for various labels, as shown in the "Comments" column of *Table 1*, are only suggested starting points based on published information and some informed guesswork. Actual recording curves varied wildly over the years, and the final decision on which EQ settings are correct belongs to your ears alone.

Some recordings have curves that are quirky enough to require special networks. So, for example, there are two switch settings for a high-frequency rolloff at 1580Hz: one for American Decca 78s and Folkways, Elektra, and early Columbia LPs, one for American Columbia, King, and Brunswick 78s. (By a stroke of happy fortune, the same network value generates 1580Hz when the midrange turnover frequency is set for 300Hz—late Brunswicks—but 3750Hz when the midrange is at 1000Hz—early Brunswicks.) Similarly, there is a 3180Hz rolloff setting for the RCA/Victor/Bluebird family of 78s, and a different one for other discs that use this frequency.

TABLE 2

COLUMN	HEADING	FORMULA
a2	EQ Name	[enter name]
b2	C1 (nF)	[enter C1]
c2	C2 (nF)	[enter C2]
d2	R1 (K)	[enter R1]
e2	R2 (K)	[enter R2]
f2	Real C1	=b2/1000000000
g2	Real C2	=c2/1000000000
h2	Real R1	=d2*1000
i2	Real R2	=e2*1000
j2	formula 1	=(h2*(f2+g2))+(i2*f2)
k2	formula 2	=((j2*j2)–(4*h2*i2*f2*g2))^0.5
L2	T3	=(j2+k2)/2
m2	T4	=i2*f2
n2	T5	=(j2-k2)/2
o2	f3	=1/(L2*2*3.14159)
p2	f4	=1/(m2*2*3.14159)
q2	f5	=1/(n2*2*3.14159)

These formulas were written for the spreadsheet program in Microsoft Works. If you plug them into a different program, you may need to change the syntax to fit that program's preferences.

I've included a couple of settings for playing acoustic 78s. These should ideally be reproduced flat. However, as there is virtually no signal present in the high frequencies, I've given you the option of rolling these off at 7.5, 10, or 17kHz to reduce noise. The latter setting, effectively flat for real-world recordings, is also appropriate for early European electrical 78s, which used no high-frequency pre-emphasis.

You don't need to include all possible combinations, of course. Use as many as you please, given the number of switch positions you have available (or how many you feel like soldering).

The spreadsheet runs in Microsoft Works. If you'd like to design your own networks, you can download it from http://members.aol.com/beansnicker/multieq.wks. Microsoft Works spreadsheets won't usually run on other programs. To plug the formulas into your own software, you need the text file containing the cell entries at http://members.aol.com/beansnicker/multieq.txt (*or see Table 2*). Note that these formulas use the Works syntax, which you may need to edit slightly to fit your program's language. (BeanSnicker? That's my small record label, which reissues 78s.)

SUBTLETIES

I mentioned before that you could pop these networks in between just about any low-noise amplifiers with enough headroom to avoid overload. In fact, you can use just about any quiet, fast IC op amps without further ado. If, however, you choose to use something fancier, such as a tubed stage or a discrete transistor circuit, you must consider some subtleties.

The first is the output impedance of A1. A quick glance at the schematics will tell you that A1's output impedance is in series with R1. Properly speaking, it constitutes part of R1.

With most ICs this is no big deal. Within the audio band, their output impedance is typically an ohm or two, a negligible fraction of R1. In discrete circuits, however—especially with tubes—this impedance can make a difference.

A tubed stage whose output is a cathode follower has a fairly low output impedance. A 6SL7/12AU7 stage I've used runs about 500Ω. This won't affect the curves much, producing an error of a bit less than 0.1dB below 50Hz in the RIAA curve. The ultra-finicky, of course, will prefer to compensate by lowering the value of R1 a like amount.

When describing high-frequency curves, some sources list the time-constant of the rolloff point rather than its frequency; others describe how many decibels the signal is attenuated at 10kHz. Here's a table of conversions:

F5	dB at 10k	TC (µs)
1,580	−16	101
2,122	−14	75
2,500	−12	64
3,180	−10	50
3,750	−9	42
4,000	−9	40
5,600	−6	28
7,500	−4	21
10,000	−3	16
17,000	−1	9

Amplifier stages without followers are a whole 'nother ball game. Take a pair of 12AX7 common-cathode sections in parallel, as described in a recent *audioXpress* article[2]. I calculate this amplifier's output impedance to be about 28k5. This would require dropping R1 to 15k7, which is pretty low.

If I were using these tubes as my input stage, without a follower, I'd scale up the networks' resistances and scale the capacitances down by the same factor. If you double all the resistances, for example, R1 ideally would be 88k4. Subtracting the output impedance of the tubes leaves you with 59k9 as the actual resistor, a more reasonable figure.

WAIFS AND STRAYS

Another problem develops at the input of A2: capacitance. Again, IC op amps present little problem; even FET-input units have only a few pF at the input. Discrete FET circuits and tubes, however, can present more of a problem. A 6SL7, for example, has a grid-to-plate capacitance of 2.8pF. Multiply this by the stage gain, add the grid-to-cathode capacitance and some strays in the wiring, and you're up to 100pF.

In the RIAA circuit this won't make much difference (it's only about 0.3% of C2), but it will shift the 17kHz rolloff for acoustic records down to 16kHz. The truly finicky may again wish to compensate by using a smaller capacitor for C2, trimming it up so that the total (including A2 and strays) equals 1.8nF. (See the sidebar for suggested trimming procedures.)

AC AND DC

Finally, we've assumed up to this point that the preamp circuit is DC-coupled. This is certainly feasible with ICs (especially if you use a servo in the feedback loop to null out offset voltages). It may not be possible to do that with all discrete circuits, however, and it's almost impossible with tubes.

Figure 5 shows the (idealized) network with a coupling capacitor added, and a resistor to provide the input device of A2 a path to ground. Straightforward enough, but there are again subtleties.

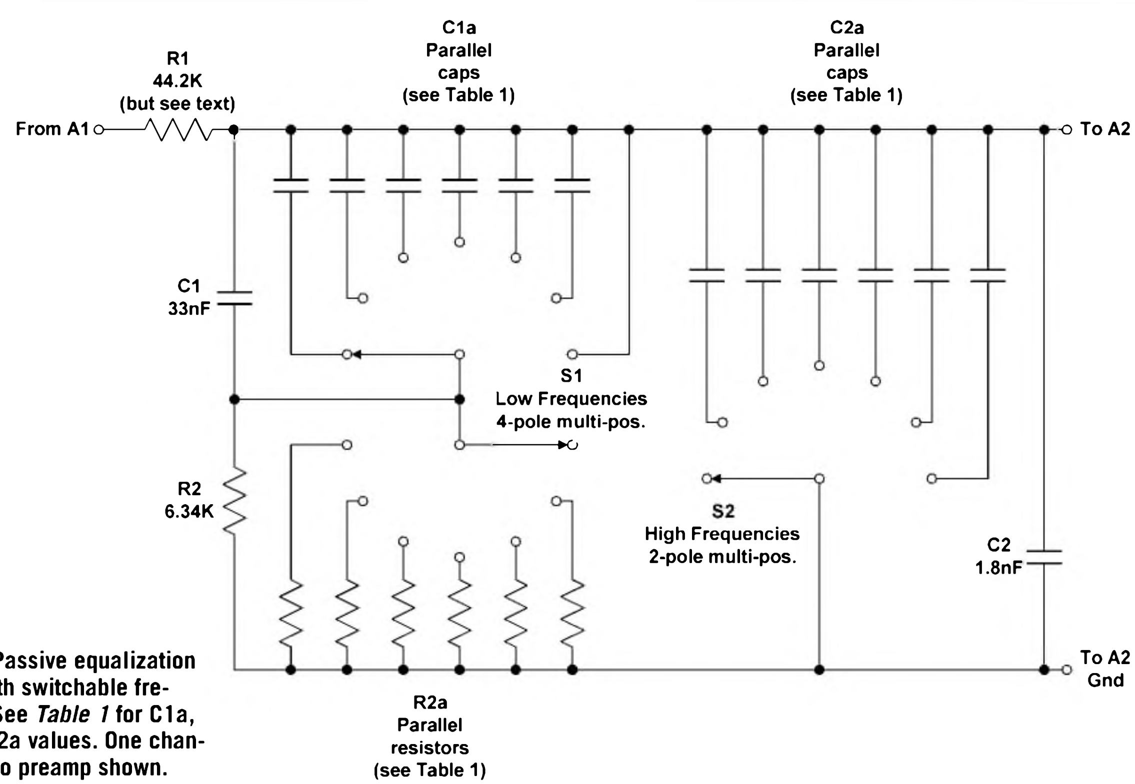

FIGURE 4: Passive equalization network with switchable frequencies. See *Table 1* for C1a, C2a, and R2a values. One channel of stereo preamp shown.

The first is that the coupling resistor R_c affects impedances in both directions. If, as shown, R_c = 200k, that value will appear in parallel with the amplifier's output impedance, lowering it. Still's paralleled-12AX7 circuit, for example, would now have an output impedance of 24k9. Follower output won't be noticeably affected. A cathode follower with an output impedance of 500Ω would drop to 499Ω, a negligible change.

Simultaneously, R_c appears in parallel with the load. In the tweakable network, for example, the minimum load at high frequencies now becomes 36k2 rather than the 44k2 of the network with DC coupling. You should take this into account when figuring the required drive current from A1.

Finally, a coupling network can cause problems if its time constant is too close to the audio band; the coupling capacitor's reactance interacts with the equalization network. If, for example, you include a subsonic filter in the design, you should not make the RC network part of it. Setting the bass rolloff frequency to 12Hz (as part of a third-order 16Hz Bessel filter) results in significantly reduced output at audible frequencies. Instead, you should place the single-pole filter elsewhere in the preamp, either at the output of A2 or, if you want to minimize warp frequencies in the active circuits, at the input

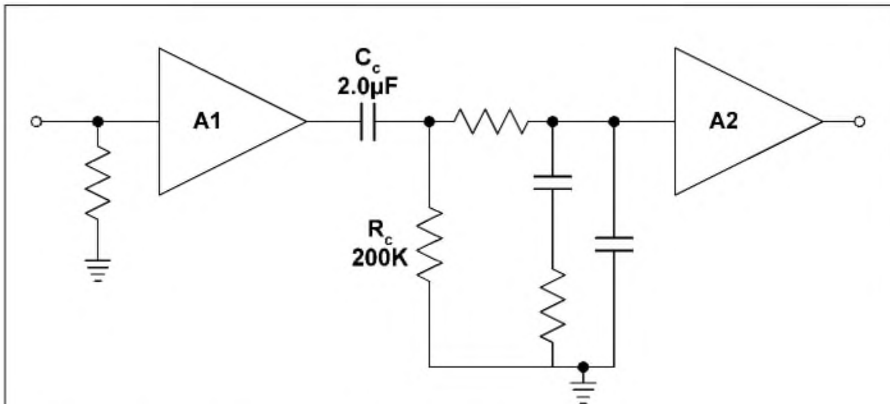

FIGURE 5: Passive preamp circuit with AC-coupled amplifier.

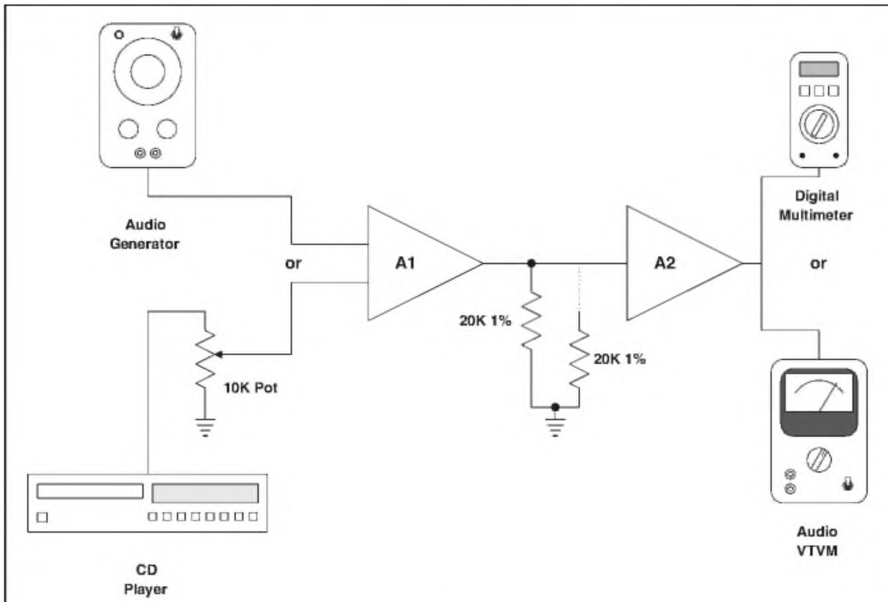

FIGURE 6: Setup for measuring output impedance of A1. The dashed line denotes a clip-lead that is removed during the test.

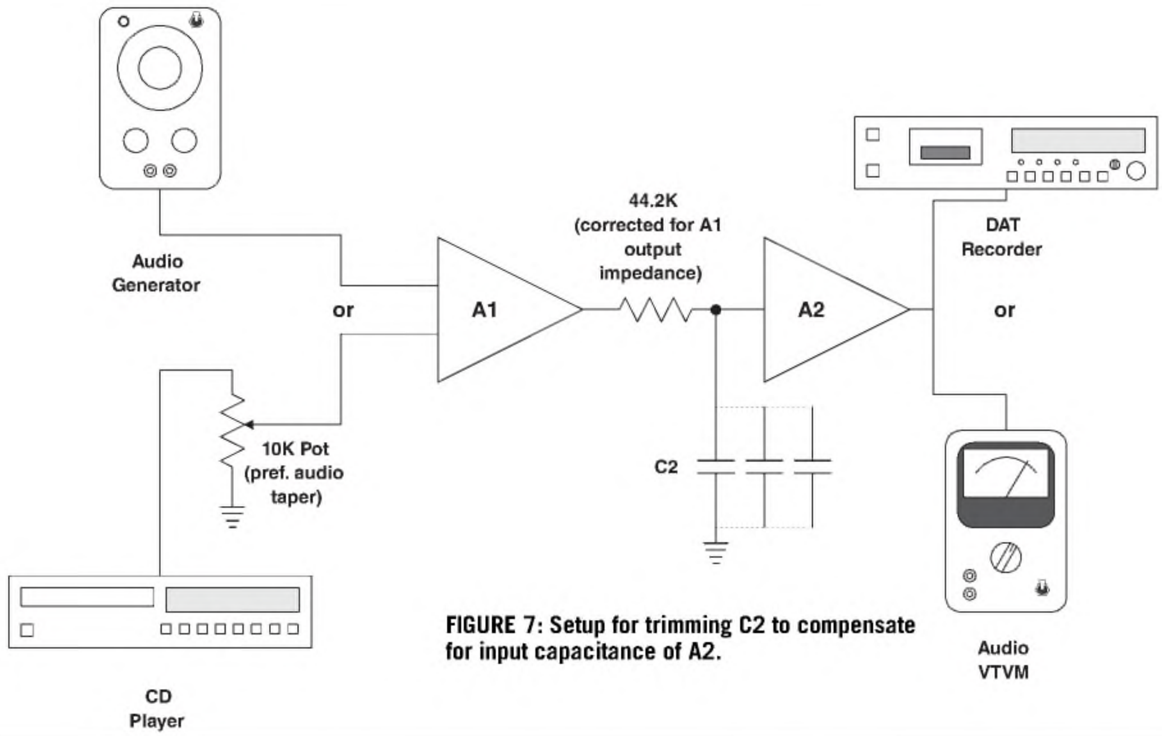

FIGURE 7: Setup for trimming C2 to compensate for input capacitance of A2.

of A1. The value shown generates a rolloff frequency of 0.4Hz, far enough from the audio band to minimize interactions.

WRAPPING UP

In this article, I've given you the information to design a passive phono EQ network for virtually any curve employed during the pre-RIAA glory days of disc recording. There's a wealth of lovely music out there to discover, and I guarantee you it'll sound better if it has the right equalization.

As I've emphasized, these networks will work with just about any type of amplifiers, from IC chips to ultra-fancy tube circuits, provided you take cognizance of the amplifiers' own characteristics and trim up the network accordingly. But what of the constraints on the amplifiers themselves—noise and overload?

Ah, that's another article entirely. Stay tuned! ❖

REFERENCES

1. Lipshitz, Stanley P., "On RIAA Equalization Networks," *Journal* of the Audio Engineering Society 27:6 (June 1979), p. 458.

2. Still, Joseph Norwood, "High-Quality ($180) Control Unit," March 2001 *audioXpress*, p. 36.

A Great First Amplifier Project

Here is an excellent beginner's project for those just getting started in this hobby.

By Rick Spencer

PHOTO 1: Completed amp on Hammond chassis box—painted with black "wrinkle coat."

This "little" amp is easy to construct and has only about 25 parts in each channel. The system transformers are very inexpensive, and the total cost involved for a complete stereo version should be very minimal! Most of the parts used in this project are available from Antique Electronic Supply. The tubes involved are the least costly of most name brands that I could find and yet they will still deliver the sound that vacuum tubes are noted for.

The voltage amplifier and phase inverter tube is a 12SL7, and the outputs are 12V6s. These tubes have the same electrical characteristics as their counterparts—the 6SL7 and the 6V6. The only difference, of course, is the heater voltage. With the type of transformer used for the power supply, and with the help of one small 12.6V transformer, the tubes will be supplied with all of the necessary voltages to operate the amplifier.

Again, as in my 6550 single-ended amp project (Sept. '01 *audioXpress*), this article does not contain any complex math. It gives you only the information you need to complete the amplifier. No special or "high end" components are used because I don't believe that a beginner in this hobby should suffer "sticker shock" when starting his/her first project.

If you wish to experiment and try different capacitors, resistors, or other components, in order to obtain a better sound, please feel free. After all, this hobby is all about having it your way. The amplifier (*Photo 1*), when constructed with the parts listed, gives a warm and full range of sound and will be pleasing to most new hobbyists—especially those of you used to listening to small solid-state equipment. It has the kind of sound that tubes are famous for and gives an output of just around 12W of power per channel. This is actually more than the famous single-ended 300B will deliver, even though the 300B is in a class all its own. The amp will drive most speakers with an efficiency of 90dB or more to a room-filling volume.

For a first-time project and introduction to the types of circuits involved with tube audio, the results should be very rewarding indeed. If you follow the schematic diagram (*Fig. 1*) and use careful construction techniques, the amp will work when you first power it up and should give you years of listening enjoyment. It will also be very easy to service and repair in the future because you built it yourself!

CIRCUIT DESCRIPTION

The circuit here is very simple and yet is effective for the types of tubes used. These 12V6s operate in push-pull class AB1 with around 285V on the plates and will give you the warm, rich, and classic sound of the amplifiers from many years ago. It was used, in the 6V6 version, in the same types of circuits as the 6Y6 and the EL84.

All of these tubes have been around for a long time and many audio veterans got their start in audio electronics using one or the other of them. So, for a lot of old-timers in this hobby, this circuit should bring back a lot of memories with its pure simplicity and its pleasing sound.

The 12SL7 operates both as a voltage amplifier and a phase inverter, and has a 12.6V heater that draws only about .15A. The 12V6s have a heater rating of 12.6V at a draw of .23A, and this is supplied by the heater transformer, which is adequate for all of the tubes used in this small amplifier. The Hammond part number for this transformer is #167L12.

The heaters use DC voltage to reduce the possibility of any hum that might otherwise be generated from using AC instead. This AC-to-DC conversion is done by the full-wave bridge and then is smoothed by the filter capacitor shown in *Fig. 1*. The wiring hookup for the heaters allows equal voltages to all of the tubes, a technique that has always worked great in all of my circuits.

The switch marked S3 allows the tube

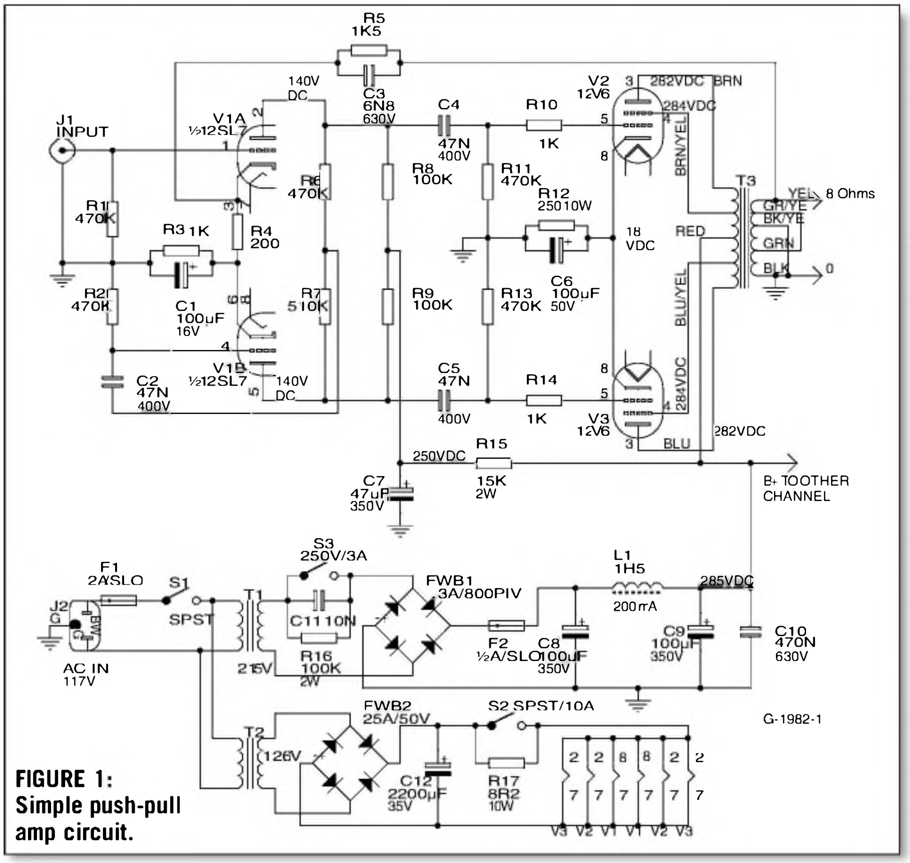

FIGURE 1: Simple push-pull amp circuit.

heaters to warm up before letting the B+ hit the plates and thereby saves the cathodes from the stripping effect that can occur. In this circuit you don't need to use a relay to turn on the B+ because the switch is rated for the voltage which is furnished by the B+ transformer. The resistor and capacitor across the switch points stop any flashover and also keep the "pop" out of the speakers when the switch is turned on.

The circuit uses cathode bias for the output tubes, which helps to keep the amp simple (i.e., no extra power supplies). All of the resistors are ½W rated unless otherwise noted, and all of the pinout numbers are shown on the schematic diagram. For a grounding technique I recommend the "star" method (in other words, connect all of the indicated grounds to one central wire).

I used a length of bare copper wire for this and connected it to chassis ground at only one point right next to the input jacks. You can find the hookup wire used for this project at most hardware stores. It is type TFFN, rated for 600V, and is 18 gauge. Other types of circuit wire are available from Antique Electronic Supply.

I didn't use any special bypass capacitors in the circuit in order to keep the cost down and to keep it simple. The only bypass cap is in the power supply. You may use them, of course, in case you want to try to "tailor" the sound. Remember that the main purpose of this project is to build an amplifier that is inexpensive and very simple, but if you choose to take this or any other project to a higher level of involvement, then have fun! (Actually, some amplifiers are in a constant state of upgrades and other improvements—sound familiar?)

The 12V6 output tubes will really pump out the power without pumping out too much heat. These power tubes were used in a lot of old car radios because their low heater current draw and their reduced plate-current was needed. If you wish to have a small fan blowing fresh air across your tubes, then you will find that they will last even longer. I have small fans circulating air around all of my tube equipment, and because they are rated for 12V and are running on 6V, the noise is almost nonexistent!

CHASSIS LAYOUT

Try to set up the chassis as illustrated in *Photo 2* with the two channels in a kind of "mirror image" of each other. This simplifies most of the wiring sequences. In other words, after you install a part on the left channel, you simply go over to the right channel and repeat the process. If you assemble the chassis components in this manner, you'll avoid making the usual mistakes that can be made when you don't have the aid of a pictorial diagram or layout chart, such as the kind that Heathkit was famous for.

You will notice the "star" ground wires in the photo. They look like a sort of upside down "t." This makes all of the ground connections very convenient and keeps most of the lead lengths to a minimum.

Also, before connecting the leads of the resistors and capacitors, try to insulate them with heat-shrink tubing or other type of insulation. This will help reduce the chance of any shorted wiring, which can cause a lot of damage, and it also serves as a safeguard when testing the chassis for voltages (just in case you slip and misplace a test probe!). You can actually make your test probe much safer to use if you slip a piece of heat-shrink tubing over the bare metal to cover all but the very tip of it. Remember, there is some high voltage on your chassis that can be very unforgiving of mistakes!

The chassis shown in the *Photo 3* was a prototype and a friend of mine liked it so much he wanted me to install it in a case or enclosure. This "sub-chassis" will be in the case with his power supply and the cooling fan when completed. The final chassis box for the amplifier I'm building for myself is a Hammond #H1444-22, which is only 12 × 8 × 2″ and is perfect for this little amp.

The completed unit is quite lightweight, as tube amps go, and won't take up much room on your system rack or shelf. For us Californians who have to worry about electrical usage nowadays, this amplifier draws only about 65W of total power when going full tilt! This makes it very inexpensive to operate.

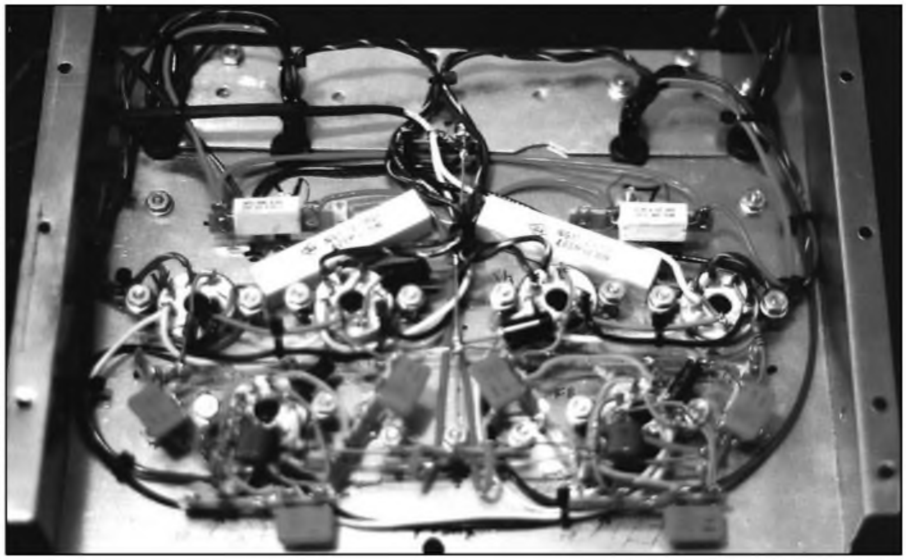

PHOTO 2: Prototype chassis showing "mirror" image channel wiring and component placement.

TESTING TIPS

After you finish wiring the heater circuit, you can test it for proper voltage at all of the tube sockets (*Photo 4*). Do this with the B+ fuse removed from the holder—just for safety sake. You will find that the voltage will be higher than 12.6V because of no load on the transformer.

After you install the tubes, retest the heater voltage again. This time leave the soft-start switch, S2, "off" for a few seconds and then turn it "on." You will notice the voltage will be about half of the normal heater rating, and when you turn the switch "on" it will slowly rise to where it should be. Check all of the heaters for that orange glow they produce to make sure they have all come on.

Switch S2 is a sure, simple way to protect your tubes from that initial surge that occurs when the resistive elements are cold. Even though the tubes in this amp aren't really expensive, it's a good idea to try to make them last as long as possible by treating them gently. When checking all of the other voltage readings on your amp, remember that they will proba-

TABLE 1
PARTS LIST

RESISTORS	SOURCE
R1, R2, R6, R11, R13 = 470k ½W	AES
R3, R10, R14 = 1k ½W	AES
R8, R9 = 100k ½W	AES
R7 = 510k ½W	AES
R4 = 200 ½W	AES
R5 = 1k5 ½W	AES
R12 = 250 10W	AES
R15 = 15k 2W	AES
R16 = 100k 2W	AES
R17 = 8.2 10W	AES
CAPACITORS	
C1 = 100μF 16V	Mouser
C2, C4, C5 = 47nF 400V Wima (MKP 10)	AES
C3 = 6.8nF 630V Mylar or Poly	Mouser
C6 = 100μF 50V	Mouser
C7 = 47μF 350V	AES
C8, C9 = 100μF 350V	AES
C10 = .47 630V Solen	AES
C11 = .01 1kV	Mouser
C12 = 2,200μF 35V	AES
TRANSFORMERS	
T1 = Hammond #261M6	AES
T2 = Hammond #167L12	AES
T3 = Hammond #1608	AES
L1 = Hammond #T156R (choke)	AES
RECTIFIERS	
FWB1 = 3A 800V	Mouser
FWB2 = 25A 50V	AES or Mouser
TUBES	
V1 = 12SL7	AES
V2, V3 = 12V6GT	AES
(Tube sockets, hdwe, and so on, available from Antique Electronic Supply)	
SWITCHES	
S1 = SPST 125V 3A	
S2 = SPST 125V 10A	
S3 = SPST 250V 3A	

Antique Electronic Supply, 480-820-5411, www.tubesandmore.com
Mouser, 800-346-6873, sales@mouser.com

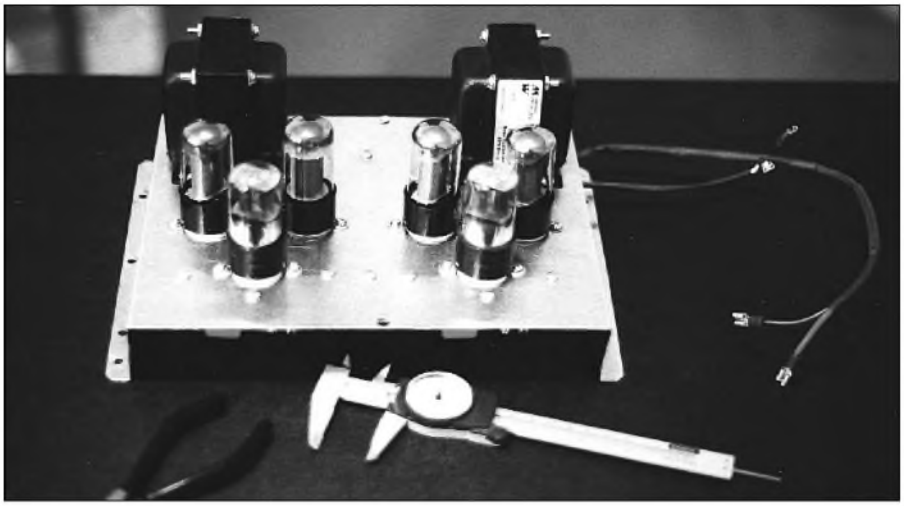

PHOTO 3: Prototype chassis complete with tubes.

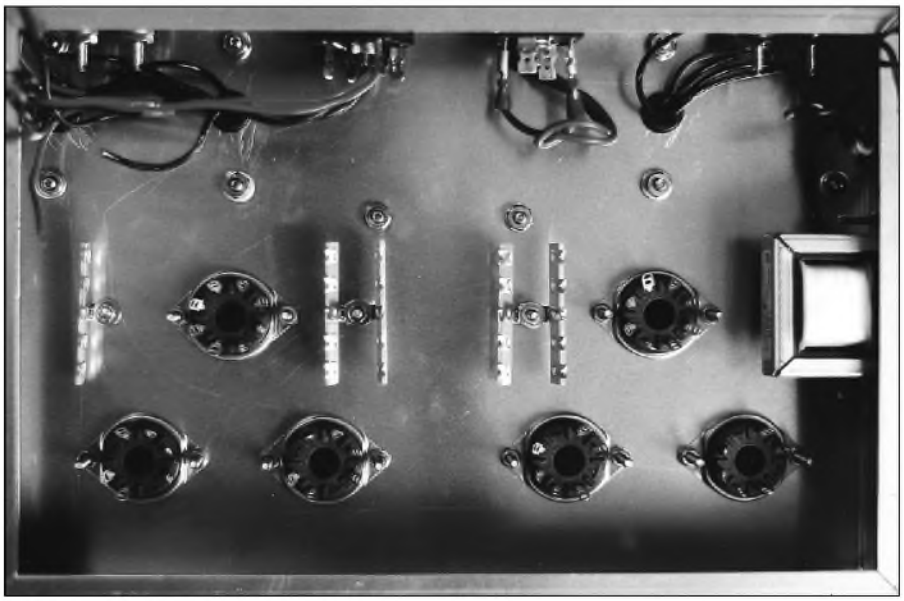

PHOTO 4: Underside of new chassis box with sockets, terminal strips, and power-supply components mounted.

bly be within 10% of the listed parameters. This is usually because of the difference in the line voltage from your utility company.

By the way, when wiring in the Hammond output transformers, use the wire codes shown on the diagram for the primary connections. This should prevent any wild oscillation upon startup. This means that you should avoid the hassle of having to unsolder the leads to the output tube sockets, reverse them, and resolder them again. But in case you do, please remove the tube from the socket first!

One final note about the transformers used here: If you decide to use 6V6s and 6SL7s for your amplifier, the Hammond power transformer listed will supply all of the necessary voltages because it has a 6.3V secondary included. I am using this same transformer, #261M6, along with Hammond transformer #167L12, because I preferred to use the less expensive version of the tubes that were available from Antique Electronic Supply.

So, if you use the 6.3V tubes, then you only need to purchase the single transformer. Also, change resistor R17 to 2Ω and cap C12 to 4700µF. Everything else will remain the same. Furthermore, even though this is a fairly safe and sane project with a mild B+ on the chassis, be sure to think about each move twice before probing around in your circuit. Getting zapped is never any fun.

FINAL TESTING

Every time you test or use your new amplifier, be sure to use the proper start-up sequence. Turn on S1 first, then after about ten seconds turn on S2. After the heaters have warmed up for around 20 or 30 seconds, turn on S3. Now your amp is ready to be put into service. To power down, turn off S1 first then reset S2 and S3 to off. Now you're ready for the next time.

Remember to always have a pair of speakers or dummy resistors connected to the speaker terminals while testing your amp. Radio Shack has a good 8Ω resistor (#271-120) for this purpose. This resistor is a great way to test an amplifier if you don't care to listen to all of the hum, pop, and other noises that are produced when you touch certain parts of your circuit with the test probes.

To test for overall frequency coverage, I used an audio-frequency generator, and this amp seems to handle the audio spectrum quite well. I really think that the way an amplifier sounds to you personally is much more important than any test results, so, after allowing it to warm up for a while, I listened to Toni Braxton's album *The Heat*. This album has a great deal of punch in the midbass range, and the guitar on track #3, entitled "Spanish Guitar," sounds very crisp and lifelike through this amplifier. I think Toni has one of the most powerful yet totally controlled voices in the music industry today.

My other test album was Jane Monheit's *Never Never Land*. Engineer Tom Schick and mastering pro Gene Paul did a fantastic job of recording this young jazz artist! Her voice is very clear and natural and her phrasing is letter-perfect. The soundstage on this recording is a great way to test your new amp, or you can listen to your own favorites. Allow about 10 to 20 hours for your new amplifier to break in and settle down.

Good luck in building your version of this project. This "little amp" will serve you well for many years and the experiences you gain from constructing it can be stepping stones to other higher powered and more complex equipment in the future. Happy listening! ❖

The Sessions Monitor, Part 2

Part 1 of this article described the design concept of the Sessions Monitor System and the construction of the bass cabinets. Part 2 continues by describing the design and construction detail of the satellite and the final assembly of the system. **By Steve Philipczak**

After I had chosen the drivers for the satellite (Part 1), I needed to decide how to do the passive two-way crossover. Although I have had reasonable success with my own software program for designing passive crossovers, it is not as powerful and thorough as LinearX Systems' LEAP (Loudspeaker Enclosure Analysis Program), so I decided I would enlist the LEAP design services at Madisound for a project of this caliber.

SATELLITE PASSIVE CROSSOVER

Brian Kane at Madisound was very helpful on several of my first projects, and I also spoke with Tom Roberts and gave him the basics of what he would need to design the crossover for the satellite. In all fairness to Madisound, it would have been best to send them a "mock-up" cabinet so they could take some acoustic measurements with my cabinet/baffle before designing a crossover. At some point, I will probably do this.

I must say, however, that Mr. Roberts did an excellent job with the crossover, and I'm almost tempted not to fool with it. I also purchased the drivers for the satellites through Madisound. I've dealt with several other suppliers and all of them have been reasonably good, but I've never had any problems with Madisound and can highly recommend them.

Figures 5 and *6* show the LEAP on-axis response curves for the system and the individual drivers. According to the graphs, the low-end rolloff (3dB down point) of the 15W8530K01 appears to be about 55Hz in a .75ft^3 (roughly 21 ltr) sealed enclosure, with two drivers. This is excellent performance for a 5½″ driver. Since I was augmenting the bottom end with a subwoofer system, I modeled the satellite on a smaller, 15 ltr enclosure with a normal amount of fill to fit the space requirements of the Sessions Monitor.

The crossover schematic (*Fig. 7*) reveals the design to be a fairly simple second-order filter, with only one capacitor and no resistors in series with the tweeter. Using an MTM design often has the added benefit of increasing efficiency for the midrange drivers, thereby eliminating the need to pad down the tweeter.

MEASURING THE BASS-MID DRIVERS AND BOX MODELING

After the Scan Speak 15W8530K01s arrived, I broke them in for 24 hours on a signal generator set to 25Hz. I then tested all four drivers on the Woofer Tester, the same as I had done for the Adire Shivas in Part 1. *Table 4* shows the results. The other three drivers were remarkably close, again showing the manufacturer's good quality control.

I imported the Thiele/Small parameters into BassBox Pro and did a box model based on 15 ltr. The results can be seen in *Table 5* and *Fig. 8*. The Q_{TC} is close to .8, and the 3dB down point is roughly 62Hz. This would pose no problem when integrating to the bass system, which I had intended to cross over at 100Hz or so. Refer to Part 1 for more design details of the satellites and how they fit in with the whole system.

The SEAS E-011 tweeters arrived from Madisound along with the Scan

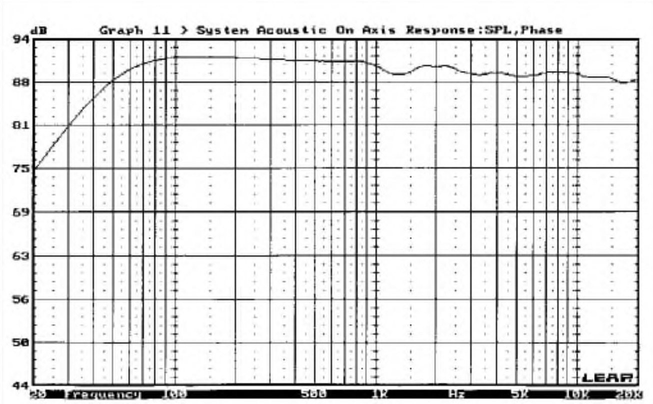

FIGURE 5: LEAP on-axis response of satellite.

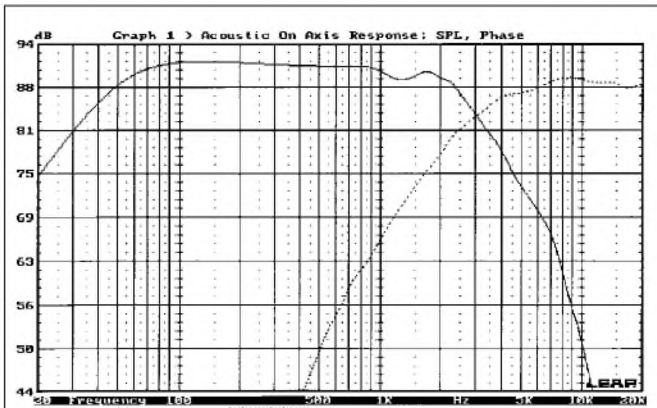

FIGURE 6: LEAP on-axis response, mid woofer and tweeter.

Speak midwoofers. They all look truly impressive and well built. These drivers are somewhat pricey, but I think you will be satisfied with their build and sound quality and top-flight engineering. It's a real blessing that we DIYers have access to such quality.

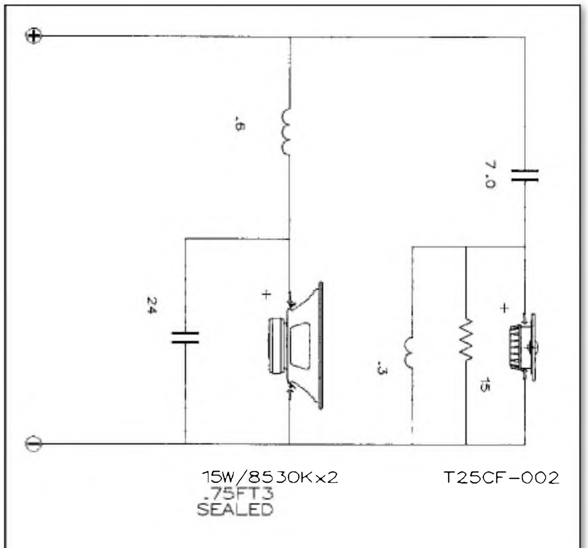

FIGURE 7: LEAP satellite-crossover schematic.

BUILDING THE SATELLITES

Figure 9 is a detailed drawing of the satellite. The top, bottom, and sides are of ¾″ MDF (medium-density fiberboard), and the front panel is composed of two layers of ¾″ MDF backed by a layer of ⅛″ particleboard. The particleboard backer resides inside the outer cabinet shell. The top and the side views show a piece of 6″-diameter stiff cardboard tubing that is halved lengthwise at the back part of the cabinet. You can find this material at most building-supply stores under the brand names of "Sonotube" or "Quick-Tube."

The "dead" corners behind the tubing are filled with sand, which makes the cabinet very solid and has the added benefit of reducing internal standing-wave resonance modes. I picked up this idea from Vance Dickason's excellent book, *Loudspeaker Recipes Book 1* (available from Old Colony Sound Laboratory). The cabinet is braced from top to bottom with one piece of ½″-thick maple (but a good-quality piece of ½″ plywood such as Baltic Birch would do as well).

A small horizontal shelf brace also sits beneath the semicircular cardboard tube to support the bottom of the tube piece as well as to provide some extra cabinet stability. To allow a place for the input binding posts, I did not extend the cardboard tubing to the bottom of the cabinet. You may recall from Part 1 that the passive crossover resides on top of the bass cabinet, mounted to the "T tracks," and not inside the satellite.

I rounded the outer edges of the front panel with a radius somewhere between 1″ to 1¼″, but in order to have a really positive effect on baffle edge diffraction, the radius needs to be close to 2″, so a 1″ radius is largely cosmetic, although it certainly can't hurt. I do not have the equipment to do a radius larger than a 1¼″ accurately, since you really need a CNC router for this. Should you choose not to do the radius at all, I don't believe it will affect the performance very much.

The grille is a stand-off type that I will describe later. The back is recessed ½″ exactly like the bass cabinets. The sides, top, and bottom are finished with wood veneer (my cabinets are natural cherry), and the front panel is painted gloss black.

SATELLITE CONSTRUCTION

Begin construction by cutting the top and bottom ¾″ MDF pieces, but before cutting, measure the top of your completed bass cabinet and make the width of the top and bottom satellite pieces exactly the same. After these are cut to size (11⅛″ deep), rout or cut a rabbet ⅛″ deep by ¾″ wide into the side edges of the top and bottom pieces. Now cut the sides out of ¾″ MDF to exactly the same depth as the top and bottom (11 ⅛″), by 19¼″ long. Adding the ⅛″ rabbets will make the cabinet 20″ high.

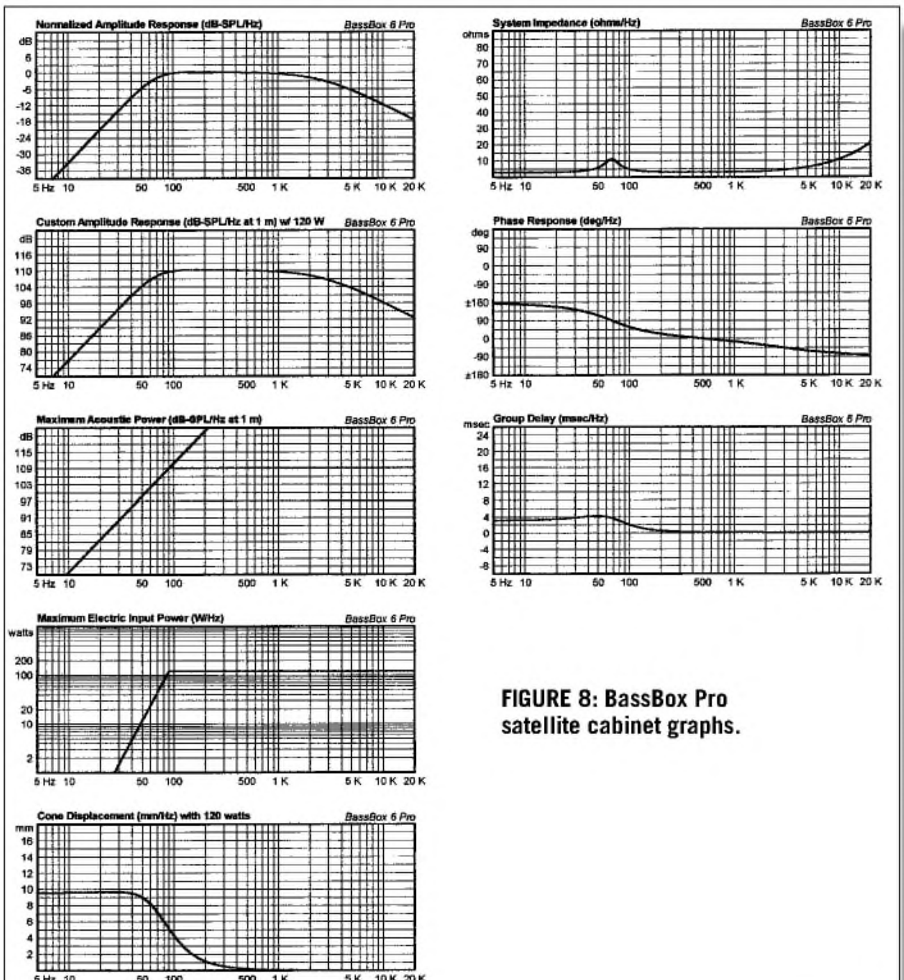

FIGURE 8: BassBox Pro satellite cabinet graphs.

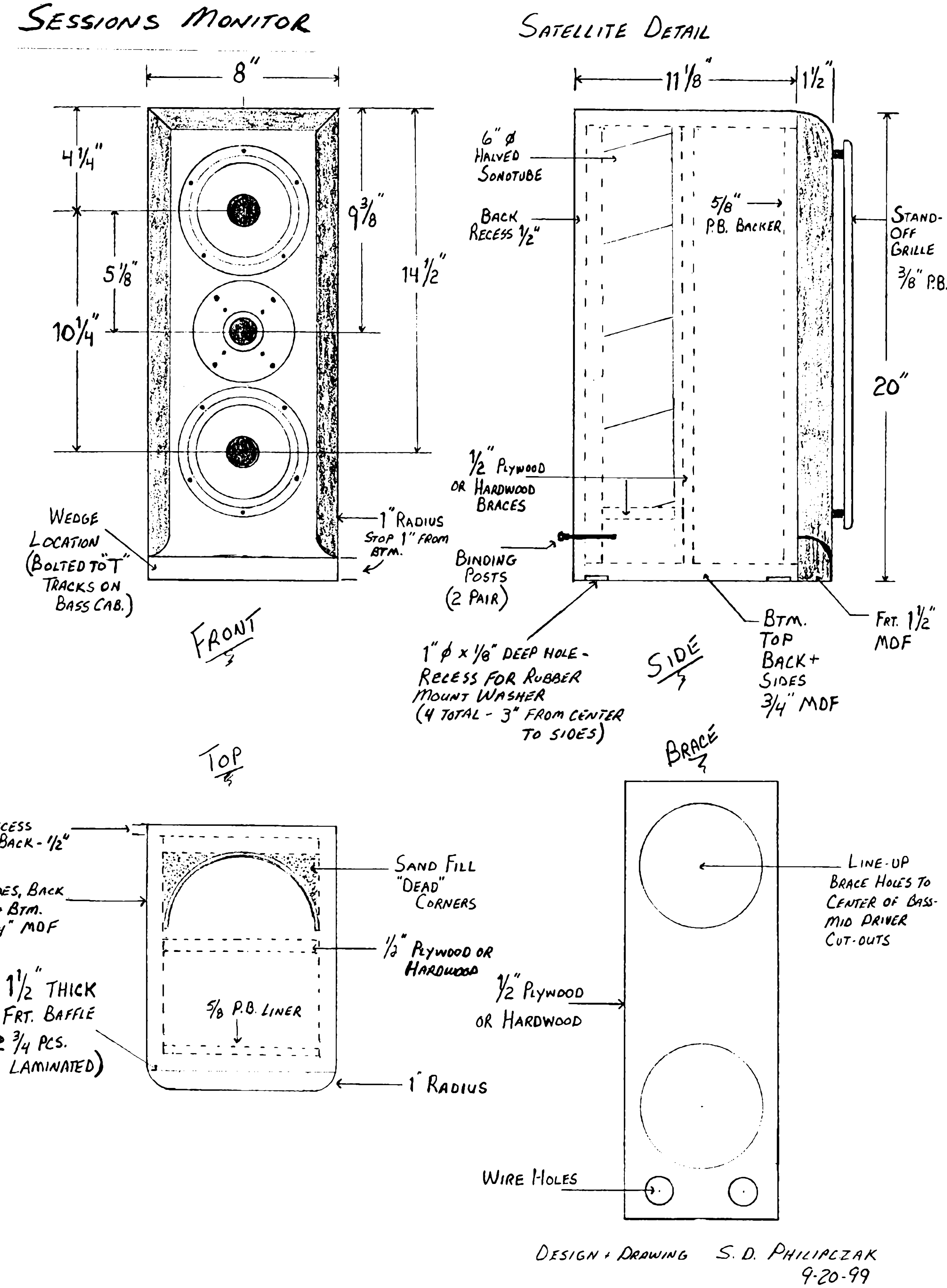

FIGURE 9: Satellite detail.

Rout or cut dado grooves ¼″ deep by ¾″ wide in the top, bottom, and sides for the back panel, so that it is recessed ½″. The center of this channel should be ⅞″ from the back edge. Cut the back panel to appropriate size (about 19″ by 7″ if your dadoes are ¼″ deep). Use your cabinet for reference.

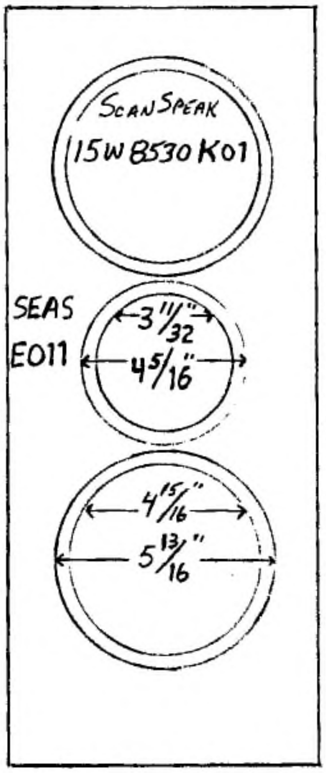

FIGURE 10: Satellite driver cut-out diagram.

Refer to *Fig. 9* and note the 1″-diameter flat-bottom recess holes on the bottom piece for the cone-type faucet washers that will rest on top of the small cone points attached to the bass cabinet T tracks. These recessed holes (also visible in *Photo 12*) should be located exactly 3″ from the side-to-side centerline to mate with the T-track spacing. Use a 1″ Forstner bit for these holes, if you have one. Make them about ⅞″ deep. After you have drilled the recesses, temporarily screw the cabinet together and check that it is square.

TABLE 4
WOOFER TESTER: SCAN SPEAK 15W8530K01 RESULTS

Peak Instruments
340 E. First St.
Dayton, OH 45402

RAW DRIVER OR CLOSED-BOX RESULTS

f_S is 33.9084Hz	Q_{TS} is 0.4986	
cone area is 0.0097m^2	Q_{ES} is 0.5364	
V_{AS} is 0.8955ft^3	Q_{MS} is 7.0877	
V_{AS} is 25.3677 ltr		
Re is 5.6000Ω;	Rm is 79.5971Ω;	R1 is 21.1126Ω
Voice coil inductance Le is 0.3074mH		
Efficiency is 84.49dB 1W/1m;		BL is 5.05 Tesla meters
Cm is 1920.132 microns/Newton;		Mm is 11.4767 grams
Rm (mechanical) is 0.345 mechanical ohms (Ns/m)		

NOMINAL IMPEDANCE MEASUREMENT RESULTS

R min is 0.00	R max is 0.00	R average is 0.00

Before proceeding further, you should have the cardboard tube halves cut and ready to use as references, since certain parts of the cabinet interior are actually built around them. Incidentally, because it is possible to have some variation with the cardboard tube, I did not provide dimensions on the drawing for the brace locations. You will need to cut the tube to length and keep the ends perpendicular to the sides.

I accomplished this by making an extended height fence for my table saw's miter gauge. Set your table saw's rip fence to cut the tube 15″ long. Then clamp the miter gauge with its extended fence to the table at a point where the tube's radius extends over the high point of the blade. Raise the blade only as high as necessary to cut through the tube. Turn the tube toward you as you're cutting, keeping it against both fences and down to the table.

Be sure to wear some ear as well as eye protection during this operation, since this actually makes more racket than cutting wood. This will give you a perfect cut. Now you can cut the tube in half lengthwise with a hand jig saw, giving you a half tube for each cabinet.

POSITIONING THE TUBE

Lay the dry-fit cabinet on its back, drop the half tube in place, and push it up against the top of the cabinet, rotating it so that its front edges are the same distance from the front of the cabinet on both sides. Make a mark on both sides about ¾″ from the tube's front edges. This will be the centerline for the vertical-brace dado. Also make marks at the bottom of the half tube on both sides and on the back panel to locate the top of the shelf brace's dado groove.

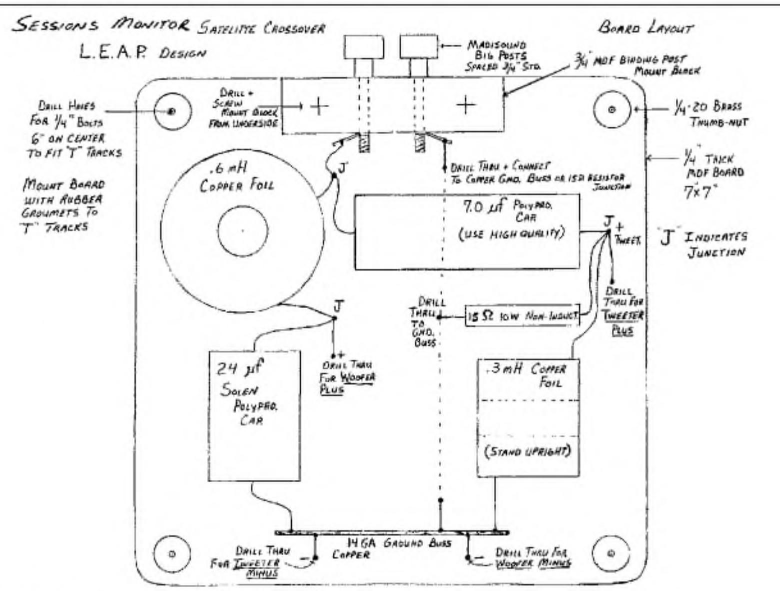

FIGURE 11: Satellite crossover physical layout.

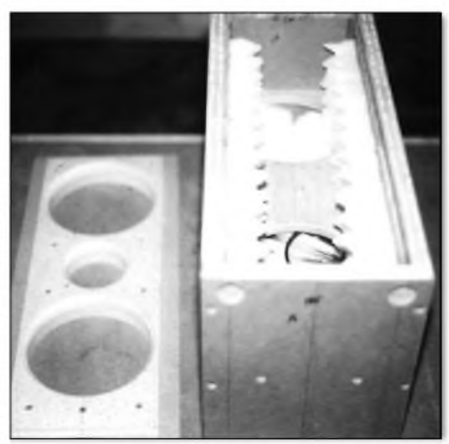

PHOTO 12: Satellite cabinet, ready to glue on front.

After you have made all these marks, disassemble the cabinet and rout or cut the dadoes for the vertical and shelf braces ½″ wide by ⅛″ deep. Remember to cut the dadoes for the vertical brace into the top and bottom as well as the sides, and also dado the back panel for the shelf brace. *Photo 13* shows how the dadoes and rabbets should look. If the outside diameter of the tube you use is exactly 6″, you will have a somewhat loose fit of the tube at the sides. If this occurs, line the sides where the tube sits with some ⅛″ or ¼″ hardboard (masonite).

Using the inside dimensions of the cabinet and allowing for the depth of your dado grooves, cut a piece of ½″

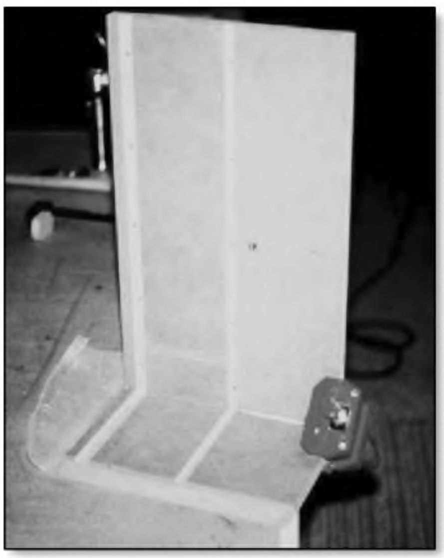

PHOTO 13: Start of satellite construction. Note dadoes for back and braces.

PHOTO 14: Open view of top of satellite showing half section of cardboard tube. "Dead" back corners to be sand-filled.

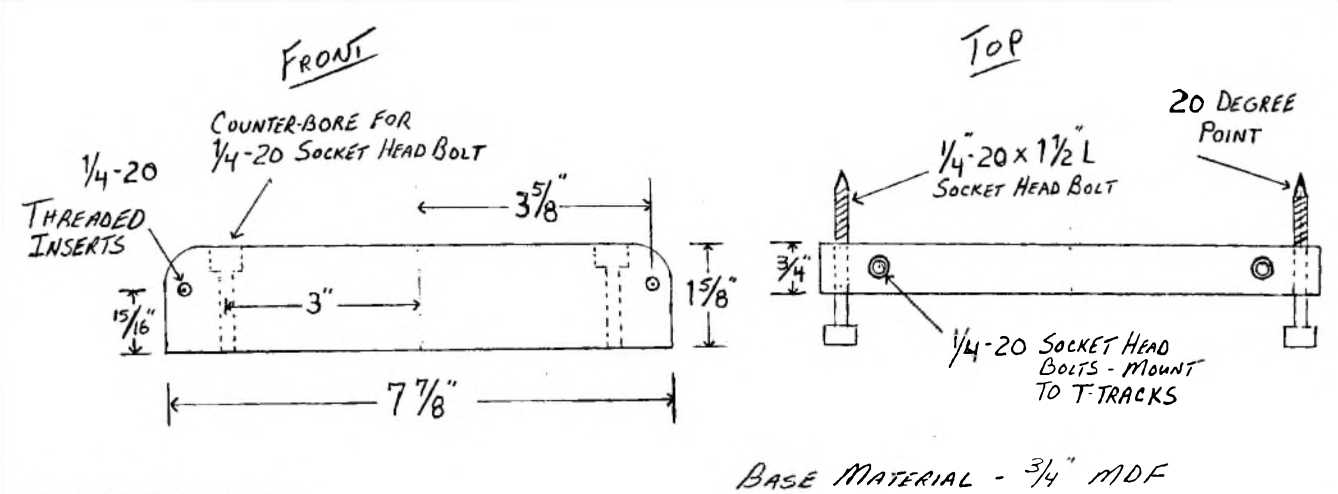

FIGURE 12: Pin block detail.

hardwood or plywood for the vertical and shelf braces. Make the shelf brace deep enough so it contacts the back of the vertical brace. Note that the vertical brace's back edge will be about ½″ forward of the half tube's front edges. This places the vertical brace at a point that is not centered in the cabinet's inside cavity. Once all the dadoes are cut, you should dry-fit the cabinet again with the braces in place. If all is well, you can disassemble the cabinet again and prepare to start gluing it together.

Before you start to assemble the cabinet, examine the back panel to locate and drill the holes for the satellite's binding posts. The two pairs of posts will be mounted side by side, centered in the cavity below the shelf brace. You need two pairs of posts, one pair for the tweeter and another for the paralleled midwoofers. After you drill holes and mount the posts, solder lengths of your choice of wire for internal wiring. Make sure these wires are long enough to enable easy removal of the drivers later on if necessary. I soldered two sets of wires to the midwoofer posts, one set for each driver.

CABINET ASSEMBLY

Refer again to the brace drawing in *Fig. 9*. Cut air-passage holes for the midwoofers and access holes for the driver wires as shown. Start assembling the cabinet by first gluing and screwing one side to the bottom; then glue in the back panel. Next, glue in the shelf brace, making sure it is seat-

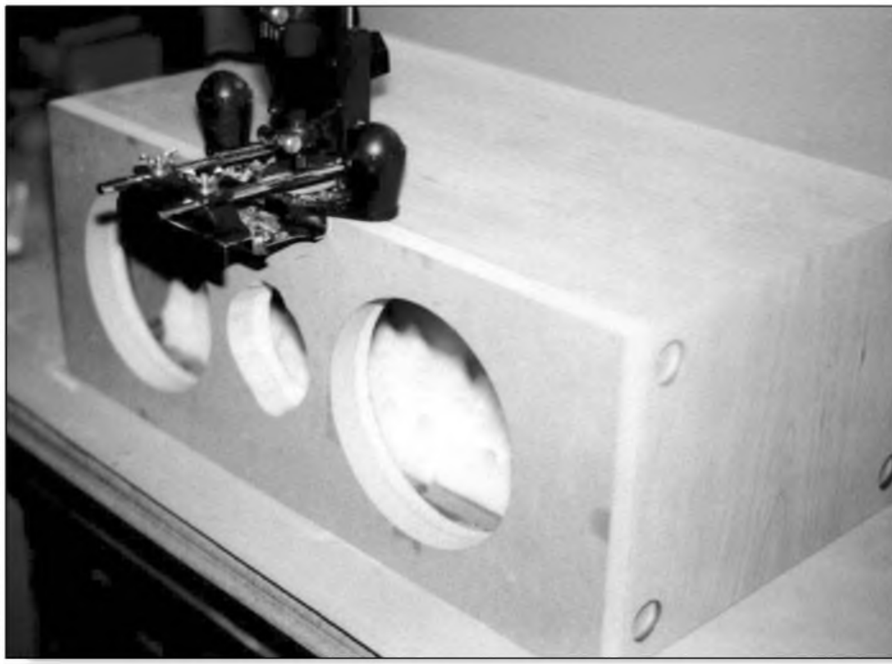

PHOTO 15: Cutting the reveal groove with Dremel tool on router base. Note: veneer is applied at this point.

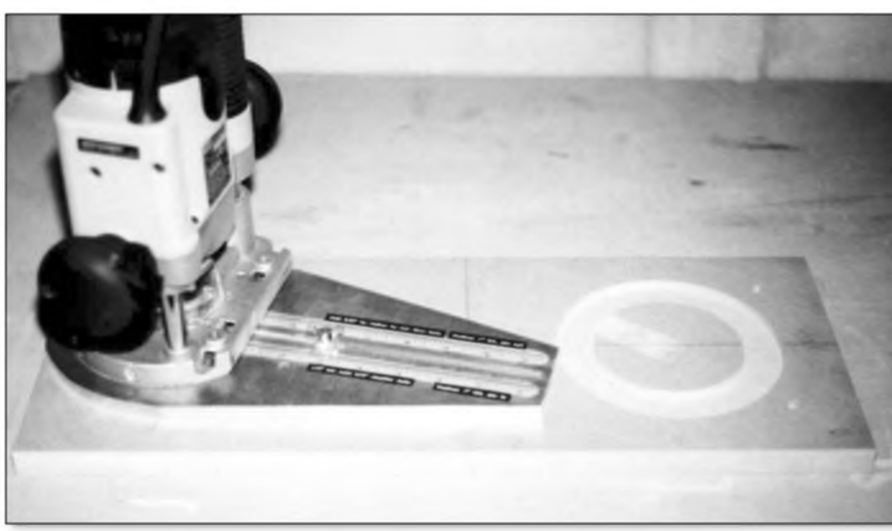

PHOTO 16: Routing front of satellite for drivers with homemade circle cutter.

ed into the side and back-panel dado grooves (it should clear the dado for the vertical brace on the sides). Next, glue and screw the other side in place. Do not yet glue in the top, but make sure it still fits. Check that all stays square during the assembly. Seal the back-to-side and back-to-bottom corners with silicone sealant at this time.

At this point, you need to glue in the half tube. I use Weld Bond for gluing the cardboard. Remove any wax coating that might be on the edges and the back of the tube where it needs to be glued. Put a line of glue on the back of the tube where it meets the back panel. Set the tube in the partially assembled cabinet and make sure it's down to the shelf brace. Tack it in place to the back and sides with some small brad nails. Glue the tube to the sides and the shelf brace, too. Allow this assembly to dry overnight before you continue.

Once the Weld Bond is dry on the tube, seal the tube to the sides and the shelf brace with silicone sealant. Roll up the driver wires and tuck them into the cavity below the shelf brace. Check to make sure the vertical brace slides freely in its dado grooves all the way down and into the bottom dado. Recheck that the top piece still fits properly. If not, relieve the brace as required.

PHOTO 17: Gluing the front panel with driver cut-outs on to the first layer of MDF.

Now glue the vertical brace in place. Pull the driver wires back through their access holes, using a coat hanger with a hook on the end if you can't get them with your fingers. At this point, the cabinet should look like *Photo 14*. Next, pour sterilized play sand into the open back corners behind the tube.

PHOTO 18: Satellite front showing edge round-over and gussets for driving mounting screws.

Screw on the top. Wait for all the glue to dry, then seal the top of the tube to the top of the cabinet with silicone by reaching through the top air-passage hole in the brace. Seal the rest of the inside corners with silicone sealant, but stop about an inch short of the front edge for now. The basic satellite shell is complete. Time to start on the front baffle.

Tap on the cabinet with a rubber mallet while doing this to settle the sand and pack in as much as possible.

Apply Weld Bond to the top edge of the cardboard tube and wood glue to the top-piece rabbets and dadoes.

PHOTO 19: Painting satellite fronts.

SATELLITE FRONT BAFFLE

Construction of the front baffle is best handled in stages, so if you have a problem routing the driver holes, you will not need to redo a whole assembly, but just one piece. As stated earlier, the front baffle consists of two layers of ¾″ MDF laminated together, backed by a 5/8″ piece of particleboard. Start by cutting a piece of particleboard to fit the inside front of the cabinet shell. Like the bass-cabinet front backers, this should be a good fit.

When satisfied, mark a centerline on the backer. Cut access holes for the drivers, using the dimensions in *Fig. 9* for reference. Remember to subtract ¾″ from the top to compensate for the wall thickness of the shell. The center of the top access hole should then be 3½″ from the top of the backer.

If you follow the spacing of the drivers exactly as *Fig. 9* shows, the driver diameters will be *very* close together. The "bridge" between the midwoofers and tweeter will be only about 3/16″. The drivers in the satellite should all be flush-mounted. Unless you have a very accurate circle gauge for your router, you may not wish to work that closely. You can certainly add a little more space between the drivers without hurting anything.

Place the top midwoofer just as I did, then move the tweeter and bottom midwoofer down a little. If you make this adjustment, draw a rough sketch with your numbers on it; you'll need this for reference to continue from here. Cut the midwoofer access holes in the backer with a 5½″ diameter, and cut the tweeter access hole with a 3 3/8″ diameter. Set the backer aside for now.

Cut a piece of ¾″ MDF for the middle layer of the front baffle assembly, making it ¼″ larger in both width and height than the outside of the cabinet shell (8¼″ × 20¼″). Temporarily screw the 5/8″ backer to this piece, leaving the same amount of "overhang" all around. With a pencil, trace the access hole locations onto the middle MDF piece (*Photo 12*). Remove the backer, and rough-cut the access holes in the MDF close to the trace with a jig saw. Now, glue and screw the backer to the middle layer, being careful to clean up any excess glue around the edges.

When the glue is dry, use a flush trim bit on your router to blend the access holes in the MDF middle layer to the particleboard backer. I also routed a 45° relief edge on the back side of

PHOTO 20: Aluminum block for sanding stand-off mounts.

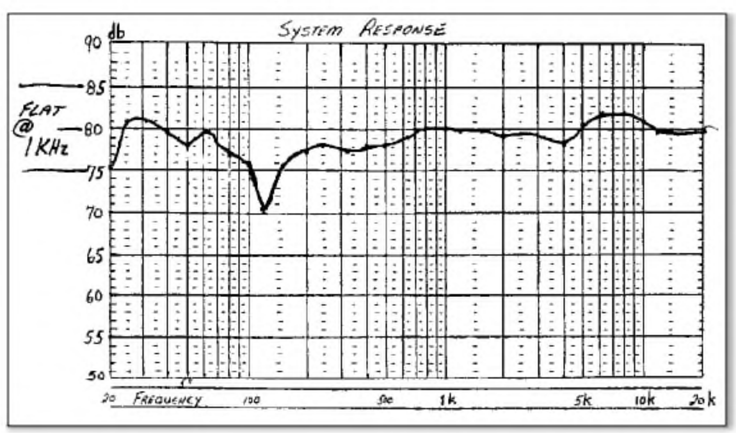

FIGURE 13: Finished system in-room response, measured at 36″ from front on tweeter axis. Dip at 125Hz is a room null at that distance.

the holes to help prevent driver "loading." Drop this assembly into the cabinet and check all fits. The middle MDF piece should overhang the outside of the cabinet shell about ⅛″ all the way around (it does not have to be perfect). Remove the sub assembly for now and prepare for gluing.

PHOTO 21: Back side of stand-off grille, showing mounts.

STUFFING AND SEALING

Before gluing the front subassembly onto the cabinet, stuff the cabinet with about 6 ounces of Acousta-Stuf. I also lined the side walls in front of the brace with some acoustic foam (*Photo 12*). When you've finished stuffing and lining, run a thick bead of silicone sealant ½″ back from the front edge and all the way around the inside of the cabinet. Apply wood glue to the front edges of the cabinet and the side edges of the front panel assembly backer.

Set the front-baffle subassembly in place and clamp it thoroughly to the cabinet. Screw on the front through the sides, top, and bottom and into the particleboard backer (as you did for the bass cabinets). Avoid screwing into the center area of the midwoofer holes, since there is not much material to the particleboard backer at that point. Reach through the access holes and smooth out the silicone sealant while it is still wet. Allow the cabinet to dry overnight before proceeding to the next step.

Once the basic cabinet is assembled with the first layer of MDF and particleboard backer in place, it is a convenient point at which to apply the veneer. So fill and sand all the screw holes and apply the veneer of your choice. Make sure the front edges of the veneer are straight and fit tightly against the back of the overhang edge of the MDF front piece. Blend and smooth the corners of the veneer. The next step is to rout the first MDF layer piece flush with the veneer using a flush-routing bit.

When that's done, remember to remove the veneer where it covers the recessed holes on the cabinet bottom. At this point I always cut in a reveal groove where the MDF and veneer meet, since they will have different finishes (the front will be painted). This will enhance the appearance quite a bit and is well worth the extra time and trouble. I do this with a Dremel tool on a router base with an

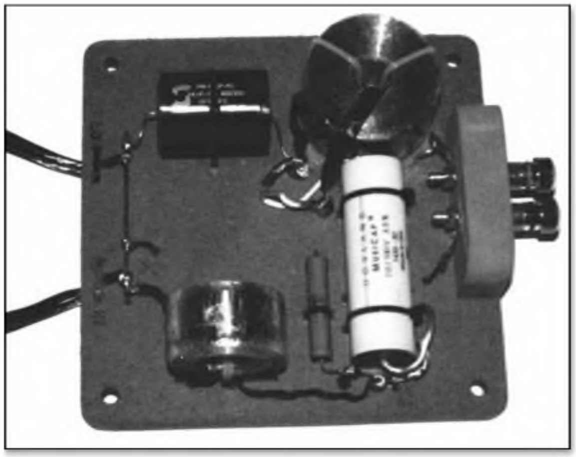

PHOTO 22: Passive two-way satellite crossover.

edge guide, using a small-diameter cutter (*Photo 15*).

Cut this groove mostly in the MDF, and only slightly into the veneer. You needn't cut the groove very deeply; 1/16″ is enough. The width of the groove looks best if it is less than 1/8″. I use a 5/64″-diameter solid carbide end mill (available from Reid Tool). If you do not have a setup like this, you can simply clamp a straight edge to the cabinet and carefully cut the reveal groove in with a small hand-saw. Now you are ready to make the actual front panel.

FRONT PANEL

Cut a piece of ¾″ MDF to the same dimensions as you first cut the middle layer (8¼″ × 20¼″), and mark a centerline on it from top to bottom. Using your figures or mine for the spacing of the drivers, rout the driver holes to flush mount them. I use a home-made jig for this (*Photo 16*). For the cut dimensions, see *Fig. 10*. Remember to account for the thickness of the compressed gasket you are using when you figure the depth of the outside cut. I always make the outside cut first, then do the through-holes.

Once you have cut the driver holes, check them with the drivers and the access holes in the cabinet. The holes should line up with the front panel overhanging the cabinet by about 1/8″ all the way around. Fashion a grille from your choice of material (5/8″ particleboard works nicely if you can find it), and cut holes in it for the drivers. I made a stand-off type of grille for the Sessions Monitor that I will detail later.

Drill the holes for the grille mounting sockets in the front panel, being careful not to go all the way through. Then drill a smaller hole for a #6 wood screw through the socket holes. This way you can hide the screws that help fasten the front without needing to fill and finish them. Make sure the grille fits and that all else is ok, then glue on the front panel, screwing through the grille holes and clamping around the outside and through the driver holes (*Photo 17*). When the glue is dry, flush-rout the front panel to the middle layer of MDF.

You can now rout the radius on the front edge if you're inclined to put one on. See in *Photo 18* that I did not take the radius all the way down to the bottom of the cabinet, but I stopped about an inch above, which provides a better aesthetic blend into the bass cabinet. Note also in *Photo 18* the small gussets in the midwoofer holes. I made these out of solid maple to provide added substance for the driver-mounting screws. This concludes the basic construction of the satellite cabinet.

FINISHING THE SATELLITE CABINET

The easiest way to finish the satellite is to start with the veneer. Apply your favorite finish, or try my method as detailed in Part 1 in the bass-cabinet construction section. Wait for the finished veneer to dry thoroughly before proceeding, since you must mask over it. Mask the veneer carefully, making sure not to cover the reveal groove anywhere. Also, mask the driver holes to protect the inside of the cabinet from spray. Sand the fronts with 220 dry sandpaper.

You must fill the "end grain" of the MDF before you paint the front. You can use wood filler for this, but I found it easier to use a good grade of automotive primer-surfacer, three coats of which will fill the "end grain" nicely with minimal hassle. You can find this product at auto-body supply stores. It is lacquer-based, and dries very quickly, allowing you to sand it within a couple of hours. Lay the cabinets on their backs to make your spray painting go easier (*Photo 19*).

If you wish to be fancy, you can paint the fronts with several coats of gloss black lacquer and then fine-

TABLE 5
BASSBOX PRO SATELLITE CABINET BOX MODEL

BOX PROPERTIES
DESCRIPTION
Name: Sessions Monitor satellite
Type: closed box
Shape: prism, square (optimum)
Serial No: 1
Comment: drivers tested on woofer tester
Finish: hand-rubbed wipe on polyurethane
Veneer: natural cherry
Core material: ¾″ MDF
Joinery: rabbet, dado, and slot
Bracing: one vertical, one horizontal (solid maple)
Box parameters

Vb =	15 ltr
QT_C =	0.787
QL =	7
f3 =	62.58Hz
Fill =	normal

DRIVER PROPERTIES
DESCRIPTION
Name: 15W/8530K01
Type: standard one-way driver
Company: Scan Speak A/S
Serial No: sampled 01232-78 + 01232-76
Comment: Revelator
Piston: reinforced, coated-paper cone
Suspension: Large rubber surround. 4″-diameter spider
Dust cap: coated hard paper
Frame: cast frame
Voice coil: 38mm aluminum voice coil on Kapton former

Configuration
No. of Drivers = 2
Mounting = standard
Wiring = parallel
Drivers sum coherently = Yes
Mechanical parameters

f_S =	34.3Hz
Q_{MS} =	6.85
V_{AS} =	24.52 ltr [49.04]
Cms =	1.856mm/N [0.928]
Mms =	11.61 g [23.23]
Rms =	0.367 kg/s [0.734]
X_{MAX} =	6.5mm
Xmech =	9mm
P-Dia =	4.363″ [6.17]
Sd =	95cm^2 [190]

Electrical parameters

Q_{ES} =	0.522
Re =	5.6Ω [2.8]
Le =	0.33mH [0.165]
Z =	8Ω [4]
BL =	5.185 Tm [0]
Pe =	60W [120]

Electromech. Parameters

Q_{TS} =	0.485 [0]
no =	0.183% [0.366]
1W SPL =	85.14dB [88.15]
2.83V SPL =	84.5dB [90.52]

sand and buff. I painted my front panels with gloss black epoxy enamel and they look just fine like that. Allow the fronts to dry well and then mask off the binding posts and paint the cabinet backs flat black. After all the paint is dry, apply foam gasket tape to the driver flanges and mount the drivers.

MAKING THE STAND-OFF GRILLES

I decided to make stand-off grilles for the Sessions Monitor satellite (refer to Part 1, Photo 3) to help reduce the negative-diffraction effects that normal grilles can have. To do this easily, I used a small ball-and-socket type of grille fastener (from Meniscus), the flanges of which are ½″ in diameter. From my local Sears, I then obtained some ⅝″-long nylon spacers with ½″ outer diameter and ⅜″ inner diameter.

To use these, first glue the "ball" ends of the fasteners into the nylon spacers with gel-type instant adhesive. Then fill the other end of the spacers with pieces of ⅜″ dowel. Cut the dowels close to the end of the spacers. To ensure that all of the stand-offs were exactly the same length, I used an aluminum block with a ½″ hole drilled in it to the proper depth. I then inserted the spacers (ball-end down, of course) into the block and sanded them flush. A smaller hole drilled into the other side of the aluminum block allowed me to eject the finished pieces. *Photo 20* shows the stand-offs at various stages of assembly, along with the aluminum block.

Next, with a ½″ Forstner bit, I drilled slight recesses into the back side of the grille at the stand-off locations. Then I drilled clearance holes for #6 screws through the recesses, and countersank the front of the holes. I then mounted the stand-offs to the grille frame with short #6 wood screws. *Photo 21* shows the back side of the grille with the stand-offs installed. I did not glue the stand-offs in place, for if they are just screwed in, you can replace them if damaged, or reuse them if the grille is damaged.

I installed the sockets for the fasteners into the front panel, their black color blending in well with the painted front. I then covered the grille frame with standard acoustically transparent grille cloth. I also rounded over the front edge of the grille frame with a ⅜″ radius for appearance's sake. This completes the construction of the satellite, and you are now ready to assemble the passive crossovers.

SATELLITE CROSSOVERS

Assembly of the passive crossover for the satellites is fairly straightforward. *Figure 7* is the schematic drawing from Madisound's LEAP design, and *Fig. 11* shows the physical layout for my crossovers. *Photo 22* shows the finished crossover assembled and wired point-to-point on a ⅜″ MDF board that is 7″ square. Note that the .6mH coil is lying flat and the .3mH coil is upright and located at the opposite corner to reduce coupling.

I used some premium-quality components in my crossovers, mainly because of the quality of the system drivers and also because there aren't that many components. Both coils are

CFAC copper-foil type inductors. The 7µF capacitor is the only component in series with the tweeter, a Hovland MusiCap. The resistor in the tweeter circuit and the midwoofer capacitor are not as critical, because they are paralleled with the drivers. I used a 15Ω noninductive resistor and a 24µF Solen polypropylene capacitor for the woofer circuit.

I put some foam padding underneath all the parts, and mounted them to the board with tie wraps. You can obtain the crossover parts from Madisound, Meniscus, or Parts Express. The leads going to the satellite binding posts should be about 6″ to 8″ long (refer to *Photo 22* again). The mounting holes for the board are drilled 3″ from center to the sides to mate up with the ¼-20 bolts installed in the bass cabinet T tracks (Part 1, Photo 7). I used some rubber faucet washers under the board to help isolate it from vibration.

I installed the board with ¼-20 knurled brass thumb nuts (available from Reid Tool or any good hardware store), so that it could be easily removed and yet solidly mounted. The binding posts are Madisound's "BIG POSTS" and are mounted to the board with a ¾″ piece of MDF screwed and glued to its rear. I did not supply a separate list for the crossover parts, since they are detailed here and on the physical-layout drawing of *Fig. 11*. Once you have assembled your crossovers, there are a couple of small details left to attend to, and then it will be time to assemble the total system.

THE WEDGE AND THE "PIN BLOCK"

Refer again to Photo 6 (Part 1), to get an idea of what the pin block and the wedge should look like. These are a couple of ideas I had that are optional, but highly recommended. The wedge makes a nice aesthetic blend by eliminating a 90° shelf where the satellite meets the bass cabinet. The pin block basically pins the satellite cabinet against the wedge, preventing any minute front to back movement of the cabinet due to driver-cone movement. You can make the pin block now. The wedge, however, must wait until you do some testing of the finished system.

Make the pin block out of a single piece of ¾″ MDF, with the top corners rounded for appearance (*Fig. 12*). The pins, or spikes if you will, are ¼-20 socket-head Allen bolts, 1¼″ to 1½″ long, with points machined on the ends. You can do this yourself if you have a bench grinder and a power hand drill. Put the socket end of the Allen bolt in your drill chuck and set the direction to reverse. Tap the bolt around to eliminate as much of the run-out as possible. Hold the end of the bolt lightly against the grinding wheel at approximately a 20° angle until you get a nice sharp point.

Install ¼-20 threaded inserts in the pin block for these bolts. The inserts should be horizontally in line with the centers of the satellite-cabinet sides to keep them out of the line of the T tracks so they will not interfere with the holes you need to drill to mount the pin block to the bass cabinet. Counter-bore a hole on each side of the top of the pin block 3″ from center for ¼″-20 socket-head bolts to mount the pin block to the T tracks. Paint the pin block flat black and set it aside for now.

FINAL ASSEMBLY AND ADJUSTMENTS

With all the major work finished, you can assemble and evaluate the Sessions Monitor system. You will need eight 1″-diameter cone shaped faucet washers for the bottom of the satellite cabinet. With some instant adhesive, install these into the recesses you made earlier, cone facing down, of course. The center holes in these will mate to the small Tenderfoot cones on the T tracks of the bass cabinet.

The Tenderfoot aluminum cone points (available from Audio Advisor) have center holes drilled into the flat sides. You will need to enlarge these holes (#6 or #7 drill) and tap them with a ¼-20 flat-bottom tap. Be careful when you are drilling to avoid cutting through the points. Install the Tenderfoot cones into the T tracks of the bass cabinet with short ¼-20 hex bolts. Make sure you can tighten the cone points to the tracks before they bottom out in the tapped holes; if they do not become tight, shorten the bolts accordingly.

If all looks well, screw the cone points down to the tracks, but do not tighten them yet. Make sure you can slide the cone points back and forth in the tracks. Rest the satellite cabinet on the cone points with the rubber washers over the cone-point tips. Enlist another pair of hands for this if needed. You should now be able to slide the satellite cabinet back and forth with the cone points moving in the T tracks. Move the satellites to the front edge of the bass cabinets for now.

Install the completed crossover boards at the very back of the T tracks. Do not install the pin blocks at this time. Connect the active crossover to your system and the speaker cables from each amplifier to their respective cabinets. I suggest you set the active-crossover frequency at 100Hz as I did, although you can certainly experiment with this later. Where you set the controls on the active crossover will depend on your room, the respective gains of your amplifiers, and the fact that the bass system is less efficient than the satellites. The User's Guide that comes with Marchand's XM-9 gives you some good starting guidelines.

Balance the system as best as you can. If you have test equipment such as sweep generators, spectrum analyzers, and so on, by all means use them to get a flat response over the crossover frequency. Do this as a starting point, but let your ears be the final arbiter. You will probably need to move the speakers around to find the best location in your room, so do not install the large cone-point feet into the bottom of the bass cabinets yet. Instead, use three furniture-type "glide feet" with ¼-20 threads.

Once you are satisfied with the positioning, install the large cone points. Listen to the system this way for about a week, or until you think the tweeters are fully broken in. The tweeters will smooth out a bit given some playing time. My woofers and midwoofers were broken in before I tested them, but if yours are not, you'll prefer to wait until they come around, too.

SATELLITE POSITIONING

After the drivers are broken in, you need to enlist two understanding friends to help you decide on the final position of the satellite. Time-adjusting is a technical subject I will not go into here. Whether or not you believe such adjustment is worthwhile, you may as well try to achieve it, since you can move the satellite on the Sessions Monitor. Time-adjustment may be even less of an issue with an active crossover, and is probably less critical between midrange and high-end than between midrange and bass. You cannot physically move the tweeter on an MTM design anyway.

Put yourself in the listening position and play full band pink noise through the system with a generator or a test CD. Have your helpers slowly slide the satellites back and listen for a smoothing-effect change. Repeat the procedure as needed until you determine you have found the ideal location for the satellites. Leave them in this position for now and proceed to make the front wedges.

The distance from the front of the satellite to the front of the bass cabinet will be the depth dimension for the wedge. If you put a radius on the front edge of the satellite, then figure the height of the rear side of the wedge to meet the end point of the radius. Refer to Part 1, Photo 4 for some visual clarification on this. Once you have the depth and height dimensions of the wedge, you can calculate the angle at which to cut the top.

I constructed the wedge from two pieces of ¾″ pine glued together, cutting the angled front on my table saw. I then covered the wedge with a material called Foamtastic, available from most arts and crafts stores. This material is about ⅛″ thick, and it comes in various colors. Before covering the wedge, I drilled two counter-bored holes 3″ from center to mount it to the bass-cabinet T tracks.

When the wedge is finished, you will need to remove the satellites, the cone points, and the crossovers from the T tracks. Install the wedge all the way to the front of the bass cabinet and secure it with the appropriate length of socket head bolts and ¼-20 hex nuts. Make sure that the wedge is secured to the bass cabinet and that the mounting bolts do not bottom out into the T tracks. Shorten the bolts if needed.

Reinstall the cone points and the satellites, and slide the satellites up against the back of the wedge. Now you can install the pin block. Leave approximately a ⅛″ gap between the pin block and the back of the satellite. Bolt the pin block securely to the bass cabinet, and, as you did for the wedge, make sure the pin block's mounting bolts do not bottom out into the T tracks.

Install the ¼-20 spike bolts you made earlier and tighten them until the spike enters the back of the satellite sides. Do not over-tighten them; when you feel some resistance, and about one third of the spike is buried, it's time to quit. Now tighten the small Tenderfeet down to the T tracks. Carefully use a pair of slip-joint pliers for this if needed. Install the passive crossovers again and hook up all connections. This completes the construction of the system. Time to enjoy the fruits of your labor by doing some serious listening!

LISTENING IMPRESSION

When my system was complete, I placed it in my somewhat large (3200ft^3), purpose built room that I constructed in my basement several years ago. The dimensions of the room are not exactly ideal, since I needed to work around some support poles when I built it. Nonetheless, it is a good "sounding" room in general. Speakers that sound mediocre in our living room sound much better in my listening room. The front of the speakers are 43″ forward from the wall behind them. Almost all speakers tend to image better when placed away from walls.

The problem with moving them forward is the loss of bass reinforcement from the rear wall. Since the woofer in the Sessions Monitor is mounted fairly low (another part of my design criteria), it will have bottom-end reinforcement from at least one boundary (the floor), no matter how far from walls it is located.

The speakers are placed 7′ apart (measured from the inside bottom corner of the bass cabinets), and the listening position is 9½′ from the speaker fronts, which are toed in approximately 18°. The walls behind and to the sides of the speaker area are treated with Sonex. When hooking up the system, the output of your preamp goes to the input of the active crossover. The low-pass output from the active crossover goes to the amplifier, which will be connected to the bass system; the high-pass output goes to the amplifier that is connected to the satellite's passive crossover.

The User's Guide that is supplied with the Marchand XM-9 explains this and anything else you will need to know. One of the big advantages of biamping is separating the frequency ranges actively and before the signal is amplified. Loss of power and phase anomalies associated with large coils in a passive three-way crossover are eliminated entirely.

MEASUREMENTS

After doing some casual listening to the system, I did some preliminary measurements. A near-field measurement on the satellite with warble tones and an SPL meter (Radio Shack model 33-2050) revealed the satellite to be 3dB down at 63 to 64Hz, just as predicted with my Box Model.

Figure 13 is an in-room response curve, again generated with warble tones and an SPL meter placed on the tweeter axis, 36″ away from the satellite cabinet. The dip at 125Hz is most likely a room null or due to the placement of the meter. The same frequency at the listening position is actually up 1dB. It should also be noted that my SPL meter is not accurate much above 10kHz, but from that point on up the scale, the response is of course entirely governed by the tweeter.

It is difficult to describe the sound of a speaker system into which you have put a lot of your own time and effort without being somewhat prejudiced. That said, I will do my best to describe the sound objectively. My first impression was...large. The soundstage is wide, deep, and high, and the imaging in general is spacious. It is a big speaker, and it sounds it.

The tweeter axis is a little higher than a seated listener's ear level, which seems to produce stage height. When you stand to listen, the sound does not change much, if at all. The bottom end is robust and solid, and yet very well defined. An excellent test track for this is Patricia Barber's cover of "She's a Lady" on her *Modern Cool* album (Premonition 741).

This track features Patricia's voice, along with a consistent finger snap, some guitar accent, and a very well recorded upright acoustic bass. If there is anything rattling on your speaker cabinets (or your house for that matter), this track will show it. "Let it Rain," from the same album, is another good test track for both piano and voice.

CLEAN AND OPEN MIDRANGE

The midrange is clean and open, and percussion is especially "snappy." A good recording to show this is Medeski, Martin, and Wood's "Friday Afternoon in the Universe" (Gramavision GCD 79503). When Billy Martin hits snares and toms, they are quite convincing. Both male and female voices sound very natural. John Gorka's "Can't Make Up My Mind" from his *Between Five and Seven* album (High Street 72902) is a good recording of his unique and powerful voice, and it comes through with force on the Sessions Monitor.

A good female voice for speaker testing is one of my favorite artists, Alison Krauss. Not to take anything away from her violin playing, which is also excellent, her voice is very well recorded on several of her latest efforts. Try any of the first three tracks on her *Now That I've Found You* collection CD. Her voice and talent are outstanding.

At first I thought the top end of the system was a bit too bright, but I gave the SEAS tweeters some more time to break in before deciding to make any changes. After a few weeks, I concluded that the high end had smoothed out a bit, so I decided not to make any changes after all. The high end now sounds "airy" and smooth, with very little trace of harshness.

You may wish to include an L-pad in the passive crossover for the satellite, depending on your room and tastes. For me, it was not necessary. My tastes lean mostly toward acoustic music and jazz, and the Sessions Monitor does very well with both. I also do listen to some classical music. Full orchestra on the system has convincing weight, and strings and horns sound real.

To sum up, the engineering put into the drivers used in the Sessions Monitor system was worth the effort and expense. My hat is off to the good people at Adire, Scan Speak, and SEAS for pushing the edge of the art, as drivers just keep getting better and better.

I thoroughly enjoy listening to music through this system. There's really nothing lacking, and there is a sense of ease about the sound in general. Both amplifiers and all drivers are not being pushed anywhere near their limitations.

The opinions of other audiophiles who have listened to the system at length are not unlike my own. Many have compared it favorably to some of the best commercial systems they've heard. Listening sessions with friends have lasted many hours without fatigue. These sessions were partial inspiration for the name, "Sessions Monitor," along with the s's in sessions being a takeoff on the driver-manufacturer names (Shiva—Scan Speak—SEAS).

FINAL THOUGHTS

Would I change anything? Not much, really. As stated earlier, I think vertical braces in the bass cabinet would be a good idea. You could also, perhaps, make the satellite cabinet an inch or two deeper, which would give you a little more volume and consequently a little more bass extension, perhaps allowing a slightly lower crossover frequency to the bass system. I can't say whether this would effect much of an improvement in the system, though. It seems best to not make the midwoofers do too much.

You could experiment with different slopes or different crossover frequencies on the active crossover. You could also experiment with wiring the Shiva woofer's voice coils in parallel, bringing the efficiency of the bass system more in line with the satellite. Doing this would, of course, change the Q and probably raise the f_3 somewhat, but it might help in a room that is particularly "lossy" in terms of bass.

You could probably optimize the passive crossover to the satellite cabinet a little better by sending the actual enclosure to Madisound. I will probably do this at some point, especially as I will soon duplicate the system for someone else. Another consideration is to purchase Scan Speak 15W8530K01 drivers from North Creek Music Systems. They sell Scan Speak drivers that are pretested and hand-sorted into matched pairs at a slightly higher cost.

The nice part about the Sessions Monitor is that it lends itself to versatility and refinement. You can match the bass system to just about any type of room. You have controls on the active crossover, and if the bottom end sounds a bit underdamped, you can employ the resistive damping option that the second voice coil of the woofer offers (see Part 1).

The satellite's passive crossover is externally mounted and is easily modified or replaced. All items mounted to the top of the bass cabinet are movable, and yet solidly mounted, thanks to the T-tracks. If the exposed passive crossover is a practical or aesthetic problem, you can make a small cover box out of solid wood and mount it to the middle of the crossover board with a single bolt.

If you decide to build the Sessions Monitor for yourself, I wish you luck and lots of fun. It's a nice project to build; you can do it in stages to keep it interesting. I made sure that all items required to build the system were available from speaker supply companies, typical local home centers, and woodworker suppliers. Most of these advertise regularly in this magazine and are very helpful in correspondence if you need it. Although I would not recommend the Sessions Monitor for a first time project, it is not at all that difficult to build. No fancy joinery is involved, just simple dadoes and rabbets.

Outside of the assistance I received from Madisound on the satellite's passive crossover, this project was entirely a one-man effort in both design and construction. I invite any constructive criticism or suggestions from those who are better versed than I am. Further refinement of the design is no doubt a worthwhile endeavor. ❖

SUPPLIERS

Audio Advisor
(small and large Tenderfeet aluminum cone points)
(800) 942-0220
www.audioadvisor.com

Old Colony Sound Laboratory
(*Loudspeaker Recipes Book 1*)
(888) 924-9465
www.audioXpress.com

Hovland
(Musicaps)
(209) 966-4377
www.hovlandcompany.com

Reid Tool
(Carbide end mills and brass thumb nuts)
(800) 253-0421
www.reidtool.com

North Creek Music Systems
(Scan Speak drivers and miscellaneous components)
(315) 369 2500
www.northcreekmusic.com

Solen
(Polypropylene Capacitors)
(450) 656-2759
www.solen.ca

Choosing and Using Electronic Parts: A Survival Guide, Part 2

We continue our examination of electronic circuits with a look at some common circuit components. **By Charles Hansen**

RESISTORS

Resistor elements can be made from carbon, metal or oxide films, or resistance wire. Overheating is the major cause of resistor problems, resulting in deterioration over time until at some point the resistor fails, usually resulting in an open circuit. Resistors generate their own internal heat and serve as "watt wasters" in order to establish the proper operating voltages in electronic circuits. Their combined total in any piece of equipment can be the major contributor to heating, especially when large power resistors are used.

Figure 4 shows the relative failure rates for resistors versus temperature. *Figure 5* shows the relative failure rates for resistors versus power. Note the large jump in failure rates when you use values larger than 1MΩ. From a reliability standpoint, it makes sense to use two lower values in series rather than one resistor of 1MΩ or more.

CARBON COMPOSITION RESISTOR

Carbon composition resistors are very reliable electronic components, with excellent RF performance. They usually fail when overheated or overly stressed due to shock or vibration. Composition resistors are considered to be general-purpose resistors, available in tolerances from ±5% to ±20%. They should not be used in critical applications where environment changes can be expected. The effects of humidity, temperature, and pressure, as well as normal aging, can cause a composition resistor to decrease as much as −15% below its specified tolerance.

Every resistor acts as a noise generator whose mean square AC noise voltage is directly proportional to the value of the resistance. Johnson noise, also called thermal noise, occurs in all conductors as a consequence of the random motion of electrons. In addition, at low frequencies, the noise power varies inversely with frequency. A common term for this type of noise is 1/f noise, or excess noise, since it exceeds Johnson noise at frequencies below a few hundred cycles. The root-sum-square total of noise currents (including those in tubes or semiconductors) will determine the noise floor in audio equipment.

Figure 6 shows the relationship between resistance and total current noise in the fixed resistors discussed above. As you can see, carbon composition resistors have a much higher excess noise value than the other types of fixed resistors, and it increases with time and temperature.

Carbon composition resistors are also sensitive to applied voltage, and have voltage limits based on case size rather than power ratings. The point where the applied voltage limit changes from power to voltage is called the critical resistance, listed in *Table 1*.

FILM RESISTORS

Film resistors are made by applying a thin layer of resistive material to an insulated form. The most commonly used film resistors can be divided into types according to the materials used: carbon-film, metal-film, metal-oxide and cermet (a combination of ceramic and metal materials), which have the best reliability of all the resistor types. The spiral pattern cut into the resistive film increases the inductance, limiting their use at high frequencies. Like the carbon composition types, film resistors also have critical voltage limits.

CARBON FILM, METAL OXIDE, AND METAL FILM

Carbon film resistors have improved noise and stability performance over the composition type, and tolerances of ±5%.

Metal oxide resistors are a bit noisier than carbon film types, but have tighter tolerances, down to ±2%, and better stability.

Metal film resistors have very tight tolerances (1% to 0.1%) and excellent temperature stability. If a metal film resistor is exposed to excessive voltage, ion migration can occur between adjacent resistance stripes. This can cause an increase in resistance in the more positively biased portion of the package. Moisture and elevated temperature can accelerate this ion migration, and can produce localized changes in the metal film thickness or width.

THERMISTORS

Thermistors are temperature-compensating devices whose resistance change with temperature is designed to track that of other devices, such as semiconductors. Ideally, the unavoidable de-

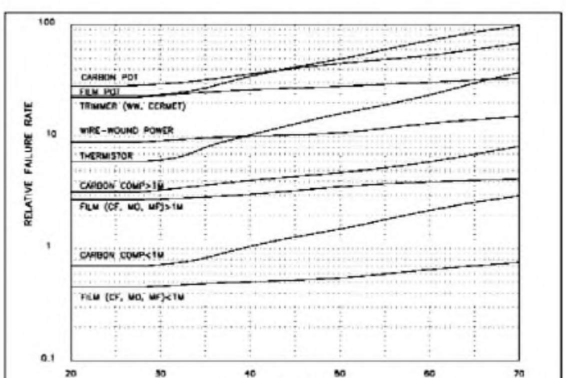

FIGURE 4: Resistor failure rate versus temperature.

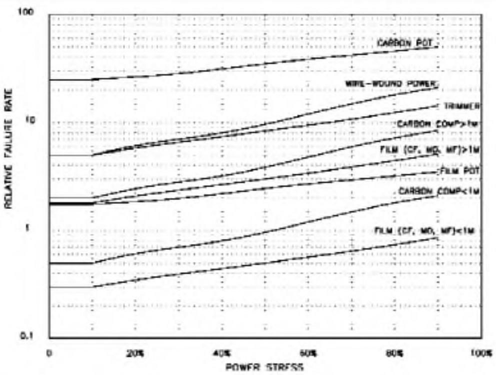

FIGURE 5: Resistor failure rate versus power.

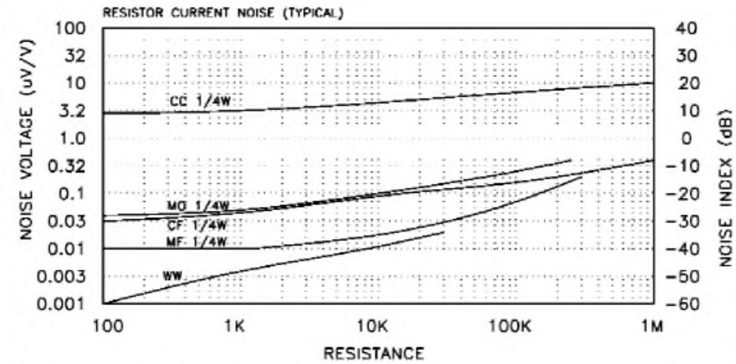

FIGURE 6: Current noise for fixed resistors.

crease in forward voltage drop with temperature in diode and transistor junctions is offset by the deliberate opposing change in resistance of the thermistor.

WIREWOUND RESISTORS

The wirewound resistor is the most stable resistor type, with commercial tolerances down to 0.1% or less. Wirewound resistors are constructed by winding a nichrome resistance wire around a ceramic insulated form and covering the package with a protective coating. Flameproof versions are available to meet fail-safe requirements.

Conventional wirewound resistors are highly inductive. Special noninductive types are available, with the winding applied in opposing directions to cancel the inductance, although this technique is not completely effective at high frequencies.

Their predominant failure mode is an open in the wire resistance element, more prevalent as the resistance increases. Turn-to-turn shorts are also possible due to accumulation of dirt, dust, or a break in the outer coating. Moisture can cause corrosion of the end connections.

VARIABLE RESISTORS

High-wattage variable resistors, or rheostats, were once used in audio to adjust filament and plate voltages, and are still used in L- and T-pads for adjusting remote loudspeaker volume. We will concern ourselves here with the low-wattage variable resistors commonly referred to as controls or potentiometers (pots). A potentiometer uses a continuous resistive element with a sliding contact that traverses the element in a straight or circular path. A pot is normally adjusted by a rotary shaft, or by means of a lead screw that is adjusted by a screwdriver.

Variable resistors may have carbon, cermet, film, conductive plastic, or wire-wound resistor elements. Small, precision (sometimes multi-turn), adjustable resistors are called trimmers or "trimpots" and are used for fine adjustments and calibrations in low current applications.

Most variable resistors have a linear "taper," in which the change in resistance is linear with wiper travel. Audio controls are available with an audio taper, in which the change in resistance is logarithmic with wiper travel (*Fig. 7*). Noise voltage is more complex than with fixed resistors due to the additional noise variable introduced by the wiper.

Carbon composition elements are the most common type of potentiometer. They are analogous to carbon fixed resistors in that they use a mixture of powdered carbon and organic binders, fired on a ceramic substrate. They can be easily formulated in any type of taper (linear, audio, log, reverse log, and so forth).

Although they have (theoretical) infinite resolution, they change value with temperature and humidity. Excessive humidity may cause an increase in resistance. A composition resistor element may wear after extensive use, and the wear particles may cause high resistance short circuits.

Cermet elements are a mix of fine metal and metal oxide particles. They have excellent temperature stability, but low adjustability wear. This limits their use to trimmer pots in which frequent adjustment is not required.

Conductive plastic is made by compounding a low-friction plastic (polyethylene and/or polypropylene) with carbon and other materials. The resulting resistor element is a very smooth film, allowing extremely high resolution, low noise, and extended service life. Many high-end audio controls—in both linear and audio taper—use conductive plastic elements.

Wirewound elements are made the same way as fixed wirewound resistors, and have the same high degree of temperature stability. Since each turn of wire adds additional resistance, they

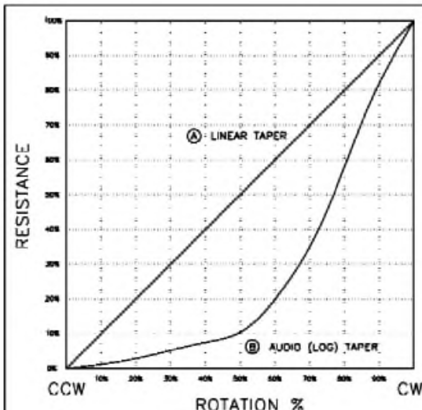

FIGURE 7: Potentiometer tapers.

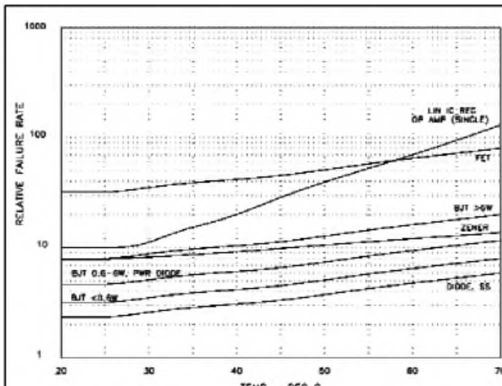

FIGURE 8: Semiconductor failure rates versus temperature.

change value in discrete steps as the wiper moves from one winding to another. Disadvantages are a lower life expectancy and higher electrical noise due to abrasion. They are only available with a linear taper.

Multiturn trimpots have two advantages over single-turn types. First, the lead screw mechanism uses a slip clutch at each end making it impossible to overpower the end stop and damage the pot. Second, the small adjustment screw opening in the case (sealed by an O-ring in the better pots) keeps dirt and contaminants out of the wiper/element.

The number of contact points directly determines the current-carrying capacity of the contact, its noise, the apparent contact resistance, and its operating smoothness. Higher quality pots use a "multi-finger" contact group of aligned contact springs, each providing a discrete point of electrical continuity on the resistive element. Inadequate contact force will result in poor contact resistance, while too much force will cause excessive contact and film wear, decreasing the useful life. While lower-cost pots use base metal nickel-chromium and nickel-tin alloy wipers, precious metal alloys give the best electrical performance and life.

In instances where the pot wiper is connected to one end of the element, designers must consider the effect of an open wiper on circuit operation. The bias spreading adjust pot in power amplifiers must be positioned so that an open wiper will decrease the bias current (away from Class-A) to prevent overheating the output devices.

Motorized potentiometers are often used in stereo equipment with remote-control capabilities. These consist of standard potentiometers with a drive motor integrated onto the pot stack. Multichannel home theater receivers use digital integrated-circuit pots.

SEMICONDUCTORS

The manufacturing process for semiconductors of all kinds is carried out in a tightly controlled clean-room environment. Despite the best efforts to prevent contamination, semiconductor manufacture is a highly specialized multi-step process, and minute flaws can be introduced along the way.

Semiconductor materials provide the ability to control their conductivity by controlling the number of free electrons in the material. The leakage current flow in a semiconductor increases exponentially with temperature. In addition to the leakage current effects, the aluminum metallization used to make internal circuit connections reacts with the other materials. This can cause electromigration and degrades the connections, eventually causing them to fail.

Uncontrolled leakage current flow results in internal heating of the device

TABLE 1
MAXIMUM CONTINUOUS WORKING VOLTAGE FOR CARBON COMPOSITION RESISTORS

P-RATED (W)	MAX V DC OR V RMS	R-CRITICAL
1/8	150	180k
1/4	250	250k
1/2	350	245k
1	500	250k
2	500	125k

junction, which in turn produces more leakage current. The device may reach a point where, if the external impedance does not limit the current flow, the device will fail due to thermal runaway. The monolithic structure of modern semiconductors ensures that the die itself will always fail short. If sufficient fault current is available, the device may then fail open.

This "burnout" is often a catastrophic failure (eruption) that masks any evidence of the original cause of failure. The current required to open a device may range from as little as 60mA for a low-voltage UHF Schottky diode to kiloamps for a really large diode or thyristor.

Photoelectric effects can cause improper operation if light gets through the device package due to voids or other defects. Photocurrent increases with higher light levels.

Other degradations can cause misoperation: decreased diode peak inverse voltage (PIV), decreased gain, increased noise, increased reverse leakage current, and increased junction voltage drop. A change in junction capacitance or reverse recovery time seldom occurs.

Low-noise devices are characterized by larger junction areas than general-purpose devices. If the gate of a FET or FET-input op amp is driven positive so that it is caused to draw current, the noise performance of the device can be permanently degraded. Precision devices that are exposed to electrostatic discharge (ESD) or high levels of RF energy can also be permanently degraded, even if failure does not occur. It's not good practice to use the microwave to dry the Walkman™ you dropped in a puddle!

Figure 8 shows the relative failure rates for semiconductors versus temperature, while *Fig. 9* shows the relative failure rates for semiconductors versus electrical stress. Note the vertical axis scale. Semiconductors have failure rates an order of magnitude higher than those of passive components.

Electrical stress is not used as a failure rate factor for microcircuits. You must calculate junction temperature and apply the temperature factors. Digital ICs have their electrical stress pre-defined by their fixed 3.3 to 5V DC supplies.

DIODES

The principal failure modes encountered in diodes is a short. Small point contact or RF diodes may open at the lead attach point, but the most common die failure is a short. As with most failure modes, if sufficient current is available a shorted diode will burn open.

ZENERS

Zener, or regulator, diodes work in two different ways. True zener diodes are available only below 7V DC. The farther you go below 5V DC, the worse a zener diode performs. They have a negative temperature coefficient (TC), and the zener voltage changes considerably with current. Zeners with voltage ratings of 5.1 to 5.6V perform best, with near-zero TC and very little change in reverse voltage over a fairly wide range of currents.

Zener diodes rated more than 7V DC are called avalanche diodes. An avalanche diode's voltage varies with current, and the TC increases in the positive direction with increasing voltage ratings. If zener voltage is a critical parameter, temperature-compensated types with tight tolerances are available.

SMALL SIGNAL TRANSISTORS

Transistor degradation can be characterized by high leakage, high saturation resistance or voltage, low beta or transconductance, or low breakdown voltage. PNP bipolar junction transistors are subject to inversion, a phenomenon associated with high temperature reverse bias (HTRB). A P-type region forms at the base, resulting in an extension of the collector region, causing high reverse leakage current.

POWER SEMICONDUCTORS

The limitation on the power-handling capability of diodes, bipolar transistors, and MOSFETs is the peak junction temperature. Power devices have all the degradation mechanisms of small-signal devices, with added deterioration introduced by thermal cycling. Power devices can also fail when the mechanical connections to their heatsinks degrade (reduced mounting screw torque, deterioration of the heatsink compound, damage or oxidation at the mounting surfaces).

Bipolar power transistors, being minority carrier devices, have an additional limitation, called second breakdown[4]. *Figure 10* shows the active-region safe operating area (SOA) curve with its four limiting regions for a 2N3772 NPN power transistor at 25°C.

The inner DC limit is bound by the horizontal collector current line (20A), the diagonal thermal limit power parabola line, and the vertical collector voltage line. The thermal limit lines move outward for single pulse applications of decreasing duration, based on the thermal time constant of the transistor.

You can also see additional breakpoints in the 500ms and 100ms pulse lines, which represent the second breakdown limit, where V_{CE} must be reduced below the level that is required for purely thermal reasons. Second breakdown is caused by the tendency of a bipolar transistor to develop localized hot spots when the collector current flow becomes concentrated at high V_{CE}. Exceeding the second breakdown limit causes thermal runaway in the collector region, which can destroy the device.

The SOA curve is based on the ab-

TABLE 2
DERATING FACTORS FOR COMMON ELECTRONIC COMPONENTS

COMPONENT	PARAMETER	DERATING FACTOR
Cap, ceramic or mica	Voltage (peak)	0.80
Cap, film or paper	Voltage (peak)	0.70
Cap, aluminum	Voltage (peak)	0.85
	Ripple Current (average)	0.85
Cap, tantalum	Voltage (peak)	0.78
Diode	Peak Inverse Volts (PIV)	0.80
	Current (average)	0.87
Diode, zener	Power (average)	0.90
IC, linear	Voltage (peak)	0.80
IC, regulator	Voltage (peak)	0.80
	Current (average)	0.80
	Power (average)	0.50
IC, digital	Fan-out, frequency	0.85
Resistor, film	Power (average)	0.80
Resistor, wirewound	Power (average)	0.67
Resistor, variable	Power (average)	0.70
Relay contacts:	Current (peak)	
Resistive load		0.75
Reactive (L or C) load		0.40
Motor Load		0.20
Filament/lamp load		0.10
Transistor, FET, or bipolar	Voltage (peak)	0.80
	Current (average)	0.80
	Power (average)	0.50

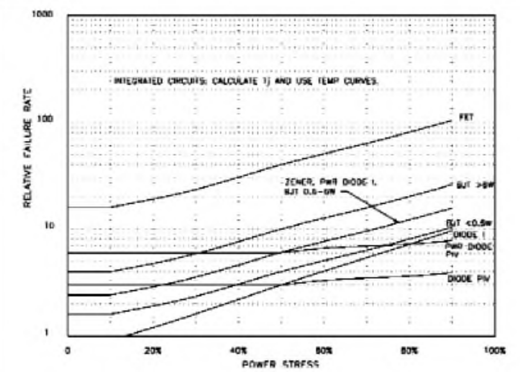

FIGURE 9: Semiconductor failure rates versus electrical stress.

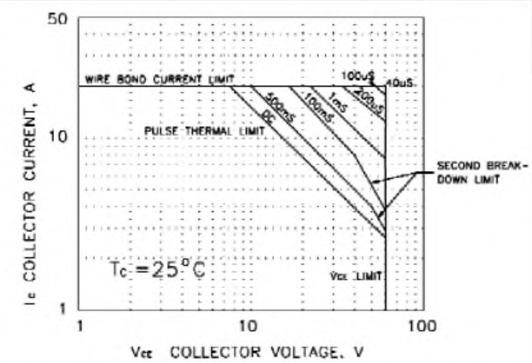

FIGURE 10: Active-region safe operating area curve for 2N3772.

solute maximum voltage, power, and current ratings for the device. Operating anywhere on the power limit line will produce the maximum junction temperature, which is not a good practice. The derating factors in *Table 2* should be applied to this curve. Further power derating is required for operation above 25°C, and some manufacturers provide additional ambient limit lines for the thermal limit portion of their SOA curves. You can see that making the most effective use of a power transistor without seriously degrading its reliability is not a trivial task.

MICROELECTRONICS

The failure rate in *Fig. 8* for op amps is for an average single unit. The actual failure rate, as with all integrated circuits, is based on the number of transistors in the design. FET-input devices are very sensitive to static electricity.

Now that the features and interconnects in digital integrated circuits have gone sub-micron (less than one-millionth of an inch), with more and more layers, electromigration has become a real problem. Despite the fact that the current in these devices is milliamps or less, the current density (amps/square mil) is high enough that the moving electrons collide with the aluminum metallization, producing molecular level holes in the chip's interconnections. When the current density in the reduced area becomes high enough, the aluminum can melt, causing a failure of the device. Another failure mechanism occurs when the displaced aluminum atoms cross over an area of insulation in the chip, causing a short circuit.

Part 3 wraps up this look at electronic circuit components. ❖

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The Revenge KR842VHD Amplifier

Don't tell this author tube technology is passé. Using a latest-generation tube from Czechoslovakia, he's put together a high-performance SE amp. **By Mauri Pännäri**

PHOTO 1: The completed amp.

This basic high/medium power SE amp uses the best-quality components available on the market, such as the new KR842VHD high dynamic power triodes and Lundahl C-core transformers. I put special effort into choosing the driver tube: the old telephony workhorse C3m in triode mode and capacitor-coupled. The pre-driver 12BH7 is a proven low-gain component for good and detailed sound, too. Simple, although not very cheap, the result is a quiet, dynamic unit, sonically different from the usual 300B designs.

INTRODUCTION

Hi-fi (and lately, tube hi-fi) has been my hobby for about 35 years. There have been many things to see—from some clever innovations to circuit and theory nonsense. I thought, as many others probably have, that the tube times were totally over and the development ceased until the '80s, when I found Jean Hiraga's book of Japanese tube trends in my hand.

At that time I was partly involved in tube technology at Telecom Finland (now Sonera Corporation). I headed the repair work of professional speech transmission measuring equipment, older oscilloscopes, and other tubed test gear in the repair center lab. This kind of equipment used the best tubes and tube types available on the market, without consideration of cost.

Many used samples were available, so I began to experiment with some promising circuits and came to the well-known single-ended push-pull (SEPP) bridge design, which was invented here in Finland in the mid-50s by a legendary national radio and hi-fi guru, Tapio M. Köykkä (1911–1994). SEPP can be traced to his two patents from that time, which were immediately copied in the US and Germany by Electro-Voice and Philips. This background is not commonly known, and the origin of the concept is most often attributed to Electro-Voice. This circuit is the mother of current OTL tube designs of a particular category.

My then new SEPP 4W 6BX7 triode construction was such an excellent performer that I had to sell my Yamaha 1010 top receiver right away. I thereafter played happily for many years with my SEPP (it's still in full service at my summer house). However, around that time I happened to have some 4300A carrier-wave transmission telecommunications tubes made by Standard Telephones and Cables (STC) in England, and was inspired at last to try a single-ended design with these old direct-heated triodes, which are very scarce today. This is actually the same tube as the 300B with identical electrical STC data to the STC 4300B.

I also saved some STC 4328A drivers from work. This is like the WE 310A but with a 7.5V heater. Both of these STC types were unused spare parts for telephone 12-channel open-wire carrier-wave telephony transmission amplifiers from the '50s. The 4300A was the open-wire line transformer power feeder (I have a plan to build a STC-tubed replica of the WE 91B amp in the future).

TUBE SELECTION

Over four years ago I selected my own SE driver tube for a 300B-type DHT by listening to a variety of tube types and manufacturers in my capacitor-coupled circuit, as samples were easy for me to get. At last I decided: Siemens C3m in triode mode with excellent sound and good swing capability. This long-life steel pentode was common in the '50s and '60s as a channel amplifier in German speech carrier-wave telecommunications equipment and also in measuring equipment, despite its expense. The market price now is upwards of $50, available from major dealers. The steel-shield construction holds a glass envelope within and has a mesh anode structure.

I ordered my Tango U-808 output transformers from Japan, because there

ABOUT THE AUTHOR

Mauri Pännäri (M.Sc. e.e.) has worked 30 years in professional telecommunications repair, construction, measurement, planning, and research at Telecom Finland (now Sonera Corporation). Hi-fi has been his hobby from the age of 15, when he built his first tubed amp with 2× EL84s. He enjoys classical music, as well as rock and roll, as played through his vintage 300 ltr 15″ Tannoy Monitor Golds. He mainly spends his free time with his family at a well-stocked inland lake in the southern part of Finland.

were none available in Finland at the time. I have been happy with my 300-type SE for years, although I've tried various other circuits beneath.

During the 1999 Hi-fi Fair in Finland I happened to meet Dr. Riccardo Kron and his wife, founders and owners of KR Enterprise, which was offering some of its top Czech amplifier models in Finland. We discussed at length DH tubes, vacuum transducers (the high power tube component from KR), and using them in constructions. After that, my wife and I visited the Krons in Prague, seeing and hearing much about DH tube manufacturing in the old Tesla tube lab premises they now own.

In addition to the existing models, we listened to a new tube, the KR842VHD, which was just in the release phase in the factory, and is now used in one of KR Enterprise's top amp models. It is also available to constructors worldwide via dealers (also here in Finland) as a next-generation 300B replacement or a basis for something better. I was very impressed by its sonic potential.

I decided to build a totally new SE construction, which I present here (*Photo 1*). I was, however, curious and plugged my new 842 tubes into my old 4300 design. After trimming the DC heater voltage for bigger current, they flashed their sonic potential with 300V/70mA supply only.

The dynamics I heard again caused me difficulty in choosing output transformers of comparable quality for the new construction. I decided to purchase the whole iron set from Lundahl (Sweden), as I noted some good references in various articles and on the net, and the prices were reasonable. Mr. Lundahl was also very helpful and the delivery was quick.

DESIGN

The KR842VHD output tubes work here (*Fig. 1*) with 420V/95mA with a fixed bias of about –62V, which is provided with a separate 3VA transformer via 180kΩ resistors to the grids. The rectified 5.9V AC heater voltage is just right and will accommodate other 5V/1A 300-type tubes as well. The tube is loaded with Lundahl's LL 1623, a C-core multi-winding quality SE iron.

My version is specified to 90mA max, but can be used, according to Mr. Lundahl, for even 105mA flux with essentially higher main inductance than the following standard version with a bigger air gap for 120mA. Its numerous secondaries are coupled for 3kΩ anode load with a printed circuit board, "PCB_C," available from the manufacturer (this can be done by wiring also). There is nearly 2dB local feedback by connecting the output secondary to the cathode. This is a trick that non-feedback purists can use with positive properties and was long ago used even with a 1:1 ratio in some commercial push-pull designs, such as Quad, MacIntosh, and Finnish Köykkä. It increases the DC magnetization flux of the output transformer by about 5%, which is marginal.

I have had very good results with high-mu tubes in my other SE trials. If you have mu around 20, you can have 5–7dB of feedback and a very fixed bass end from a modest power SE. Here the practical influence is quite small with the tube-mu around 4. This tube has the

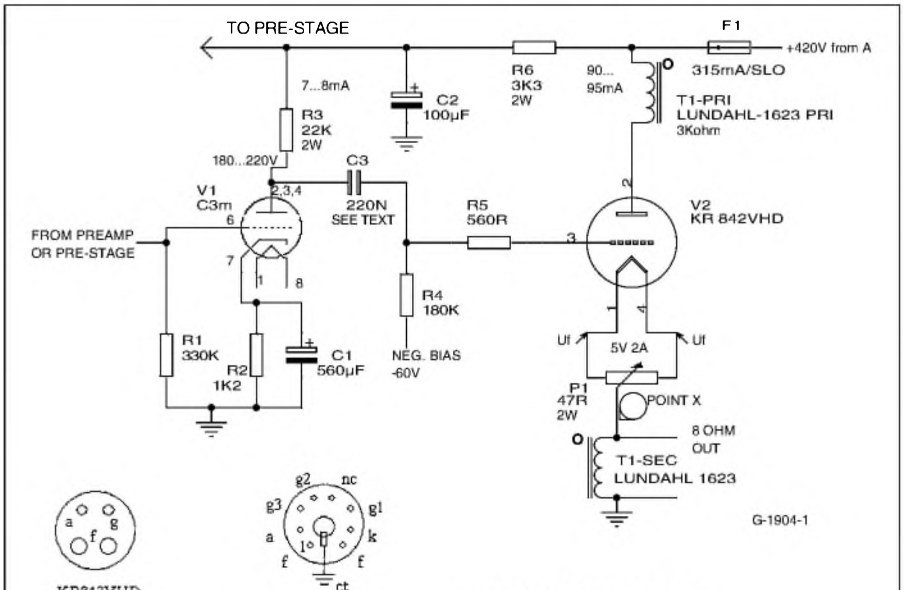
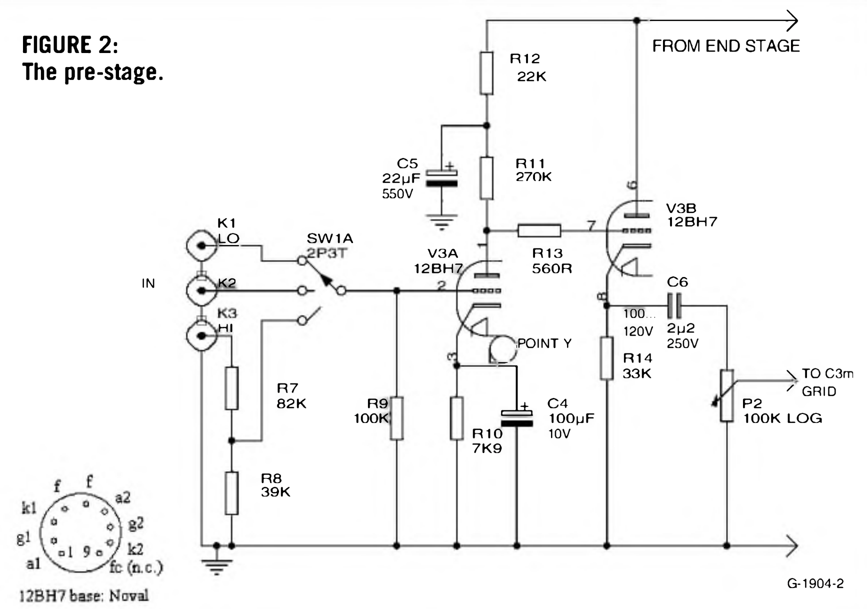


FIGURE 1: The end stage.

FIGURE 2:
The pre-stage.



same base configuration as 300B-types with the UX4 base, although the inner construction is totally different with eight separate flat cathodes.

CIRCUIT OPERATION

The signal coupling capacitor is of great importance. I used proven Wima FKP1, 0.22μ/1250V foil polypropylenes ("blue" series), which were readily available. You can use your favorite audio special capacitors here, maybe spending several tens of dollars, if you consider it worth it.

I saw no reason to change my good driver tube and concept, after the preliminary short successful trial with the 842s to check the combination sound. This stage with C3m as triode is conventional with an anode load of about ten times Ri and reasonable current, to feed the output tube, which here works in pure Class-A1, meaning you don't go to the grid current drive region on purpose. Actually, the driver I use here is C3o, which is the 6.3V version of C3m, but these are extremely rare and you might better use C3m by adding a 2× 3V/3VA aid transformer, separated 3V halves in series with the main transformer's series coupled 6.6V windings to get near 20V AC (note and turn the 3V phases if necessary).

The stage has a gain of 15 and good linearity, so sensitivity around 3V RMS is obtained at the grid for over 45 RMS volts out for the output tube. This tube needs a special base called a "locktal." For triode mode, just connect electrodes a, g2, and g3 straight together (pins 2, 3, and 4). If you have a suitable good preamp with enough output capability, you can build these two stages per channel as a mono or stereo end block, as was my earlier mentioned 300-type design. An extra pre-stage is still needed if you want to have a complete amplifier with line level inputs, as I decided to include here.

The pre-driver stage (*Fig. 2*) does not need much gain. I have tried SRPP with a wide variety of tube types in the past, as well as semiconductor constant-current loading, and decided here to keep on the basic gain stage with a big value anode resistor using a sonically well-proven tube. So I turned again to a favorite of mine—the Philips 12BH7. With low current of 1mA from 350V and an anode load of over 40 times Ri, the first stage fits straight and practically as unloaded, to a 3–4mA cathode follower using the other triode in the same tube. You will have some 100–130V DC at the cathode as a guide value.

The first stage gain is about 12, offering a final sensitivity of 300mV. I provided one input with 10dB lower sensitivity (and a little bit lower input impedance) for high output sources (e.g., a CD player without a volume control), because it is against my design principles to put the volume control before the whole tube gain chain and have all the front-end noise and possible residual hum present at the output all the time. So the 100kΩ volume potentiometer lies behind the cathode follower coupled with a big polypropylene (Wima "red" MKP used here), a proven solution and good compromise (you would have the capacitor in a separate tube preamp in most cases, anyway). This allows me also to parallel-record to my old reel tape (the rec. connectors wired from the top of the volume control are not shown in the schematic).

You have only a few volts signal normally at the cathode output related to the space of several tens of volts available, with respectively small distortion. If you choose, you can put a cermet pre-

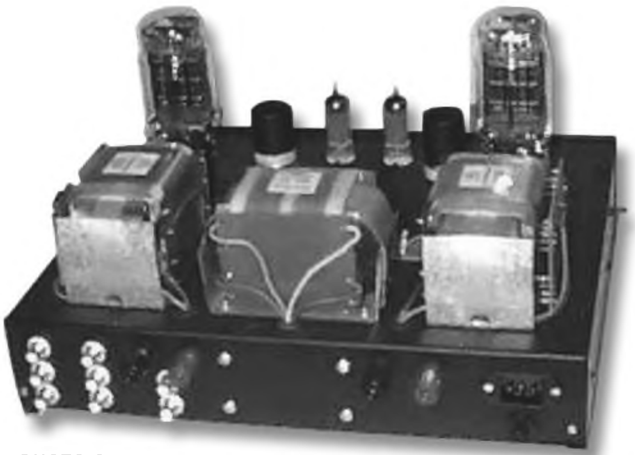


PHOTO 2: With top cover removed.

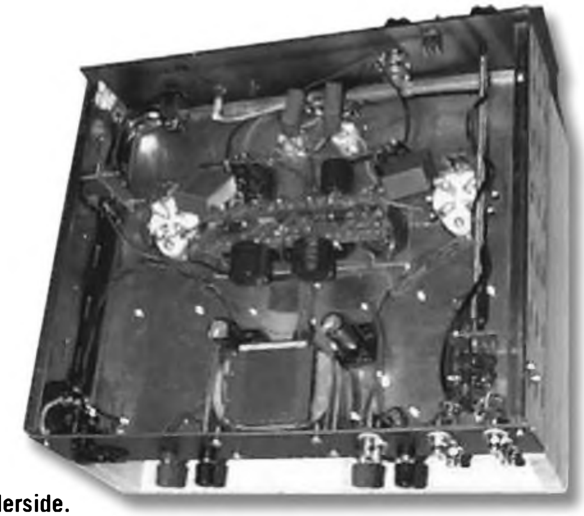


PHOTO 3: Amp underside.

set/balance potentiometer in place of the voltage divider to adjust the input to the maximum level used frequently.

There is not much more to say about the basic circuit. It is elementary and proven with no tricks. Sound relies on top-quality components and tubes and their sonically proven match.

CONSTRUCTION

I constructed the amplifier around an old 19″ rack case, which I narrowed and shortened, and used a new cover/back of 1.5mm aluminum as the mounting template, painted black. You can use any suitable case (such as a Hammond), or make your own. The Lundahl transformers are no Miss World beauties (*Photo 2*), so I constructed an aluminum cover for them as shown in *Photo 1*.

The Lundahl LL1638 choke is under the chassis. All irons have 90° physical axis relation, more as a good design factor for these low leakage C-core irons. The input selection switch is located near the inputs in the back corner behind a rotation bar, so no screened wires are needed there (*Photo 3*). You could also consider using small relays for this purpose, activated by a front panel switch.

The power supply is conventional (*Fig. 3*). Separate 5.9V windings from the universal (115/230V) Lundahl LL1648 mains transformer feed the output tube heater rectifiers. I finally replaced the 47Ω potentiometer (P1) with two 33Ω/2W fixed resistors, but you can install these at the outset. The other two 6.6V heaters are in series—the center earthed—to feed 13.2V for the driver and front end tubes.

You need a small 2× 3V/3VA aid transformer to provide 3V extra to both ends of the center earthed 13.2V for the C3ms, reaching over 19V total; 19–21V is acceptable here. The grid bias is provided by a separate 30V/3VA transformer, and mine gives around –60V DC with no load. This depends on the mains voltage and the transformer and may need some trimming.

Typically these small PC-board transformers provide about 35% overvoltage with no load as used here. They can be conveniently fixed by a strip of PC board with holes at the ends and two 40mm-long bolts upside down against the case, wires welded directly to the pins. Both the small transformer primaries are behind the fuse F1—although not drawn in the schematic—from where I also dropped the mains switch and pilot light.

You can cut at the point X in the output amplifier schematic (*Fig. 1*) to add a 10–50Ω/1W aid resistor in series to lift the negative bias voltage if needed, roughly with –1V/10Ω add. It may be a good idea to put some 100Ω here in the initial phase and then come down to reach the anode current value in the region of 90–100mA. I finally left 33Ω for 95mA, not wanting to change to a new 36V AC secondary bias transformer. This can also be done, but needs a voltage splitter/potentiometer for adjusting for the bias needed.

The aid resistor also has a bonus of stabilizing the current to some degree, and I noticed no sonic or measurable side effects as left unshunted. The semiconductor-rectified anode supply uses 400V capacitors for switched-mode power supplies in series with paralleled voltage dividers, forming a pi-filter with the choke. The driver and the pre-stage have additional RC-filtering. Purists can use tube rectification here with possible choke input, selecting a different mains transformer accordingly and getting a delayed anode voltage as a bonus.

A trick I've recently seen is to use two semiconductor diodes and two tubes or a double tube in a Graetz bridge arrangement with all the good and bad properties of tube rectifiers. You need only one anode voltage winding then. GZ34 could be good, or you could use two color TV boosters, e.g., EY500s. The anode voltage will be about 30–40V lower. I have not tested them in this amplifier. You could also try hexfreds.

I have switched mains on to my old 4300s (without any anode delay or heater softening) several times daily for nearly five years with exactly the same kind of power supply for 300V as here. It has been useless to research information about the meaning of delayed anode voltage instead of the "would be good or necessary" nonsense frequently offered, and this gives me the impression the subject is not very meaningful in our 80-year tube time frame. The same holds for soft heater start-up. I seldom saw them in professional equipment where selenium or diode rectification provided immediate full anode voltage to "shock heated" tubes, when heading the centralized repair work of tubed telephony and measuring equipment in the beginning of my career.

OPTIONS

Those who choose to, and have skills to experiment, can try some changes of interest. If it is impossible to get C3m tubes, you could try ECC81 or its corresponding telephony type 33A/101K, which was sonically the second best driver for my tastes for my earlier 4300 design. It has a gain of about 40 here and needs different load and biasing. You can try 30–40k at the anodes with 5mA. You can also parallel-couple the halves.

With this tube there is sensitivity without a pre-stage for a CD player. However, I recommend you keep on the C3m for best sonic results. You can, if you prefer, use an SRPP pre-driver built around the two front-end tubes; it was my original idea to install two double tubes here for stereo to maintain flexibility to easily work with various front ends. Cathode-follower haters can leave one tube or parallel-couple the tube halves for a normal gain stage in each channel.

E182CC (7119) is a good, proven tube candidate for the front end, too. Still, you could try semiconductor current generators as anode loads (e.g., "Camille cascodes," which withstand, however, only about 300V). I have prototyped this end amp with an E182CC having 3mA constant-current semiconductor loads from 120V in the test phase, and the result was fine, although maybe with a more "sterile" sound influence. There are a lot of options available, which I leave to you.

After having tried various tricks and possibilities during my years of experi-

TABLE 1

THE REVENGE KR842VHD AMPLIFIER PARTS LIST

(PER ONE CHANNEL, ALL RESISTORS ½W UNLESS OTHERWISE NOTED)

END AMPLIFIER	
R1	330k
R2	1k2
R3	22k/2W
R4	180k
R5	560R
R6	3k3/2W
P1	47R/2W wirewound linear potentiometer, see text
F1	Fuse T315mA (slow)
C1	560µF/16V electrolytics
C2	100µF/550V electrolytics
C3	0.22µF/1250V polypropylene Wima FKP, see text
T1	Lundahl SE transformer LL1623/90mA
PRE-STAGE	
R7	82k
R8	39k
R9	100k
R10	7k9, see text
R11	270k
R12	22k
R13	560R
R14	33k/1W
P2	100k stereo log pot cermet or other high quality
C4	100µF/10V electrolytics, tantalum
C5	22µF/550V electrolytics
C6	2.2µF/250V polypropylene Wima MKP 10
SW	2 × 3 switch, e.g., Elma gold-plated
POWER SUPPLY	
T1	Lundahl universal mains transformer LL1648
L1	Lundahl 10H anode choke LL 1638
T2	Transformer 2 × 3V/3VA out, see text
T3	Transformer 30V/3VA out, see text
B1	Diode bridge 1000V/4A, see text
B2, B3	Diode bridge 100V/20A
B4	Diode bridge 200V/1A
C7–C10	330µF/400V electrolytics
C11	100µF/100V electrolytics
C12, C13	10mF/16V electrolytics
R15–R18	100k/2W
F2	Fuse T1A (slow)

menting, my ears felt as though they were returning home with my 12BH7 "normal" stage here. For skilled experimenters the field is endless, and there really are nuance differences in sound, the best remaining unfound (luckily). Also, the working point of the voltage gain stage makes a difference.

High-current enthusiasts may increase the first stage current. I have noticed special "nuance switches" in some designs for pre-stage bias. Here, however, you should not lift the cathode follower's DC voltage much over 100V to remain safe with the maximum positive DC + AC under 200V, as specified for the 12BH7 tube as heater grounded! For other tubes, see their data sheets for safe operation in this respect.

There is the point Y in the preamp schematic (*Fig. 2*). If you cut there you can eliminate the cathode capacitor and experience a little bit different sonic result—at the expense of smaller, but sufficient, gain. My last trial was with the Philips Miniwatt E80CC, which fits pin-to-pin, but you must change the cathode resistor R10 in each channel from 7.9k to 3.4k to achieve 100-110V at the anode. This tube sounds very relaxed here without the cathode bypass. This is an excellent preamp, even though it was built separately for different purposes.

If you want to use other UX4-socket 5V output-tubes, there is a wide variety to try. The sound is typically different in each case. Remember to adjust the bias for the anode current needed; the heater is sturdy as such for any current at 5V up to about 2.5A DC.

With these good irons and driver tubes, I recommend the new KR842VHD high dynamic tubes for something sonically better than usually heard. If you are not sure what to do, follow the types and values presented here; the result is good. Remember the safety factors!

The bandwidth of the amp is from 10Hz to over 30kHz within 1dB. I don't like to go any further than this.

I had the opportunity to listen to the best KR amplifiers in Prague and also here at a local dealer (Ideale High Fidelity, Finland), and to me they have their recognizable, detailed, and more dynamic sound, different from my traditional 300s. In my opinion, my design has reached the same authoritative level, which fits my purpose. The power, over 13W, is also more than sufficient for my 15″ 95dB/W Tannoy Monitor Golds, allowing play within the first and cleanest SE watt in normal listening and also reaching live-like orchestra sound frontiers when necessary. ❖

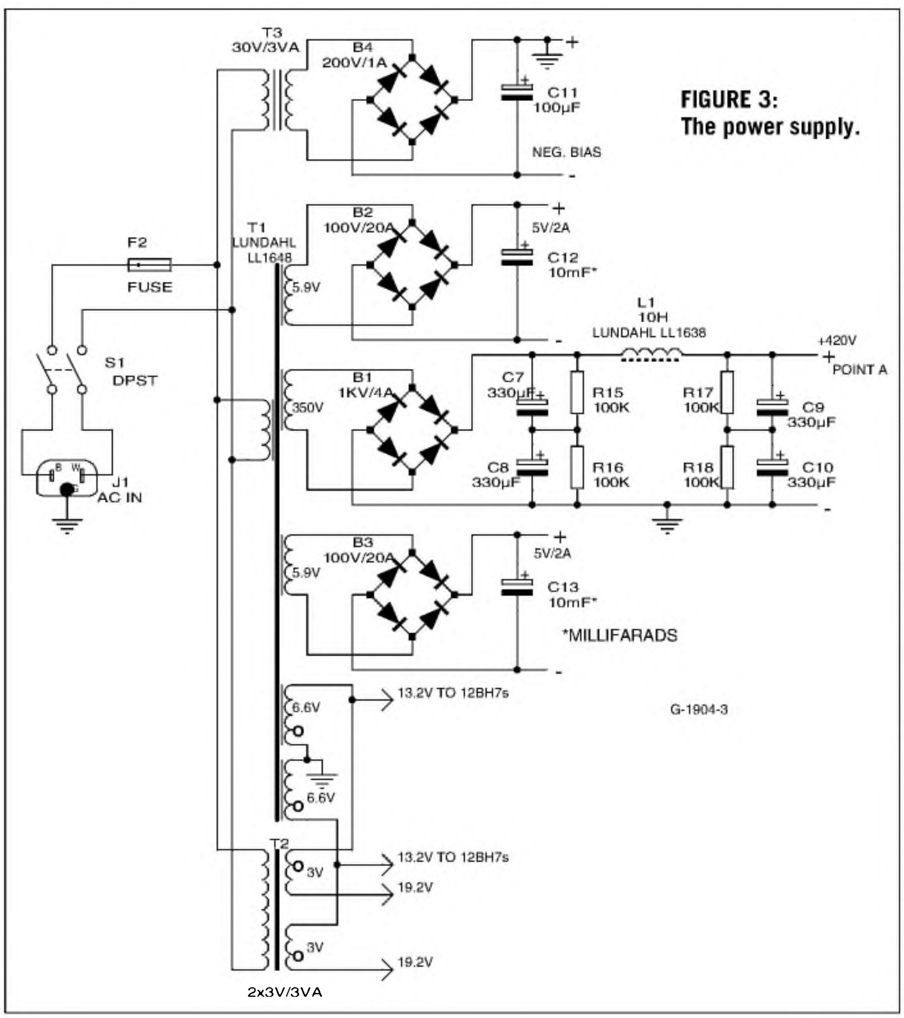


FIGURE 3:
The power supply.

COMPONENT SOURCES

www.kr-enterprise.com
KR842VHD tubes, including list of dealer sources

Billington Export Limited
Gilmans Industrie Estate
Billingshurst, West Sussex
RH14 9EZ, England
billingtonexport@btinternet.com
(see also advertisements in *GA*)
C3m tubes/bases

www.lundahl.se
Lundahl transformers, including list of dealer sources

A First-Order 3-Way

Part 2

Now that we've whetted your appetite with the design and testing of this unit, it's time to consider the construction details. **By J. L. Markwalter, Jr.**

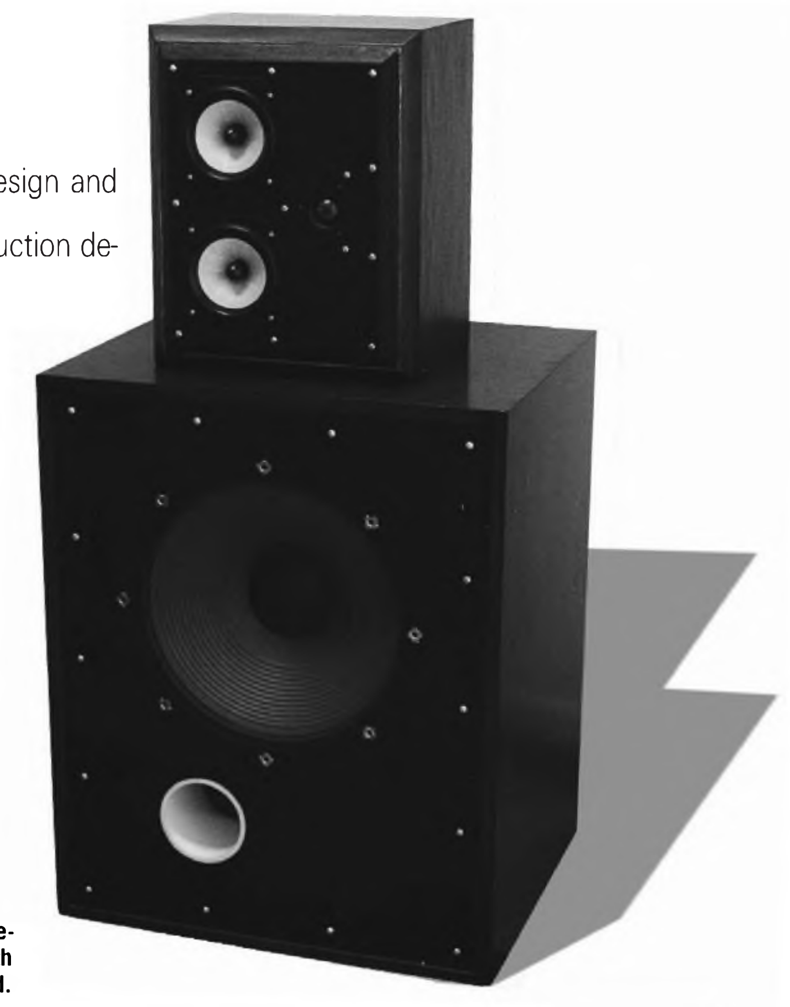


PHOTO 1: Three-way system with grilles removed.

Both enclosures are constructed of ¾″ plywood. I used lauan plywood because it is not prone to internal voids that arise from knots in inner plies and is made up of many plies. Having many plies makes any voids that may exist less significant. Para-ply is also a good choice for the same reasons.

I do not recommend fir plywood, which is very prone to internal voids because of knots and typically has relatively few plies. Particleboard is not recommended, either, because it is brittle and tends to flake and usually splits when a screw is driven into the thickness. It shouldn't make much difference what kind of ¾″ plywood you use as long as it has many plies and appears to be relatively free of voids

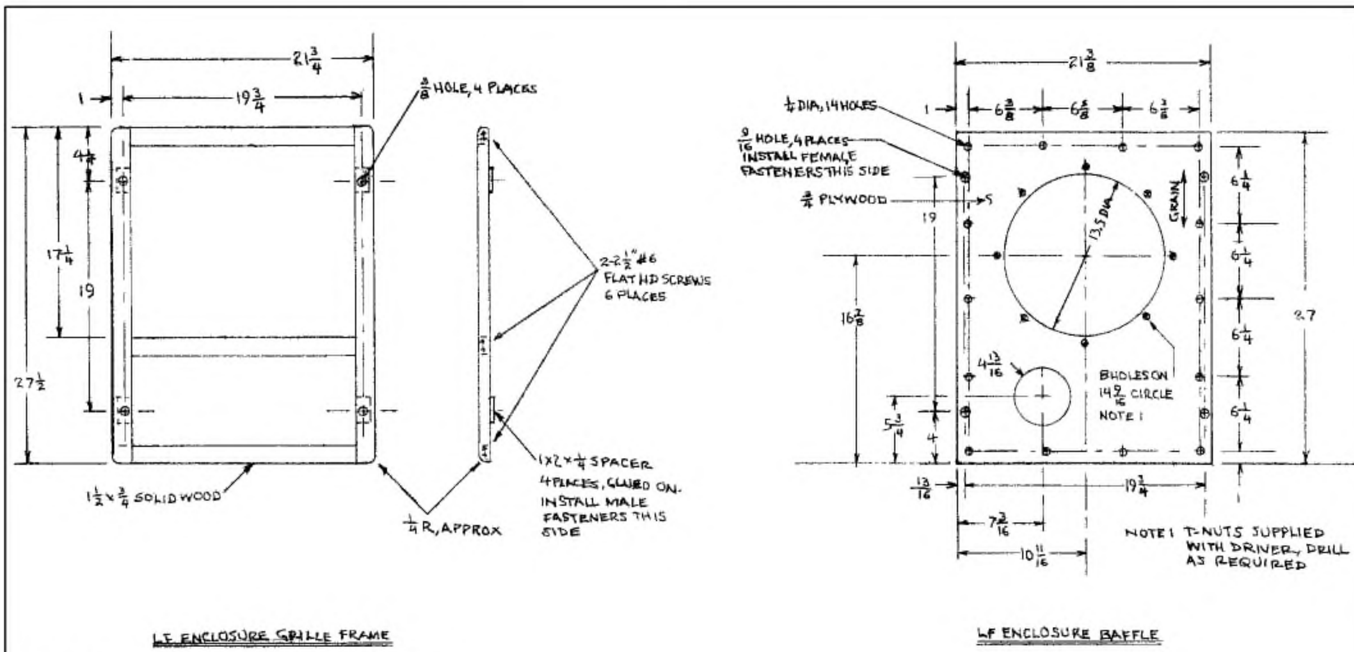


FIGURE 3: LF enclosure grille frame and baffle.

usually apparent in exposed thickness at the edges.

Actual plywood thickness is usually less than the stated thickness; my lauan measured 0.7″, which is 0.05″ less than ¾″. Thickness variation is the reason the drawings are dimensioned with inside dimensions where thickness variations could make a difference.

Observe the grain orientation as shown in the photos. You can make the four enclosures using three sheets of 4 × 8 plywood, with some left over, by judicious layout of the parts.

Glue and screw all joints in the final assembly. In general, place screws no farther apart than 3″ and use 2 or 2¼″ #6 flat-head screws in most places. I prefer Philips-head screws since they are easier to drive with an electric screwdriver. I definitely recommend an electric screwdriver (to save your hands) since you will use 200 or more screws.

Pilot-drill all holes; use a small bit of about 1/16″ or 5/32″. Pilot drilling facilitates holding hole locations more accurately. A combination drill bit is recommended for final drilling of the screw holes. It makes the screw lead hole, the hole for the screw upper body, and the countersink for the screw head in one operation.

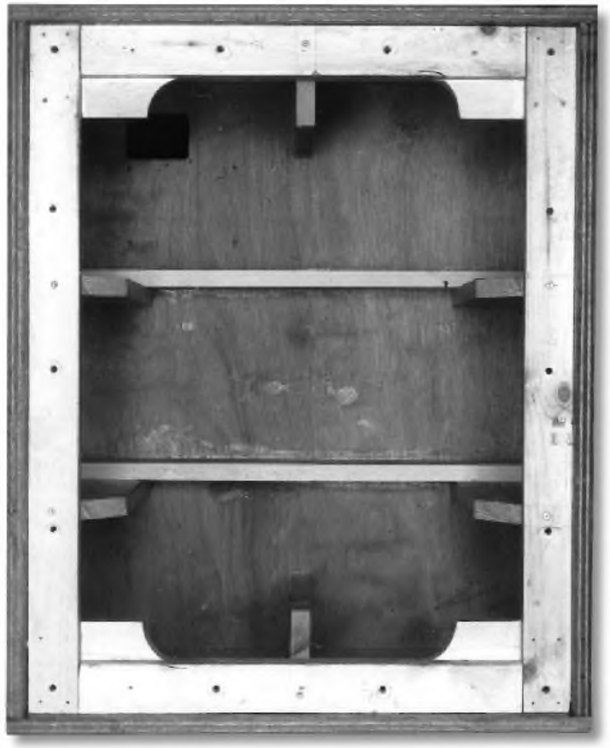


PHOTO 2: Low-frequency enclosure body.

I used Elmer's Carpenters' Wood glue. Buy a quart and use a narrow paintbrush to apply it.

Cut all parts accurately and square, and be sure the thickness is cut square also. A table saw is pretty much necessary. Try to hold dimensions better than 1/32″; this is not as hard as you might think.

Clamp parts during drilling and in final assembly. I use pipe clamps and corner clamps. Check inside front corners with a square before drilling and during final assembly. The enclosure joints should be tight; gaps should not be visible when you screw the parts together. In some cases you may use very slight trimming to close gaps, but if you do so, use the table saw to preserve squareness. As a last resort close the gaps with glue in the final assembly.

Internal bracing, used only in the low-frequency enclosure, is made of 1 × 4 trade size lumber. Actual size is likely to be ¾″ × 3½″, which is not critical. However, the lengths of the braces should be accurately cut and the ends square. Two-inch members, where used, should have accurate width and will require ripping wider lumber down to size.

LOW-FREQUENCY ENCLOSURE

The low-frequency enclosure is a box structure with a removable front (baffle) on which the driver and bass reflex vent are mounted (*Fig. 3*). This arrangement affords flexibility for possible later changes since only the baffle must be replaced, but the box retained. I believe most experimenters will find this a desirable feature.

Refer to *Photo 2*. I recommend that you cut all parts before starting assembly (*Fig. 4*).

Assemble the front parts consisting of stiffeners A, E, and F into a frame as a subassembly. Glue and screw all parts together being sure the outside is square and dimensionally correct. Allow the glue to set before using the frame in the trial assembly.

Do a trial assembly of the top, bottom, and side panels with the rear panel in place to observe fit. Use clamps to hold the parts together (no glue) and drill the screw holes holding the panels together. Put in a few temporary screws. Put the front frame in place temporarily, flush with the enclosure front and hold in place with clamps. The rear panel and the front frame control the squareness of the enclosure body.

If the body parts fit, disassemble and begin final assembly using glue and screws. Do not place the front frame while gluing but keep the interior front corners square with corner clamps or temporary braces while the glue sets.

After the body glue has set, temporarily install the body stiffeners B, C, and D using only screws. The fore and aft stiffeners B and C control the depth of the front frame in the enclosure and also maintain the planar quality of its baffle mounting surface. Make sure the stiffener ends are in contact with the rear panel when driving the screws.

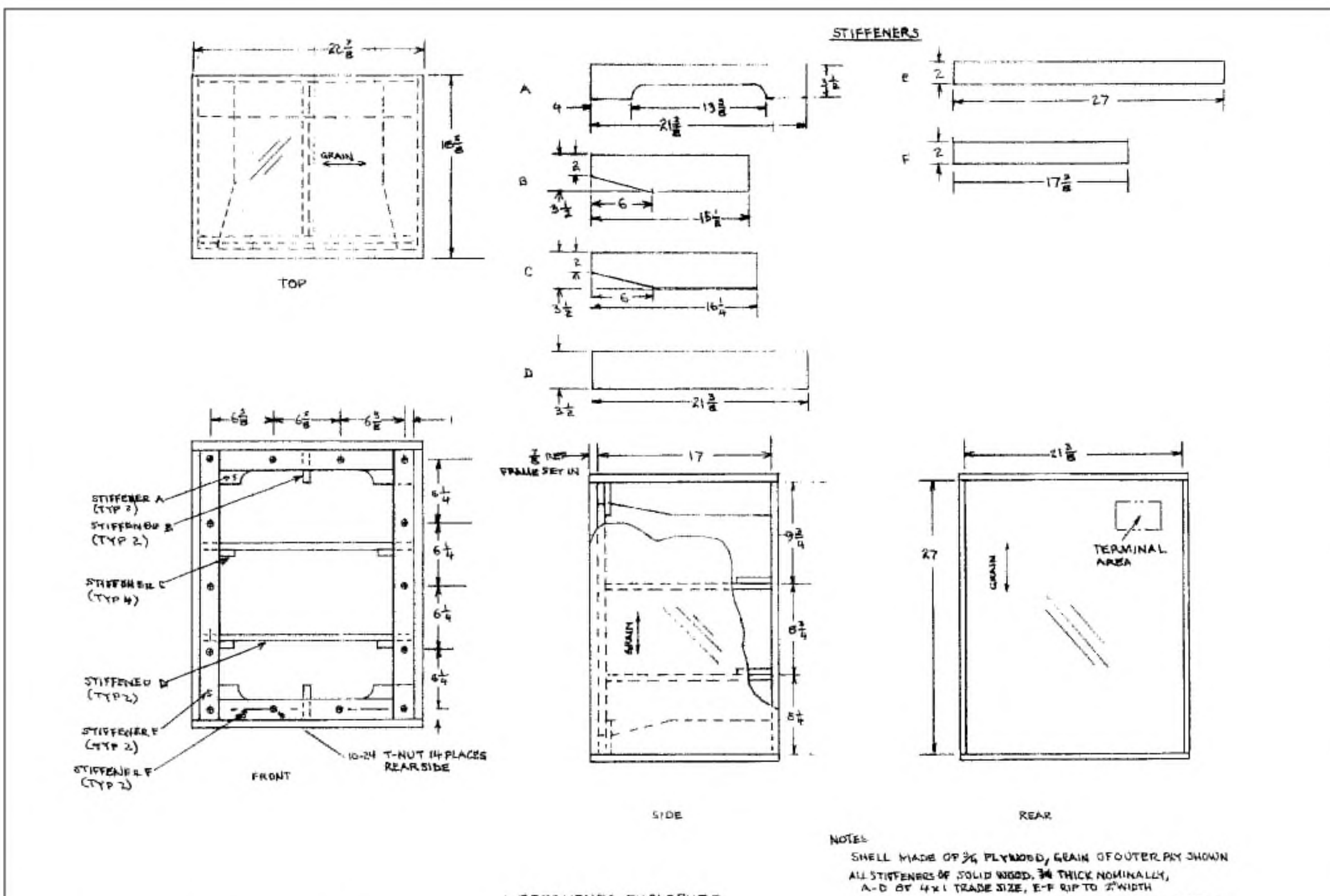


FIGURE 4: Low-frequency enclosure.

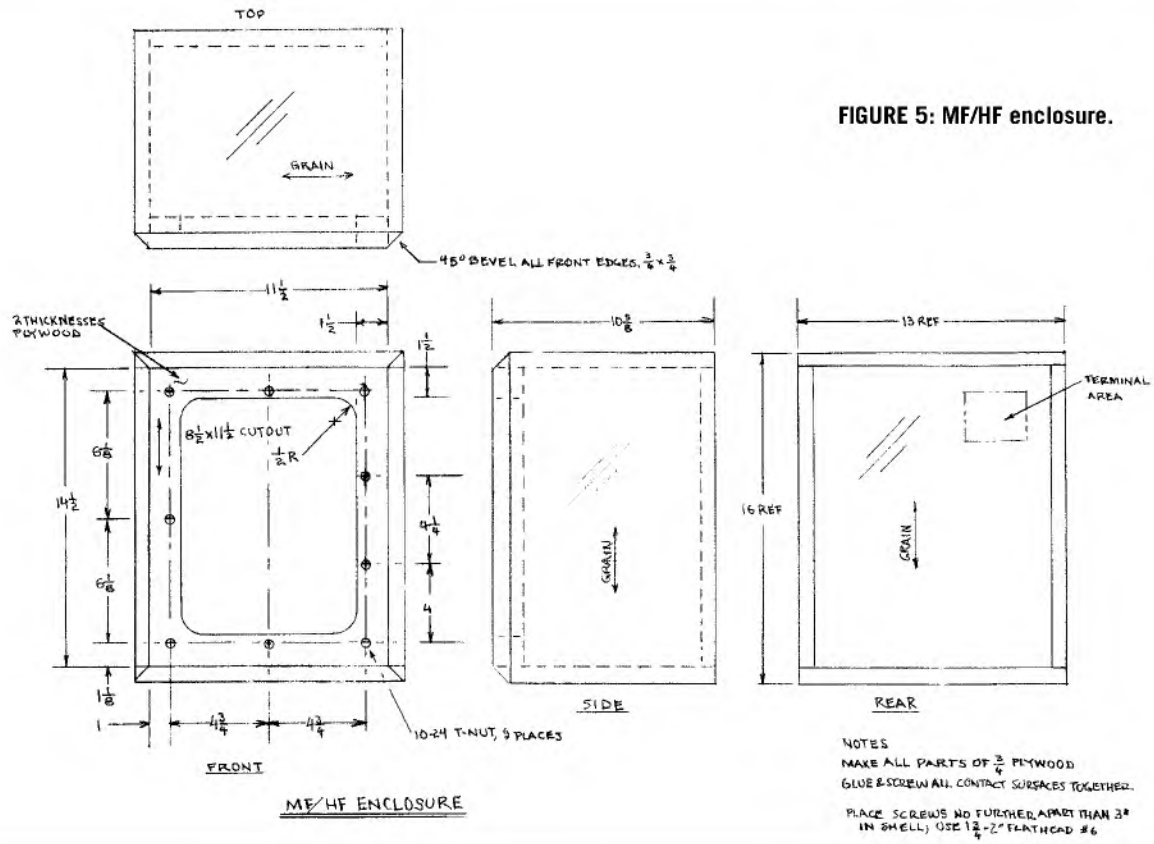


FIGURE 5: MF/HF enclosure.

Push in the frame, which should rest on the ends of all stiffeners. The frame front surface should set in the enclosure 1/8″ ± 1/16″. If for some reason the frame does not make contact with all the stiffener ends, check their lengths as measured from the rear panel.

If all is well at this time, permanently install the stiffeners and the front frame with glue and screws.

Cut the speaker baffle to size. You might need to trim a little so you can freely install and remove it. Cut the holes for the driver and vent pipe, and drill the holes for the driver mounting T-nuts. The baffle mounting screw holes are to clear 10–24 machine screws and are not countersunk. I cut the driver and vent holes with a saber saw.

Refer to *Photo 3*. The frame is made of ¾″ × 1½″ solid wood held together by screws (only). Spacer pads are glued to the rear side for the fasteners. The corners are rounded approximately spherically to make the grille cloth fit nicely.

The grille cloth is somewhat stretchy so it fits easily in the corners. I pulled the cloth around the frame to the back side and stapled it in place.

After you complete all exterior work on the enclosure, install the driver and vent on the baffle. Mount the crossover network inside near the cabinet terminals and leave enough hookup wire slack for convenient connection to the driver. Staple about 1½″ of polyester batting (usually three layers) on all interior surfaces except the baffle. Close up the enclosure.

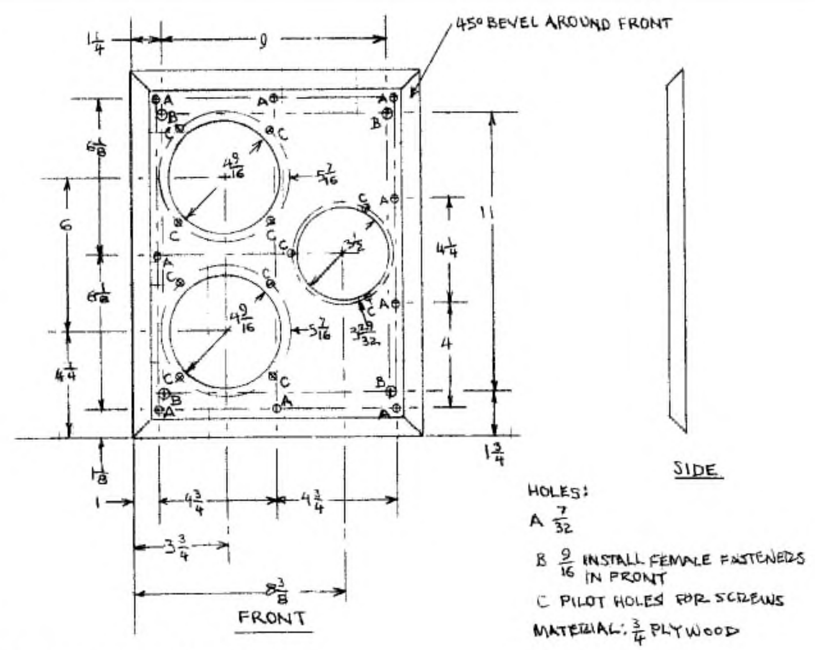


FIGURE 6: MF/HF enclosure baffle.

MID/HIGH-FREQUENCY ENCLOSURE

The mid/high-frequency enclosure (*Fig. 5*) is a simple box structure with a removable front (baffle, *Fig. 6*) on which the two midrange and single high-frequency drivers are mounted. This arrangement affords flexibility for possible future changes, since only the baffle must be replaced and the box can be retained.

Refer to *Photo 4*. I recommend that you cut all parts before beginning assembly.

Assemble the parts making up the front frame (*Fig. 7*) by gluing its two

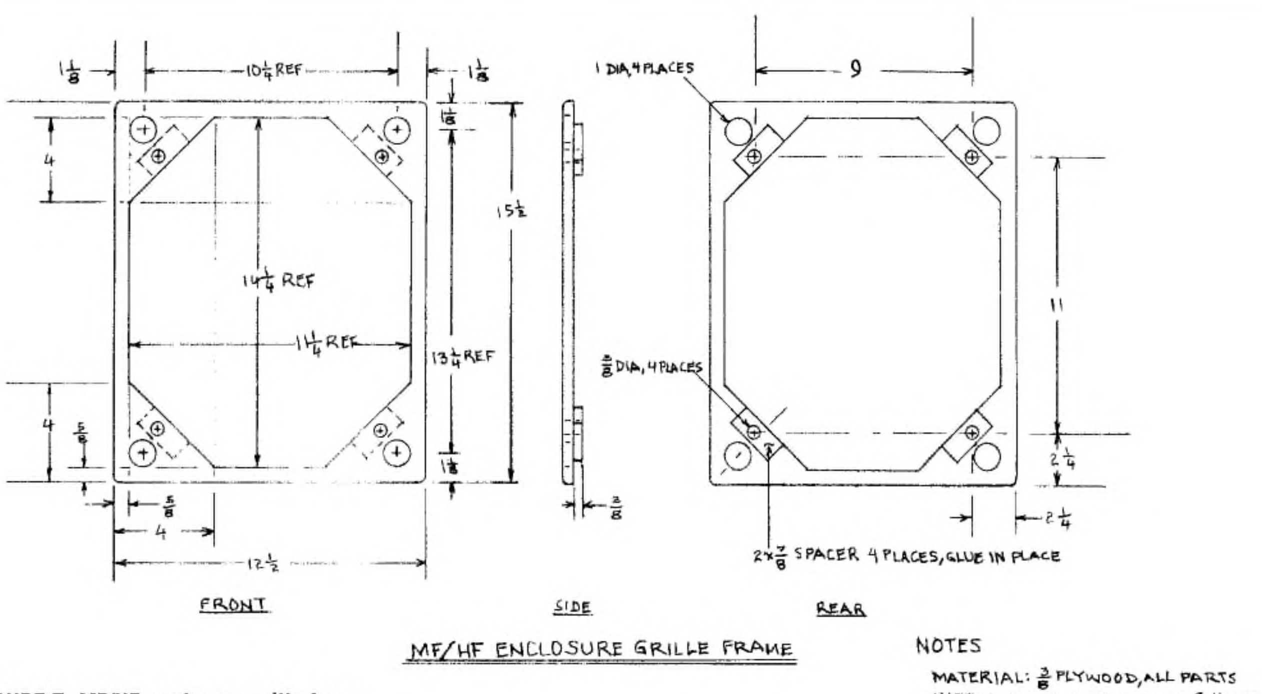


FIGURE 7: MF/HF enclosure grille frame.

layers of ¾″ plywood together. I suggest you cut the two parts somewhat oversize before gluing. The final sizing and cutout should be made after gluing.

Do a trial assembly of the top, bottom, and side panels to the rear panel with front frame in place to observe fit. Use clamps to hold the parts together (no glue) and drill the screw holes holding the panels together. In the top, bottom, and side panels, drill holes for the frame only into its inner member to avoid the enclosure bevel, which must have no holes in it. Use a few screws to hold the parts together to observe the parts fit.

If all parts fit satisfactorily, disassemble and do the final assembly using glue and screws.

Cut the speaker baffle to size and make the holes in it (*Photo 5*). Bevel after doing the hole work because the hole measurements will be easier to

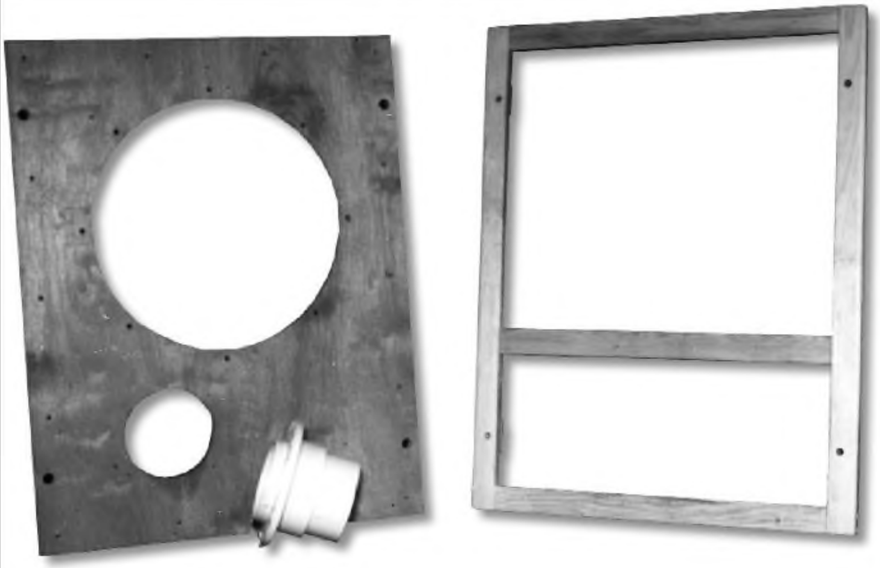


PHOTO 3: Low-frequency enclosure baffle and grille frame.

PHOTO 4: MF/HF enclosure body.

make before beveling. The baffle mounting holes should clear the body of 10–24 machine screws and are not countersunk. I cut the driver openings with a saber saw.

Refer to *Photo 6*. The frame is made of a single piece of ⅞″ plywood with four spacers glued to the back for the fasteners. I used para-ply. The corners are rounded approximately spherically to make the cloth fit nicely.

After you finish all exterior work on the enclosure, install the three drivers on the front surface. Mount the crossover network inside near the cabinet terminals and leave enough hookup wire slack for convenient extension to the drivers. Staple about 1″ of polyester batting around the top, bottom, and sides, and fill the remainder of the enclosure about ¾ full with batting beginning from the rear. No batting is attached to the baffle. Close up the enclosure.

FINISHING THE ENCLOSURES

The finish applied to the enclosures is purely a matter of personal preference. My enclosures, made of lauan, could have had the finish applied to it and might have looked OK were it not for the screws. I drove the screw heads slightly below the surface and applied wood filler over them, which you should do in any event. Nevertheless, filled screw heads would have appeared unsightly through any kind of varnish finish. I could have painted the enclosures as is often done for utility enclosures. *Photo 1* shows the completed unit sans grille cloth.

I wish that mine be presentable as furniture, so I decided to veneer them. If you have never done veneering before, you might start with a test sample such as a scrap of plywood. I learned the hard way and won't go into detail about veneering (because I don't know much about it).

I ordered the veneer from Constantine's, which included a little booklet with the order that was quite helpful. I chose cherry veneer and applied it with contact cement. It is usually impractical to apply veneer to a completed cabinet using regular wood glues because of the high clamping pressure needed. Regular veneer comes in

PHOTO 5: MF/HF enclosure baffle.

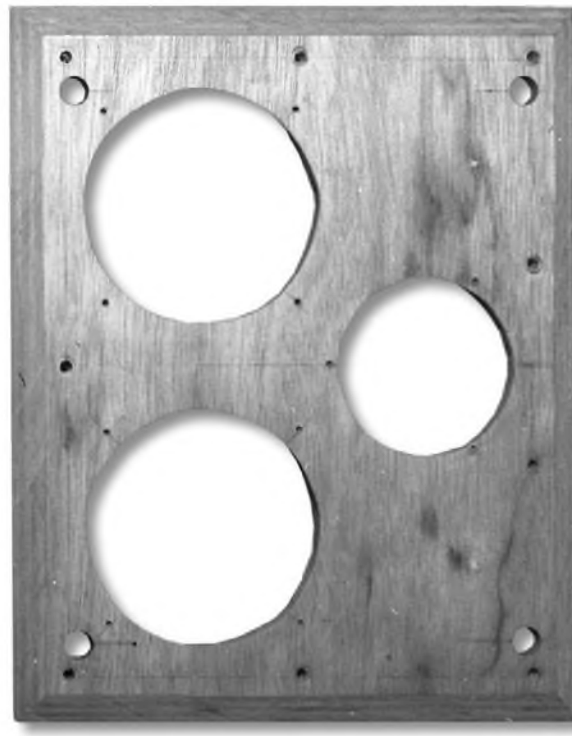


PHOTO 6: MF/HF enclosure grille frame.

PARTS LIST

PART	DESCRIPTION	SOURCE	PART #
batting	polyester, ½″ approx.	fabric stores	
grille guides	male and female, 12 pr/pkg	Parts Express	260-267
grille cloth	67″ wide	Parts Express	260-335

sheets about 1/40″ thick and may have a "potato chip" effect, i.e., curls in two directions, making it very difficult to apply flat.

I used this kind of veneer on the low-frequency enclosures, but chose thick veneer (1/16″) for the mid- and high-frequency enclosures which did not seem to have the "potato chip" problem. The thick veneer was much easier to use. Since you're using contact cement, you must bend the veneer away from the surface receiving it as it is being applied because the cement "grabs." Use a roller to bring the veneer into contact and to press out non-contacting spots.

I initially bought contact cement and solvent from Constantine's, but later used Weldwood contact cement and Xylene with about equal results (they are available locally). You will use a gallon or more of Xylene, mostly for cleaning the paintbrush, and it is not expensive. Constantine's is a good source for all veneering needs, however. ❖

What **music** selections do you use to **test** your **audio setup?**

TEST TRACKS

Simply describe your seven favorite pieces (not to exceed 1000 words); include the names of the music, composer, manufacturer, and manufacturer's number; and send to:

TEST TRACKS
Audio Amateur Corp.
Box 876
Peterborough, NH 03458
editorial@audioXpress.com

We will pay a modest stipend to readers whose submissions are chosen for publication.

The Ultra Fidelity Computer Sound System, Part 4

This part of the series examines the circuit design, including the all active "tri-amped" crossover system, pull down "T" attenuator, power amplifiers, and system performance. **By R.K. Stonjek**

Active crossovers have many advantages over passive. The area of particular concern for the computer sound system involves power handling. *Figure 25A* shows an amplifier clipping a low-frequency tone. *Figure 25B* shows the same tone after passing through a passive high-pass crossover. Note that although it is the low-frequency speaker that is being overdriven, a substantial error signal is being passed to the tweeter. This not only sounds horrible, but is also liable to burn out the tweeter.

DESIGN STRATEGY

Figures 25C and *D* show a low-frequency tone and a high-frequency tone together. The low-frequency tone is of a high amplitude; the high-frequency tone is of a relatively low amplitude. Even though the low-frequency tone is still below the clipping level, the high-frequency tone is clipping. The tweeter fizzles and burns again. In an active system, an overdriven bass driver does not affect the treble.

Note that if the tweeter's overall impedance is higher than the bass, then the treble voltage handling of the amplifier will also be higher. The illustrated situation no longer occurs. As tweeters almost always require attenuation to match the bass, using only series instead of series/parallel shelving resistors would raise the tweeter's overall impedance. This raises the overall power handling (for program material) by as much as 6dB.

Inexplicably, speaker manufacturers still insist on matching the tweeter's overall impedance with that of the bass. This increases the distortion and causes unnecessary destruction of the tweeter, particularly when you use amplifiers with only moderate power output.

The second problem prompting the active decision is the power handling of the subbass. While the frequency response of a bandpass box is determined by its tuning frequency rather than the size of the driver used, larger-diameter units are more efficient, and cone movement below the lower tuning frequency is substantially less.

The cone movement is the main worry. To overcome this problem, we place a high Q notch filter just outside the passband (*Fig. 26*). With a –3dB point of 34Hz, the attenuation is greater than 60dB at 23Hz. This is sufficient. Sound cards roll off below 20Hz quite steeply, as does the response of the amplifier modules used, so a notch filter is perfect for the job.

As mentioned in Part 3, the sub-bass needs to be close to the listener, ever more so since the crossover point rises above 80Hz. To achieve this, the sub-bass is designed to be wide and flat so the main speakers and monitor can sit on top of it.

Apart from these particular concerns, active crossovers have a number of other advantages such as greater efficiency and closer tolerance. In the close-in computer environment, the system seems to be more sensitive to phase anomalies such as the crossover from bass/mid to tweeter. With the better phase performance of the active crossover and the phase-aligned tweeters on the recommended enclosures, the crossover point melts away completely as far as the listener is concerned.

INTEGRATED OR OPEN DESIGN?

The overall strategy is for an integrated yet open design. You can use the standard unit, once built, as a launching

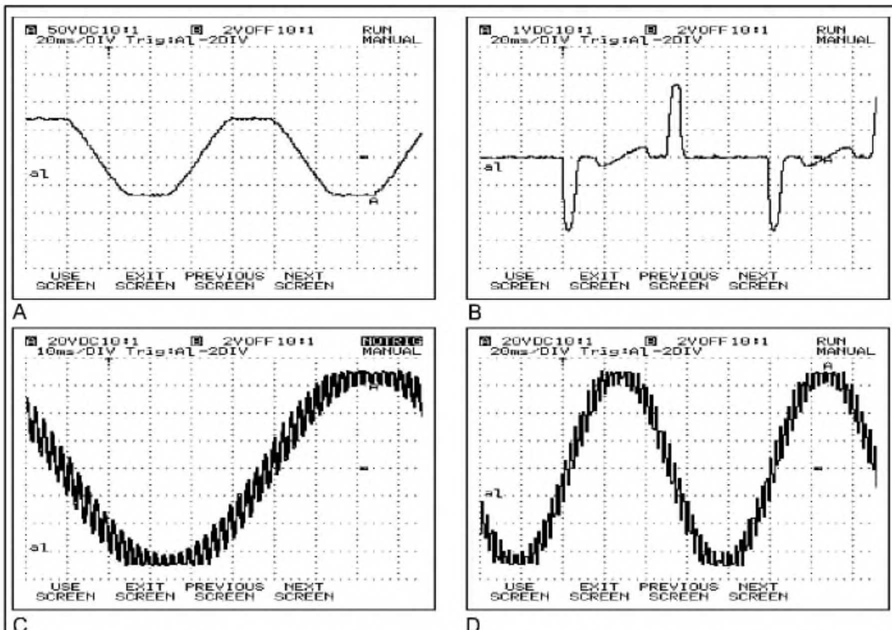


FIGURE 25: A) Amplifier clipping, B) Passive high-pass, C and D) 10Hz + 10kHz at clipping.

pad for a wide variety of amplifier/ speaker combinations. I made the design modifications as easy as possible so that you can adapt, modify, or adjust the overall system for your specific requirements.

The speakers, active crossovers, and amplifier are matched as far as practicable. If a hobbyist wishes to use different speakers, however, they can be matched to the system simply by changing the shelving resistors (R36, 37, 42, 43, 46, and 47) and adjusting the crossover frequency where necessary.

If you plan to use alternative amplifiers, you can take the signal directly from the naturally buffered output of the active crossover (at the shelving resistors). A valve amplifier with 2 or 3W output might improve the treble.

You can substitute potentiometers for the shelving resistors if you require individually adjustable shelving.

The recommended case can accommodate a second PCB, allowing the addition of two extra drivers for four-speaker systems. You can use the "spare" subbass amp for a center speaker in a 5.1 (AC3) system.

For this computer sound system, I added automatic on/off switching using the computer's 5V line. This means that the sound system will power up automatically whenever you switch the computer on.

PCB DESIGN

Before printed circuit boards there were wires connecting one component to another. Way stations, called tag strips, were used to mount a couple of components here and there. As circuits became more intricate, the wiring became complicated and construction tedious. To overcome this problem, printed circuit boards (PCBs) were developed.

At first the PCB circuits were drawn by hand. Templates were used for the various shapes—mainly donuts and straight lines. Curved lines were just too difficult, especially if you didn't get it right the first time. Later, when I came into electronics, we had transfers that we carefully placed onto plastic transparencies. The lines connecting donuts were on rolls, so you rolled them out. They had to be straight.

When computer-aided design tools became available, the whole process became much simpler. In particular, editing a design was made much easier. You could be more casual with the first draft—try different things and see how they look.

Oddly, computer-design tools have emulated the limitations of the hand-drawn designs rather than going back to the original intention of PCBs—to replace wires. All of the lines connecting donuts are still rigidly straight. More

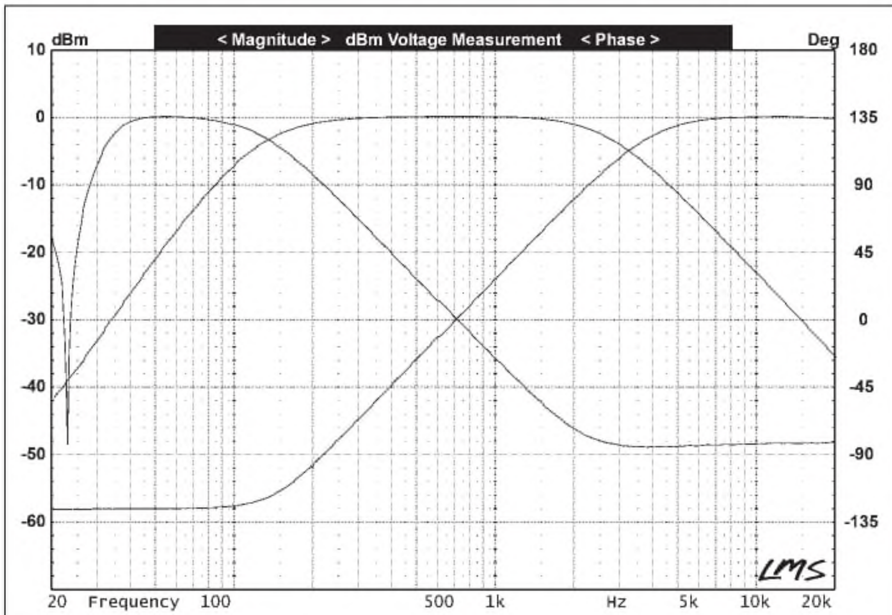


FIGURE 26: Active crossover ignoring relative shelving.

recent computer software is a little more flexible.

The PCB layout which I have designed uses straight and curved lines where appropriate (*Fig. 29*). I've averted complication by being flexible with the positioning of noncritical components. Naturally, high and low voltage and signal and power lines are kept well away from each other.

POWER SUPPLY

Figure 27 shows the overview of the amplifier design. The solid-state switch is made up of a triac driving optocoupler (IC1) and a 6A mains triac (S1), while 5V is tapped from a USB or other external port (PS/2 or keyboard) and passed to the optocoupler via current-limiting resistor R1 (*Fig. 30*). If you use a component other than MOC3021, you will need to adjust the value of R1 to suit.

When you switch on the computer, the optocoupler switches the gate of S1 and the mains is switched on to the transformer T1. Note that the triac does not switch like a relay, but acts as a high impedance series resistor. This means that if there is no load, the voltage at the output of the triac will be the same as the input (mains). Lethal voltages can be passed by the triac even when you expect it to be switched off.

When the transformer and power supply are connected, there will be about 10% of the mains voltage still flowing—around 25V in a 240V system. This results in around 1V at the secondary of the transformer. No appreciable current is drawn by the circuit when the computer is switched off (a couple of milliamps). This small idle voltage makes the system less noisy during switch on—less of a "pop."

R4 gives the amplifier some isolation while maintaining safety. The 18-0-18

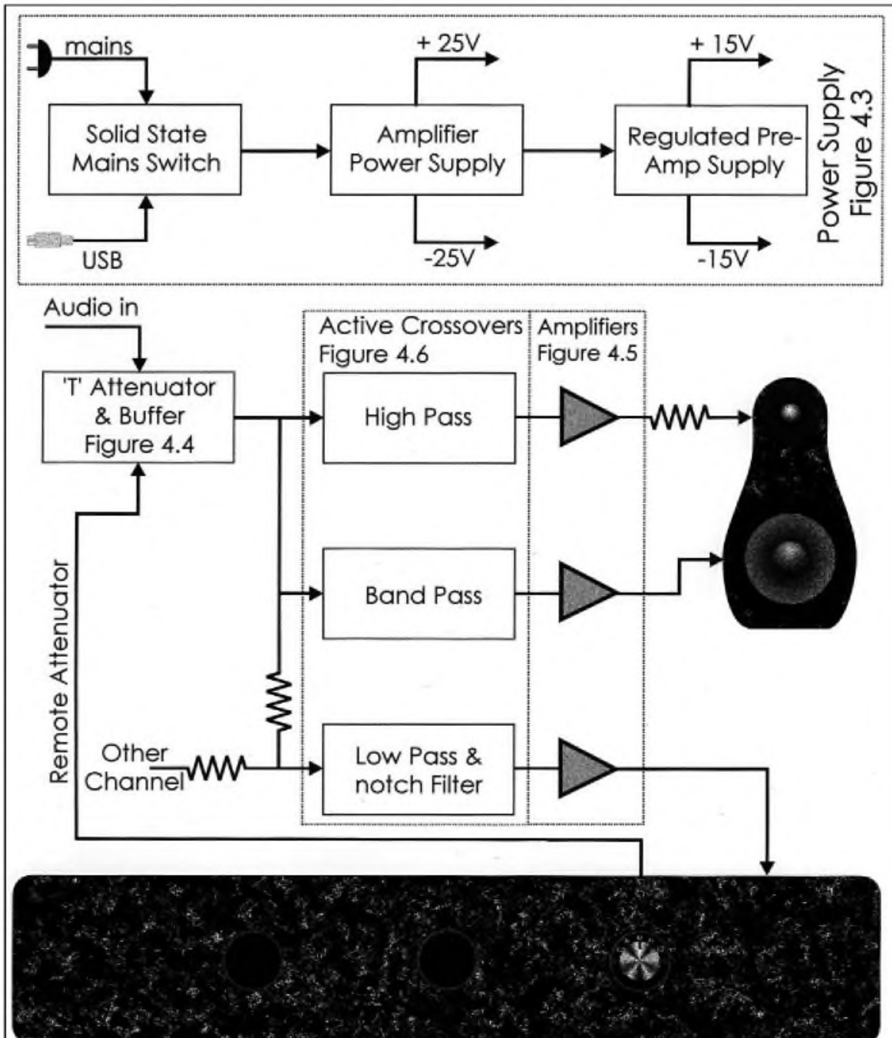


FIGURE 27: Overview

secondary of the transformer, once rectified via the bridge rectifier (S2), produces around ±25V DC, which is enough for the power amplifiers.

Regulators S3 and S4 give the required ±15V for the preamp and active crossover. Current-limiting resistors R5 and R6 protect the circuit from inadvertent PCB shorts or where ICs have been inserted the wrong way around.

INPUT STAGE

The input stage (*Fig. 31*) consists of optional capacitors C8 and C9, which are required only if a DC line voltage is anticipated. All the sound cards I tested have capacitor-coupled output and therefore don't require any additional series capacitors.

IC2a acts as an input buffer and attenuator driver. R9 and C10 form a low-pass filter. Because the sound source is a computer that has a clock running at many megahertz, the possibility of radio frequencies finding their way into the circuit cannot be discounted. The ICs

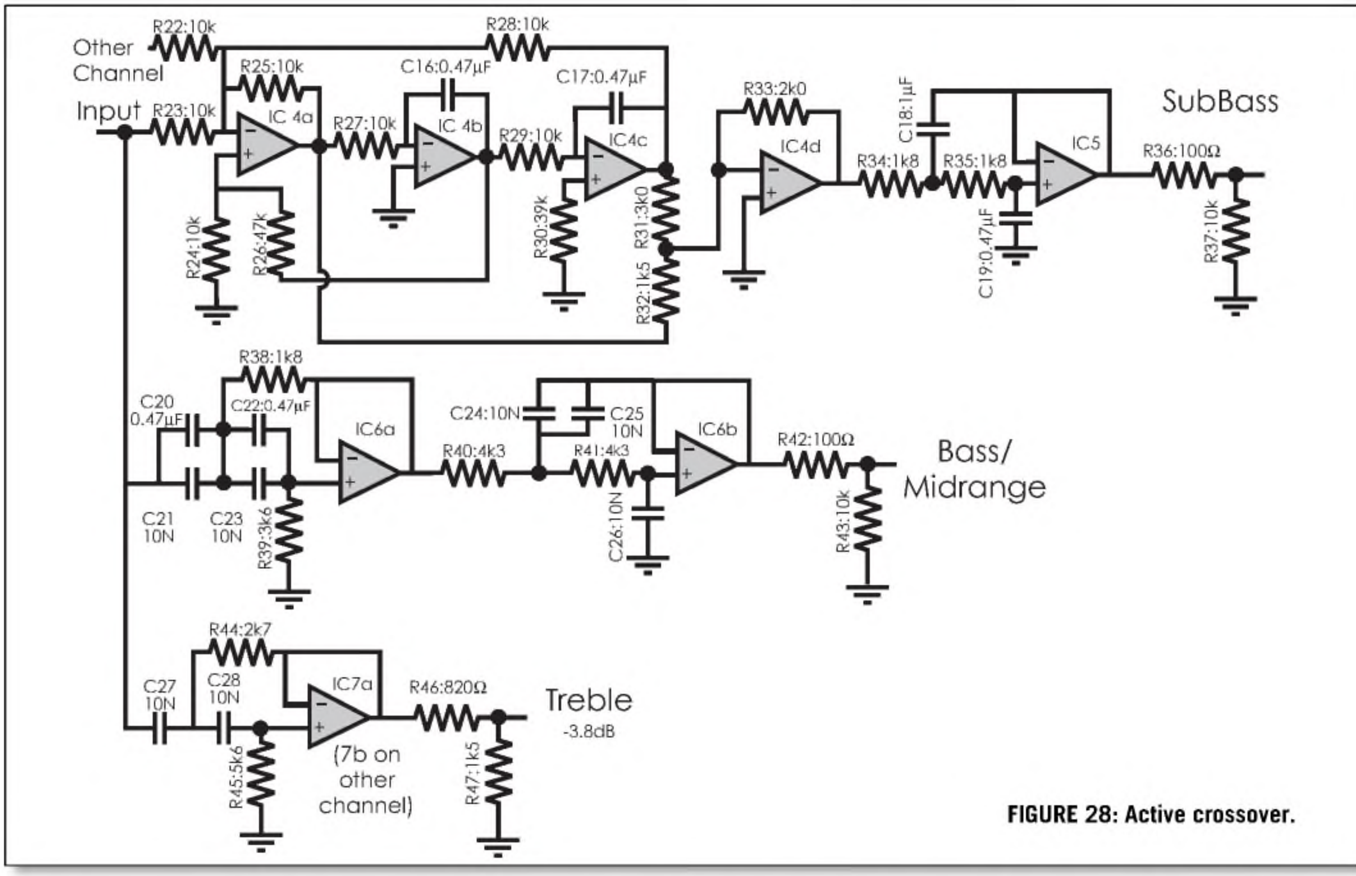


FIGURE 28: Active crossover.

used have a –3dB point of around 450kHz, so there is a need for a low-pass filter. The exact cutoff point of the low-pass filter varies with the attenuator setting, but remains between 50 and 80kHz.

The attenuator is a five-step pull-down "T" type. Attenuator noise and frequency response are no higher than the driving IC (*Table 2*/OPA2604). The trick to this attenuator is that it switches between a virtual earth at the negative input of IC2b and a real earth. R16 is an optional sixth step.

Using five steps you get an even –12, –6, 0, +6, +12 decibel spread. As the sound source is a computer, fine adjustment of the level is possible directly in the software. The attenuator is handy for matching the amplifier circuit with the sound card's output level, and for quick adjustments of the sound level. IC2b sets the overall gain (R18) and buffers the signal for the active crossovers.

ACTIVE CROSSOVER

The active crossovers (*Fig. 28*) for the bass and treble are the fairly straightforward second-order type. *Figure 26* shows that there is a bigger than usual dip at the bass/treble crossover point (–5dB). This is to compensate for a rise in both the bass and the treble response close to the crossover point. This is where integration really is an advantage. You can accurately tune the active crossover for the speakers used.

There is only one subbass speaker, so R22 and R23 act as a mixer adding the left and right channels together. IC4 is a notch filter with a very high Q and a resonant frequency around 23Hz. This circuit allows the subbass to be driven to quite healthy levels even when the signal contains near infrasonic frequencies.

IC5 forms the subbass's low-pass filter. R36 and R37 are not absolutely necessary, but are added to make it easier for an advanced builder to add separate shelving attenuators. By varying the ratio of R36 to R37, you also vary the output of the subbass. Likewise, R42 and R43 for the bass/midrange and R46 and R47 for the treble.

If you are not using the recommended drivers, you can vary these shelving resistors to suit a wide variety of speaker combinations. If you were using a separate subwoofer amplifier and large speaker, for instance, then you might increase the attenuation. The resistor combination should not show a collective resistance of less than 600Ω; preferably keep it to around 2kΩ.

POWER AMP

The LM1875 power amp module (*Fig. 32*) is robust and complete in itself, and requires few external components. The manufacturer's design notes specify a resistor and capacitor (Zobel network) on the output, but I could see no reason for this. The feedback capacitor, C13, prevents high-frequency oscillations.

Capacitors C11 and 12 limit the amplifier's low-frequency response. On the subwoofer amplifier, C11 is omitted and there are three extra C12s to give better low-frequency response.

OPERATIONAL AMPLIFIERS

Table 2 shows a comparison of the op amps considered for this project. The shaded area indicates the best performance for a particular parameter. Dual op amps are considered for practical reasons. Dual versions of single op amps are: LM1458(dual)-LM741(single), NE5532-NE5534, TL072-TL071, and OPA2604-OPA604.

The NE5534 is perhaps the first truly "hi-fi" operational amplifier featuring high slew rate, low noise, low distortion, and an ability to drive relatively low impedance loads (600Ω). Other devices used in hi-fi were either borrowed from other specialist areas such as low-noise meter drivers and photo diode drivers such as the AD711; were rated for general audio use such as the LM741 or the TL071, or never made it into the amateur audio consciousness, such as the OP27 and MC33078.

So much was the NE5534 talked about and specified in quality projects in the early 1990s, that when I considered an op amp for the Ultra Fidelity Concert series, I chose this device with little thought for careful comparisons. I found the device to be sensitive to "latch up," a problem that generally occurs when the input circuit does not have a low enough DC path to earth. The device will "snap to a rail"—swing all the way to one voltage rail or the other. I found that these devices were also failing more often than they should.

FIGURE 29: PCB.

When designing the Ultra Fidelity Concert Mk.3 in 1995, I ordered 30 NE5534ANs for testing—ten each from each of the local suppliers, in turn sourced from manufacturers such as Texas Instruments and Signetics. Ten of the devices failed during testing. The Texas Instruments device seemed to sound clearer than the others, though I could find nothing in the data or tested specifications to support this.

The Burr-Brown data sheet (for the OPA2604) discusses the audio quality of op amps and also mentions: "Op amp noise is described by two parameters—noise voltage and noise current. The voltage noise determines the noise performance with low source impedance. Low noise bipolar-input op amps such as the OPA27 and OPA37 provide very low voltage noise. But if source impedance is greater than a few thousand ohms, the current noise of bipolar-input op amps react with the source impedance and will dominate. At a few thousand ohm source impedance and above, the OPA604 will generally provide lower noise."

The current noise is quite high for

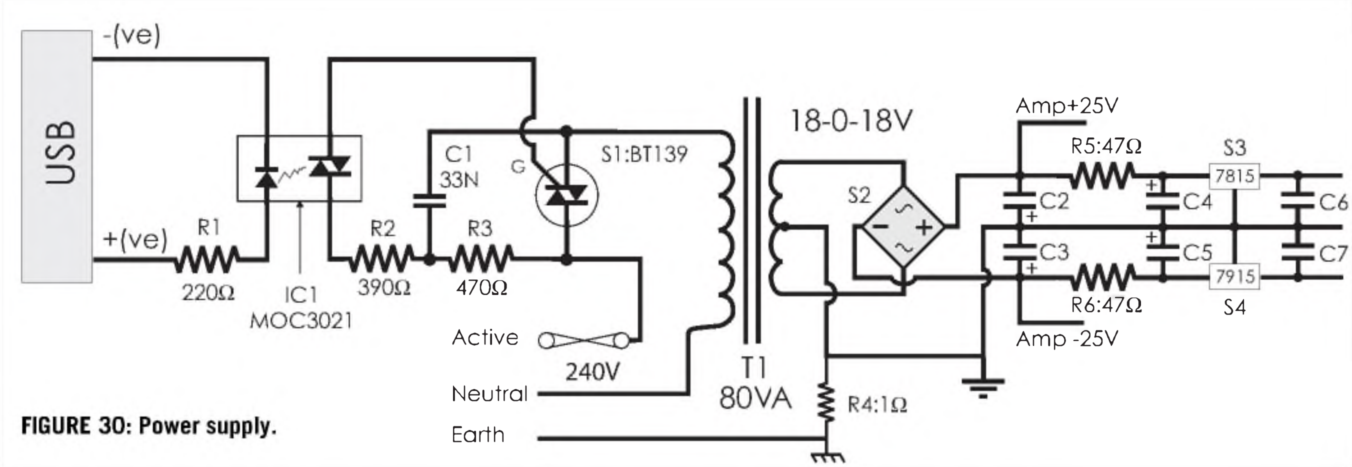


FIGURE 30: Power supply.

the NE5532 at 700 fempto amps (*Table 2*). Note that this specification is often given in pico-Amps ~1pA = 1,000fA. The NE5532 has a differential input voltage limit of ±0.5V. Zener diodes come into effect if this limit is exceeded. The failure of these diodes during latch-up may account for the high overall failure rate during testing.

After these failures, I looked around for other op amps and tried the MC33078 Motorola device. I had no failures at all and have had none since. The datasheet (see *Table 2* for datasheet download addresses) specifies the differential input as "either or both input voltages must not exceed the magnitude of Vcc or Vee," i.e., the supply rails, usually ±15V (total swing is from one rail to the other, or ±30V). This, along with 28.5% less current noise, makes the Motorola product my choice for bipolar audio devices. There are dual and quad versions.

In the mid-1990s, there seemed to be much hype for new FET op amps. The TL072 has been around for a while, but is inherently noisy. The AD712 offers very high slew rate and low distortion. My Rotel CD player had AD711s in the audio output stage. The noise is high enough for the "sound of the sea" to be heard in the background. When I substituted them for the Burr-Brown devices, the OPA604, the noise went away.

The Burr-Brown device features three times the slew rate, less than one-tenth the distortion, one-hundredth the current noise, three times the bandwidth, and an extra 20dB channel separation. It is a very long way ahead of equivalent bipolar devices (*Table 2*).

On the Burr-Brown datasheet you can read all about their commitment to the sound quality of their op amps, the special "distortion rejection circuitry," and a comparison between bipolar and FET distortion: "FETs, like vacuum tubes, have a square-law I-V transfer function." The 0.0003% distortion speci-

Parameter / Device	LM1458 (Dual LM741)	NE5532	MC33078	TL072	AD712	OPA2604	Units
Voltage Noise f=1kHz		5	4.5	18	18	11	
Current Noise f=1kHz		700	500	10	10	6	
Distortion		<0.01	0.002	0.003	0.0003	0.0003	%
Slew rate	0.5	9	7	13	20	25	V/μs
CMRR (Common Mode Rejection Ratio)	90	100	100	86	88	100	dB
PSRR (Power Supply Rejection Ratio)	96	100	105	86		100	dB
Channel Separation		110	120	120	120	142	dB
Full Power Bandwidth		140	120	>150	200	>400	kHz
Differential Input Voltage	±30	±0.5 (Shunt Diodes)	Vcc to Vee	±30	±20	±1.0	V

DataSheet Downloads	
LM1458	http://databook.stts.edu/National/Web/htm/nsc02674.htm
NE5532	http://focus.ti.com/docs/prod/productfolder.jhtml?genericPartNumber=NE5532
MC33078	http://www.onsemi.com/pub/prod/0,1824,productsm_ProductSummary_BasePartNumber=MC33078,00.html
TL072	http://focus.ti.com/docs/prod/productfolder.jhtml?genericPartNumber=TL072
AD712	http://products.analog.com/products/info.asp?product=AD712
OPA2604	http://focus.ti.com/docs/prod/productfolder.jhtml?genericPartNumber=OPA2604
LM1875	http://www.national.com/pf/LM/LM1875.html#Datasheet

TABLE 2

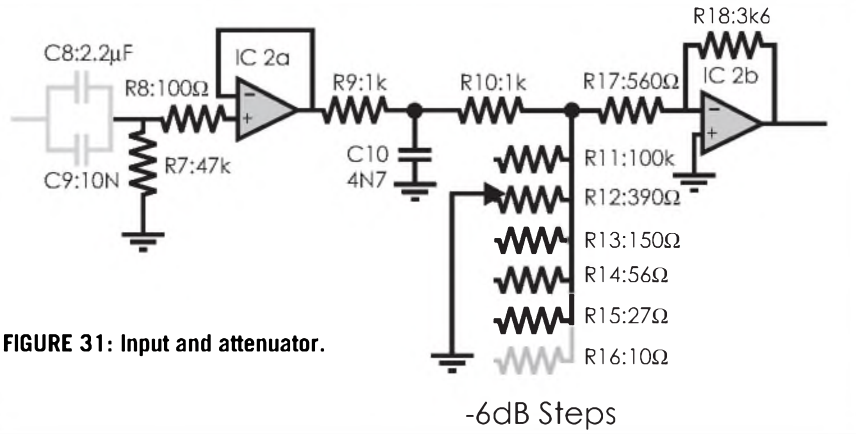


FIGURE 31: Input and attenuator.

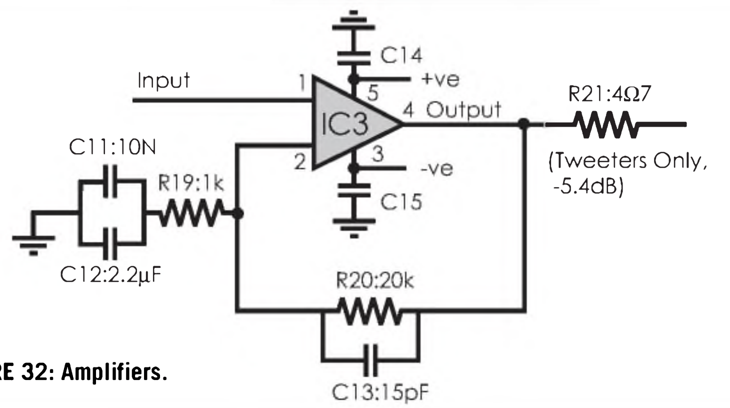


FIGURE 32: Amplifiers.

fied is actually a measurement of the distortion+noise!

CAPACITORS

Polystyrene capacitors have always been known as an excellent choice for high-quality audio applications, but the standard devices have many practical failings. The clear plastic case is difficult to work with. One dab of the soldering iron and they're dead—the internal foils short-circuit.

The size of the capacitor varies with their value, manufacturer, and even batch. The tolerance is not all that low—typically 5%. Low-voltage capability is not usually an issue in the circuits to which they are best suited—in our case, the active crossovers.

All of these problems are solved with the LCR polystyrene capacitors. The 7 × 7 × 12mm case is the same for all values and is robust and easy to handle. The tolerance is just 1%. I've specified these devices along the signal path.

Note the relatively low priority sound card designers give to capacitor selection. Even looking at the better-quality Creative and Turtle Beach sound cards, you discover a sea of electrolytic capacitors—yuk! When consumers become serious about computer sound, matching products will follow.

Upgrading your sound card's capacitors cannot be discounted, but does the risk justify the reward? Proceed at your own peril! Hint: rather than replacing existing capacitors, try bridging them with high-quality polystyrene capacitors of a value within the bridged capacitors tolerance; e.g., 1µF ±5% has a tolerance of ±50N. A 10N capacitor would be perfect.

MKT-type polyester capacitors are used everywhere else except for the power amp decouplers (which are monolithic ceramic) and the amp's feedback capacitor (which is an NPO ceramic).

In Part 5 I will detail the construction of the Ultra Fidelity Computer Sound System. ❖

In the mean time, interested parties may obtain the printed circuit board from Old Colony Sound Lab, provided there is sufficient interest in the project. Most PCB printers will print from a Corel Draw! Graphic. Instruct your PCB printer to download the 88k zip file containing PCB graphic and hole drilling guide from *audioXpress*' website at http://www.audioXpress.com on the *audioXpress* magazine page.

Product Review

Golden Ears Training

Reviewed by Mark Florian

Available from Old Colony Sound Laboratory, PO Box 876, Peterborough, NH 03458-0876, 603-924-9464; Fax 603-924-9467; custserv@audioXpress.com; price $220.

As long as humans have had ears, the war has raged back and forth: "Did you hear that?" "Hear what?" "That... That!" "No, there's nothing out there." Sure enough, the next moment, the deaf one was eaten by some hungry prehistoric beast.

Much further on in mankind's evolution, he discovered and built electronics for the reproduction of audio, and the war continued. This time between the meter-readers and the golden ears. "Did you hear that?" "Hear what?" Capacitors, resistors, wire, steel pins on tubes, tubes versus transistors, even cabinet materials. No material was left out of the debate.

INTRODUCTION

Dave Moulton, a long-time recording engineer, teacher, and composer, decided it was time to develop a course to teach you to improve your hearing and discernment of what you hear. Aimed primarily at those on the recording side of the disk, it's of benefit to those on the playback side as well, as you seek to tweak your systems to the highest level.

The course consists of eight CDs divided into four volumes: Volume one is on frequencies; volume two on effects and processing; volume three on time domain-delay and reverb drills. Volume four consists of master frequencies on one-third and dual-octave drills. In addition to the CDs, a 99-page manual is included with instructions, answer sheets, and answers. That's right, answers...feedback. You get to check your guesses and score yourself.

Do you really have golden ears or do you just think you do? Regardless, after you finish this course, your ability to discriminate between subtle details and pinpoint one-third octave frequencies that are out of balance will be highly acute. I'll cover each of the volumes in turn and then finish up with an overall summary.

The stated goals for this course are rather high: "Working through these drills...will permit you to gain the equivalent of five to ten years of critical listening experience in a matter of weeks." Also, "You will come to know, by ear, how the audio energy is distributed across the spectrum, approximately how loud two sounds are relative to each other, the kinds of signal processing going on, and so on. Almost as important, you will be able to easily detect when others are reduced to random knob-twiddling and hype." "...you should be able to listen to an equalizer switched in and out on a recording and describe how the equalizer is set, in terms of frequencies and the amount of boost or cut, by ear alone!"

The first step is "calibration of your system." System listening levels are set using pink noise. Having a SPL meter is handy, but not necessary. This is important because once set, you should leave the volume alone for the remainder of the exercises. Also important is to make sure all tone controls, loudness contours, and so on are set flat. After levels are set, five tones (1kHz, 10kHz, 100Hz, 15kHz, and 40Hz) are played to make sure you can hear each.

VOLUME ONE—FREQUENCIES, THE EQUALIZATION DRILLS

"The equalization drill sets each consist of ten examples. Each example is about ten seconds long and consists of either pink noise or music. The example begins with the sound played normally. After about three seconds, the sound is altered by boosting or cutting the amplitude of one or more octaves in the spectrum, using a graphic equalizer. After about four seconds of equalized sound, the sound is returned to its normal state. Your task is to identify which octaves of the spectrum were boosted or cut."

In the beginning, the tests are restricted into three groups 31–500Hz, 250–4000Hz, and 1–16kHz to simplify the process until you get the sound of each into your ears. As with any advanced learning course, the exercises become progressively harder as you gain more skill in listening.

In the first six drill sets, one of the frequencies is boosted by 12dB. In the following six, one is cut by 12dB. Set 13 boosts one out of all ten one-octave bands by 12dB using pink noise as the source. Set 14 repeats the previous set except now music is used.

The process for each set is the same. A warm-up is first, followed by the first set with pauses in between each frequency so you can write down your answer. At the end, you play the set again and check over your answers. Then compare them to the answer guide while listening to the set for a third time. This repetition helps you to associate the sound of the frequency with its name, much like learning the names of the colors.

Dave recommends taking a break after three drill sets, which is 30 sounds. When repeated three times each, it's 90 sounds. If you don't, this becomes very monotonous very fast.

Very intense listening is quite fatiguing, so take a break. When you begin the next session, it's suggested you repeat the last exercise and then add two more. This "one step back, two steps forward" approach helps you review what you've learned and anchors the sound with its name in your mind.

The second CD in volume one no longer restricts the answer to one of three bands. Here a band is boosted or cut 12dB in one of all ten octaves. In the first two drill sets one frequency is boosted 12dB. In the following two, one is cut 12dB. In the following ten sets, one is boosted or cut 12dB.

As in the first CD, both pink noise and music is used. The process for listening, answering, and reviewing is the same.

VOLUME TWO—EFFECTS AND PROCESSING, THE A/B DRILLS

"Each A/B drill set consists of five examples. Each example is a pair of recorded excerpts of music. The first recording (A) is the 'reference' and the second (B) is a clone of the first with some sort of signal processing or audio anomaly added. Your task is to identify the signal processing applied to the B recording."

"To assist you, we have limited the number of possibilities to a menu of 31 possible signal processing changes, grouped into six families: amplitude change, distortion, compression, equalization, stereophony, and time-delay/reverberation. Also, we have included 'no change' as an additional answer, just to keep you honest. Instead of a warm-up drill, the first half of CD 3 demonstrates all of these effects for you. The balance of CD 3 and all of CD 4 are A/B drills."

The purpose of these drills "is an effort to instill critical listening paranoia in you and to teach you how to hear and identify small differences between two versions of the same recording." Unlike the previous drills in which the information was recorded in mono, these drills are in stereo. So not only must you identify the effect used, but which channel changed as well. I'll discuss each of the six preceding families.

Hearing Amplitude. "Since louder sounds often sound better, it is essential to know when loudness is the only difference between two signals." Another twist in these drill sets is that differences of only 3dB are used and the level can increase or decrease gradually over the test.

Hearing Distortion. THD is grouped into two categories: 10–30% and 1–10%. Dave points out "that the perception of distortion is significantly affected by the music being played and also by the extent to which harmonic distortion is dependent on level."

Hearing Compression. Both fast and slow compressor release times are demonstrated on different source types.

Hearing Equalization. Since this is where we started, these drills are much more extensive than those found in volume one. The boosts/cuts are now 6dB instead of 12dB. In addition, only one channel is boosted or cut, leaving the other normal. Also combinations are used such as the lows being boosted while the mids or highs are cut.

Hearing the Stereo Field. Faults commonly occurring in stereo mixes include reverse image, mono summation, polarity reversal, and pseudo-stereo. Some of the tests are faded in or out, for example, stereo to mono and mono to pseudo-stereo, which was generated by "using a 10-band graphic equalizer with alternating octaves boosted and cut and the settings of the two channels reversed."

Hearing Time Domain Differences. In these exercises, channel-to-channel time differences are varied from 1 to 50ms. Within the tests, reverb both ungated and gated is added or deleted and time delays of 1, 5, 15, 30, and 50ms are used between left and right.

After the demonstration tracks for each of these effects, the drill sets consist of all six categories mixed up for each set. So, for example, one set might include equalization, compression, stereo, amplitude, and another equalization. You may be asked to name not only the effect, but its degree as well, according to the previous categories.

It becomes quite difficult at this point to choose from so many options and to identify its degree. By the way, all of these drills use music! No more easy pink noise. There are a total of 18 drill sets.

VOLUME THREE—TIME DOMAIN-DELAY AND REVERB DRILLS

"This volume is devoted to the development of your ability to discriminate time intervals. Specifically, the first CD focuses on the perception and identification of short delay times, while the second CD focuses on the perception and identification of predelay and reverberation decay times." The manual goes on to point out a couple of key factors regarding our perception of time in sound.

First, and most important, a delay of 50ms represents a very important discrimination boundary. Remember that a delay of 50ms corresponds to a frequency of 20Hz. When you hear several sounds that are more than 50ms apart, you think of them as distinct, individual events. If these sounds occur within 50ms however, you think of them as one. So your brain doesn't distinguish between them, but blends them together. Thus the range of 30ms to 70ms (plus or minus 20ms on either side of 50ms) is very critical.

Second, you perceive delays between two channels shorter than 0.7ms to originate between them—in the middle. Delays greater than 0.7ms are perceived to originate from the "earlier" of the two sources. Rather remarkable when you think about it!

The delay drills consist of 35 different delay times, ranging from none to 170ms, on three different sources: kick drum, pink noise, and vocal music. The whole set contains four kick-drum drills, three pink-noise drills, and ten music drills. In addition to identifying the amount of delay, you must also note which channel is delayed. When listening to these 17 drill sets of delay drills, it's very important to sit equidistant from both speakers, in the median plane. Otherwise, you may not be able to hear them.

The reverb drills consist of "six different predelay times, ranging from 0 to 100ms, and eight different reverb decay times, ranging from 0.3s to 5s. These are demonstrated three times: First with a kick-drum sound, second with a drum machine loop, and finally with a brief phrase of piano music. Each reverb time is demonstrated with all six predelay possibilities. There are four kick-drum drills, three drum-kit drills, and seven music drills." There are 14 drill sets of reverb drills.

VOLUME FOUR—MASTER FREQUENCIES—1/3 OCTAVE AND DUAL OCTAVE DRILLS

"Volume four is devoted to the development of your ability to discriminate frequency bands to a sophisticated level. The first CD focuses on the perception and identification of bands centered about each one-third of the audio spectrum, while the second CD focuses on the simultaneous perception and identification of two separate octave bands...Identifying two octaves at once is a fairly high-level cognitive task, and requires the developed mental ability to consciously 'scan,' or break into component parts, the audio spectrum. This is the skill you need to be able to equalize sound and predict the sound of any given equalization effectively. When you can successfully hear and identify two octaves at a time, you can easily manage all the various equalization tasks.

"For the one-third octave drills, there are 27 separate bands that I've arbitrarily separated into groups of low (31 to 315Hz), mid (315 to 1250Hz), and high (1.6 to 16kHz). These are demonstrated twice: first with pink noise and second with music. The first six drills are pink noise and music drills for low, mid, and high frequency ranges. These are followed by three broadband pink noise and five broadband music drills. For the dual octave drills, I've simply prepared a demonstration of typical combinations to help you get the idea. There are three pink noise and nine music drills."

Each drill set in this volume contains ten sounds to listen to. These exercises are the most difficult of all.

SUMMARY

After having listened to most of the exercises in volume one, I can say that the next time you are in a grocery store or at happy hour or an amplified concert, the frequency will pop in your head and so will the boost/cut amount. Without even thinking! You will develop a whole new dimension to your listening. You will also surprise those around you by naming the frequency.

I regularly carry rubber earplugs when going to concerts indoors these days—they are so loud. I've noticed that you can even pick out bass humps while wearing them!

These exercises in the four volumes do take a considerable amount of time to work through. But you shouldn't rush it, or try to hurry through it. Remember how long it took to learn to ride a bicycle? Or hit a baseball? Consistent effort over a period of time is the best way to master these exercises and achieve a new way of listening.

I, for one, will be setting aside some time each day to work my way through them. After all, the ear is the final audio tweak, and tuning it will help in evaluating everything else! Highly recommended.

Manufacturer Response:

I'd like to thank Mark Florian for his thorough and essentially accurate review of my Audio Ear Training Course, Golden Ears. I think he's got the idea rather well. One feature he's missed that I believe adds significantly to the value of the course is that the slate identifications of each drill set come before the track ID, so that when any given CD is played with the RANDOM or SHUFFLE function, the listener can take the drills blind (IDs come at the end). This means that, in essence, the listener can never "use up" the drills (except by memorizing them all!). I still take the drills on a fairly regular basis (I've been doing them for about 30 years!) using this function, and they still help, as ear maintenance.

Also, I don't recommend listeners do them with headphones (some of the stuff becomes unreasonably hard), and I don't recommend listeners try them in a car, especially while commuting!

The only other comment I have is that these drills have been around for a long time now, and they've been used by a LOT of recording engineers. They were a central part of the curriculum at National Public Radio's remarkable Music Recording Workshops held throughout the eighties, and I've used them every place I've taught, including SUNY/Fredonia, Berklee College of Music, UMass/Lowell, Emerson College and the School of the Museum of Fine Arts. They're still in use at most of those places.

Oh yes, and buyers can feel free to contact me with questions. I've often done tutorials over the phone for listeners who got confused or stuck. It's part of the deal. ❖

David Moulton
Moulton Laboratories
Groton, MA 01450
978-448-6828
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Product Review

The Rane Active Crossovers

Reviewed by Justus Verhagen

PHOTO 1: AC 22 front panel. From left to right: on/off switch, power indicator, master input gain, low-pass gain, low-pass mute switch, delay control, crossover frequency selector, and high-pass gain. Repeated for the second channel.

Rane Corporation, 10802 47th Ave. W., Mukilteo, WA 98275, 425-355-6000, FAX 425-347-7757.

In the 13 years that I have been involved with audio/music as a major hobby, many pieces of equipment have come and gone. Few were here to stay for more than two years. Besides my modified Dynaco MK-IIIs, my Rane AC 22 two-way active crossover has been a component that has stayed—and will stay, for many years.

ABOUT THE AUTHOR

Justus Verhagen has a Ph.D. in neuroscience and is currently employed as a post-doc in the Department of Experimental Psychology at the University of Oxford, England. There he researches neural coding of sensory systems in monkeys. He has been working on audio equipment for 13 years and is using engineering methods (modeling, design and measurement) more and more. His biggest audio-related interests are non-dynamic type loudspeakers and tube amplifiers, as well as good music.

Before I went "active," I considered active crossovers as something only for the professional and car audio branches. Boy, was I wrong. Let me summarize some of the possible acoustical advantages of active filtering:

1. reduced intermodulation distortion (amps see smaller frequency spectrum)

2. reduced power drain from amps (idem)

3. optimization of amps for each low- and high-pass section, e.g., solid-state for low, and tube for high frequencies.

But even besides these advantages (some think they are merely theoretical), active crossovers have practical features that make them ideal for the DIY speaker builder:

1. you can instantaneously change the crossover frequency

2. no more need to invest in caps, coils, and resistors

3. you can separately adjust the volume of each section of each channel

4. you can delay the output of one section relative to the other.

The Rane AC 22 does all that. But best of all, it can be obtained for about $150 on the second-hand market!

TECHNICAL DETAILS

The AC 22 is a two-way stereo active crossover employing state-variable Linkwitz-Riley filter alignments. Its bigger brother, the AC 23, has a three-way configuration, but the rest is the same.

The Linkwitz-Riley alignment consists of two second-order Butterworth filters in series for each high- and low-pass section. This results in 24dB/octave rolloff slopes and in-phase output of the sections.

The advantages of the steep slopes are that each speaker driver acts more specifically within its passed frequency domain and that comb-filter effects are reduced due to the smaller frequency overlap of the output of the drivers (assuming they are physically displaced). The advantage of the in-phase outputs is that the sections sum correctly (on-axis) around the crossover region.

PLUG AND PLAY?

The AC 22 comes in two flavors: balanced/unbalanced TRS (tip-ring-sleeve) inputs (the large microphone plugs) and unbalanced TRS outputs (AC 22) or active balanced inputs and outputs via XLR plugs (AC 22B). Although this may sound confusing, it turned out to be very easy to hook the unit up to the RCA outputs of my CD-player and RCA inputs of my amps: Radio Shack sells gold-plated adapters (RCA to TS) into which you plug your cable's RCA male and then plug it into the input of the AC 22. If you are lucky enough to have balanced XLR inputs and outputs, buy the AC 22B.

Rane has a website (www.rane.com) with a plethora of information, including excellent notes on how to connect components. Print it out and keep it in a safe place after memorization. The units are powered by a remote power supply, the RS 1. You can also use the AC 22 as a mono three-way crossover, and more recent versions have the option for a mono sub output.

Photo 1 shows the front panel of the AC 22. The crossover frequency selector (ALPS pots) has 41 discrete settings between 70 and 3.6kHz, allowing for accurate frequency selection and matching between the two channels. The delay control moves the low section backwards relative to the high output (0–2ms delay), which allows for alignment of drivers if that's not physically feasible.

If needed, you can modify the AC 22 to instead move the high-pass section backward. Make sure to read the online manual on how to accurately align the sections. My AC 22 is ten years old and all the switches and pots still work flawlessly.

The rear panel has an input and two outputs for each channel and a connection for the external power supply. The modern versions additionally have a switch for the two-channel versus mono-sub configuration. The input impedance is 20kΩ, output is 100Ω. These values are likely to yield good impedance matches.

Gain is off to +6dB for the input and output sections, but note that Rane recommends not to get much gain from the unit. RFI, infrasonic and ultrasonic filters are built-in, limiting its frequency response from 20Hz to 40kHz, with a THD and IM distortion of 0.02% (both +4dBu, specs provided by Rane).

FEATURES

- Stereo 2-Way or Mono 3-Way
- Linkwitz-Riley Alignment with 24 dB per Octave Slopes
- Adjustable Delays
- Mono Subwoofer Switch and Output
- Infrasonic, Ultrasonic, and RFI Filters
- Low Output Muting (2-Way)
- Input & Output Level Controls
- UL/CSA/CE and 100/120/230 VAC Remote Power Supplies

AC 22 FEATURES

- ¼″ TRS Inputs and ¼″ TS Outputs
- Active Balanced/Unbalanced Inputs & Unbalanced Outputs

AC 22B FEATURES

- XLR Inputs & Outputs
- Fully Active Balanced Inputs & Outputs

LET'S PLAY!

I have compared the AC 22 with a DIY tube-based crossover and my home-built minimalist Linkwitz-Riley active crossover. The latter is built around the good-sounding OPA-604 opamps from Burr-Brown and polypro caps. My Rane AC 22 employs Texas Instruments' TL072cp for all but the delay sections, and these are older dual JFET opamps.

I found the three units to perform equally well, despite the higher degree of complexity of the AC 22. I ended up using only the AC 22 and selling the tubed one, due to the AC 22's high degree of flexibility.

This doesn't mean you can't upgrade the AC 22. Various people have suggested two mods that are both effective and cost nothing. First, you'll need to download the PDF schematics from rane.com. Next, localize electrolytic caps C4 and C6 on channel 1, and C10 and C12 on channel 2 (in my older model they are called C110, C112, C210, and C212, respectively). These are DC-blocking output caps, and can safely be taken out to be replaced by jumpers.

The second mod consists of bypassing the infrasonic (subsonic) filters. In these days of rumble-free media, that's one stage too many. This filter is based around the second amp in the dual JFET TL072cp op amps, which are indicated as Z100. The input stage is built around the first amp of the dual op amp.

To bypass the filter, remove the op amp and bend pins 5, 6, and 7 up, so that they are out of the circuit. Next, solder a short wire between pin 1 (output of input stage) and 7 (output of the now inactive filter stage), reinsert the opamp, and voilà, you're done. Make sure the wires don't touch the base of the steel casing, though.

To my ears, there were subtle improvements of the sound quality: more air around instruments and voices; better sense of depth, especially during complex musical passages; a higher level of detail; and perhaps a generally smoother sound.

Another possible mod is to replace the op amps, but beware: the circuit may start to oscillate (OPA-604 is known for doing this now and then) or too much current may be drawn if you choose the wrong one. Check for compatibility and perhaps with Rane.

ALTERNATIVES?

Rane introduced an economy-class line in 1996 called Mojo. The MX 22 and MX 23 are cheaper and simpler than the AC crossovers. They have only XLR connections, stereo controls, and no delays.

Other brands, such as Ashly and Marchand, also make affordable active crossovers. You may choose to check them out.

My next step will be digital signal processing (DSP). Models have appeared on the market that have both equalization and crossover functions (Rane, DBX, Yamaha, and BSS). As yet, these models cost about ten times more, though.

Note that further significant advances have been made in the last few years: equalizers from SigTech and TacT not only extensively adjust in the amplitude domain but also in the time domain. If you have a few grand to spend, you may want to look into them.

In conclusion, the Rane AC 22 (or 23) provides an excellent way to step into the active crossover scene. It's cheap (new or used), flexible, dependable, and sounds good. In the long run it has saved me time and money, and added lots of musical joy. Highly recommended!

ACKNOWLEDGMENTS

I received many helpful suggestions on the mods for the AC 22, especially from Rane's own technical consultant, Chris Duncan. It should be noted, that these were suggestions from Chris, not from Rane.

Manufacturer's Response:

Rane Corporation appreciates Mr. Verhagen's thorough and accurate review and offers some additional information.

Regarding our design, rather than cascading two second-order Butterworth filters, as originally done by Siegfried Linkwitz and Russ Riley, our implementation of their crossover alignment is built around a proprietary fourth-order state-variable circuit, first developed by Rane and presented at the 74th Convention of the Audio Engineering Society in 1983 (preprint 2011; available for download at http://www.rane.com/pdf/linriley.pdf). Since then, it has become the de facto *standard method for professional audio active crossover designs.*

*The importance of the state-variable approach is that it guarantees the high-pass and low-pass outputs are always exactly the same frequency, and it produces a simple and affordable method for making the circuit independently adjustable in frequency with no interaction to gain or Q (quality-factor). Anyone interested in additional information about Linkwitz-Riley designs is welcome to browse or download (*RaneNote 107 Linkwitz-Riley Crossovers *at http://www.rane.com/note107.html).*

The modifications Mr. Verhagen suggested are good ones for fixed home-type applications, although please note that by removing the non-polarized output electrolytics, there will now be a small amount of DC voltage on each of the outputs. If you disconnect or reconnect the unit while it is powered up ("hot plugging") you will hear a sharp crack through the system if the amplifiers are on and turned up. The sudden application of this small DC voltage appears as a momentary step input and will pass right through any subsequent DC blocking capacitors, thus causing an audible result.

Removing the infrasonic filter (note that the term "subsonic" means less than the speed of sound, while "infrasonic" refers to frequencies below the normal range of human hearing, and is the correct term) does not hurt anything. In the professional audio world, particularly in live sound applications, they are necessary to control wind-noises, cord movements picked up by sensitive microphones, and other low-frequency anomalies that shouldn't be passed on to the power amplifiers and loudspeakers.

Since this review was written, Rane has introduced its newest pair of active crossovers, based on the MX 22 and MX 23 designs, with additional features. The model numbers are SAC 22 and SAC 23; info is available at http://www.rane.com/sac22.html. Check them out. ❖

Dennis A. Bohn
V.P. Research & Development
Rane Corporation

WEBSITE LISTING

Anker's Appliances' Electrostatic Loudspeaker Page is a website devoted totally to all things electrostatic. The site is a great reference for those who wish to try an easy-to-build hybrid electrostatic speaker.

Matthew Anker
http://home1.gte.net/res0f2t3/index.htm

SOLUTION TO VIOLENCE

The other day I had the opportunity to talk to an FBI agent while in his home on a technical service call, and I found out that he was also an audiophile. He told me that in his years of investigating and studying criminals—especially those of a violent nature—he had never found one that was an audiophile. He concluded that, as far as he was concerned, those who love the musical arts—especially those who care enough to have their music reproduction sound as close to the original as possible—have a very calm inner self (his words) and are never inclined towards any sort of violence or terrorism.

I personally have heard it said that if the eyes are the windows of the soul, then music is surely the voice of the soul.

It is too bad that those evil idiots who crashed the planes into the World Trade Center weren't audiophiles. If they had been they would have stayed at home and listened to their audio systems, and all of those innocent people would still be alive today!

Rick Spencer
Clovis, Calif.

SUGGESTIONS

I read Charles Hansen's article on the one-chip amplifier with great interest (August '01 *aX*). I congratulate you on a very informative article, and would like to suggest one more.

Any chance you could write an article on building a preamp with MC phono stage using IC op amps? This would be particularly useful because there are so few preamps available with MC phono capability. Second, there are several IC manufacturers who have suggested circuits for a phono stage. Finally, it would be extremely beneficial to read an article about actually building such a circuit and then providing valuable insight, from someone with Mr. Hansen's expertise.

Perchance an article like this has been written. If so, please direct me to it.

I love your magazine, I would like to see fewer reviews on preassembled, store-bought components and more reviews on kits and one-off designs. I like to see construction articles on turntable kits (original designs or Euro-kits, Origin Live and/or Teres), FM tuners (are there any out there?—certainly there are lots of ICs), and the already mentioned MC preamp.

Richard Jenkins
jenkinr@napapipe.com

Charles Hansen responds:

I personally have no interest in making a moving-coil phono preamp. However, a number of articles on the subject have been published:

Discrete Circuits
"A Pre-Preamp for Moving Coil Cartridges," Richard Marsh, *TAA* 1/82, p. 7; correction letter, *TAA* 2/82, p. 50.

"A Moving Coil Preamp, Part 1," Erno Borbely, *TAA* 4/86, p. 7; "A Moving Coil Preamp, Part 2," *TAA* 1/87, p. 30.

Op-Amp Circuits
"Moving Coil Preamplifiers," Walt Jung, Audio IC Op-Amp Applications, 2nd Edition (1978), pp. 127–129; Howard Sams (out of print).

"Single-Ended RIAA Pre-Preamp with 30/40dB Gain," Walt Jung, *Analog Devices System Applications Guide* (1993), Fig. 8.25, pp. 8-37 to 8-39; Analog Devices Corp.

"Three High Accuracy RIAA/IEC MC and MM Phono Preamps," *Analog Devices Applications Reference Manual* (1993), AN-124, p. 4-23; Analog Devices Corp.

"A Low-Noise Pre-Preamp," Frank Mee, *AE* 5/99, p. 10; letter, *AE* 6/00, p. 50.

SYLVANIA DATA

I wish to thank Thomas Sheckels for being fascinated with the Sylvania 7695/7754 vacuum tube audio amplifier circuit published in September '01 *audioXpress*, p. 96. In 1955–1962, I was responsible for all technical data that went into *Sylvania Receiving Tube Manuals* and engineering data sheets, as well as answering all correspondence of a technical nature. In regards to the 7695/7754, when I reviewed the data for this tube, it was obvious to me that it would be a good candidate for a single-ended, push-pull circuit to permit either transformerless or (low impedance) output line operation.

I submitted the circuit to the engineering lab at Sylvania, and Bill Stempler (B.S.-E.E.)—a good audio man—"ran" the test data, which I then included in the data sheet of the 1959 *Sylvania Technical Manual*. This circuit was to all intents and purposes a secondary application and seemed a logical addition to the typical and traditional single-ended, push-pull high-impedance output circuit. Also, at this time there was some experimentation with high-impedance voice coils and there was high availability of 600Ω primary to 4, 8, and 16Ω secondary transformers. Anyway, I thought someone might find this circuit application useful. So that's the story of how this strange circuit was published in the *Sylvania Receiving Tube Manual*.

I also wish to congratulate Mr. Rick Spencer for his excellent single-ended 6550 construction article in the same issue. This amplifier is designed to provide very conservative operation of the 6550s, and the very simple straightforward design is to be commended. Mr. Spencer's many years of experience is indeed insightful; his design substantiates the truism that a simple and straightforward design is the best approach to obtain high-fidelity sound. I also want to thank Mr. Spencer for his kind words on my 100W amplifier article (*GA* 3/00).

I commend Mr. Edward T. Dell's decision to provide the reader with a much more inclusive coverage of audio components. I once compared your magazine to *Audio Anthology*, but your magazine is much more comprehensive than *Audio Anthology*. Since your magazine is a monthly publication (not an annual publication), it obviously requires a great amount of "sweat" every month for you and every member of your staff to maintain this excellent coverage of the audio field.

The articles maintain the same high quality with every issue, and the article contributors are in possession of greater qualifications. Indeed, the college-trained and advanced-degree types are becoming more numerous with each publication; and to their credit they write at a level that reaches all the readers. Again, you have re-energized your magazine and renewed the interest of the readers and many contributing authors.

Joseph Still
Bel Air, Md.

FFT ANALYZER

I just finished reading your article entitled "A High-Quality Two-Channel Chip Amp" (August '01 *aX*, p. 12). Thank you!

I am looking to purchase a good-quality FFT. What test equipment did you use to make the FFT analysis measurements? Do you have any suggestions or comments on FFTs currently available (Picoscope, and so on)?

Russell J. Torlage, CTech
SOTA Instruments, Inc., Canada

Chuck Hansen responds:

I use the Pico ADC-216 because it is a 16-bit DSO optimized for audio testing. I reviewed this instrument in the 5/00 issue of Audio Electronics *("DSO Trio, Part 3," p. 34).*

HELP WANTED

I am hoping you can help me. I am in South Africa and I need a schematic diagram of a Conrad Johnson PV-3 preamplifier to repair a used faulty model. ❖

Jerry Swart
syrinx@yebo.co.za

2001 *audioXpress* Index

Subject

Author

Classifieds

VENDORS

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Forte Acoustics Inc. 19 Hanover Place, PMB164
Hicksville, NY 11801 (516) 249-9632

AD INDEX

WANTED

TUBE HIFI, HORN SPEAKERS, and HIGH END. Altec, Marantz, JBL, Mcintosh, Electrovoice, Fisher, Scott, Dynaco, Leak, Eico, Audio Research, CJ, Quad, Bedini, etc. Sonny, Phone: 850-314-0321 Fax: 850-314-0284 **sonnysound@aol.com**.

Yard Sale

FOR SALE

Hundreds of electronic parts and dozens of back issue audio magazines. E-mail roymottram@aol.com for list.

2 Plitron power transformers (107026201) 750VA 2 × 40V AC—never been used ($133), asking $100 plus shipping. Ashby, ashby123@mediaone.net.

All issues 1991–1996 of the *Journal* of the Audio Engineering Society free to anyone for only the shipping cost. J.L. Markwalter, (941) 764-8713, or markwltr@sunline.net.

Grado reference platinum—eight hours classical use, $200. ADC XLM-III. New-old-stock. Five hours use plus new spare astatic stylus, $70. C. Bauza, (787) 864-0007, PO Box 810, Guayama, PR 00785-0810.

Dynaudio D-52AF midrange dome drivers, unused, in original packaging, $150 pair or trade for Scan-Speak drivers. Ed, (425) 562-9848 or ed@clearsoft.net.

One Altec 511B horn (black), $50. One H-811B horn (gold), $50. Pair of Altec 806A drivers (blue)—need new diaphragms, $65/pair. JBL K120 12″, $100. Jack, (732) 463-1472.

WANTED

Dynaudio 15W75 speakers. I need three but will take one or two if need be. Will pay full price if perfect. Ernie Holmes, (916) 762-7039, ernieh@lanset.com.

Looking for schematics of Paragon 12 and 10e preamplifiers. Will pay for copying and shipping. Reinhard Hoffmann, Wiembecker Str. 1, 32657 Lemgo, Germany, phone/FAX: 05261 87586, in.audio@t-online.de.

Pre-recorded, two-track reel-to-reel tapes. Andy Pennella, (203) 882-2305, janekaufman@worldnet.att.net.

Glass Shard

Breadboarding With Tubes

I'm fairly new to the hobby of vacuum tube audio circuit experimentation, but am already convinced that tube circuits definitely sound better, particularly for electric guitar amplification.

When I'm experimenting with any solid-state design—whether it's my own design or from a published schematic—I always breadboard a prototype circuit before I permanently build it. This prevents surprises, gives me a better feel for what the circuit is doing, and allows quick component value changes. In building permanent versions, all the specific parts are thus readily at hand; just pull it out of the breadboard and it's ready for permanent mounting.

Tubes obviously do not lend themselves to breadboarding...trying to push a 9-pin mini tube into a DIP breadboard will very literally leave you with glass shards! So to get the best of both worlds, I solder 0Ω jumpers (available through Digi-Key) straight down from the solder lugs of 7- and 9-pin panel mount tube sockets. You can then plug this assembly into a breadboard (*Photos 1* and *2*). These 0Ω jumpers have good, stiff leads that will support the tube, and are thin enough to fit the breadboard perfectly.

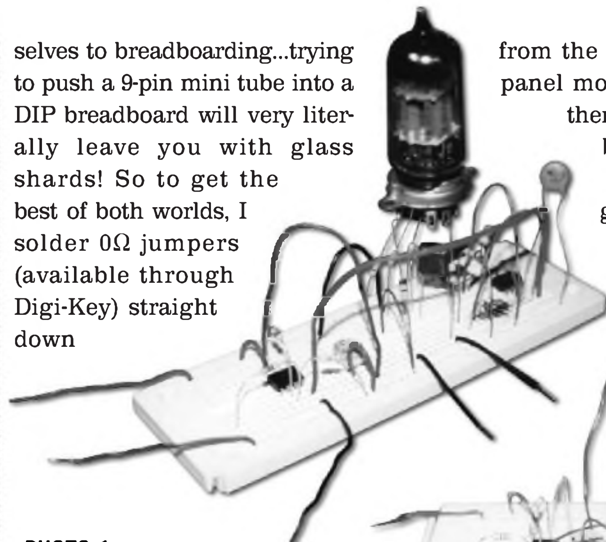

PHOTO 1: Breadboarding a tube.

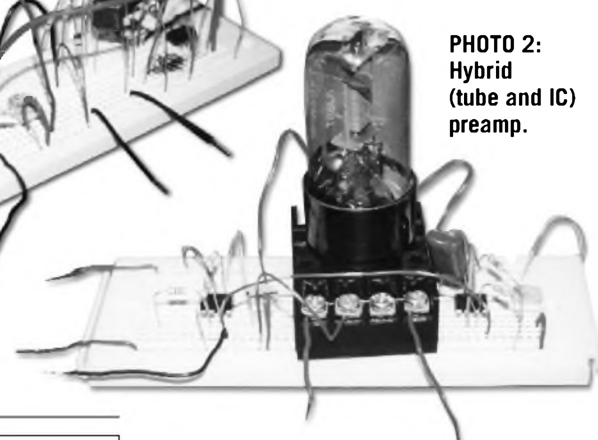

PHOTO 2: Hybrid (tube and IC) preamp.

For bigger octal base tubes, I use octal relay sockets; attach thin, solid wires to the screw terminal; then plug the other end into the breadboard. A big consideration with these techniques is the plate voltage. I do not know what voltage rating a typical DIP breadboard has, but I have experienced no problems up to 160V DC. Beyond that, you'll need to proceed very carefully, because I doubt a breadboard will put up with 470V DC without a smoking, arcing meltdown! ❖

Nick Cinquino
Schaumburg, Ill.

Editor's note: For safety's sake, I think I'd use insulated wire, perhaps color-coded, for tube socket tab connections to the breadboard. Those voltages can short easily or bite the experimenter.—Ed.