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Fig. 6
Paul W. Klipsch
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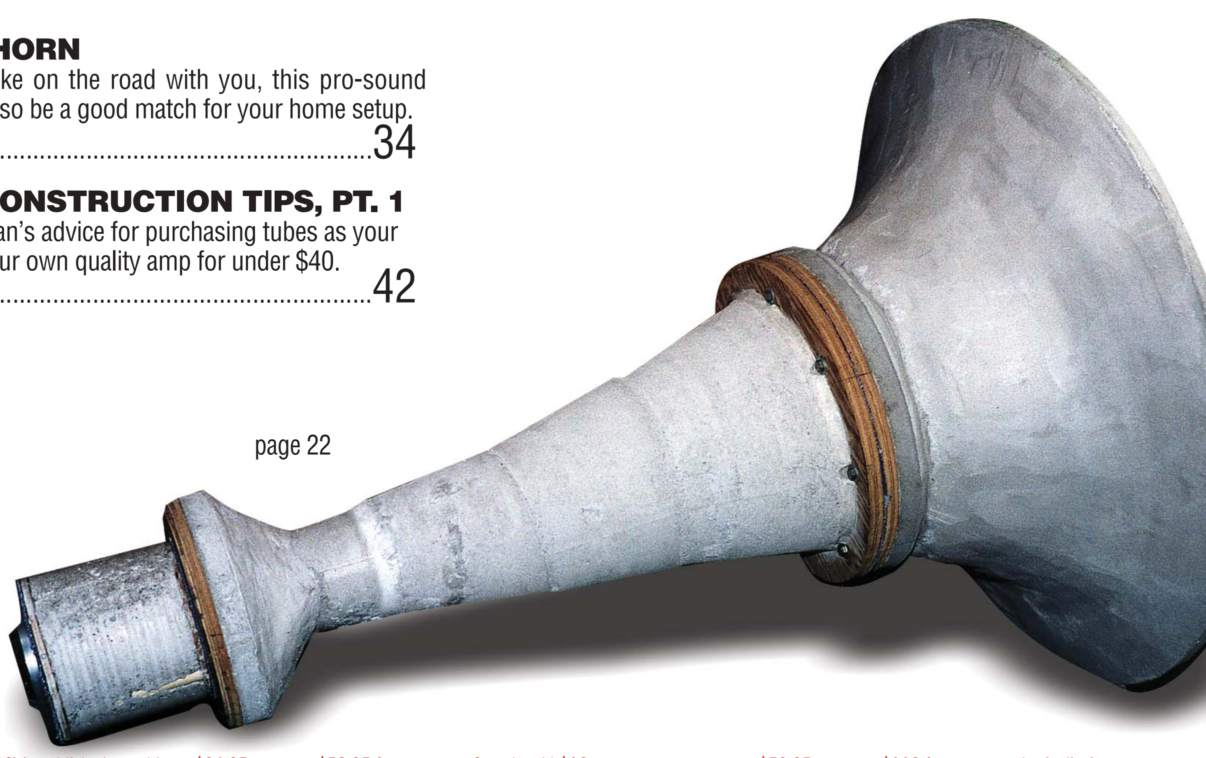

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DEPARTMENTS

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CODE Articles and letters identified with the following symbols:
solid state tubes speakers

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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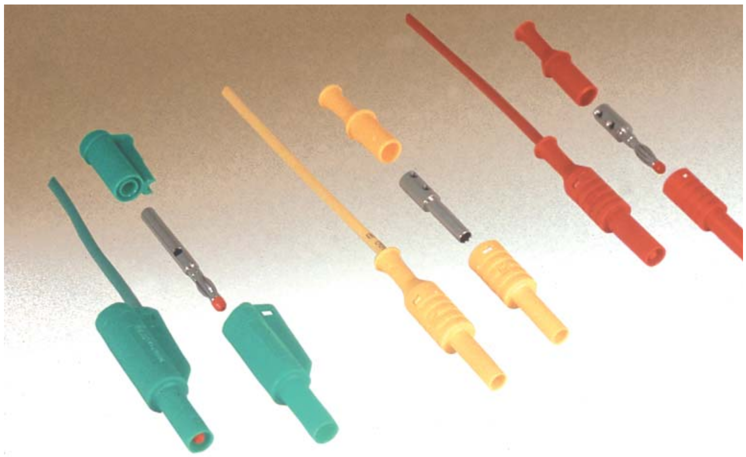
■ NEW BANANA PLUGS BY CAL TEST

Cal Test Electronics has expanded its product line to include three solderless 4mm Safety Banana Plugs. The Sheathed Banana Plug, both Stacking and In-Line, as well as the Sheathed Jack, In-Line, provide a fast, easy way to repair test leads or build custom patch cords. Available in six colors, each plug consists of three parts: two snap-together housing pieces and one metal contact. For more information, visit www.caltestelectronics.com, or fax Bill Hansen at (714) 921-9849.



■ GROMMES HIFI REOPENED

Grommes-Precision, after having moved to a new address (Grommes-Precision, 1331 Estes Ave., Gurnee, IL 60031), has reopened their home audio division, Grommes Hifi. Its first product, an updated version of their popular model 260A monoblock from 1957, is the model 360, and unlike the original, has no global feedback and a true balanced input. Some of the future releases include a passive line stage preamp with real drive, a headphone amplifier, and a phono preamp. For more information, contact Albert Shippits at 1-800-SINCE-46, or e-mail sales@grommeshifi.com.

■ ULTRAVIDEO AND ULTRAAUDIO BY ACCELL

Two new series of products have been released for use in audio and video applications by Accell Corporation. UltraVideo, consisting of composite, component, S-video, and F-pin cables, features 24K gold-plated connectors and contacts, UltraFlexa high quality jackets, and more. UltraAudio products include analog audio cables, subwoofer cables, digital audio coaxial cables, and oxygen free copper cables terminated with gold-plated pin connector. To learn more, visit www.accellcorp.com.

■ QUAD—THE CLOSEST APPROACH

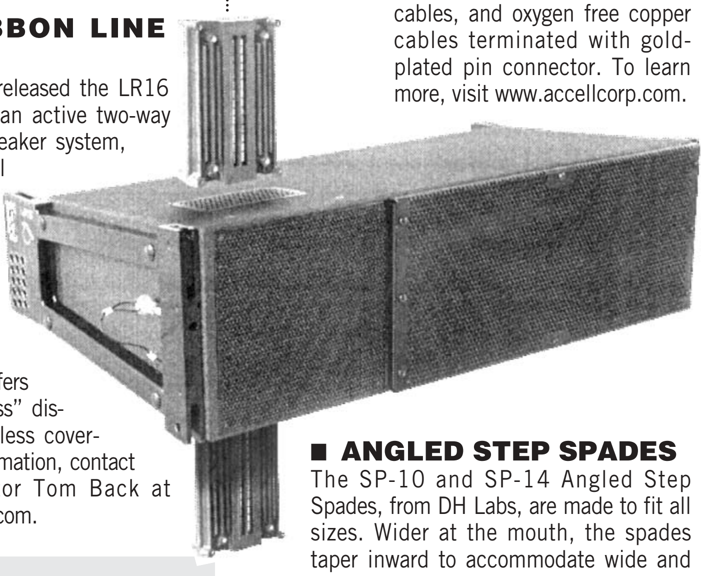
Quad—The Closest Approach, by Ken Kessler, gives the history of one of the oldest and most respected companies in hi-fi. The book gives the background of all their products from the people involved at the time, and includes contributions from two of the company's former Managing Directors as well as E.A.R.'s Tim De Paravicini and valve aficionado Gordon Hill. Available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458-0876, 603-924-9464, custserv@audioXpress.com/ #BKIA1.

DUTCH AUDIO SHOW

A Dutch DIY audio showcase (DDIY2004) will be held April 17 in Apeldoorn, Netherlands. For further information, visit http://members.fortunecity.com/blue4/DDIY 2004 ENG.htm

■ LR16 RIBBON LINE ARRAY

Alcons Audio has released the LR16 ribbon line array, an active two-way line-source loudspeaker system, for use in vertical arrays. Loaded with the Alcons RBN601 ribbon driver for HF reproduction, and able to handle 1000W peak power, the LR16 offers even and "spike-less" dispersion with seamless coverage. For more information, contact Managing Director Tom Back at back@alconsaudio.com.



■ ANGLED STEP SPADES

The SP-10 and SP-14 Angled Step Spades, from DH Labs, are made to fit all sizes. Wider at the mouth, the spades taper inward to accommodate wide and standard binding posts. The spades, manufactured from pure copper and gold plated, have barrels angled at 30° for easy connection to an amplifier or receiver. To learn more, contact DH Labs (Phone/Fax) 561-745-6406 or visit their website at www.silversonic.com.

■ NEW HEADPHONES FROM ULTRASONE

The newly designed HFI-650 and HFI-550 stereo headphones, by Ultrasone AG, offer depth, dimension, and detail with up to 98% reduction in electromagnetic field emissions. The HFI-650 is foldable, circumaural, and has a frequency response of 10Hz–25kHz. The HFI-550, designed with the musician, DJ, and broadcaster in mind, has a more rugged transducer than the HFI-650, with lower impedance and improved sensitivity. To find out more, visit www.ultrasone.net or www.earsaudio.com.

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WBT

A Mini SE Amp

For about $10 worth of tubes and some small transformers, you can build this neat little amp in one weekend. **By Rick Spencer**

One day while trying to think of a new project to build, I remembered reading an *aX* letter a while back from a reader who had requested an article about a small single-ended class A amplifier that used parts available from Radio Shack or other local electronics stores. I have always been a fan of SE amps—and of odd tube types—so I decided to see what I could find on the shelves in my hobby room that might fill such a request.

After searching for a while, I decided to build an SE project with the 12L6s I had found. Now, first of all, don't get excited and think that a 12L6 is the same tube as a 6L6 with a different heater rating! This tube is in the same basic category as a 12W6, a 12CS5, a 12DB5, and—except for the heater—a 6DG6. The normal operating voltages are 200V DC on the plate and 125V DC on the screen. It was frequently used in many old television sets as the audio output tube and is capable of nearly 4W of single-ended power. (I can still remember the great sound that came from an old Zenith set my parents used to own.) The mini-sized amp in this article is only about two-thirds the size of my "little amp" project (*aX* 12/01) and uses a very simple circuit.

I used a 12SN7 for the voltage amplifier, and coupled each 12L6 to a Hammond output transformer, which is small and inexpensive. If you like to experiment, you can even modify the circuit to use other tubes which will deliver a different sound and more power to your speakers. You'll find some of the parts at Radio Shack, while others are available from Mouser and Antique Electronics Supply. I used a black ABS plastic box for my amp, but if you prefer to use a Hammond aluminum chassis then a #P-H1444-16 will do just fine.

The plastic box has been a pleasure to work with. It was easy to drill and punch and leaves no rough edges or sharp filings. The box is even textured with a sort of wrinkle coat finish that doesn't require any painting (*Photo 1*). It measures only 9 × 6 × 3″ and held all of the components of the power supply and amp quite nicely.

Now, some hobbyists may shy away from plastic, fearing that the heat generated by the vacuum tubes would cause a problem, but I found that this amp has a low heat output and the integrity of the box wasn't bothered in the slightest. It's hard to cause a short circuit with a test probe and it's difficult to get zapped if you're touching the chassis when testing!

This easy-to-construct project should take about 16 to 20 hours to complete and can be a great learning experience, especially if you are a beginner in this fascinating hobby. If so, it can be a great confidence booster and can perhaps help you to better understand basic vacuum tube circuits. So, good luck and have fun!

GETTING STARTED

Start by obtaining all of the parts necessary to complete the project, because this will allow you to properly lay out the chassis in the manner you desire. Drill and punch all of the holes needed to attach everything to the box. If you wish to copy the design of this amp, you will find that the topology will come out just fine. If you do use a small plastic box, as I did, just remember to be careful with the component arrangement so you don't cause any shorted connections, and try to insulate the bare leads where possible.

PHOTO 1: Small size of amp and plastic chassis box.

If you look closely at the parts layout in *Photo 2*, you will see that the sockets and terminal strips are spaced to allow for easy wiring hookup. The choke, heater transformer, filter capacitors, and the time-delay module tend to make it look a bit more crowded, but this was the only way I could fit all of the parts inside. If you use the Hammond aluminum chassis, you will have much more room to mount your components.

I installed everything on the bottom of the chassis top plate first and added the filter capacitors last. This gave me lots of space to work in when connecting the heater and amp circuit wiring. The rear panel has the speaker binding posts, input jacks, primary circuit fuse, and the off-on switch installed on it and also has the power cord exiting through a hole with a rubber grommet in it.

The front panel contains the volume controls, complete with aluminum knobs, and a cool-looking blue LED pilot light. I didn't place the main power switch on the front panel for the sole reason of maintaining its clean look with only the control knobs and LED showing.

The chassis top has the HV power and output trannies and, of course, the vacuum tubes arranged in an open, uncluttered fashion. I turned the Hammond #125CSE outputs at a 90° angle to eliminate any magnetic coupling with the high-voltage Hammond #T261M6 transformer. I painted all of the transformers to match the chassis box color and then used brass screws to mount them and the tube sockets to the top plate.

I used an in-line type of fuse for the B+ circuit inside the chassis area. I like to always use a fuse in the HV rail to prevent the destruction of the B+ transformer in the event of a short in one of the filter caps or similar problems.

Wire in the heater circuit first and use the hookup method shown in *Fig. 1*. This allows equal voltage to all of the tube heaters, even though these tubes don't use much heater power. The actual total draw is only 1.50A.

Mount both of the 12L6 sockets in the same direction (key to the left, bottom view), because this will assist you in your wiring layout of both channels and will reduce the confusion that can occur when having to "flip-flop" the wiring scheme. Once you know that you have one channel correct, simply move over to the next one and complete it.

THE CIRCUITS

The single-ended circuit I used here is really simple and contains only the parts necessary to complete the amp and give a smooth, warm, and a somewhat surprising sound. This easy-to-build, low-parts-count circuit has been used on many single-ended amplifiers over the years and is capable of delivering a decent hi-fi sound without the use of "high end" components. It is also a very trustworthy and trouble-free design that should last for many years.

I have included some notes on the sound quality of the amplifier at the end of this article. One-half of the 12SN7 is used as the voltage amplifier for each channel. The input signal enters through the RCA jack (J1), then travels through the volume control, which controls the amount of voltage that reaches the grid of a triode section of the 12SN7. The boosted voltage is then coupled by way of capacitor C2 to the grid of the power tube, which is the final amplification stage.

The Hammond 125CSE couples the plate of the 12L6 to the speaker and matches the plate resistance to the voice-coil resistance, in this case 8Ω. If you are using the 12L6s, as I did, with 8Ω speakers, then use the color codes you see on the diagram (*Fig. 1*). The 12L6 receives its cathode bias by way of resistor R8, which is bypassed by capacitor C3. I chose not to use any feedback from the output secondary windings, but, if you wish to experiment, feel free to try to tailor the sound to your liking.

The power supply uses a choke-input filter setup in order to obtain the 200V of B+ needed. You will see on *Fig. 1* how the choke is wired into the circuit with the first filter capacitor installed after the coil. The initial resistance of the Hammond #158Q choke is around 110Ω and supplies the drop in voltage (under full load) needed to allow 203V DC to enter the primaries of the output transformers.

The final reading at the plates of the 12L6s is 195V, which is close enough. (All of the operating voltages are shown in *Fig. 1*.) The B+ for the screens is set to the correct value by means of resistor R10. (You can disregard the two 10Ω resistors in *Photo 1* that are paralleled and tied into the plates of the 12L6s; these were part of an experiment.)

Most single-ended amps are more susceptible to hum from their power supplies than are push-pull amps because push-pulls can cancel out the hum in their circuits, so I used a large value capacitor in the B+ rail. Its rating is 680µF × 250V. I had it in stock and it works very well in keeping the hum to a minimum, but your amp might do OK with a lower value so you could start out with around 200µF and add more if needed. It also helps to use DC on the heater of the 12SN7. I found that I needed to place my ear 12″ away from the speaker to hear any hum, and since that is not my usual listening position, it doesn't bother me at all.

The blue LED pilot lamp on the front panel is the "ultra bright" type, so I used a dropping resistor to reduce its output brilliance. I didn't want it so bright that it would overpower the glow of the heaters in the vacuum tubes, especially the 12L6s, which have a really nice glow about them. I used a 100k ¼W resistor to obtain the perfect amount of light from the LED, and it is a great indicator that the amp is on when you first glance at it from across the room.

The 12L6 has a heater that was meant for "string series" operation, and it needs a full 10 to 12 seconds to warm up. Even though the tubes used here are inexpensive, it is against my better judg-

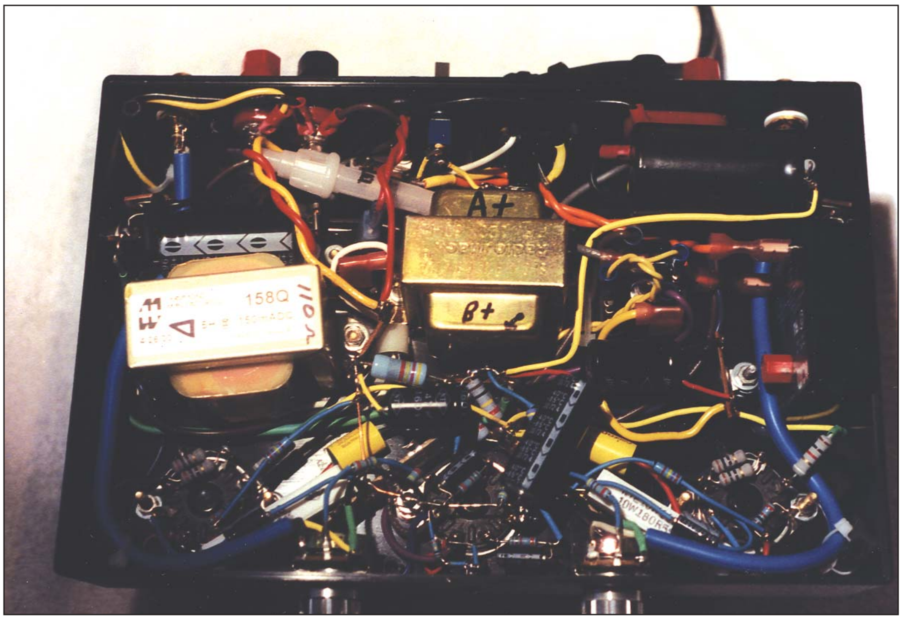

PHOTO 2: Underside of amp showing components and wiring layout.

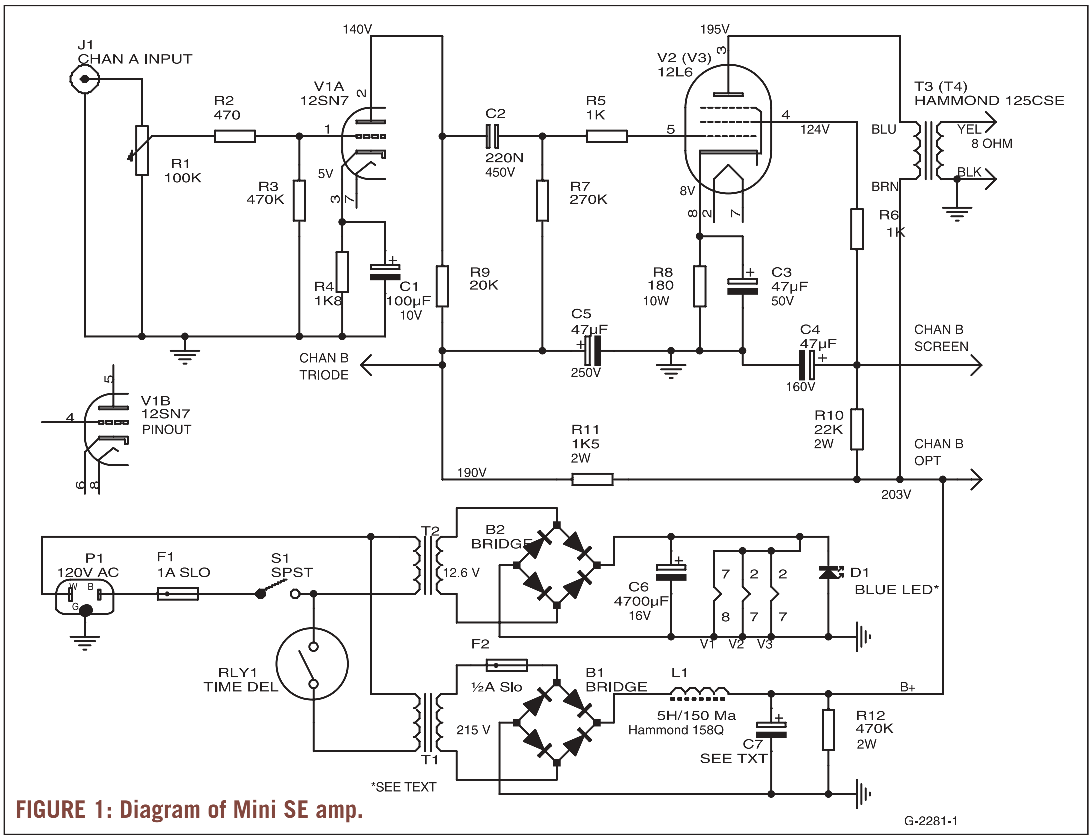

FIGURE 1: Diagram of Mini SE amp.

ment to allow the B+ to hit any tube before the cathode warms up sufficiently, so I installed a time delay relay in the primary circuit of the B+ transformer. This little modular relay is sealed in epoxy, about 2″ square, and has an adjustable time capability. I picked it up from a local appliance parts store, but if you have trouble finding one, you can always use a separate switch for the transformer. Just remember to wait about 30 seconds before turning it on.

I can't even hear the slightest pop or thump when the B+ is applied to the tubes, which is about 30 seconds after the main power switch is engaged. I used a Radio Shack transformer, rated for 12.6V at 3A, for the heaters. I have found over the years that all of their small transformers have performed quite well, and I've never experienced any mechanical hum or vibration from them. Not bad for cheap trannies.

I used a couple of terminal strips at the socket for the 12SN7 to keep the wiring fairly neat, and I used the socket terminals—numbers 1 and 6—on the 12L6s for additional connection points. This is possible because there are no connections to the tube at these pins. Be sure to use shielded cable when wiring the input jacks to the volume controls and use good grounding and soldering techniques.

I used some bare copper wire for all of the grounded connections, then attached it to the ground at the input jacks. The power transformer used here also has a 6.3V winding; so if you choose to try other tubes for this amp, such as a 6V6, a 6Y6, or a 6BQ5, then you can wire the power supply as the correct input load type and then reduce

PHOTO 3: Amp in operation with Dynaco tube CD player and Parts Express speakers.

the B+ voltages to their proper values by means of dropping resistors of sufficient value and power ratings. Speaking of power ratings, I used a 10W resistor for the cathode bias resistor to prevent heat buildup under the chassis. I realize that a 5 watter would probably have been OK, but why take the chance?

By the way, I used metal film resistors throughout the amplifier circuit. If you find that the sound from your amp is a little on the "bright" side, you might try some carbon resistors to take the edge off and mellow it out. I used "Auricaps" for the coupling capacitors because I love the clean way they sound, but if you want to save some money then use the "orange drop" capacitors available from AES.

After you have finished the assembly and wiring of your amp and are sure that everything looks OK, power it up with all of the tubes in place. Be sure to have your speakers connected. After the amp has warmed up for a couple of minutes, use a DVM to measure the operating voltages at the points shown on the diagram in *Fig. 1*. Of course, you will do this with the negative lead from the meter to a ground connection point and then use the positive lead to touch the reference points noted on the diagram.

If you are using test leads with long pointed bare metal ends, it would be a good idea to slip a piece of heat-shrink tubing over the tip and leave only a small portion of it exposed. Shrink the tubing with a heat source and you will have a much safer test probe. Be careful here because even though the B+ is only around 200V, it can still get your attention if you are careless! If your meter readings are within 5% or so of the ones on the diagram, you can be fairly sure that all of the circuits are performing OK.

In case you do find a problem such as a bad solder joint causing an erroneous reading, power the amp down, discharge the power caps, and correct it, power up again, and take another reading. If the terminal on one of the tube sockets needs resoldering, be sure to remove the tube first! Now turn on and activate the audio source that you are plugging into the input jack (CD player, FM tuner, tape deck, and so on) and adjust the volume controls to balance the output of both channels as desired. With the individual level controls on this unit, no balance control is required. You are now ready to enjoy the sound of your new single-ended amplifier.

Incidentally, this small amp takes up less than a square foot of shelf space, is very lightweight, and uses only 45W of electrical energy.

NOTE: If you like to experiment with power supplies and want to use all Radio Shack transformers, you could use a backwards wiring technique with a second heater tranny and wire it as a voltage doubler type of circuit. Then adjust your B+ voltages accordingly. Most experienced hobbyists already know how to construct this type of power supply. In this article I'm not including this information because I'm trying to keep the project as simple as possible.

THE SOUND

I'm not going to try to tell you that this little SE amplifier will blow your socks off and sound better than any single-ended 2A3 ever could. That would really be a bit of a stretch! What I will say is that this amp really surprised me with its sound and the way it handles most music. I was not ready for the warm, smooth sound when I first started it up.

Most amps require a fairly good break-in period before sounding their best, but this little guy came out of the corner swinging! First of all, the specs on the Hammond output trannies must be really conservative because the fullness of the music I heard you can only hear from bigger iron.

The stated frequency range is from 100Hz to 18kHz, but in no way does this amp sound puny or lacking in sonic energy as you might expect. It has a real "peppy" and "happy" sound! Except for music with a lot of very low bass passages, it seems to handle whatever I put into it with relative ease. It is well known that the output transformer has a very large effect on the final sound of any

PARTS LIST

PART SYMBOL & NAME	NUMBER	SOURCE
T1 = Hammond, main power B+	T261M6	AES
T2 = 12.6V × 3A, heater power	273–1511	Radio Shack (RS)
T3, T4 = Hammond output transformer	125CSE	AES
Ll = Hammond 5H, 150mA choke	158Q	AES
FWBl = Bridge rectifier, 600V × 6A	583-BR66	Mouser
FWB2 = Bridge rectifier, 50 × 10A	583-BR1005	Mouser
LED = Blue color	276-0316	RS
S1 = SPST switch, main power	275-0635	RS
F1 = Fuse 1A slo-blo	270-1021	RS
F2 = Fuse ½A slo-blo	270-1018	RS
TDR = Timed delay relay; 24 to 240V AC; Delay on Make: ICM model #ICM102; time range is 1sec to 10mins; From appliance parts co.		
V1 = Dual triode, vacuum tube	12SN7	AES
V2, V3 = Beam power Amp tube	12L6	AES
R1 = Volume control, 100k, audio	271-1722	RS
(Aluminum knobs for controls)	274-0424	RS
R2 = 470R, ½W		
R3 = 470k, ½W		
R4 = 1.8k, ½W		
R5, R6 = 1.0k ½W		
R7 = 270k, ½W		
R8 = 180R, 10W power	280-CR10-180	Mouser
R9 = 20k, ½W		
R10 = 22k, 2W power		
R11 = 1.5k, 2W power		
R12 = 470k, 2W power		
(You can order all of these fixed value resistors, except for R8, from AES, and all of the resistors, including R8, from Mouser.)		
C1 = 100µF × 10V capacitor	140-XAL10V100	Mouser
C2 = 220N × 450V capacitor (Auricap)	C-AUD22-450	AES
** If Orange Drop **	C-RD22-600	AES
C3 = 47µF × 50V capacitor	C-ET47-50	AES
C4 = 47µF × 160V capacitor	C-ET47-160	AES
C5 = 47µF × 250V capacitor	C-ET47-250	AES
C6 = 4700µF × 16V capacitor	C-ET4700-16	AES
C7 = 680µF × 250V capacitor	5985-250V680	Mouser
MISC.: Chassis boxes, tube sockets, speaker binding posts, terminal strips, RCA jacks, hook–up wire, and other hardware is available from Antique Electronics. Other hardware available at Radio Shack, or other local electronics stores.		
CATALOGS: AES phone 1-480-820-5411		
MOUSER phone 1-800-346-6873		

tube amplifier, and all I can say here is, "Thank you, Hammond engineers!"

The midrange is very open, and the treble, though somewhat rolled off above 12kHz, is downright smooth. The bass is very good, although not as good as my 6550 SE amp (*aX* 9/01), as you would expect, but when I played Holly Cole's album "Blame It on My Youth," track #4, the plucked bass sounded full and fairly solid. Her voice came through crystal clear and natural.

Another surprise was the violins and bowed basses on the Philips album #416361-2, by John Williams and the Boston Pops. The bowed basses on track #2, "Clair de lune," really filled up the room—although not a very large room—and then the goosebumps came. When track #4, "The Swan," started, Martha Babcock's wonderful cello almost floated me away to another time and place.

Another album that I used to test the performance was what many audiophiles consider one of the best recordings of plucked bass sound of all time, the Three Blind Mice CD #GS-CD006. Tracks #5 and #8 came through this mini-sized amp sounding very natural, and, while not earthshaking, were notably accurate. For such a small low power amp, the only words I can find to sum up the experience are—"WOW, Big Surprise!"

Originally I used the GE brand of 12L6 tubes, which you see in *Photos 1* and *3* and are date-coded 1962. They sounded great, but after Steve at AES sent me a pair of "Dumont" branded tubes, dated 1971, and I gave them a three-hour break-in period, they gave me an even better quality of sound.

In *Photo 3* you will see the amp in use with a Dynaco CDV-1 vacuum tube CD player with 2.0V of output. This CD player drives the amplifier to full output with both volume controls at their 1:00 position and with minimum distortion. (Actually, I never pay any attention to distortion in single-ended amps.) Also in the photo, you will see the amp driving the Parts Express #BR-1 speakers. Although not really high-efficiency speakers, they are playing very loud with the amp's volume controls at the 2:00 setting, and they sound very nice.

If you have some good full-range speakers that are at least 90dB efficiency, I think you will be satisfied with the results you will obtain from this amplifier. If you choose to become deeply involved in this project and wish to achieve as good a sound as possible, you can upgrade to a bigger (and better) single-ended output transformer. There are quite a few on the market now. You can always use the "high end" tube sockets, resistors, wiring, volume controls, and other such components. The possibilities are almost endless.

Again, this amp project doesn't really take very long to complete, and I'm sure that someone out there much wiser than I can probably improve on it. So, take your time, don't rush, enjoy yourself. Follow the diagram closely, and, when finished, your version of this amplifier should work when you first power it up, and give you many hours of enjoyment. The finished product will also give you the pride of knowing that you built it yourself. Happy building and happy listening. ❖

A Bridge Adapter

This versatile device will open up a number of new audio applications for you. **By Dick Crawford**

Bridged amplifiers are becoming more common. Recently I was testing some switching amplifiers which had bridged (balanced) outputs. I found that I needed to make balanced measurements on the bridged outputs, or I would get faulty distortion measurements. But most distortion instruments use a single-ended input. So I needed a balanced-to-single-ended adapter of very low distortion.

That was the genesis of my bridge adapter. I decided to add some circuits which would provide balanced outputs with single-ended inputs. You can use this to make a bridged power amplifier from two single-channel amplifiers. Finally, you can use the bridge adapter as a balanced input to balanced output, with selectable gains or attenuations.

SOME RESULTS

Figure 1 shows the intermodulation distortion (IMD) of the bridge adapter, using 19.2kHz and 20kHz two-tone intermodulation signals. The IMD is a minimum of about 1PPM (0.0001%) at about 6V peak-to-peak (2V RMS) and rises to about 6PPM (0.0006%) at signal levels above 30V peak-to-peak (10V RMS). This means that the bridge adapter is useful for measurements of IMD down to 0.001%, up to 30V peak-to-peak. The IMD increases rapidly above about 31V peak-to-peak, as the op amps start to clip. I tried to measure the total harmonic distortion (THD) at both 1kHz and 20kHz, but the THD was below what I could reliably measure (about 0.004% THD) until the output was above 30V peak-to-peak.

I also measured the common mode rejection (CMR) of the bridge adapter as 50dB at 1kHz and 46dB at 20kHz. This decline in CMR at 20kHz is partly due to the 5% tolerance of C2, C6, C8, and C12. These CMR measurements are mostly good luck, as analysis indicates that with 1% resistors and 5% capacitors, the CMR could be as bad as 24dB at 1kHz and 22dB at 20kHz. This is still good enough, but you could improve the worst-case CMR at 1kHz to 44dB by using 0.1% Holsworthy resistors (available from Mouser).

All of this is in a box of 4″H by 6″W by 2″D, with a single printed circuit (PC) board of 1.6″D by 5″W. There are two identical channels (for glorious stereo). There are no pots or adjustments, as all the gains and attenuations are switch selected. The power supply is on the PC board, and is powered by an external AC to AC adapter that plugs into an AC wall outlet.

WHY A BRIDGE?

Bridged amplifiers can deliver as much as four times the power into a loudspeaker as single-ended amplifiers. With a bridged amplifier, the loudspeaker is connected between the outputs of two amplifiers. These outputs are balanced (or differential), so that when one output goes positive, the other goes negative, and vice versa. Thus the signal to the loudspeaker is the difference between the two outputs.

This can give as much as twice the voltage to the loudspeaker as the single-ended amplifier. Twice the drive voltage means four times the power. Because of this, bridged amplifiers have been popular in automotive applications for years. Lately they have become more common in other low-voltage applications, such as computers and portable equipment.

PHOTO 1: The front panel of the author's bridge adapter.

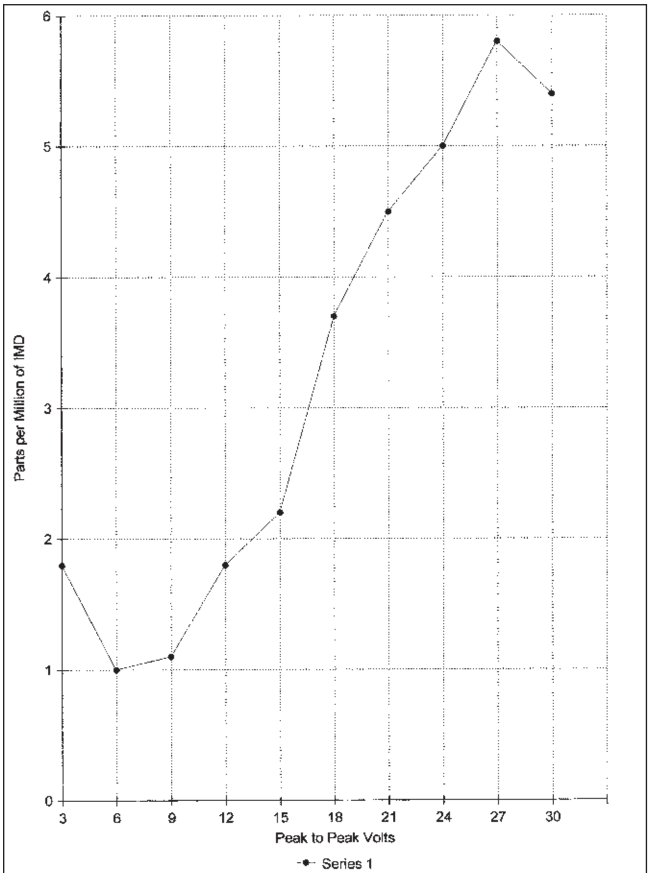

FIGURE 1: IMD of bridge adapter.

A-2243-1

Bridged amplifiers seem well-suited to switching amplifiers, partly because switching amplifiers often require only a single supply voltage. Switching amplifiers have significant advantages, such as less heat, smaller size, and lower cost. But switching amplifiers tend to have greater distortion than conventional amplifiers. Some of the designs for switching amplifiers use clever distortion-reducing schemes, wherein the distortion is of the same amplitude and phase at each of the outputs.

When a signal is the same at both outputs, it is called a common mode signal. If you measure either output, you will find significant distortion, but if you measure the difference between the two outputs (which is how the loudspeaker is connected), then the distortion is lower. In order to accurately measure the distortion of this amplifier, you need a measuring system that mostly ignores the common mode distortion. The bridge adapter does this.

You can make bridged amplifiers by using two single-ended amplifiers with balanced inputs. Often the circuitry needed for bridging the amplifiers is part of an amplifier, but in many cases you will need an adapter which takes a single-ended input and provides a balanced output. Balanced signals are often used with microphones because balanced signals can reduce pickup of hum and noise. Balanced signals have the same benefit for large signals. You can use the bridge adapter to convert single-ended inputs to balanced outputs, with low distortion, up to 10V RMS per output (20V RMS balanced).

Sometimes balanced amplifiers are called differential amplifiers. Sometimes the outputs of a balanced amplifier are defined as +phase (in phase) and –phase (out of phase). I use +phase and –phase for the front panel.

DESIGN

The schematic of channel 1 is shown in *Fig. 2*. All of the amplifier stages are inverting, as this usually gives lower distortion than non-inverting amplifiers. In order to reduce noise, the feedback resistors are a low 3kΩ (R10 and R20). The gain is set by the ratio of the feedback resistor to the input resistors:

Maximum single-ended gain = R10/R9
(R20/R19 are the same as R10/R9).

You can select the single-ended gain by switching in resistors, which, in effect, change R9 and R19. I've chosen to alter the gain in 6dB steps from +6dB to –36dB. I chose 6dB steps for the gain because this makes the calculations of the conversion from single-ended to balanced a little easier. I chose the wide range of gains to cover the large range of signals available from a power amplifier. All of the gains

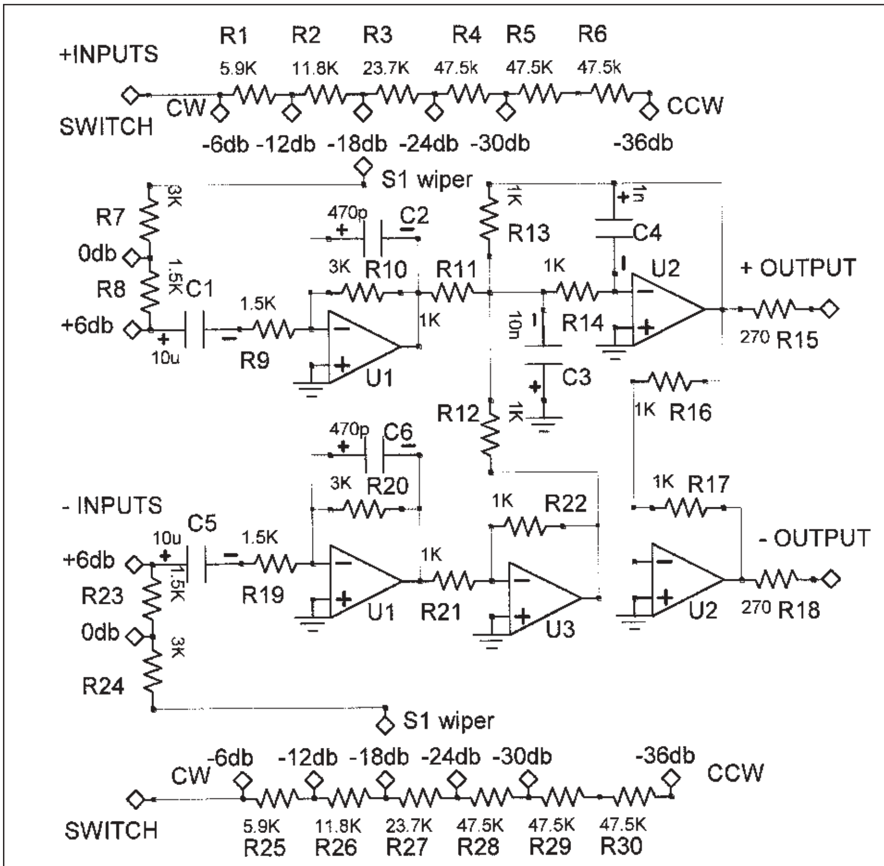

FIGURE 2: Schematic of channel 1 of bridge adapter.

A-2243-2

are governed by 1% fixed resistors, as this helps preserve the common mode rejection.

Channel 2 is identical, except that the part numbers are different. Resistor part numbers are increased by 30, capacitor part numbers by 6, and op amp part numbers by 3. The gain of channel 2 is adjusted by S2, and this is independent of channel 1.

The input amplifier is AC-coupled, which means that you need a large (10µF) film capacitor to keep the low-frequency rolloff below about 10Hz. AC coupling is needed because the outputs of some bridged amplifiers have a large DC component. The high-frequency response is rolled off above about 50kHz.

The reason for the high-frequency rolloff is that switching amplifiers (Class D or T) often have significant high-frequency switching signals in their outputs. Distortion measurements

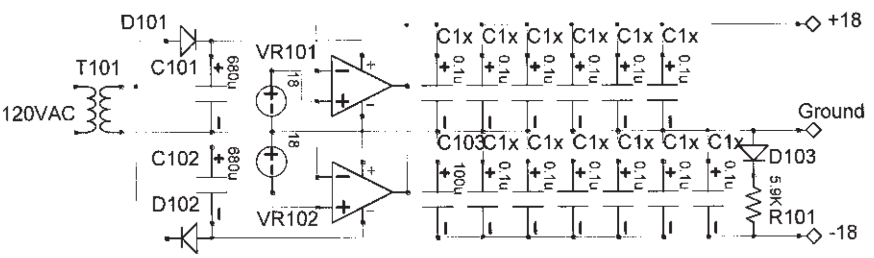

FIGURE 3: Schematic of power supply.

A-2243-3

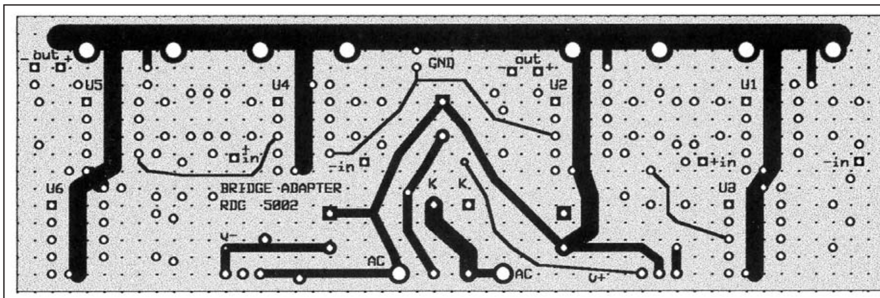

FIGURE 4: Component side of PC board.

A-2243-4

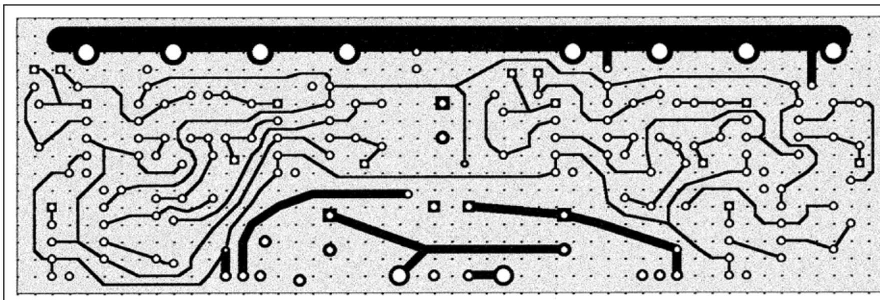

FIGURE 5: Circuit side of PC board.

A-2243-5

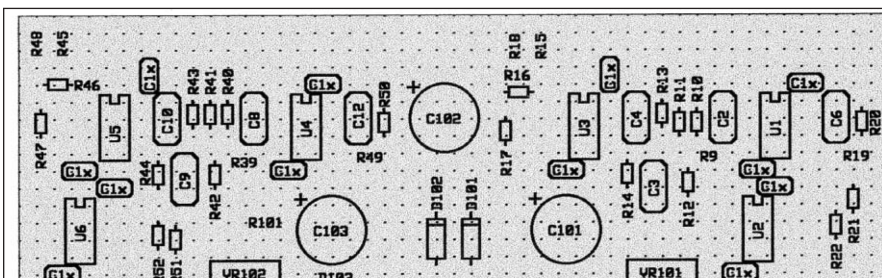

FIGURE 6: Component loading of PC board.

A-2243-6

will sometimes interpret these switching signals as distortion. So it is advisable to reduce these signals by means of low-pass filters.

There are both in-phase and out-of-phase signals in the two halves of U1. U3 inverts the phase of the –input, thus making it in-phase with the +input. The two signals from the +input and the –input are summed in half of U2. This summing amplifier also acts as a two-pole low-pass filter.

There is a slight peaking at 20kHz in the summing amplifier to compensate for the slight loss at 20kHz in U1. The result is an overall high-frequency response that is flat within 0.1dB at 20kHz, and 3dB down at about 50kHz. The low-frequency response measures about 1dB down at 20Hz.

The output of the summing amplifier is in-phase with the +input. The other half of U2 is an inverter providing an output which is in-phase with the –input. This gives inputs which are in-phase and out-of-phase, as well as outputs which are in-phase and out-of-phase. This provides for both single-ended and balanced inputs and outputs. All inputs and outputs in a channel have the same gain.

GAIN

The gain of the bridge adapter can be confusing. When used as a single-ended input and single-ended output amplifier, the maximum gain available is +6dB, and the minimum gain is –36dB. This is the way that I have labeled the front panel (*Photo 1*). For balanced inputs and balanced outputs, you can calculate the gain from the difference signals of both inputs and outputs. This gives twice the output signal as the single-ended case, and thus the maximum gain is +12dB and the minimum gain is –30dB.

POWER MEASUREMENTS

The optimum single-ended output level seems to be about 2V RMS, which gives a good compromise between distortion (which is minimized by a small output signal) and signal-to-noise ratio (which is maximized by a large output signal). At 2V RMS output, the minimum single-ended input level (at +6dB gain) is 1V RMS, and the maximum input level (at –36dB gain) is about 126V RMS. For differential inputs with single-ended outputs the numbers are the same: 1V minimum differential input and 126 maximum differential input, for 2V RMS single-ended output. The 126V figure corresponds to 1985W (that was a good year for watts) into 8Ω, and, of course, twice that (almost 4kW) into 4Ω. This seems adequate for the measurement of bridged power amplifiers.

If you are willing to use an output signal level of more than 2V RMS, the maximum input signal level (and power measurement capability) increase. At 10V RMS output, the maximum power is 25 times that above, or nearly 50kW. Theory says that the bridge adapter can handle this much signal (of course, the 8Ω load resistor is not part of the bridge adapter), but, strange to say, I don't have an amplifier that I can use to test this.

I should mention that some high-power bridged amplifiers operate directly from the AC line, without a power transformer. Testing these "hot chassis" amplifiers is potentially hazardous (pun intended) and requires some special isolation transformers and great care. I've measured these and I have the scars to prove it. I do not recommend that amateurs try measurements on "hot chassis" electronics.

CONSTRUCTION

You can build the bridge adapter without the switches, and for applications where the gain range is limited (such as driving single-ended amplifiers in a bridge configuration), this is fine. But after building a breadboard without switches, I preferred more gain options, so I added the switches.

If you are going to use the bridge adapter for measuring the output of bridged amplifiers, then the switches are helpful. The resistors that determine the gain are off the PC board, either on the connectors or the switches. I think that the rank amateur could build the bridge adapter without the switches, but I recommend the experience of building a couple of electronic projects for someone building the switch version.

PRINTED CIRCUIT BOARD

Both channels and the power supply are on the PC board. The connectors are octal phono plugs from Radio Shack, and the ground tabs of all those phono plugs solder directly to the PC board. This forms a good ground connection and also a good mechanical support for the PC board. The octal phono plugs are then mounted to the front panel, holding the PC board in position.

Also on the PC board are the ±18V power supplies for the bridge adapter, and this gives lots of output capability. As shown in *Fig. 3* (the schematic for the power supply), the regulators for

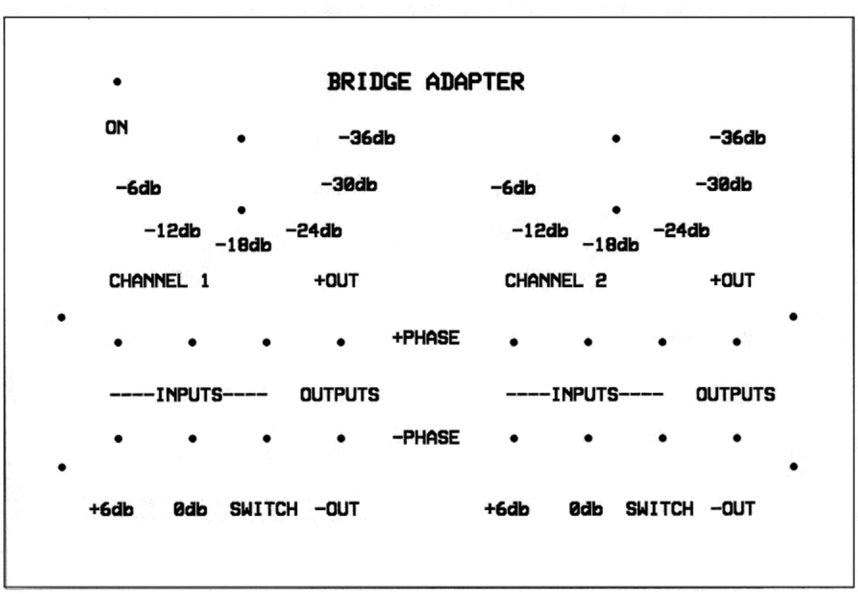

FIGURE 7: Front panel layout.

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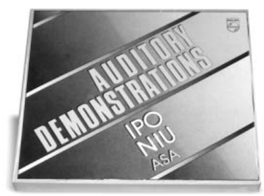

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the ±18V power supplies are fed from unregulated ±25V supplies, which are generated by an 18V AC to AC adapter, rectifying diodes, and 680µF electrolytic capacitors.

The three electrolytic capacitors in the bridge adapter are all premium units chosen for long life. The current drain of the +18 and −18V supplies is about 100mA, and at this current all of the power-supply components are lightly loaded, which should result in a long life. The power-supply schematic also shows an LED (D103), which is mounted to the front panel and lights when the unit is on.

Also shown in the power-supply schematic are the bypass capacitors, C1x. There are two bypass capacitors for each op amp, one for the +18V supply and one for the −18V supply. I label the bypass capacitors as a generic C1x because labeling all of them in *Fig. 1* would add confusion. The bypass capacitors are all the same value, so there is no need to separately identify them.

The PC board is 1.6″ deep by 5″ wide. The component side is shown in *Fig. 4*, the circuit side in *Fig. 5*, and the component loading diagram in *Fig. 6*. It is easier to load and solder the PC board first.

To save space, I mounted all of the resistors vertically. That is, I bent one of the resistor leads about 135°, forming a V, and then soldered the open ends of the V into the designated PC holes. The capacitors are loaded onto the PC board in the usual manner. The voltage regulators are mounted so that the part with the lettering faces the front panel.

The rectifying diodes (D101 and D102) and electrolytic capacitors have their positive polarity indicated by square holes on the PC board. The positive (cathode or banded end) of the diodes also have the letter k (for cathode, as the Germans spell it) at the positive end. The op amps' orientation is indicated by a square hole at pin 1. I recommend sockets for all the op amps, with the socket indentation being on the pin 1 end.

S1 AND S2

The switches (S1 and S2) are separate from the PC boards, and are mounted directly to the front panel. The recommended front-panel layout (*Fig. 7*) has a dot indicating the position of a hole, which accommodates a tab on the switch; this tab prevents the switches from becoming loose with use. I recommend using these holes and switch tabs, and hiding them with a large knob.

The switches are wired by bending the resistors in a V shape, and soldering the resistors in the proper eyelet. R5 and R6 are wired together to form a V, as are R29 and R30, R35 and R36, and R59 and R60. (The series combination of these resistors is used to increase the maximum input voltage at −36dB gain.)

When wiring S1 and S2, observe the CW (clockwise) and CCW (counterclockwise) indications as shown on the

PARTS LIST FOR BRIDGE ADAPTER

CAPACITORS			
C1, C5, C7, C11	10µF/100V film Mouser 5989-100V10	4@$2.92	$11.68
C2, C6, C8, C12	470pF/5% npo ceramic Mouser 581SR151A471J	4@$0.26	$1.04
C3, C9	10nF/2% film Mouser 140-pf2a103f	2@$0.43	$0.86
C4, C10	1nF/2% film Mouser 140-PF2A102F	2@$0.43	$0.86
C101, C102	680µF/50V electrolytic Mouser 647-UPW 1H681MHH	2@$1.71	$3.42
C103	100µF /100V electrolytic Mouser 647-UPW2A101MHH	1@$1.38	$1.38
C1x	0.1µF X7R/50V ceramic Mouser 581-sr215c104k	12@$0.14	$1.68
CONNECTORS			
Octal RCA Phono	Radio Shack 274-370	2@$2.29	$4.58
DIODES			
D101, D102	1A/200PRV Mouser 583-1N4935	2@$0.08	$0.16
D103	Green LED Radio Shack 276-069	1@$2.29	$2.29
ENCLOSURE			
4″H × 6″W × 2″D	Radio Shack 270-1806	1@$4.39	$4.39
INTEGRATED CIRCUITS			
VR 101	18V pos. regulator Mouser 513-NJM7818FA	1@$0.36	$0.36
VR 102	18V neg. regulator Mouser 513-NJM7918FA	1@$0.56	$0.56
U1–U6	NJM 5532D Mouser 513-NJM5532D	6@$0.55	$3.30
PRINTED CIRCUIT BOARD			
1 at estimated $20.00			$20.00
RESISTORS—ALL 1% METAL FILM (MF) UNLESS OTHERWISE NOTED			
R1, R25, R31, R55, R101	5.9kΩ Mouser 271-5.9k	5@$0.09	$0.45
R2, R26, R32, R56	11.8kΩ Mouser 271-11.8k	4@$0.09	$0.36
R3, R27, R33, R57	23.7kΩ Mouser 271-23.7k	4@$0.09	$0.36
R4–R6, R28–R30, R34–R36, R58–R60	47.5kΩ Mouser 271-47.5k	12@$0.09	$1.08
R7, R10, R20, R24, R37, R40, R50, R54, R101	3kΩ Mouser 271-3k	9@$0.09	$0.81
R8, R9, R19, R23, R37, R39, R49, R53	1.5kΩ Mouser 271-1.5k	8@$0.09	$0.72
R11–R14, R16, R17, R21, R22, R41–R44, R46, R47, R51, R52	1kΩ Mouser 271-1k	16@$0.09	$1.44
R15, R18, R45, R48	270Ω Mouser 271-270	4@$0.09	$0.36
SWITCHES			
S1	two pole, six position Mouser 10WW026	2@$1.57	$3.14
TRANSFORMER			
T101	AC to AC adapter, 18V secondary Mouser 412-218054	1@$5.49	$5.49

schematic, else the front panel gain markings will be incorrect. Solder resistors R7 and R24 to the wiper tab of S1, and R37 and R54 to the wiper tab of S2. After mounting the switches to the front panel, I used hookup wires to connect the switches to the PC board and phono connectors.

CONNECTORS

The connectors have their ground leads soldered directly to a ground plane on the PC board. If the ground tabs of the connectors are first bent together, then the PC board will slide between the ground tabs and indent into place at the correct location on the ground plane. Solder all the connector grounds to the ground plane, being careful to keep both connector boards aligned so that they will mount properly to the front panel.

Resistors R8, R23, R38, and R53 are soldered to the phono connectors, going between the +6dB terminals and the 0dB terminals. C1 has one lead soldered to a +6dB terminal and the other lead soldered to one end of R9. The other end of R9 is soldered into a square hole (marked R9 on *Fig. 6*) on the PC board. Likewise, C5 is connected to R19, C6 is connected to R39, and C11 is connected to R49.

In a similar manner, one end of R15 is soldered into a square hole (marked R15 in *Fig. 6*), and the other end is connected to the +output connector. R16, R45, and R46 also go from the PC board to the output connectors. R9, R39, R15, and R45 are from the top of the PC board, while the others go from the circuit side (bottom) of the PC board.

FRONT PANEL

Figure 7 shows a layout for the front panel, which I copied onto an 8½ by 11 transparency using a standard copying machine. Then I temporarily centered the transparency over the metal front panel and marked the location of all the holes onto the front panel, using a center punch. Remove the transparency. Then drill ⅛″ pilot holes for all hole locations and enlarge these holes to the desired size with a Unibit. Check to see whether the connectors and switches fit properly into the holes.

Glue the transparency onto the front panel using your favorite transparent glue. I used GE Silicone 1 clear Kitchen and Bath, which worked well enough, but there are probably better adhesives available. Let the glue set. Using a hobby knife, cut out the holes in the transparency so that they are the same size as the holes in the front panel.

Mount the switches first, using lockwashers. Solder the leads going from the switches to the PC board and then mount the board and the LED. Wire the LED to the board (*Fig. 7* shows the location of holes for the LED (D103) and its associated resistor (R101)). Cut the connector from the 18V AC leads of the AC to AC adapter.

Drill a 3⁄16″ hole in the left side of the plastic enclosure and pass the 18V AC leads from the AC to AC adapter through that hole. Tie a slip knot about 4″ from the end of the 18V AC leads. Strip, tin, and then solder the ends of the 18V AC leads to the PC board. The polarity is unimportant, so simply solder one lead into each of the AC holes on the board.

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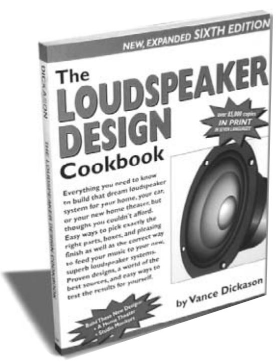

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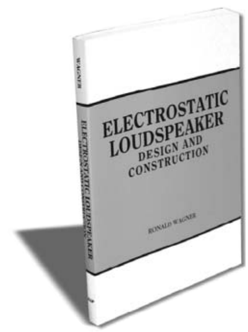

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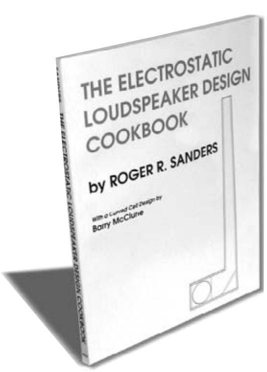

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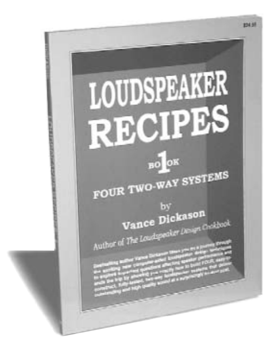

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TESTING

The purpose of the testing is to determine that each stage of the bridge adapter works. To do this you will need a DMM (digital multimeter) and some clip leads. First, plug the AC to AC adapter into an AC wall outlet that is turned on. With the DMM check the DC voltage between ground and the positive lead of C101; it should be between +22V and +35V.

Repeat for the negative lead of C102—between −22V and −35V. Check that the output of VR101 is between +17 and +19V and that the output of VR102 is between −17 and −19V. Be sure that the cases of VR101 and VR102 don't become hot; warm is OK. Hot is when you can't hold your pointing finger on the part for a full second.

While monitoring the +18V, plug in U1, which should not become hot, and the +18V should change less than 0.1V. Repeat for U2, and then for U3. Touch U1, U2, and U3: all should be simply warm. Check that pins 1 and 7 of U1, U2, and U3 all read less than 0.1V DC. Check the AC voltage at pins 1 and 7 of U1, U2, and U3; all should read less than 0.1V AC.

Set S1 and S2 for −24dB gain and connect a clip lead from the cathode end (striped end) of D102 to the +switch input. This will put 18V AC onto the +switch input. Move the DMM to the +output and measure the AC voltage. It should read between 1 and 2V AC.

Move the DMM to the −output; it should read within 0.1V of the +output. Move the clip lead to the −switch input. The DMM should read the same AC voltage within 0.1V AC. This completes the testing of channel 1. Repeat this process for channel 2.

These steps test the functioning of the power supply and the op amps, but it will not guarantee that all the resistors are in the right place or are soldered properly. Double-check the resistor values against the loading diagram, and resolder all the PC solder joints, the switch solder joints, and the connector solder joints.

PARTS LIST

In the parts list, you can see that the parts cost less than $70. The most expensive item is the PC board at $20. For another $20, you could substitute a metal case, which should reduce the 60Hz hum pickup to below 1µV.

HUM AND NOISE

Being in a plastic enclosure, the bridge adapter is susceptible to stray magnetic fields, and this may mean hum. Hum is bad for THD measurements, as most THD measurement instruments cannot distinguish between hum, noise, or distortion. Try to place the bridge adapter away from large power transformers, and well away from its own AC to AC adapter. Be sure to ground the metal front panel to the printed circuit ground, or you may have more hum.

I put a ground lug under one of the 4-40 nuts used to mount the PC board, and then soldered that ground lug to the PC ground. The ground for the bridge adapter will be affected by the ground leads of the cables connected to it. A good way to minimize the hum and noise of the bridge adapter is to use phono cables that are bundled together, so that ground loops are minimized. The ubiquitous stereo hookup cables are a good example.

On my workbench, which is surrounded by instruments and Variacs and AC line cords, I measured a hum level at the output of 4.4µV at maximum gain. At lower gains this reduces to about 1µV. This is well below the hum level in most preamplifiers and power amplifiers.

THERMAL

I measured the temperature rise of the voltage regulators and op amps at about 30° C. The temperature rise of the PC board measured about 10° C. These are moderate temperature rises and should not degrade the life of these devices. Still, I drilled ten holes of ½″ diameter around the sides of the enclosure to improve cooling.

CONCLUSION

The bridge adapter is a versatile unit which you can use to interface to, or from, bridged amplifiers. Its huge input signal capability, large output drive, low noise, and low distortion make it suitable for many applications. ❖

An Experimental Throat Module

This author's third article on concrete horns features an experimental throat module that interfaces with his two previously built units.

By Robert Roggeveen

While I encourage readers to read my previous articles on concrete horns (*aX* 9/02 and 8/03)—the 30″ diameter mouth and the extensions—this article stands on its own merits as a research inquiry into the ratio of driver piston area to throat cross-sectional area for a cone driver. I built this module and attuned it to the physical attributes of the Siare 16 VR, namely the driver's cone area, phase plug, and the driver mounting frame diameter. Resonant frequency is 180Hz for this 5″ fiberglass cone driver that Bruce Edgar researched extensively in his midrange horn article (*Speaker Builder* 1/86).

I will show that by narrowing the throat opening to a 19:1 ratio, the useable bandwidth is still substantial with markedly improved acoustical output. This information may transfer to other stiff, high Bl factor midrange cone drivers with a phase plug.

DESIGN CONSIDERATIONS

Designing an axisymmetrical expansion limits reflection problems to the care and attention paid to a smooth graduating horn surface from the throat to the mouth, particularly when building a concrete module. You notice reflections by their effect on the horn's frequency response, causing dips, peaks, and decay of acoustical energy. A loudspeaker may produce incoherent phasing, which will cause an irregular frequency response, including cancellation of frequencies in the bandwidth.

Frequency cancellation will take place when wavelengths from the center of the diaphragm and the outer areas differ by one-half wavelength (Abraham Cohen, *Hi-Fi Loudspeakers*, first edition, 1959; a most educative book). A phase plug helps equalize the various path lengths from the diaphragm to the throat. In addition, horn drivers may show frequency cancellation due to driver-to-throat interface conditions, regardless of whether the driver has a phase plug or not.

Bruce Edgar showed that by enlarging the throat and introducing a space between the loudspeaker and the mounting face of the throat, you could overcome a peak in response with expanded bandwidth (*SB* 1/86, p. 14). However, by introducing a gap between the driver and the throat plate, air compression is depleted.

Considering the throat size, for large drivers a piston to throat area ratio of 2:1 is generally employed. Variance appears to be quite forgiving (Bruce Edgar, *SB* 6/90, p. 91). For midrange drivers, ratios from 3:1 to 4:1 are used. Josef Merhaut designed a mid- to high-frequency range horn-loaded electrostatic loudspeaker that has a 10:1 ratio (*Journal* of the Audio Engineering Society, November 1971).

The length and the shape of a horn play an important role in supporting the low end of the reproduced bandwidth. In general, the length of the horn will support a wavelength of twice the horn length (Cohen). The shape of the horn length from conical to cantanoid will influence the bandwidth and throat resistance (Daniel J Plach, *Journal* of the Audio Engineering Society, October 1953).

The mouth size is a further determining factor of the lower cutoff frequency. For a circular horn, the mouth diameter must be at least one-third the wavelength of the sound to be transmitted from the horn.

Or stated another way, the dimension that circumscribes the outside of

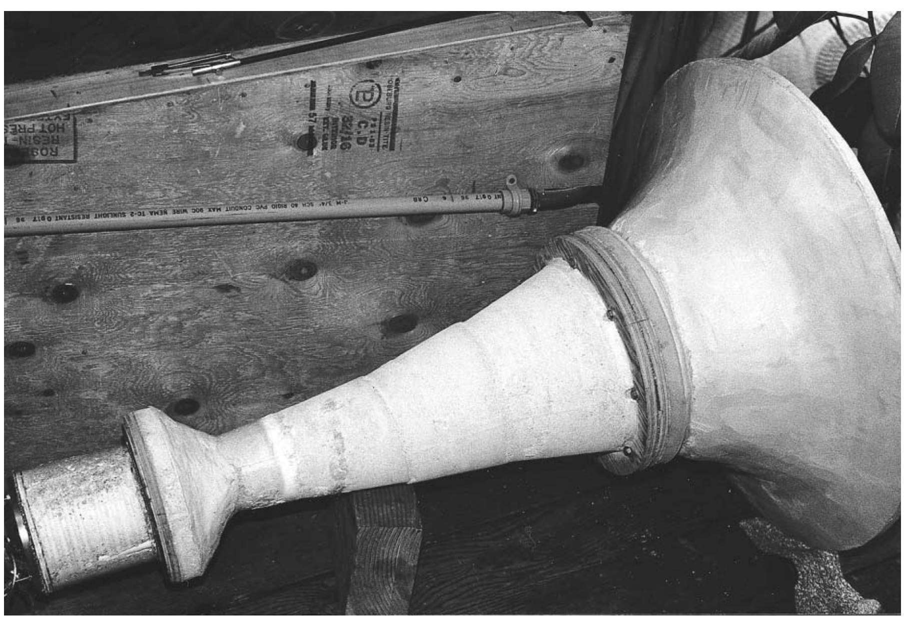
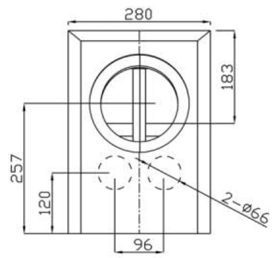


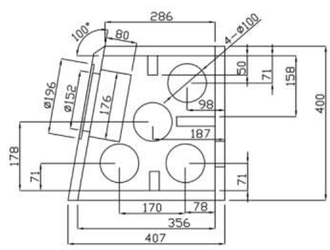
PHOTO 1: The completed horn with extension.

Swans Tempus Kit

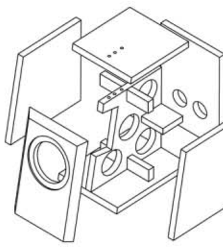
Swans proudly presents their first European-style, independently developed loudspeaker kit. Due to exceptionally high standards for a kit speaker, TEMPUS is a top-level, world-class performer in all regards.
German sound magazine KLANG+TON featured the Tempus in the June 2002 issue, commending its excellent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



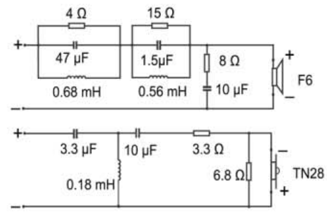
Front View



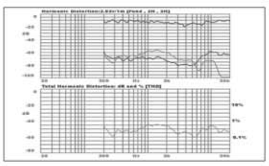
Section Side View



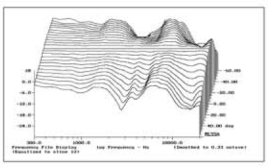
Explosion View



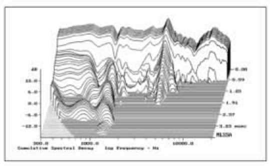
Filter Circuit



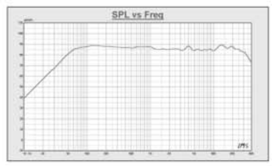
Tempus Harmonic Distortion



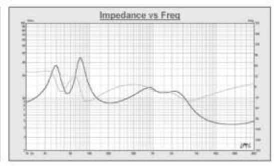
Tempus Horizontal response +60deg ~ - 60deg



Tempus Cumulative Spectral Decay



Tempus Frequency Response



Tempus Impedance

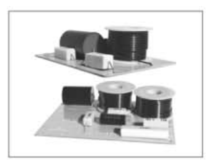
The Tempus project was initially conceived as a private work of the acoustic arts, to be executed independent of all professional affiliations. Despite a lack of traditional marketing and initial editorial commentary, the design has since received wide acclaim in the independent press and has gone on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

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

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

The new Swans Tempus set includes:

- 2 x 1.1"/28mm Swans TN28 soft-dome tweeters with open-air housing, Neodymium motor, ferrofluid cooling, decompression chamber, integrated grille and German-made silk dome;
- 2 x 6.5"/165mm Swans F-6 shielded high efficiency midbass drivers, with Kevlar/paper curvilinear cone, phasing plug, and alloy basket;
- 2 x completed crossover networks with premium-grade components;
- Two sets heavy input terminals;
- Two pairs internal cable sets;
- Complete manufacturer's instructions.



All Tempus kits are thoroughly pre-tested by Swans and come in matched acoustical sets.

Price:$599 with Tempus parts kit only;
$999 with Tempus kit and cabinets;
$1299 assembled Tempus.

hear different.

"Tempus"

SPECIFICATIONS	
Frequency Response:	48Hz-20kHz
Sensitivity (2.83V/m):	86dB
Nominal Impedance:	8 ohms
Power Handling:	10-120W
Dimensions(HxWxD):	406x286x408 mm

for more info please
Call: (323)-881-0606
SWANS SPEAKER SYSTEMS, INC
2550 Corporate Place, Suite C103
Monterey Park CA 91754
www.swanspeaker.com
www.theaudioinsider.com

the horn must be at least equal to one wavelength of the sound to be transmitted (Cohen). This is the point, though, where you don't hear that wavelength! The frequency where the horn will be heard at –3dB is at the cross-sectional area where the mouth size is reduced 25% (Cohen).

TAKING SHAPE

I did not want to produce an air gap to remedy the phase problems and forego the principle of a compression horn. Nor did I wish to increase the throat size, because this will shorten the horn, effectively raising the lowest wavelength the horn will support. If at all I want to lengthen the horn to support lower frequencies. I studied the cross-sectional drawing of a compression driver (Cohen).

The high-frequency driver has a phase plug causing what looks like a ring-shaped wave front launch pattern—such as that of an EV TW35 high-frequency horn—around the base of the phase plug. In comparison, the bottom of the phase plug of a compression driver is concave and follows the shallow shape of the diaphragm. The sound is expelled from the edge or parameter of the diaphragm (Cohen).

In relation to the Siare, what would the bandwidth be if I narrowed the throat to such a launch pattern? What would it do to the frequency response and acoustical energy? That means I'll work with the phase plug of the Siare. The ring-shaped wave front launch pattern will be just outside of the phase plug at the base. The throat is as near as reasonable to that launch plane to keep cancellation of frequencies to a minimum.

This design will work with the phase plug as an initial horn loading of the sound waves in an acute expansion; that of the axial cross-section of the phase plug set against the tubular edge of a throat ring. After that, the sound wave will propagate along a conical expansion to a 2.12″ diameter cross-section, conveniently chosen to mate with another horn unit I built (*Fig. 1*). The pitch of the conical expansion follows the pitch the 30″ diameter mouth tractrix curve has at the 2.12″ diameter connection point. The length from 2.12″ to 1.5″ at this pitch is about 5.5″. The tracery of the tractrix curve is such that it behaves much like an exponential expansion at the initial length of the horn near the throat (Edgar, *SB* 2/81, p. 10).

This design then has a compromise: the throat expands conically for the first 7″ or so. Let's see what happens! I felt safe with this conical horn shape so close to the throat since its length was only 12% of the total length of the horn.

THROAT RING

Experientially I arrived at the following: the throat ring is 1.5″ in diameter, which corresponds to an area of 1.767 in^2. The 1″ diameter thickness of phase plug at the base translates into a 0.785 in^2 area. Since the phase plug fits in the ring, subtracting one from the other results in a cross-sectional area of 0.982 in^2. That is a throat shape ring opening of a ¼″ width, with an outer dimension of 1½″, and an inner diameter of 1″!

The driver piston area has a 5″ diameter, which is 19.634 in^2. Subtracting the 1″ diameter phase plug leaves 18.849 in^2 of piston area. That is a ratio of 19.2:1. With some error in the actual cone piston movement, it is fair to say that the ratio is 19:1.

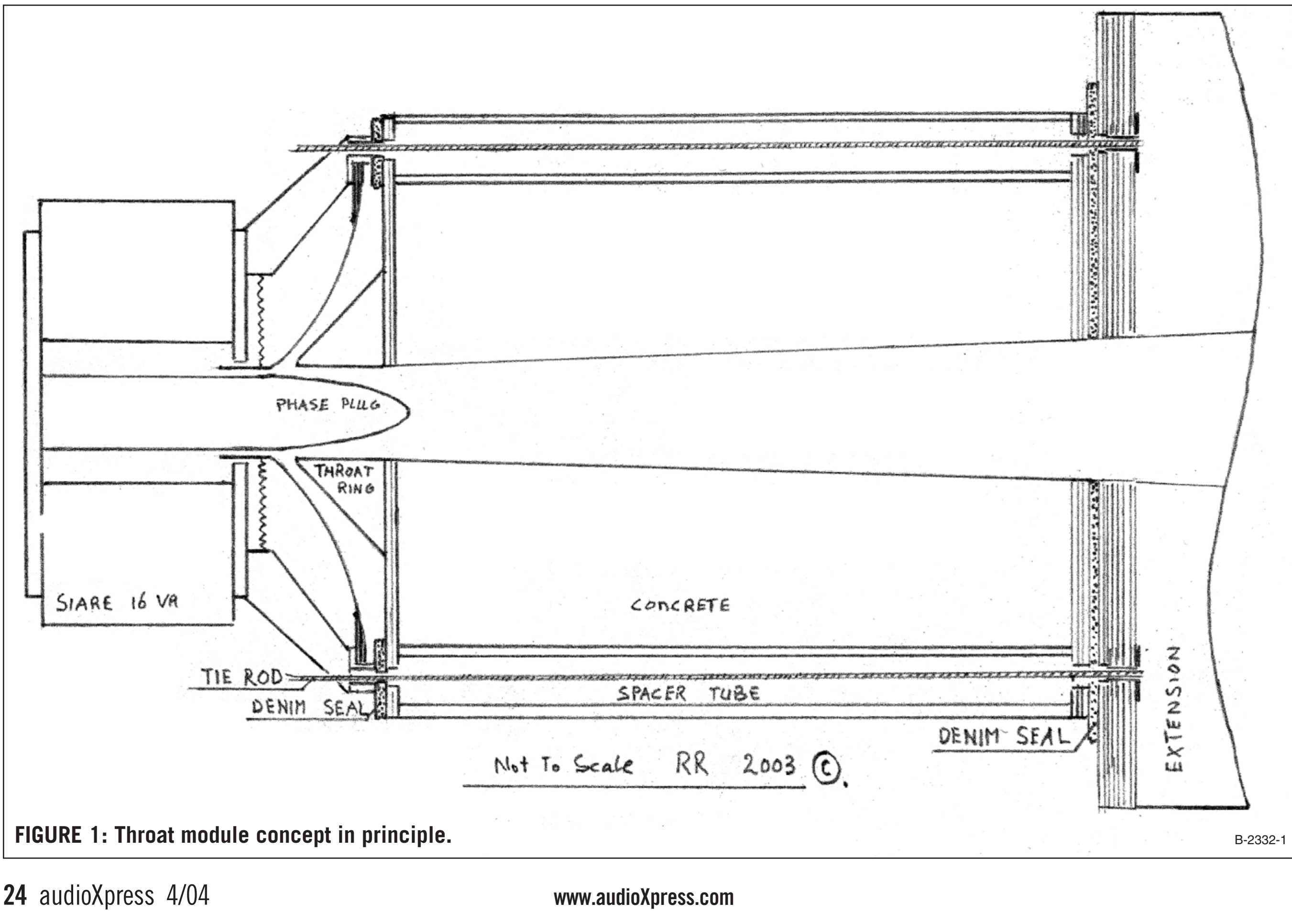


FIGURE 1: Throat module concept in principle.

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With this article, I describe the making of those first several inches of concrete horn expansion. It is connected to the horn extension and the large 30″ tractrix mouth with gaskets made from cotton denim to form a horn made up of three parts (*Photo 1*). As with other projects, such as Bruce Edgar's midrange, I have not utilized a driver back volume.

TIN CAN MOLD

This module is made of concrete, steel, PVC tubing, and wood, and poured in one action. The diameter of the module must accommodate the dimensions of the driver mounting face configuration on one end and the extension on the other. I decided on a diameter of 6½″.

It's easiest to make a tubular mold out of a tin can, and the 6½″ diameter corresponds nicely with an empty Planters peanut can I had on hand. The height of the can is 6½″. You may use a can of similar size or larger if need be, but not less than the driver's mounting dimensions. This will also be the diameter of two plywood end disks used in this project.

Use a can opener to remove both the top and the bottom of the empty peanut can. A tube is now apparent. You will be cutting this tube open lengthwise and on axis with sheet metal sheers. Before cutting, though, draw with a felt-tip marker a 2 × 4″ oval midway on the tube and orient it lengthwise on-axis. This will be the mortar application hole. Now cut the tube lengthwise through the long axis or the oval. Then cut out the oval shape.

PLYWOOD DISKS

I use the best grade of plywood I can buy and thus reduce the risk of voids,

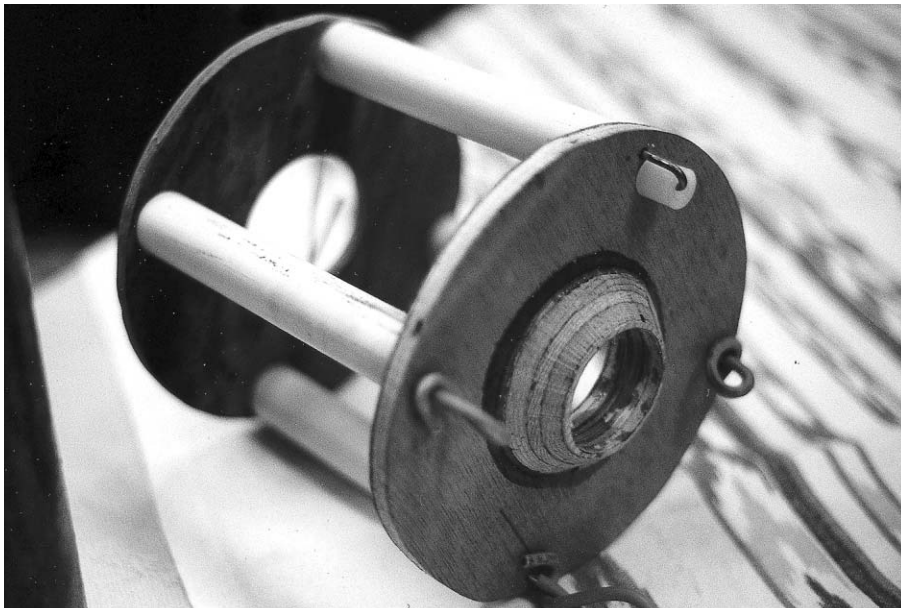

PHOTO 2: Close-up of wooden ring you must fashion.

which might cause air leaks. Or, you may fill the voids in cheaper grades. Using a protractor, draw two circles with a 6½″ diameter. Mark the centers with an ink pen, and then cut these disks with a saber saw.

Center the two disks and the Siare over one another and mark the mounting holes of the driver on the wood. Then drill these four holes, slightly oversized. Drill a pilot hole in the center of the disks. Use a keyhole cutter or similar large-diameter round saw to cut out the center holes individually. One disk has a 1½″ hole, the other a 2.12″ hole.

Place the two plywood disks at each end of the metal can, holding them loosely in place with large hose clamps. You may choose clothes dryer air hose clamps. Or you can easily make one out of two or three automotive hose clamps, which is what I did.

MOUNTING PLATE

The cavity produced when the driver is mounted flush with the driver mounting plate is significant. To counter phasing problems noted earlier, I chose to shape the space such that the exhaust from the Siare is expelled along an inner ring just outside and along the base of the phase plug, right over the voice coil gap. The cone moves back and forth, so allow some leeway if you fill that space with a solid, so that the sound waves are forced to the ring area at the base of the cone (*Fig. 1*).

In comparison, the bottom of the phase plug of a compression driver is concave and follows the shallow dome shape of the diaphragm where the sound is expelled from the edge or parameter of the diaphragm (Cohen). At that area, the Siare's cone is nearly 1″ inside the mounting plate plane, where the surrounding and mounting frame is located.

TUBULAR RING

Using ⅝″ plywood, you need to make a ring with a steep 45° level on the outside circumference, and a 1½″ hole at the center (*Fig. 1*). The apex is at most one saw blade thick so that a cross-section of the ring looks like two triangles with their 90° corners at the inside corners spaced 1½″ apart (*Photo 2*).

Making this is rather tricky. Draw a 1.75″ diameter circle on ⅝″ thick plywood and mark with ink the centerpoint of the protractor. I next used a saber saw, set at 45°, and cut the circle, with the 45° facing outward, which is where any error may be forgiven. Then, with a 1½″ diameter hole saw, cut out the center, holding onto the wood with wooden thongs.

Glue the base of this ring on the outside of the driver mounting plate, centering both 1½″ holes. I used epoxy for strength. The outside is the side where the driver will be mounted so that the ring taper will face inward into the driver plane space.

CONICAL THROAT CONE

Make a cone out of a craft paper disk section of 30″ diameter stock, of a dimension used as the last cone for the extension (*aX* 8/03), and measure at what cone length the cross-section decreases from 2⅛ to 1½″. This corresponds to roughly 5½″ in cone length. I will save you the trouble of figuring all this out, but you still need to work with paper that has a 15″ radius. This is easily done, using a strip of plywood 17″ long. Drill a hole 1″ from one end to fit a nail, and do the same at the other to fit a pencil.

Place this arrangement on the workbench and use sturdy paper to draw the arc on the long side of the paper. A piece of letter-size paper will do, although I'd double up on the thickness for strength. Also, draw the radial lines on the paper near the overlap for the glue strip and to keep the pitch accurate as you adjust the cone relative to the disks. Once drawn, cut it out and use paper clips to fit it to the two plywood disks that you made earlier. Fit these disks in the can at 5½″ as measured from the inside of the plywood disks. Considering the peanut can, the disks are at the ends of the can, abutting the folded over crimped tin edge.

Then mark the cone for gluing with a ¼″ glue strip, again along radial lines. I used Super Glue adhesive. Make the cone long enough so that it will protrude outwards from both ends of the disks by ¼″ or so (*Photo 3*). Epoxy the cone to prepare it for the mortar application. Let it cure. For much more in-depth description on how to make cones and the tractrix curve, read my previous two articles (*aX* 9/02, 8/03).

TUBE SPACERS

Place the two plywood disks over the epoxied and cured paper cone. The cone will be longer than needed, for stability. When fit snugly, and parallel to each other, the two disks will be about 5½″ apart, as measured along the axis.

Measure the distance at the edge. It should be the same. If not, your ply-

PHOTO 3: Cone inserted in disk.

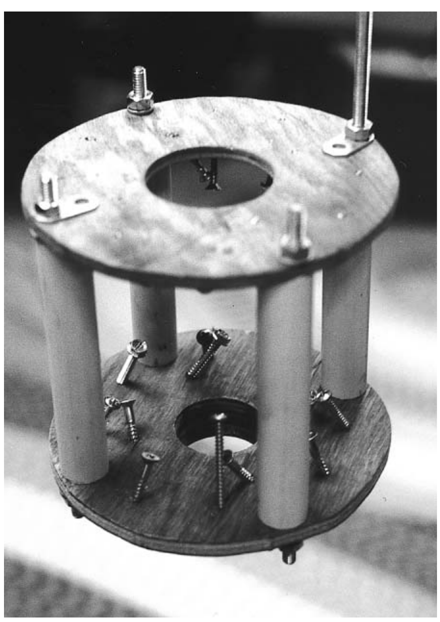

PHOTO 4: Throat module with anchoring screws.

wood piece is warped, and I wouldn't use it because the gaskets are then more prone to air leaks. In other words, the mounting surfaces for the module must be flat to optimize the benefits of both compression and horn propagation principles.

Cut four lengths of 5½″ of hot water tubes or PVC as square as possible; they will interface with the wooden disks so a good fit is necessary. I used ½″ hot water tubes. Sand the ends smooth. You will embed these tube spacers in the concrete to make a void for the Siare and module mounting tie rods to move through.

TIE RODS

The rods are used for the mold and pour process. Use ¼″ diameter all thread stock and cut four 7″ long sections. With nuts and washers on each end, tie the two plywood disks together spaced apart by the length of the four tubes, which should correspond to the length of the conical horn expansion. Fit the cone into the statically held disks and check for movement. It is okay to be a bit robust with the cone to make it fit as long as it keeps its conical shape.

If there is still movement, you may adjust the tubes accordingly, but only slightly, as with a little sanding. If more tube length is needed, cut new tubes. If the size and pitch of the rod stock is the same as the T-nuts embedded in the extension section, you can use these rods when connecting the module to the extension.

I used fine thread ¼″ T-nuts, so I separately obtained mounting rods. I further wanted the mold concept again, which includes the four-spacer tie rods, so I did not mind dedicating these.

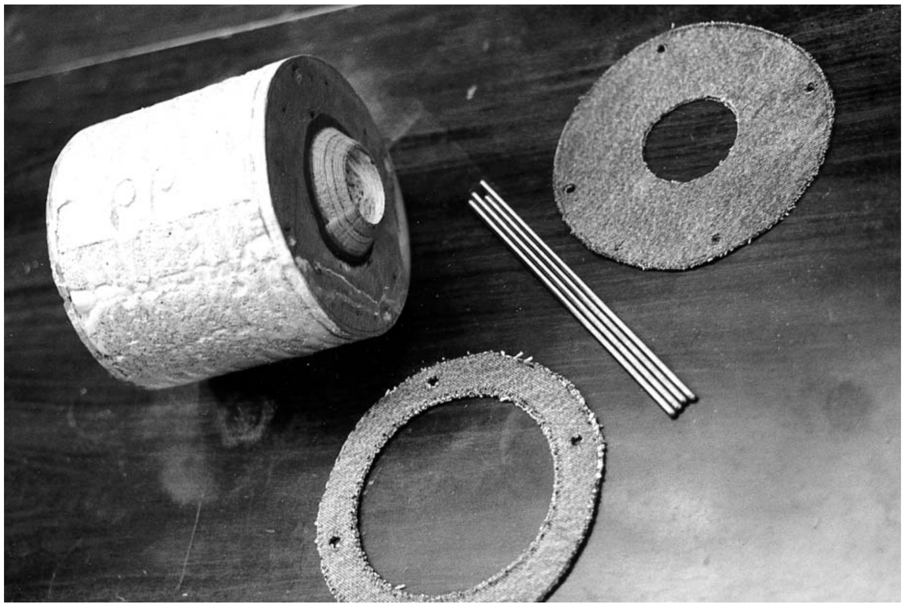

PHOTO 5: Denim seals to be used between the expansion and module and between the driver and module.

ANCHORING SCREWS

Anchoring screws are inserted at 45° angles into both disks on the inside, and placed randomly, but not in the way of the paper cone or the four tube spacers or the tin mold enclosure (*Photo 4*). The 45° angle allows for better purchase of screw, wood, and concrete. So take the unit apart, pre-drill holes, and screw in various wood screws. Then put the module back together.

WRAPPING THE VOIDS: (CONE AND SPACERS)

Wrap the metal can around the disk and tube ensemble and draw the clamps tight. How does it look? Are the disks parallel? Fit the epoxied cone in the disks. Does the cone fit snugly? Place the tube spacers so that the outer edges of the four-spacer pipes are touching the tin can.

This is the last check and dry run before you encase everything. Do you like what you see? Is the unit twisted? Turn it correctly so that the spacer tubes are perpendicular to the wooden disks. You can still make adjustments. Make peace with your work and diligence.

It occurred to me that the wooden disks are rather thin. I had theorized that because of the high level of acoustical stress, the harder medium is preferred. As a result, I kept the wood to a minimum. I used some baling wire to tie together the heads of the anchor screws, effectively linking both disks. This reinforcement is embedded in the mortar and is meant to provide tensile strength.

POURING THE THROAT MODULE

Set up your tools: a tray to mix the mortar, a spatula or serving spoon, a tamper with a ⅜″ to ½″ head with a 1′ long stem, water, and screwdriver.

After smearing bonding paste on both disks where the screws are, tie down the clamps over the tin tube ends crimping onto the wooden disks. Is the pour opening large enough? The 2 × 4″ oval worked fine, but a larger opening is also okay. Make it to suit your hands and space needs, but keep the opening lengthwise and at the apex as it is laid on the side.

Make at least two dry runs of the process from assembly up to actual mortar application. It helps with thought sequencing and raising confidence in successful completion of the module.

Using a good grade mortar, pour enough in the tray so you won't run out, mixing in water in small increments to arrive at a moderate, firm, plastic stage of concrete. Use a serving spoon to toss the mortar into the oval hole. You can toss only small portions because of the size of the oval, but that is okay. Add two or three spoonfuls at a time.

The mortar will likely fall on the cone. Use the tamper to shove the mortar around the paper cone, dropping the mix to the bottom of the mold. With the tamper, work the mortar around the screws on the disks and the PVC spacers.

Add more mud. Work at a steady, unhurried pace, but not too slowly, in case the consistency of the mortar changes. It is important not to pound directly onto the cone with the tamper. The cone is strong and will do fine for this job but won't stand repeated poking with a steel rod. You wouldn't detect damage until you unpacked the mold and found an indentation protruding into the now cured horn expansion! A sorry finish.

Continue to throw three to four shovelfuls into the mold at a time and tamp and prod carefully. Tamp up and down the length of the mold, firmly yet gently. Work around the screws, rods, and spacer tubes. Fill up the tin mold to the rim of the oval. Mark that space with a code, a number, letters of your choice, or maybe a logo. It's your module!

Clean your tools and let the concrete cure. You are finished with this task.

MAKING SEALS

You need to make two seals to mount the module. One seal between the expansion and module, the other between the driver and the module. I used the cotton denim material from a pair of jeans. Use pieces that have no seams.

Cut five 6¾″ circles. The driver seal is made of three layers, while the other seal has two layers. One seal has a hole in the center that corresponds to the 2.12″ diameter end of the module, the other to the dimensions of the mounting face of the Siare. I oversized the width of this ring by a half inch (*Photo 5*).

I used two layers between the module and expansion because the surfaces were pretty flat in relation to one another. I used three layers between the driver and the mounting plate because I needed compensation for the slight wobbly planar relation. This requires some sewing skills, threading the edge together, the center hole edge, and the four islets. Make a snug-fitting layering job.

Use a sturdy sewing needle and an 18″ section of coarse strong thread. Make a clumping knot at one end and thread the other through the eye of the needle. Start by placing a few sewing pins around the surface to keep the layers from moving relative to one another. Then make a loop threading pattern sticking the needle ⅛ to ¼″ from the edge through the denim, bringing the thread to the back, around the parameter over to the surface and sticking the next threading ⅛ to ¼″ along the parameter of the seal. Pull the length of thread through to the knot.

Once you have made a couple of loops, pull them tight, but not so much that it deforms the denim disk shape. This takes a little practice, but by the time you've gone around the whole parameter, you will be ready for more tight spaces. Sew the inner circle and eyelets in a similar manner. Work at your own pace and try not to become discouraged.

After sewing the edge together, place the seal over the module centering its 2.12″ hole with the denim's. Place a spare piece of plywood slightly larger than the 6½″ diameter module over the denim and hold the seal in place as you turn the module over and rest the plywood on a table.

Place a tie rod in a mounting hole. It will come to rest. Wiggle it a bit from side-to-side so it falls past the wooden disk onto the denim. Tap the rod a couple times with a hammer. Repeat for all four mounting holes. Lift the module off the denim, revealing four indents in the denim. Use an ink pen to mark those indentations and a grommet puncher to take out the marked areas. I used this tool to make the seal for the interface between the 30″ mouth and the extension, but I found it too big for the module tie rods, so I used a single paper hole punch. This is not as sharp, but with repeated squeezing, I made the holes.

This seal is meant to be an air seal. It will do fine once compressed when all units are connected.

BREAKING THE MOLD

The mortar is dry. You can now loosen the two clamps to remove the tin can. This exposes the exterior of the module. Next, pull and tear out the epoxied paper cone, exposing the conical horn propagation. Let the module dry some more, a day or two.

ADJUSTING THE MOUNTING HOLES

Since the tube spacers leave room for adjustment at the wooden disk end, drill or enlarge the holes as needed. Find out by inserting the mounting rods into the T-nuts embedded in the extension-mounting plate. Lower the module over these rods and ascertain how the module mounting holes need to be adjusted, both at the module to expansion interface and the driver mounting plate.

Sealing the module is important to keep the concrete from shedding dust. Using the same material as the bonding agent, I sealed the wood as well as the module.

CENTERING THE HORN THROAT

Centering the module onto the expansion can be elusive. I used a light bulb inside the horn to look down the throat for shade lines and balance of circumference and made pencil marks as reminders.

For the driver-to-module alignment, I laid the driver down on its magnet with the mounting rods through the holes of both driver and module and looked from the expanded end into the throat, aligned it, and made marks to remind me later. This may mean rotating one unit relative to the other to get the best fit. This takes time, and because you work with the driver, be very careful. Also watch out for dirt particles, especially when the driver is laying on its back.

Once you have made all the alignment marks, insert the mounting rods into the T-nuts, fit the cotton seal, lower the module, and set it. Fit the next cotton seal and place the driver, setting it according to the marks you made. Hand-tighten the nut and washer first. Then, use an open wrench, making a quarter turn at a time going around the circumference of the driver.

How tight do you make it? Not that tight, but enough to compress the cotton seals slightly, and so the module does not sag when placed horizontally. Use common sense. If you over-tighten, the T-nuts in expansion are pulled up into the wood, leaving room behind it, unnecessarily causing play in the T-nut to wood fitting that you can't access for repairs.

LISTENING

This module, connected to the extension and mouth sections, raises the acoustic energy level as compared to other concrete tractrix horns I built, using the same driver. The difference is the length of the horn, the throat size, and, I believe, the methodology used incorporating the phase plug as described. The result of dB level testing across the bandwidth was published earlier (*aX* 8/03).

The bandwidth is lowered to about 200Hz and useful to about 5kHz. High frequency suffers from the relative rough surface area of the concrete. There is a peak at 1kHz, which I have not been able to tame, so I need to examine this further.

I listened to it with a replica of a Jensen Imperial folded bass horn I built from drawings found in *Speaker Enclosures* (p. 98, Alex Badmaieff and Don Davis, 1968; a very informative book). The bass horn has a Jensen P15LL 15″ driver. For the high end, I used the Electro-Voice TW35 horn tweeter, with 6dB passive crossovers at 200Hz and 5kHz.

The midrange is very pronounced, maybe too much for some listeners. The voice is very well articulated with detail and punctuation, and the voice harmony and chorus are wonderful. The piano in combos is clear and intimate. Large orchestra and jazz band music is richly textured.

I have listened to this in mono setting because of practical reasons. I have only one horn as described in this article, and I did not wish to be distracted by problems of matching and spacing. I believe the midrange is what our ears are most sensitive to, and this horn delivers.

The finished horn rests on the floor. It faces into our half sphere geodesic dome living room, which has no divisions and is 36′ in diameter and 19′ high at the center. So I can take distance from the horn speaker arrangement. It weighs 120 lbs, with the driver, and looks like a gray whale, which not everyone will want in their living room. However, for those of you who persevere and build your own, the sound from a horn system is unequaled. It could be enclosed, but that is not where my energies are.

Just a quick glance at the marketplace shows that a French-built horn system goes for multiple tens of thousands of dollars (*Stereophile* and *Audio* 9/03). I hope that my research can add to the body of information, and encourage people to experiment with horns.

I thank the editors for the opportunity to tell my story.

Happy horn building and listening. ❖

Borbely-Clow Super Buffer

This preamp project gives you many options for recording various formats. **By Sherm Clow**

Over the years (well, by now, over the decades), I've accumulated many recorded formats and an equal number of playback devices. These days everything ends up on CDR, and I have an excellent burner, the HHb 850. As a result, I'm something of a magnet for anyone with an older format recording wanting to get his/her musical history onto the current CD format.

This all came into solid focus during a CD production of the Smoke Blues Band, a local group from the 70s. This nostalgia project required collecting material from cassette, 2-track master reel-to-reel, standard 4-track reel-to-reel, and even an old reel from a 1&7/8 machine. I spent a lot of time swapping cables back and forth in a system that did not have a sufficient preamp to accommodate all the devices.

IN SEARCH OF THE HOLY PREAMP GRAIL

Most studios solve this cable swap problem with a patch panel. However, I had acquired an old Macintosh 2205 amplifier that played great (love those output transformers!). So I needed some sort of buffer for the digital devices playing through the amp as well as a way to connect a lot of playback and recording machines together.

I first examined several op-amp designs for a preamp. There were good ideas there, but an *audioXpress* article (5/02) by Erno Borbely caught my eye. As he described his development of an all-FET line amp, he mentioned an intermediate step that produced what he called a "Super Buffer." The topology looked straightforward, reasonably simple, and compact, and he offered a kit of parts (EB-602/403) on his Web page (www.borbely.com). So I ordered the kit and received it within a week, shipped from Germany.

SPARE PARTS, SURPLUS FINDS, AND PROJECTS NEVER COMPLETED

Now that I had decided on the active part of the preamp, I needed switches pots, power supplies, cabling, and a few whiz-bang features. I acquired a couple of 8-position, two-pole Grayhill switches

PHOTO 1: Front view of the Borbely-Clow Superbuffer 2003.

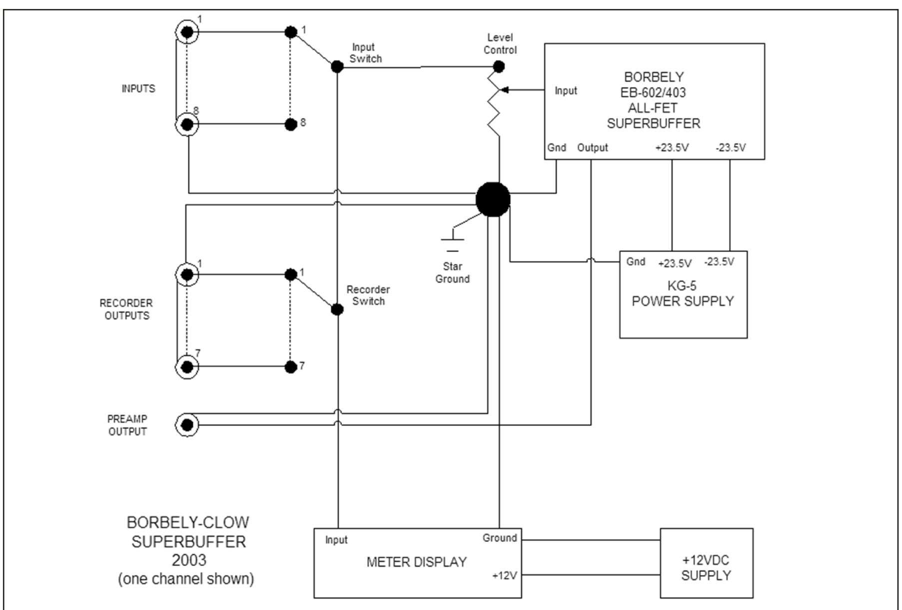

FIGURE 1: Block diagram (one channel shown).

ABOUT THE AUTHOR
Sherm Clow is a Senior Telecommunications Analyst and a subscriber to Audio Amateur publications all the way back to (just about) the beginning. He is currently engaged in a desperate attempt to record every note of the vibrant underground jazz scene in Salt Lake City.

from a local surplus store that also provided 4-pair phono panels originally designed for circuit board mount but also with screw connections for panel mount.

I already possessed an Alps volume control that I had ordered long ago from Old Colony. A bunch of raw Mogami audio cable was available for signal connections. I examined a power supply, also purchased years ago from Old Colony, designed by Gary Galo and Walt Jung in *Audio Amateur* (4/90). The only component I needed to purchase was a rack-mount chassis, which I obtained from Sescom.

For a whiz-bang feature, I decided to adapt a Velleman LED VU meter kit to the preamp. My HHb 850 has LED meters active during playback. I've noticed that most modern CDs have very little dynamic range; some rock CDs show as little as 3dB for most of the songs! No wonder people complain about CD sound.

Once in the digital domain, engineers can do anything to the music, and I'm actually amazed that some of these super-compressed productions sound as well as they do. Anyway, I wanted some way to monitor the dynamic range of recordings with the preamp, thus the addition of the Velleman VU meter.

LITTLE PLASTIC BAGS AND BEEFY TOROIDS

The Borbely super buffer kit arrived containing a zillion little plastic bags, one for each component value or part number! Well, it did speed assembly. Everything went together smoothly, although there were a couple differences from the original article. However, the supplied kit documentation included an updated schematic as well as a setup procedure for the completed kit. My only complaint was the lack of 6dB gain resistors with the kit. Thankfully, I had some available.

The power supply was designed for ±15V DC but I needed ±24V DC for the super buffer. Borbely supplies an excellent, but somewhat exotic, power supply for the super buffer; however, it is quite expensive, and I had the Galo/ Jung supply waiting for an application. The Avel-Lindberg toroidal transformer was hefty enough to supply sufficient rectified voltage so I could modify the regulators for a higher output to the preamp. With a little resistor swapping, I was able to get about ±23.5V DC for the super buffer.

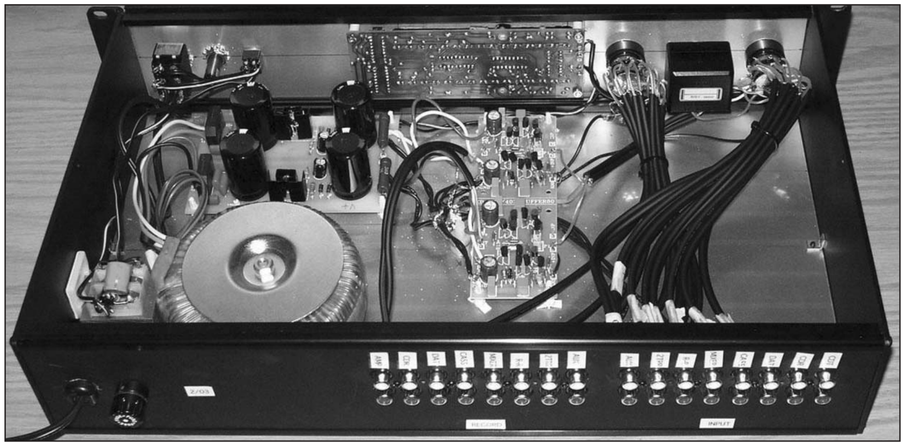

PHOTO 2: Back view of the unit.

WIRING MADNESS

I guess I could have saved some interior space and some hassle using simple hookup wire, but I had all this cool Mogami around, so I used it for all the input and output connections. The feed to the output switch was taken before the volume control. I used a switch for the outputs to avoid several simultaneously connected recorders loading the signal. One position on the switch was left open (null) for those pure listening occasions.

The grounds from the input/output connector panels, power supply, and super buffer board were taken to a single star ground. I also stripped a wall wart for the VU meter 12V DC supply (switched, for those times when I'm tired of flashing lights).

The Alps volume control is 100k. Borbely recommended 20k, but the larger value doesn't seem to cause any problems.

HEAVY METAL HACKING

The most difficult part of this project was the drilling and cutting for the connector panels and the VU meter display. But with a little patience and fortitude (and powerful drills, saws, and large files), the operation was completed. Everything ended up mounting rather nicely. The chassis itself was a bit of a puzzle; instructions were not supplied. But I did finally figure out the system. One complaint was the insufficient screws supplied by Sescom—one more bag, please!

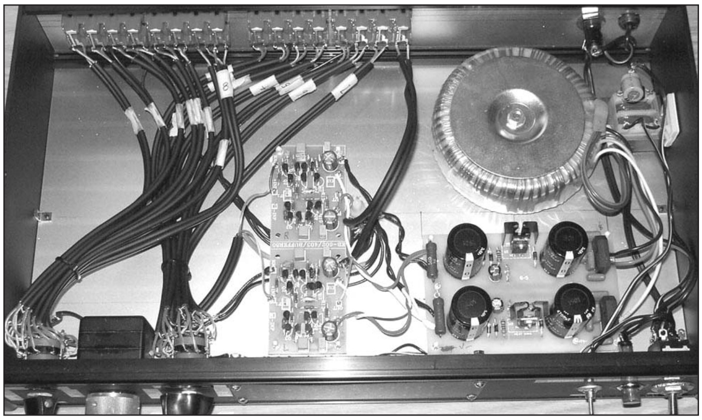

PHOTO 3: Overhead view of the device.

I used a Brother P-Touch printer and computer software for labeling. I used white-on-clear tape, which provided a decent look to the front panel.

LISTENING TO THE ANGELS SING

Of course, the best part of this type of project is listening to the final product. And this super buffer does sound gorgeous! Very detailed, particularly on those CDs produced with sound quality in mind (Audioquest, Mapleshade, and so forth). I do a lot of live recording. This preamp gives me the chance to delve deeper into my recordings, listening for those things that tell me what I did right and what I need to improve.

And, as I hook up more equipment to the preamp, there is less hassle swapping wires around. The switches feel solid and have (nearly!) enough positions to accommodate the equipment I wish to use.

GILDING THE LILY

There's always room for improvement. Purists will demand a higher quality volume control and better switches. The I/O panels could be improved (at a much greater cost!). However, this preamp is already indicating deficiencies in other parts of my system, so I don't need to improve parts quality at this time.

One feature I neglected in the original design is some sort of monitor switch so I can compare the signal source and the output of the recording equipment. This switch is at the top of the list of future mods and will involve the input of the CDR, the most used recording device.

THANKS AND CREDITS

A big thanks to Erno Borbely for his JFET designs and kits. And thanks to Old Colony for supplying significant kits and parts. I just wish they would bring back some of the old favorites (at least a circuit board for the Galo/Jung power supply). Also thanks to Raelco, a parts hound's heaven. ❖

The Tuba 24 Horn

PHOTO 1: The Tuba 24 horn-loaded sub.

Rules are made to be broken in the design of this horn-loaded subwoofer. **By Bill Fitzmaurice**

One of the decisions that I needed to make when I started to prototype our speaker line was whether to use separate subwoofers. I've personally shied away from them, because I prefer to lug as little gear as possible from home to gig and back again. But having separate subs and mid-bass cabinets has become accepted practice in pro-sound, so I agreed to go that route.

The subs would need to be horn-loaded to keep their SPL ratings close to those of the DR horns that they'd be mated with, but I didn't wish the subs to be much larger—if at all—than the respective DR mid-bass cabs that they'd be augmenting. Since the DR250a prototype measured about 22 in^3, I decided that 24 in^3 would be the goal for its matching sub. Thus the moniker Tuba 24 (*Photo 1*).

FEATURES

I took the basic design cue and name from the familiar brass instrument, since the cabinet closely resembles a tuba (or perhaps more accurately a sousaphone) with the horn wrapping around the "player," in this case a woofer. As in a tuba, a nearly conical taper that flares quite rapidly at the terminus allows the longest pathway and largest mouth within the least amount of space.

The driver is mounted on an axially configured baffle that bisects the driver chamber into a sealed rear chamber and a larger than usual throat chamber, which serves as a low-pass filter just before the horn throat. While it's wise to minimize the throat chamber volume in a wideband horn for maximum high-frequency loading, the opposite tack enhances the low end in this application.

The requirements for pro-sound subs are quite different from those of hi-fi and home theater, because pro-sound doesn't have a lot happening in the lowermost register. To illustrate this, take a look at *Fig. 1*, a real time analysis (RTA) readout. Taken at a live "ZZ Top" concert from the FOH (Front of House, where the mixing console resides) position, with the overall level at 125′ from the stage a bit over 105dB, I can testify that there was more than enough low-frequency impact to satisfy any bass freak. However, the highest individual ⅓-octave band SPL levels, at 100dB, were in the 60–80Hz range.

At 50Hz the level had dropped by 9dB, and at 40Hz it was down about 25dB. The speaker complement for this concert included two clusters of eight dual 18″ woofer subs to either side of the stage, so these results were not colored by the lack of low-frequency capability from the sound system. Further, I took RTAs all summer long at over 20 concerts, and these results were consistent from act to act. While hi-fi and home theater aficionados may need subs that are flat to 16Hz, pro-sound just doesn't have that requirement.

RULES OF HORN DESIGN

Having decided upon the basic geometry, the size of the box, and its performance characteristics, I began the design process. I wasn't ten minutes into the job when I discovered that what I wanted to accomplish would not be allowed if I followed all of the accepted rules of horn design. Not being a fan of rules in general, I did what I always had in the past when told "you can't do that—it's against the rules." I did what I set out to do, making up my own rules as I went along. Here's how the journey went.

1. First choose your driver, and then design your horn around it.

Because this sub was going to be marketed for club usage and not pro touring sound, I figured that a reasonable goal would be to achieve maximum SPL to perhaps 80Hz, with effective horn-loading (higher than direct-radiator efficiency) to 50Hz and usable response (–10dB from average without boundary reinforcement effects) to 32Hz. That would usually mean using a 12 or 15″ driver for the desired Fs and cone area, with a horn length of no less than 6′. I soon found that obtaining such a pathway within the given cabinet size limits could only be done if the driver chamber was kept quite small, so small that even a twelve wouldn't fit.

ABOUT THE AUTHOR

Bill Fitzmaurice (BA, University of New Hampshire) has been a professional musician since 1966 and has been constructing instruments, amplifiers, and speakers for just as long. Vice president of DeltaSounds Loudspeakers Inc., Fitzmaurice is the author of "Speaker Builders Loudspeakers for Musicians" and over 30 magazine articles dealing with speakers and electric instruments, as well as the action-adventure novel "Operation: Sergeant York." Bill and his wife of 30 years reside in Laconia, NH.

A ten would work, but if conventional wisdom was correct, I had a problem: the lack of pro-sound horn-loaded subs on the market that use tens. In fact, the most popular driver size for the application is 18″, although its usage usually results in horns shorter than what I consider optimal. I decided that the benefits of a longer horn would outweigh those of a bigger driver, and I was not going to make the box any larger, so a ten it would be. So much for rule number one.

After laying out the design on paper, I used a computer-modeling program to predict its performance. The program, Hornresp™ (Freeware by David McBean, http://www.users.bigpond.com/dmcbean/), confirmed my contention that a longer horn would be of more benefit than a larger driver. [I have noticed in using this program that altering driver specs has little effect on the predicted performance, so I'm not so sure that part of the program is as accurate as it might be, but for the most part it does predict within a reasonable margin of error what a horn will do at the lower end of the frequency spectrum, as you can see in SPL chart 1 (*Fig. 2*).]

I did my initial testing using a driver I had on hand, a Carvin™ PS-10 (an OEM from Eminence™, similar to their Beta-10). The measured specs of my sample were Fs 55Hz, Qts 0.5. While the Fs is within accepted traditional design parameters, the Qts is almost double the normal specs considered acceptable for horn loading. But I'd used this same driver previously in horns with good results, so rather than follow the "rules," I once again deviated from the norm. The main reason why has to do with rule number two.

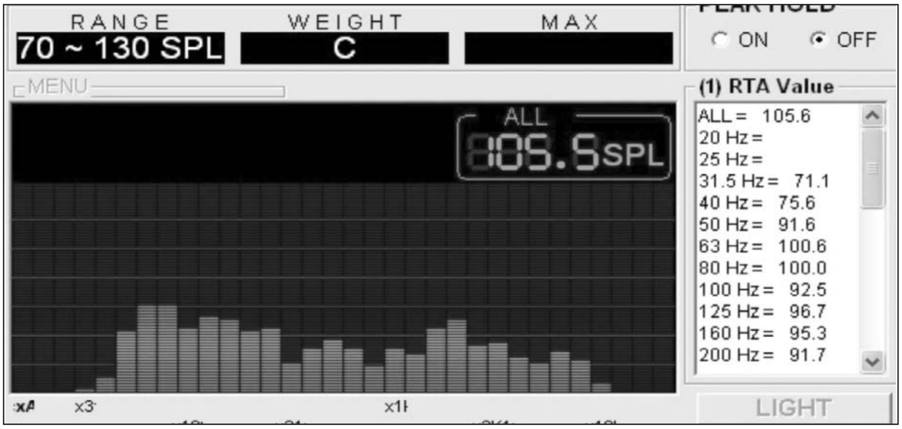

FIGURE 1: RTA of "ZZ Top" in concert.

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2. A horn won't work below its Fc (corner frequency).

A horn will obtain maximum sensitivity near Fc when the box resonance, Fb, approximates Fc; for that reason designers traditionally have used drivers with Fs close to the Fc. However, the combination of the air mass of the horn and the resonant chamber produced between the cone and the baffle when the throat size is smaller than the cone area causes the effective driver resonance to be lowered to Fs(h), the driver/horn resonant frequency. This Fs(h) may be an octave or more lower than Fs, which can result in a low Fb; the lower the Fb is relative to Fc the worse the sensitivity near Fc.

I've found no evidence that the cause of this phenomenon was ever properly identified by horn designers in the past, but they did recognize the symptom, and gave a name to it: "throat reactance." They also found a cure, in what is referred to as "reactance annulling," which is accomplished by using a small sealed rear chamber to push the Fb back up to the Fc. This offsets the low Fs(h), and the lost sensitivity near Fc is regained. The technique works best with low Qts drivers, which are inherently well suited to small rear chambers, and therefore have long been specified for horn-loaded applications.

When reactance annulling is used, there is a rapid response loss below Fc, as the 12dB rolloff of the horn and 12dB rolloff of the sealed rear chamber both kick in at roughly the same frequency for a combined slope of 24dB/octave. This rapid rolloff gives credence to the theory that horns won't pass sound below the Fc. However, if you don't require maximum SPL to the Fc, it's perfectly all right to have the horn Fc at one frequency and the Fb at a lower frequency.

While the horn gain below Fc will drop at 12dB/octave, at some point the sensitivity loss will cease as the driver continues to operate as a direct radiator. In this scenario, a large rear chamber with a high Qts driver will deliver the best low-frequency performance, and the driver in direct radiator mode will operate to the Fb before it begins to roll again. As the Fb is primarily determined by the Fs(h), the driver may operate to an octave or so below its Fs, depending on the horn.

The resulting "stepped" response curve (*Fig. 2*) is not necessarily what you'd want for your hi-fi or home theater, but for live sound it fits the bill perfectly. An impedance spike at 70Hz (*Fig. 3*) denotes the lowest frequency where the horn operates with maximum loading, below which response falls off at 12dB/octave until the driver starts operating in direct radiator mode, which occurs at about 50Hz. Normally a driver with a 55Hz Fs would not work much below 50Hz, but in this case the horn pushes the driver Fs of 55Hz down to an Fs(h) of 28Hz. With the rear chamber sealed, the Fb ends up at 32Hz (seen as another spike on the impedance chart), and sure enough the driver works down to 32Hz, below which the SPL rolls off at 24dB/octave as the 2nd order high-pass functions of the horn and the sealed rear chamber combine for a 4th order slope.

To be sure that I was getting the most from the Tuba 24, I tested it with two other drivers. A P-Audio™ SN10-MB with a higher Fs and lower Qts worked better above Fc, as expected, but performance below Fc was decidedly inferior. An Eminence™ Kappa-Pro Ten with both a lower Fs and lower Qts was not even close to the PS-10, both above and below Fc. The only disadvantage of the higher Qts driver is that with inputs in excess of 200W it will distort; for very high power usage it would be better to use a lower Qts driver and accept some loss of bass response. However, if maximum bandwidth and sensitivity at moderate power levels is the goal, the inexpensive PS-10 works quite well despite specs that would indicate otherwise. Rule number two is history.

3. Folded horns must use flat reflectors at the horn bends.

Traditional theory states that bends in folded horns should have flat reflectors installed at a 45° angle to the pathway. I pretty much had already dealt a deathblow to that rule with my Snail and DR horns, which proved that maximum broadband response is obtained from a rounded horn bend with a large inner radius. However, I wasn't so sure they would be beneficial in a pure bass horn. In fact, I preferred to have a fairly sharp acoustic rolloff characteristic to minimize the low-pass crossover components requirement, so I decided to build prototypes both with and without reflectors to see what happened.

The result was that within the desired passband—up to 200Hz—there was no difference whatsoever between the Tuba 24 versions built with and without reflectors, both flat and curved. I ended up putting a 45° reflector on the lower rear corner of the box as a convenient spot to mount casters, but it had no sonic effect. Not only is rule number three dead, it is also buried. RIP.

4. You can't combine a 6th order dual chamber vented box with a horn.

This rule doesn't actually appear in any of the writings I've seen, but it is a de facto rule since no one I know of has done it yet, which was a good enough

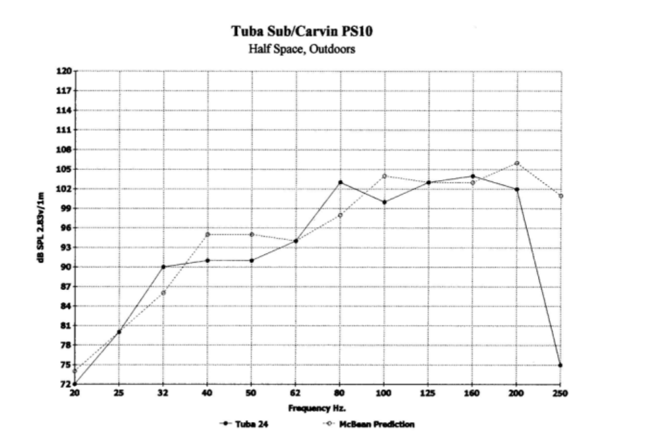

FIGURE 2: SPL chart 1. The Tuba 24 SPL, predicted and actual.

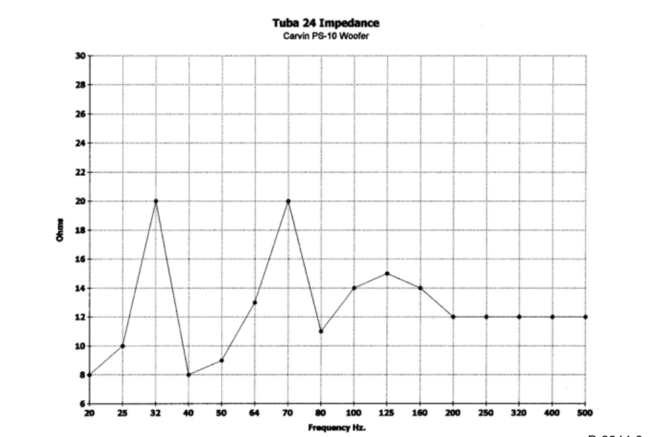

FIGURE 3: Tuba 24 impedance.

reason for me to try it. I wanted to vent both a front and rear chamber, tuning them an octave or so apart as in a 6th order reflex, and then have both vents enter the horn throat; a number of the accompanying pictures reflect that attempt. It didn't work, and I figured out the reason why.

With a fairly short horn of relatively high Fc, you can use a ducted vent to tune the rear chamber to a frequency well below Fc—ideally the Fs(h)—venting the rear chamber to the air or even into the horn throat, and obtain useful output from the vent. However, if you vent the rear chamber directly into the throat of a horn and attempt to tune the rear chamber to an Fb within about a half-octave or less of Fc, the air mass of the horn completely swamps the air mass of both the rear chamber and duct and tuning is not possible.

If you choose to tune the Fb within a half octave or so of Fc, vent the rear chamber directly to the air, even though the horn's output will overshadow that of the port to the point of making it moot. My Snail and DR horns have rear chambers tuned at least an octave below Fc and they work well, but in the case of the Tuba 24 there was no performance improvement gained from venting the rear chamber, so I left it sealed to prevent the driver from unloading below Fb. It may be possible to combine a 6th order dual chamber vented box and a horn, but I haven't figured out how to do so . . . yet. Hey, you can't win them all.

CONSTRUCTION

Now that I've covered the process behind the design, here's how you can build your own Tuba 24. For materials, I used ½″ spruce plywood throughout; the self-bracing design makes thicker materials unnecessary. You can use Baltic birch if you wish, but be aware of the added weight; there are better than two full sheets of plywood in this box, so it's no flyweight by any means. All

PHOTO 2: The baffle.

PHOTO 3: Attaching the first horn plate to a side.

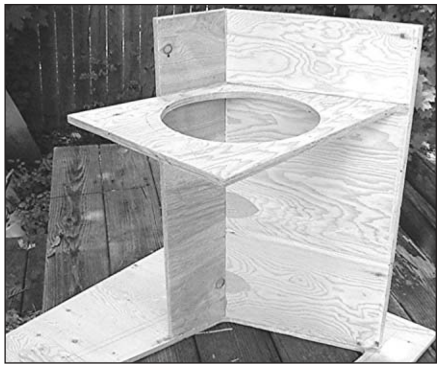

PHOTO 4: Attaching the baffle to the assembly.

joints are butted and secured with drywall screws and construction adhesive.

I strongly recommend using a polyurethane-based adhesive that expands as it cures to seal joints tight, as any air leaks in the box would seriously degrade its performance. Pilot holes and countersinking for the screws is a must, as is accurate sizing of parts. For instance, whereas all of the parts except the baffle and sides are the same approximately 23″ width, I recommend that you rip enough plywood to make them at the same time on a table saw to ensure they are the same width.

The pictures and text don't match perfectly, for reasons previously noted. Follow the text precisely, using the pictures only as a rough guide to the construction process. The actual sizes of all parts are figured by dead reckoning, as there is no such thing as ½″ plywood that truly measures a half-inch thick. Start by cutting out the cabinet sides, both measuring 24″ × 24″. Draw the parts layout pattern of *Fig. 4* onto one of the sides.

Then draw where the top, back, reflector, and bottom intersect, and lay out the horn panels starting at the mouth working backwards from horn plate 6 to horn plate 1. The measurements shown at each bend of the horn are clearance dimensions, while the material used determines the thickness of each panel and consequently the length of each part.

Cut out a porthole, which is framed by the first four horn plates, in the cabinet side, using a saber-saw and starting with a plunge cut. The vertically aligned baffle is exactly the same size as the porthole, so use the porthole as a template to cut out the baffle (*Photo 2*) and cut a 9″ diameter hole in it, using the woofer as a guide for alignment. Install driver mounting "T" nuts if desired.

Next, cut out horn plate 1 (*Fig. 5*). [Horn plate 1 as shown in the picture is incorrect, as I altered the design later.] One corner of horn plate 1 is cut away to provide a pathway for the driver front wave to enter the horn, which starts its route in the gap between horn plates 1 and 5.

[The configuration of horn plate 1 as shown produces a compression chamber on the front wave of the driver, which acts as a low-pass filter. This gives the lowest possible Fs(h), and thus the lowest possible frequency extension. However, at input levels of 200W or more, high acoustic pressure in this throat chamber can result in distortion. You can relieve the pressure by making the compression chamber exit larger, but Fs(h) will go up, and the Fb along with it. For electric bass or home usage I recommend making horn plate 1 as shown, but for the maximum possible output in PA duty do not extend horn plate 1 beyond the baffle.]

Attach horn plate 1 to the side, using a guideboard and clamps to hold it in position while driving the screws (*Photo 3*)—a procedure you should use whenever possible. Cut out the second horn plate, attach it to the assembly, and then attach the baffle to the two horn plates (*Photo 4*).

[Note in *Photo 5* that another horn plate section has been attached to the assembly. This is actually a continuation of horn plate 1, as initially I made horn plate 1 from two separate pieces of plywood.]

At this point, start making the horn braces—which are pieces of plywood—again determining their sizes by dead reckoning. The first sets of braces are attached to the first two horn plates, roughly halfway across their spans (*Photo 6*). Drill these braces full of holes

PHOTO 5: An initial horn plate configuration, later amended.

PHOTO 6: Attaching braces to the horn plates.

PHOTO 7: Attaching braces to a horn plate prior to plate installation.

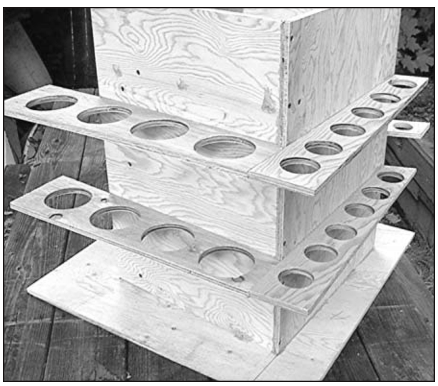

PHOTO 8: Continuing installation of braced horn plates.

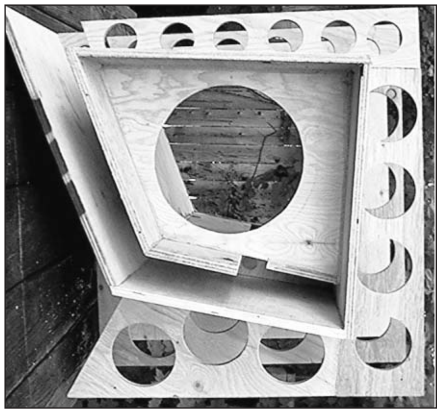

PHOTO 9: View with all horn plates installed.

PHOTO 10: A porthole with flange installed.

"Swiss cheese" style—either with various size hole saws or via saber-saw—to both lighten them and to allow sound waves to travel freely through them.

The braces on the third and subsequent horn plates are doubled, dividing the span into approximate thirds, attached to their respective horn plates prior to installing the plates to the assembly (*Photo 7*). Install the subsequent horn plates and braces sequentially (*Photo 8*) until all are in place (*Photo 9*). Note that you may joint the braces end to end, but this is not a structural requirement, and a gap of an inch or so between where one brace ends and the other begins is perfectly OK. Also note that the 45° reflector at the joint of the back and bottom is not shown in *Photo 9*, so keep that in mind when making the last two sets of braces. You can construct the lower back with or without the reflector as you may wish.

In sequence, add the top, back, reflector, and bottom to the assembly.

Temporarily screw the remaining side to the assembly; this will allow you to trace on it the intersections between it and the first four horn plates from the inside, reaching through the porthole. Remove the side and use the tracings as guides to pre-drill screw pilot holes. Use scraps of plywood or ¾″ stock to rim the porthole opening, producing a flange (*Photo 10*).

On the prototype, I initially put portholes on both sides; the view of *Photo 10* is through the porthole leading to the throat chamber, with the driver in place. In practice, the second porthole is not required, only the one leading to

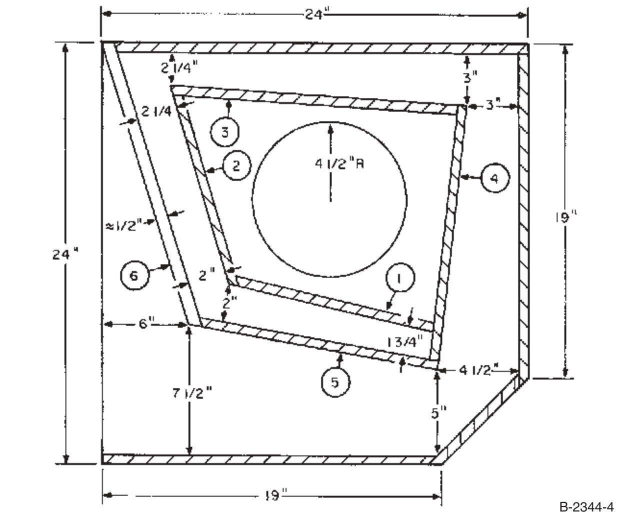

FIGURE 4: The Tuba 24, cutaway side view.

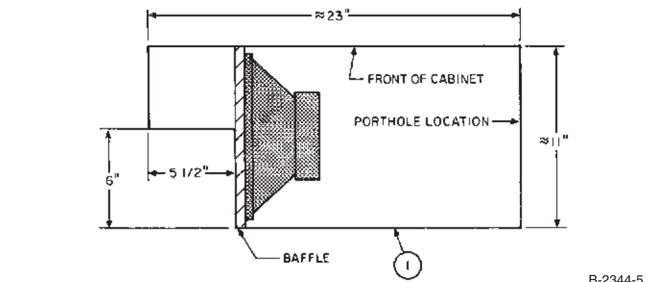

FIGURE 5: Horn plate 1.

the rear chamber.

Attach the second side to the assembly. When the adhesive has set, sand or rout the exterior edges, rounding over the corners as may be required for any protective hardware. Apply your finish of choice, handles, and protective hardware and install appropriate jacks, of airtight design only, on the porthole cover; "Speakon"™ jacks are a good choice here. Install the driver, either with bolts into previously installed "T" nuts or with screws; this driver is not particularly heavy, so screws are a viable option.

PERFORMANCE

I tested the cabinet both with and without stuffing in the rear chamber and there was no difference, so don't bother with any damping material. A single pair of casters mounted on the reflector (*Photo 11*) makes the cabinet easy to roll about when tipped back a bit, while a pair of table leg "cups" (*Photo 12*) will keep another wheeled cabinet placed atop the Tuba from rolling off. Another option is to mount a speaker mounting pole receptacle either in or atop the cabinet, for "flying" a mid-bass box above it; just make sure that you don't disrupt the airtight horn path.

With its high average impedance (*Fig. 3*) and rapid acoustic rolloff, you can run the Tuba 24 in parallel with a mid-bass cabinet without a crossover. However, for better loading, use a 10mH inductor in series with the driver as a 100Hz 1st order low-pass filter. Make sure it is a high-current capacity iron or ferrite core inductor with less than 0.5Ω DC resistance. [If you wish to include a high-pass filtered output to go to a mid-bass box, a 200µF/100V non-polarized capacitor will provide 6dB/octave filtering from 100Hz for an 8Ω load.]

Mount the inductor securely on the inside of the porthole cover; wire the driver positive lead to it and then to the positive pole on the jack, and the driver negative lead to the negative lead of the jack, using 12 or 14 gauge speaker wire. Rim the porthole flanges with weatherstrip and screw the porthole in place.

A logical question is how the Tuba 24 performs paired with a high-quality mid-bass/HF cabinet. The answer is found on SPL chart 2 (*Fig. 6*), which traces the combination of a performance of a Tuba 24 and a DR250a with sealed rear chamber (*Photo 13*). On the same chart is a modeled response of the benchmark 8 × 10 commercial electric bass cab, the Ampeg™ SVT; the Tuba 24/DR250a combination is considerably more powerful. In fact, the Tuba 24 has a higher SPL and lower frequency extension than the typical commercial 18″ sub, even some that are horn-loaded and considerably larger. For PA use you'd be hard-pressed to find any commercial gear anywhere near the equal of a DR250a/Tuba 24

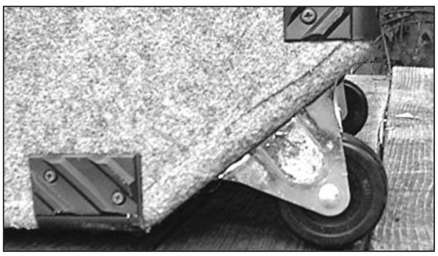

PHOTO 11: Using the reflector for caster mounting.

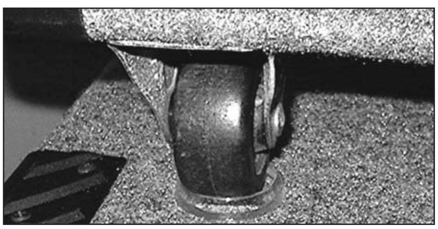

PHOTO 12: Using table leg "cups" to secure a stacked cabinet.

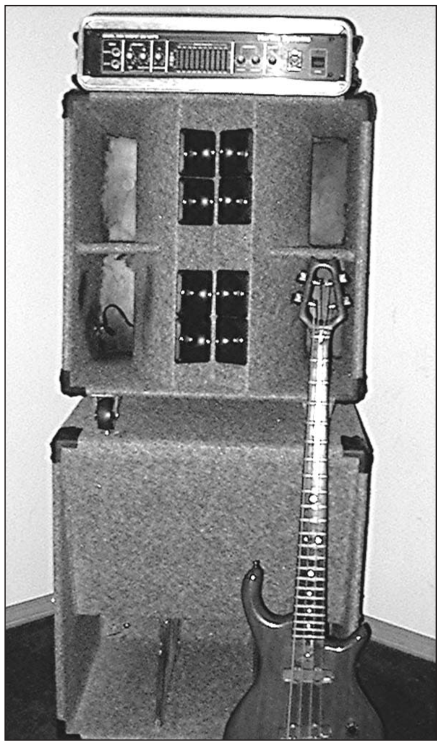

PHOTO 13: The Tuba 24/DR250a combination.

combination for less than $1,000 per box, and quite a few at three times that price still come up short.

Finally, there will be those of you who'd like to know whether the Tuba 24 makes any sense for hi-fi or home theater. The answer is yes. The preceding SPL charts reflect measurements made out of doors. SPL chart 3 (*Fig. 7*) shows the measured in-room response facing the center of a wall, 8″ out, and with corner placement, the horn mouth facing the corner. The additional loading enhances performance below 60Hz quite nicely, making the Tuba 24 eligible for SET amplification and a good match with high-efficiency extended range mid-bass drivers.

On the other hand, if Tuba 24 is a bit more than you need and you don't care to press the WAF (Wife Acceptance Factor) too hard, you might wait for my next sub project, the Tuba 18. As the name implies, it's a Tuba sub with an 18″ outside dimension. Designed specifically for in-home wall or corner placement, it manages to squeeze a 7′ horn into a box small enough to even qualify for auto-sound use.

The rear chamber is so small that a ten won't come close to fitting inside, so it uses an eight. But it still will manage an average 100dB/1W SPL from 40 to 400Hz. I can hear the traditionalists already bemoaning, "you can't do that—it breaks all the rules." Well, when it comes to my breaking loudspeaker design rules, as the old saying goes: "You ain't seen nothing yet!" ❖

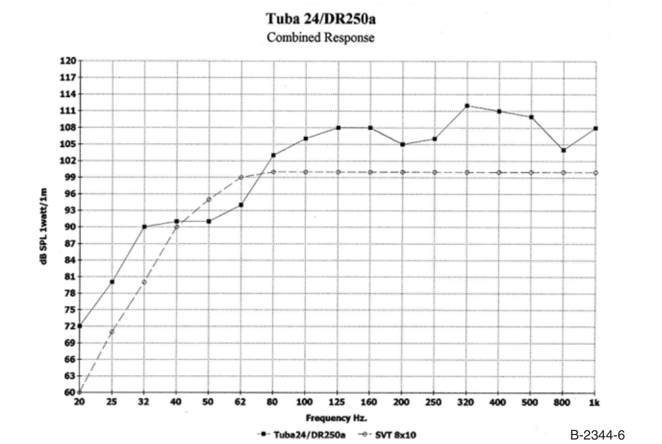

FIGURE 6: SPL chart 2. Tuba 24 and DR250a combined response.

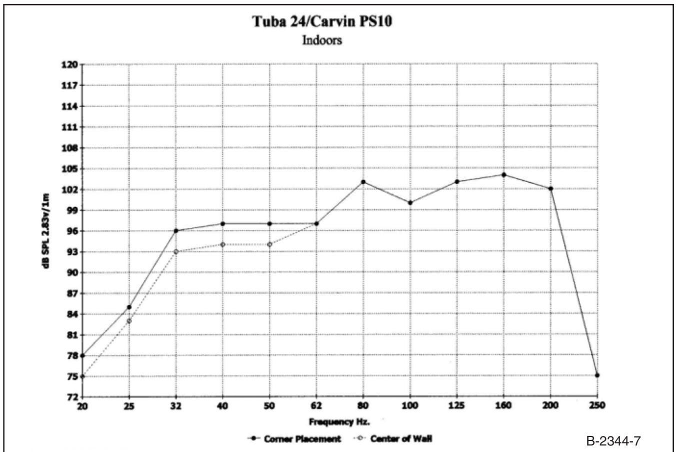

FIGURE 7: SPL chart 3. Tuba 24 in-room response.

Tube Audio Construction Tips Part 1: Tubes

Here's some valuable advice from an experienced audio constructor to help you get started with tubes. **By Graham Dicker**

The idea of this series is to pass along to others tricks of the trade that I have discovered over the years before they become lost forever. I hope that this will encourage others to share what they have learned as well, and provide many ideas for readers to experiment with. The key here is not to just read about it, but to pick up a soldering iron like the rest of us with a passion for things that glow in the dark, and build something. This is where the real joy comes from, and at the end of the project you can listen and look at your own work of art.

I hope that I can share how to build a tube amplifier for almost nothing. I am not talking about building a nice WE300B SE job in which the tubes set you back a cool $1000 plus another $1000 for the output transformers and other bits, but a respectable amplifier costing little. With a little prudence—and an approach to the project like Scrooge himself—you can make a tube amplifier for very little, and have a lot of fun along the way. Over the last 40 years of building tube amplifiers, I have found that it takes about as long to build a small amplifier as a large one. With this in mind, I now spend most of my time building single-ended Class A monsters using transmitting tubes in the output.

Many of my secrets are not without technical drawbacks. I accept any criticism in advance for this, but you, too, can make a 50W per channel SE class "A" amplifier for under $40 with my handy hints, should you wish to take up the challenge.

Photos 1, 2, 3, and 4 courtesy of
Dabney Crump
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Suite # 150
Centennial, CO 80112
(303) 790-1830

PURCHASING TUBES

This is the start to your quest. If you do not have any tubes to use, you need to get some. The Internet is a great place to purchase them. You can either purchase to plan, or if you want to experiment, there are a large number of TV service caddies (*Photo 1* and *Photo 2*) with hundreds of tubes in each available for the very low cost of around $10 US per 100 tubes. These caddies come from RCA, GE, Sylvania, and others and yield a variety of useful power output tubes for audio, vertical, or horizontal deflection (sweep tubes), and signal level triodes and pentodes, often including 12AT7, 12AU7, and others in the kits (*Photo 3*).

In my opinion, these caddies are the best value around. You can never have too many. If you live some distance from the vendor and the shipping costs are too expensive, then you can request of some vendors that they package and ship the tubes without the case, thus saving you a bundle in postage.

Used, untested tubes from Internet auction sites are often not a bargain. Over the last five years that I have purchased tubes, those that were not tested prior to purchase or guaranteed by vendors offered a low usable yield. Most tubes were 70% in emission or lower, and many had open circuit filaments.

NIB and NOS tubes often turned out to contain failed tubes in the original cartons. TV servicemen, in particular, often put dead tubes back into the same carton that the new tube came from. This is not always the case, but a usual giveaway is an altered box label or a cross on the carton (*Photo 4*). Another method was to bend the pins on the base of a tube so that it could not be in-

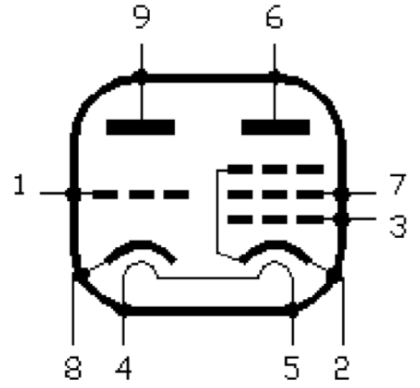


FIGURE 1: 6GW8 tube pinouts.

PHOTO 1: RCA tube caddy.

serted into a tube socket.

Those that were genuine unused NOS and NIB tubes rarely turned out to be duds, and almost all tubes that I have purchased have been okay. It's a good idea before using older tubes to run them up on filaments only for a few days first. I installed octal, noval, and 7-pin sockets on my bench test power supply for just such a purpose. Often a tube tested new from the box will test low, but after a few days of filaments only, it will often re-test 100% and be perfectly okay.

From my experience, however, genuine NOS tubes seem to outlast new, brand-name manufactured tubes, by as much as 10 times the life span. While I have paid around $100 US for a good

PHOTO 2: Tube caddy opened showing contents.

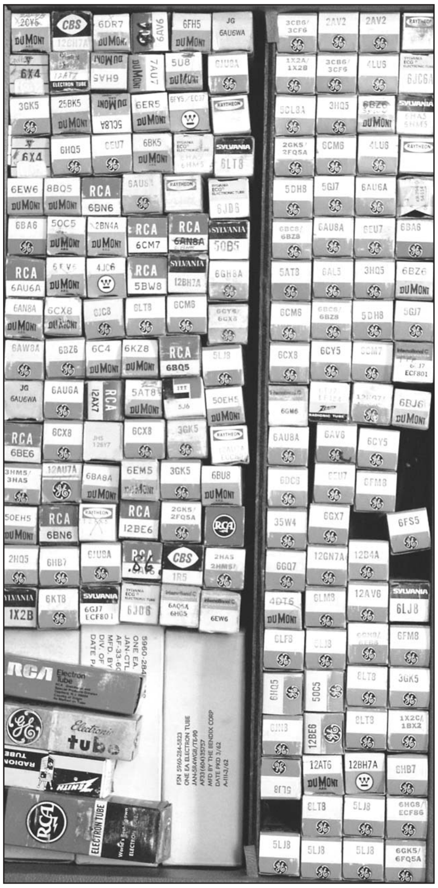


PHOTO 3: One-third of the tubes in caddy.

Mullard 1960 vintage NIB EL34, in practice I have found it is better to pay the premium and not re-valve the amplifier as often. You'll often find an added bonus with these vintage tubes in the improved sonic qualities. There are also newly manufactured tubes that come with good credentials. The choice is up to you.

Europe—and, in particular, Lithuania—have some entrepreneurs who sell new ex-military tubes. I often source tubes from these people with great success. I have found most of these to be new in original boxes. Prices for these imports are most reasonable, and if you purchase some spares as well, you will amortize the shipping costs over more tubes. This results in a lower cost per tube. Kwtubes (www.Kwtubes.com) is one of my regular suppliers for 6V6, 6L6, EL34, and GK-71 tubes. These sell for around $6–10 US each and represent good value for the money.

PHOTO 4: Suspect tubes in caddy.

I prefer to use transmitting tubes in the output stage (*Photo 5*). Good ones are available at low cost with good anode dissipation limits. Some of my favorites include pentodes and tetrodes such as the ceramic 4CX250B, the 813, and its Russian equivalent—the GK71, the RCA 4-125A, the Philips 3-300, and, for the audiophile, the Eimac 304TL or GI-7B ceramic triode.

Most of these tubes cost from $12 to $25 a pair, and will represent the major portion of your investment. You can purchase low-level tubes such as the EF91/6AM6 for around two for a dollar. An EF86 will be a little more at $5, while 12AX7 and 12AT7 twin triodes sell for around $5 to $7 each for good ones.

TUBES FOR PROJECTS

For a first project, a single-ended single tube design is a good place to start. You have a number of choices for a suitable tube. To keep this project affordable and simple, you can use a combination tube with a triode and a pentode (*Fig. 1*), such as the 6GW8 or the 6BM8 (*Photo 6*), which are readily available on eBay for around $2 each. I have also used the 6GV8 TV tube (vertical oscillator and output) with good results. If you are a purist and like triodes, the 6CG7 by itself or in parallel is a particularly good choice. You'll find most of these tubes in the TV service caddies mentioned previously.

I have used RF pentodes for audio in applications where noise and distortion are not a big issue. Half a century ago, I discovered—much to my surprise—that many of the UHF TV tuner triodes and pentodes actually performed better than the respected 12AX7 triode. One of the hidden gems is the 6AM6/EF91, which performs almost as well as the EF86.

Another alternative is to use two separate tubes—one a power amplifier, the other a voltage amplifier (tube selection is endless). A junk box search or the local TV repairman is another good place to start. Hamfests and boot sales are also good sources of bargain tubes.

If you do not like TV-type tubes, you can use a 6V6, 6L6, or an 807 as an output tube. Octal tubes such as these (5-pin in the case of the 807) are often overlooked and are readily available as NIB for around $5 US each. If purchasing specific tubes for a project, always buy some spares at the same time. ❖

PHOTO 6: ECL86/6GW8 tube.

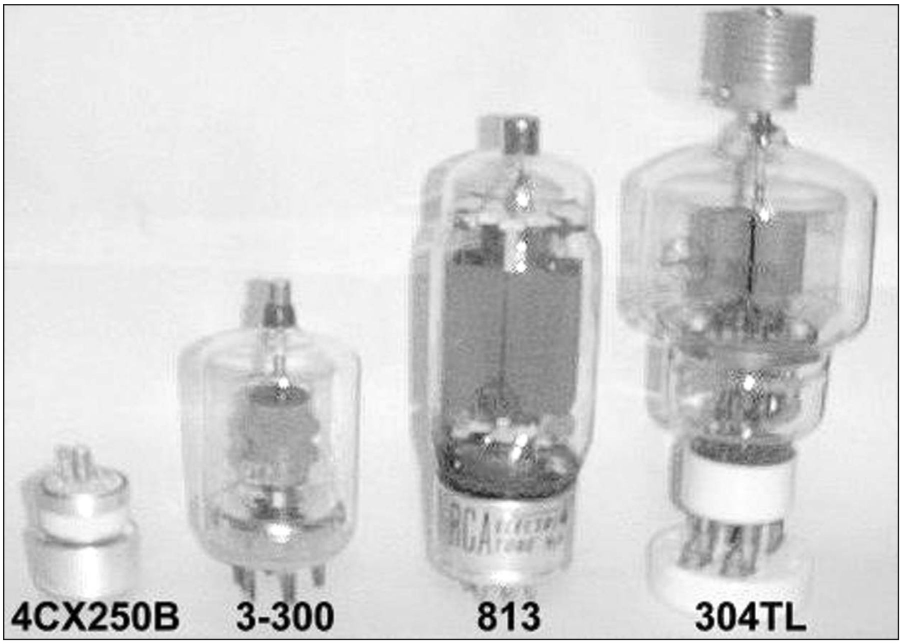


PHOTO 5: A range of transmitting tubes for audio use.

New Chips on The Block

By Charles Hansen

AD5241 Digital Pot

The AD5241 and AD5242 I^2C® compatible digital potentiometers contain a single and dual channel, 256-position, digitally controlled variable resistor. Operating from 2.7 to 5.5V DC or ±2.7V DC supplies, these devices are ideal for multimedia, video, and audio applications.

Stable 30ppm/°C, low noise settings result from internal SiCr thin film resistor technology used for the resistor segments. Internal power ON reset to midscale position speeds up initial circuit adjustment.

Two programmable logic outputs are available to drive digital loads, gates, LED drivers, analog switches, and so on, to save micro-controller pin usage. Additional package address decode pins allow multiple packages to share the same 2-wire I^2C bus. Devices are available in the 1.1mm thin TSSOP 14- and 16-lead packages.

FEATURES

Replaces potentiometers, trimmers, or variable resistors in new designs

2-Wire I^2C compatible digital interface

10k, 100k, and 1MΩ end-to-end terminal resistances

Low-drift 30ppm/°C absolute resistor temperature coefficient

APPLICATIONS

Audio Level Control
Video Gain Adjustment
PC Peripheral Adjustments
Optical Network—Control Loop Settings

For additional technical information, email the Digital Potentiometer Group at digital.pots@analog.com. Device pricing (1k pieces): $.99 (AD5241, single) $1.75 (AD5242, dual).

Analog Devices Inc.
804 Woburn St.
Wilmington, MA 01887
(781) 937-1428
fax (781) 937-1021
www.analog.com ❖

The Wolfson Stereo Volume Control and DAC

VOLUME CONTROL

The WM8816 is a highly linear stereo volume control for audio systems offering unsurpassed distortion and noise performance, with total harmonic distortion plus noise (THD+N) at 0.001% (–100dB). The design is based on resistor chains with external op amps, which provides flexibility for the supply voltage, signal swing, and noise floor. The gain of each channel can be independently programmed from –111.5dB to +15.5dB through a 3-wire digital serial control interface.

Audible clicks on gain changes are eliminated by changing gains only when a zero crossing has been detected in the signal. The device also features peak level detection, which can be used for automatic gain control. The WM8816 operates from a single +5V supply and accepts input signal levels up to ±18V.

The WM8816 is available in a 16-pin SOIC package. It is guaranteed over a temperature range of –40° to 85°C.

APPLICATIONS

Audio amplifiers/preamplifiers
Home entertainment systems
Consumer audio
Mixing consoles
Audio recording systems

For a White Paper on digital volume control technologies, visit the Wolfson web site at www.wolfson.co.uk. Device pricing $2.90 (10k pieces).

STEREO AUDIO DAC

The WM8722 is a high-performance stereo DAC designed for audio applications such as digital broadcast receivers, TV, and set-top boxes. The WM8722 has two stereo analog outputs, one at line level and one with digital volume control and soft mute. An on-chip tone generator can be routed through the line or variable outputs.

The WM8722 supports data input word lengths from 16 to 24 bits and sampling rates up to 96kHz. The WM8722 consists of a serial interface port, digital interpolation filter, multi bit ΣΔ (sigma-delta) modulator and stereo DAC in a small 20-pin SSOP package.

The 3- or 2-wire serial MPU-compatible control port provides access to all features, including tone generation, mute, attenuation, and phase reversal.

The programmable data input port interfaces to popular DSPs, audio decoders, and S/PDIF or AES/EBU receivers with normal (right-justified MSB first) or I^2S 16/20/24-bit outputs.

APPLICATIONS

Digital TV
Digital broadcast receivers
Set-top boxes

FEATURES

Sampling frequency: 8kHz to 96kHz

2- or 3-wire serial MPU-compatible interface for 16/20/24-bit input data

Soft mute; de-emphasis; volume control

On-chip tone generator (1Hz – 32kHz, 0.1 – 25.5s)

Stereo analog inputs

–80dB THD, 92dB SNR

Power supply 5V, 50mA; can operate at 3V

Device pricing $2.06 (10k pieces)

Wolfson Microelectronics
20 Bernard Terrace
Edinburgh EH8 9NX
United Kingdom
Tel: +44 (0) 131 667 9386
info@wolfson.co.uk ❖

Product Review

Alesis ML-9600 Mastering Recorder

Reviewed by Charles Hansen

Alesis, 12555 Jefferson Boulevard, Suite 285, Los Angeles, CA 90066; www.alesis.com (see web site for dealer network map), E-mail info@alesis.com, (310) 821-5000, Fax (310) 306-2650, price $1000, dimensions (W×H×D): 19″× 3.5″ × 11″ (483mm × 88mm × 279mm), weight: 16.55 lb (6.2kg), warranty: 1 year, parts and labor.

The Alesis MasterLink ML-9600 is a high-resolution two-track hard disk mixing and mastering recorder. It can record from analog, or AES/EBU (XLR) and S/PDIF (coax) digital inputs, or from its internal optical drive, all to its internal hard drive. Data sample rates available are 44.1kHz, 48kHz, 88.2kHz, and 96kHz, with word lengths of 16, 20, or 24 bits.

From the audio files stored on the hard drive, you can burn CDs using standard "Red Book" (16-bit/44.1kHz), or high-resolution CD24 formats up to 24-bit, 96kHz. You can choose any combination of digital resolutions (16-, 20-, and 24-bit) and sample rates (44.1, 48, 88.2, and 96kHz) with full Audio Interchange File Format (AIFF) compatibility.

You can assign audio files to any of 16 playlists, which can each store up to 99 tracks. You can store any mix of sample rate and word length files in a playlist. Once on the hard drive, you can edit your recording and apply finishing tools—such as high-resolution parametric EQ compression, limiting, and normalizing using the onboard digital signal processing (DSP). You have total control of fade-ins, fade-outs, track gain, start points, track cropping, and track split and join.

Then you can burn the finished recording to standard data CD-R in either industry-standard Red Book or the high-resolution CD24 format. CD24 is accepted by many DVD-Audio mastering facilities. You can also load either Red Book CDs or CD24 files onto the hard drive from the CD-RW drive.

PHOTO 1: ML-9600 front view.

PHOTO 2: ML-9600 rear view.

The operating software is stored in flash RAM. You can download software upgrades via Internet, burn them to a CD-R on your computer, and load the new software via the ML-9600 CD-RW drive. However, a new software installation will reformat your hard drive, so you should back up all your playlists and audio files. This particular unit had software version 1.24. The latest version is 1.25, which adds support for another Sony CD-RW drive and support for hard drives larger than 30GB, even though the ML-9600 will not recognize more than 30GB at this time.

The 49-page reference manual is thorough and well written, with individual chapters covering playlists, DSP, the recording process, and CD player mode.

INSIDE THE ML-9600

The ML-9600 is constructed of heavy gauge steel, with provisions for rack mounting. The rack "ears" are furnished with the unit and fasten to each side with three screws. Various areas of the chassis have vents to enhance cooling. Four large rubber feet support the chassis.

Photo 1 shows the front panel. The CD-RW drive tray is at the top left. Just below it are the remote control IR receiver window and the power switch. Next come the CD format button (High Definition or CD) and the group of record mode buttons that select sample rate, input source, and word length. The Create CD button is just to the right of the record mode group.

The vacuum fluorescent digital display is to the right of the CD-RW tray. Four cursor buttons are located below it in an oval grouping. To the right of the cursor buttons are the five Playlist select, Track, and DSP buttons. Just below are another group of five Playlist Edit buttons and the Utility button. The latter controls meter mode, loop mode, playlist backup and restore, Red Book start and end time offsets, file sorting, hard drive format, and allows you to access system information (software versions and drive information).

The hard disk "transport" controls are arranged in a stack at the right of the display. Play/Pause are incorporated in one switch, and the Record button is to its right. The final front panel controls are the HD/CD mode button, and a headphone volume control and ¼″ stereo phone jack. The ML-9600 acts as a conventional CD player in CD mode. The display format changes and the recording functions are locked out. In HD mode, the recording functions are enabled.

The rear panel (*Photo 2*) has an IEC power receptacle on the right. The third pin of the AC receptacle is connected to the chassis. Both balanced XLR and unbalanced phono jacks are provided on the right side of the rear panel. They are grouped as analog inputs, analog outputs, and digital inputs and outputs.

Photo 3 shows the ML-9600 from the front with the cover removed. This particular unit has a Sony ATAPI CD-RW optical drive, and a Maxtor IDE 20GB hard drive that allows storage of up to 30 hours of CD Red Book audio. In the 24/96k high-resolution format, you can store up to 9.4 hours of CD24 format audio on the hard drive.

The display and user interface board occupies the entire length of the front of the unit, incorporating a cutout for the CD-RW optical drive. The hard disk drive is located below the CD-RW. The ML-9600 uses a computer-like switching power supply, located on the left rear of the chassis. The CD-RW and hard drive receive the usual +5V and +12V power.

The long PC board at the rear of the chassis handles audio processing, connecting to the two drives with the usual IDE ribbon connectors. The XLRs are mounted on a smaller board that connects to the main board via pin and socket connectors.

TOPOLOGY

A schematic was not furnished with the unit, but a $50 CD-ROM service manual is available from the Alesis Parts Dept., (310) 821-5000. Analog audio input/output, including the balanced-to-unbalanced signal conversion, is handled by eight NE5532 dual low-noise op amps. An AKM AK5393 enhanced dual bit 108kHz 24-bit 128× oversampling ADC with integral anti-aliasing filter immediately converts all analog input to the selected sample rate and word length.

Analog output is processed at the same selected sample rate and word length by an AKM AD4393 96kHz 24-bit 128× oversampling DAC. This chip also provides digital de-emphasis for the 44.1kHz and 48kHz sample rates. Both the ADC and DAC utilize differential analog signals. The analog output is buffered by another 5532 and sent to the front panel stereo headphone jack. The balanced outputs are 75Ω, while the unbalanced outputs are 150Ω.

PHOTO 3: ML-9600 interior view.

The digital inputs accept standard AES/EBU (balanced XLR) and S/PDIF (unbalanced coax) formats and are processed by a Cirrus Logic CS8414 96kHz digital audio receiver chip. Optical formats are not supported. A Cirrus Logic CS8404A 96kHz digital audio interface transmitter drives the digital output jacks through 75Ω interface transformers.

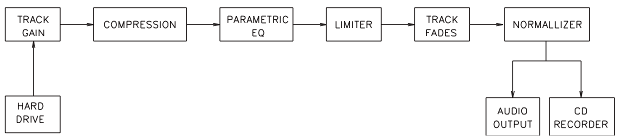


FIGURE 1: DSP block diagram. A-2282-1

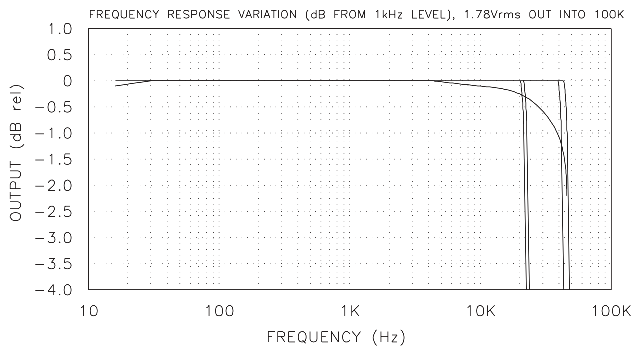


FIGURE 2: Frequency response. A-2282-2

Digital signal processing is provided by an Analog Devices ADSP-21065L SharcTM 32-bit DSP chip. Two 1M × 16 SDRAM chips provide support memory. The flash memory chip is a socketed Alesis-labeled device. Photo 4 shows the neat layout of the main multi-layer PC board and the DSP chip.

USING THE ML-9600

The ML-9600 is furnished with a full-feature remote control, but is very easy to operate from the front panel controls. In CD mode, the recording capability is locked out and Red Book 16-bit/44.1kHz is automatically selected.

TABLE 1
MEASURED PERFORMANCE

PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Freq Response (44.1/48kHz):	20Hz–20kHz, +0/–0.3dB	20Hz-20kHz,+0/–0.1dB
Freq Response (88.2/96kHz):	20Hz–40kHz, +0/–0.5dB	20Hz-40kHz, +0/–0.2dB
Total Harmonic Distortion:	<0.002%, 1kHz, –1dBFS	0.0025%, 1kHz, –1dBFS
IMD – CCIF (19+20kHz):	N/S	–89dB (0.0035%)
Signal to Noise Ratio:	–113dB, A-weighted	<–100dB ref 2V RMS, A-wtd
Maximum Output, Balanced:	+19dBu=0dBFS	6.92V RMS (+19dBu)
Maximum Output, Unbalanced:	+5dBV=0dBFS	1.78V RMS (+5dBV)
Maximum Output, Headphone:		5V RMS (10k), 1.6V RMS (32Ω)
Analog Input Impedance:	15k balanced	30k balanced
	10k unbalanced	10k unbalanced
Analog Output Impedance:	75Ω balanced	76Ω balanced
	150Ω unbalanced	128Ω unbalanced
		68Ω headphone
Power Requirements:	40W maximum	

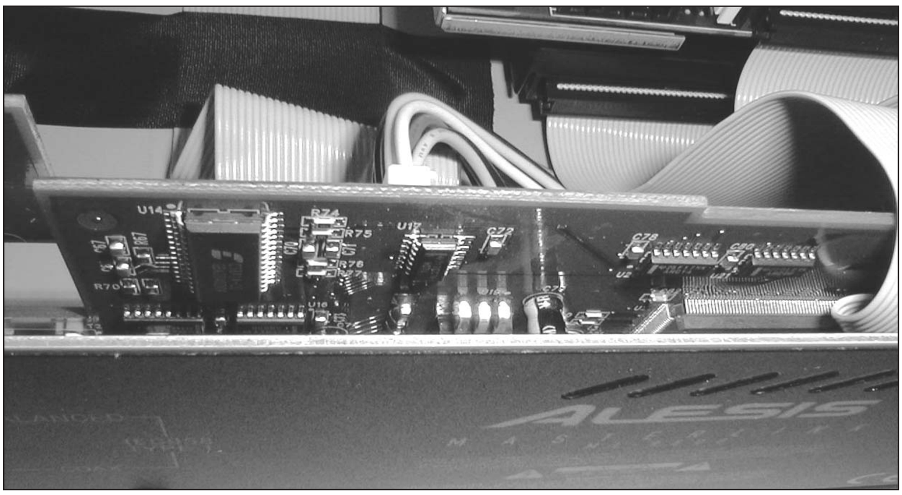


PHOTO 4: DSP Chip on main PC board.

The ML-9600 has some operational limitations when used as a CD player. There is no numeric keypad on the remote, so you cannot program individual tracks. You must use the skip buttons to go sequentially from track to track, and you can't program a series of individual tracks for playback. You don't have the usual track time and total time displays unless you change the default display mode. Keep in mind that this is a mastering recorder, not a consumer CD player.

You can move CD tracks onto the hard drive, then produce up to 16 playlists of up to 99 tracks each in any order you prefer. You can then use the ML-9600 as an audio jukebox if you wish. The ML-9600 does not record on CD-R to any of the DVD-Audio formats, but the CD24 high-definition AIFF format is recognized by most DVD mastering facilities.

When using the analog inputs, you can select any sample rate and word length you desire. The balanced inputs are rated for +4dBu nominal, with 15dB headroom (+19dBu maximum). The unbalanced inputs are the consumer –10dBV nominal with the same 15dB headroom, or +4dBV maximum. The analog outputs have the same dB ranges.

There are no provisions for analog input level control on the ML-9600. You need to limit the analog maximum peaks by means of external audio equipment. The analog inputs are converted to the selected digital format, and the internal hard drive records this digital data, which is also re-converted to analog and sent to the analog output jacks in real time. This allows you to monitor the audio the way it will sound when played back from the hard drive. In fact, you can connect the ML-9600 into the same tape loop that you would connect a cassette tape deck, easily integrating it into most audio systems.

If you intend to burn standard Red Book CDs from your analog source, the manual suggests recording files in 24-bit 88.2kHz format for hard drive storage. This allows the cleanest sample rate conversion to 16-bit 44.1kHz of all the high-definition formats, using a proprietary high-quality dithering algorithm.

For digital input, the ML-9600 accepts any valid AES/EBU or S/PDIF data input and automatically switches to external clock mode. You can use either the balanced or the unbalanced digital input, but not both. The ML-9600 will output digital audio data at the clock speed determined by the connected source. This means you could conceivably play 44.1kHz recorded audio at 48kHz (if a DAT was connected, for instance), which would make the audio pitch too high. You cannot, however, record a Red Book CD placed in the internal CD drive to hard disk at any higher sample rate in an attempt to upsample the audio.

Editing features are extensive. Since editing is destructive (the original audio file is overwritten), it is best to back up the file before editing.

FUNCTIONS

You can rename any playlist (up to eleven characters) and/or rename any track (up to eight characters). The default format is "Playlist XX" and "Song XXX." You must use the cursor keys to scroll up and down through an alphanumeric character set and change one character at a time. Tedious work! (You may recall how difficult it was to come up with unique 8-character filenames when DOS was the only PC operating system.)

You can take each track and crop the head and/or tail to eliminate noise or unwanted audio. The ML-9600 can play five seconds forward or backward at a very slow speed (like "rocking the reels" on an analog tape deck). This allows you to precisely place new start and end points on the track.

You can record one long analog performance, then divide the single track into separate tracks using the "Track Split" feature. You can construct a single track from different performance takes by dividing a number of tracks at precise points into segments. You can combine the segment elements in a playlist and audition them (they will play back seamlessly). Then you can paste the elements into a single new track using "Track Join" if you are satisfied with the edit. The audio output is never muted during any mode of playback or edit.

The DSP section of the ML-9600 is non-destructive. It operates on the audio file data read from the hard drive without altering it (*Fig. 1*). You can apply track gain, compression, high-resolution 3-band fully-parametric EQ, limiting, fade-ins, fade-outs, and gain "normalizing" using the onboard digital signal processing (DSP).

The compressor is very versatile—adjustable threshold, ratio to 20:1, make-up gain up to 65dB, attack and release time to 9.9sec, and five choices of "knee" behavior at the threshold. The three EQ bands are each adjustable over 20–22kHz, ±18dB, with variable Q. The limiter is a 65dB range "look-ahead" peak limiter that prevents you from exceeding the maximum input limit, thereby avoiding the associated digital clipping. The fades have two logarithmic and one linear "shape."

You can copy and paste any DSP parameter of the current track to any other track. If you really muck up the sound, you can reset the DSP to the default values and start over. If you are satisfied with the DSP results, you can "render" the settings to an audio file. This is a destructive process (overwriting the original track) so the usual backup recommendations apply. You can also apply DSP to only certain portions of your recording using the Track Split and Track Join features.

One interesting feature is "Normalizing," which allows you to scan a track for the highest peak value, determine the ratio between the peak and full-scale, and multiply the gain so the highest peak value of the track is equal to 0dBFS. The normalizer is applied to the hard disk data in real time.

The CD burning process occurs at a rate between 2× and 4.5×. The ML-9600 ignores any remote-control input during the CD burning process. This allows the full digital resources to be devoted to recording.

A full 650MB Red Book CD-R will record in about 19 minutes, while a full CD24 CD-R takes 36 minutes. There is little incentive to use 20-bit word lengths for CD24. The CD24 AIFF file format orders 20-bit audio samples as 24-bit, so the maximum record time is the same for any given sample rate. You can record 29.6 minutes of 16-bit 96kHz data or 19.7 minutes of 20-bit or 24-bit 96kHz data on a 650MB CD-R. However, the hard drive stores 20-bit data in true 20-bit fashion, so 20-bit files give you more hard disk recording time than will 24-bit.

The ML-9600 supports 700MB CD-R (80 minutes), but not all CD players will read them. MP3 is not an option.

LISTENING TEST

My first test was a listening audition of

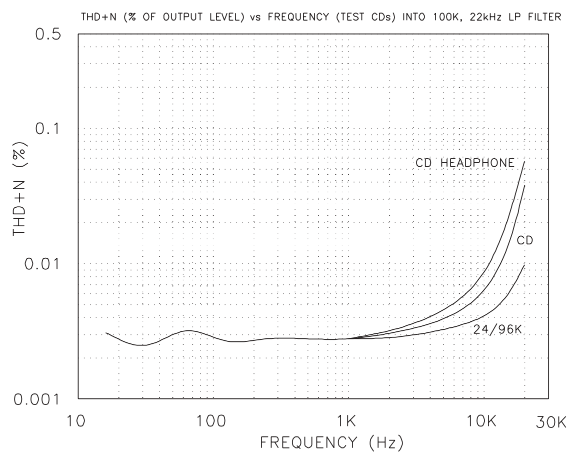


FIGURE 3: THD+N vs. frequency. A-2282-3

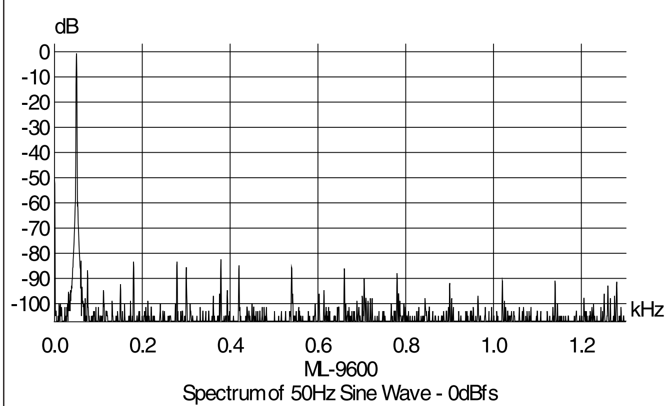


FIGURE 5: Spectrum of 50Hz sine wave. A-2282-5

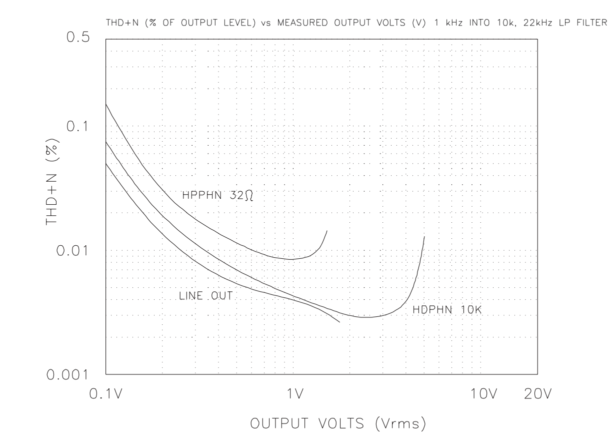


FIGURE 4: THD+N vs. output voltage. A-2282-4

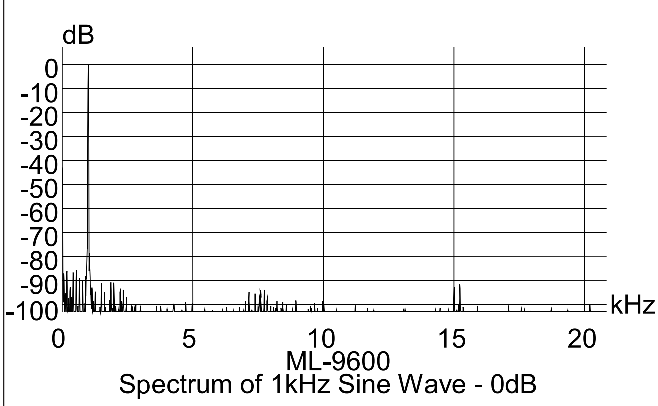


FIGURE 6: Spectrum of 1kHz sine wave. A-2282-6

the ML-9600 as a CD player compared to my modified Rotel 970BX CD player (*aX* 2/03, p. 26). The ML-9600 is an entirely satisfactory CD player (ignoring the lack of a numeric keypad on the remote and the limited digital display). It has very solid bass performance and a pleasing midrange. Only at the highest frequencies did I find it lacking compared to the Rotel.

While the highs are just a bit rolled off, they also have a noticeable edginess to them, almost imparting a sibilance on percussions such as chimes and triangles. The sound stage is also a bit more forward and not as spacious as the Rotel. These characteristics may improve with operating hours, as was my experience with the Rotel when it was new. And the ML-9600 is visually entertaining, with its dancing VU level meter bars.

The first recordings I made were direct digital copies from CD to the ML-9600 hard disk, using the CD-RW tray. This is a reasonably fast process, running at 4× rate. You first select a CD track you want to copy, then select the destination track in the playlist. The Track Move button starts the track copy process. Be careful to select a new track in the playlist each time, so you don't overwrite the previously recorded track. You can also move an entire CD to a playlist with the Move CD feature.

Next, I made a digital duplicate of the same CD tracks using a 75Ω S/PDIF coax connection between the Rotel CD player and the ML-9600. I listened to the playback from the ML-9600 hard drive in comparison to the Rotel. The same listening comments apply to these two digital copy methods as they did for the original CD played directly on the ML-9600. As you would expect, there were no audible missteps in either digital copy process.

I have a few LPs for which I also have duplicate commercial CD versions. The best test of the ML-9600's analog-digital-analog prowess is to record the analog LP files to the ML-9600 hard drive in nine of the twelve sample rate/word length combinations (I skipped 48kHz) and see how they sound as compared with the analog LP. I used shorter clips of what I considered to be key segments on the LPs, to allow faster reviews of the different data rate combinations.

I also went through the illogical step of connecting the analog output of my Rotel CD player to the ML-9600 analog input, and copying the analog audio to the hard disk in digital form (Would that be "DAD" on the old CD labeling standard?). I made the recordings at 16/44.1kHz and 24/96kHz. I copied all these various audio files, from both the analog and digital sources, to a single ML-9600 playlist, which I then auditioned in great detail.

First, I believe the digital "sweet spot" for analog LP copies lies at 20-bit/88.2kHz and above. The Red Book rate 16/44.1kHz copies are the least satisfying, while both the 24/88.2kHz and 24/96kHz copies come close (but not quite equal to) the LP playback itself. I also think that sample rate is more important than word length. The 16/96kHz copies were more interesting than the 24/44.1kHz copies. I think this is because the standard 0dBFS output level is identical in both formats at 2V RMS.

20-bit and 24-bit content extends the response further down into the noise floor rather than providing more headroom. Even if a CD player could play back 24-bits of resolution, it would likely exceed the noise floor of the analog playback circuitry. A higher sample rate, however, extends the nonlinear and nonharmonic products of the steep output filters beyond where they might have audible consequences. This appears to be the case with the ML-9600.

When I recorded the CD player analog output to the ML-9600 at 16/44.1kHz, there was some obvious degradation of the audio as compared with the original CD. The highs were rolled off, drier sounding, and less involving. The 24/96 recording of the Rotel analog output was more faithful to the original CD.

Finally, I burned the entire playlist I produced in the previous steps to Red Book CD-R for playback on my Rotel CD player. After you press the Create CD button, the ML-9600 will first determine that the CD-R you want to use is recordable, then makes sure the playlist will fit on the disk. Then the CD-RW drive goes through an initializing period where it does self-calibration and processes the table of contents (TOC) from the playlist.

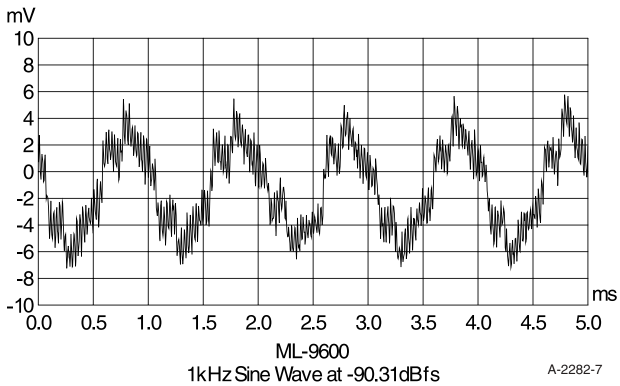


FIGURE 7: Undithered 1kHz sine wave at –90.31dBFS.

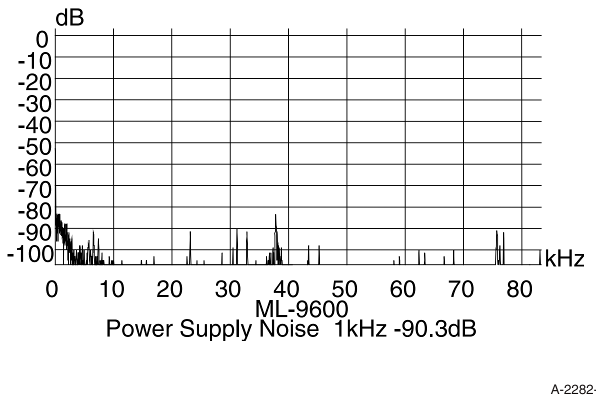


FIGURE 8: Wideband spectrum of 1kHz sine wave at –90.31dBFS.

The next step involves a "rendering" process whereby the selected DSP is applied and the audio is re-recorded to a special area of the hard drive set aside for this process. If you elect to burn a standard Red Book CD, the appropriate sample rate conversion and/or noise shaping is applied to all tracks other than those that already exist in 16/44.1kHz format. The high sample rate and long word length files take a fair amount of time to render. My 41:27-long playlist took 20 minutes to render and 10 minutes to burn to CD-R. If your playlist consists of only Red Book CD tracks without any DSP, the rendering process is not required.

Finally, I set about listening to the finalized CD-R on the Rotel CD player. The three analog LP 44.1kHz recordings at the three different bit rates all sounded the same. Perhaps the ML-9600 rendering process, when faced with any 44.1kHz sample rate audio file, merely truncates any extra bits.

Some of the LP songs I recorded to CD-R were not quite as good as some of the commercial CDs, while others were noticeably better. What surprised me was the high maximum recording level used on some CDs in comparison to the same songs on LP. I was fairly conservative during the analog recording, rarely exceeding +2dB (unbalanced inputs), while the CDs spent a lot of time near 0dB, with lots of hits on the MAX point on the level indicator. Compared with the LP, there sometimes appeared to be more compression used in the transfer from master tape to CD.

The dynamic range of my LP recording was limited by the LP surface noise, while the dynamic range of some of the commercial re-releases on CD were intentionally compressed more than I thought was necessary. I'm not sure why—the master tapes are a known entity in terms of dynamic range. However, the commercial CDs that seemed to preserve the original LP dynamic range were nicely done indeed. As with any recording process, the care given in preparation and attention to detail pays off at the end.

MEASUREMENTS

I operated the ML-9600 for one hour with a loud music CD before making any tests. I performed many tests using the CBS Labs CD-1 and Pierre Verany test CDs (both available from Old Colony Sound Lab). I used my distortion test set sine-wave oscillator for tests where a variable signal level was required or test CD digital data was not available. I did not evaluate the ML-9600 digital outputs. *Table 1* summarizes the measured performance and the manufacturer's specifications.

The unbalanced analog input impedance measured 10k (DC resistance), while the balanced analog XLR inputs measured 30k at 1kHz and appeared to be capacitively coupled. Pin 2 of the XLR jack is hot. The unbalanced analog

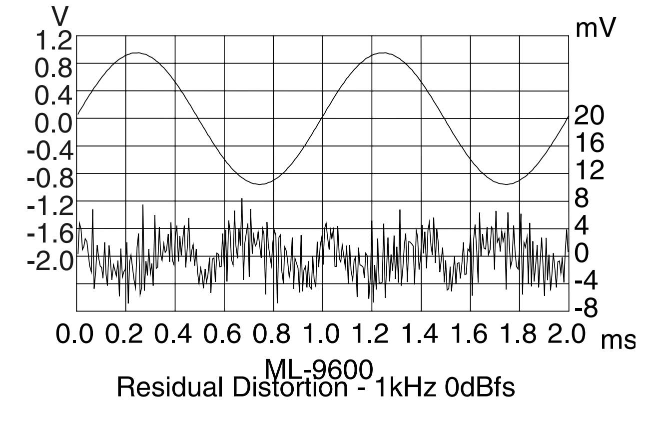


FIGURE 9: 1kHz sine-wave residual distortion.

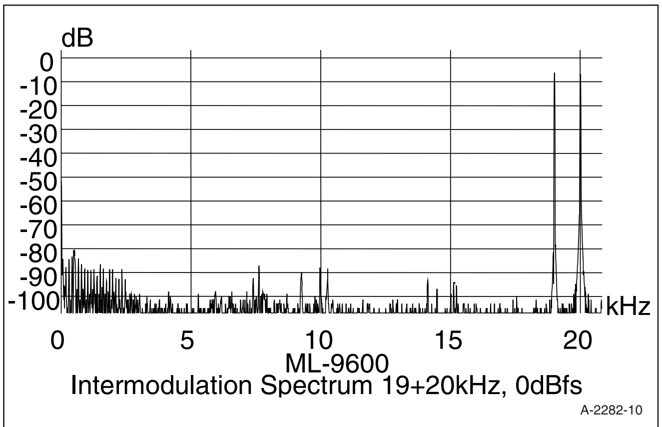


FIGURE 10: Spectrum of 19kHz + 20kHz intermodulation.

output impedance at 1kHz was 128Ω, increasing to 135Ω at 20Hz and 20kHz. The balanced analog output measured 76Ω, and balance between the two channels was perfect. The headphone jack output impedance measured 68Ω.

All analog outputs had normal polarity, a positive-going pulse producing a positive-going analog output (XLR pin 2 hot). The analog outputs are muted by a relay when power is off, and during some phases of operation.

The DAC frequency response for the ML-9600 was ruler-flat from 17Hz to the point where the digital output filters for each sample rate sharply rolled off the response (*Fig. 2*). The 0dBFS indicated unbalanced analog output was 1.78V RMS at 1kHz, or about −1dB below the CD Red Book standard of 2V RMS. The balanced output at 0dBFS was 6.92V RMS, reflecting the difference between the pro audio dBu and the consumer dBV standards. The −1dB level display indicator came on at 1.76V RMS and 6.85V RMS, respectively. I also measured the frequency response of the headphone output, which is the curved line at both ends of the graph, measured with a 0dBFS analog input signal and the input sample rate set to 96kHz.

Crosstalk between channels out to 16kHz was excellent and below the noise floor of my test equipment. Likewise, the signal-noise ratio was greater than 100dB referred to the Red Book 2V RMS with A-weighting.

THD+N vs. frequency at both the

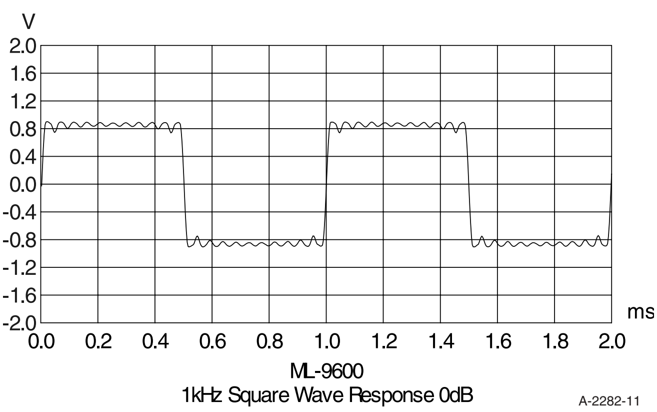


FIGURE 11: 16-bit 44.1kHz square-wave response at 1kHz.

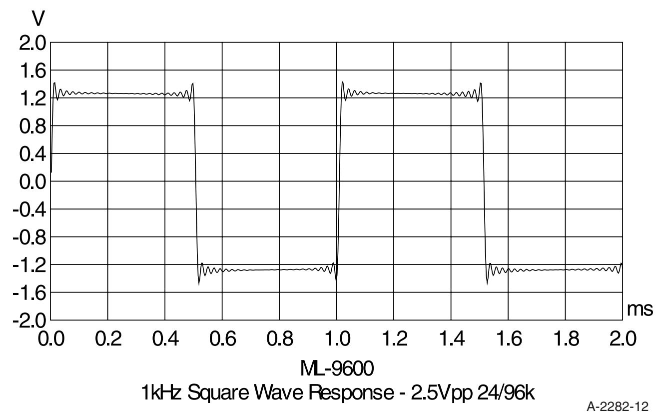


FIGURE 12: 24-bit 96kHz square-wave response at 1kHz.

standard Red Book format, and the maximum 24-bit 96kHz format are shown in *Fig. 3*. I recorded the 24-bit 96kHz tones onto the ML-9600 hard drive from the oscillator in my distortion test set. I engaged my distortion test set 22kHz 4-pole low-pass filter to remove out-of-band noise. I did not attempt to measure THD+N beyond 20kHz for the 96kHz data due to the lack of a suitable LP filter in the test set.

The upper curve is the distortion from the headphone jack with Red Book format data (10k load). At the lower frequencies the distortion is at or near the measurement floor of my distortion test set.

Figure 4 shows the THD+N vs. output voltage using the CBS test disk. The bottom trace is the unbalanced line output, the middle trace is the headphone output (both at 10k load), and the top trace is the headphone output with a 32Ω load. There is no clipping evident because of the CD test signal source. You can, however, drive the ML-9600 into clipping from the analog inputs. When I overdrove the unbalanced analog input, the distortion went straight up just above the point where the +4dBV level indication occurred. It does not sound pleasant!

An analog input signal of 1.78V RMS produced an identical analog output level, showing excellent analog-digital-analog conversion performance. The headphone output at the maximum volume control setting (10k load) was 5V RMS.

The spectrum of a 50Hz sine wave at 0dBFS is shown in *Fig. 5*, from DC to 1.3kHz. The calculated THD+N based on the first five harmonics is 0.0022%, with no significant harmonics and only low-level noise artifacts. The measured THD+N is 0.0027%. I repeated the spectrum analysis at 1kHz 0dBFS, as shown in *Fig. 6*, where THD+N also measured 0.0027%.

Figure 7 shows the reproduction of an undithered 16-bit/44.1kHz 1kHz sine wave at –90.31dBFS. At this level the signal consists of ±1 bit of data, producing two different voltage levels that are symmetrical about the horizontal axis (time). These discrete voltage steps are not totally obvious, due to out-of-band high-frequency noise (see the next graph).

The spectrum analysis of *Fig. 8* represents the undithered Red Book 1kHz sine wave at –90.31 dBFS from DC to 84kHz. The reason for this graph is that I wanted to determine whether the switching power supply in the ML-9600 was introducing any high-frequency noise. You can see switching power-supply components at about 38kHz and 76kHz. I used the –90.31dBFS signal to keep the analog output from muting. The noise is below –80dB and is probably of no real significance, even while using the 96kHz sample rate mode.

The distortion residual waveform for a 1kHz sine wave at 0dBFS is shown in *Fig. 9*. The upper waveform is the analog output signal, and the lower waveform is the monitor output (after the THD test set notch filter), not to scale. This distortion residual signal shows a low level of third harmonic overlaid with noise. THD+N at this test point is 0.0027%.

Figure 10 shows the spectrum response to equal level of 19kHz and 20kHz signals, each at –6dBFS, from DC to 20.8kHz. The 1kHz intermodulation difference product measures –89dB, while the 18kHz and 21kHz products are virtually nonexistent.

I viewed the response of the ML-9600 to 1kHz square wave test frequencies. The 16-bit 44.1kHz square wave in *Fig. 11* is from the CBS test disk. The 24-bit 96kHz square wave in *Fig. 12* was one I recorded from my function generator at 2.5Vpp. Both show the characteristic Gibbs Phenomenon ringing associated with the limited bandwidth produced by the digital filters. The 24-bit waveform is a better representation of the original square wave.

The ML-9600 ignored defects on the Pierre Verany Test CD# 2 out to track 30. At track 31 (1mm defect), the unit put out a series of audible clicks at each pass through the defect. The unit easily met the Red Book requirement of 0.2mm max.

I am very impressed with the ML-9600. Not only is the digital recording capability first-rate, but also the moving parts (the ones most likely to wear out) are easily obtainable computer hardware. ❖

Book Review

Audio Transducers

Reviewed by Dick Campbell

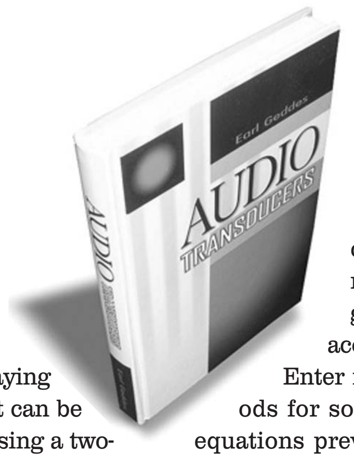
Dr. Earl R. Geddes is a world-renowned expert on all things audio, so when he decides to write a book we all pay attention. In this case, it is a book on "Audio Transducers," that thorny wicket between electrical signals and physical reality.

The story is told in 300 pages and 13 chapters, with the final chapter, on psychoacoustics, contributed by Dr. Lidia W. Lee, an accomplished audiologist. Dr. Geddes takes a bit of time at the end of the preface to explain the logic of ending a book on audio transducers with a chapter on psychoacoustics. Three cheers! It's the same reason I include a big dose of psychoacoustics in my audio course at WPI—if you do not understand how we hear things, you cannot get beyond first base in this field.

The *raison d'être* of this book is to focus on the science of transducers rather than their practical aspects. It is helpful if the reader has a working knowledge—or at least a comfortable familiarity— with partial differential equations, integral calculus, and matrix algebra. The last one mentioned in this list is absolutely essential because most of the transduction equations are expressed in this form.

The first chapter reviews many of the basic techniques of engineering analysis, such as orthogonal functions, Fourier transforms, electrical resonance, Newtonian mechanics, and coupled systems. The coupled system section is the launch pad for all of the following matrix-based equations. Here we first see the traditional equations for an electrodynamic loudspeaker driver where that troublesome stuff called "back EMF" causes a mixed-domain equation that stretches clear across my blackboard.

The central feature of my long equation is a transformer that has one side facing the electrical system and the other side facing the mechanical system. Sometimes called a gyrator, it models domain-shifting the "physical" way. The transformer ratio is Bl:1, the impedance ratio is Bl^2, the whole being terribly convenient for those experienced in circuit theory.



Now comes Dr. Geddes saying that every such domain shift can be expressed mathematically using a two-port transfer matrix, referred in this book as a "T-matrix." Domain shifting the slick mathematical way. On paper it even looks nicer than a transformer. An old, well-practiced idea, actually, but the explanations and the examples of its application in this book are made very clear and easy to understand.

Following this first example of the application of the T-matrix is a section on basic matrix manipulation leading to a sorting out of series and parallel circuit element representations. At this point it becomes clear how to "stack" matrices to solve almost any complex system.

The book proceeds onto transduction mechanisms, acoustics, and simple radiation. Then come loudspeaker enclosures both closed and ported. Ever see a closed box expressed as a T-matrix? See page 96. A ported box? See page 101. Radiation load on each side of the diaphragm? See pages 111 to 125. Each of these is elegantly expressed as a composite T-matrix.

It should be noted that matrix-based equations lend themselves to be solved rapidly by a variety of computer programs expecting matrix-structured input, such as MatLab. Also, once you get used to matrix forms of transducer equations, certain physical relationships and interactions become more obvious.

My favorite chapter covers waveguides. This is a 37-page frontal assault on horn theory and how badly wrong it can go without the correct mathematical representation of acoustic propagation confined to a flared conduit. Dr. Geddes has published journal articles on this subject before (referenced in the book). At the midpoint in this chapter are three review points of his waveguide theory including higher-order modes, wavefront geometry, and acoustic loading.

Enter now numerical methods for solving the waveguide equations previously derived. This part begins with the piecewise sectioning of the waveguide and brings in T-matrices of considerable complexity to properly deal with modal propagation. Then follows a treatment of mouth diffraction and a discussion of diffraction horns.

Having now assembled a big stack of linked T-matrices, it is straightforward to compute an accurate spatial acoustic output from a given input, namely the velocity profile at the throat of the waveguide. Can you do this backwards? Can you compute the throat velocity profile required for a given set of radiation conditions? Of course, since the T-matrix stack members are bidirectional. This is a sure-fire way to get the optimal phase plug design.

The short chapter on crossovers is revealing due to the effective use of polar maps of angular radiation data due to multiple drivers. This plotting technique is introduced earlier in the book to illustrate beam-width as a function of wavelength, related to the size of the radiator (ka product). These plots are like slices of conventional line polars but stacked vertically, run out on the horizontal axis as a function of ka, then shaded as a function of beam-width level slices. In the crossover chapter, we see, typically, five 6dB shadings from on-axis white (0dB) to essentially null regions off-axis in black (more than 30dB down).

Since a passive crossover is easily represented in matrix form, placing one into the loudspeaker system matrix stack means you can then correctly use acoustic responses as one or more poles of the overall desired crossover response. Thus ducts and fabric layers may be brought into play, or even the driver response itself. This can be a risky business if you do not fully characterize the out-of-band system response, but at least this method is potentially the most accurate. As the author concludes, "Crossover filters must be matched to the specific set of drivers being used . . . simply using electrical filter concepts does not work."

The chapter on room acoustics deals with small rooms and primarily their modal behavior. There is a short treatment of the transition from modal to statistical behavior, the need for directivity as a control element, and the influence of wall impedance on modal behavior. Finally, there is a rather complete discussion on source placement in rooms.

The chapter on transducer arrays is an extensive discussion of multiple driver topology with several examples and many polar maps. The conclusion on the radiation pattern for arrays using wavefront curvature is non-intuitive and against popular belief, but like revealing the end of a movie, I will not tell you what it is here!

Next come 22 pages on the subject of distortion providing a detailed discussion of loudspeaker component nonlinearity and the extensive mathematical treatment that goes along with the subsequent analysis. Once again, the psychoacoustic background for the audibility of loudspeaker distortion is clearly written to the credit of Drs. Geddes and Lee. This is the part that is so important and so frequently left out.

There are stated "Perception Principles" that, when applied to reality, give rise to new hypotheses for a new approach to quantifying nonlinearity. Signal-based distortion metrics become irrelevant. Of course, in the case of loudspeakers, the whole subject must include a discussion of thermal compression that closes the chapter.

There is a wonderful short chapter on microphones in which T-matrices once again appear. Subjects include noise, enclosures, pattern control, and arrays. The point of this chapter is to illustrate that much of the same mathematical treatment that is used for loudspeakers is directly applicable to microphones.

The penultimate chapter has to do with measurements and begins with a mild disclaimer that maybe this subject does not belong here. Quickly, the prospect of discussing actual measurements with actual instruments is dispelled. No, this starts with the philosophy of measurement and how we can figure out what is important to measure in the first place. In electrical engineering, we can start with Maxwell's equations and get right down to E = IR by successively tossing out "unimportant" terms. Likewise, we can start with the complete T-matrix of a loudspeaker system, expand it into an impressive algebraic equation, and simplify by tossing out "unimportant" terms with the goal to reveal what should be measured.

It is a process of deconstructing a total, complete measurement of everything into a series of useful incremental measurements that can be reassembled into the whole. This process includes the technique of parameterization. The actual measurements that are required to accomplish this goal are given in detail. The chapter concludes with short discussions on measuring component nonlinearity and thermal parameters.

Finally, ear physiology, masking, loudness, and binaural hearing are the subjects of the last chapter on psychoacoustics. This is a quantitative treatment that reveals what research audiologists think about. The last figure in the book is our old friend, a polar map, of IID (Interaural Intensity Difference) around the human head.

The bibliography is doubly useful because many of the entries are noted below with remarks about the content and usefulness of the reference.

This is a fabulous book and no one—repeat, no one—in this business should be without a copy on their bookshelf. Drs. Geddes and Lee have made a major contribution to our art. With some expansion (glossary, table of variables, problem/solution examples) this book could be an excellent textbook in teaching electroacoustic engineering.❖

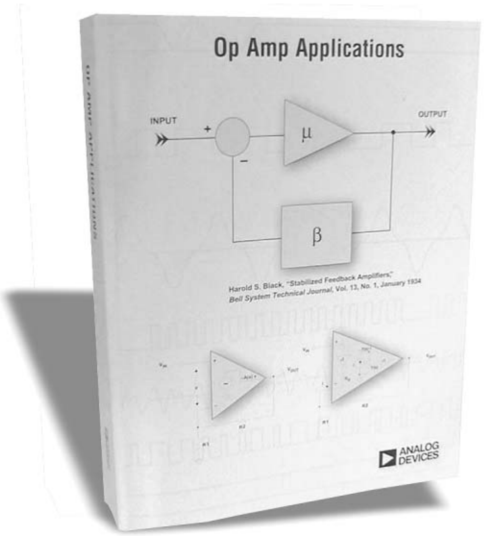
Audio Transducers is available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 603-924-9464, Fax: 603-924-9467 e-mail custserv@audioXpress.com, BKGL1.

Dick Campbell teaches acoustics and audio engineering at Worcester Polytechnic Institute and operates an active consultancy in audio and loudspeaker system design.

Book Review

Op Amp Applications

Reviewed by Charles Hansen



Op Amp Applications, *edited by Walt Jung, Analog Devices, Inc., July 2002, ISBN 0-916550-26-5, 960 pp., $40 plus S&H*

This eight-chapter, 960-page soft cover book was produced for the Analog Devices 2002 Amplifier Seminar North American and European tours. It is now available through ADI's distributors (Arrow, Avnet, Future, and others). You can obtain further ordering information at 1-800-262-5643 or at www.analog.com/seminar. The order code is OP-AMP-APPLIC-BOOK.

The book was edited by Walt Jung, and compiled with the input from many other ADI engineers: James Bryant, Joe Buxton, Wes Freeman, Paul Hendriks, Walt Kester, Charles Kitchin, Scott Wurcer, and Hank Zumbahlen. It is loaded with application examples, and gives performance tables and selection charts for the ADI op amps recommended for many applications.

The purely "audio" portion of this book, while substantial, is really only a small portion of the total. But the breadth of detail, amplifier types, and the application content should interest even those who only dabble in audio electronics.

HISTORY

Chapter H is an extensive history of the technology that led to the development of the op amp. It's a fascinating path from the two-element Fleming diode that evolved from Edison's filament electric lamp (1904), to Lee De Forest's "audion" triode (1906), which provided the means for amplification, to Harold Black's 1934 Bell Systems Journal "Stabilized Feedback Amplifiers" paper that led to the first phase of vacuum tube op amp development in the early 1940s. The term "operational amplifier" actually came after World War II, in 1947.

Technology marched forward at a rapid pace after the war. Vacuum tube op amps evolved from the early amplifiers. Analog computing was developed further, often using more accurate chopper-stabilized amplifiers, and the first true vacuum tube op amp products; many based on the Bell Labs M9 amplifier, and then the first commercial op amp, the PHILBRICK K2-W.

With the invention of the germanium transistor in 1947, and later the much improved silicon variety, solid-state versions of the op amp came into being. The first devices were discrete modular transistor designs, then hybrid packages. Jack Kilby and Robert Noyce independently invented the integrated circuit, and Bob Widlar designed the first monolithic op amp, the Fairchild μA702.

The history continues from Widlar's famous μA709 to the modern low-noise, high-performance, high-speed op amps we see today. I don't want to diminish the technical content but, to me, this chapter alone is worth the price of the book.

CONTENT BREAKDOWN

Chapter 1 is entitled Op Amp Basics. It starts with the ideal op amp and generalized feedback circuits: the non-inverting amp and follower, the inverting amp, and the differential amp. Next, it looks at non-ideal static errors and common-mode dynamics. Power supplies play an important part in op amp design, so both single and dual supply issues are discussed. Finally, there is a device selection matrix, showing the decision drivers that can influence the priorities when selecting an op amp (*Table 1*).

The next topic is the topologies of voltage feedback (VFB) and current feedback (CFB) op amps, with an interesting example of CFB using vacuum tubes. The structure of the monolithic op amp starts with single-supply issues. Input stages are well covered: bipolar inputs and bias compensation, super-beta, and JFET. A discussion of rail-to-rail input requirements, and input overdrive/overvoltage (OV) protection follows.

Output stages include NPN, NMOS, complementary-bipolar (CB), "almost rail-to-rail" bipolar junction transistor (BJT), and CMOS. The important practical considerations of open-loop impedance, output surge protection, and the offset voltage trim processes typify the depth of coverage that is found in this book. Finally, there is a look at the op amp process technologies developed by ADI.

One look at the data sheets from various manufacturers shows that a detailed discussion of op amp specifications is absolutely necessary for the poor design engineer, since the significance of any

TABLE 1

FUNCTION	PERFORMANCE	PACKAGE	MARKET
Single, Dual, Quad	Precision	Type	Cost
Single or Dual Supply	Speed	Size	Availability
Supply Voltage	Distortion, Noise	Footprint	
	Low Bias Current		
	Power Dissipation		

spec parameter is highly dependent on the final application. The parameters covered in this section include:

Input offset voltage V_{OS}
Offset adjustment (internal and external)
Drift and aging
Input bias current I_B
Bias current cancellation and total error calculation
Input impedance for VFB and CFB
Noise and signal gain
Open-loop gain and its non-linearity
Frequency response
Slew rate
Full-power bandwidth (FPBW)
Settling time
VFB gain-bandwidth product (GBW)
CFB optimum feedback resistance
Bandwidth flatness
Input noise current (i_n) and voltage (e_n)
Source impedance and noise
Frequency characteristics of noise (1/f, popcorn, rms noise over bandwidth)
Total noise calculations
Noise effect of resistance and reactance elements
Distortion, total harmonic distortion (THD), and THD+noise (THD+N)
Common-mode rejection ratio (CMRR) and power supply rejection ratio (PSRR)
Power supply decoupling
Power supply voltage and power dissipation (Pdiss)

The next section delves into precision op amps: a definition of their characteristics, an error budget analysis, chopper stabilization, and noise considerations.

From here, the book moves to high-speed op amps: VFB bandwidth and slew rate calculations, the folded cascode topology, and the ADI current-on-demand and quad core g_m topology. The CFB is also covered, with the effects of feedback caps, and noise comparisons between VFB and CFB circuits.

KINDS OF AMPS

Chapter 2 covers Specialty Amplifiers, starting with instrument amps (in amps). The authors present the definition of the in amp, difference amps, and the 2-op amp and 3-op amp configurations. Next come the details, such as single-supply considerations, error sources, noise sources, bridge amplifier error budgets, performance tables, and a discussion of input OV protection. Finally, there is a generous selection of applications.

Programmable gain amplifiers (PGA) are next, with design issues, the effects of gain switch Ron, differential input PGAs, and some application examples. For applications where galvanic isolation is important, such as the medical electronics field, the next section on isolation amps is a must. There is a discussion of the various methods for analog isolation (transformer, optical, capacitive coupled) and digital isolation (transformer, optical).

Chapter 3 explains how to use op amps with data converters. The introduction talks about data sampling techniques, applications, and trends. An op amp selection criteria table is included. The first section presents analog-digital converter (ADC) and digital-analog converter (DAC) specs. The pertinent topics are transfer functions and DC errors, quantization noise, ADC input-referred noise, calculation of output noise compared with ADC input-referred noise, converter dynamic performance; signal-noise and distortion ratio (SINAD), signal-noise ratio (SNR), and effective number of bits (ENOB); analog BW, aliasing, harmonic distortion, worst harmonics, THD and THD+N, spurious-free dynamic range (SFDR), and two-tone intermodulation distortion (IMD).

Next is a section on driving ADC inputs, with the key op amp specs and requirements. There is a discussion of high-resolution sigma-delta (Σ-Δ) measurement ADCs, multiplexed (MUX) data acquisition systems, single-supply ADCs with scaled inputs, buffered input and buffered differential ADCs, switched-capacitor CMOS ADCs, single-ended (SE) ADC drive circuits, DC coupled gain setting and level shifting, and differential input ADCs. Finally, there is the important topic of input overload/OV considerations.

Every DAC or ADC needs a conversion reference, so reference stability is an important topic. This section covers ADC/DAC reference input drivers and successive-approximation (SAR) ADC references. The final section in this chapter involves DAC outputs. The general considerations are presented first, followed by differential to SE conversion techniques, SE current-to-voltage (I-V) conversion, differential I-V conversion, and low pass (LP) filters for audio DACs.

Chapter 4 covers sensor signal conditioning. The introduction encompasses sensor characteristics, output formats, and the process control loop. Treatment of the bridge circuit involves amplification and linearization of bridge outputs, driving remote bridge circuits, minimizing system offset, and AC and DC excitation. Next come strain, force, pressure, and flow measurements and their signal conditioning. High-impedance sensors include photodiodes, charge outputs sensors (piezo, accelerometers, hydrophones), pH monitors, and chemical and smoke detectors. Finally, we find the temperature sensors: linearizing techniques, thermocouple cold junction compensation, platinum RTDs, thermistors, and semiconductor sensors.

FILTERS

Chapter 5 presents analog filter circuits. The introduction covers the idealized filter response, the transfer function, and key filter parameters. Next we delve into the S-plane, pole-zero response, f_0, and Q. Applicable filter types are low pass (LP), high pass (HP), bandpass (BP), band reject (BR) or notch, and all-pass (AP). The third section covers phase response, time domain response, and impulse and step response.

Section 4 includes the standard responses: Butterworth, Chebyshev, Bessel, linear phase, transitional, Gaussian, all-pole, elliptical, maximally-flat delay, and inverse Chebyshev. There are 11 pages of response curves and 11 design tables for the various filters. Next up are frequency transformations: LP-HP, LP-BP, LP-BR, and LP-AP.

Section 6 contains real filter realizations: single pole (SP) passive RC and SP active, passive LC, the integrator, general impedance converter, active inductor, frequency-dependent negative resistor (FDNR), Sallen-Key (S-K) or voltage-controlled voltage source (VCVS) LP, HP, and BP; multi-feedback (MFB) LP, HP, BP; the state variable filter, the biquadratic (biquad), the dual-amp bandpass (DABP), the twin-T notch, the Bainter notch, the Boctor notch, the 1-BP notch, and the first- and second-order AP. There are 19 pages of design equations for all of these filters.

Section 7 covers practical problems in filter implementation, beginning with passive components (R, L, and C). There are capacitor and resistor parameter comparison charts, and a discussion of the limits of op amps; gain variations with frequency, GBW requirements, and input and output impedance. Next, the book looks at distortion from input capacitance modulation in FET op amps, and Q enhancement. The final section has a number of filter design examples. Of interest to audio types are the fifth-order anti-aliasing filters (passive LC, S-K, MFB, state variable, and FDNR), and the seventh-order CD DAC reconstruction filters (passive LC, FDNR).

Chapter 6, which is the longest chapter in the book at 210 pages, presents signal amplifiers. The first section covers audio amplifiers. First are the microphone preamps; single and differential input solid-state, electret mike amps, and transformer-coupled mike amps. The RIAA phono preamp is presented next, with the RIAA idealized response. The two RIAA equalization (EQ) networks, "N1" and "N2," are shown with a discussion on the sensitivity to device tolerances, parasitic capacitance, and resistor and capacitor parameters. The phono preamp topologies presented in this section are the active feedback preamp (AC and DC coupled), and the passive EQ preamp.

The book covers line stages next. Line receivers are shown in balanced, SE, and transformer-coupled configurations. The line drivers that the authors present are balanced, SE, current-boosted composite amps, inverter-follower differential drivers, cross-coupled, transformer output and mixed-feedback transformer output types. The topic for Section 2 is buffer amplifiers, and driving capacitive loads and the types of compensation required to do so.

APPLICATIONS

The third section details video amps. The intro covers video signals and specifications for NTSC and PAL. Video amp design considerations look at BW, signal transmission, and line termination. Video line driver and distribution amplifier driver topologies include the inverter-follower, cross-coupled differential amp, and the fully integrated differential amp.

Video line receiver topologies include the four-resistor differential receiver and the active FB differential receiver. Some design examples are the cable-tap (through-loop) amp, high-speed clamping amp, high-speed video multiplexing, and single-supply video amps. Communications amplifiers come next, with noise and distortion specs, automatic gain control (AGC), voltage-controlled amplifiers (VCA), and two application examples: a CATV upstream driver and an xDSL upstream driver.

Section 5 presents miscellaneous amplifier ideas and applications: a high-efficiency video line driver, wideband noise generator, precision rectifiers, parallel amps for low noise, a pulse generator, clamp amps, a sync inserter, and a negative resistance buffer.

The final section of this chapter discusses the composite amplifier; the two-op amp composite, a chopper-stabilized amp, voltage-boosted rail-rail output, and gain-boosted input composite amps. The final amplifier is a Walt Jung-designed "Nostalgia" vacuum tube input/output composite amp with a discrete transistor folded-cascode, cascade current-mirror in between the input and output tube stages.

HARDWARE

Chapter 7 is a very practical presentation of Hardware and Housekeeping Techniques. Section 1 begins with passive components and their comparison charts again. The parameters of capacitors include dielectric absorption (DA), parasitic elements, dissipation factor (DF), tolerance, temperature variations, and other effects. Resistor topics include temperature coefficient (TC), parasitic elements, thermoelectric effects, voltage sensitivity, aging, excess noise, and a discussion on the use of trimpots. Stray and mutual inductance, ringing, parasitic elements, and quality factor (Q) are discussed, plus, inductors finally get covered in-depth.

Section 2 covers the otherwise obscure printed circuit (PC) board design issues; conductor resistance, kelvin feedback, return currents, ground noise/loops, grounding (star, analog, digital), ground planes, skin effect, ground isolation, static PC board effects, leakage, and the use of input guard tracks on PC boards.

Section 3 covers the all-important topic of op amp power supplies (PS). The authors investigate fixed and adjustable linear IC regulators, low-dropout regulators; charge-pump voltage converters, inverters, and doublers; and linear post-regulators for switching power supplies. Next comes PS noise reduction and filtering. Pertinent topics include filter caps, inductors, ferrites, resistors, post-regulators, bypass and decoupling caps; grounding and component separation consideration, and PC board layout.

Section 4 looks at op amp input protection: OV, common-mode limitations, clamp protection, clamp diode leakage, CMOS channel protection, high common-mode input instrument amps, and inverting amplifier protection. Then the section covers the output phase reversal problem, and input differential protection and series resistance. The last topic in this section is electrostatic discharge (ESD) protection and testing. Section 5 investigates thermal considerations. The intro looks at thermal basics and follows with the concept of thermal resistance (θ) as it pertains to cooling and heatsinking.

Section 6 covers electromagnetic interference (EMI) and radio frequency interference (RFI) mechanisms. It looks at EMI/RFI noise sources, coupling paths, passive filters, shielding, cables and their shields, and ground loop contributions. Then it covers RFI rectification sensitivity in emitter-coupled and source-coupled op amp input stages. There is a discussion of reducing this sensitivity in op amp and instrument amp inputs. The section also looks at output stage EMI/RFI protection, PC board design, logic level protection, and gets into microstrip and transmission line termination.

Section 7 wraps up the book with Simulation, Breadboarding and Prototyping. The first topic is analog circuit simulation with SPICE and IC SPICE models, and macro models vs. micro models. ADI's ADSPICE op amp macro models are discussed in depth: input and gain/pole stages, frequency-shaping stages, output stages, model transient response, and noise models. The book considers CFB amplifier models and ways to deal with PC board parasitics.

Under breadboarding and prototyping, the book examines the "deadbug" and solder-mount breadboard methods, and milled prototype PC boards. There is a discussion of sockets, DIP and SOIC packages, and evaluation op amp boards (general purpose and dedicated). This section on real-world hardware issues is invaluable. No book is complete without a thorough index, and this one covers 45 pages, indexing both key words and ADI device numbers.

CONCLUSION

I found this to be a fascinating and technically valuable book. It has been thoroughly researched, fully footnoted with lots of references, and is written in an interesting and engaging style. *Op Amp Applications* is a must-have for anyone interested in the history of technology and/or the practical application of op amps over the proverbial range of "DC to daylight." ❖

Ask *aX*

FORMULA SEARCH

I understand that there is a formula that defines the size of a chamber required to couple a loudspeaker drive unit to a horn which involves drive unit parameters. Can somebody tell me what the formula is?

Also, in Augspurger's part 3 article on transmission lines (*SB* 4/00), he mentions an R/C rolloff produced by the springiness of the air in the chamber. Are there any formulas that can calculate this?

David Field
difield@lineone.net

G.L. Augspurger responds:

Mr. Field's questions deserve a little discussion about coupling chambers in general. Let's take a look at four types of systems:

1. High Frequency Front-Loading Horn

Figure 1 *is a simplified diagram of a front-loading horn. Assume that the horn is large enough to minimize mouth reflections above its flare cutoff frequency. Consequently, the acoustic load at the horn throat is largely resistive. You are concerned with chamber A that couples the driver diaphragm to the horn throat. Why bother with a coupling chamber when a simple dome tweeter mounted on a waveguide can deliver excellent response? The answer is efficiency.*

Theoretically, the coupling chamber is not necessary. If you could drive a nearly weightless diaphragm with a super-efficient motor, then electric power could be converted to acoustic output at about 50% efficiency. However, to achieve maximum efficiency using practical technology, the coupling chamber is a necessary evil.

Figure 2 *shows the response of a TAD 2″ diaphragm driver on a straight round horn. The horn has a 350Hz flare cutoff and a mouth diameter of about 7″. It is mounted on an infinite baffle. The upper (bold) curve represents the commercial driver, which has a coupling ratio of about 8:1. That is, the area of the actual horn throat, which is buried inside the driver, is one-eighth that of the diaphragm.*

The lower curve shows what would happen if you shortened the horn to provide a 2″ diameter throat and mounted the diaphragm directly to the horn. Conversion efficiency drops from about 25% to less than 10%. Note that both curves represent acoustic power response. On-axis response will rise substantially above 1kHz as the horn becomes more and more directional.

How large should the coupling chamber be? Assuming that the trapped air acts purely like a spring, then two effects can be analyzed:

First, the air spring resonates with the mass of the diaphragm, resulting in a response peak at the resonance frequency. I found that Olson's classic text Acoustical Engineering *devotes almost two pages to this subject, explaining in great detail how the coupling chamber volume can be chosen to ". . . thereby increase the efficiency over a wide range."*

The problem is that it doesn't work very well. In almost any practical horn system, the peak is so narrow and occurs at such a high frequency that it is useless unless a number of other parasitic resonances are bundled into the design.

Second, since the resistive horn throat is driven through the coupler air spring, an acoustic high-cut filter is formed. You can calculate the break frequency in Hz from this simple formula

$$f = c/2\pi hT$$

where c = the speed of sound, h = the height of the coupling chamber, and T = the cou-

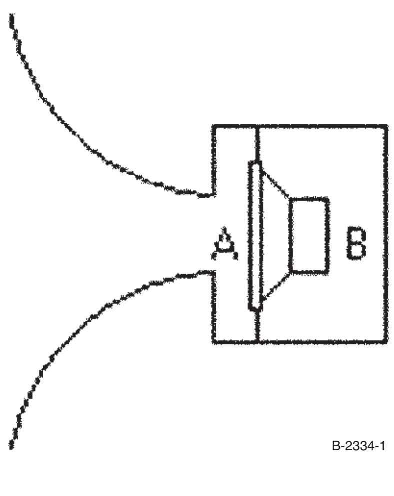


FIGURE 1: Front-loading horn.

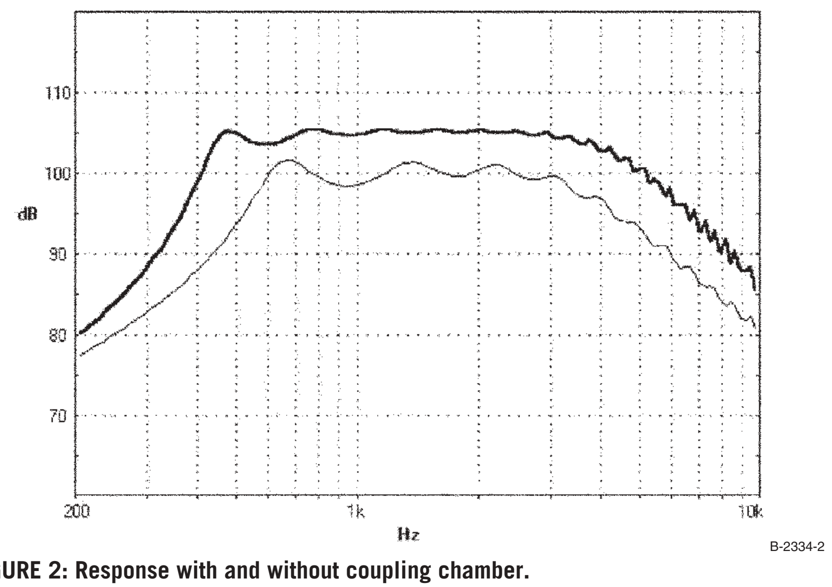


FIGURE 2: Response with and without coupling chamber.

pling ratio (in this case, the ratio of coupler cross-sectional area to throat area). The formula works with any consistent units of measurement, i.e., meters and meters/sec, feet and feet/sec, or inches and inches/sec. A typical high-frequency driver might have a 10:1 compression ratio and a coupling chamber height of 0.02", resulting in a break frequency of 10.7kHz. Reducing the height of the chamber will push the cutoff still higher, but may not allow enough clearance for maximum diaphragm excursion.

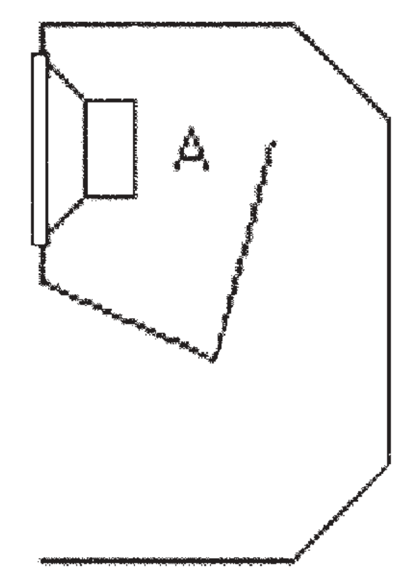


FIGURE 3: Rear-loading horn.

2. Low-Frequency Front-Loading Horn

Although the preceding comments also apply to low-frequency designs, their usefulness is limited. Practical bass horns are simply not big enough to control mouth reflections. They behave more like tuned pipes than true horns, and their throat impedance is not very resistive.

This applies even to the mighty Klipschorn, which has the advantage of full corner loading. However, in one version of the K-horn, Paul Klipsch did introduce a high-frequency resonance by making the exit of the coupling chamber smaller than the horn throat. Computer modeling verifies that a little extra gain in the 500Hz region was achieved.

3. Low-Frequency Rear-Loading Horn

By eliminating the rear chamber and allowing the speaker to radiate freely, you arrive at the design of Fig. 3. *In effect, delayed rear radiation is added to front radiation and the result is a messy comb filter. However, if the horn is big enough, the comb filter shows up as one deep notch with reasonably flat response above and below. The old JBL dual-woofer C55 is a good example.*

Alternatively, if you make the coupling chamber relatively large and set the horn length correctly, then the system behaves like a vented box that is mostly vent and very little box. Jensen used this "reactance annulling" technique in its 1954 Laboratory Reference system. Low-frequency performance was good for its time, but not by today's standards.

4. Transmission Line

Yes, a damped transmission line (resistive load) driven through a coupling chamber (air spring) benefits from the resulting acoustic high-cut filter. And yes, you can use the coupling chamber formula to calculate the filter break frequency, but not very accurately.

One problem is that throat impedance may differ from the characteristic impedance of air. Another is that you can't extrapolate performance from that of an equivalent straight pipe because the speaker is damped differently and behaves differently. A safer approach is to base your design on one of the transmission line alignments published in part 3 of my article. ❖

XpressMail

SUB HUNGER

As an audio enthusiast, I go back several decades, and was weaned on glass "valves." I have found several of the articles in your magazine over the years quite interesting, but my reaction to "A Hegeman Subwoofer" (12/03, p. 12) is the first one that really grabbed me and won't let go. I really enjoy elegant solutions to real-world problems, and this one is unique.

I have been trying to brute-force the bass smoothness with electronic crossovers, custom equalization, and brute power in my system using old subs with electrostats. This has worked OK to date, but, although 1/4 wave stubs are a bit complex to build, they are a much more natural solution. I have been ready to upgrade to more modern and more linear drivers in my quest to ripple rugs just a little anyhow.

I followed your references to find out more, and found out about Don Morrison and his website—yet another source of glee. I learned several things just from perusing his website. He preaches the same gospel I do, but with much more humor. And I thought I was an island surrounded by the golden ears of the world. Hooray!

Anyhow, I still wind up with a thirst for more design info. I figured out that the stub/tunnel lengths were measured along the center of the tunnel path around the folds/corners. I also noticed the choice of about a 10 inch^2 area for the tunnel cross-section. I assume this is not arbitrary. Is there a relationship between the speaker cone area and the total port area? Or is there some other reason?

I figured out that the tunnel resonances were spaced at square root of 2 frequency multiples, which made sense to me. I wondered why there were four tunnels. Why not three or five? I think I know why you don't use just one. I learned that the original Hegeman had six. Ouch!

And what determines the frequencies of choice for the tunnels? I assume it ties in to speaker resonance, but how? Does the plenum/rear chamber volume affect/raise the Fs as most enclosures do? Did the 12″ speaker boxes have the same tunnel frequencies and cross-section areas? I notice the Fs of the speakers you chose is 28Hz. And I noticed the fairly high Qs in free air, too. I assume the multiple tunnels and port tuning prevent the need to match a single tunnel design to a specific speaker.

I would love to build one of these for a Brahma 10″ for my fairly large family room, but I have chosen to stay away from foam surrounds for longevity's sake. And the Brahma, although a great-looking driver with impressive distortion results, seems to be out of production now (although there are still a few left). I had planned for two 10″ long-throw sub cabinets, but that cabinet seems a bit much to make two of. I suppose once I commit to building one, the other one is almost free. Two drivers also allow a stereo power amp, which is easy to obtain. I still need to look up and check out the other speakers you mentioned—such as the Audiomobile.

One reason for all the questions is that I had chosen 10″ long-throw drivers for their quickness and relative rigidity over 12″ and larger to meld better with my electrostats. I was heading toward sealed boxes again, but am wondering whether minor modifications to your layout might make more sense. Decent sound level at very low frequencies requires either cone area or cone excursion. I wonder how a pair of 10″ drivers would work in this cabinet.

I notice the tunnel openings are directly behind the driver cone. How important is this? You mention coupling of the tunnels to the driver. And how important is the plenum volume? I am also thinking about some dado cuts in the (slightly thicker!) separator and side walls for alignment of tunnel walls and strength. It simplifies the jig problem, and this doesn't seem to be a good application for biscuits because of the thin walls. You just used glue, which sounded adequate. With all those tunnel walls, the cabinet must be pretty stiff and low on resonances. But I bet they are heavy, too.

You entice us with comments about what you learned by building some 12″ and a few 10″ cabinets, but didn't expand on it. I am hungry for more! What would you do differently next time?

Thanks for a great article on an exciting concept. I think I'm hooked. It may be a busy winter.

Bill Irwin
Audio Clinic
Geneseo, N.Y.

Cornelius Morton responds:

Glad that you enjoyed the article. I will try to answer your questions in order, but first a comment on the Brahma 10″ sub. It is still in production as the Mark II, same as the original but with a few improvements, see http://www.acousticvisions.com/~acoustic/products/subwoofer_drivers/adire_brahma/ for availability. As of mid-December, they are available at the same price. The high-density foam surround feels more like rubber than foam and should last a very long time. Your comments on 10″ subs were my reasons for going the 10″ route. The extreme linearity and Xmax of 27mm one way of the Brahma 10 convinced me to try it, and the resulting sound requires my recommendation. Pumping-wise the Brahma ten should hold a 90dBi SPL to about 18Hz as it pumps 1620cm^3 per stroke.

I frequent the Don Morrison website for a breath of sanity; talking to him is even more refreshing and he does live by his Sound Rules.

The pipe lengths are measured along the centerline of the pipe to the center of the stops, which are angled at 30° to slightly broaden the pipe response. The felt facing is used to damp any high-frequency trash that might be generated due to hf cone resonance and such due to speaker construction or crossover leakage.

The total mouth area of the four pipes is a compromise between cabinet size and area. Drivers with long Xmax and high Qms *seem to like larger areas. A value of 80% of the effective cone area for the total pipe area seems good in these cases. I have gone as low as 50% for short-throw wide-range drivers.*

The frequency spacing of the pipes is at a square root of two interval and allows a pipe Q of around 3. This spacing also allows the first and second pipes to aid pipes three and four, as pipe three is the second harmonic of pipe one and pipe four is the second harmonic of pipe two. A pipe with a length of one-half wavelength closed at one end appears to be closed at both ends.

As these are low Q pipes, the closure is only partial, probably around 50%, but even a 50% reduction of the volume of pipe one or two improves the efficiency of pipe three or four. Similarly, when pipe four is active, the activating frequency is the third harmonic of pipe one or an odd number of quarter wavelengths for pipe one, which boosts the tuning efficiency of pipe four.

Additionally, pipe two acts as a phantom pipe at its third harmonic or 90Hz. This can be seen as a fifth bump when measuring impedance before adding the stuffing. All of this interaction can make the initial tuning of this contraption a bit interesting. Tuning can look from bad to awful and getting worse until you try one more thing. At about this time your arms are probably ready to fall off from lifting the driver in and out of the cabinet—the Brahma ten weighs 38 lb! As for the number of pipes, four at the spacing used will cover the useful band of the sub, providing a rather level impedance across the band.

In choosing the pipe frequencies, I like to set the second pipe so that Fs is in or near the low-frequency edge of the band for the pipe. As for Fs changing due to cabinet volume, in a standard enclosed or vented cabinet the Fs change is due to the reduction of compliance—mechanical capacitance—of the suspension. In this cabinet, the trapped air acts like a resistance in series with the cone instead of a spring and seems to have little effect on Fs.

The 12″ units had a low-frequency tunnel tuned to 18.5Hz and a tunnel cross section of 3.25″ by 4.5″ for a tunnel area of 14.625 in^2 and a total of 58.5 in^2. They were also about 25% larger in each dimension.

About a stereo pair: if your electrostatics will work with a low-frequency crossover of 100Hz or less, a stereo pair of subs is not needed. Summing the low-pass outputs of a low-level crossover to drive the sub amp will be fine. The Brahma ten will put out an SPL of 93dBi at 25Hz and over 95dBi at 35Hz, all at 1m. Another 3dB may be added to these figures when the subwoofer is sitting on the floor, or near the floor, due to it firing in to half space; then the room effect will probably add another dB or so to that.

This takes a bit of power. Adire lists a 700W requirement for full excursion in a large box. A friend has an Ashley theater amp that we ran in the bridged mode—800W. We managed to clip the amp without bottoming the driver—a 2″ peak-to-peak excursion is an awesome sight—and it quite adequately energized his 3600 ft^3 listening room!

You could put two 10″ drivers in one box, but the throat area for each tunnel would be doubled or two sets of tunnels made. The box will be a bit unwieldy and ugly, I fear. If two drivers are needed, I suggest two boxes.

The plenum needs to be large enough to accept the driver, but not so large in volume as to reduce the efficiency of the pipes. Placing the pipe throats directly behind the driver ensures that each pipe affects the entirety of the cone evenly.

I also considered dado cuts for wall placement. That would require seven cuts per side that were precisely aligned for six sides if both edges of the walls are to be trapped, or three sides if only one edge is trapped. But the wall alignment is not the critical one; the alignment of the side edges and separator edges are critical when putting on the front, back, top, and bottom.

Small variations in pipe area caused by alignment variations of up to ⅛″ are undetectable, while a variation of 1⁄32″ in edge alignment will be very frustrating while it is being either shaved off or built up to match the rest—especially when you know the opposing side will also require adjustment. That is what the alignment jig shown minimizes. Biscuits would be fine for the front, back, top, and bottom, but by that time I am usually just anxious to finish, so out come the glue and weights!

These boxes are basically honeycomb structures with four built-in resonators that are excitable at the open throat; the walls do not effectively excite them. Rapping on the sides results in the sound of a solid block except in the plenum area, which has a resonant frequency that is well above the operating range of the sub and is damped with fiberfill. These are about as dead a cabinet as can be found. The finished cabinet-less driver weighs 100 lb, with another 38 lb for the driver.

If you estimate the moving mass, cone, voice coil, and suspension as 100 grams, the inert mass would be 637 times the moving mass. Laying the unit down on a frictionless surface and driving to the full excursion of 54mm peak-to-peak then, the unit would move about 0.084mm peak-to-peak. Normal surface friction would reduce that further, or mounting it to fire upward, then the kick would try to move the earth. In either case no exotic coupling to the floor is needed, and spikes would mark a concrete floor and tear up a wooden one!

In addition to the 10 and 12″ units, I have built an 8″ unit using the Audiobahn AW800Q driver. While it sounds better than conventional reason would expect, it is down –8dB at 50Hz when driven with a constant amplitude signal. The rolloff is very smooth and the system Q is about 0.5, which contributes to the sound quality for bass strings and so on. The Xmax is only around 12mm one way, which, when combined with a piston area of 190cm^2, really limits the low end.

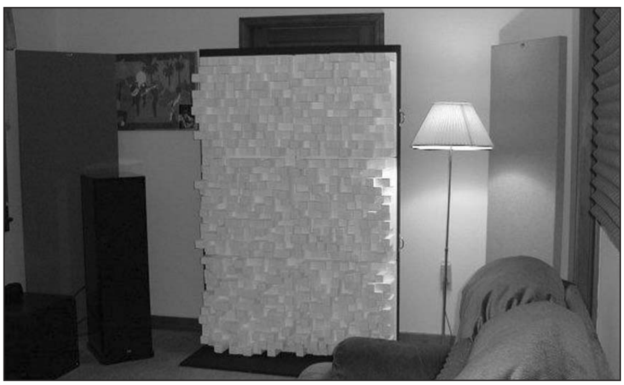
But when combined with the popular small speakers, 5 or 6″ midrange, and 1″ domed tweeters—stand mounted—it provides a very pleasing low-range extension. As the dimensions are about 75% of the 10″ unit, it is more easily hidden.

Note: All SPL calculations are based on pumping 214cm^3 at 50Hz to achieve a 90dBi SPL as noted on S. Linkwitz's excellent website at http://www.linkwitzlab.com/.

Hope that this helps. Thanks for your interest.

AUDIO PARADISE

At the conclusion of Stephen Harner's article, "Acoustical Shrinkage of Large into Small Rooms" (*aX* 7/03), Mr. Harner mentions in the last sentence (p. 31) that he has a unique solution to the standing wave problem. I believe that readers like myself would be very much interested in reading such an article.



Kindly pass on to Mr. Harner that, after reading his article, I significantly altered my plans for the acoustical treatment of my listening room by including a considerable amount of diffusion absorption. The accompanying photos show the listening room setup. I have not yet installed the ceiling diffusers.

Ray Segura
New Orleans, La.

Stephen Harner responds:

Thank you for your comments. I am submitting a second article about the standing wave paradox for publication. Regarding the alterations to your listening room, marvelous!

Your room does illuminate the double-edged sword of acoustical treatments. The upside is the wonderful sound-field that your room obviously presents, while the downside is the problems with aesthetics and ergonomics. It appears that your doors are partially blocked by the diffusers. For such audacity, I applaud you.

As far as standing waves are concerned, you might try placing your two subwoofers in front of your two wall diffusers. If the two subwoofers are in phase, they will cancel out the lowest (left/right) resonance, while the central placement from front to back will eliminate the second resonance.

Thank you for this glimpse into your audio paradise.

FEEDBACK EFFECTS

Regarding Dennis Colin's balanced primary feedback (12/03, p. 24): I've rocked down that exact road for all those same reasons and it sounded terrible at high output in spite of weighing 85 lb. I can only guess that the push and pull stop working together past class-A, giving a very crispy-crunchy broken glass sound. Try putting valve cut-off in a simulator! Whatever. Dennis has done a very thorough "small signal analysis" (and I have had all his waveforms on my test bench in the past), but when he turns it up on music there will be a very loud disappointment.

The only solution—against my will—was a zobel network across the primary to correct the phase shift due to leakage inductance and damp the ringing at its source, which feedback just cannot do because the phase has shifted. Mine needed 1.1nF/1kV series 1.5k/10W, which doesn't suck power. The feedback resistor from the secondary is compensated as usual (e.g., 470pF/5.6k), but the cleanest sound definitely resulted from no first-stage-plate-compensation because that reduces hf loop gain. And, yes, you do need an output secondary R/C to ensure no-load stability. Mine needed 47nF series 47Ω/10W, which doesn't suck power.

I've been building tube amps since 1958 (35mW), and my final "Colossus" (50W) is giving the sharpest, cleanest, most attacking tops ever, even right into grid current. (Don't you love pushing it?) But Dennis should try his for himself of course. That's why we live.

Jim Carlyle
chrisjim@e3.net.nz

Dennis Colin responds:

Jim, your letter comes at a most welcome time—I've just completed the amp. Before any audition, I spent six months resolving the very issues you described, which were visible on the scope. First, some changes: I used the Plitron PAT-4142-00 toroid transformer, at half its nominal Z and with its separable primary leads rewired for 67% UL (nearly triode but more power). Second, I returned the primary F8 to the input 12AX7 plates, not cathodes.

The issues:

1. *Tuned Zobel needed to damp this transformer's 400kHz parallel resonance Z peak. One Zobel across full primary was NG; common-mode resonance allowed by*

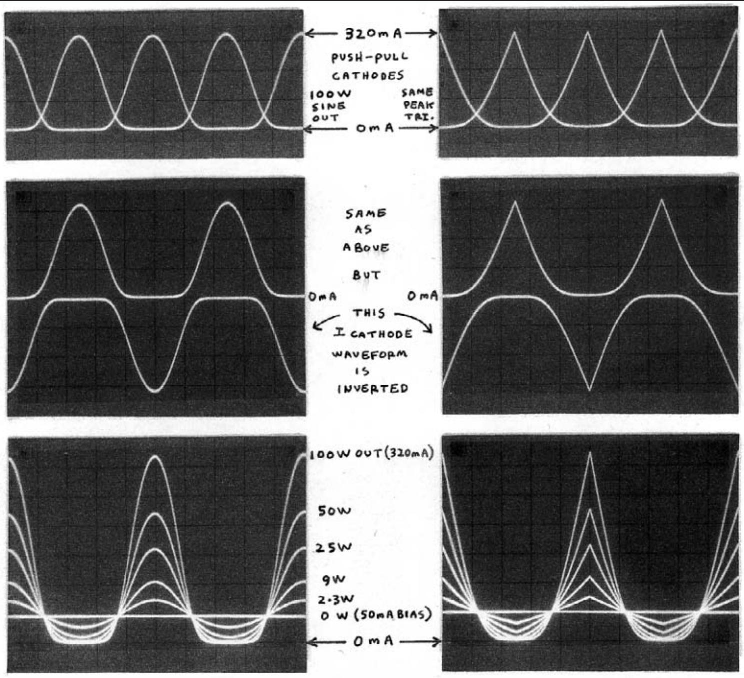


FIGURE 1: The Class A to AB transitions with sine and triangle waveforms. Note the absence of sudden cutoff.

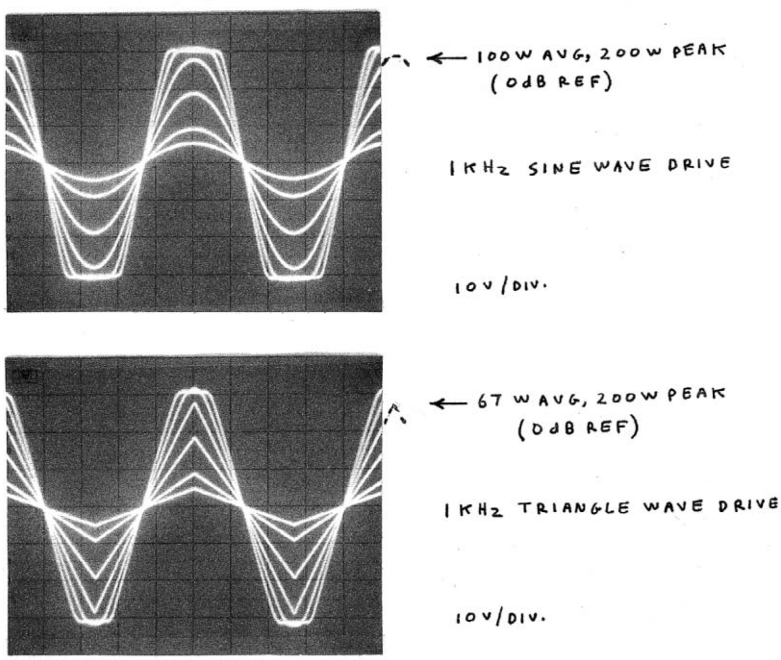


FIGURE 2: The smooth overdrive characteristics.

leakage. I used two Zobels, each plate to B+, and 1.5nF series with parallel 100mH, 300W 5W.

2. *"Push" and "pull" were not complementary (as you surmised), due to measured 1.2mH leakage* difference *between one primary half-to-secondary and the other; thus, different phase shifts on push and pull. As you said, this causes distortion in class AB, where each half conducts a (smoothly approaching cutoff) half-cycle. In push-pull AB, these would add up to a sine wave (canceling the half cycles' harmonics), but the differing phase shifts "tilted" a 20kHz 100W output, with about 10% 2nd harmonic (inaudible). But what would be* very *audible is the 10% 2nd order IM. Even-order IM rectifies, so a 10–20kHz band of, say, musical transients produces a "spectral downwash" from DC to 10kHz! Even smooth pink noise then produces that crunchy broken glass sound.*

   I inserted a 1.2mH inductor (no saturation at 1A), damped by a parallel 1k 5W, in series with the plate lead of the primary half having 1.2mH less leakage to secondary than the other one. This did the trick. Now, 100W 20kHz THD is about 3% (mostly 3rd harmonic, symmetrical) and decreases rapidly at lower power. 100W THD, 20Hz–5kHz is less than 1%.

   I think this may be an overlooked problem in class AB tube amps. Note that (1) the HF asymmetry (tilted wave, strong 2nd harmonic and 2nd order IM) was there without *any FB as well, (2) a Hammond 1650T transformer also had this leakage imbalance, and (3) reversing the plate leads reversed the 20kHz tilt direction (showing it was the transformer, not tube mismatch, and so on).*

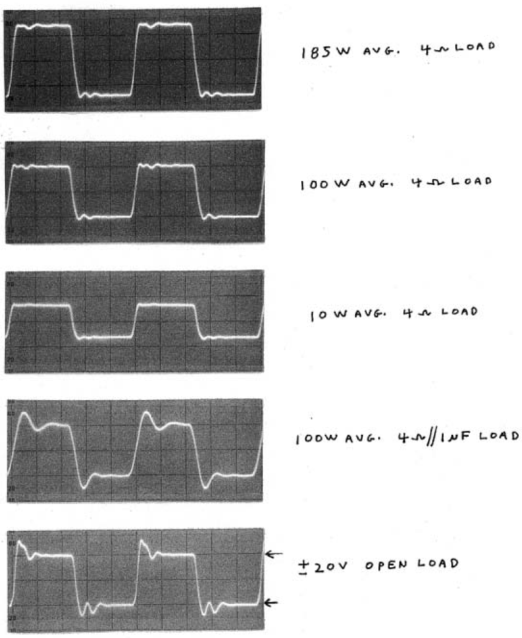


FIGURE 3: 10kHz square wave responses up to 185W average.

3. *Like you, I used no first stage HF comp; I used usual FB lead capacitance. Dominant rolloff is transformer primary capacitance plus Zobels. I didn't need secondary RC; the amp is no-load stable, and is also stable with a pure 3.3µF load. (This required serious labor optimizing driver BW and FB loop compensation!)*

   The amp sounds extremely clean up to clipping (120W per channel sine). SACD music is very transparent with no hint (yet) of HF roughness nor transient blurring. Clipping is very smooth without recovery problems; I use a 6SN7 cathode follower direct-coupled to the KT88 pair (per transformer half) grids; there are no coupling caps here, it can drive into AB_2.

The half-year I spent exploring and addressing these loop-affecting transformer parasitics (that you so aptly described) was well worth it: the sound appears to be of the highest clarity and resolution, and not from a lack of searching for faults (the tubes are biased, I hope I'm not).

I, too, built my first tube amp in 1958 (±1dB, age about 15). My father, Robert Colin, would call this a "Colincidence." There's something to be said for the 45 years of experience you and I are fortunate to have accumulated. Thanks again for your letter.

I've included some waveform photos (Figs. 1–5). ❖

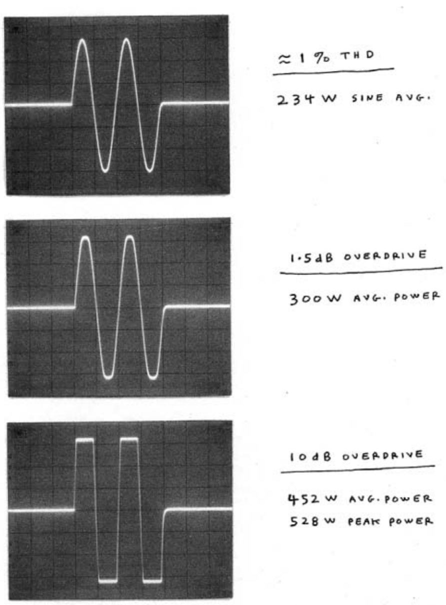


FIGURE 4: Clean recovery from a 452W 10dB overdriven sine wave (in bridged monoblock).

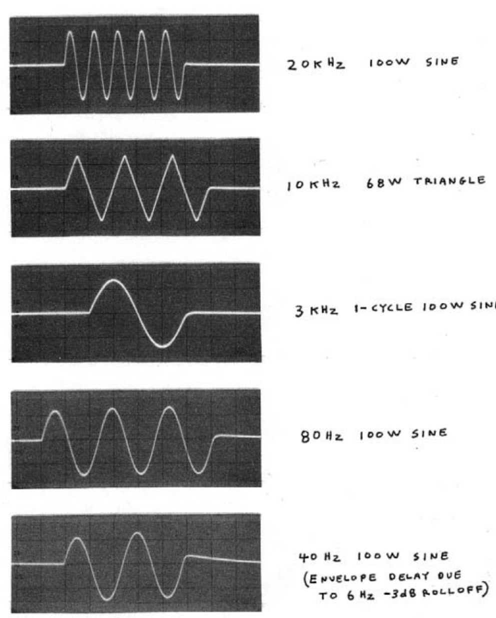


FIGURE 5: Various high-power tone bursts.

Showcase

A Dutch Tube Amp

My dream during the last ten years of my extensive amplifier and tube experience (55 years) was to build a parallel single-ended direct-heated triode amplifier with adequate power, without the bustle of 5 or 8W connected to horns and so on. My amp generates 65W RMS in 5Ω in my 89dB XANADU DS30 S silver wired 3-way loudspeakers. I prefer to build monoblocks instead of a stereo unit, mainly because of the weight issue.

Menno van der Veen, a good friend who lives in the same city, has the knowledge, reputation, and capability to design a perfect OPT for my goal. The blueprint for an SE 5k and 90W transformer already exists; however it has never been produced.

They (Menno and the Dutch distributor AMPLIMO B.V.) took care of this as a friendly gesture; I received two output transformer-type VDV HQ5090s (PAT-HQ5090). After ten years collecting parts, and then prototyping and listening for two more years, I was ready to build my own amp.

DESIGN

The chassis is made from 18mm hard press MDF. A friend of mine who is a teacher at a woodworking school made this beautiful chassis according to my design. After the inside has been clad with thin red copperplate for screening purposes, the outside of the chassis is sprayed in a "champagne" two-component metallic glossy lacquer.

With some help from the Dutch distributor, I designed the power supply of each monoblock with a special custom-made toroidal transformer, which has all the necessary voltage with sufficient power, is 20cm in diameter, and weighs 9kg. Every 211 or 845 amp should be made with this universal transformer.

Rectifying is done by a bridge of four EY500A tubes or the Svetlana 6D22S. In this way the HV supply gets an adequate 30-second delay, without any help. Two 6-chamber special-wound

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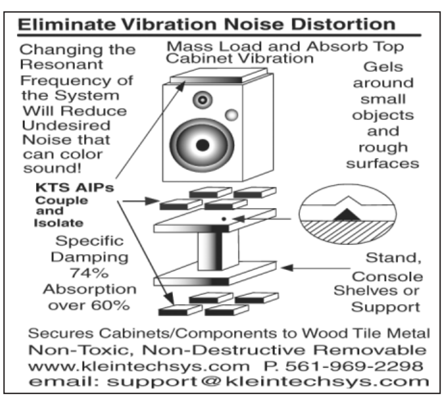
PHOTO 1: Front view of the ALFI PSE monoblock amplifier HQSE 65. You can see the special table made for these 35kg monoblocks. The weight is due, for the most part, to the giant power transformer and the van der Veen toroid output transformer type HQ5090 with a diameter of 18cm and a height of 9cm. Its weight is 8kg. The OPT is also lacquered a second time in the same color as the amp itself. The power transformer diameter is 20cm and height is 6cm; its weight is 9kg.

PHOTO 2: A rear view of the two amplifiers, showing the engraved black acrylic back panel with the WBT "Signature" platinum loudspeaker connectors. You can clearly see the large OPTs and the four rectifier tubes.

PHOTO 3: The front of the amplifiers shows the 24-carat gold-plated metal plate, engraved with inscriptions and my ALFI logo. The meter is switchable for the left 211, zero, and the right 211. Bias set is 60mA by 1250V DC for the 211s. One of the knobs includes a yellow LED status indicator.

Classifieds

VENDORS



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CLASSIFIEDS

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chokes are connected as a double pi filter with only glass-isolated O.I.Ps between and after them.

Both transformers, power and outputs, are mounted, one on top of the other, with a space of approximately 4cm, showing one of the great advantages of toroids: no hum! The amp itself is true PSE, and the 211s work with 1250V DC. This was possible after I added some special precautions in the transformers. The tubes are General Electric's VT4-C, which I bought some years ago when tube audio was cheaper. The giant 211 sockets are NOS, which I dismounted and mounted again to gold-plate the holder and re-silver-plate the spring clips.

Input goes to an SRPP stage with a 6CG7, and further on via the only coupling cap in this amp, a silver PIO, on to 2 × 5687s, connected as two cathode followers with gain, each DC-coupled to the grids from the two 211s. You must deal with driving the highly capacitive grids from these monsters. The output is generated in the excellent 5k toroid OPT VDV HQ5090 (PAT-HQ5090) with secondary 2, 4, and 8Ω. I used Super WBT "Signature" connectors such as the model Signature platinum for the speaker outputs.

On the front are two round knobs, one of which is illuminated with a yellow LED as pilot, and a meter in the center measuring the plate current from each 211. One knob turns on the amp, the other is a meter switch to select the left or the right 211 tube to be measured. The meter is high quality (a Russian moving coil type), but I re-constructed and re-scaled it for my application. I placed two yellow LEDs inside to illuminate the scale, and mounted one tiny red LED in the scale as bias point. The complete meter is also lacquered with a champagne color.

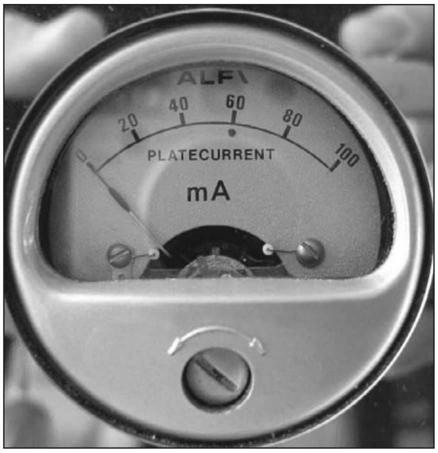

PHOTO 4: Detail meter for the bias of the HQSE 65. The meter originally was a black Russian moving coil type that was re-constructed and re-scaled for measuring plate current. The scale has a tiny red-glowing LED as the 60mA bias point, and is illuminated by two yellow LEDs.

IN ADDITION

The story is not yet over. An HQ5090 output transformer is on its way with primary and secondary windings with gold-plated silver wire! The amp is completely wired (including the power supply) with special gold dotted 1mm and 0.5mm silver wire. All small components are Vishays, Caddocks, Holcos, Tantalums, Jensens, and so on, delivered by partsconneXion, Canada. All connectors are WBTs. The amps are hardwired on mounting boards.

But finally, was it worth all the effort? The sound quality is excellent and is a great pleasure to listen to—no stress, no hiss, good definition, and much space in the sound produced. It is also very precise—approximately the sound of WE300B SE but with much more power. And the expressiveness in this sound is great.

The amp is never in trouble; the sound is very natural. Many folks around me in the audio scene—in audioclubs and in sessions given for the audio diehards— all say, "This is the end, it's not from this planet."

It is the most beautiful amp I have ever made. ❖

Bert Fruitema
Netherlands

TECHNICAL SPECS

Output: 16V RMS pure sine in 4Ω resistor

Output impedance: 2, 4, 8Ω

Frequency response: –1dB 16–23kHz
–3dB 8–44kHz

Input sensitivity for full power: 1V RMS

Tubes: 4XEY500A/6D22S, 1X6CG7, 2X5687, 2XVT4-C/211

Dimensions: length 50cm, width 35cm, height (chassis) 14cm, all over including tubes 30cm.

Weight approximately 35kg each