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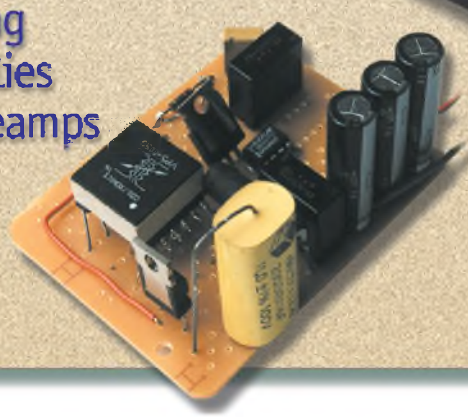

BUILD A **50W** COMPOSITE AMP
KEN MILLER

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Editorial

Mom Discovers Music

By Edward T. Dell, Jr.

When your mom's best friend drives her Bronco toward your mother's drive, you can hear her from a block away. Another crazie, you're thinking, going deaf early. Not.

Your mom gets into the vehicle and her whole attitude about music changes. Apparently the music inside seems less loud than outside. The system is big—one of those great Crutchfield specials—but what matters to mom isn't the gadgetry, it is the sound quality. It is unlike anything she has heard ever before. Mom is like a very large proportion of Americans, living in the middle of a country with the greatest advances in sound reproduction quality in the history of the world, but a chance ride in a friend's car opens the doors on a pleasure mom did not dream existed.

Within a week, mom has done some homework and has a brand new home theater system in her home. While those of us who love all the technicalities of the craft of audio enjoy sitting around talking about the details, we often completely forget the absolutely prime issue: "*It's the* music *stupid!*"

Had not my neighbor in my college dorm brought home recordings from the local library (on the streetcar), I might never have discovered music's charms, or at least not as soon as I did. Hearing high-quality music for the first time also changed my perception of what music in the home could be like, and ignited an appetite that persists to this day.

Those who know that we enthusiasts have an impressive array of equipment, or that we talk about music with an enthusiasm ordinary mortals do not seem to possess, have a whole array of "reasons" why a good sound system is not for them.

"I don't think my ears are as good as yours" is one I often hear. "That stuff is way too expensive for my budget. I hear guys spend more than the price of a car for just their speakers." There are lots of others.

Sound for most people is what comes out of a ghetto blaster, or a combo system, some of which are surprisingly good. I recently sat through an hour of chamber music examples in a classroom, played from CDs on one of these amazing little combos. At intermission I asked the teacher if I might unhook the speakers from the center chassis. I unrolled the little wires and placed the little two-ways about six feet apart. The sound spread into a quite believable image that was almost totally different from the chamber music we had been hearing previously.

Fortunately, the course included live rehearsals and performances by a really first-rank group in the afternoons and evenings, but just properly using a very modest system moved what we heard from it a little closer to the real thing. For most of the population, aural wallpaper or elevator music is the norm. Given the nearly universal appetite for some kind of musical "white noise" abroad today, it is not surprising that expectations are low—and that most people never suspect anything better is possible.

The system you have in your home has a power to offer some people a recognition of an unrealized appetite which might be awakened by an evening's music. Most people have no clue that they might really love music if they ever had a chance to hear the real thing. Our systems are not Carnegie Hall or a concert by the Stones. But most of them are capable of a lively hint.

Are you keeping all this to yourself? You're probably shy or reluctant to don missionary robes, but if you realize your own depth of feeling about what music can do, isn't it worth the risk of some very soft-sell hospitality?

I don't know how many of the audio clubs that still flourish across the land provide a meeting where guests are actively invited to share a couple of hours of excellently reproduced music. I doubt words are really a needed part of that equation.

The marvel about music listening is its ability to compound our pleasure as time goes by. It gets better, and hearing grows in competence and power, despite what we know about the aging process and our ears. If the sample a friend hears in your home triggers a response, think of the dividends that can appreciate over the years if he or she reacts by pursuing the pleasures and satisfactions music offers.

Not everyone like mom is going to install a new audio system a week later. But ponder, if you will, the potential in your carefully and lovingly crafted system which can give the same pleasure you enjoy to others.—**E.T.D.** ❖

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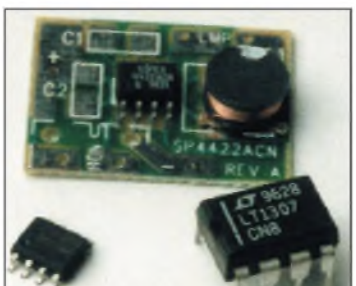
CONTENTS

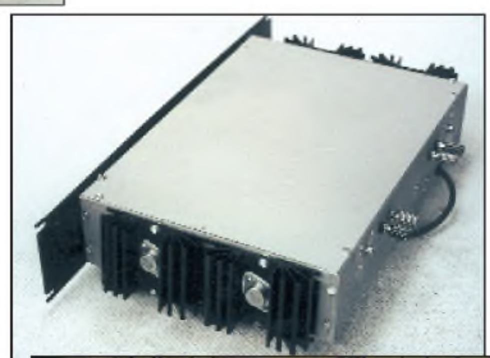
VOLUME 32 NUMBER 4 APRIL 2001

FEATURES



page 16

REVIEWS



page 6

page 81

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page 58

DEPARTMENTS

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

IN EVERY ISSUE

page 54

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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A 50W/Channel Composite Amplifier

Here's a stereo amp construction project that promises ultra-low distortion.

By Kenneth P. Miller

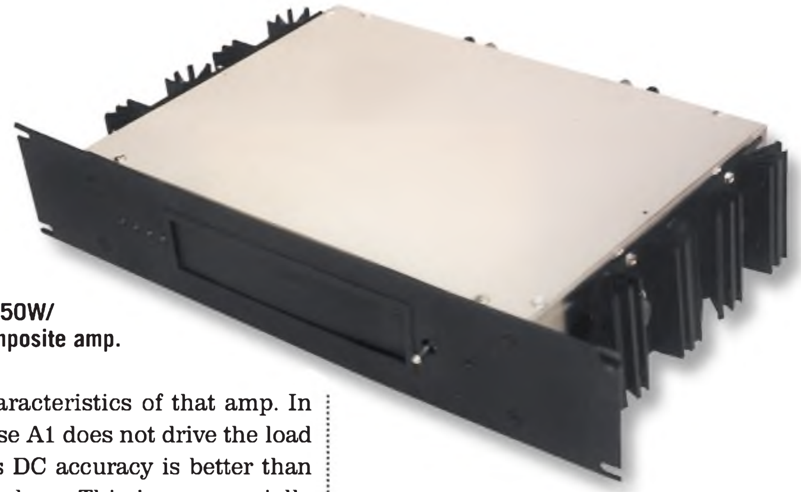

PHOTO 1: A 50W/ channel composite amp.

A composite amplifier consists of two or more operational amplifiers cascaded together with a negative-feedback loop around the entire network. Composites (also called nested-feedback amplifiers), used in electronics for a number of years, widen the bandwidth at high gain, boost slew rates, and lower distortion. *Figures 1*, *2*, and *3* are examples of composite circuits. Other possibilities exist, but these are probably the most common.

Figure 3 shows the block diagram of the circuitry I used in this project. I relied heavily on information in the *1994 Burr-Brown Applications Handbook*[1] to design my amplifier, and the circuit of *Fig. 3* is thoroughly discussed in bulletin AB-028, which is the main reason I chose it over the other two. In addition, it seems to offer higher loop gain, a desirable feature here, since my goal is ultra-low distortion. (*Photo 1* shows the completed amp.)

OTHER FEATURES

The composite offers other advantages. For example, you can combine the best qualities of both amps used in the compound, and DC performance of the composite is excellent. Since A2 is in the feedback of A1, the composite retains the DC characteristics of that amp. In fact, because A1 does not drive the load directly, its DC accuracy is better than that of A1 alone. This is an especially welcome feature in a power amplifier.

Thermal feedback in a power amplifier driving heavy loads can cause errors due to self heating, but the composite eliminates this problem. This is not just theory. My composite amplifier has a DC output-offset voltage of ±1mV, stays there regardless of the load placed on the output transistors, and uses no "zero adjustment" pot. Another advantage in my amp is the slew-rate boost: in the composite, the slew rate of A1 is multiplied by the gain of A2.

Obviously, the composite must be stable for you to benefit from its use. With composites, you should include the rolloff characteristics of both A1 and A2 in the AC analysis, and also have some means of providing phase compensation for the loop. Two amplifiers in a loop virtually guarantees oscillation, since the individual poles combine to produce a two-pole rolloff. Fortunately, there is a way to stabilize this composite.

STABILITY

In *Fig. 3*, A1 and A2 are identical, compensated op-amps, which makes it easy to predict the composite rolloff needed for AC analysis. I have learned from building this amp that the open-loop gains of A1 and A2 need not be the same. However, in this type of composite, it is essential that A1 and A2 have the same unity-gain frequency, F_T. At F_T, each must have a phase margin of 90°. These requirements are easy to meet if you use a dual, compensated IC op-amp.

Remember, though, that this is to be a 50W power amp, so A2 in this project is a discrete unit to handle the power. Making the discrete amp unity-gain stable at some specific frequency with the required phase margin of 90° turned out to be a tough job. Indeed, most of the design effort in this project was directed toward meeting that requirement.

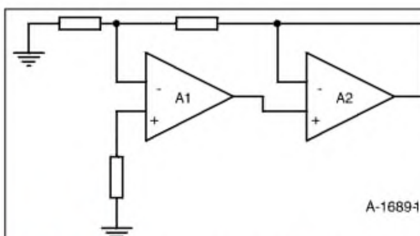

FIGURE 1: Sample composite circuit.

ABOUT THE AUTHOR
Kenneth P. Miller recently retired, so now he has more time for his favorite hobby, the design of audio power amps and associated test equipment. His other interests include loudspeaker crossover networks, model railroading and alternative energy. Kenneth P. Miller, 2032 Aud. Co. Rd. 389, Mexico, MO 65265, e-mail: peteanne@socket.net.

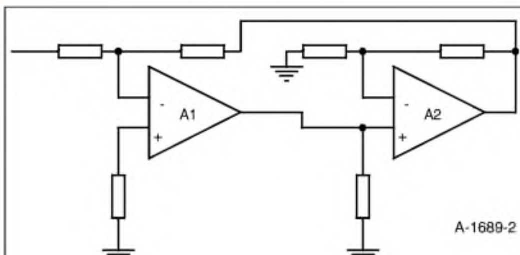

FIGURE 2: Another sample composite circuit.

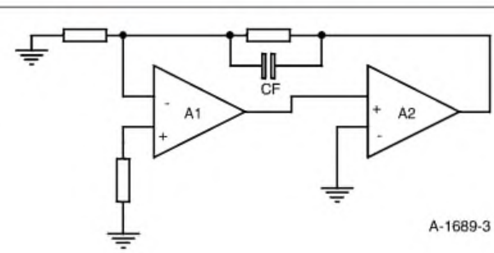

FIGURE 3: Composite amp circuit used for this project.

As things worked out, 2MHz was the highest unity-gain frequency I could achieve in A2 while still meeting the phase-margin requirements. All that's needed now to complete the composite is a compensated IC op-amp having an F_T of 2MHz. For this, I selected the OPA-121. Since this project is a power amp, you must also consider slew-rate needs.

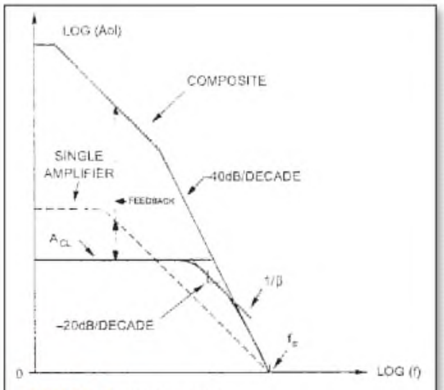

FIGURE 4: Bode plot.

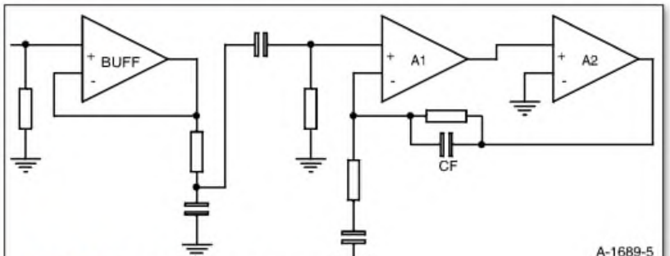

FIGURE 5: Full block diagram.

The intended output power is 50W into 8Ω up to 20kHz, which requires a minimum slew rate of 3.55V/µs.[2] The OPA-121 has a slew rate of 2V/µs. The composite solves the apparent slew rate shortage of this op-amp because it is multiplied by the gain of A2, as mentioned previously. The voltage gain of A2 at 20kHz is about 130, so the effective slew rate of A1 is 260V/µs.

Unfortunately, I was not able to utilize all that speed because the slew rate of A2 itself is about 12V/µs, and this slew rate sets the power bandwidth of the amp. Referring again to *Fig. 3*, capacitor Cf stabilizes the composite, but for the stabilization method to work, the closed voltage loop gain of the composite must be ten or greater. I use a voltage gain of 30 in this amp. (Both the mechanism of stabilization, as well as the reason for the minimum gain, require an explanation that is well beyond the scope of this article. I refer you to the previously mentioned application bulletin for more details.)

The size of Cf is found using the following equation: $Cf = \sqrt{10\,Acl}\,/2\pi R2\ F_T$, where Acl = 31, R2 = 8200, and F_T = 2MHz. Solving the equation, Cf = 170pF. I used 180pF. You can determine the amp's bandwidth by using the value of Cf and R2 in the equation, Fbw = 1/2πCf R2 = 107kHz.

You now have almost enough information to make a Bode-plot of the composite amplifier. All that's missing is the open-loop gain (Aol) specs for A1 and A2. Aol for the OPA-121 is typically 120dB, and for A2, it's 78dB. The Bode plot of the amplifier is shown in *Fig. 4*.

BODE PLOT

In *Fig. 4*, the heavy, solid lines are for the composite amplifier. For comparison, the dotted lines show the AC characteristics of a typical conventional power amp operating with the same closed-loop voltage gain. At any given frequency, the difference between Aol and Acl is the amount of global feedback. The more feedback, the less distortion.

This Bode plot provides a visual display of the superior distortion-suppressing capabilities of the composite arrangement. The Acl of the composite is down 3dB at about 105kHz, which is the effect of the stability capacitor, Cf, which rolls off the response of the composite so that it eventually intersects the Aol curve at a rate of 6dB/octave, ensuring stability.

The 105kHz bandwidth is too wide for

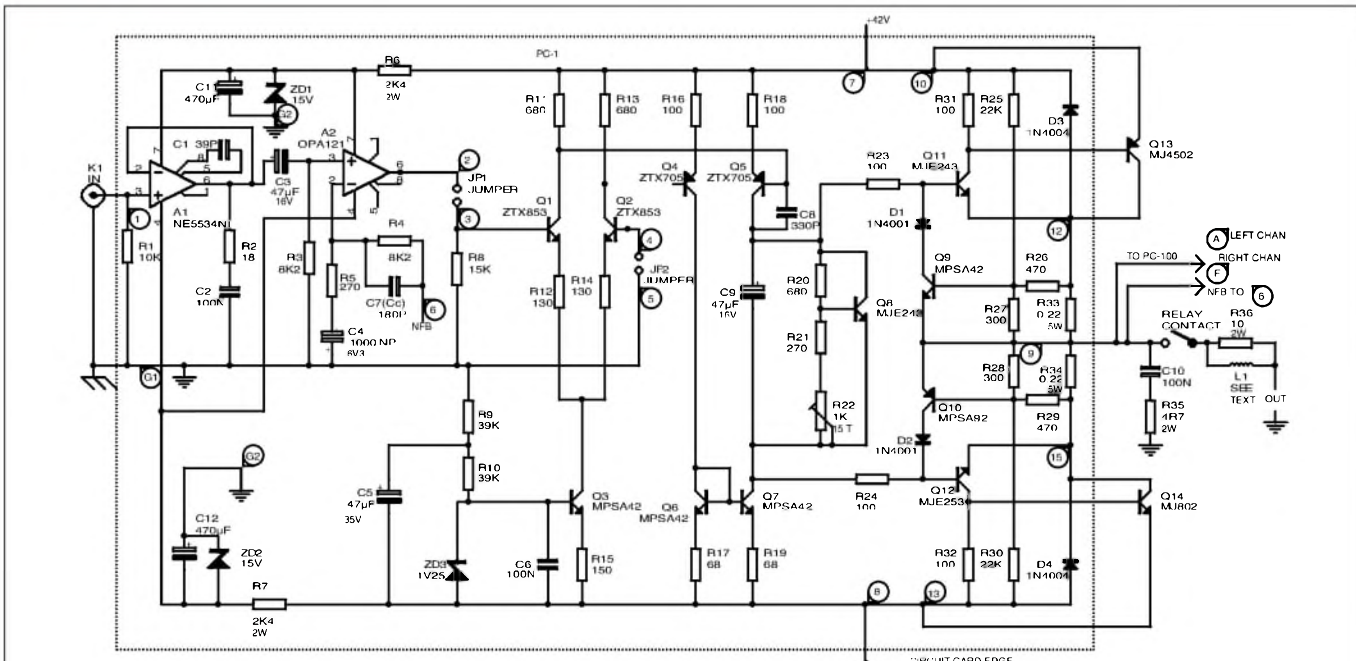

FIGURE 6: Amp schematic. Dashed line is boundary of printed circuit board, PC-1. Circled number or letter indicates numbered pad on foil side of circuit board.

the best low-noise performance.[3] I would prefer it to be about an octave lower. Normally, it is possible to restrict bandwidth by using a small capacitor across the feedback resistor, but in this composite that space is already occupied by Cf. So I added a unity-gain, low-pass, active filter ahead of the composite.

The active device also serves as a buffer. The preamp used with this power amp is an unknown factor, and its source resistance could interfere with the filter were this buffer not included. In addition, the 10kΩ input resistance may be too low for some preamps, and the buffer permits you to substitute a more suitable load.

The active device here is the 5534 op-amp, a very quiet unit that also sets the noise figure for the amp. *Figure 5* shows the full block diagram of this amp with the buffer/active filter included.

AMP CIRCUIT

Figure 6 is the amplifier's detailed schematic. *Table 1* shows the parts list. *Figure 7* is the circuit layout, and *Fig. 8* is the amp stuffing guide. Most of the circuitry is conventional and requires no explanation, but there are a couple of exceptions. Transistor Q8 provides temperature compensation for the output-stage bias current. It is physically connected to the case of Q11 with a short 4-40 machine screw and hex nut.

Bias current should be adjusted for optimum class-B operation at 11.5mA using R22. Single-slope volt-amp limiting is provided by Q9, Q10 with associated circuitry of D1, D2 and resistors R25–30. The amplifier output is routed to the speaker jack through normally open relay contact, "R," shown here as a SPST switch.

POWER SUPPLY

Figure 9 is the schematic of the power supply and relay pull-in circuits. *Table 2* shows the parts list, *Fig. 10* is the PCB layout, and *Fig. 11* is the stuffing guide. A101 is a quad comparator with two comparators per channel used to sense the DC-output offset condition of the main amplifier. One comparator is biased from the positive power supply to +80mV, while the other one associated with that channel is biased from the negative power supply to –80mV.

TABLE 1
AMPLIFIER PARTS LIST

REFERENCE	DESCRIPTION	PART #	SOURCE
A1	5534N op-amp	NE5534N	MCM
A2	OPA-121	OPA121KP-ND	D-K
D1, D2	1N4001		
D3, D4	1N5401, 3A, 100 PIV	#1N5401GICT	D-K
JP1, JP2	Jumper		
K1	RCA jack FEM		
L1	See text-10 turns #16 magnet wire wound on 1" diameter form		
Relay contact	(Part of RL1 or RL2)		
ZD1, ZD2	15V, 500mW zener	#1N5245BDICT-ND	D-K
ZD3	Zetex 1.25V ref. diode	#ZRA125AO3	D-K
CAPACITORS			
C1	39P		
C2, C6, C10	0.1µF, 50V	100N	
C3, C5	47µF, 16V		
C9	47µF, 35V		
C4	1000µF NP, 6.3V		
C7 (Cc)	180P		
C8	330P		
C11, C12	470µF, 35V		
TRANSISTORS			
Q1, Q2	Zetex NPN	#ZTX853	D-K
Q3, Q6, Q7, Q9		MPSA42	MCM
Q4, Q5	Zetex PNP Darlington	#ZTX705	D-K
Q8, Q11		MJE243	MCM
Q10		MPSA92	MCM
Q12		MJE253	
Q13		MJ4502	MCM
Q14		MJ802	MCM
RESISTORS			
R1	10k		
R2	18		
R3, R4	8k2		
R5, R21	270		
R6, R7	2k4, 2W		
R8	15k		
R9, R10	39k		
R11, R13, R20	680		
R12, R14	130		
R15	150		
R16, R18, R23, R24, R31, R32	100		
R17, R19	68		
R22	1k, 15 turn, top adj.	#CT9W102	D-K
R25, R30	22k		
R26, R29	470		
R27, R28	300		
R33, R34	0.22, 5W		
R35	4R7, 2W		
R36	10, 2W		

*See also "miscellaneous" section in Table 2

Each of the two comparators shares a common input from the amplifier output via resistor R101 from the left channel, R102 from the right. Capacitor C101 along with R101 and C102 with R102 form low-pass filters to keep the AC portion of the output signal from activating the comparators. As long as the power amplifier output remains within 80mV of ground potential, there is no output from the comparators, and the power transistor associated with that channel (Q101 or Q102) will conduct and pull in the relay for that channel. The outputs of each pair of comparators are wired OR via LED 1 and 2, or LED 3 and 4. I mounted these LEDs on the front panel for visual indication (*Photo 2*).

The main amplifier circuitry is completely DC coupled, and in the event of a failure the output could settle at a DC output high enough to damage the loudspeaker. The comparator/relay circuit prevents this. I had hoped that this same circuit would also remove all turn-on noise at the instant of power application. Unfortunately, it is not 100% effective for that purpose. Although there is no "thump" when you first apply the power, occasionally there is a soft "tick" or "swish." There is never any noise when you turn off the power.

As I already mentioned, ultra-low distortion is the goal of this amplifier. As always, it is best to make the amplifier as linear as possible before applying global feedback. British engineer Douglas Self in his excellent book, *The Audio Power Amplifier Design Handbook*[4], has identified eight distortion mechanisms in power amplifiers and ways to minimize them. I have taken great care to follow his advice in the design of this amplifier.

Distortion measurements are a problem for me with this amp. For one thing, my own analyzer test set has a residual distortion of about .01%, which is quite inadequate for measuring this amp. On the other hand, better test equipment does not appear to be an answer, either. The huge levels of global feedback used in this amp put the distortion products well into the noise floor, where even the best equipment can't retrieve them.

Now, how does the amp sound? In a word, invisible. It interfaces so seamlessly between preamp and loudspeaker that it just slowly fades away behind the sound stage, leaving only the music to catch your attention—music that is a step closer to the original.

In the absence of a signal, the amplifier is dead quiet with no audible hum or hiss. With 50Ω source resistance, A-weighted noise is more than 100dB below full-rated output. Maximum power output, one channel driven, 1kHz, 8Ω load, is 75W; and 90W with a 4Ω load under the same conditions. Frequency response at 1W output using the suggested low-pass filter is ±0.5dB, 20Hz–20kHz. Half-power (–3dB) frequencies are 2Hz and 58kHz. Sensitivity is about 650mV for 50W output.

CONSTRUCTION

For a nominal fee the local heating and air-conditioning shop made my chassis. Material is galvanized sheet

PHOTO 2: Front panel.

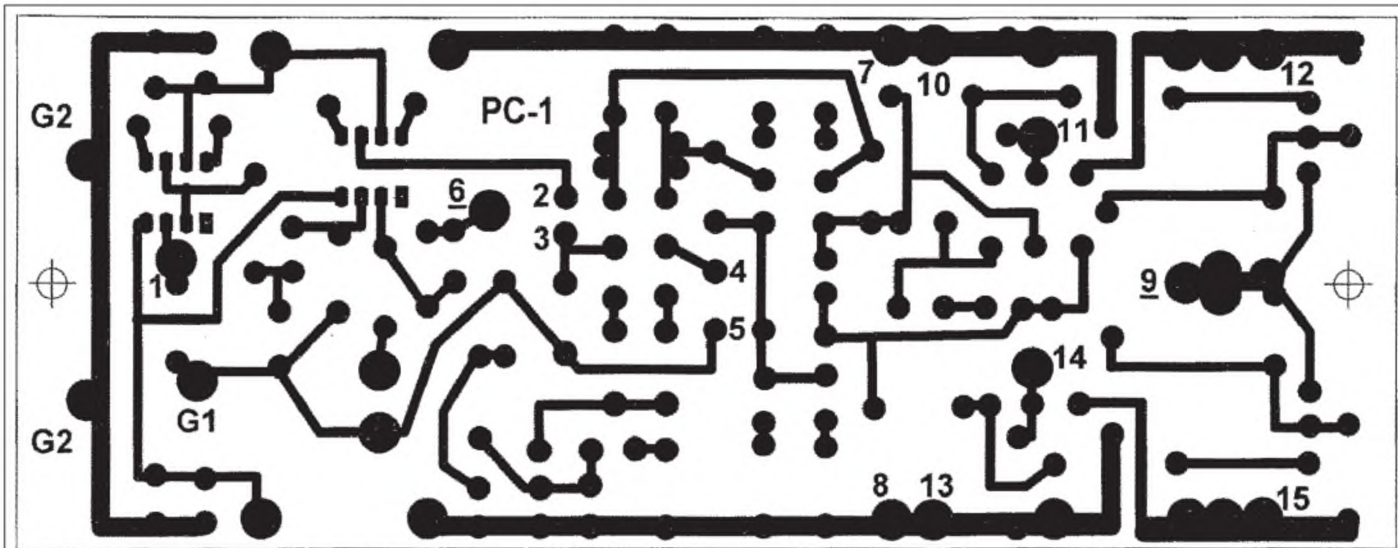

FIGURE 7: Amp PCB. (100%)

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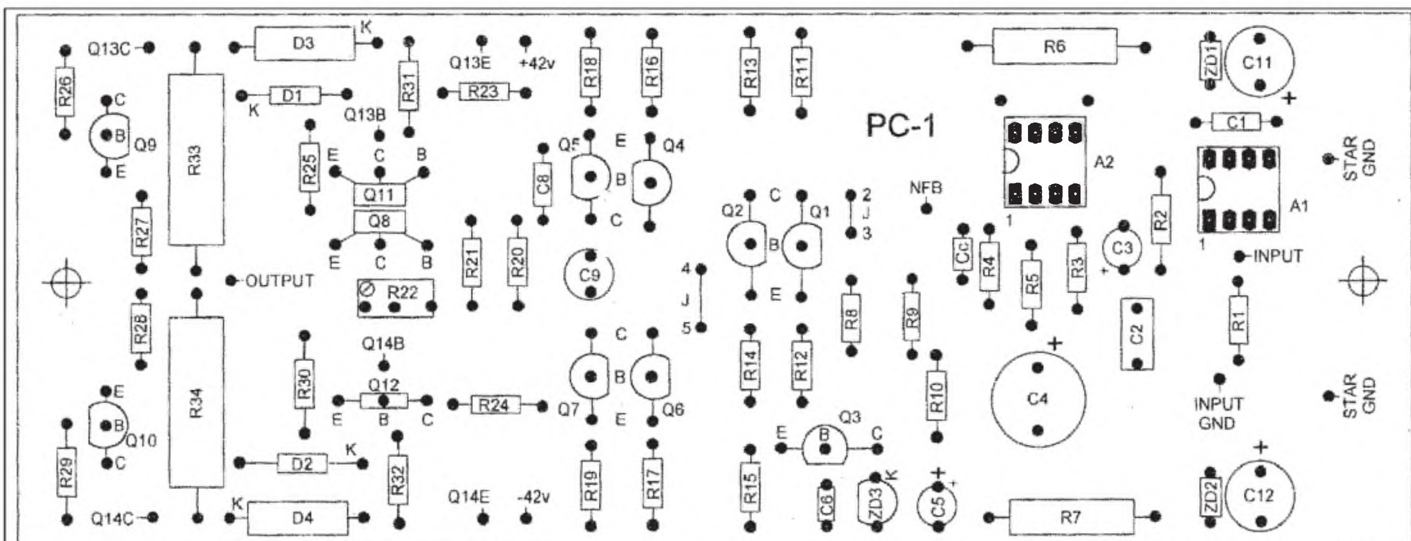


FIGURE 8: Amp PCB stuffing guide. (100%)

A-1689-8

metal, which is quite ugly, so I primed it with zinc chromate and painted it light gray. Box size is 14″ × 10″ × 3″ with a ⅛″ lip all around the perimeter for attaching the top with hex-head sheet-metal screws. This lip is also provided at the front and back corners of the main chassis.

The front, bottom, and back are of one piece of sheet metal, bent in a U shape (*Photo 3*). Both sides, as well as the top, are separate pieces, attached to the main chassis with ⅛″ × 6 hex-head sheet-metal screws. I rack-mount equipment, so I attached a 3½″ × 19″ faceplate to the main chassis with #10 sheet-metal screws, and reinforced the front corners of the chassis with a 3″ length of 1″-square extruded aluminum tubing. Angle aluminum or iron would do just as well.

Figure 12 shows where I mounted the major components inside the chassis. Exact spacing between components is not critical and is therefore not included. Moreover, you may prefer to use exotic input/output jacks, which usually take up more space than the ordinary kind. The chassis specified should be large enough to handle custom hardware as well as accommodate any necessary relocation of parts.

GROUNDING AND WIRING

Although minimal distortion is the goal of this project, the highest priority is safety, and for that reason you should use a three-conductor power cord. It is not necessary to sacrifice safety to achieve a system free of ground loops and annoying hum. To do so, I used Radio Shack #274-688 for the star-point ground. This is a terminal with five eyelets, with the middle one having a bracket for attachment to the chassis. I located this terminal close to where the power cord enters the chassis and bolted it to the bottom using #6 machine screws and serrated edge lock washers, scraping the paint off the chassis at this point to get a good electrical connection. I soldered a short piece of bare wire across the bottom of the terminal strip to tie all eyelets to chassis ground. I crimped and then soldered a ring-type "solderless" connector to the green lead of the power cord and fastened it to the terminal-strip mounting screw with another lock washer and a 6-32 hex nut. It is important that the "cold" or common side of the input phono jack not contact the chassis. The same holds true for the common terminal-output speaker jack.

When you fasten these parts to the chassis, use an ohmmeter to check that they are insulated from it. If you wait until the ground wires are attached to make this measurement, it is not possible to verify this condition. With one exception, all grounding was done with green, insulated, #18 gauge, stranded wire. The center tap of the secondary of

PHOTO 3: Side and rear views of completed amp.

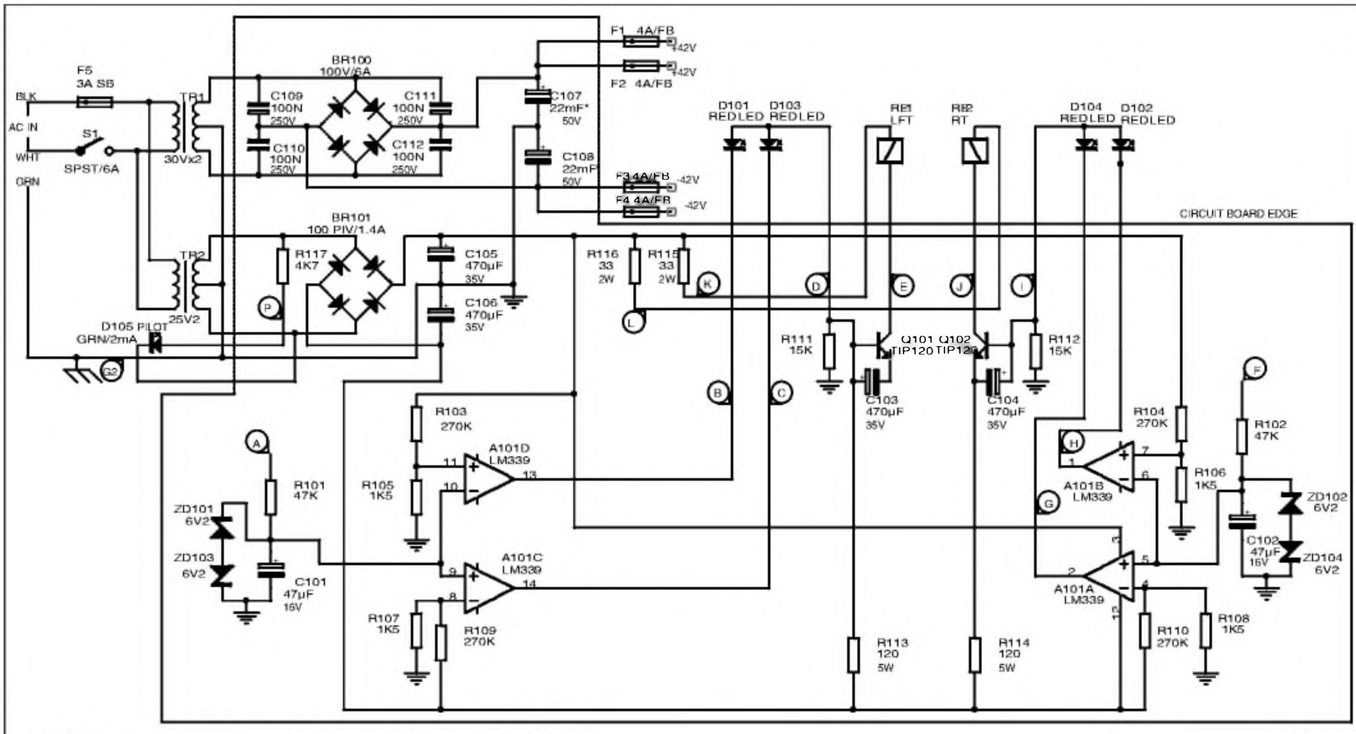


FIGURE 9: Power supply /protection relay circuit.

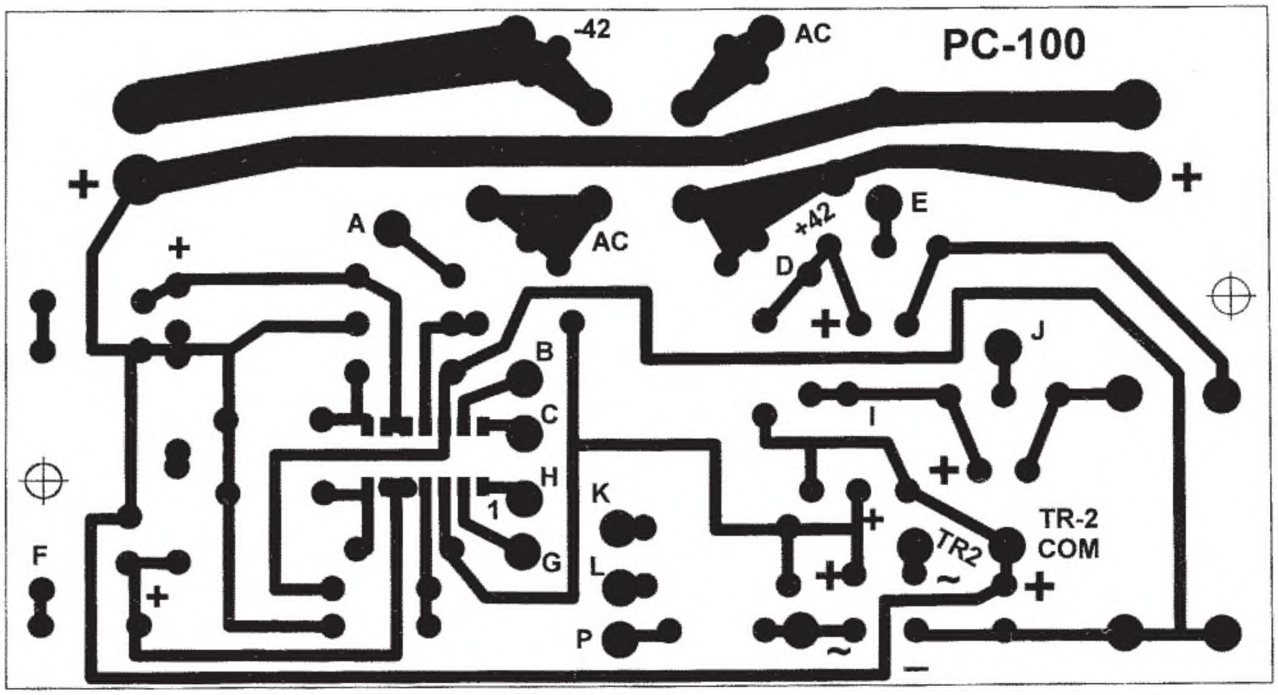


FIGURE 10: Power supply/protection circuit PCB. (100%)

the toroidal power transformer is connected directly to an eyelet on the star-point ground strip.

If you orient parts as shown in *Fig. 12*, you will need to splice a few additional inches of wire so this lead reaches the intended point. Solder the center-tap lead of transformer TR2 to the "GND" trace on PC 100, which is near the smaller of the two bridge rectifiers on this board. Also on PC 100 is another trace marked "GND" that you must wire to the star point. Because of the width of this trace, it may be necessary to use a soldering gun to avoid a cold-solder connection here.

The common side of each phono jack is wired to the star point, as is the common terminal speaker jack. On each amplifier board, PC 1, there are two ground connections, G1 and G2. Wire G1 to the common side of the input phono jack, and G2 to the star point. There are two eyelets on this board marked G2, but

TABLE 2
POWER SUPPLY PARTS LIST

REFERENCE	DESCRIPTION	PART #	SOURCE
A101	LM339 quad comparator	LM339	MCM
BR100	100V/6A, 400 PIV	GBPC604	D-K
BR101	1.4A, 100 PIV	276-1181	RS
F1, F2, F3, F4	4A Fast fuse	270-1010	RS
F5	3A Slow Blow fuse	270-1025	RS
Q101, Q102		TIP120	MCM
RE1	Relay, SPST RT, 12V DC coil	275-226	RS
RE2	Relay, SPST LFT	275-226	
S1	SPST/6A power switch	275-651	RS
TR1	D3051 230/115 Pri., dual 30V sec., 330VA		
TR2	115 Pri, 25.2 sec.transformer	273-1366	RS
ZD101, ZD102,ZD103, ZD104	6.2V zener, 500mW	1N5234BDICT-ND	D-K
CAPACITORS			
C101, C102	47μF, 16V		
C103, C104, C105, C106	470μF, 35V		
C107, C108	22kμF, 50V	P6946-ND	D-K
C109, C110, C111, C112	100N,0.1μF, 250V AC	P4605-ND	D-K
DIODES			
D101, D102, D103, D104	Red 2mA LED	276-310	RS
D105	Pilot LED, green, 2mA	276-303	RS
RESISTORS			
R101, R102	47k		
R103, R104, R109, R110	270k		
R105, R106, R107, R108	1k5		
R111, R112	15k		
R113, R114	120, 5W		
R115, R116	33, 2W		
R117	4k7		

MISCELLANEOUS PARTS
Fuse block, 4 gang, D-K, part #3539K-ND
Fuse block, single, D-K, part #3536K-ND
Heatsink, 50W cooler, D-K, part #HS117
IC socket, 8 pin, D-K, part #AE9808
IC socket, 14 pin, D-K, part #AE9814
Input jacks, RS, part #274-346
Power cord, RS, part #278-1258
Speaker jacks, RS, part #278-307
Terminal strips, RS, part #274-688

only one of them should be used. Use whichever one requires the shorter length of wire to the star point. This will depend on whether you are wiring the left or the right channel, and will become obvious when you do it. These ground leads carry very little current, so I used #24 gauge wire for them.

Capacitor C10 and resistor R35 form a zobel network at the output of the power amplifier. These parts are not on the printed circuit board. I mounted them on another Radio Shack five-eyelet terminal strip. I used one 6-32 machine screw to fasten both this terminal strip and the protection relay to the chassis. You can solder the zobel network to any of the four ungrounded eyelets.

One end of resistor R35 needs grounding, but do not use the terminal-strip mounting bracket for that purpose. Ground the zobel network with a wire to the star point. There are more leads back to the star than there are points. Double up wires to the eyelets where you can. If you run out of room, solder to the jumper wire connecting the eyelets together.

This RC zobel network plays an important part in the stability of the amplifier and is needed for that function whenever the amplifier is energized. You must wire it so it is directly attached to the amplifier output at all times, which means it needs to be wired "upstream" of the normally open relay contact. If by mistake you wire it "downstream" of the relay contact, the power amplifier may be unstable for the five or six seconds it normally takes for the relay to close.

PARTS PLACEMENT

Fasten PC 100 to the chassis bottom using two threaded standoffs. The four-unit fuse block for the dual-polarity

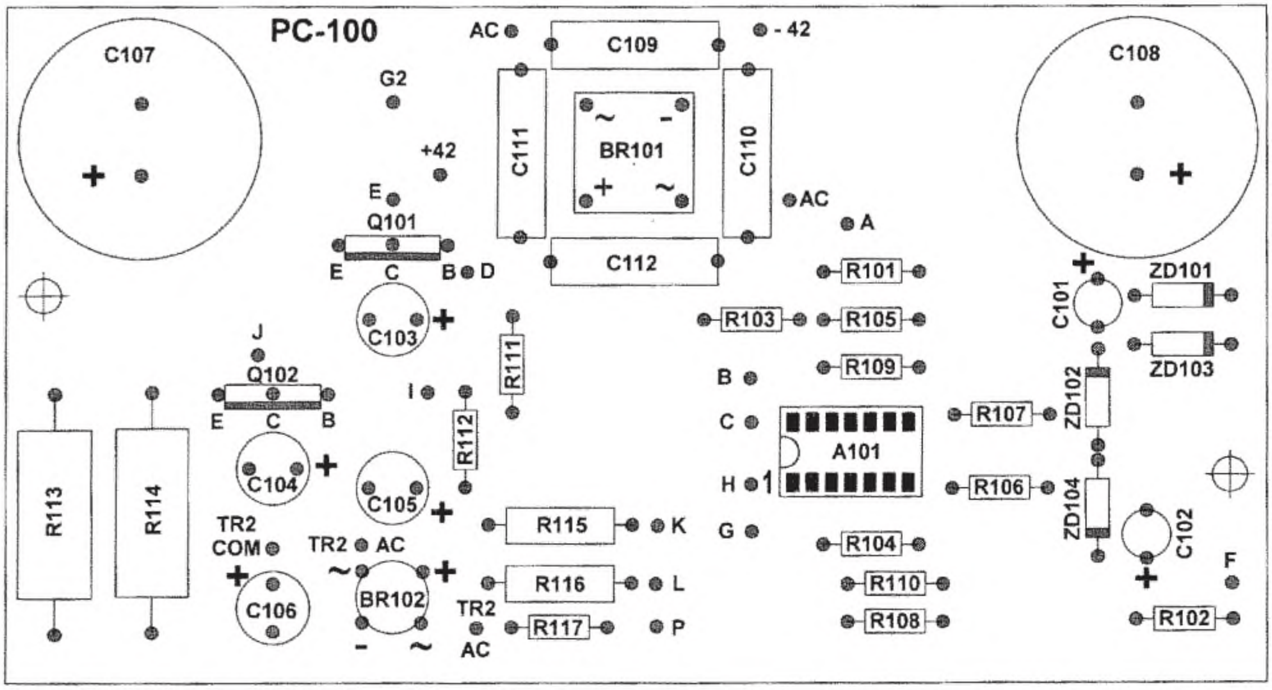


FIGURE 11: Power supply PCB stuffing guide. (100%)

A-1689-11

power supply is fastened to the back plate of the chassis approximately centered behind PC 100. I left enough space between the fuse block and the circuit board to permit easy fuse replacement.

Glue the fuseholder for the 120V AC main power to the top of transformer TR2 with silicone RTV adhesive. The secondary leads of the toroid transformer are long enough to reach their destination on PC 100, but too short to allow easy removal of the board for visual inspection or troubleshooting, so I lengthened these leads by about 4″.

The heatsinks specified are factory-drilled for the TO-3 output transistors. Before installing these transistors, place the heatsinks on the 3″ × 10″ end panel (removable in mine) at the desired location and, using a sharp pencil, trace the hole locations of the base, emitter, and collector (case). Use the eyeball method to find and then centerpunch these holes. Drill them using a 5⁄32″ drill bit, and then de-burr. Find the four smallest rubber grommets in the Radio Shack assortment and try one of these in the holes just drilled. If the hole needs to be enlarged, use a test hole in a piece of scrap sheet metal or thin plastic.

Once you find the proper size drill bit, enlarge only the holes for the base and emitter leads. De-burr the enlarged holes and install the four grommets. If they fit too loosely, fix this with silicone

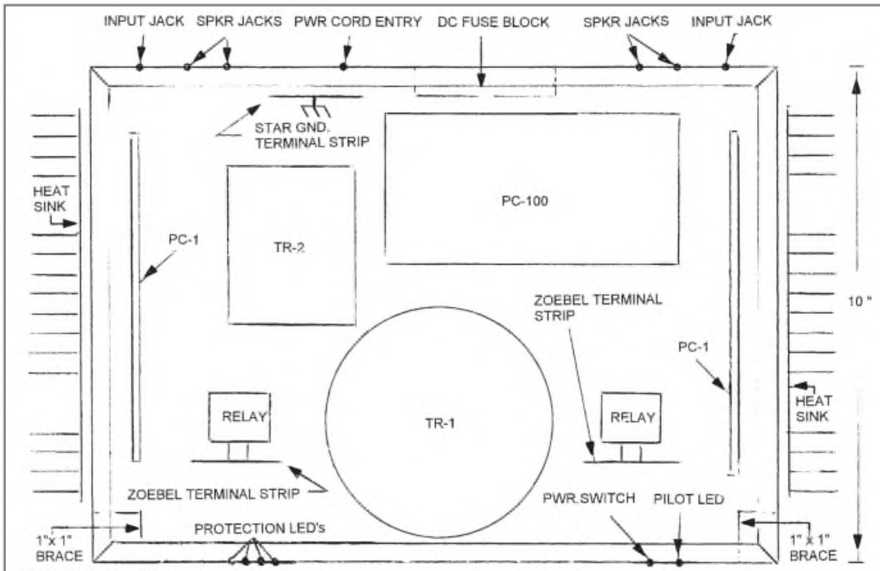


FIGURE 12: Parts layout.

A-1689-12

rubber glue. Next, enlarge the collector/case holes to ¼″; then use a universal step bit to enlarge the top hole to 7⁄16″ and the bottom to ⅜″ (*Photos 4* and *5*).

I used mica insulators and silicon heatsink compound between the transistor case and heatsink. Fasten the power transistors to the heatsink using 1″ 6-32 machine screws, nylon bushing, #6 washer, lock washer, and hex nut. I soldered a 12″ length of #24 wire to the base pin, and a 12″ length of #18 wire to the emitter pin. I crimped and then soldered a ring-type "solderless" connector to one end of another 12″ length of #18 wire and fastened it to the upper 6-32 screw with another lock washer and hex nut.

Then thread the base and emitter leads through the grommets and the collector/case lead through the upper, larger hole. Fasten the heatsink to the sheet metal. I found the factory-made flanges on the heatsink inconvenient to use, so instead, I drilled holes at top center and bottom center of the heatsink and used ⅜″ × 6 sheet-metal screws to hold them to the sheet metal. The heads of the machine screws holding the threaded standoffs in place prevent the heatsink

PHOTO 4: End plate prepared for heatsinks.

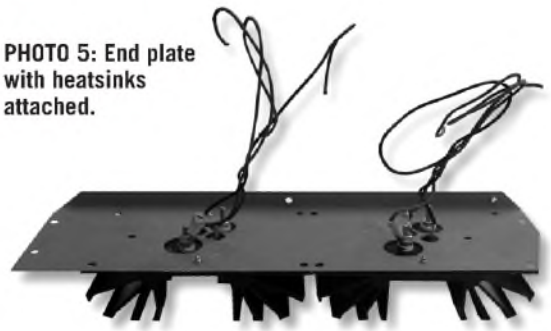


PHOTO 5: End plate with heatsinks attached.

PHOTO 6: Completed amp module.

from pulling up flush against the sheet metal. To overcome this, I used a pair of flat washers as spacers between the heatsink and the sheet metal.

Once this mechanical assembly of the output transistors is complete, you can trim the leads to the appropriate length and then solder them to the correct points on PC 1. The leads from the power transistors should be inserted into PC 1 from the foil side of the board, and then soldered. All other wires to this board should be inserted from the component side.

You should orient PC 1 with the input stage near the back or rear of the chassis as close to the input phono jack as possible. Check that this condition exists before completing the wiring for this circuit board. Wiring should be as short and direct as possible and yet long enough to allow easy removal of the board and heatsinks for visual inspection and/or troubleshooting (*Photo 6*).

CIRCUIT BOARDS

You should have good soldering skills, be able to read schematic diagrams, and own and know how to use a VOM, signal generator, and oscilloscope if you are interested in building this amp. I plan to have blank, drilled, etched circuit boards available for those who don't care to make their own. These will remain available as long as there is sufficient demand.

As long as I'm on this topic, I'd like to mention a better way to stuff the boards than the method used by most project builders. I divide the circuitry of the amplifier into sub-circuits, or sub-assemblies, according to the function performed. (A typical sub-circuit contains at least two, but rarely more than four, components.) Then I stuff and solder in place the parts of the designated sub-circuit, power up the board, and take DC voltage measurements to verify that it is working properly.

I repeat this procedure with the next sub-assembly, and so on, until the board is completely stuffed. A mistake produces a voltage reading significantly different from the one shown for that step, and you can quickly find it since it must lie somewhere in the last two, three, or four components stuffed.

During the stuffing process, the main amplifier board is powered from a dual-polarity, 18V source using the small Radio Shack transformer. The use of a much lower-than-normal supply voltage usually prevents damage to components if you make a mistake. I even use this low voltage supply to initially test the completed amplifier. Once you know the amp is working properly, replace the board supply with the normal high voltage source.

My stuffing procedure is very detailed. It is also lengthy, and so I do not include it in this article. If you plan to etch your own boards, the stuffing guides are available from me for one dollar to cover the cost of postage and handling. If you buy the drilled, etched boards, the stuffing guide is part of the package.

The finished project takes up a minimum of space. Despite the compact size, the interior is roomy enough to allow easy access to all parts. Moreover, the board layouts are spacious and use no surface-mount parts. Best of all, this amp offers a level of performance one step closer to that elusive goal, a straight wire with gain. Finally, many thanks to Rod Cavin, who supplied all the PC board artwork for this project. ❖

NOTE:
Most of the parts listed in the parts lists are readily available from any supplier. However, some parts are available from specific suppliers; D-K = Digi-Key, MCM = MCM Electronics, and RS = Radio Shack. All resistors are carbon composition, ¼W, 5% tolerance, except as noted. All capacitors are radial lead. Main transformer is available from Avel Transformers, 47 South End Plaza, New Milford, CT 06776.

REFERENCES

1. *1994 Burr-Brown Applications Handbook.* (Available from Digi-Key Electronics.) See Fig. 9, p. 202. Also available at www.burr-brown.com under applications, AB028, Fig. 9.

2. Jung, Walter, *Audio IC Op-Amp Applications*, Howard W. Sams, p. 44.

3. Motchenbacher & Fitchen, *Low Noise Electronic Design*, John Wiley & Sons, Chapter 7.

4. Self, Douglas. Available from Old Colony Sound Lab, 888-924-9465, www.audioXpress.com. Also see "Distortion in Audio Power Amps, Parts 1-3," *AE* 2/99, 3/99, 4/99.

Switching Power Supplies for Tube Preamps

Is the lack of high-voltage power supplies holding back today's tube market? This industry insider examines this question and offers some solutions. **By Eric Barbour**

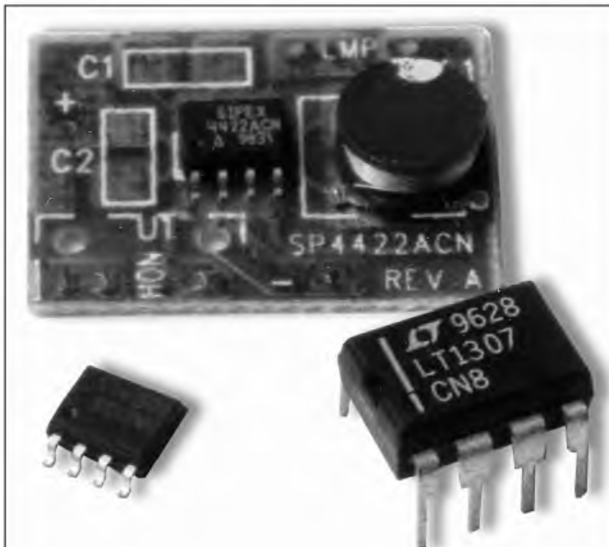


PHOTO 1: Clockwise from top: Sipex SP4422 IC mounted on Sipex evaluation board; Linear LT1307 switching-supply IC; IMP803LG EL-lamp IC.

Experimenters in the tube-audio field commonly use power-supply designs that have not changed since the 1930s. Since power transformers are commonly available for such duty, very little in the way of innovation in high-voltage power sources is seen in the pages of *aX* or other publications. However, if you are interested in a tube device that can be operated with DC power, such transformers are of little value. And, being specialized products for an "obscure" market, they are not manufactured in large numbers, resulting in high prices—even if bought in commercial-lot quantities. This drives up the price of tube-audio equipment, especially small preamps or pro-audio processing modules.

SMPS

The switch-mode power supply (SMPS) has become a staple of the world of commercial electronics. Savings in manufacturing cost, size, weight, and efficiency are the common reasons. Unfortunately, such designs usually produce no more than 24V DC. Getting the higher voltages needed for vacuum-tube plate supplies is a quandary nowadays, since voltages above 24V are rarely needed. Such supplies do exist as commercial products, though at very high prices.

The main issue is the magnetic device. A ferrite-core transformer with the necessary ratio is not really available off the shelf, and is likely to be a custom item. Hobbyists and small OEMs are at a disadvantage, since the cores and bobbins are also not readily available to them. Also, some skill and knowledge is needed to properly wind and use such transformers. (You could dig a good transformer out of a defunct low-voltage SMPS and use it "backwards"—assuming you had the requisite patience and ability!)

Some SMPSes provide efficiency simply by using "flyback" effect in a ferrite-core inductor. Although this means the supply will not give isolation from the input power, there are often cases where the device is powered by an external "wall wart" DC power supply or from batteries. This makes safety certification for consumer products easier, and allows a small OEM to market the same product worldwide without having to change AC input voltage taps or power plugs; simply enclose a suitable wall wart, or let the local customer or dealer provide the correct one.

Such a device could use a small SMPS to make plate voltage from a 6V or 12V DC supply, which is also used to run the tube heaters and other circuits. This method can be a boon for the maker of audio processors for musical use, such as tube distortion pedals for guitar. There are many such gadgets available; unfortunately, most use a single 12AX7 with plate voltages as low as 12V, simply to save cost. "Plate starving" a tube in this manner doesn't really give the true sound quality a tube is capable of, and would be foolish in the case of a preamp for high-quality audio applications.

I decided to examine the available IC devices that are capable of making voltages above 50V DC without custom transformers or special techniques (*Photo 1*). Although there are many SMPS chips available from various manufacturers today, rarely do their application notes discuss how to apply them to make voltages above 24V. Even the few ICs that claim to be capable of boosting voltages without limit are not usually capable of more than about 50–60V when used with a simple inductor, due to internal switching-speed or other limits.

An exception is the class of special devices which produces high-voltage AC for use in powering electroluminescent (EL) backlights for liquid-crystal displays. Such EL lamps are commonplace in industrial controllers and rack-mount electronic devices, since inex-

ABOUT THE AUTHOR

Eric Barbour is a consulting engineer and is currently owner of Metasonix. Eric is the Senior Editor of Vacuum Tube Valley Magazine and has been one of our regular contributors since 1991. He was the winner of the Antique Wireless Association's 1999 Gerald F.J. Tyne Award for his efforts in documenting the early history of electronics. Barbour holds a BSEE degree from Northern Arizona University and is a member of IEEE, the Audio Engineering Society, the Antique Wireless Association, and the Tube Collectors' Association.

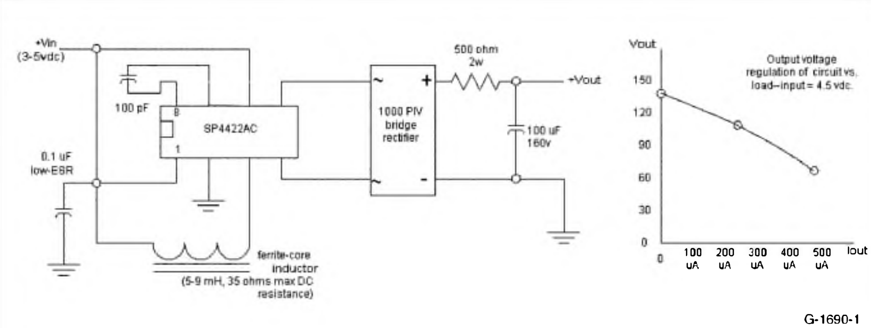


FIGURE 1: Switching HV power supply using Sipex SP4422AC EL-lamp driver IC.

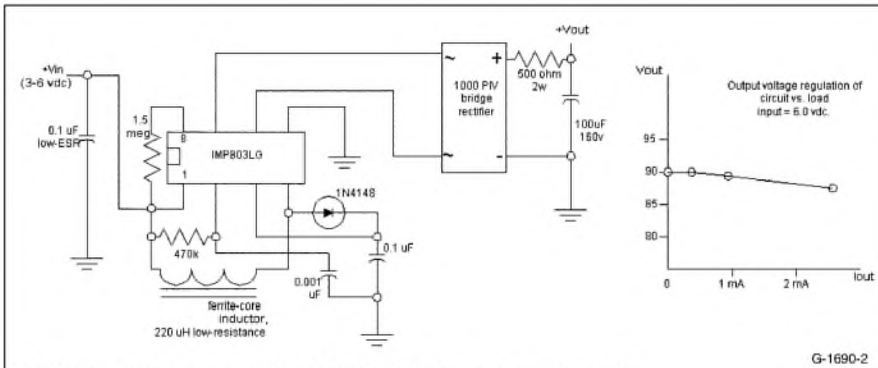


FIGURE 2: Switching supply using IMP803LG EL-lamp driver IC.

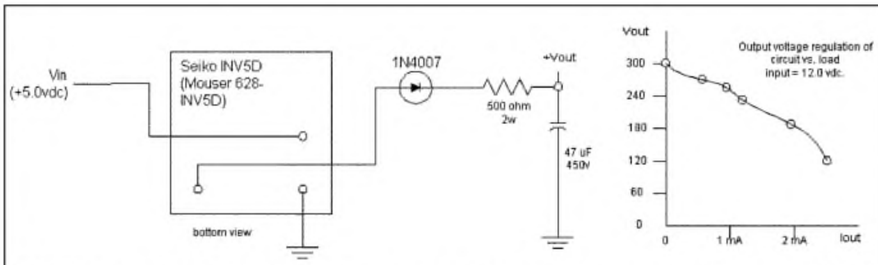


FIGURE 3: Applying Seiko INV5D EL power inverter as HV power supply. G-1690-3

pensive alphanumeric LCDs can be impossible to read without backlighting. EL panels are inexpensive and easy to apply to this job, so there are some special devices made to power them. After much research, I obtained samples of the two most usable devices for powering small-signal tubes.

IC SOLUTIONS

Figure 1 shows a basic circuit using Sipex's SP4422AC. Like the other specialized ICs meant to drive EL panels, the SP4422AC has push-pull outputs, requiring the use of a bridge rectifier. (Grounding one of the outputs would destroy the device.) Unfortunately, such circuits are meant to power the purely capacitive load of EL backlights, and are rated in terms of the size of the panel they can power, in square inches or square centimeters.

As the chart in *Fig. 1* shows, loading this device with a rectifier, resistor, filter capacitor, and load resistance causes a rapid drop-off in voltage with increasing load. The SP4422AC drops below 90V DC when sourcing more than 400µA. This is not much, although it would be enough to power a single EF86 or other small preamp pentode.

Also note that most of the ICs in this article are only available in surface-mount packages, since they are intended for use in portable electronics. The SP4422AC sample came in an SO-8 package and required the use of an SO-type breadboard fixture. (I operated it at different input voltages, since the spec sheet said the best efficiency would occur at other than the maximum-rated 6V DC input. As it turned out, best voltage regulation occurred at about 4.5V DC input.)

This circuit would be just right for powering an EF86-based music device, such as a small distortion box or preamp. A 6V battery supply or "wall wart," using a linear regulator or resistor to drop the input voltage to the optimum 4.5V, while using the full 6V for the tube heater, would allow for low manufacturing costs while still giving the EF86 a good plate supply.

Using these ICs in a manner not intended by their manufacturers, you actually must breadboard them and test them with a load. The circuit in *Fig. 2* gave much better load performance on the bench than the SP4422AC. The IMP Inc. IMP803LG proved to be a superior device for our purposes, even though its original data sheet did not have panel-driving specs superior to those of the Sipex chip, and in spite of using a smaller-value switching inductor to make the high voltage.

Its output voltage is lower at zero-load, yet it produces almost ten times the current when loaded down to a 20% voltage drop. The IMP803LG is also a surface-mount-only device. It worked fine with 6.0V DC input, which gives it yet another advantage for powering tubes. Such a power supply could operate at least two 12AX7 dual triodes (assuming an idle current of 450µA per 12AX7 and a 100k plate-load resistor).

Please note that the ICs I just mentioned are not available from hobbyist distributors. Only a determined DIYer would be able to obtain them, either as free engineering samples or in quantity from an industrial distributor such as Marshall or Arrow. I offer the previous circuits mostly for possible use in a commercial product. PC board fabrication would be a must for applying them properly, since they do not appear to be available in DIP or other through-hole mounting form.

EL LAMPS

If you wish to use EL-lamp inverters to power tubes, there are simpler options. Some of the firms that make EL panels also offer small, self-contained modules that produce the required high AC voltage. Seiko ECD is a major maker of EL panels and inverters. Best yet, they make these devices available through

Mouser Electronics, at reasonable prices, to allow the hobbyist to purchase small quantities.

Figure 3 illustrates how to use Seiko's INV5D inverter as a DC power supply. This is a tiny PC board with a transistor "blocking" oscillator mounted thereon, using surface-mount devices. It was meant for use while mounted to a larger PC board, yet can also be attached to a terminal strip via its leads. The INV5D produces more voltage than other EL devices. It would be suitable for powering a single 12AX7 triode, while two triode stages might represent an excessively heavy load.

I wouldn't recommend drawing more than 2mA from it. Since its output is single-ended, a voltage doubler could be easily applied for more voltage, although this would require derating the current output by 50%. As with all the circuits shown here, lifetime tests are strongly recommended to ensure that the devices are working within their ratings, especially for commercial application.

The EL-lamp drivers, above all, operate at audio frequencies of about 200–600Hz. Although this should be easier to filter than 60Hz AC, none of these circuits produce anything resembling a sine wave. So, for powering tube preamp stages, the additional RC filtering in *Fig. 4* is strongly recommended.

OTHER SOLUTIONS

More conventional high-frequency switching supplies also must be examined. The main issue is magnetics—most specialized ICs are designed as down-converters for lower than input voltages, for producing low negative voltages, or (increasingly in recent years) for boosting battery supplies to typical 5V or 12V logic levels. These application limitations allow for the use of simple circuits with simple ferrite inductors. Still, a few such ICs work with the few small transformers available pre-made.

Figure 5 comes from the full data sheet for the Linear Technology LT1307. The transformer was made available by Dale Electronics, by special order only. The data sheet does not discuss the intended application for

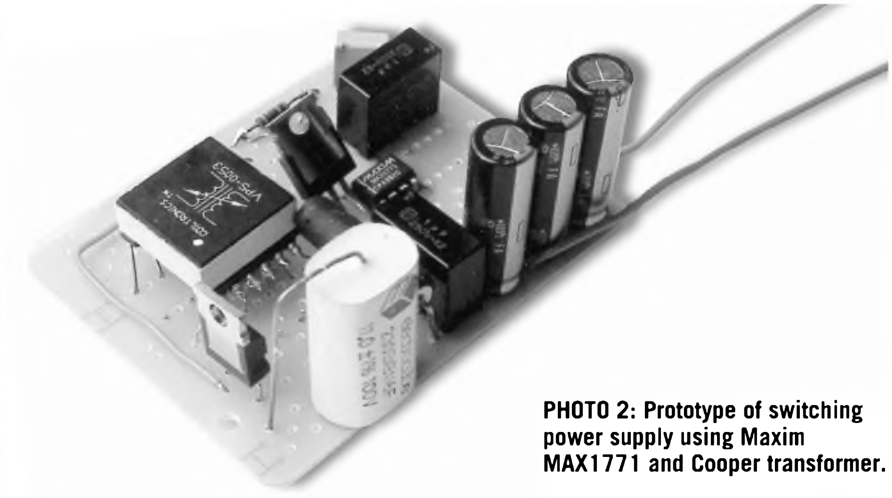


PHOTO 2: Prototype of switching power supply using Maxim MAX1771 and Cooper transformer.

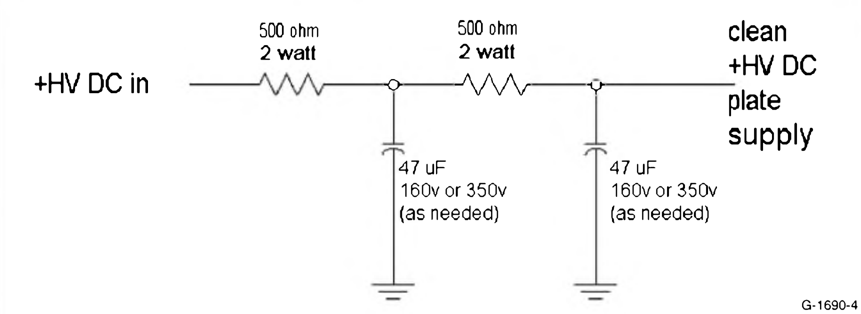


FIGURE 4: Additional filtering recommended for all circuits shown in this article, if HV is used to power vacuum-tube preamps. This circuit should reduce audio noise from any of the supplies described to less than 1mV peak-to-peak or better.

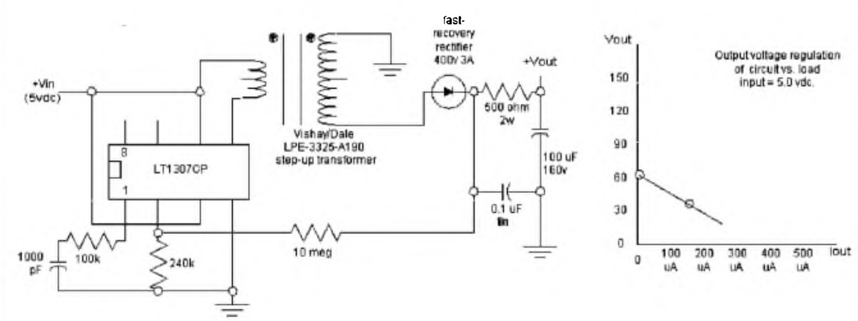


FIGURE 5: Switching HV power supply using Linear Technology LT1307CP DC/DC upconverter IC with Dale 1:12 voltage boost transformer.

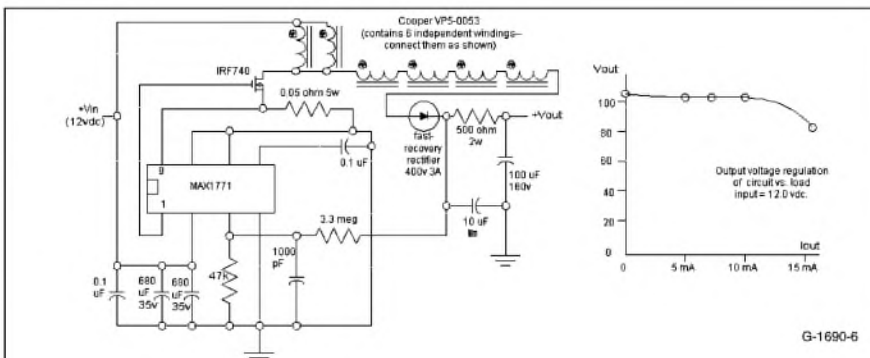


FIGURE 6: Switching HV power supply using Maxim MAX1771 DC/DC upconverter IC with Cooper voltage boost transformer.

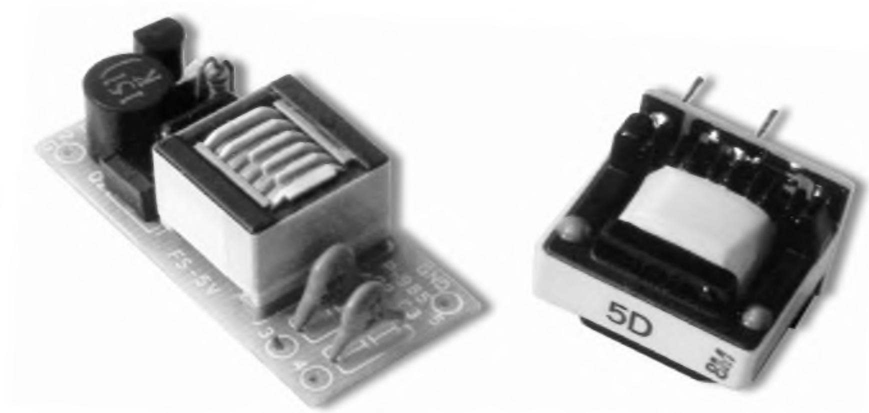


PHOTO 3: Left, JKL fluorescent lamp power module; right, Seiko EL-lamp power module.

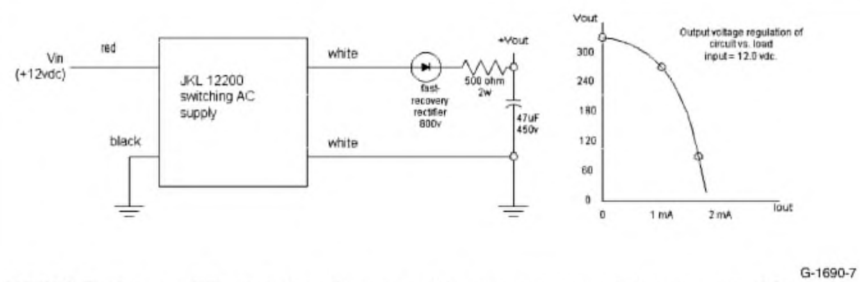


FIGURE 7: Using a JKL miniature fluorescent lamp power supply to generate DC.

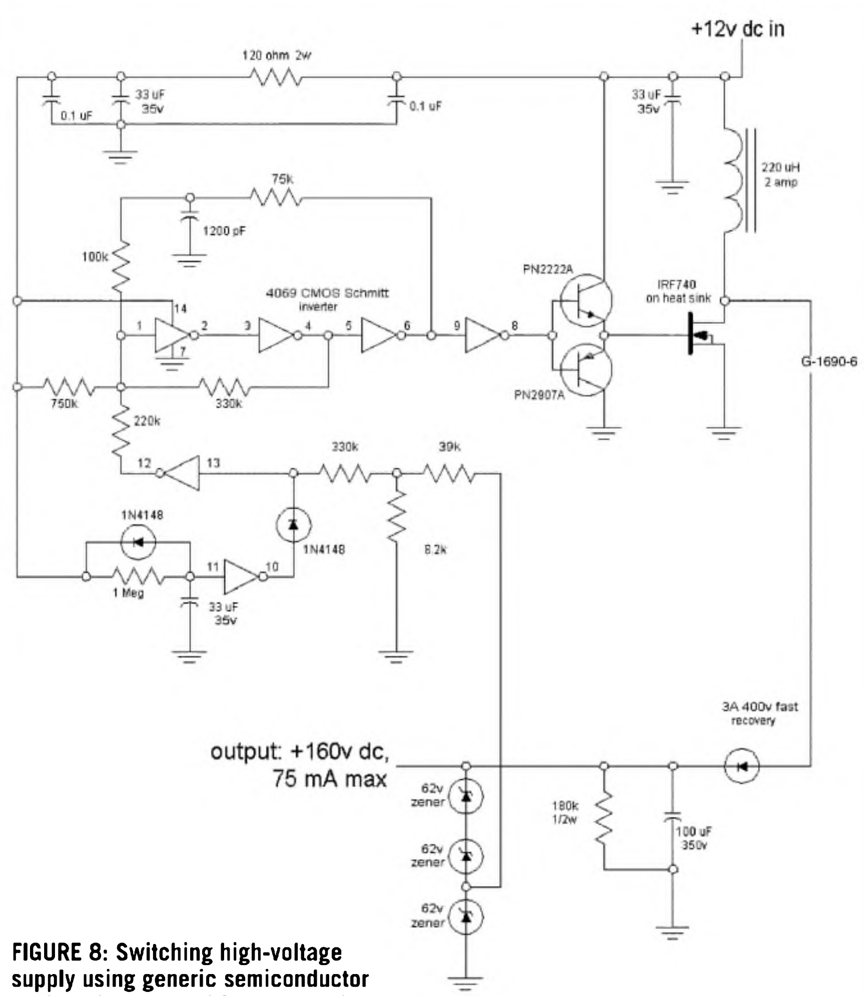


FIGURE 8: Switching high-voltage supply using generic semiconductor devices (courtesy of Steve Bench).

this circuit, although it does appear to be designed for maximum efficiency (possibly to provide high voltage for a vacuum-fluorescent display in a battery-powered device).

The converter operates at about 600kHz, and is apparently capable of operating on as little as 1.0V input. Unlike the EL-lamp ICs, the LT1307 is available in DIP form; unfortunately, the Dale transformer is surface-mount only. My rough breadboard did not deliver very good performance, though it must be said that a properly designed version, mounted on a PCB with short interconnections, might perform better due to the high switching frequency.

Figure 6 uses one of the very few small ferrite-core transformers, which is available off the shelf—the Cooper Electronic Technologies VP5-0053. It has six independent windings, and by wiring some in series and some in parallel, you may obtain high voltage. The circuit comes from a Maxim "Design Showcase" sheet, giving an un-

SOURCES

Cooper Electronic Technologies Incorporated
(formerly Coiltronics)
6000 Park of Commerce Boulevard
Boca Raton, FL 33487
(561) 241-7876; Fax (561) 241-9339

Dale Electronics Inc.
East Highway 50
(PO Box 180)
Yankton, SD 57078
(605) 665-9301; Fax (605) 665-1627

IMP Inc.
2830 N. First St.
San Jose, CA 95134-2071
(408) 432-9100; Fax (408) 434-1215
Website http://www.impweb.com

JKL Components Corporation
13343 Paxton St.
Pacoima, CA 91331
(818) 896-0019
Website http://www.jkllamps.com

Linear Technology Corporation
1900 McCarthy Boulevard
Suite 205
Milpitas, CA 95035-7414
(408) 428-2051; Fax (408) 432-6331
Website http://www.linear-tech.com

Maxim Integrated Products Inc.
120 San Gabriel Dr.
Sunnyvale, CA 94086
(408) 737-7600; Fax (408) 737-7194
Website http://www.maxim-ic.com

Sipex Corporation
22 Linnell Circle
Billerica, MA 01821
(508) 667-8700; Fax (508) 670-9001
Website http://www.sipex.com

usual application for their MAX1771 switching boost converter IC with external drive MOSFET (*Photo 2*). The original design gave 90V DC and was supposed to be capable of 700mA in short pulses. Its intended use was as the HV source in an automotive electronic ignition.

I modified it to give the maximum output voltage while still having good DC load handling. Unlike the other circuits, this one is capable of running several tubes at once, although at greater cost. Output ripple is high under load, so extra filtering is strongly recommended for audio circuits. The IC operated at about 50kHz. The MAX1771 is available in DIP form, although the transformer is surface-mount only.

Like the Seiko EL module, there are high-frequency ways of generating high voltage with minimum work, though at greater cost (*Photo 3*). JKL sells its miniature fluorescent lamps through various hobbyist distributors, including Jameco Electronics and Fry's Electronics. To power their lamps, JKL also offers small modular DC-AC inverters. *Figure 7* shows how to use their largest module, the 12200, to make DC. The output is transformer-coupled, so it could be used with a voltage doubler made with fast-recovery rectifiers.

Shown in *Fig. 8* is a suggestion for more power, also without a custom-wound transformer. This idea is courtesy of well-known hobbyist Steve Bench and was taken from his website (members.aol.com/sbench101/). His design is basically similar to that incorporated in specialized switching-supply ICs, only built-out using a CMOS inverter as an oscillator. Suitable high-current inductors are readily available (examples: Mouser 580-1422435 or 542-5254).

Although I have not constructed this circuit, other experimenters have, and report that it works as advertised. An output of 75mA at 158V is claimed. Some care in its construction and use is advisable, since CMOS and high voltages do not mix very well. You should not attempt to build this circuit without some experience in such designs. Specialized ICs commonly have some built-in protection circuitry, whereas this circuit is relatively free of protection, especially against reverse voltages and MOSFET failure.

FINAL NOTE

If applied properly, some of these circuits can help reduce the cost of OEM tube-audio devices. Since many of the ICs and magnetic components are only available in quantity, they may not be suitable for you.

As an industry insider, I am concerned about the future of the glass vacuum tube, and the ability of this tube-audio hobby to attract new members. If demand for new tubes declines, the few remaining factories making them will shut down their production lines. I hope that OEMs will use these circuits in low-cost products, thereby helping reverse the common perception of tube equipment as costly and difficult to use, and keeping this industry alive for the foreseeable future. ❖

A Bass-Boosting Network

The author describes the design and testing of an unusual passive bass-boosting network that allows you to build a bookshelf speaker system with an acceptable f_3 and excellent transient response.

By Peter Lehmann

The attempt to achieve high-fidelity bass reproduction from loudspeaker systems designed for bookshelf use has always intrigued me. For a given type of woofer or full-range driver, my goal was to design a sealed-box system with the characteristics of f_3, or the frequency corresponding to half-power in the bass range, and internal volume that approached the characteristics of a vented-box system design. My preference for a sealed system was based on its generally superior transient-response characteristic and more gradual rolloff of response for decreasing frequency below f_3.

My idea was that boosting bass response by passive means would be easiest when attempting to alter a less abrupt rolloff characteristic. The rate of rolloff of response for decreasing frequency below f_3 of vented systems is about twice that of sealed systems. By conventional methods of construction and for a given internal volume and type of driver, f_3 of a vented system can be an octave or more below that of a conventional sealed system.

An additional goal was to keep the construction as simple and inexpensive as possible. If I found that my basic concept of a method of passive bass-boosting was leading me down a blind alley, I would at least have the consolation of knowing I didn't empty my pockets in the endeavor. Also, there is a certain aesthetic appeal to a simple and inexpensive solution of a problem.

ABOUT THE AUTHOR
Peter Lehmann is self-taught in electronics. His BA degree with a major in philosophy helps him to be innovative. A patent is pending on circuitry he developed for improving the efficiency of incandescent lighting.

PARALLEL CONNECTING

The bass-boosting network (*Fig. 1A*) shows the basic principle of my method. It also would seem to be the most simple design of the required configuration. LS1 is the primary driver whose output is boosted. It is either a full-range driver or a combination of woofer and tweeter.

Choke L1, in series with woofer LS2,

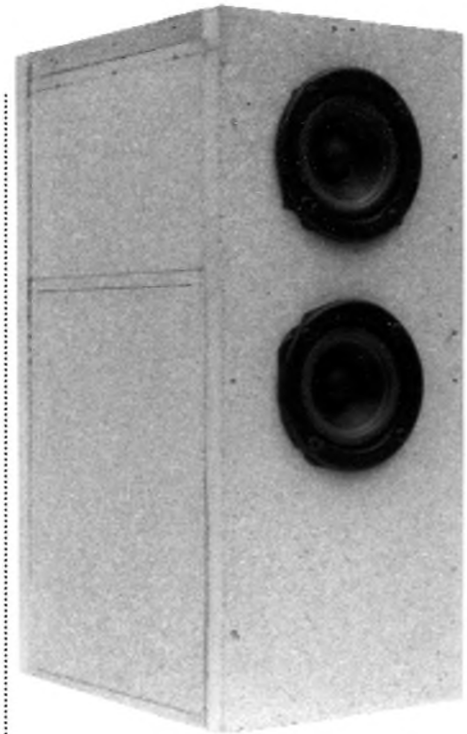


PHOTO 1: Testing enclosure with internal volume conforming to the golden ratio, W = 5.25″, H = 14″, D = 8″.

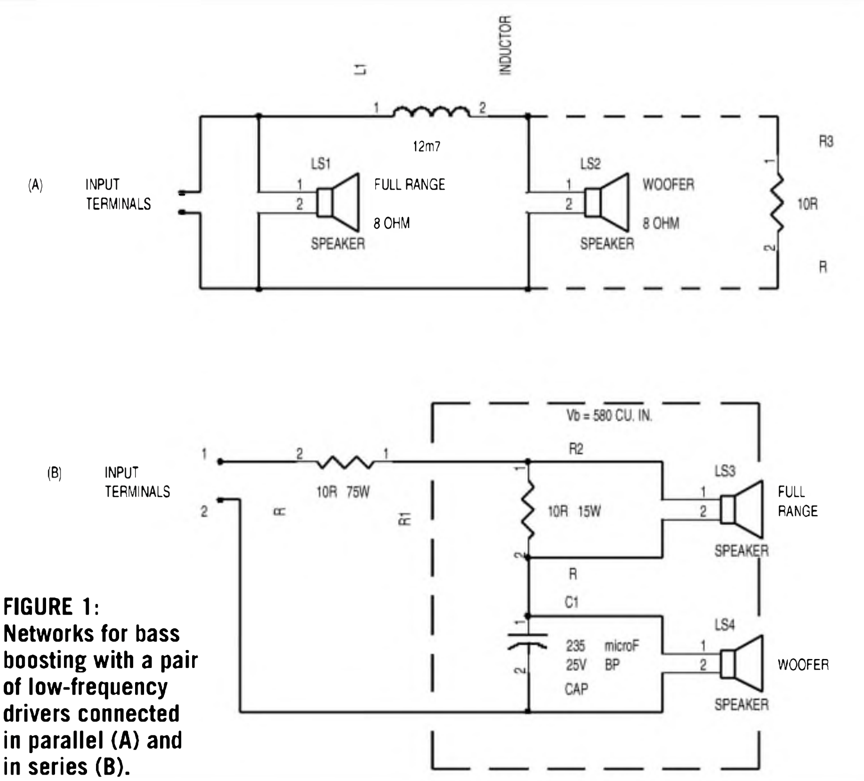


FIGURE 1: Networks for bass boosting with a pair of low-frequency drivers connected in parallel (A) and in series (B).

comprises a first-order low-pass filter network connected in parallel with LS1 to the input terminals of the speaker system. Resistor R3 is optionally connected in parallel with LS2. The first and second input terminals of LS1 and LS2 are positive and negative, respectively.

With resistor R3 disconnected, the inductance of choke L1 in *Fig. 1A* results in a half-power acoustic output of LS2, namely, an output-signal frequency of 100Hz. I selected this frequency of 100Hz for the half-power point since 100Hz is, respectively, one octave above and below the lower and upper limits of the bass-frequency band of 50–200Hz.

As the output-signal frequency progressively decreases from 200Hz, the reactance of choke L1 decreases, and the wattage consumed by LS2 increases. Since the intensity of acoustic output of the *Fig. 1A* system is directly proportional to the product of electrical wattage (converted to acoustic wattage) and the total surface area radiating sound, the result is an increase in the level of acoustic output for signal frequencies below about 200Hz. Where the cones of LS1 and LS2 have identical surface areas and the reactance of choke L1 is negligible with respect to the impedance of LS2, then the maximum boosting effect is about four times, or 6dB, greater than the acoustic output level of the system with LS2 inactive.

Where choke L1 in *Fig. 1A* is an air-core coil constructed according to formulas developed by Thiele, it has the following dimensions. If the direct-current resistance (DCR) of the coil equals 0.4Ω, then the height and outside diameter are 2.4″ and 7.2″, respectively. The corresponding wire gauge for this coil is 10 AWG. Thiele's technique for designing air-core coils can be found in *The Loudspeaker Design Cookbook*, by Vance Dickason.[1]

The bulk, weight, and somewhat involved construction of choke L1 as an air-core coil detracted from my desire to experiment with the boosting network of *Fig. 1A*. An alternative and costly course I might have taken was to purchase several metal-core inductors I could combine to produce inductance values about equal to my estimate of those required of choke L1.

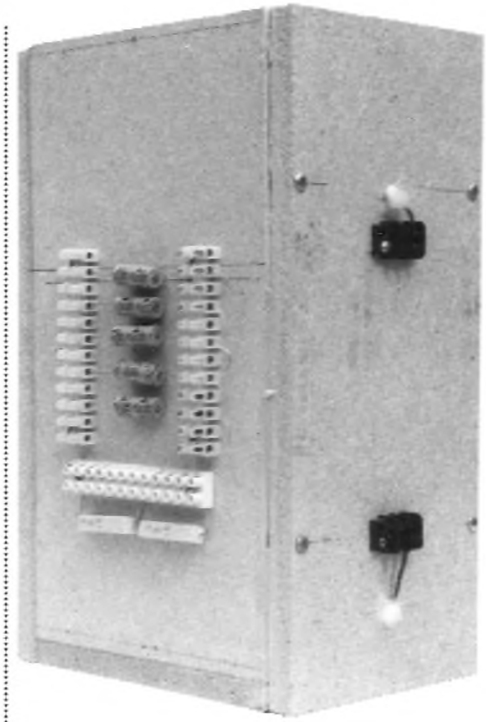


PHOTO 2: Block and European-style connectors mounted on exterior of testing enclosure for ease of changing connections and components of network.

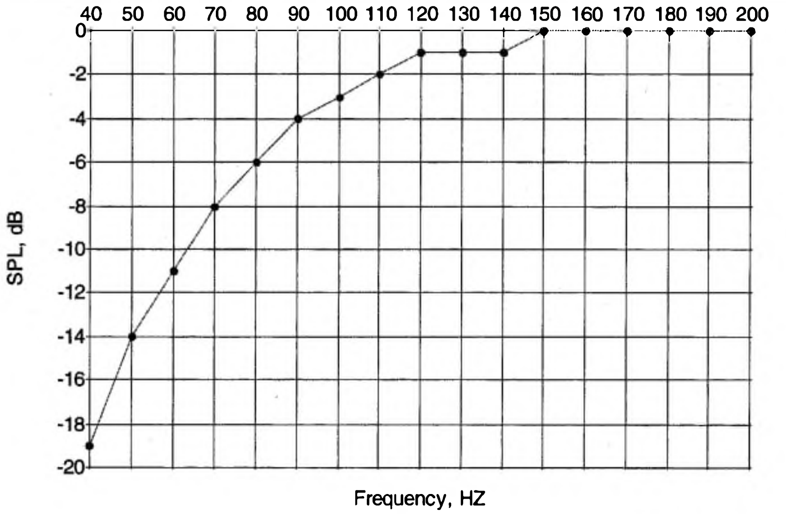


FIGURE 2: SPL versus bass frequencies of a sealed system with an Audax HT080M0 driver, $V_B = 290\ in^3$.

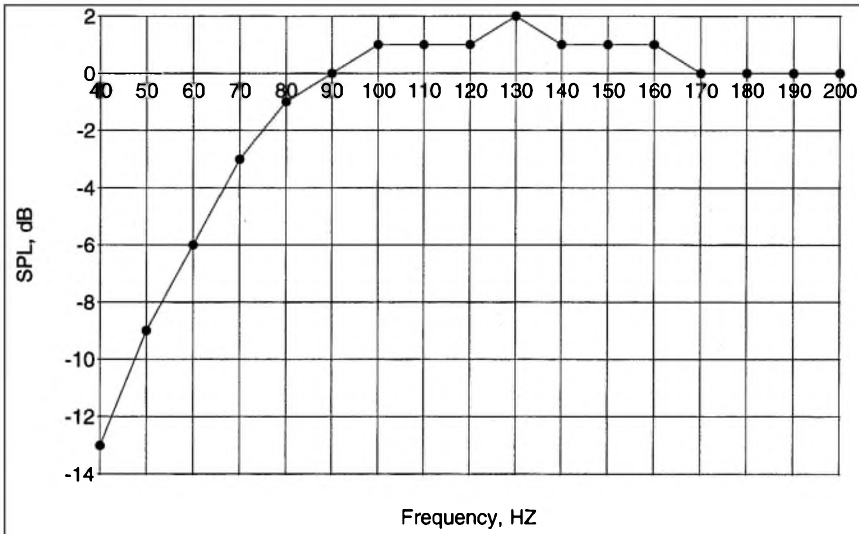


FIGURE 3: SPL versus bass frequencies of the sealed system of *Fig. 1B*, including a response-shaping network.

A flaw in the design of the closed box speaker system of *Fig. 1A* is that the impedance of LS2 abruptly rises to a peak value at the resonant system frequency, f_C. I thought that this increase would result in excessive boosting of the acoustic output of the system. That is, the impedance of LS2 at f_C, generally equal to more than double the nominal impedance of 8Ω, causes the acoustic outputs of LS1 and LS2 to be more nearly in phase, and—by voltage division of the low-pass filtering network—increases voltage drop across the input terminals of LS2.

Connecting resistor R1 in parallel with LS2 smoothes this peak. The adverse effect of connecting resistor R1 is increasing power loss and attenuated boosting of the system as frequency decreases from about 200Hz (as a result of current division between resistor R1 and LS1 and LS2).

SERIES CONNECTING

Figure 1B shows an alternative bass-boosting network I constructed and tested, which includes a first-order low-pass filter consisting of capacitor C1 connected in parallel with LS4. This type of filter, which utilizes shunting capacitance, is much easier and less expensive to construct than the filter of the *Fig. 1A* network, which uses a choke coil.

With capacitor C1 connected in parallel with LS4, drivers LS3 and LS4 necessarily are connected in series. Assuming very low internal resistance of the output stage of the power amplifier connected to the input terminals of this system, the input voltage is mostly divided between resistor R1 and drivers LS3 and 4 and the associated parallel resistance of R2 and capacitance of C1. As the frequency of the input signal decreases from 200Hz, electrical wattage, converted to acoustic output by LS2, rises with the increasing reactance of capacitor C1.

Photo 1 shows a pair of Audax 3″ full-range drivers (#HT080MO) mounted on a sealed box constructed from ⅝″ particleboard that I used to design the network of *Fig. 1B* by cut and try. I selected this driver model to experiment with since I was interested to find out whether or not such a relatively small driver could adequately reproduce bass tones. Also, the small frame of this model allowed me to design an enclosure with the narrow front-face width desirable for a bookshelf speaker system.

The vertical spacing between the two drivers was purposely kept to a minimum to allow for accurate nearfield measurement of the sound-pressure level (SPL) of the acoustic-output sum of the pair. The center-to-center distance between the pair of drivers equals 5″.

Photo 2 shows how I mounted a pair of barrier terminals to the exterior back of the box and three European-style terminal strips to one exterior side. I connected the input terminals of each driver to the first and second barrier terminals, and the resistors and capacitors of the bass-boosting network to the terminal strips for flexibility of interconnection and switching component values.

CONNECTION DETAILS

As shown in *Fig. 1B*, the first termination of voltage-dividing resistor R1 is connected to the first input terminal of the network, and the second is connected to the first (+) input terminal of LS3. The second (–) and first (+) input terminals of LS3 and 4, respectively, are connected. Completing the series circuit, the second (–) input terminal of LS4 is connected to the second input terminal of the network. Resistor R2 and capacitor C1 are connected in parallel with drivers LS3 and 4, respectively.

Drivers LS3 and 4 of the *Fig. 1B* network are simultaneously active for signal frequencies in a segment of the bass-frequency range. This simultaneous radiation means that the net internal volume, V_B, of the sealed box must equal twice that required for one driver to obtain the required Q_{TC} value.

Q_{TC} is the Q of the speaker system at its resonant frequency, f_C. Where Q_{TC} is a given constant, the volume V_B is directly proportional to characteristics of the driver pairs designated Q_{TS} and V_{AS}. The total Q of a driver at its free-air resonant frequency is designated Q_{TS}. The volume of air with a compliance equal to that of the suspension of the driver is designated V_{AS}.[2]

One of my design goals was to minimize the size of the speaker-system cabinet of *Fig. 1B*. Since my design requires two drivers operating in a segment of the bass-frequency range, this goal severely restricts the maximum allowable values of Q_{TS} and V_{AS}; for the Audax HT080MO driver, these values are 0.6 and .067ft^3, respectively. This value of Q_{TS} is somewhat higher than the average for drivers operating in the bass-frequency range.

However, a Q_{TS} equal to 0.6—increasing the required internal volume of the sealed system—is more than adequately offset by the relatively low value of V_{AS}. This low V_{AS} results from the small surface area of the HT080MO driver's cone exerting a less forceful compression of the air mass it acts on. Selecting a Q_{TC} equal to 0.71 results in a corresponding value of V_B equal to two times 290 in^3. A cabinet of this internal volume is acceptable in a bookshelf system.

DECREASING VALUE OF F_3

Selecting a Q_{TC} of 0.71 results in the lowest possible value of f_3 obtainable for values of Q_{TS} less than 0.71.[3] That is, if you mount a single driver of the type used in *Fig. 1B* on a sealed box with V_B equal to 290 in^3, then you obtain the lowest possible value of an unboosted f_3. It seemed to me that attempting to lower the value of f_3 of a sealed-box system could be best accomplished by starting out with the lowest value of f_3 obtainable by conventional means.

The three conditions affecting the acoustic-output intensity of the *Fig. 1B* system are: (1) the factor of voltage division between resistor R1 and the total impedance of the series connection of drivers LS1 and 2, and the corresponding parallel connections of resistor R2 and capacitor C1, respectively; (2) the relative phase angles of the acoustic outputs of drivers LS1 and 2; and (3) the distribution of wattage between the drivers.

Using phasor algebra, I calculated the voltage-division factor and relative phase angles of the pair of acoustic outputs at the frequencies of 50Hz and 200Hz. At both these frequencies, I measured the impedance of the HT080MO driver mounted in a sealed box with net internal volume of 290 in^3 as equal to about 8Ω. For my calculations, therefore, I modeled the driver impedance of 8Ω as equal to the DCR specified for this type of driver, and, in series, the capacitive and inductive reactance for the frequencies of 50Hz and 200Hz, respectively. The DCR of the HT080MO driver equals 5.9Ω, and the reactance modeled in series equals 5.4Ω.

OUTPUT-STAGE VOLTAGE

The voltage supplied by the transistorized output stage of a power amplifier connected to the input terminals of the *Fig. 1B* system is divided between the internal resistance of the output stage, resistor R1, driver LS3 and parallel-connected resistor R2, and driver LS4 and parallel-connected capacitor C1. Where the power amplifier has an average damping factor of 100 for an 8Ω load impedance, then the internal resistance of the amplifier's output stage equals 0.08Ω.[4]

I therefore neglected the amplifier's internal resistance when calculating the ratio of the voltage drop across the first input terminal of LS3 and the second terminal of LS4 to the voltage across the input terminals of the speaker system. Where this ratio is 0.5, half the total wattage consumed by the network is dissipated by resistor R1. Where the ratio is greater or less than 0.5, then more than half the total wattage is dissipated by resistor R1.

Where the frequency of the signal applied to the input terminals of the network equals 50Hz, I calculated the ratio of the resistance of R1 to the total impedance of the network to be 0.47. Therefore, for the frequency of 50Hz, a maximum transfer of power for conversion to an acoustic output occurs. Where the frequency equals 200Hz, that ratio increases to 0.6, with the result that the ratio of power dissipated by resistor R1 with respect to total wattage consumed is greater than when the frequency equals 50Hz.

PHASE ANGLES

The phase angles of the acoustic outputs of drivers LS3 and 4 I took as equal to those of the current passing through the voice coil of the respective driver. Since capacitor C1 of the network is in series with LS3 and in parallel with LS4, the phase angle of the current passing through LS3's voice coil leads that of LS4 to an extent dependent on the relative value of C1's reactance.

For example, when capacitor C1 is initially charging, the current of LS3's voice coil is at a maximum, and current is divided away from LS4's voice coil to a maximum extent. For the signal frequencies 50Hz and 200Hz, I calculated that this current lead equals 36° and 91.1°, respectively. That is, the interference of the acoustic outputs of drivers LS3 and 4 is progressively less constructive as the signal frequency increases from 50Hz to 200Hz.

As the frequency of the reproduced signal increases from the low end of the bass range, capacitor C1 increasingly divides current away from LS4 until the driver's acoustic output ceases and the total radiating surface area is halved. Given an initial state where both drivers consume equal wattage and their acoustic outputs are in phase, and a final state where the frequency of the reproduced signal has increased to the point where LS4 ceases producing an acoustic output while the wattage consumed by LS3 doubles, then the intensity of acoustic output is cut in half, or by –3dB.

I found that the impedance of the Audax HT080MO driver mounted in a sealed box with V_B equal to 290 in^3 reaches a peak value of 25.6Ω at an f_C of 110Hz. At this frequency, the reactance of capacitor C1 equals 6Ω. Excessive SPL of the speaker system's reproduction of signal frequencies approaching 110Hz is minimized by the susceptance of capacitor C1 equal to about four times the admittance of LS4, and the conductance of resistor R2 equal to about three times the admittance of LS3. At frequencies near 110Hz, the ratio of the resistance of resistor R1 to the total impedance of the network as it decreases and approaches 0.5 is partially negated by the parallel-current paths of resistor R2 and capacitor C1.

As frequency decreases from f_C, the reactance of capacitor C1 and the current division to LS4 increase. As a result, this network's bass boosting in the required range of frequencies is not limited in the same way that the connection of resistor R3 in parallel with driver LS2 of the *Fig. 1A* network limits boosting at frequencies below about 200Hz.

The speaker system of *Fig. 1B* operates with least efficiency in converting electrical wattage to sound when the reactance of capacitor C1 relative to the impedance of LS4 is negligible. To predict the maximum attenuation effected, I first calculated the quotient of the impedance of resistor R2 in parallel with LS3 divided by the network's total impedance.

The network's total impedance for this calculation is equal to resistor R1 connected in series with the parallel connection of resistor R2 and LS3. The modeled impedance of LS3 for this calculation equals 5.9Ω of resistance in series with 5.4Ω of inductive reactance. Secondly, I calculated 20 times the base-10 logarithm of the quotient, which equals –9.1dB.

To confirm the above calculation experimentally, I measured the SPL-versus-frequency characteristic of a control speaker system and the system of *Fig. 1B* for the frequency range of 1kHz–12kHz. The control system was a single HT080MO speaker mounted in a sealed box with a V_B of 580 in^3 without a response-shaping network. On average, the acoustic output level of the *Fig. 1B* system was at –9dB with respect to that of the control. This means that where the system of *Fig. 1B* and the control system produce an identical intensity of acoustic output, the former consumes eight times the wattage of the latter.

MEASURING BOOSTING

Figure 2 shows the frequency-response characteristic of a conventional sealed-box speaker system with a single HT080MO driver and a V_B equal to 290 in^3. The Q_{TC} for this system therefore equaled 0.71. I placed the microphone of a sound level meter (SLM) as close as possible to the dustcap of the driver to take these measurements according to the nearfield technique.[5] For this speaker system, f_3 equaled 100Hz, and the rate of rolloff below f_3 equaled about –11dB/octave.

The graph of *Fig. 3* shows the frequency response characteristic of the *Fig. 1B* system. For frequencies below about 1kHz, the Q_{TC} of this system also equals 0.71. Again using the nearfield technique, I placed the microphone of the SLM vertically equidistant between the radiation axes of the pair of drivers, and half an inch back from the exterior edges of their cones. The bass-boosting network resulted in lowering f_3 to 70Hz, and the rolloff below f_3 equaled about −12dB/octave.

COMPARABLE VENTED SYSTEM

For the sake of comparison, I explored on paper achievable values of f_3 and V_B of a vented system for the Audax HT080MO speaker, given an average leakage loss, Q_L, equal to 7.[6] Since Q_{TS} of this speaker equals 0.6, my choice was restricted to designs of the nonflat alignment class. Of the three types of nonflat alignments described, the only acceptable one was C4 generating a ripple of magnitude 0.46dB.

The other two nonflat alignments, BB4 and SQB3, would produce peak magnitudes of 3.73dB and 8.87dB, respectively, which I thought were nonflat to a nonacceptable extent. Given the C4 alignment, f_3 equals 47Hz, and V_B equals 429 in^3. For a frequency of 50Hz (referring to *Fig. 3*), when the vented C4 system is at half-power, then the *Fig. 1B* system is at one-eighth power, or one-fourth the power level of the vented system—a significantly lower figure. The cabinet size of the vented system would be about three-quarters that of the *Fig. 1B* system, which is not a considerable difference.

LISTENING TEST

A comparative music listening test with the *Fig. 1B* systems and a commercial bookshelf acoustic-suspension system was interesting. The commercial speaker was an Acoustic Research Model TSW110, which by coincidence has almost the same proportions and size as the *Fig. 1B* system. The Model TSW110 has a 6.5″ woofer and a 0.75″ dome tweeter. The f_3 of the TSW110 equals 68Hz, which, again, is almost identical to that of the system of *Fig. 1B*.

With all the tone controls of the integrated amplifier driving the systems nulled, the TSW110 seemed to have fuller bass response than the *Fig. 1B* system, but that impression might have been due to a slight amount of boominess of the TSW110. Using the system of *Fig. 1B* with the amplifier's bass tone control set several gradations above null resulted in a fullness of bass reproduction comparable to that of the TSW110 with the bass tone control nulled, and it also lacked the somewhat boomy quality of the TSW110.

The most significant difference between the two systems was the superior transient response of the *Fig. 1B* system. The distinctness and discernible pitch of bass tones it reproduced was noticeably better than that of the TSW110. The TSW110 had a brightness of reproduction that was somewhat lacking in the system of *Fig. 2*, which could be attributed to the TSW110's dome tweeter. ❖

REFERENCES

1. Audio Amateur Press, Peterborough, NH, 1997, sect. 7.80, p. 122.
2. Dickason, op. cit., section 1.15, p. 11, and section 1.50, pp. 17 and 18.
3. Dickason, op. cit., section 1.71, p. 21.
4. V. Capel, *An Introduction to Loudspeaker and Enclosure Design*, London: Bernard Babani (publishing) Limited, 1993, p. 101.
5. Dickason, op. cit, sect. 8.81, pp. 135–137.
6. Dickason, op. cit., Chapter 2.

A Pair of Computer Speaker System Designs, Part 2

With the enclosure shells completed, it's now time to assemble the units. **By G.R. Koonce**

PHOTO 1: Completed CSA system.

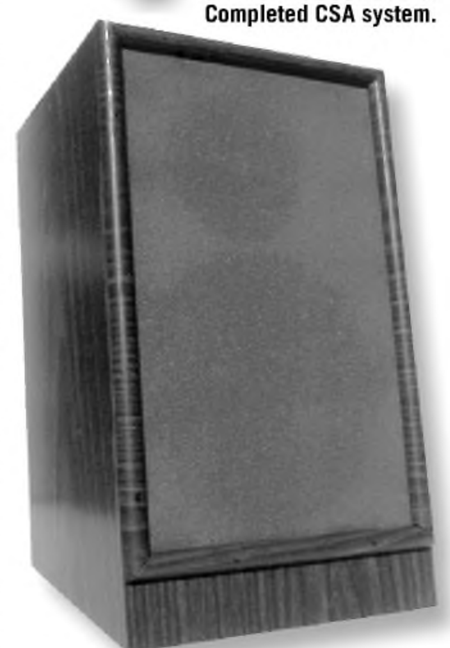


PHOTO 2: Completed CSB system.

Let's take a look at the details of CSA construction. *Table 1* shows the dimensions of the ¾″ (1.91cm) particleboard pieces making up these boxes. The layout of the odd-shaped pieces, along with the port hole diameter and location, appear in *Fig. 10*. Front and side views of the box (*Fig. 11*) show the locations of the various items and the hole sizes required. Save the cutouts from the holes for the tweeter and the port, since you can clean them up and use them to terminate the side-to-side dowel. *Table 2* lists the materials required to build two boxes, and *Table 3* the components used in each box.

Front and back views of the assembled boxes before mounting the drivers (*Photo 3*) illustrate the following:

1. Build the front panels mirror-imaged with respect to the tweeter offset.

2. Cut notches for the tweeter terminals on the wider side of the front panel and mark the locations of the driver-mounting holes. Drill holes with a #46 (0.081″) drill bit for mounting each driver with #6 × ¾″ (1.91cm) pan-head sheet-metal screws.

3. File a radius on the back of the woofer hole between the woofer-frame struts for proper woofer breathing. Also file a small chamfer, about ⅛″ (0.32cm), on the front of the woofer hole to allow mounting with caulking cord.

4. Mount the ⅝″ (1.59cm)-diameter × 6⅝″ (16.83cm)-long side-to-side dowel using a tweeter cutout at one end and a port cutout at the other. Enlarge the cutout center holes to ⅝″ (1.59cm) for the dowel, and mount the cutouts with two #6 × 1¼″ (3.18cm) flat-head brass wood screws. Alternatively, just glue and clamp.

5. To secure the back, mount the nominal 1″ × 1″ strips, actually ¾″ × ¾″ (1.91cm) wood, to provide a flush back with the foam-tape back gasket installed. Make the vertical strips full height and the three horizontal strips cut to fit tightly between the vertical. Drill the near-center horizontal strip for the front to back dowels, and notch it at each end to allow passage of a #18 zip cord. Make these notches so that the outside end is 1″ (2.54cm) in from the inner face of the side. Mount these strips against the sides, top and bottom, with 1¼″ (3.18cm)-long aluminum nails positioned to avoid interference with the back screws.

Install ½″ (1.27cm)-diameter front-to-back 7″ (17.78cm) dowels. Keep them below the surface at both ends. Mount the back with #8 × 1¼″ (3.18cm) brass flat-head wood screws countersunk flush with the rear surface of the back board. I used two screws in each horizontal 1″ × 1″, and four screws along each side. Apply 1⁄16″ (0.16cm)-thick foam tape to all the 1″ × 1″ strips, except for the notches in the near-center strip, to provide a back-panel gasket.

Cut a flare on each end of the 2″ (5.08cm) inner diameter × 5½″ (13.97cm)-long PVC port ducts, using a hobby grinder with a small sanding drum. Start these flares about ½″ (1.27cm) down the duct, and leave 1⁄32″ (0.08cm) of PVC thickness at the duct ends. Omitting these flares will mistune the

ABOUT THE AUTHOR

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

TABLE 1
CSA WOOD DIMENSIONS—QUANTITIES FOR ONE BOX

PIECES	DIMENSIONS (SEE FIG. 11)	FUNCTION
2	¾″ × 8½″ × 13¼″ (1.91 × 21.59 × 33.66cm) with bottom cut at an angle—see *Fig. 10*	Sides
2	¾ × 6¾ × 10¾″ (1.91 × 17.15 × 27.31cm)	Front &back
1	¾ × 6¾ × 8½″ (1.91 × 17.15 × 21.59cm)	Bottom
1	¾ × 8½ × 8¼+″ (1.91 × 21.59 × 20.96+cm)	Top—see text
1	¾″ (1.91cm) tweeter-hole cutout	S-S dowel block
1	¾″ (1.91cm) port-duct hole cutout	S-S dowel block
2	½″ (1.27cm)-diameter × 7″ (17.78cm) long dowel	F-B dowels
1	⅝″ (1.59cm)-diameter × 6⅝″ (16.83cm) long dowel	S-S dowel

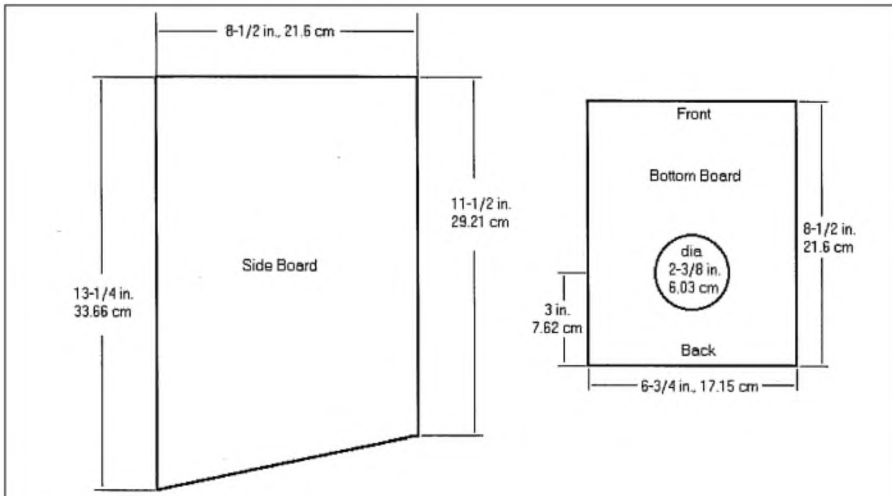


FIGURE 10: View of side and bottom boards for CSA.

boxes. Install a 6″ (15.24cm) × 17″ (43.18cm) piece of nominal ½″ (1.27cm) fiberglass, covering the bottom half of both sides and the bottom, cutting a hole to clear the port duct. Now glue the port ducts into the boxes flush with the bottom board.

Install a 6″ (15.24cm) × 16″ (40.64cm) piece of nominal 1½″ (3.81cm) thick fiberglass covering the top half of both sides and the top (*Photo 4*). After the crossover is assembled on the back, mount a 4¾″ (12.07cm) × 4¾″ (12.07cm) piece of nominal ½″ (1.27cm) fiberglass at the bottom of the back. Install a 2″ (5.08cm) × 5″ (12.7cm) piece of nominal ½″ (1.27cm) fiberglass over the coils on the top of the back (removed for *Photo 4*).

Install the tweeter with silicon rubber on the back of the faceplate, and the woofer using caulking cord rolled and pulled down to about ⅛″ (0.32cm) diameter. Add a diffraction ring to the tweeter according to the following instructions, noting that the tweeter in CSA has about a 3.7″ (9.4cm) diameter × 0.125″ (0.32cm)-thick faceplate. *Table 4* lists the number of rings to use based on the thickness of your material. The inside diameter of each ring is about 3.7″ (9.4cm) as to fit snugly on the tweeter faceplate. *Table 5* shows the outer diameter of each ring, based on the number of rings you are using.

After cutting out the rings, place the bottom (largest) ring around the tweeter and cut it along the interface with

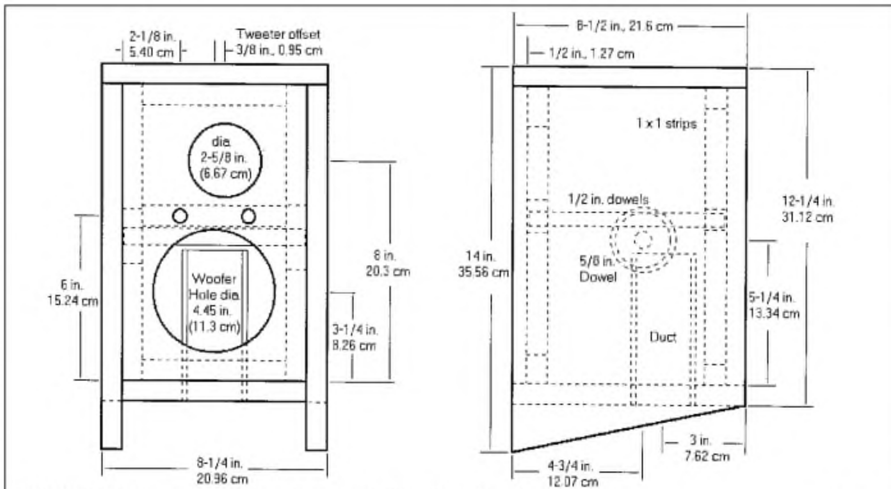


FIGURE 11: Front and side views of CSA, showing locations. B-1762-11

TABLE 2
CSA MATERIALS FOR TWO BOXES

QTY	MATERIAL	FUNCTION
1 roll	Nominal ½″ (1.27cm) fiberglass	Box damping
Some	Nominal 1½″ (3.81cm) fiberglass	Box damping
Some	Parts Express #260-762 latex carpet	FP damping
8	#6 × ¾″ (1.91cm) pan-head sheet-metal screws	Mount tweeter
8	#6 × ¾″ (1.91cm) pan-head sheet-metal screws	Mount woofer
28	#8 × 1¼″ (3.18cm) flat-head brass wood screws	Attach back
4	#6 × ¾″ (1.9cm) round-head brass wood screws	Mount input strip
8	#6 × 1¼″ (3.18cm) flat-head brass wood screws	Mount S-S dowel block
1 box	Aluminum nails, Nichols Hy-Tensil nails, 1¼″ (3.18cm) long	General assembly
1 box	Steel nails, Elco Hardened Steel Panel nails,1⅝″ (4.13cm) long	Box assembly
3yd (2.74m)	Stick-on vinyl	Cover boxes
6′ (1.83m)	Piece of ⅝″ (1.59cm) half-round	Grille trim
1 roll	Thin grille cloth	Cover front
1 tube	Silicon rubber	Sealing & CO part mounts
8′ (2.44m)	high quality nominal 1 × 1 strip	Back mount
1 roll	1/16″ (0.16cm)-thick × ¾″ (1.91cm) foam tape	Back gasket
Some	Sheets of thin cardboard	Diff. Ring
1′ (30.48cm)	2″ (5.08cm) ID SCH40 PVC tubing	Port ducts
Some	Small brads	Mount FP trim
1 roll	Mortite caulking cord—stock #F-4	Woofer seal
Misc.	Wire, screws, and terminals	Package CO

TABLE 3
CSA COMPONENTS, EACH BOX

QTY	ITEM	MODEL/FUNCTION
1	5″ (13cm) woofer, shielded	Vifa M13SG-09-08
1	¾″ (1.9cm) dome tweeter, shielded	Vifa D19TD-03-08
1	8Ω, 15W L-pad, ¾″ (1.91cm)-long shank	Tweeter level control
1	2 or 3Ω, 10W resistor	R1—Woofer low-pass
1	10µF, 100V capacitor	C1—Woofer low-pass
1	0.5mH, 0.22Ω air-core coil	L1—Woofer low-pass
1	0.4mH, 0.15Ω air-core coil	L2—Tweeter high-pass
1	2.5µF, 250V capacitor (100V min)*	C2—Tweeter high-pass
1	1µF, 100V capacitor*	C2—Tweeter high-pass
1	0.1mH, 0.1Ω air-core coil	L3—Tweeter filter
1	4 or 6Ω, 5W resistor	R2—Tweeter filter
1	Two-terminal barrier strip	Terminate input wires

*C2 is two capacitors in parallel for 3.5µF the woofer frame so that it will lie flat on the front panel. Do the same with all the other rings. Then glue the rings together using rubber cement. The cement makes the diffraction ring thicker than expected, so if the top ring extends above the top of the tweeter faceplate, omit it.

Install two layers of black latex carpet for front panel damping. Cut the bottom layer 11¼″ (28.58cm) × 7¼″ (18.42cm), notch the corners, and install it so that the edges fold up to cover the sides of the box's front lip. Cut the top layer to fit within the bottom layer.

Cut the woofer hole with a 5.5″ (13.97cm) diameter in both layers, and the tweeter hole 4.2″ (10.67cm) in diameter in the bottom layer and 4.6″ (11.68cm) in the top (*Photo 4*). This damping material will cover much of the diffraction ring around the tweeter. Testing has shown that the FP damping material is not a substitute for the diffraction ring, and that the ring functions properly when partially covered!

Build the crossovers on the back panel in mirror-image pairs, with the L-pad mounted to the opposite side of the tweeter offset (*Fig. 12* and *Photo 4*). Cut the plastic flanges on L1 at right angles down nearly to the wire level in two places so it will sit flat on the back and clear the 1″ × 1″ strip. Mount the L-pad

(to p. 36)

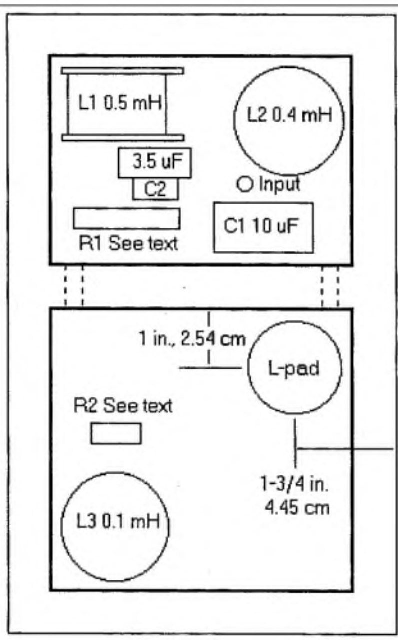


FIGURE 12: CSA crossover layout. B-1762-12

(from p. 33)
with its ¾″ (1.91cm) threaded shank by cutting a ¾″ (1.91cm)-diameter counter bore on the outside of the back to take the nut. Mount the L-pad with the lock washer on the inside, and the flat washer and nut on the outside.

Cover the L-pad threads with silicon rubber to ensure an airtight box and eliminate the possibility that the L-pad will rotate. The nominal L-pad setting is just ½ turn (180°) counterclockwise (CCW) from full clockwise (CW) (0dB). Secure the single #18 zip cord that passes through the notch in the center 1″ × 1″ into its proper location. Seal the input wire through a hole in the back (#1 [0.228″] drill) with silicon rubber and terminate it on a two-terminal barrier strip mounted near center height with brass screws.

Wrap the boxes in stick-on vinyl and then staple the grille cloth to the front edge of the grille area. Then miter nominal ⅝″ (1.59cm) half-round strips to fit around the grille area, covering these with the same stick-on vinyl and attaching them with small brads.

DETAILS OF CSB CONSTRUCTION

Figure 13 shows the sizes of the odd-shaped pieces of ¾″ (1.91cm) particleboard used to construct CSB. As noted earlier, I recommend that the top and front fill pieces be cut large to allow accurate fitting later. Other pieces are defined in *Table 6*. *Figure 14* shows the front and side views of CSB, identifying the locations of the various components. If you plan to cut the bottom flat to use the speakers with stands, the exact height of the side panels will be a function of the actual thickness of the particleboard. I find there is quite a variation in the actual thickness of nominal ¾″ (1.91cm) particleboard.

The woofer is extremely heavy, so it is mounted with #10-32 × 1¼″ (3.18cm) round-head machine screws (hole size #10 [0.193] drill). The four mounting holes for this woofer place the screw heads right against the rubber surrounds, so to avoid damaging it, you need to put washers under the screw heads and tighten the nuts without turning the screws. You may need to file a flat edge on the washers to clear the outer edge of the speaker frame. *Table 7* shows the materials required to build two boxes, and *Table 8* lists the components used in each box.

Photo 5 shows the front and side views of the front-panel/bottom-panel assembly. Also note the following:

1. The woofer really dominates the major portion of this enclosure, with its very large rear shield. The shield is vented, so not as much box volume is lost as this picture seems to indicate.

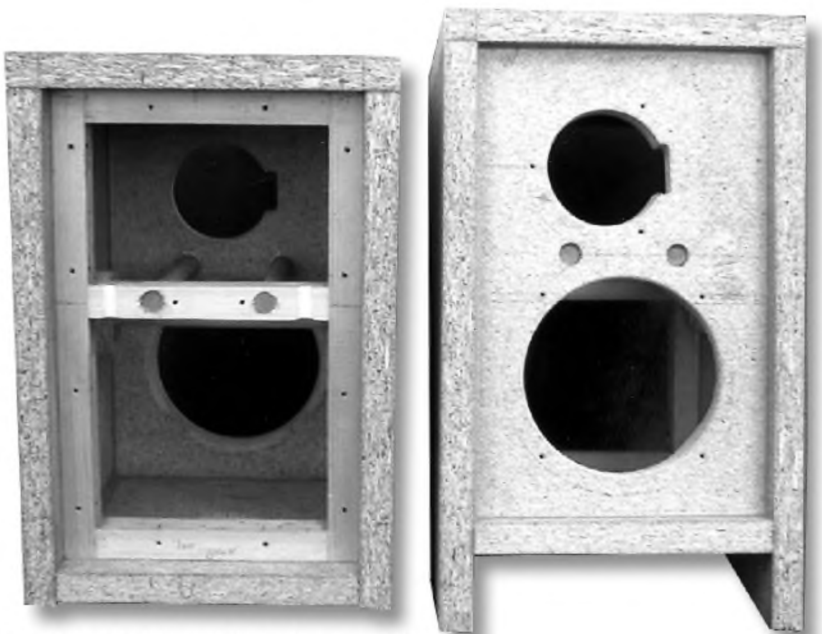


PHOTO 3: CSA inside and front views, no drivers.

2. Notches for the tweeter terminals are sawed and filed toward the wider side of the front panel. The two front panels are mirror-imaged relative to the tweeter offset. Drill the tweeter mounting-screw holes with a #46 [0.081″] bit for #6 × ¾″ (1.91cm) pan-head sheet-metal screws.

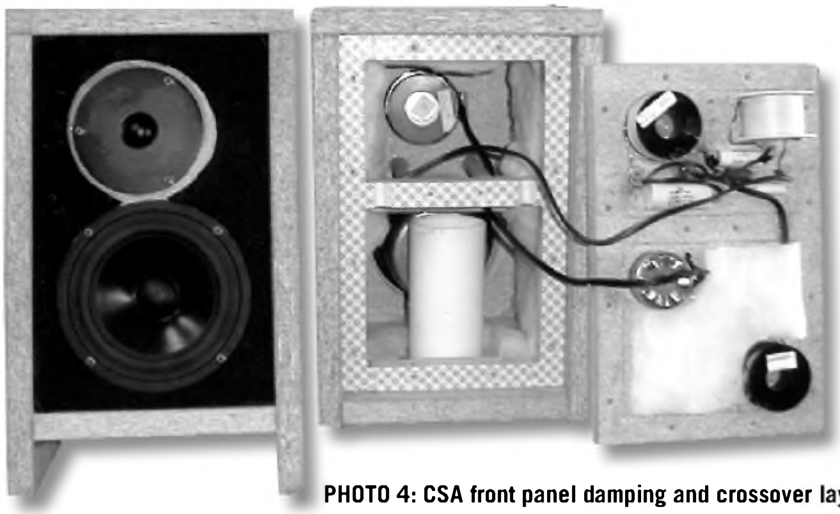


PHOTO 4: CSA front panel damping and crossover layout.

3. The woofer-mounting screws are in a vertical/horizontal pattern, and washers and double nuts are used on the back. Any other positioning of the mounting screws will make getting the nuts on these machine screws almost impossible.

TABLE 4
NUMBER OF RINGS IN DR VERSUS MATERIAL THICKNESS

MATERIAL THICKNESS	NUMBER OF RINGS IN DIFFRACTION RING
Less than 0.018″ (0.46mm)	Use thicker material or two layers per ring
0.018″ - 0.025″ (0.46 - 0.64mm)	Use five rings
0.026″ - 0.032″ (0.66 - 0.81mm)	Use four rings
0.033″ - 0.042″ (0.84 - 1.07mm)	Use three rings
Above 0.042″ (1.07mm)	Use thinner material

TABLE 5
OUTSIDE DIAMETERS FOR EACH PIECE IN DIFFRACTION RING

	3-RING DR	4-RING DR	5-RING DR
1st ring (top)	4.1″ (10.41cm)	4.1″ (10.41cm)	4.1″ (10.41cm)
2nd ring	4.6″ (11.68cm)	4.5″ (11.43cm)	4.4″ (11.18cm)
3rd ring	5.1″ (12.95cm)	4.8″ (12.19cm)	4.7″ (11.99cm)
4th ring	None	5.2″ (13.21cm)	4.9″ (12.45cm)
5th ring	None	None	5.2″ (13.21cm)

4. File a radius on the back of the front panel for the woofer between the mounting-screw areas.

5. Note the holes for the back-to-front ½″ (1.27cm) dowels.

6. The front panel is rather weak because of the large holes in it, so you should handle it carefully until it is installed into the box.

Once the basic box structure is assembled and the glue set, install the side-to-side dowel after making sure the mounting blocks will not interfere with the back 1″ × 1″ strips. The front-to-back dowels are ½″ (1.27cm) in diameter and 5⅝″ (14.29cm) long. The side-to-side dowel is ⅝″ (1.59cm) in diameter and 6¾″ (17.15cm) long. The ¾″ (1.91cm)-thick particleboard blocks for mounting the side-to-side dowel are 1⅛″ (2.86cm) wide × 2½″ (6.35cm) long, with the holes for the dowel in the centers of the blocks.

Install these blocks with the longer dimension vertical and each one attached with glue and two tempered aluminum nails through holes drilled with a #46 (0.081″) drill. Having enough room to swing a hammer to install these nails is difficult, and using brass screws might be easier if you can drill the needed holes in this tight area (*Photo 6*). Just gluing and clamping is also acceptable.

Install the 1″ × 1″ (actually ¾″ by ¾″ [1.91cm]) wood strips. The side strips are full height, and the three horizontal strips are installed to fit tightly between the two vertical strips. Install all strips so that their back faces are positioned to make the back fit flush with the foam tape used to gasket the back.

TABLE 6
CSB WOOD DIMENSIONS—QUANTITIES FOR ONE BOX

PIECES	DIMENSIONS (SEE *FIG. 14*)	FUNCTION
2	¾″ × 7⅛″ × 13¾″ (1.91 × 18.1 × 34.93cm) with bottom cut at an angle—see *Fig. 13*	Sides
2	¾″ × 6⅞″ × 11½″ (1.91 × 17.46 × 29.21cm)	Front & back
2	¾″ × 6⅞″ × 7⅛″ (1.91 × 17.46 × 18.1cm)	Bottom
1	¾″ × 7⅛″ × 8⅜+″ (1.91 × 18.1 × 21.27+cm)	Top—see text
1	¾″ × 6⅞″ × 1½+″ (1.91 × 17.46 × 3.81+cm) with bottom edge cut at an angle—see *Fig. 13*	Front fill—see text
2	¾″ × 1⅛″ × 2½″ (1.91 × 2.86 × 6.35cm)	S-S dowel blocks
2	½″ (1.27cm)-diameter × 5⅝″ (14.29cm) long dowel	F-B dowels
1	⅝″ (1.59cm)-diameter × 6¾″ (17.15cm) long dowel	S-S dowel

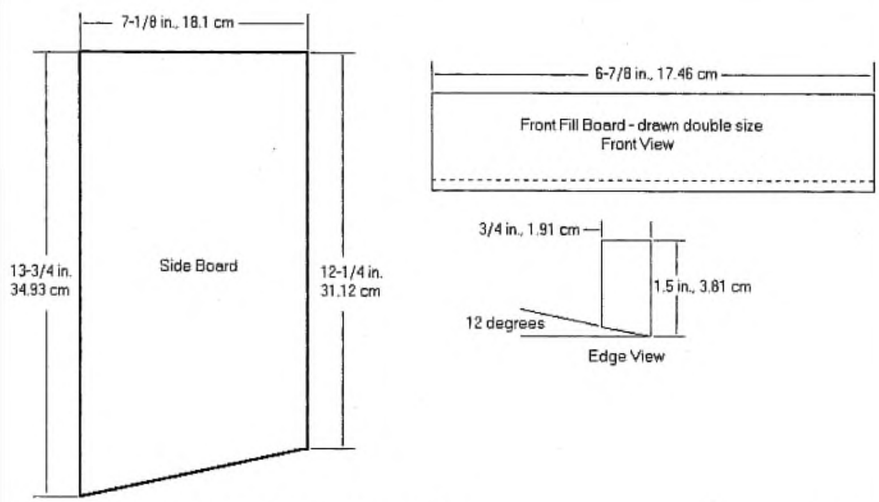


FIGURE 13: Layout of special pieces for CSB. B-1762-13

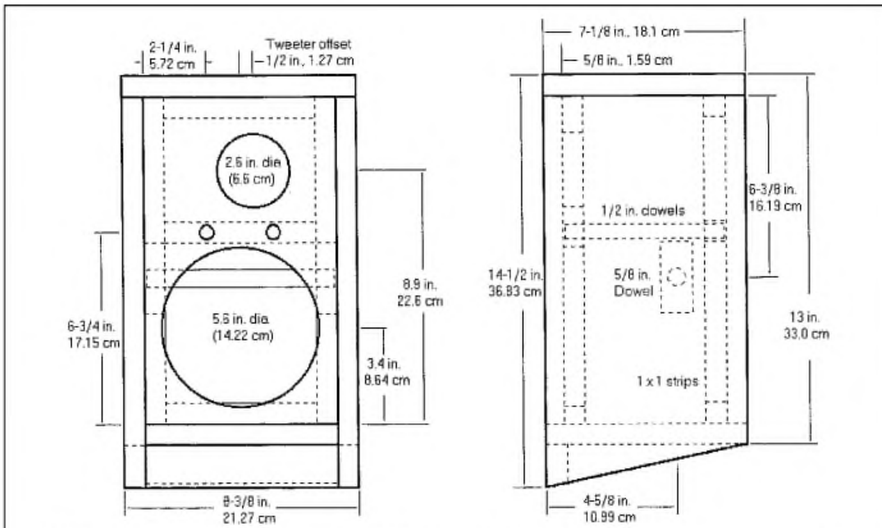


FIGURE 14: Front and side views of CSB, showing locations. B-1762-14

The horizontal strip terminating the front-to-back dowels has holes (½″ [1.27cm]) for the dowels and also notches filed near each end to accommodate an #18 zip cord. You should position these notches with their outer end about 1″ (2.54cm) from the inside edge of the side panels and facing the back board. Install the strips against the top, bottom, and sides of the box with aluminum nails located so that they will not interfere with the back mounting screws (*Photo 6*).

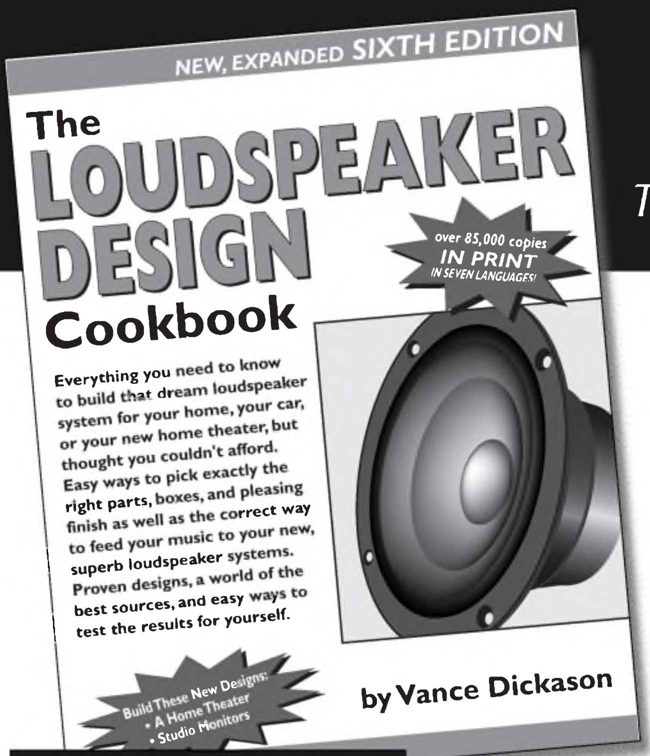
TABLE 7
CSB MATERIALS FOR TWO BOXES

QTY	MATERIAL	FUNCTION
1 roll	Nominal ½″ (1.27cm) fiberglass	Box damping
Some	Nominal 1½″ (3.81cm) fiberglass	Box damping
8	#6 × ¾″ (1.91cm) pan-head sheet metal screws	Mount tweeter
8	#10-32 × 1¼″ (3.18cm) round-head machine screws	Mount woofer
16	#10-32 hex nuts	Mount woofer
16	#10 flat washers	Mount woofer
16	#8 × 1¼″ (3.18cm) flat-head brass wood screws	Attach back
12	#8 × 1¼″ (3.18cm) flat-head steel wood screws	Attach back
4	#6 × ¾″ (1.91cm) round-head sheet-metal screws	Mount input strip
1 box	Aluminum nails, Nichols Hy-Tensil nails, 1¼″ (3.18cm) long	General construction
1 box	Steel nails, Elco hardened steel panel nails, 1⅝″ (4.13cm) long	Box construction
3yds (2.74m)	Stick-on vinyl of your choice	Cover boxes
6′ (11.83m)	Piece of ⅝″ (1.59cm) half-round	Grille trim
1 roll	Thin grille cloth	Cover front
1 tube	Silicon rubber	Sealing & CO part mounting
8′ (2.44m)	High-quality nominal 1 × 1 strip	Back mounting
1 roll	1/16″ (0.16cm) thick by ¾″ (1.91cm) foam tape	Back gasket
Some	Small brads	Mount trim
Misc.	Wire, screws, and terminals	Package CO

TABLE 8
CSB COMPONENTS FOR EACH BOX

QTY	MATERIAL	MODEL/FUNCTION
1	6½″ (16.5cm) woofer, shielded	Eminence 615F2025
1	1″ (2.5cm) dome tweeter, shielded	Audax AW025M1
1	8Ω, 15W L-pad, ¾″ (1.91cm)-long shank	Tweeter level control
1	8Ω, 10W resistor	R1—Woofer Zobel
1	20μF, 250V capacitor (100V min)	C2—Woofer Zobel
1	15μF, 220V capacitor (100V min)	C1—Woofer low-pass
1	0.7mH, 0.28Ω air-core coil	L1—Woofer low-pass
1	0.6mH, 0.24Ω air-core coil	L2—Tweeter high-pass
1	5.6μF, 250V capacitor (100V min)*	C3—Tweeter high-pass
1	1μF, 100V capacitor*	C3—Tweeter high-pass
1	Two-terminal barrier strip	Terminate input wires

*C3 is two capacitors in parallel and measured 6.5μF



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Secure the back with a total of fourteen #8 × 1¼″ (3.18cm) flat head wood screws countersunk to be flush with the back. Use a #41 (0.096″) drill for the screws into the centerline of the 1″ × 1″ strips, and a #15 (0.18″) pass drill through the back. The recommended screw positioning is as follows:

1. Top and Bottom: one screw in 2⅛″ (5.4cm) from each side of the back.

2. Center strip: one screw in 2¾″ (6.99cm) from each side of the back so they are between the two dowels.

3. Each side: down from the top 1⅛″ (2.86cm), 4¼″ (10.8cm), 7⅜″ (18.73cm), and 10⅜″ (26.35cm).

Because of the crossover coils mounted at the top of the box, all the back screws from the center strip upward (eight per box) should be brass, while the bottom six screws can be steel. Of course, all the screws can be brass if you prefer.

I made the gasket for the back panel from 1/16″ (0.16cm)-thick double-sided foam tape with the protection strip left on. Cover the backs of all 1″ × 1″ strips with tape except for the notches filed in the center strip. Cut holes in this tape around each mounting-screw hole so that the screws will not tear up the tape when installed (*Photo 6*).

Glassing for these boxes consists of a 30″ (76.2cm) × 4½″ (11.43cm) strip of

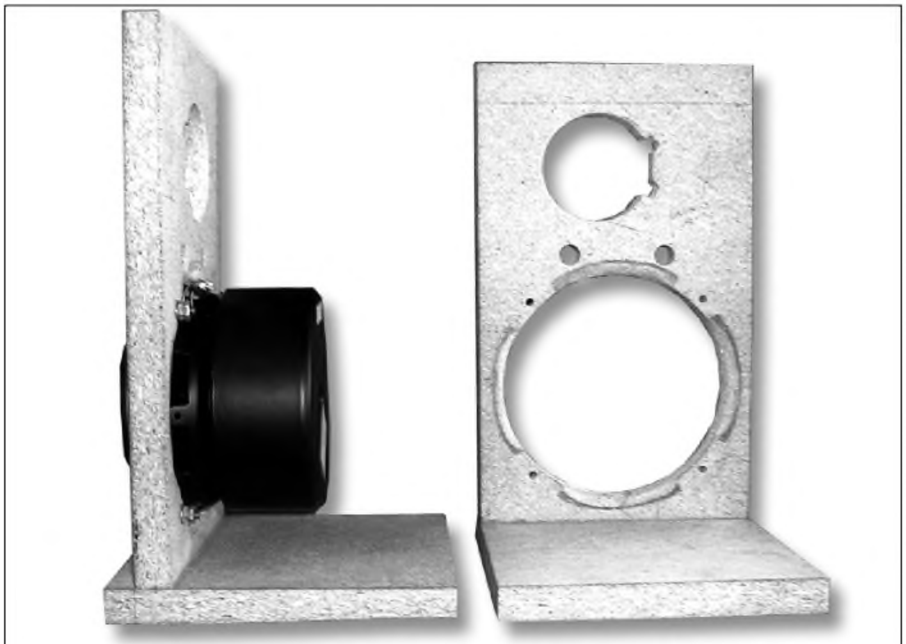

PHOTO 5: CSB front panel/bottom board assembly.

nominal ½″ (1.27cm) fiberglass covering the bottom and both sides of the box. In addition use a 6½″ (16.51cm) × 4½″ (11.43cm) piece of nominal 1½″ (3.81cm) fiberglass to cover the top. After crossover construction, use a 5″ (12.7cm) × 3½″ (8.89cm) piece of nominal ½″ (1.27cm) fiberglass to cover the lower back where no crossover components are mounted.

Also, use a 6″ (15.24cm) by 3″ (7.62cm) piece of the same material to cover the components on the upper back, gluing it to the two coils and—tucking it down between the coils—gluing it to capacitor C3. Note that the piece of fiberglass covering the upper portion of the back panel was removed for *Photo 6*.

At this point, you should loosely install the back and fill the nail holes in the front panel so you can sand the boxes. Only the top, sides, and the front fill board need to be sanded, since you will cover the board edges around the front panel.

Mount the tweeter by inserting #6 × ¾″ (1.91cm) pan-head sheet-metal screws into the drilled holes. Ensure a proper tweeter seal by installing silicon rubber only on the back of its faceplate so you can cut it if necessary to remove the tweeter. Also install the woofer with silicon rubber, again restricting it to the face of the front panel in case you must remove the woofer. Each of the four woofer-mounting machine screws has a washer on the outside, filed to clear the woofer frame, with a washer and two nuts on the inside. Remember to tighten these screws by turning the nuts while preventing the screws from turning. The second nut is tightened against the first to lock its position. When CSB is buttoned up, the woofer cones will be very stiff if you have the box sealed properly.

CROSSOVER LAYOUT

Figure 15 shows the crossover layout. To mount the two large air core coils at the top of the back, it is necessary to cut the plastic bobbin nearly flush with the winding level at two points 90° apart. The crossovers for the two boxes were built mirror-imaged, with the L-pad always mounted on the opposite side from the tweeter offset. Thus when the boxes are finished, you can identify the tweeter-offset side by the L-pad location.

Mount the L-pad with its ¾″ (1.91cm) threaded shank by providing a ¾″ (1.91cm)-diameter counter bore on the outside of the back to take the nut. The L-pad is installed with the lock washer on the inside and the flat washer and nut on the outside. Cover the L-pad threads with silicon rubber to ensure an airtight box and eliminate the possibility that it will rotate. The correct L-pad setting to match the modeling work is about ¼ turn (90°) CCW back from full CW (0dB).

The wires that pass through the notches in the horizontal 1″ × 1″ strip should be secured into position. *Photo 6* shows the mirror-image crossover from the layout in *Fig. 15*. Removed for this photo is the nominal ½″ (1.27cm) fiberglass used to cover the CO components at the top of the back. Mount all the crossover components to the back panel with silicon rubber.

Install the front-panel damping material, which consists of 6″ (15.24cm) of nominal ½″ (1.27cm) fiberglass, cut out

PHOTO 6: CSB front panel damping and crossover layout.

for the woofer and tweeter and covering the top portion of the front panel. I actually used two strips of 3″ (7.62cm) wide fiberglass for convenience (*Photo 6*).

Now put the finish on the boxes. For CSB I used stick-on vinyl. Note that the dome tweeter is unprotected and the fiberglass front panel damping material will get stuck to the vinyl. To solve this problem, I cut a piece of thick cardboard to fit the front-panel area, with lips on the edges to hold it up flush with the front of the box to protect the tweeter.

Apply the stick-on vinyl so that about 2″ (5.08cm) overhangs the back and it covers the bottom of the box. After covering the front fill board with vinyl, cut out the grille area around the cardboard protective insert. When finished, the vinyl covers about 2″ (5.08cm) on the back, the entire front edge and fill board, and wraps under the box.

Next, staple the grille cloth around the front edge of the box. Finally, miter the half-round strips to fit the grille area, cover them with stick-on vinyl, and install them with small brads. Mount a barrier terminal strip low on the box to terminate the input wires, keeping them away from the crossover coils.

USING THE SPEAKERS

As always, you should experiment with the placement of both systems for best sound. I generally liked the systems with the tweeters offset to the outside and the boxes facing straight ahead. The L-pads allow setting the tweeter level to match your room acoustics. Good listening. ❖

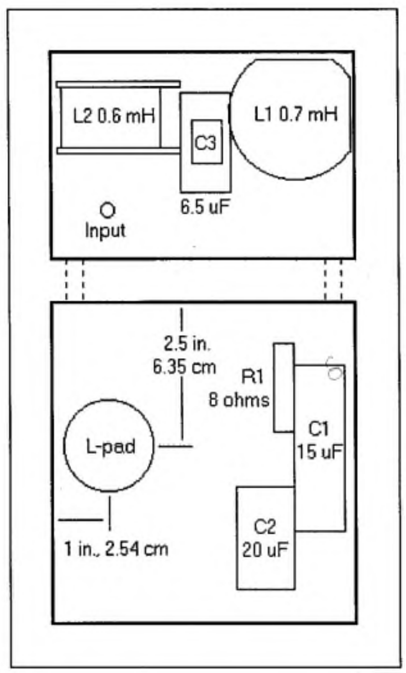

FIGURE 15: CSB crossover layout.

SOURCES

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Digital Class-D Subwoofer Amp, Part 2

Part 1 introduced readers to the specifications of the SA1 digital amplifier kit. We continue here with a look at the theory behind the company's design. **By Thomas O'Brien**

Efficiency is power in (from the power supply) divided by power out (into the load). Because of switching losses, which heat up the FETs, the output stage wastes some power to heat. As power-supply voltage increases, so does heat dissipation. The equations that calculate efficiency are simplified because there are many parasitic losses and complex component interactions that are ultimately negligible. Efficiency is a number from 0 to 1 (0% to 100%).

Efficiency of the output stage is:

$$n_{out} = \frac{1}{\left(\frac{V_S I_{SW}}{P_O} + \frac{R_T}{R_L}\right)}$$

EFFICIENCY OF THE SYSTEM

Powering the modulator circuits and FET drivers reduces efficiency. In the SA1, +12V is used for these circuits, and requires 200mA, or 2.4W. The 2.4W comes from a DC/DC converter that is 75% efficient, so the DC/DC converter requires 3.2W. Even if the output stage were 100% efficient, at 3.2W output into the load, the amplifier as a system would be only 50% efficient. In this example, half the power is wasted in the modulator.

Fortunately, as the output power rises, so does efficiency. At 100W output and 90% output-stage efficiency, the output stage (not including the FET drivers) dissipates 11W (111W × 90% efficiency = 100W out). If the other circuits in the amp dissipate 3.2W, total dissipation is 100W + 11W + 3.2W = 114.2W (87% efficiency).

Efficiency improves as output power approaches maximum. As the amplifier goes into clipping, the average output spends more time in the maximum and minimum pulse-width areas, where the output stage is most efficient.

Consider the difference in the size of the heatsink for a 90%-efficient Class-D amplifier versus a 45%-efficient Class-AB amplifier. The heat dissipated in the 90%-efficient amp is 10% of the output power, and 55% in the 45%-efficient amp. The Class-AB amp requires 5.5 times the heatsinking!

Efficiency of the complete amp system is:

$$n_{SYS} = \frac{1}{\left(\frac{V_S I_{IDLE}}{P_O} + \frac{R_T}{R_L}\right)}$$

Idle current of the amp is:

$$I_{IDLE} = I_{12V} + I_{SW}$$

I_{12V} is the current used by the 12V DC/DC converter. DC/DC conversion from the main supply voltage (36V in the SA1) to 12V is about 75% efficient. For example, if the 12V supply provides 200mA to the FET drivers and modulator circuits, the DC/DC converter would require 89mA from 36V. The calculation used to figure the supply current for the system can also be applied to DC/DC converters (I_S, later in this article).

The 5V supply is linearly regulated from the 12V supply. This is simpler, but less efficient, than using an additional DC/DC converter, and is done to drive the ADC with a low-noise power supply. The 3.3V supply is linearly regulated from the 5V supply, but the voltage drop (only 1.7V) is small enough that efficiency of this conversion is not an issue.

POWER-SUPPLY CONFIGURATION OPTIONS

36V was chosen to drive 4Ω with more than 100W. The recommended supply is regulated, current-limited, switching-power supply. You can use

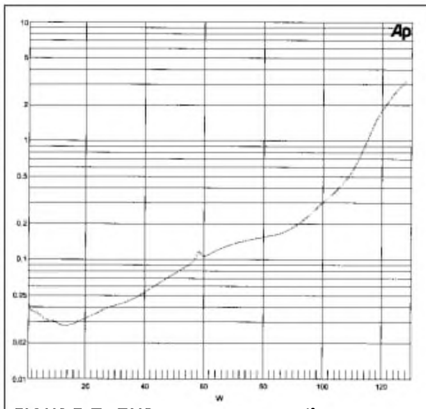

FIGURE 7: THD versus power (input = 100Hz). All figures use load = 4Ω, with Astrodyne power supply.

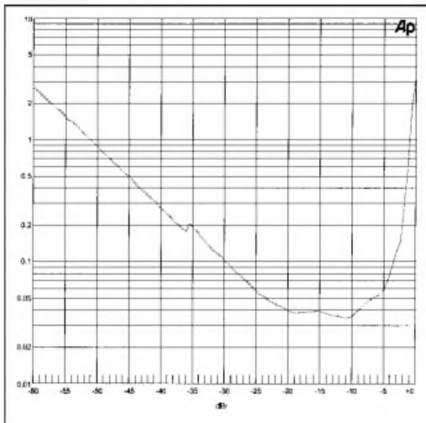

FIGURE 8: THD versus input amplitude (input = 100Hz).

RECOMMENDED POWER SUPPLY

The recommended power supply for SA1 is from Astrodyne.

ASTRODYNE LPP-150-36 (+36V @ 150W)

a linear supply, but it should be regulated to maintain good audio performance. A linear supply may be less expensive, but takes more space, is heavier, and requires custom circuitry to regulate.

If you use a separate 12V supply, the 12V DC/DC converter circuit must be bypassed. See the SA1 kit for details on this modification.

If you use a main supply of more than 40V, the 12V DC/DC converter should not be used because it has a maximum input voltage of 40V.

If lower power output is required, and no external 12V supply exists, you can use a main power supply of as little as 18V. Another option is to use a 12V-only supply, connect the main supply to the 12V, and bypass the DC/DC converter, which has a minimum input voltage of 18V.

POWER-SUPPLY CURRENT LIMITERS

Although switching power supplies are more expensive than linear ones, their outputs are likely to be regulated and current-limited. Regulation ensures that the voltage won't "dip" with increasing load current, and that the 120Hz ripple found in linear supplies is not present. The current limiter in these supplies causes the output voltage either to disconnect momentarily or to dip, keeping the output current below a set value. This protects the power supply and amplifier from damage due to short circuiting the outputs or overdriving the audio input.

You can build over-current protection into the amplifier, but this adds to the series resistance of the output and increases the complexity of the output stage. It was omitted from the SA1 because the recommended power supply is current limited.

It is important that the power supply doesn't use a current-limit circuit that shuts down the supply until AC power is cycled. Suppose that you're watching a DVD that includes an explosive scene. Along comes a loud sound, and the subwoofer turns off. Not only that, but there is no power switch, so you must unplug the unit and then plug it in again to get it working!

POWER-SUPPLY CIRCUIT

The power-supply current required to run the SA1 is not calculated in the same way as for a Class-AB amp. Usually, the maximum current from the power supply is the same, for example, as the peak current through the load at the top (or bottom) of a sine wave driving the amplifier at 1kHz. However, take into account the efficiency of the amplifier, so the power in is nearly the same as the power out. Since the efficiency is high and the voltage driving the amplifier is higher than the output voltage, the required supply current is lower than the output current.

The peak current into the load is:

$$I_P = \sqrt{\frac{2P_O}{R_L}}$$

and the power-supply current at maximum output power is:

$$I_S = \frac{P_O}{nV_S}$$

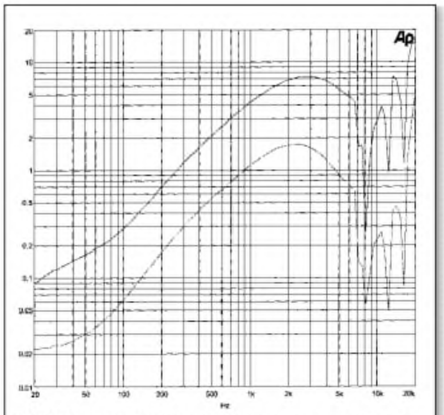

FIGURE 9: THD versus frequency (output = 50W, 100W).

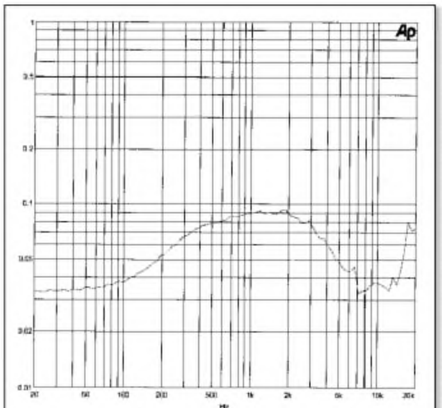

FIGURE 10: THD versus frequency (input = –20dB).

MAXIMUM OUTPUT POWER

The system is designed to run from 36V, but a supply as low as 18V is permissible. The maximum power output before clipping in a bridged configuration Class-D amplifier is:

$$P_O = \frac{PW_{RANGE}^2 V_S^2 R_L}{2R_T^2}$$

PW_{RANGE} is the range of pulse widths from the highest output to the lowest. In the case of SA1, the pulse-width range is $\frac{7}{8}$ (limited digitally to exclude the top and bottom 16th). During peak output voltage, the maximum pulse width is $\frac{15}{16}$, and the A-side of the bridge operates at $\frac{15}{16}$ pulse width. Meanwhile, the B-side operates at $\frac{1}{16}$ pulse width, so $\frac{14}{16}$ of the supply voltage is across the load. The pulse width is limited to ensure controlled operation of the output stage during peak output voltage.

R_L is the load resistance, which is 4Ω. R_T is the total resistance through the FETs, output filter inductors, and load. R_{FET} is R_{ON} (series resistance) of the FET when it is fully turned on. The IRFZ44 FET's R_{ON} is approximately 0.03Ω, and the inductors are about 0.02Ω each (R_{DC}).

The total resistance around the output loop is:

$$R_T = R_L + 2R_{FET} + 4R_{IND} = 4.14.$$

This translates to 0.14Ω output resistance, and the maximum output power of SA1 into 4Ω is:

$$P_O = 0.09V_{SUPPLY}^2.$$

With a 36V supply, the maximum output power into 4Ω is more than 115W.

The LT1776 DC/DC converter has a maximum input voltage of 40V. To provide a higher supply voltage for the output section, you must use a separate +12V supply, and the LT1776 circuit must not be populated. With a 48V supply, the SA1 can output 207W into 4Ω.

POWER OUTPUT AND CLIPPING

The amplifier can output more than the so-called "maximum output power" because this is defined by where the amplifier is at 1% distortion (THD+N). This is where the amplifier starts being overdriven, and the output waveform looks as though the top (and bottom) of the signal is flattened out. If the input signal (sine wave) is large enough, the output will look more like a square wave than a sine wave. In this case, the distortion is high, and the limit of the power to the load is based on the maximum current output of the power supply.

The output devices are stressed during clipping, resulting in extra heat. The audio quality is poor when the amplifier is clipping, but if the power supply is current-limited, clipping should not damage the amp. Power into the load during severe clipping approaches instantaneous peak power.

MORE POWER

IRFZ44 FETs can handle a maximum of 60V, and the FET drivers can handle 125V. Why, then, can't you drive the SA1 with 60V, for example, to deliver 324W into 4Ω? Aside from the expense of the required power supply, the three basic reasons are overshoot, current requirements, and heat dissipation.

You can minimize overshoot on the output by slowing the output transitions (or by using additional circuitry), but you pay a penalty in distortion performance if the transitions aren't fast enough. Either way, a rule of thumb is to stay within 10% of the FET's rated voltage to allow for some overshoot.

A power-supply voltage greater than 48V is not recommended for the SA1. Suppose the efficiency of the output section is 90% into a 4Ω load. It is safe to surmise that most of the heat dissipated will be from the FETs, not the inductors. For example, also supposing that the output power is 300W using a 58V supply, then about 10% of that power (30W) would be dissipated through the FETs. Since the dissipation is distributed evenly among the four FETs, this is 7.5W per FET.

To be safe, your design should handle twice the required dissipation (or close to it). This results in additional heatsinking requirements as the power-supply voltage increases. At 207W using a 48V supply, each FET dissipates about 5W, and a 10W heatsink can be quite large (one for each FET, too). At 115W using a 36V supply, each FET is below 3W dissipation, easily provided by a slide-on "5W" heatsink.

The output inductors are current-limited as well, and can overheat and fail if they are pushed too hard. The inductors used in SA1 are rated at 7.2A RMS and 21A DC. The DC maximum current specification (A DC) of the inductors should be greater than I_P, and the AC maximum (in A RMS) should be greater than 70% of I_P.

INSTANTANEOUS PEAK POWER

A common marketing trick of amplifier makers is to claim huge "peak power" numbers. Actually, instantaneous peak power is twice the output power, unless the output is not a sine wave, or is clipped. At the top and bottom of a sine wave output, the "instantaneous" power is the output voltage divided by the load resistance. The large capacitors in an amplifier's power supply provide the momentarily higher current.

DRIVING 8Ω

Driving 4Ω is more difficult than driving 8Ω because of the series resistance of the FETs and inductors. However, 4Ω was chosen as the ideal load for the SA1 because you can deliver almost twice the power into 4Ω versus 8Ω (with the same power-supply voltage). 4Ω is a common subwoofer impedance, and maximum output power is usually first priority.

The output filter is a tuned circuit designed to drive a 4Ω load. To drive an 8Ω load, the filter capacitors must be half the value. This is described in the SA1 kit.

The maximum output power of SA1 into 8Ω is:

$$P_{OUT} = 0.046V_{SUPPLY}^2$$

SA1 KITS

The SA1 documentation kit (available from Digital Amplifier Company, 1 Turret Drive, Limerick, PA 19468) includes schematics, layout information, full specifications, and hardware description. The full SA1 kit includes the documentation kit, one modulator board, and one output-stage board. The configuration of the hardware drives 4Ω at 115W, and is for use with a single 36V supply.

CONCLUSION

The SA1 subwoofer amplifier delivers high power with low distortion using digital Class-D technology. The modular architecture allows future configuration changes, and flexible mounting in (or on) a subwoofer. The power-supply type and voltage can be varied to produce a range of power levels. High efficiency means compact size and almost no heatsinking. ❖

If you are interested in receiving schematics and tables to build Thomas O'Brien's amp, please send a large, self-addressed envelope with a loose stamp or postal coupon to Audio Amateur Corporation, PO Box 876, Peterborough, NH 03458, or visit our website at www.audioXpress.com

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An Unusual Tone Control

This article appeared originally in *Audiocraft* for August 1957, pp. 4, 48 as one of Marshall's highly acclaimed monthly columns, "The Grounded Ear." **By Joseph Marshall**

From time to time I have reported new developments by others, and I hope I will not be considered presumptuous if I now report one of my own.

One of the most troublesome of unresolved problems in hi-fi design is that presented by the so-called loudness control, which is meant to compensate for the fact that the frequency response of the ear varies with the intensity of the sound. At high intensity levels the response is relatively flat over the whole audible range. But as the intensity level is reduced the response falls off at both the high and low ends until, at very low levels, low bass sounds of the same intensity as the middle-frequency sounds become inaudible. Fletcher and Munson investigated this 25 years ago and prepared the well-known Fletcher-Munson curves from their data.

DIFFERENT TASTES

The Fletcher-Munson curves we see in hi-fi literature are actually abstract curves of several hundred diverse human ears. These abstract curves are extremely helpful in improving our understanding of the processes of hearing, but they are not universal curves applicable to all or even a majority of human ears—any more than the measurements of an average or mean American male are applicable to all or even a majority of American males.

And yet, too many engineers, both amateur and professional, have treated the Fletcher-Munson curves as if they could be applied universally. There might be some justification for such use in strictly communication media (such as the telephone) which do not pretend to provide faithful reproduction. But high-fidelity equip-

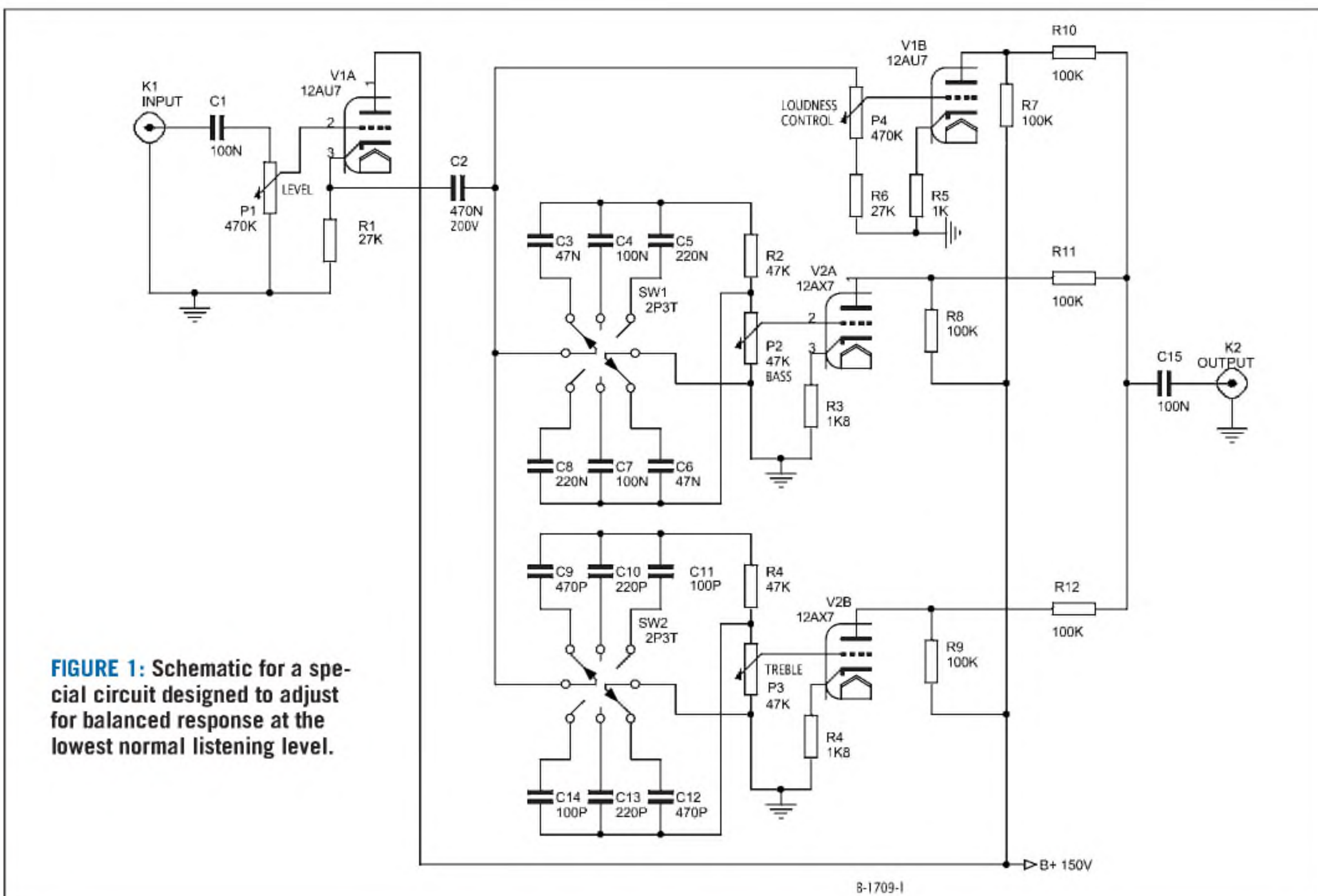


FIGURE 1: Schematic for a special circuit designed to adjust for balanced response at the lowest normal listening level.

ment is intended to furnish faithful reproduction, and a universal application of the Fletcher-Munson curves is about as likely to ensure faithful reproduction for everyone as a suit tailored to the measurements of the average American male would be likely to make every man well-dressed.

This would be true even if all listeners agreed in their preferences. But, as a matter of fact, some listeners prefer a frequency balance approximating that of an orchestra heard very close-up, while others prefer that of an orchestra rather more remote, and this difference requires different degrees of equalization as well as different sound levels.

Actually, if we look into the original Fletcher-Munson data and subsequent research efforts in this field, we will find that the one thing they make very clear is that there is a wide variation in the hearing capabilities of individual human ears. If these studies prove anything of value for high fidelity, it is that the loudness-level contours vary so much both with different ears and with different intensity levels that any equipment which hopes to provide the highest degree of faithfulness for the greatest number of people will have to include a means of varying the loudness contours pretty widely.

ADJUSTMENTS

It is significant that the original attempt to stick closely to the abstract Fletcher-Munson curves on all equipment has been a failure. Only the cheapest and simplest hi-fi equipment today offers a single loudness compensation control. In the better equipment engineers have tried to provide some range of variation.

In some instances this takes the form merely of a switch to disable the loudness control entirely; in others, the loudness control is paired with a level control and, when properly operated, the combination can furnish a sufficient variation to suit many and perhaps a majority of ears. In still others, there are contour selector switches or controls which provide a choice of slopes, as well as degrees of boost. A notably felicitous solution is found in the Marantz control unit in which a

LOUDNESS COMPENSATOR control provides some variation of both the slope and the boost, and when used in conjunction with the excellent tone controls, can meet a wide variety of requirements or preferences.

Many experts, especially in Great Britain, believe that the best way to handle the problem is by means of the tone controls. Also, many owners of American systems with loudness controls have found that they obtain the most satisfying sound by disabling the loudness control and merely adjusting the tone controls. Actually, this approach has great merit. The only trouble is that, if we desire to maintain a truly perfect balance, we must readjust the tone controls every time the volume changes substantially.

TONE-CONTROL SOLUTION

I have inclined to this point of view myself, but have been plagued with the problem of removing the disadvantage ever since I first started to fool around with high fidelity many years ago. For five years now I have been experimenting with a circuit which seems to me to come closer to a satisfactory solution than any other method I have tried. This involves providing greater versatility in shaping the response curve of the system to take care of all the factors which necessitate a modification from flat response. These factors include the hearing curves of the listener, the level at which the sound is reproduced, the nature of the listening environment, any deficiencies in the source material, and, finally, the characteristics of the reproducing system, particularly the speaker.

To achieve this I use two tone-control channels, each of which provides a choice of peaking points or slopes, and a maximum boost of between 15 and 20dB. This provides a flexibility approaching that of the equalizers used in recording and broadcasting. Among other things, it will handle the problem of loudness compensation at any given volume level very nicely, since both the slope and the amount of boost can be varied as necessary.

There remains the problem of disposing of the need for readjusting the tone compensation when the volume level is changed, and the circuit provides a means of doing this which is quite satisfactory. There are two gain or volume controls. One of these maintains the equalization as volume is increased or decreased, and the other can be called a LOUDNESS control, although it works in an opposite manner to the usual con-

PARTS LIST: ONE CHANNEL

QUANTITY	REFERENCE	PART
4	C1, C4, C7, C15	100N
1	C2	470N
2	C3, C6	47N
2	C5, C8	220N
2	C9, C12	470P
2	C10, C13	220P
2	C11, C14	100P
1	K1	INPUT
1	K2	OUTPUT
2	R1, R6	27K
2	R2, R4	47K
2	R3, R4	1K8
1	R5	1K
6	R7, R8, R9, R10, R11, R12	100K
2	SW1, SW2	2P3T
1	V1	12AU7
1	V2	12AX7
2	P2, P3	47K
2	P1, P4	470K

trol. In the conventional loudness control circuit, the system is equalized at a very loud level, and, as the volume is reduced with the loudness control, the bass and/or treble are boosted in relation to the middle frequencies.

STRIKING A BALANCE

In my circuit, the system is adjusted for a balanced or satisfying response at the *lowest* normal listening level. As the LOUDNESS control is turned to increase volume, the bass and treble boosts are progressively washed out until, at maximum volume, the response is flat. This seems to be a preferable method because the ear is not nearly so sensitive to frequency imbalance at very loud levels as it is at low levels. In any case the big problem in home listening is to compensate at low or moderate levels and this method makes possible a far more precise adjustment.

This is achieved by means of the circuit shown in *Fig. 1.* There are three parallel channels fed by the same source and tied together at the output. The uppermost channel is the flat channel, the next is the bass channel, and the bottom is the treble channel. Response of the two lower channels is shaped by networks of the Wien-bridge type. All three channels have gain controls.

It is obvious that the input to the following stage will be the sum of the outputs of all three stages. The gain control in the flat channel is the LOUDNESS control, and is normally at its minimum position, at which point the stage has no gain or a slight loss. The BASS and TREBLE channels provide a direct boost of 20dB or more. The interesting point is this: increasing the gain on the flat channel will start washing out the boost of the lower channels, and when the flat channel is at maximum the boost cannot be more than 6dB.

To provide a choice of response shapes and crossover frequencies, three switch-selected networks are used in the bass and treble channels. The values shown give excellent results, both for tone control and loudness control. Increasing the value of the capacitors moves the crossover downward.

This circuit may be incorporated as part of a control unit, replacing both the tone control and loudness circuits; it can also be used as a separate unit to precede or to follow a present control unit. The two 470k controls can be of the coaxial type to save space. The circuit provides a large versatility in adjustment.

The simplest way to initially set it is as follows: 1. Put the LOUDNESS control and the two tone controls in their minimum volume position; 2. Adjust the LEVEL control to produce the lowest normal listening level; 3. Now, with the tone controls, adjust the BASS and TREBLE for a satisfying balance and over-all sound; 4. Increase volume with the LOUDNESS control as desired. Increasing this control will progressively wash out the bass and treble boosts to compensate for the increased acuity of hearing at both ends as volume is increased. ❖

Editor's note: It might be convenient to use switched step controls for P2 and P3 so as to have resettable positions for a variety of recorded material.

Mapletree Octal 6 Preamp

Reviewed by Charles Hansen and Duncan & Nancy MacArthur

PHOTO 1: Preamplifier front view.

PHOTO 2: Preamplifier rear view.

Octal 6 Duplex Stereo Preamplifier, Mapletree Audio Design (MAD), R.R. 1, Seeley's Bay, Ontario, Canada, K0H 2N0, (613) 387-3830, hollowstate.netfirms.com. Kit: $375 US + S&H; assembled unit: $775 US + S&H, available from FSAudioweb, www.fsaudioweb.com/mapletree.html.

The Mapletree Octal 6 gets its name from the six octal tubes (2—12SN7GT, 2—12J5GT, and 2—OM4/6R7G) it uses. The manufacturer furnishes only North-American- or British-manufactured new old stock (NOS) types. Two complete preamp circuits are incorporated: a classic common cathode-cathode follower cascade (CF) and a shunt-regulated push-pull (SRPP) circuit.

Output and input switching permits any of three sources to be routed to either of these two preamp circuits, or through a passive path that allows use of the volume and balance controls but with no amplification. A separate record output (REC) enables source switching directly to a tape recorder or CD burner. Optionally, you can connect the REC output to a separate headphone amplifier. The manual also suggests using it as a second output for bi-amplification, but this would be difficult to implement since the REC output does not vary with the volume control as the line output does.

PHOTO 3: Preamplifier interior view.

CONSTRUCTION

The preamplifier is extremely rugged, with a bronze-painted heavy gauge steel chassis surrounded by an oak base. The chassis is not fastened directly to this oak base. Four urethane bumpers on the bottom of the chassis sit on brass plates that are screwed to the oak. These plates have 1.25″ round rubber feet that in turn support the oak base. This chassis "float" is noticeable when you tilt the preamp—you can feel it shift slightly in its base.

All controls and connections are on the top of the unit. Brass nameplates with orange letters on a black background provide control, connection, and switch descriptions, while a brass logo plate is located on the transformer case. While there is no finger space under the unit, it is quite light and easy to lift by means of the brass handles on each side of the tubes.

Photo 1 shows the forward part of the panel, which has the power switch, topology selector (SRPP, CF, Passive), balance and volume controls, three-position source selector, and neon indicator lamp. The six tube sockets with two grid cap leads for the output tubes sit just behind the controls. I installed the metallic spray-coated OM4s in the output tube sockets. I also placed one of the ST-shaped glass 6R7Gs in front of the unit for this particular photo.

The rear panel (*Photo 2*) has ten gold-plated RCA input and output jacks and the IEC power receptacle. The unit is furnished with a heavy power cord, and the third pin of the AC receptacle is connected to the chassis. A switch on the rear of the transformer cover allows you to select either 115V AC or 230V AC mains. The line fuse holder (with a spare fuse) is located under the transformer cover.

Photo 3 shows the amplifier with the bottom cover removed. A metal shield surrounds the input/output jacks. All wiring is point-to-point with shielded cable employed in critical signal paths. High-quality components are used throughout: Philips 1W 5% carbon film resistors, Xicon and Philips polypropylene film capacitors, Alps potentiometers, Cinch terminal strips, and an oversize Hammond power transformer.

MAD also sent the detailed 37-page assembly manual. The kit version includes a different prepunched chassis without the wood base, and with all the jacks on the left side and the switches and controls on the front rather than the top. There is a single brass carry handle and an open power transformer rather than the covered one in the factory-assembled unit. However, the circuitry is identical and the kit includes a detachable line cord, all parts, tubes, hardware, fuses, wire, and solder.

The assembly manual is excellent, with instructions for preparing wire, soldering, information on the fasteners, and so forth. All resistors are color-coded to minimize errors. Each assembly step has a box for checking off the items as you complete them. Detailed wiring diagrams as well as photos are provided for the six phases of assembly. The Checkout chapter includes a troubleshooting section in case of problems, with tech support readily available.

TUBE-POLOGY

Mapletree supplied the following NOS tube complement with the review unit:

2—Sylvania 12SN7GT
2—Sylvania 12J5GT
2—Ken-Rad 6R7G
2—Cossor OM4

All the tube boxes except the 6R7Gs were marked for left and right installation.

The user's manual is excellent and includes a complete schematic and parts list along with a detailed two-page circuit description. You can also view the schematic and circuit description at www.fsaudioweb.com/mapletree.html. I will summarize that description here for one channel.

The selected input is connected to the Alps volume and balance control set. The wiper of the volume control is connected through a series 47k5 metal-film resistor to the topology selector switch, which connects the signal to one of two active circuits or the passive path. Tracking resistors (1% metal film) are mounted to taps on the volume controls to minimize tracking errors.

The first active circuit consists

TABLE 1
MEASURED PERFORMANCE

PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Input resistance	50k (max volume)	46k
	100k (min volume)	87k–91k
Output resistance	SRPP: 5.5k (1kHz)	5k7
	CF: 600Ω (1kHz)	540Ω
Voltage gain	SRPP: 13dB (1kHz)	13.2dB, 47k load
	CF: 18dB (1kHz)	19.0dB, 47k load
	Passive: –7dB (1kHz)	–6.9dB
Volume tracking	Within 0.7dB over full range	0.5dB–0.9dB
Frequency response,	SRPP: 20Hz–20kHz (–0.5dB)	15Hz–20kHz (–1dB)
2V out, 100kΩ load	CF: 15Hz–20kHz (–1dB)	16Hz–17kHz (–1dB)
		47k load
Noise (100k load, zero	SRPP: <0.5mV	1.35mV (see text)
input, volume max)	CF: <0.9mV	2.7mV
	Passive: <70μV	80μV
Distortion	<0.1% THD+N, 2V RMS out	0.13% max, 2V RMS
IMD-CCIF (19 + 20kHz)	N/S	0.11% SRPP
		0.063% CF
Maximum output (100k)	SRPP: 25V RMS	26.1V RMS
	CF: 20V RMS	34.5V RMS

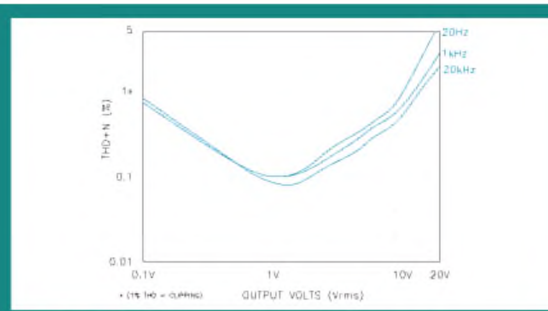


FIGURE 1: THD+N versus output power, CF mode.

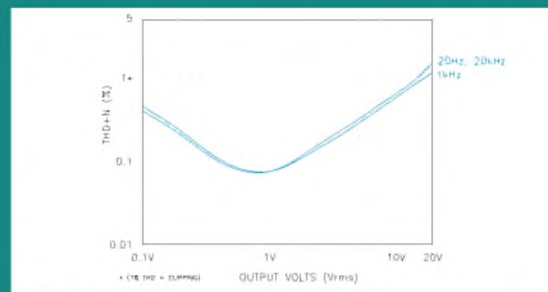


FIGURE 2: THD+N versus output power, SRPP mode.

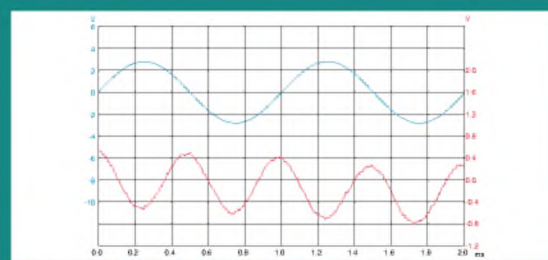


FIGURE 3: Residual distortion, CF mode.

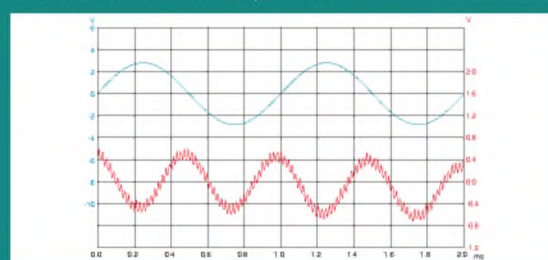


FIGURE 4: Residual distortion, SRPP mode.

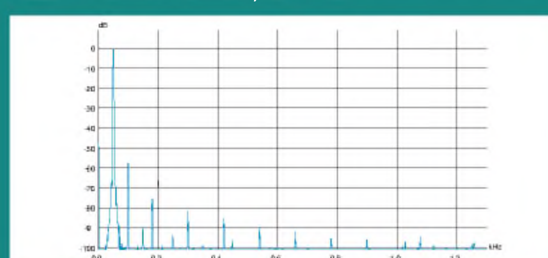


FIGURE 5: Spectrum of 50Hz sine wave, CF mode.


FIGURE 6: Spectrum of 50Hz sine wave, SRPP mode.

Reviewed by Duncan and Nancy MacArthur

The Octal 6 consists of two entirely separate active preamplifiers (common cathode/cathode follower, or CF, and shunt regulated push-pull, or SRPP). A three-position switch selects either of these circuits or a passive mode with no gain. In addition, the manufacturer supplied alternate tubes (ceramic packaged OM4s) for the CF stage; thus we listened to four different configurations.

METHODOLOGY

While a passive option can be added to any preamplifier for the cost of a switch, placing two entirely separate active circuits in a single enclosure is less cost-effective. Regardless of which active option you eventually choose, you may end up paying for a preamplifier circuit not used past the auditioning stage. You might prefer to put the money into a single preamplifier that will be used consistently.

Since our system is optimized for operation with no preamplifier, testing a preamplifier such as the Octal 6 required special care. Our DAC (a modified Philips DAC-960) has a 2V maximum output, and the amplifier used for this review (a Manley Stingray) has an input sensitivity of 185mV (maximum). Thus, an added preamplifier gain stage is not required. In addition, the combination of the grounded AC plugs of the amplifier and Octal 6 caused a minor ground loop.

We listened to the Octal 6 in the following manner:

1. Our "reference" consisted of two pairs of RCA cables joined by a pair of Radio Shack gold-plated adapters.

2. For listening tests, the Octal 6 replaced the gold-plated adapters.

3. When listening to the passive "preamplifier," we turned the volume control on the amplifier fully up.

4. For listening to both active preamplifiers, we set the volume control on the amplifier to the "12 o'-clock" position. This reduced the ground loop hum and helped match the voltage levels.

FIRST EXPERIENCE

Our first sample of the Octal 6 preamplifier had suffered shipping damage and was not operational. The transformer PC boards were not adequately secured in this version. Also, the first sample used partially insulated grid caps that in our opinion presented an unacceptable risk to the consumer.

We returned this sample to the manufacturer, who immediately fixed both problems. All other commentary in this review refers to the revamped version. If you own the previous version with the partially insulated grid caps, we recommend that you consider replacing these with the fully insulated variety.

AESTHETICS AND PACKAGING

Both samples of the Octal 6 came packed in Styrofoam "peanuts"; this type of packing is not as trouble-free as custom foam inserts, but the peanuts are acceptable if used correctly.

If aesthetics are important, then the Octal 6 is probably not the preamplifier for you. Although the Octal 6 appears to be adequately constructed, the level of "fit and finish" of this preamplifier is more typical of a "home-brew" project than of a commercial product. But if sound is more important to you than aesthetics, read on.

The Octal 6 is packaged in a four-sided wooden enclosure surrounding a metal chassis on all sides (but leaving the top and bottom exposed). This type of packaging can be quite effective for power amplifiers with few connectors and no switches, but packaging the Octal 6 in this manner forces all the input and output jacks and all four controls onto the top surface along with the hot tubes. In particular, the volume control is located close enough to the tubes that you can feel significant heat while operating the control. This construction technique does not add to the aesthetic appeal of the preamplifier.

The volume and balance controls on the Octal 6 work smoothly and quietly. The "feel" of these controls is much better than you would expect from looking at the unit's exterior. Our only caveat is that the balance potentiometer does not have a center detent; we spent some time searching for a truly balanced condition.

We ended up with the balance control noticeably off-center in order to achieve proper left/right balance. Although the "topology" selector switch would seem to allow "hot" switching, which, the manual implies, is permissible, we always turned the volume all the way down before switching topologies.

LISTENING EXPERIENCE

The Octal 6 requires a moderate amount of burn-in. As our sample had been used previously, the manufacturer recommended a burn-in period of "a couple of days;" therefore, we used the Octal 6 with a variety of recordings for about 20 hours before doing any serious listening. The sound of the Octal 6 changed markedly during this burn-in period. Out of the box, the SRPP stage sounded unbearably bright and the bass response of the CF stage was muddy and excessive. The burn-in process reduced both of these characteristics significantly.

As in earlier reviews, we listened carefully to several tracks from the *Hi-Fi News and Record Review* disk III. These included track 2: Parry's "Jerusalem," track 4: Vivaldi's "Trumpet Concerto in C," tracks 5 and 6: excerpts from Prokofiev's "Peter and the Wolf," track 7: Purcell's "Welcome, Welcome Glorious Morn," track 10: a Corkhill percussion piece, and track 14: a Rio Napo RSS demonstration. We also listened to favorite tracks from a wide variety of musical genres, ranging from classical to rock 'n' roll. In all cases the Octal 6, as well as the amplifier, was turned on at least 30 minutes before serious listening.

To put the following comments in perspective, during casual listening we could not easily determine which (if any) of the preamplifier circuits was playing. For this type of use, both the CF and SRPP stages simply provided gain. During careful listening, differences became apparent.

In a nutshell, the passive "preamplifier" is the best-sounding mode of the Octal 6 in every area. The CF preamplifier sounds like a classic tube circuit in that the midrange response is very good, the bass is slightly muddy and not terribly deep, and the highs are clean but slightly rolled-off. The SRPP has more of a "hi-fi" sound with an improved but still a little fuzzy bass response and extended, but slightly bright, high-frequency response. The midrange response of the SRPP is good but not as engaging as in the CF and passive modes.

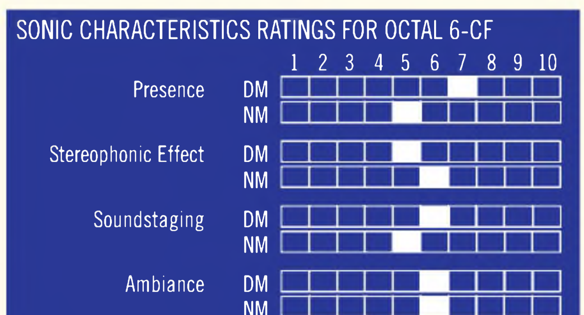
The passive stage doesn't add much to the sound of the amplifier to which it's connected. The sound in this mode was the all-around winner in the Octal 6; however, the potential buyer isn't spending $775 for the passive stage. Thus, in the remainder of this review, including the sonic characteristic charts, we have concentrated on the active modes.

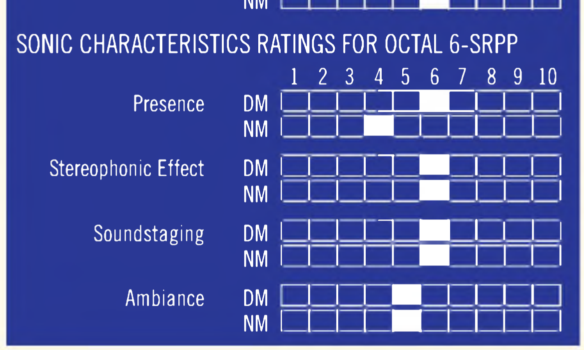
Both active stages reproduced instruments in the midrange (such as strings and winds on the Prokofiev pieces) well. The dynamics and transients in both the Corkhill piece and the Rio Napo demonstration were fairly realistic: the drums sounded like drums.

The CF stage reproduced instrumental sounds naturally as evidenced by the trumpets in the "Trumpet Concerto" and had the ability to separate the tenor and bass voices in the Purcell piece. On the other hand, Nancy noticed a subtly harsh sound with "Honey Baby Blues" (Doc and Merle Watson, *Pickin' the Blues*, Analogue Productions compact disk, CAPFG 026.) The ambience recovery on "Jerusalem" and the "Trumpet Concerto" was fair, providing some sense of the space surrounding the instruments.

Although the CF stage presented a wide sound stage, the placement of the instruments within this stage was not terribly precise. The bass was ample but a bit muddy or muffled, and the highs were smooth to the point of being slightly recessed. The OM4 tube option provided very similar sound to the standard 6R7 tubes; if anything, the sound was more euphonic with the optional tubes.

The bass response of the SRPP stage was better than the CF. The tympani and drums on "Peter and the Wolf"





were deep, realistic, and well defined. The highs were much "flatter," to the extent that they occasionally verged on harshness.

The SRPP stage provided very little sense of space on either "Jerusalem" or the "Trumpet Concerto." With the SRPP the images appeared to be "tied" to the speaker positions. The SRPP was not as natural as either of the other modes—the voices and harpsichord on "Welcome, Welcome Glorious Morn" were more mixed together and less readily identifiable.

FINAL THOUGHTS

NM: Should you buy this preamp? It depends. First, ask yourself honestly what sort of listener you are. Do you love to fiddle with different components, listening carefully to the subtle differences that each one produces? If so, the Octal 6, which provides several preamps in one box, is a good buy at $775 for the assembled version and a complete steal at $375 for the kit. (Caveat emptor: we have neither built the kit nor seen the instructions. We're assuming the completed kit will sound much like the assembled version.)

Or do you prefer to choose one component carefully for optimum sound and stick with that component for years? In this case the answer is less clear. A number of good preamplifiers are available in the under-$1000 price range, and you should audition several of these before you buy. If, on the other hand, you're willing to build the Octal 6 kit, this preamplifier will provide good sound for this price range.

DM: If "fit and finish" or construction quality is an important consideration, the Octal 6 is probably not the preamplifier for you. However, the sound quality belies the construction quality of this product. If you need the extra gain of an active preamplifier, the Octal 6 provides good sound for the money, especially as a kit. If you don't need the gain, use the passive stage or buy a good potentiometer and a few jacks and install them in the enclosure of your choice.

of a 12J5GT common-cathode gain stage that is direct-coupled to a self-biased 0M4/6R7G cathode-follower (CF) output stage. This tube was originally designed as a detector/first audio tube back in my grandfather's time, and includes two diode plates (grounded in this application). The output is taken from the cathode and capacitively coupled to the load through two 220nF polypropylene caps in parallel (440nF total).

If you select the SRPP mode, the volume control is connected to the grid of the first half of the 12SN7GT. The second half of this tube is in series and carries the same bias current. The output of the lower triode is directly coupled to the grid of the upper triode, which operates 180° out of phase with the lower half (hence the term push-pull).

The upper triode serves as the load for the lower triode and is configured as a cathode follower. Its high load resistance maximizes the gain of the lower triode. The cathode of the upper 12SN7GT is capacitively coupled to the output, again with two parallel 220nF caps.

While the CF output impedance is 600Ω, the SRPP output impedance is 5k5. The passive output impedance varies with the volume/balance controls, but is never less than 23k. The manual recommends a power amplifier input resistance of 50k or greater.

As I mentioned earlier, the record output jacks are connected to the selected input signal and do not vary with the volume control setting.

The power supply uses a full-wave fast-recovery solid-state diode bridge and a capacitor input filter to generate +337V DC. This high voltage is split in two paths, one for each channel, and stepped down through R-C filters to +200V DC for the tubes. A separate filament transformer with two series 6.3V AC windings is rectified and filtered to 11.6V DC for the 12V tube heaters. The two 0M4/6R7G output tube heaters are connected in series and grounded at their common connection.

MEASUREMENTS

I chose the 0M4 output tubes, then ran both channels in CF mode at 2V RMS for one hour into 47kΩ. The Octal 6 inverts polarity in CF and SRPP modes, but preserves polarity in passive mode. The unit was still quite cool after this run-in period.

The input impedance measured 87k left channel and 91k right channel at minimum volume. At maximum volume, both these numbers dropped to 46k. The CF output impedance measured 540Ω at 1kHz, while the SRPP output impedance was 5k7 at the same frequency.

At 20Hz the output impedance was about 10k, reflecting the increased low-frequency impedance of the 440nF output capacitor. In passive mode, the output impedance varied from 23k5 to 37k with the volume setting. The connection from any selected input to REC output was 0Ω.

The Octal 6 showed unity CF gain when the volume control was set at approximately 12 o'clock, and 1 o'clock for the SRPP mode. I made all the measurements at a volume setting corresponding to 2V RMS output with a 1V RMS test signal (6dB gain). There was a low-level hum with my ear against the speaker, but the Octal 6 was quiet from my listening position. There was no sound during power-up or shutdown.

I recorded the frequency response in both CF and SRPP mode with a 47k load. The 440nF output coupling cap will roll the low frequency off −3dB at $f = 1/(2\pi RC)$. In CF mode the frequency response was within ±3dB from 8Hz–35kHz, with 0dBr defined as 2V RMS at 1kHz into 47k. It was −0.5dB at 20Hz and −1.2dB at 20kHz.

You must be aware of input impedance when using the Octal 6 with a solid-state power amp, as Mapletree recommends 50k or more. With a 10k load, the low-frequency response was down −3dB at 37Hz.

In SRPP mode the frequency response stayed within ±3dB from 9Hz to 42kHz, with 0dBr again defined as 2V RMS at 1kHz into 47k. It was −0.6dB at 20Hz and −1dB at 20kHz. In both modes, HF response rolled off gradually above 20kHz with no additional gain peaking.

The Octal 6 provides a maximum of 19.1dB gain in CF mode, and a lower 13.2dB in SRPP at 1kHz into 47k. The CF mode gain varied only 0.2dB for loads of 10k–100k, but the SRPP gain spread was 3.6dB for this same load range, reflecting its higher output impedance. The passive mode showed an insertion loss of −6.9dB with a 47k load. Volume control tracking was good, with only 0.5dB difference between channels at 2V RMS output, increasing to 0.9dB at low levels.

Hum and noise (maximum volume, input shorted) measured 2.7mV in CF mode, 1.35mV in SRPP mode, and 80μV in passive mode, power on. Viewing the noise content on an oscilloscope showed a 60Hz waveform with noticeable spikes near each power-line zero crossing.

Right-channel crosstalk from the left output in both modes was much higher than the other direction, being only −28dB at 20kHz. This may be because the right input jack is adjacent to the left output jack. In SRPP mode only, the left-channel crosstalk from the right output actually decreased a bit with increasing frequency, from −50dB at 1kHz to −52dB at 20kHz. I never saw that before.

DISTORTION

Distortion performance was essentially identical for both channels, so I will present the right-channel data. *Figure 1* shows THD+N versus output voltage into 47k at 20Hz, 1kHz, and 20kHz, in CF mode. I engaged the test-set 80kHz low-pass filter to limit the out-of-band noise. The same data for SRPP mode is shown in *Fig. 2*.

The distortion waveform for 2V RMS into 47k at 1kHz, CF mode, is shown in *Fig. 3*. The upper waveform is the amplifier output signal, and the lower waveform is the monitor output (after the THD test-set notch filter), not to scale. The distortion residual signal shows mainly the second harmonic, with just a bit of high-frequency noise at the peaks.

You can also see from the difference in height of the peaks that the residual signal is being modulated by a lower frequency. When I increased the horizontal time base from 200μs/div to 2ms/div (not shown) the modulating waveform was the same 60Hz with spikes at zero crossing as I saw in the hum and noise signal. THD+N at this point is 0.12%.

Figure 4 shows the same data for SRPP mode. This distortion residual signal again shows mainly the second harmonic, but with a high-frequency signal riding on the sine *(to page 80)*

Marchand Tube Crossover

Reviewed by Charles Hansen and Ken & Julie Ketler

PHOTO 1: Crossover and power units stacked and connected.

Marchand Electronics Inc., PO Box 473, Webster, NY 14580, 716-872-0980, info@marchandelec.com, www.marchandelec.com. Kit, $795 US; assembled, $1695 US. Dimensions: 17″ W × 2.8″ H × 8.5″ D. Limited two-year warranty.

The Marchand XM126 Tube Electronic Crossover (CO), with its accompanying XM127 Power Supply (*Photo 1*), is a fourth-order constant-voltage design that provides low-pass and high-pass outputs for two-channel stereo operation. The XM126 is available in two-way, three-way, or four-way versions.

Replacing the R-C frequency module boards will change the CO frequency. Each frequency module contains resistors and capacitors that set the CO frequency and slope for that channel. The standard slope is 24dB/octave fourth-order constant voltage (Linkwitz-Riley), but modules with first-, second-, and third-order slopes are also available.

The review unit was the two-way version, pre-assembled and preset for a 100Hz, 24dB/octave slope. It also had the XLR balanced input option. The power cord was not furnished.

PHOTO 2: PS127 power supply interior view.

CONSTRUCTION

Both units are constructed of heavy-gauge black painted steel, with white silk-screened lettering. The CO stacks on the power supply, and the two units are interconnected by means of a short seven-pin cable with bayonet-type circular plugs. The units are fairly light, and the four rubber feet provide adequate finger space under each unit for easy lifting.

Photo 2 shows the power-supply unit with the cover removed, with the interconnecting cable in the foreground. The front panel has a power switch and three red LEDs, marked Filament, Plate, and Out (output). The rear panel (not shown) has a circular seven-pin female DC power-out connector, an IEC power receptacle with center pin connected to the chassis, a 115/230V AC mains selector switch, and a 2A fuse holder. A large toroidal power transformer occupies the left side of the power-supply chassis, with the power supply and control board to its right. The circuit ground is connected to the chassis through 50Ω.

Photo 3 shows the CO unit front panel, with left- and right-channel CO sections. Each section (in the two-way version) consists of a low-level and a high-level dB control. Each Alpha carbon film control pot is variable from −20dB to +6dB. The fully counterclockwise position is marked "OFF," and the 12 o'clock position is "0dB." A SUM toggle switch on the left side sums the two LP outputs to mono at the right low-output jack so you can drive a single subwoofer amplifier.

The rear panel (*Photo 4*) has one pair of RCA unbalanced input jacks, two sets of RCA output jacks (high out and low out), and two Neutrik balanced XLR input connectors. A slide switch selects either balanced or unbalanced inputs. The RCA jacks are high-quality gold-plated Teflon™-insulated types. Space is available for center high- and low-output RCA jacks for the more complex CO versions.

The left side of the rear panel has a seven-pin male DC power-in connector and a small 12V DC cooling fan with an on-off switch. A row of cooling vent holes is located across the top-rear of the chassis.

Photo 5 shows the CO unit with its top cover removed and the three circuit boards (in the two-way version). The board functions are (L-R): low-pass CO, high-pass CO, and the preamp board. The vertical plug-in boards on the two CO boards surrounding the three tubes are the R-C filter components that determine the CO frequencies.

KIT MANUAL

While the XM126/127 pair was fur-

PHOTO 3: XM126 front view.

PHOTO 4: XM126 rear view.

nished pre-assembled, I studied the 24-page assembly manual from the kit-builder's point of view. The manual is very complete, with clearly written instructions and computer-drawn illustrations. Some of the halftone photos are a bit grainy, but this would not affect construction.

The seven-page user's guide provides diagrams and instructions on how to connect the CO to various audio systems: two-way or three-way, single-sub, and systems with passive COs. The booklet gives detailed procedures on how to set the crossover frequencies and slopes, levels, and (for those CO units that incorporate them) damping controls.

The factory construction is very neat, and should be easy for the kit builder to duplicate. The only thing I would change is the length of the four pan-head screws that attach the rear panel to each chassis. Less than the full length of the screw comes through the retaining nuts. (I come from the aerospace industry, where at least two screw threads must be exposed in any fastener. Admittedly, the XM126 is not likely to experience a carrier landing anytime soon.)

TUBE-POLOGY

XM127 Power Supply

When power is turned on, filament power is immediately available from a +12.6V DC regulator. The assembled versions of the two-way COs sometimes use a linear IC regulator, as did the review unit. The two-way kits and all three- and four-way versions use a discrete zener/power transistor regulator, which provides the higher current needed for additional tubes. The 12.6V DC source also supplies power to the time-delay control circuitry, which uses three sections of a quad LM339 comparator IC and two relay driver MOSFETs.

The center-tapped power transformer HV winding is full-wave rectified to develop ±180V DC symmetrical about ground. These voltages are sent through a time-delay relay that, after a ten-second delay, connects the HV to the XM126 CO via the DC power-out connector. After another two-second delay to allow the tube circuits to stabilize, the output relay in the CO unit is energized. Front-panel LED indicators are lit at each stage of the power-up sequence.

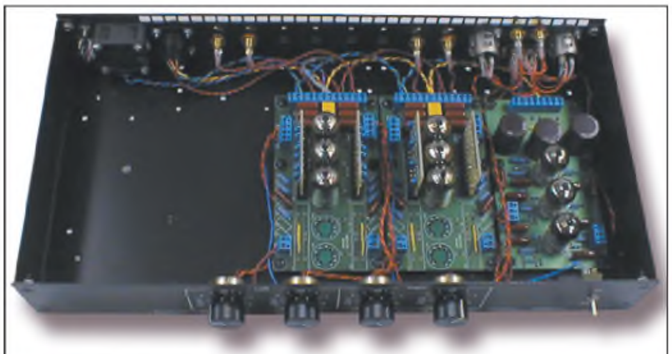


PHOTO 5: XM126 interior view.

TABLE 1
MEASURED PERFORMANCE

PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Frequency response	5Hz–100kHz ±1dB	LP 4Hz–48Hz ±1dB HP 180Hz–92kHz ±1dB
Crossover frequency	20Hz–5kHz options	100Hz (preset)
Total harmonic distortion	0.1% or better—1kHz	See *Fig. 2*
IMD—CCIF (19 + 20kHz)	N/S	0.40% CCIF, 1V RMS
MIM (9 + 10.05 + 20kHz)		0.17% MIM, 1V RMS
Insertion gain	0dB (1X)	0dB combined LP, HP
Filter slope	24dB/octave (standard)	
Signal to noise ratio	Better than 110dB	
Input impedance	500k	503k at 1kHz
Output impedance	500Ω typical	904Ω–1k3 at 100Hz
Input voltage, max	25V p-p, 8.5V RMS	
Output voltage, max	50V p-p, 17V RMS	

XM126 Crossover

The CO uses polypropylene caps and 1% metal-film resistors for the active LP, HP, and (if used) BP filters. Tables in the kit manual specify the component values for your chosen CO frequencies.

The following description applies to one channel. The selected input (balanced or unbalanced) is capacitively coupled to a 12AX7A differential amplifier that is cascaded to a cathode-follower driver tube. The cathode follower is direct-coupled

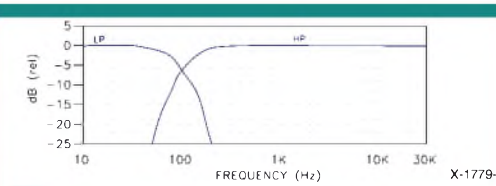


FIGURE 1: Crossover output response versus frequency.

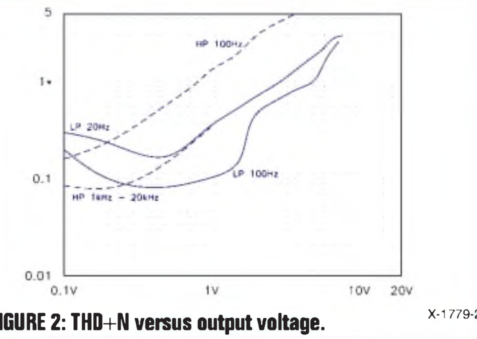


FIGURE 2: THD+N versus output voltage.

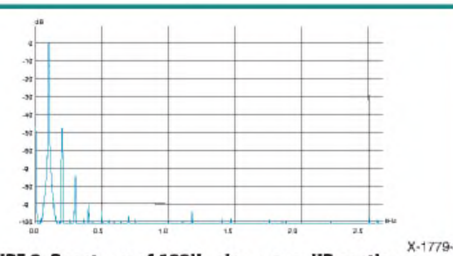


FIGURE 3: Spectrum of 100Hz sine wave, HP section.

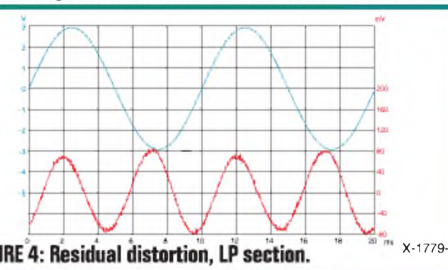


FIGURE 4: Residual distortion, LP section.

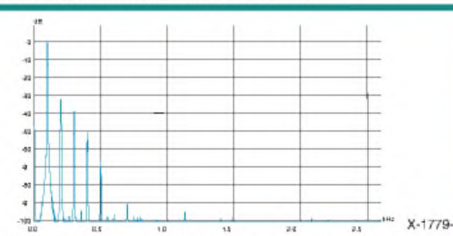


FIGURE 5: Spectrum of 100Hz sine wave, LP section.

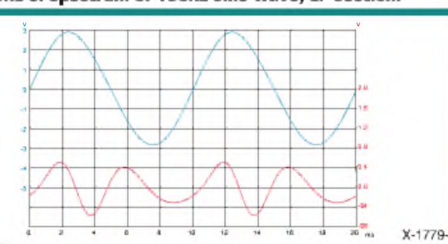


FIGURE 6: Residual distortion, HP section.

Reviewed by Julie and Ken Ketler

Please forgive us if we're getting too personal here, but we'd like to ask you a potentially touchy question. Are you currently bi-amping your audio system? If you are not, we promise not to become preachy about the benefits, but suffice it to say that when we bi-amplified our system six years ago, it was nothing short of a life-changing experience.

Perhaps we exaggerate...a little. But, oh, how much smoother our in-room bass response became when we used a bass-only enclosure (it's not quite a "subwoofer") placed in the corner. Also, the mids got a great deal cleaner when we relieved the main loudspeakers of their low-frequency duties.

Oops, see, there we go. And we promised! Our current system is split with a Marchand XM9 solid-state crossover. When Publisher Ed Dell asked us to review Marchand's XM126 tube system, visions of sibling rivalry danced in our heads, but it was impossible to say no.

SAINTS GO MARCHAND IN (TO OUR LIVING ROOM)

When the unit arrived, it was difficult not to notice that it came in two boxes! The first box housed the power supply that is approximately the same rack size as the crossover itself (in box 2). A large warning label taped to the cover of the crossover stated that a 6″ mating power cable was inside of the chassis in addition to some packing material to protect the tubes for shipping. Both needed to be removed before operation.

The Marchand XM126 can be configured as a two-, three-, or four-way system with possible crossover frequencies from 20Hz–5kHz. Although it comes standard with fourth-order Linkwitz-Riley filters, the XM126 can be configured to operate with third-, second-, or first-order slopes by replacing internal modules. The review unit we received was set up for two-way operation with a crossover frequency of 100Hz at 24dB/octave. In case you are wondering, our own XM9 unit is configured the same way!

The two-way XM126 uses a total of nine 12AX7 tubes and a solid-state regulated power supply for the filament voltage. Power is provided with a heavy-duty toroidal power transformer.

The rear of the unit holds gold-plated RCA jacks for the inputs and outputs as well as XLR input connectors for those of you running balanced setups. Since neither of us has ever been described as being particularly well "balanced," we'll stick to the RCAs. The XM126 uses a 6″ umbilical cord-like cable that connects the power supply and crossover unit. To keep cool, the crossover has a small, switchable air circulation fan.

The front face of this two-way system includes low- and high-frequency control pots for each stereo channel, intended for a "set and forget" type of operation. Also, an ON/SUM switch on the rightmost side of the faceplate allows you to combine the low-frequency outputs to mono, a handy feature if you have a single-channel subwoofer (which we do).

The XM126 also has another very nice feature that any tube audio fan will consider a turn-on. To protect the tubes from power-up transients, the XM126 employs a slow wake-up process. Once you flip the power switch, it delays application of the anode voltage until the tubes are warmed up. Then to avoid damaging the cathodes, the HV plate voltage is applied after a ten-second delay. Finally, the output relay is activated after a two-second delay. The status of these steps is shown on a 3-LED display on the front of the power supply.

HOT TUBES AT 180°

Initially, we set up the XM126 and allowed it to warm up for about a half hour. As we began to play our first CD, it was immediately obvious that something was wrong. The stereo imaging was awful!

We put in Gerry Mulligan/Paul Desmond Quartet *Blues in Time* (Mobile Fidelity UDCD 648), which is a wonderful monaural recording from 1957. Hoping to hear everything coming from the center, we were disappointed to hear the monaural "image" was smeared between the left and right loudspeakers. As if this wasn't enough, there was no signal coming from our single-channel sub in SUM mode.

The problem was fairly clear. One of the input connections was reverse-polarized! Now, was it left or right? To find out, we drove the right channel through the XM126 and the left through our XM9. The monaural "image" was pretty well centered.

Aha, but when we ran the left channel through the XM126 and the right through our XM9, we found our reversal. Although Marchand put this unit together for us, they do sell their products as kits, so we figured it was okay to dig in and re-do a couple of solder connections. No problem.

HOW DOES IT SOUND?

Oh, boy! This race ended in a photo finish; a very close call. Using *Hi-Fi News & Record Review* Test Disc #3, we used the following test tracks to evaluate the XM126:

TEST 1
TRACK 2—Jerusalem/Parry

JK: The tube crossover made the instruments sound alive as if I was sitting in the back section of the concert hall. I enjoyed the distant sound of the violins playing in front of me.

KK: This track sounds great through both units; however, the low bass is fuller and the front-to-back 3D imaging is better through the XM126.

TEST 2
TRACK 4—Trumpet Concerto in C/Vivaldi

JK: The strings have a nice, soft sound and the trumpet placement is clear with the tube crossover. Our XM9 gives a clearer, more alive sound especially highlighting the string section.

KK: Our unit sounds brighter, which often sounds like it provides better detail. However, it occasionally sounds a bit artificial on the trumpets and the "click" of the bass violin strings. The XM126 tends to soften the brassiness of the trumpets, making them somewhat easier to listen to.

TEST 3
TRACKS 5/6—Peter & the Wolf (narr)/Prokofiev

JK: The "procession" in this piece comes alive with the XM126 crossover. The trumpets have a nice full sound, and each hit of the woodblock is precise and well defined.

KK: During the first part of this selection (track 5), Sir John Gielgud introduces each instrument as a character in the story. Although I can hear very slight differences in each instrument, it is quite difficult to describe them. However, during the "procession," I find that the XM126 has a much fuller low end that rounds out this piece very nicely. Again, the XM9 has a brighter response, which causes the tambourine and woodblock to leap forward. Although this recording sounds good through both crossovers, I prefer the smoother response of the XM126.

TEST 4
TRACK 7—Welcome, Welcome/Purcell

JK: Through the tube crossover, each breath the singers take and each note they sing is presented with great clarity.

KK: Er...I'm not going to say Julie's wrong here or anything, but I prefer the XM9 in this piece. It offers better detail, making it easier to differentiate the individual vocalists in the choir. The harpsichord is also more defined and focused (in my opinion).

TEST 5
TRACK 10—Corkhill (piece 2)

JK: As in our past reviews, this drum selection scares me a bit, but I still listened critically to this disruptive piece! The drums begin at the left and crescendo in the center, placing the loud percussion instruments perfectly in front of me through both crossovers.

KK: Geez, this track goes from such a quiet volume and then explodes into a drummers' assault. I'm switching to decaf permanently as of today! My heart can't take it. With the XM9, I inadvertently tend to focus mostly upon the slap of the mallet on the bass drum. When I listen through the XM126, the buzz of the snare drums doesn't fight as much for prominence and seems clearer.

TEST 6
TRACK 14—Rio Napo RSS Demo

JK: Both crossovers give a full stereo image, surrounding me in tambourines, drums, and many synthesized sounds. I was tempted to get up and join the native tribe dancing in my living room!

KK: There she goes enjoying the music again! With the XM9, it's a bit easier to pinpoint the individual instruments. The XM126, however, gives the mix a wider, more three-dimensional quality. What can I tell you? They both sound great in different ways, and I don't necessarily prefer either one with this track.

BUT WAIT, THERE'S MORE!

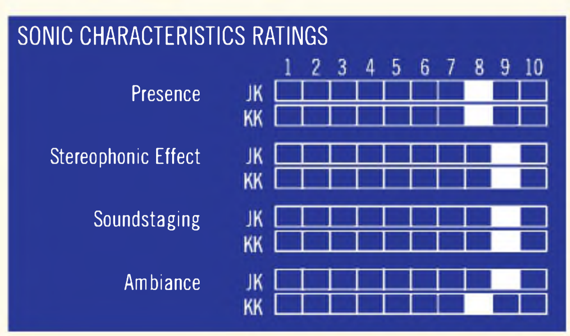
Since this was a particularly close comparison, we used three additional test tracks to help us make some determinations:

TEST 7
Indigo Girls "Let It Be Me" (from *Rites of Passage* CD)

The ingredients of this song include acoustic guitars, drums, and two harmonizing female vocals, making it a great track for testing midrange coherence.

JK: The XM126 tube crossover gives the vocals a more up-front, clear sound in this acoustic piece (by my favorite girl band). There was no noticeable noise coming from the tube crossover. In fact, all that was transmitted were their harmonized voices and acoustic guitars.

KK: Although the acoustic guitars do seem to be clearer through the XM9, all of the instruments sound a little disjointed from one another, as if they had been recorded separately! Hmm (pause for a moment of reflection). During crescendos, the lead vocals sound a bit harsh, too. The smooth tube quality of the XM126 tends to soften these discrepancies and give this track a more unified feel. Very nice.



TEST 8

Livingston Taylor "Fly Away" (from *Ink* CD)

This is a Chesky recording that was made with minimalist recording techniques, leaving plenty of air between instruments. The acoustic bass, shaker, and acoustic guitar are very detailed, as is Liv Taylor's very intimate-sounding vocal.

JK: There is a wonderful bass part in this piece. It is especially deep, yet quite clear and very pleasant to listen to through both crossovers.

KK: The overall mellow quality of the XM126 made our XM9 unit sound overly bright, particularly on the shaker and acoustic guitars! I never thought so before AB testing both crossovers. Darn it!

TEST 9

Lindsey Buckingham "Countdown" (from *Out of the Cradle* CD)

It is a highly produced track that utilizes a wonderful variety of musical textures. This is a great recording for focusing on the micro-details of how audio equipment digests a signal.

JK: This is such a fun song that even the harshest critic would get up and dance. *This* critic finds both crossovers performing perfectly, giving the music a full sound and wonderful placement of the instruments. While listening to this, both of us dropped our clipboards and danced (with our son Tommy, of course). The critical listening was over and we wanted to simply enjoy the music!

KK: Oh yeah, that's the whole point of all this audio stuff anyway, isn't it? I found that I could enjoy "Countdown" through both crossovers. However, there certainly are audible differences between the two with this track. The combination of bass guitar and bass drum is a lot warmer through the tube unit, giving the song more of a "groove." Also, Buckingham's piercing solo guitar and the crash cymbals in the drum kit are less "in-your-face." This, of course, made this track smoother.

Conversely, there's a very nice rhythm guitar part that's played up high on the neck that becomes lost in all of this tube-smoothness. That guitar part adds polish to this track, and I miss it quite a bit. Don't worry though, folks, I'll survive.

FINAL THOUGHTS

The XM126 does a fantastic job of separating its frequency bands, leaving no audible side effects at the crossover frequency. Overall, it did tend to soften the high frequencies in our system. Often this was an improvement, sometimes not. So maybe this is simply the proverbial "Tube versus Solid-State" wrestling match.

One minor complaint that we have with the XM126 is that it's somewhat susceptible to noise. Every time our furnace turns on in the basement, we hear a very audible "pop" from our loudspeakers while the tube crossover is connected. This does not happen with the solid-state XM9.

Also, when the cooling fan is switched on, it puts a very slight whine into the audio signal, which we could really only hear when standing right next to our loudspeakers. Perhaps we could eliminate this by adding more balance to our lives—for example, XLR connectors in our music system? We'd be very interested to get Phil Marchand's advice on this.

KETLERS' MUSIC SYSTEM

Sony CDP-315 CD player
Creek OBH-12 passive preamp
Marchand Electronics XM9 solid-state crossover
Valve Audio Laboratory VAA-100 30W push-pull tube amp (above 100Hz)
AudioSource Amp Two 80W solid-state amp (below 100Hz)
Custom two-way bass reflex satellites
Custom single-channel transmission line sub

to the CO filter section. Pin 2 is hot (positive) on the input XLRs.

Each HP CO consists of two stages of 12AX7A active HP filter. The final stage of each filter is capacitive-coupled to the CO level control. The signal at the level control wiper drives a cathode-follower stage that delivers the signal to the output relay on the filter board. This output relay (operating from the 12-second timer MOSFET in the PS127) ungrounds the signals and allows the audio to pass to the HP output jacks.

The LP CO filter has the same basic layout, except, with the sum switch on, the LP filter sections are summed to the right low-output jack so you can drive a single subwoofer amplifier. When a bandpass CO filter is used (three- or four-way), the BP circuit consists of two HP sections followed by two LP sections. Another option (not supplied with the test unit) is balanced outputs, using Jensen JT11-EMCF transformers and Neutrik XLR connectors. Marchand supplied nine Sovtek 7025/12AX7WAs with the XM126.

MEASUREMENTS

I operated the XM126 with pink noise at 2V RMS into 10kΩ for one hour, after which the unit was warm to the touch. The distortion was essentially the same for each channel. Data for the left channel is presented here and summarized in *Table 1*. The CO was quiet, with no noise during power-up or shutdown. The sequencer was doing its job just fine.

The XM126 does not invert polarity with balanced or unbalanced inputs (XLR pin 2 is hot). Input impedance was 503k for both channels. The output impedance at 100Hz measured 904Ω (LP) and 1k3 (HP).

The frequency response for the XM126 is shown in *Fig. 1* with a 10k load. The gain at the 100Hz crossover point (10k load) with all level controls at "0dB" was −5.7dB for both HP and LP sections. The combined LP and HP response at 100Hz results in a flat acoustic response. When I increased the load to 100k, the flat portions of the curves were +0.5dB higher (not shown).

Level control tracking was within the stated tolerance for these carbon film pots. +6dB indicated resulted in a relative change, from 0dB indicated, of +5.9dB. At −12dB indicated the output was −10dB, and I measured −17dB output with the level control at −20dB indicated. There was less than 1dB difference between the four CO level controls at each of these points.

Figure 2 shows THD+N versus output voltage into 10kΩ at 20Hz (LP), 100Hz (LP and HP), 1kHz (HP), and 20kHz (HP). The LP curves are solid, and the HP curves are dashed. I engaged the test-set 80kHz low-pass filter to limit the out-of-band noise. I expected the THD+N to be higher for the HP section since it has a much wider bandwidth over which noise can exert its effect. However, the distortion is much higher than noise, which is not visible on the HP residual signal (*Fig. 3*).

I measured the 1% THD+N clipping point with a 10kΩ load at two LP and three HP frequencies. These are summarized as follows. The clipping waveform in each case was rounded as you would expect in a tube circuit, with the lower half of the waveform flattening out first.

LP 20Hz, 1% THD+N output at 3.0V RMS
LP 100Hz, 1% THD+N output at 5.0V RMS
HP 100Hz, 1% THD+N output at 1.3V RMS
HP 1kHz, 1% THD+N output at 3.0V RMS
HP 20kHz, 1% THD+N output at 3.0V RMS

The XM126 is not happy with low impedance loads. I preset an input voltage that resulted in 1% THD+N at 100Hz with a 10k load. The THD+N increased to 5% with a load of 3k7, and exceeded 30% when I reduced the output load to 1k, primarily due to lower half-cycle flattening. This should not be a problem with most power amps, since very few have less than a 10k input impedance.

The LP section distortion waveform for 2V RMS into 10kΩ at 100Hz is shown in *Fig. 4*. The upper waveform is the amplifier 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 mainly the second harmonic with some higher frequency noise.

The LP section spectrum of a 100Hz sine wave at 2V RMS into 10kΩ is shown in *Fig. 5*, from zero to 2.6kHz. The THD+N measured 0.43%, with only a few harmonics. The second, third, fourth, and fifth measure −48dB, −74dB, −90dB, and −93dB, respectively.

Repeating the residual distortion test for the HP section produces the waveform in *Fig. 6*. The distortion residual presentation (second and third harmonic) differs depending on which half-cycle is examined, with no evidence of any noise or fuzz.

The 2V RMS 100Hz sine wave HP section spectrum is shown in *Fig. 3*, from zero to 2.6kHz. The THD+N measures 2.84%, and the harmonics, while fairly high, are limited to two through five. They measured −32dB, −39dB, −51dB, and −67dB, respectively.

I performed my intermodulation distortion (IMD) testing on the XM126's HP section. While reproducing a combined 19kHz + 20kHz IMD test signal at 2V RMS into 10kΩ, the 1kHz IMD product was 0.4%. Repeating the test with a multi-tone IMD signal (9kHz + 10.05kHz + 20kHz) resulted in a 1kHz product of 0.17%.

I did not perform square-wave testing on the XM126, since its job is to selectively alter frequency response. ❖

Software Review

Cool Edit

Reviewed by Perry Sink

Cool Edit Audio Editing Software, Syntrillium Software Corporation, PO Box 62255, Phoenix, AZ 85082-2255, 602-941-4327, sales@syntrillium.com, www.syntrillium.com.

A few months ago, I was planning to record some audio educational and promotional programs for both work and church projects. I initially assumed that I would record some material on cassette and then use a mixer and a couple of tape decks to put it all together. At least that's how I did basement tapes for a band 15 years ago. But when it came time to actually do it, I wondered: wouldn't this be much easier to accomplish if I could edit and mix it on a PC?

I investigated and discovered a very easy-to-use and highly capable Windows program called Cool Edit from Syntrillium. Cool Edit allows you to turn any PC with a soundcard and speakers into a two-track digital editor and signal processor. It turned out to be an outstanding solution to my problem.

THE PC INVADES AUDIOPHILIA

It will be clear as you read further that Cool Edit is a very useful tool for anyone doing small-scale audio production. But there are certainly applications for traditional audiophiles, too. With the availability of recordable CDs and the fact that many people would not only like to produce their own compilations but even do digital signal processing on individual selections, many of you will be interested in Cool Edit. If you have a recordable CD-ROM drive (rapidly becoming ubiquitous and inexpensive) on your computer, you can use Cool Edit to record and digitally process music tracks that are poorly equalized or mixed and then copy the edited version onto a CD for permanent use.

I discovered another thing: having Cool Edit at my disposal caused me to think more resourcefully about audio in general. Suddenly I had an easy way to record my two-year-old daughter singing "Old McDonald" (and preserve it permanently on CD), or the sound of my infant son crying just after he was born. I recorded him on my PC with a $10 microphone and put the .wav file on a website announcing his birth. Friends and relatives, hundreds or thousands of miles away, really got a kick out of hearing his voice and seeing his pictures on the net!

Which brings me to the subject of music on the Internet. Without delving into it, let me just say that with the rise of wideband internet connections, more and more people will be listening to music via MP3 and RealAudio via their PCs. I have no doubt whatsoever that both the music industry (which will distribute music electronically) and the audio industry (where the PC is invading hi-fi) are in for some major changes in the next five years. It will affect broadcasting and music as much as VCRs affected movies. The future is certainly bright for companies who have a vision for music on the web. And, of course, Cool Edit falls into that trend perfectly.

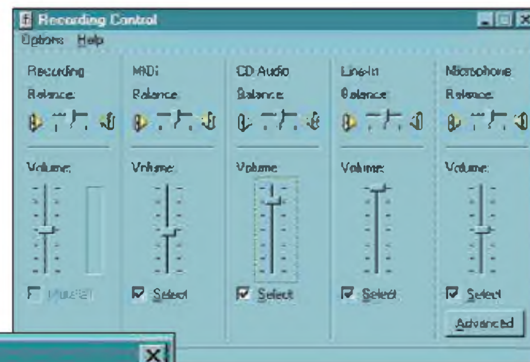


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FIGURE 1: Cool Edit's recording control menu.

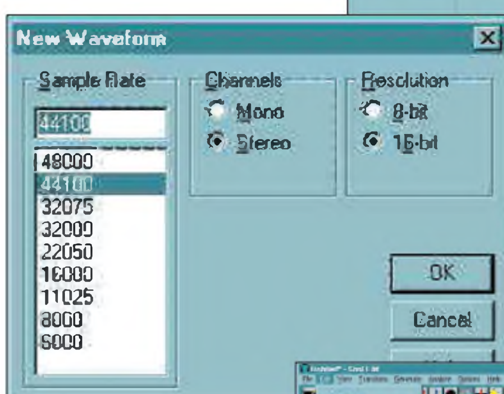

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FIGURE 2: Waveform screen.

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FIGURE 3: Multi-editing with Cool Edit.

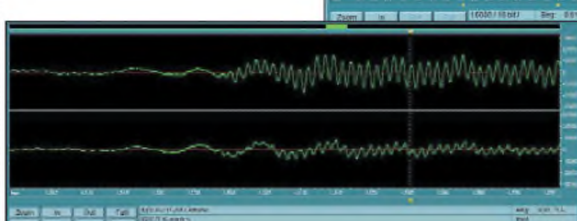

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FIGURE 4: Viewing range.

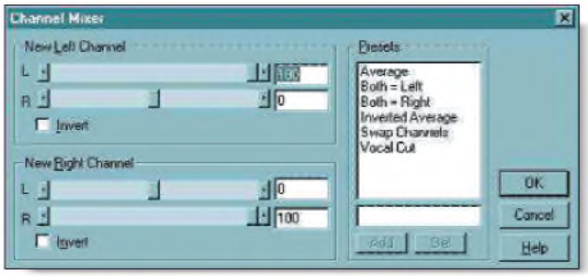

FIGURE 5: Channel mixer screen.

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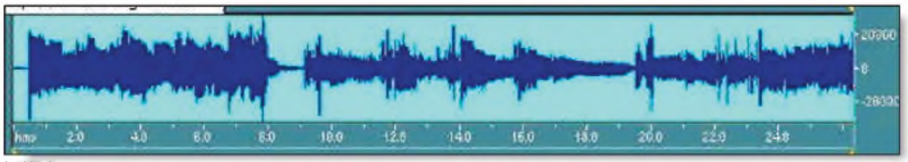

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FIGURE 6: Sample uncompressed waveform.

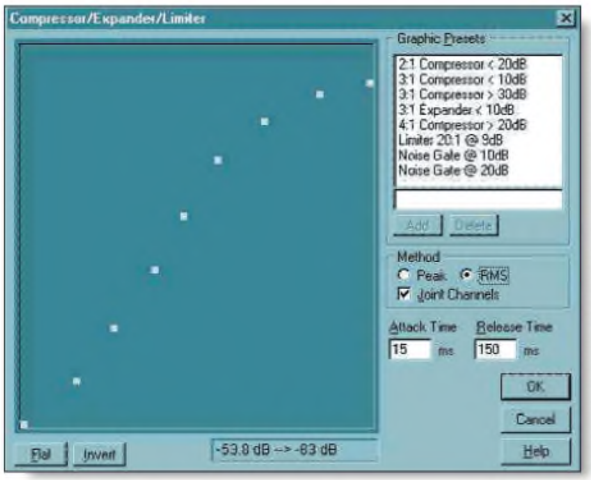

FIGURE 7: Compressor features.

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SHAREWARE

You can download the program free of charge from www.syntrillium.com. Cool Edit has 16 editing feature groups, and the unregistered shareware version allows you to use any two of them during one session. For example, you can do compression and equalization of a sound file, but if you choose to use other features such as reverb or FFT analysis, you must exit the program and start it up again.

So you can play with it all you care to, but you must register to use all the features at the same time. I tried it, and it wasn't long before I happily parted with my money! I paid my sixty-nine bucks and Syntrillium sent me a passcode to unlock the program's full functionality.

PCS AND MULTIMEDIA

These days most PCs are sold in complete packages: the CPU, hard drive, memory, keyboard, mouse, monitor, modem, CD-ROM (sometimes recordable), and some basic software including Windows 95 or 98. Also, most are multimedia, meaning they come with a sound-

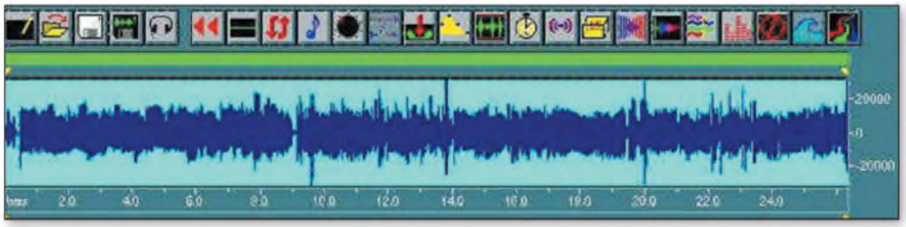

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FIGURE 8: Sample waveform compressed.

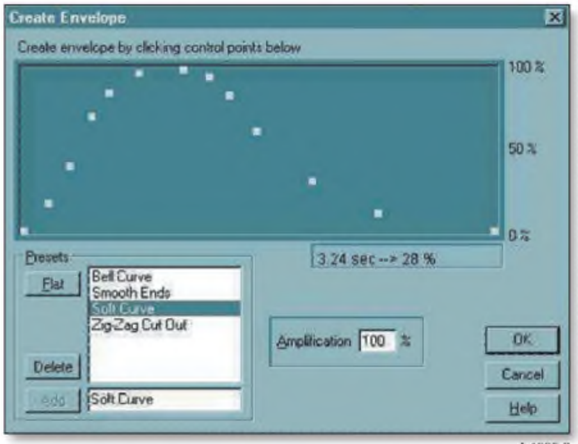

FIGURE 9: Envelope function.

A-1625-9

card and speakers, and sometimes a microphone.

Windows 95/98/NT/2000 have the ability to record and play back sounds at various quality levels, from 8kHz sampling rate and 8 bits resolution mono (4kHz bandwidth and 48dB dynamic range) to full 16 bit, 48kHz stereo, just like in a studio. It records in a file format called Windows .wav. Cool Edit supports .wav, as well as a dozen other file formats, and allows you to convert between file formats, sampling rates, and resolution levels.

HOW AUDIO WORKS ON A PC

Cool Edit captures signals from your soundcard. A recording control menu in Windows (*Fig. 1*) allows you to mix the levels of the microphone, line-in, CD, and midi as you like. A volume menu lets you play these devices at desired levels through your speakers.

So you can plug microphones and other sources into your soundcard on the back of the PC or use the CD player in the PC as a sound source. I used the CD as my signal source. When you record a waveform, you can choose 8- or 16-bit resolution and any of the sampling rates in *Fig. 2*.

FEATURES

Cool Edit does not do signal processing in real time; in other words you can't use it to equalize or filter signals live as they pass through your PC. You must record audio and edit it after the fact.

Your mouse works much the same way here as it does in a word processing program. You simply highlight part or all of a recording and then apply any of the signal-processing or editing features to the highlighted selection. How long it takes to do the processing depends on the length of the selection, the sampling rate and resolution, what kind of processing you apply, and the speed of your computer.

While the latest blazing-speed 1GHz PC may have more horsepower than most people really need for cyber surfing, spreadsheets, and e-mail, it won't be wasted with Cool Edit. Since processing and editing commands take time to execute, then the more speed, memory, hard drive space, and other forms of digital testosterone your PC has, the better off you are.

Undo. Cool Edit has an undo feature that works exactly the same way as in other Windows programs. It is extremely useful for fine-tuning processing parameters to obtain desired results. If you click on undo multiple times, you can work backwards through the most recent edits that you've made. You can conserve system resources and speed up processing time by disabling undo.

Processing and Storing Audio Files. Sound files use a lot of memory, especially for CD-quality stereo files. A 74-minute CD holds 650MB of data, which is a lot compared to the garden-variety text files, spreadsheets, and programs most of us use on our computers every day.

Multiple Simultaneous Editing Sessions. Cool Edit allows you to have any number of editing sessions open at once so you can freely cut and paste from various files at will without opening and closing them first. You can save sessions and selections to your hard drive just as you would save any other file. Note that large files (i.e., 10 or 20 minutes or more at full resolution) can occupy over 100MB and take a long time to save.

In addition to traditional cutting and pasting, you can "mix paste," or paste one file over another and effectively mix the two together. The program prompts you to set the desired levels for each of the two files.

Figure 3 shows a VU meter at the bottom of the window, as well as the play/record and zoom controls and file information. The left and right channels are displayed simultaneously, and above that, the icons representing various signal-processing and editing options available.

SYSTEM REQUIREMENTS

Windows 3.1 or higher or Windows 95 or NT, 4MB RAM, and 2MB free hard disk space

Recommended: Soundcard, speakers or headphones, and mouse

Optional: CD player and microphone

FIGURE 10: Modified waveform.

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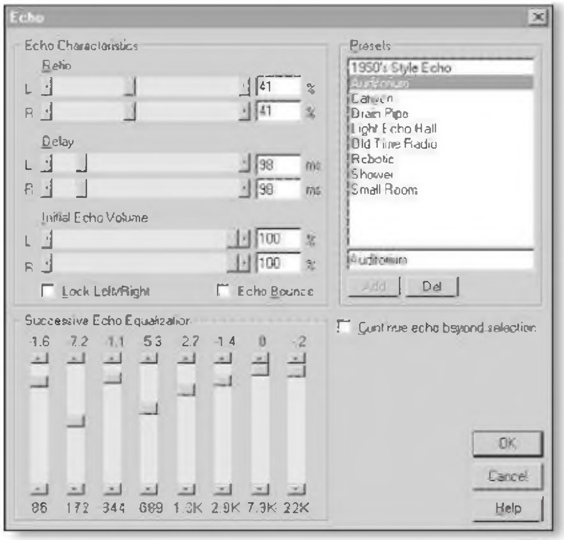

FIGURE 11: Control panel for echo function.

Zoom In and Zoom Out allow you to see the entire range of time scales from the full program length down to milliseconds (*Fig. 4*). There's a short green bar just above this waveform to indicate the length of the viewing window relative to the total length of the file.

The channel mixer (*Fig. 5*) allows you to re-mix the proportions of the left and right channels any way you like. You select from standard presets and also make and save your own.

Compression, Expansion, and Limiting Capabilities. The waveform in *Fig. 6* is uncompressed, and has a fairly wide dynamic range. The scale on the right is the number of data points, which for 16-bit recording is ±32,768 points. The ±20,000 marks are indicated here.

The Compressor/Expander/Limiter (*Fig. 7*) is extremely versatile. Preset curves/knees are available from the menu on the upper right, but you can also define any input versus output relationship that you like. You can give any curve you have defined a name and save it for future use. The vertical axis is output and the horizontal axis is input; you can define as many points as you like and put them anywhere, and also define the attack and release time.

Figure 8 shows the same waveform after being compressed according to the curve in *Fig. 7*. There are a few sharp impulses remaining, but the overall signal level is much more consistent throughout. This kind of extreme compression comes in handy for editing a telephone conversation in which the interviewee's voice is weak or there is a lot of background noise.

Envelope Function. You can select a portion of a wave (or all of it) and define an envelope for the volume level with

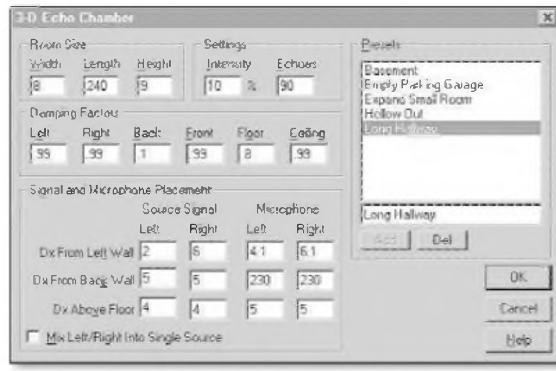

FIGURE 12: 3-D echo chamber function.

respect to time. The curve in *Fig. 9* is one of the available presets.

I modified the highlighted portion of the wave in *Fig. 10* according to the previous envelope. This feature could be used to produce custom fade in/fade out profiles, or to adjust the overall level of a program very gradually over time with very precise results.

Echo, Delay, Reverb, and 3-D Room Simulation Effects. Figure 11 shows the control panel for the echo function and the preset values for the auditorium setting. You can produce a considerable array of sound characteristics by tweaking the mixing and equalization settings. A simple time-delay function is available, with the ability to mix in the original signal in precise proportions.

Cool Edit also has a reverb function, a general room acoustics simulator. The 3-D Echo Chamber allows you to precisely define the acoustics of a virtual room, with settings for length, width, height, and damping factors for each surface (*Fig. 12*). The effects can be quite realistic, for a fraction of the price of a typical stand-alone reverb unit.

Sound Effect. The flanger (*Fig. 13*) allows you to produce many bizarre sound effects by adjusting the parameters of a time-variant phase-shifted signal mixed in with the original.

The Frequency Spectrum View. In addition to viewing waveforms, Cool Edit allows you to view the frequency spectrum versus time, with frequency on the vertical axis (note that it's linear, not logarithmic), and color intensity showing the amount of energy in any particular band. On this type of graph (*Fig. 14*) the vast majority of the spectral activity is seen at the bottom because most musical energy is concentrated below 1kHz.

Digital EQ. Digital filtering is one of Cool Edit's most powerful and useful features. This menu allows you to build any frequency response with a digital filter by simply putting points on the graph. You can literally produce any equalization curve you can imagine just by connecting the dots. You can name your curves and save them for future use. The graph in *Fig. 15* is scaled from zero to 100% (other scales are possible). If you study the points on the graph, you can see that this curve completely attenuates all frequencies between 5 and 15kHz.

To illustrate the capabilities of the digital filter, I selected a portion of the wave (from t = 23s to t = 49s) and completely filtered out all components between 5 and 15kHz. As you can see from the black areas in that band (*Fig. 16*), the filter was very effective! It totally eradicated all energy in that frequency range, while leaving everything else untouched.

Remember, you can define the filter any way you like, just by connecting the

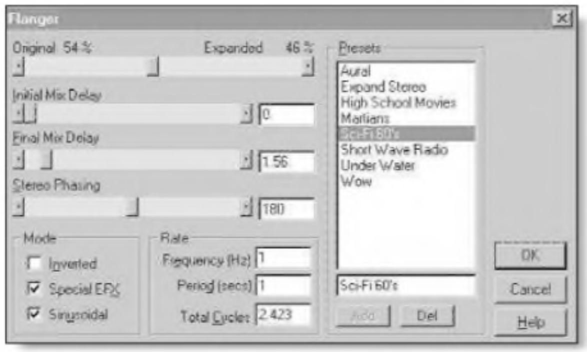

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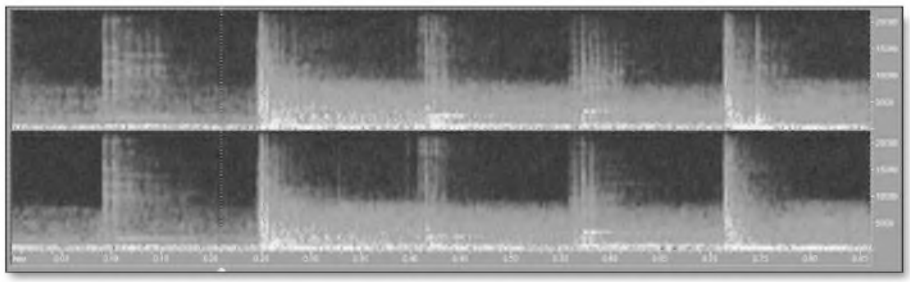
FIGURE 13: Sound-effects panel.

FIGURE 14: Unmodified frequency-spectrum graph.



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dots. This could just as easily have been a series of stripes 1kHz wide, or deep, narrow notches to filter out hum harmonics (notice the built-in 60/120Hz hum filters in the menu in *Fig. 15*), or a rolloff with any slope you wish to define. This kind of precise filtering is next to impossible using analog hardware, and certainly isn't anywhere near as flexible.

Basic Graphic Equalizer. In addition to defining digital filters by "connecting the dots," you can also use a traditional equalizer menu (*Fig. 17*). This menu was what I needed for EQ most of the time. You can also start with one equalization curve and end with another, so the EQ actually changes with time, if desired.

Digital Noise Reduction. To illustrate this capability, I recorded some music with a 45dB noise floor (*Fig. 18*). The grainy appearance of the spectral view is from the noise. The portion in the center is three seconds where there is no music, only noise.

Cool Edit does digital noise reduction (*Fig. 19*). You must "teach it" what

A COOL UPDATE: VERSION 2000

By Perry Sink

Most of this review is based on Cool Edit 96, which has now been superseded by Cool Edit 2000. The two programs are largely similar, with CE2000 containing a number of new features (*Fig. A*).

The most significant addition is the ability to open, edit, record, and convert all of the other formats it supports to/from MP3, a compression format that encodes digital audio and reduces the file size by a factor of four to over 100, depending on the desired sound quality. MP3 has taken the world by storm during the last year. I recently read that there are over 17 million MP3 downloads on the Internet every day!

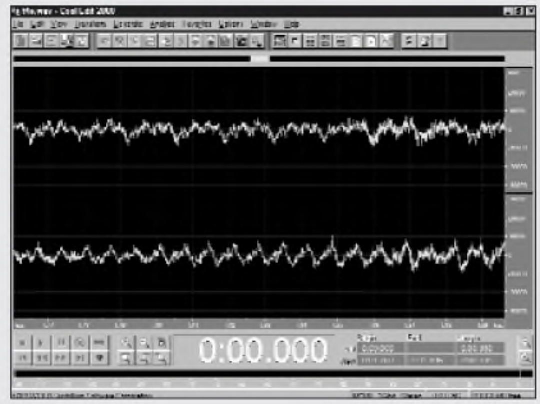

FIGURE A: Sample Cool Edit 2000 screen shot.

If you have a 56K modem, you can download an MP3 song in ten to twenty minutes. If you have a high-speed connection, such as cable or DSL, you can download a song in one or two minutes. You can also "stream" music, which means playing the file as it comes in, rather than saving it to your hard disk. Cool Edit puts a very good, very affordable authoring tool in the hands of anyone who cares to participate—instead of just being a spectator—in the MP3 experience.

Standard CD encoding consumes approximately 10M of data per minute: 16 bits (2 bytes) × 44,100 samples per second × 2 channels

= 1,411,200 bits per second
= 176,400 bytes per second
= 10,584,000 bytes per minute.

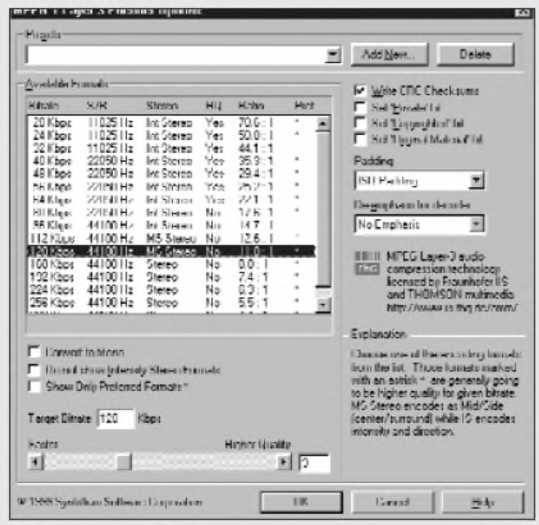

FIGURE B: MP3 file options.

So a typical song can take 30–50M of data. If you're trying to download a file that big from the Internet, forget it!

Syntrillium's MP3 encoder can work at rates of 8Kbit/sec (very poor, "CB radio" quality; 176:1 compression) all the way up to 320Kbit/sec (very high quality encoding, virtually indistinguishable from the original with 4.4:1 compression). The most common bit rate used for Internet downloads is 128Kbit/second. This gives 22kHz bandwidth and very good sound quality.

How good? It depends on the encoding mechanism. In my relatively limited experience, the encoding mechanism can make more of a difference than the bit rate itself.

Syntrillium uses the "Fraunhofer" codec, which is widely regarded as one of the best algorithms for encoding MP3. It also gives you the option of selecting the quality of the encode (*Fig. B*). You can use a low-quality conversion and do it very quickly, or you can select high quality, which takes more time.

In my opinion, a 128Kbps MP3 provides a similar level of fidelity as a very good cassette deck with metal tape and Dolby C noise reduction. An A/B comparison initially shows the copy to be indistinguishable from the original...until you spend a few minutes to listen carefully. Then you begin to identify some idiosyncrasies.

In the case of MP3, the most obvious problem is that cymbals sound slightly strange—sometimes clangy or perhaps like "shifty noise" instead of that familiar, natural "crash." The sound of the stick striking the cymbal is distorted. Sometimes you also hear "artifacts," which sound like intermodulation between various instruments. Low bit rate MP3s sound increasingly "trashy" as you go below 64Kbps. With a good conversion at 128Kbps, the problems are not at all obvious, and with increasing bit rates (i.e., greater than 200Kbps), these artifacts shrink to the vanishing point.

Cool Edit cannot, however, open files digitally from audio CDs (I believe this is intentional, because it's not hard to do technically). To open and edit an audio CD track, you must either play the CD and record the analog signal that emerges from your soundcard, or else use a separate utility to do the digital conversion.

Soundcards differ widely in their fidelity and noise levels, and are usually prone to picking up high-frequency interference from other devices in your PC. I strongly recommend using direct digital-to-digital, for obvious reasons.

With CDMASTER32, you can convert a CD to Windows WAV format and then edit the files with Cool Edit 2000. You can download CDMASTER32 and lots of shareware or low-cost software programs for audio conversion, MP3, and editing at:

http://www.mp3machine.com and
http://www.hitsquad.com/smm/.

OTHER IMPROVEMENTS

- A free plug-in for "cleanup," which removes tape hiss and LP ticks and pops. Features extensive parameter adjustment capabilities so you have tight control over the signal processing.
- Time functions displayed in a more ergonomic fashion.
- More features for customized play-lists and cue lists.
- Macros for performing repetitive tasks.
- More efficient handling of memory and files.
- More useful default editing icons for the most commonly used functions.
- Support of multiple clipboards.
- Optional studio plug-in ($49) turns CE2000 into a four-channel mixer.
- 24 bit/96kHz recording and multitrack recording and editing (up to 64 tracks!) for $399.
- Ability to start recording at a specified time and date.
- Batch process feature for identical signal processing on multiple files.
- A "find beats" feature to detect repetitive patterns in music and to help in mixing multiple tracks or songs together (i.e., DJ effects); it can also make the gridlines in the editor line up with the beats.
- Automatic removal of pauses in recorded speech.

Cool Edit 2000 represents just one facet of the biggest revolution in the music and hi-fi business since the advent of the CD. With tools like this, the "starving artist" is on a nearly level playing field with the pros. Certainly a $400 per hour recording studio includes capabilities that you won't find in a $400 software package. But most musicians have more time than money, and given enough time, could create an audio masterpiece with an advanced version of Cool Edit like CE2000 Pro.

For those with less ambitious goals—such as converting an LP to CD—Cool Edit 2000 is an outstanding tool. At $69, I can't think of why every audiophile or hands-on music lover shouldn't have a copy! The $69 price tag buys you signal-processing power that five years ago would have cost you ten grand.

I strongly encourage you to download a trial copy—and if you're at all intrigued by the possibilities, buy the full version and enjoy the benefits that your PC brings to the audio world.

kinds of signals you consider to be noise. You do this most easily by selecting portions of your program material that consist only of noise and then using the "Get Noise Profile from Selection" feature. When Cool Edit detects signals that match the level and frequency content of the noise profile, it digitally filters only those portions. The better your noise profile selection matches the noise in your program material, and the longer the selection, the better it works.

This not only works well for white or pink noise such as tape hiss, but can also be very effective for eliminating hum from a recording. You can adjust the relative severity and thresholds of the noise-reduction process.

Figure 20 shows the amount of noise remaining with the same program material as before after being digitally filtered. Because the noise had a consistent spectrum over time, this worked quite well. If the noise is inconsistent or if you have an incomplete noise model, filtering will produce strange artifacts in the processed signal. This selection did have some such artifacts, but they were not as objectionable as was the noise.

ELIMINATING HUM

For one project I recorded several telephone interviews, only to discover that one of them had a serious 60Hz hum (with a long series of harmonics also) because of a bad connection from the telephone to my PC. I was in a panic because doing the interview over again was not an option! It had taken me over a month just to schedule this person the first time. (By the way, to successfully connect a phone to a soundcard input, you need to use an isolation transformer!) I scrambled to see whether I could figure out a clever way to get rid of that hum.

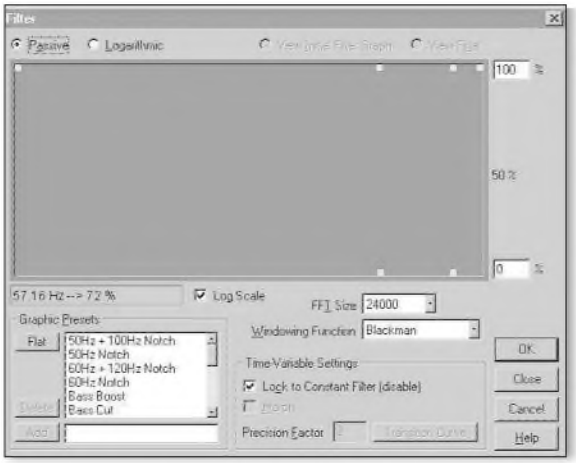

FIGURE 15: Filter menu. A-1625-15

I tried building a notch filter with the digital filter function. I put notches at 60, 120, 180, 240, 300Hz, and so on, but to my dismay I found that even though I could eliminate the lower-order harmonics, the higher ones (360, 420, 480, 540, 600Hz...) became increasingly difficult to take out—and even more conspicuous at the same time. The filtered version sounded even worse than the original because of this remaining cluster of high-order harmonics. So I turned to the noise-reduction utility.

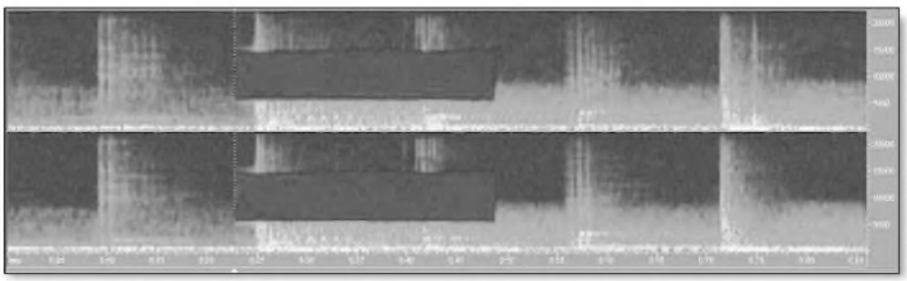
FIGURE 16: Filtered frequency-spectrum graph. A-1625-16

It took some time to figure out how best to use it, but I discovered that if I could piece together a long portion of silence (no sound except for the buzz itself) from the affected program material, the noise reducer had a complete model of the noise and it could basically hunt down all of those harmonics and leave everything else unaffected. After fine-tuning the sensitivity of the filter, the hum was gone. The man's voice did sound somewhat funny (a bit "phasey") when I finished, but my project was rescued from disaster!

Adding Distortion. Figure 21 shows an ordinary sine wave. Cool Edit allows you to add distortion to signals. You can define the linearity of the sig-

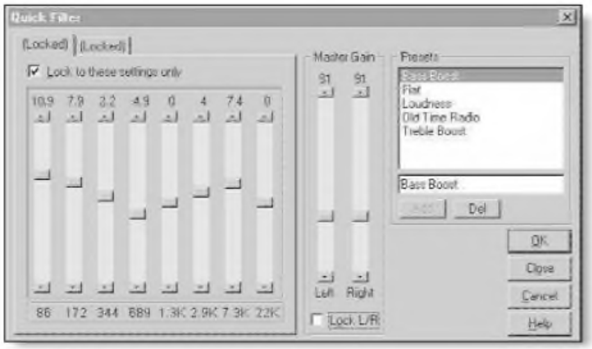

FIGURE 17: EQ menu.

A-1625-17

nal much the same way that you define the knee of the compressor. Several presets are available. *Figure 22* shows the effect of the "fuzz" setting on the pure sine wave in *Fig. 21.*

The square area with the points inside shows the zig-zag linearity definition of the "fuzz" preset. This menu allows you to imitate many of the sounds that effects pedals on electric guitars produce. Annoy your friends by making a copy of their favorite CD with a little distortion thrown in, just for fun!

Changing Pitch and Tempo Independently. Cool Edit can speed or slow the recording without changing the pitch, or change the pitch without changing the speed, or change both pitch and speed (*Fig. 23*). The pitch-shifting feature is very useful for: 1) adjusting the length or tempo of dialog–i.e., speeding up a slow talker without changing the pitch of his voice; 2) changing the pitch of an instrument that's off key; 3) making a man sound like a woman or vice versa; 4) forcing a selection to fit within a predefined length of time.

Precision settings allow you to trade off between processing time and processing quality. As with many of the routines in Cool Edit, you can define one set of parameters for the beginning of your selection and another for the end, and it will linearly adjust the settings from beginning to end as it goes. The gliding stretch tab allows the end of your selection to have a different pitch or speed than the beginning, with a smooth transition in between.

Built-In Tone and Function Generator. A tone generator (*Fig. 24*) is available with the ability to define the level of each harmonic at will. Notice the modulation and phasing capabilities. You could use this feature to produce your own audio test program or CD with sine waves, warble tones, and so forth. There aren't many function generators that allow you to produce waveforms with this level of sophistication!

Frequency Analysis & FFT. The waveform on the left in *Fig. 25* is the amplitude versus time plot of a tone with five harmonics (constructed in Cool Edit Waveform Generator). The graph on the right is a FFT frequency spectrum analysis of this wave. Notice the five peaks indicating the frequencies this wave contains. A variety of precision and range settings is available in this menu. This feature would probably not be very useful for measuring frequency response (it's too crude), but it is certainly useful for observing distortion harmonics and frequency spectrum of recordings. You could use it to determine how much actual sub-bass content is on your favorite car subwoofer test CD.

Sampling Rates. I recorded the interviews at a lower sampling rate (16kHz) than the music (which was sampled at 44.1kHz) to conserve hard drive space. When it was time to compile it all together into a complete program, I past-

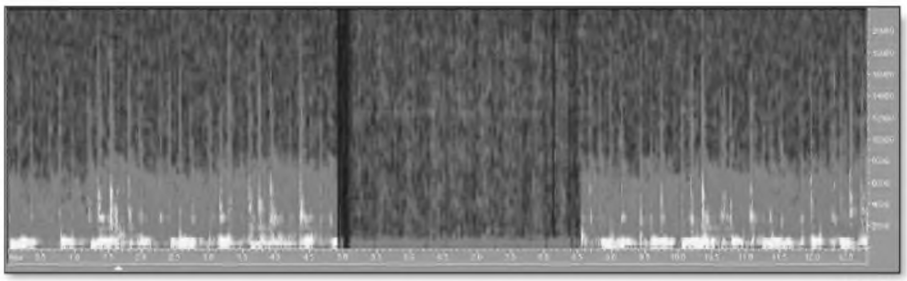

FIGURE 18: Sample graph of recorded music.

A-1625-18

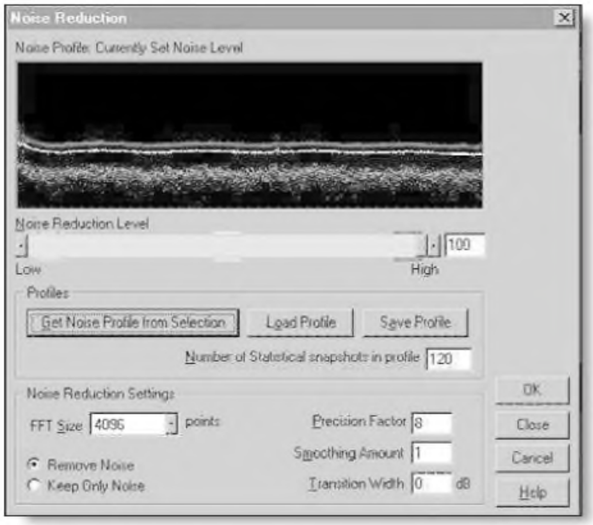

FIGURE 19: Noise-reduction panel. A-1625-19

SOUND FORMATS SUPPORTED

Windows PCM waveform (WAV)
8-bit signed raw format (SAM)
ACM waveform (WAV)
Apple AIFF format (PCM-encoded data only) (AIF)
CCITT mu-Law and A-Law waveforms (WAV)
Dialogic ADPCM (VOX)
IMA/DVI ADPCM waveform (WAV)
Microsoft ADPCM waveform (WAV)
Next/Sun CCITT mu-Law, A-Law, and PCM format (AU)
Raw PCM data
SampleVision format (SMP)
Sound Blaster voice file format (VOC)
TrueSpeech (WAV)
RealAudio
MP3

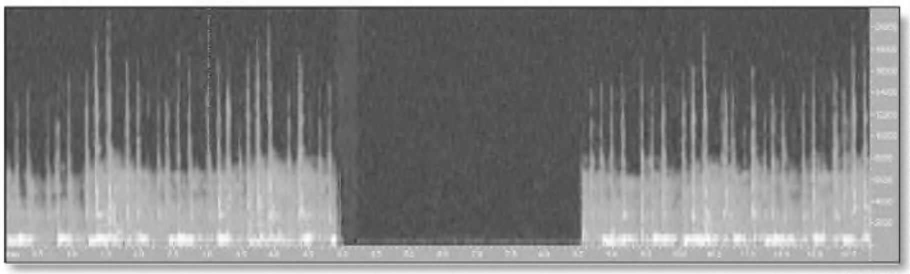

FIGURE 20: Sample filtered for noise reduction. A-1625-20

ed the interviews at the end of the music. Cool Edit automatically upsampled the mono interviews to 44.1kHz stereo with no undesirable side effects.

Of course, I was also able to equalize the speakers' voices to make them sound less telephone-like. This was helpful although not 100% effective. And as I said before, the ability to eliminate a buzz on one of the interviews came in very handy.

USING COOL EDIT IN REAL LIFE

I used Cool Edit to produce an hour-long educational audio program for CD/cassette on my PC. It included several selections of music and a series of telephone interviews, most of which needed heavy editing. Some of the interviewees had lots of "uhs" and "ums," which I easily spliced out by highlighting them and pressing the delete key. It was very easy to highlight a portion of the interview, cut it out, and save it as a separate file. In this way I could completely reorganize the order and content of the various topics discussed and reassemble them for the final version.

With Cool Edit, you can record someone talking, splice out a collection of their words and syllables, re-assemble them in any order, and use pitch shifting to change the inflections of their voice. You can literally make them say anything. For the "creative" journalist, this could come in really handy for turning an otherwise ordinary radio interview into headline news! Don't tell anybody I suggested it to you, though.

There were a few spots where I exceeded the maximum levels during recording. When you record digitally, clipping sounds much worse than on saturated analog tape, or even an amplifier. It makes a harsh clicking or snapping noise. I discovered that sometimes only part of a spoken word or syllable was clipped, and I could actually delete just the clipped portion (which might have been 50 or 100ms long). You may not even notice that "surgery" had been performed on that word.

When files are longer than 20–30 minutes, they can become difficult to manage; saving or processing long selections can be very time consuming. Cool Edit can give your hard drive a workout, and if you're not careful, you can fill it up to capacity. In most cases you can break an hour-long program into several tracks, which makes file manipulation much easier.

COPYING TO CD

When it was time for me to copy to a CD, I used a friend's computer with

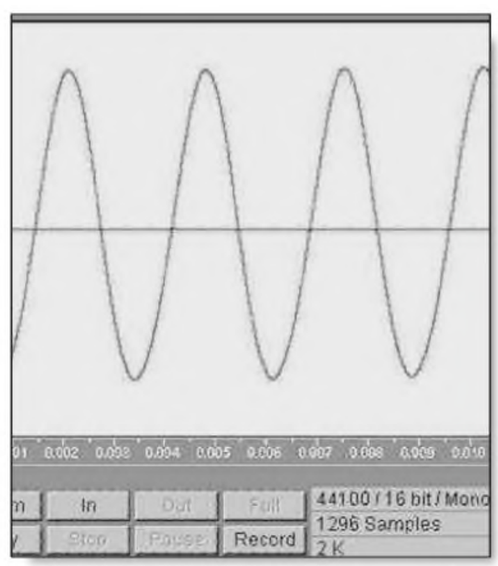

FIGURE 21: Sample sine wave. A-1625-21

an Ethernet LAN connection and a server. I copied the files from my PC to the server (which took over 15 minutes) and then copied the files from the server to the CD. His computer had an excellent program called Adaptec Easy CD Creator (www.adaptec.com), which I highly recommend. It has the ability to edit .wav files and convert them directly to standard audio CD format. (Note: If you copy Cool Edit's .wav files directly to a CD, you'll have a CD-ROM that only a PC can play. You need a program such as Adaptec Easy CD Creator to burn CDs.) It has the ability to queue multiple files and assign each of them consecutive track numbers as it records them onto the CD.

PITFALLS OF DIGITAL EDITING

You should beware that when you record digitally and then reduce the level at a later point, you "lose bits," which you cannot recover if you increase the level later on. This is much more of an issue in multitrack recording than it is with a two-channel program such as Cool Edit. So if you manipulate signals multiple times, beware that you can introduce a fair amount of unexpected digital noise in the process.

Many professional digital recording systems use 24 bit instead of 16 to lessen the severity of this problem. However, this is generally not a very serious problem with Cool Edit, and once something's recorded it stays in the digital domain until you're done. If you take precautions with respect to levels and other parameters, your end result will generally be very good.

Not all soundcards are created equal. You can spend anywhere from \$20 to over \$1000, depending on application. In one of my PCs the soundcard picked up high-pitched noise, which was probably from the video card—not acceptable for audiophile applications.

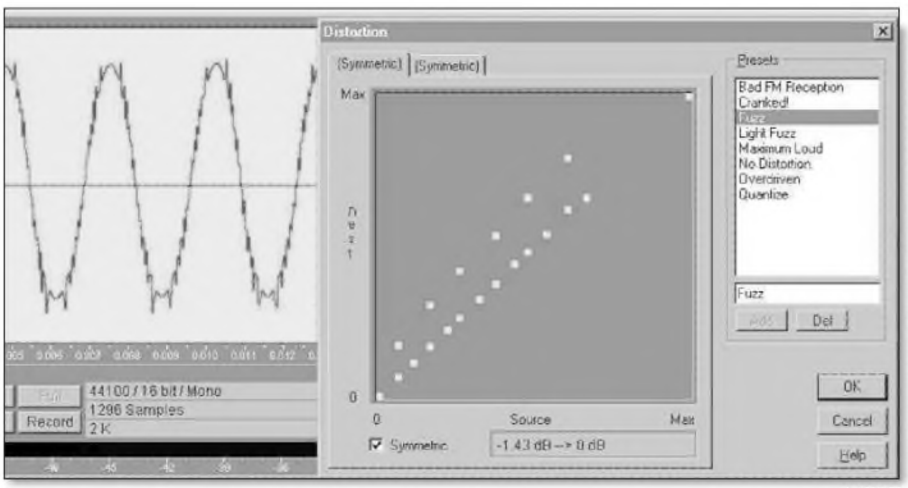

FIGURE 22: Adding distortion.

A-1625-22

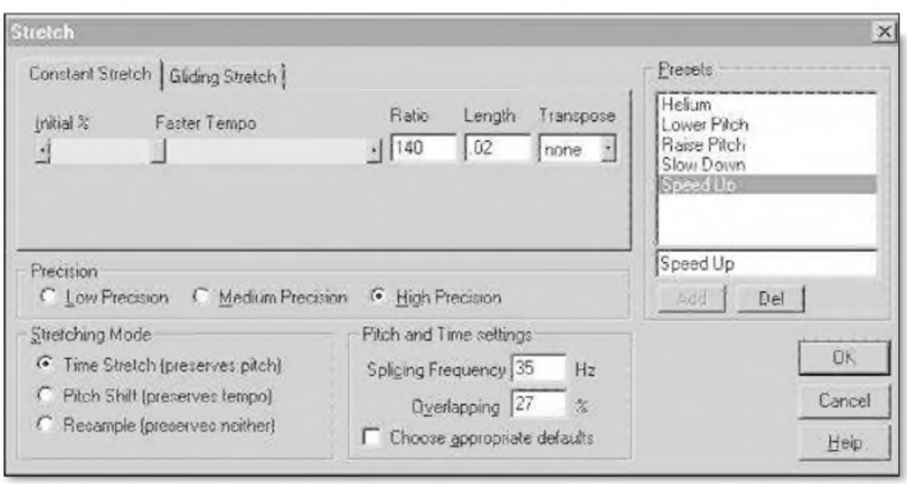

FIGURE 23: Changing pitch and tempo.

A-1625-23

Also, cheap soundcards seem to allow only a 60 or 70dB dynamic range because of a high noise floor.

MP3

In a very short period of time, MP3 has caught on like wildfire as the defacto open platform for music compression. Depending on the bit rate you select, MP3 can compress music at ratios of 10:1 or more. So a song that takes up 40MB on a CD might only take 4MB on MP3.

High resolution MP3s can sound very, very good, too. Cool Edit can encode and edit MP3s at resolutions ranging from 8 KBPS to 320 KBPS. Many audiophiles consider MP3s recorded at rates exceeding 240 KBPS to be indistinguishable from the original. Cool Edit uses one of the more sophisticated technologies, the Fraunhofer algorithm, to encode MP3s.

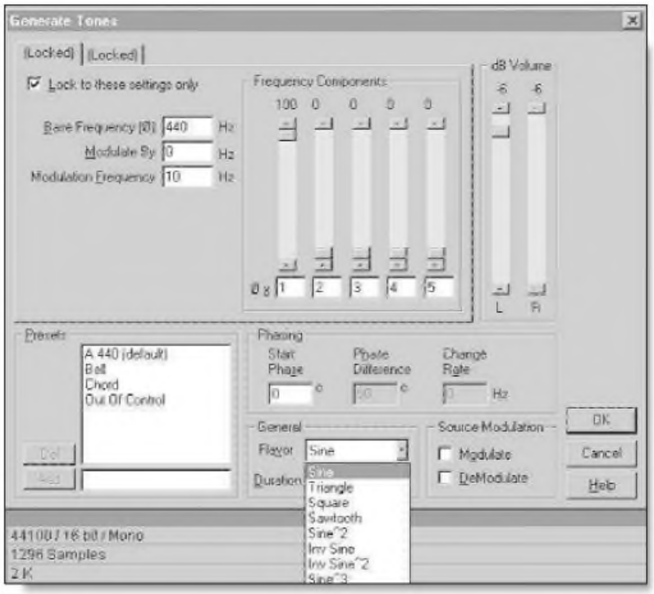

FIGURE 24: Tone generator panel. A-1625-24

Cool Edit allows you to also adjust encoding quality, meaning that at the expense of a longer processing time while you save your files, it will more accurately represent the compressed music material.

Although I didn't test this feature, Cool Edit also contains a RealAudio encoder. RealAudio is a very popular live music streaming format, and if you're doing audio applications on your website, Cool Edit is indispensable.

16 TRACK AND ENHANCED RESOLUTION RECORDING

An enhanced version, called Cool Edit Pro, handles up to 64 tracks and is designed to manage multiple gigabytes of data. It also supports 20-, 24-, and 32-bit recording to solve some of the digital mixing and dithering problems I referred to earlier. It uses the same file formats as Cool Edit. Whereas Cool Edit can record at sampling rates up to 1MHz (if your soundcard has sufficient bandwidth), Cool Edit Pro can record at 10MHz. (I suppose that even the most ardent hard-core digiphobe might admit that a 32-bit—read 192dB dynamic range—recording, sampled at a few megahertz, would be more accurate than anything analog. Of course, it may be harder to admit that it sounds better, too.)

EMPOWERING THE STARVING ARTIST

I haven't tried Cool Edit Pro yet, but at only $400 it would appear to be a very powerful tool in the hands of any amateur recording artist. You could have an outstanding recording and production studio for perhaps $10,000 by simply purchasing a state-of-the-art PC, high-performance soundcards, and a few really good microphones. With tools like this, there's no longer an excuse for your recordings to sound unprofessional. And with the rise of the web as a music medium, there will be many opportunities for audiences to be found.

There are also a number of other programs similar in scope and purpose to Cool Edit. Most of them use .wav and other popular file formats, so recordings made with one program can be used in another.

SOME MINOR CAUTIONS

The only problems I experienced were occasional hiccups with large files (that's where horsepower on your PC becomes important) and sometimes needing to be patient in saving or processing those large files. Also, on one of my PCs, the hard drive was starting to exhibit some problems (it would occasionally develop a bad sector), and heavy use of Cool Edit accelerated the problem. I replaced the drive, which was under warranty, and the problem went away. As I said earlier, you don't need a state-of-the-art PC to use Cool Edit, but if your applications are ambitious, it will use every bit of muscle available.

SUMMARY

Cool Edit is really a very ambitious and well-executed program. It is very thoughtfully designed and has nearly every feature that I could need, with the exception of a utility that allows you to record from a CD without going through the soundcard (in other words, direct to digital), which for many soundcards would make quite a difference. Another desirable feature would be a built-in utility similar to Adaptec Easy CD Creator, which moves data from Cool Edit back to a recordable CD and manages the transfer process.

If you purchased traditional analog recording, mixing and processing equipment, you could have easily spent $20,000 or more and not have as good an editor as Cool Edit. Considering it's only $69, I think every audiophile with a PC should buy it just for fun. You can download it from www.syntrillium.com (it's 8MB) in a few minutes, free of charge, and play with any of the features two at a time before you register. Happy editing! ❖

COST

You can audition all of Cool Edit's features with the shareware version. When you register your copy, Syntrillium will issue a registration number to unlock the shareware version so that you can fully use Cool Edit.

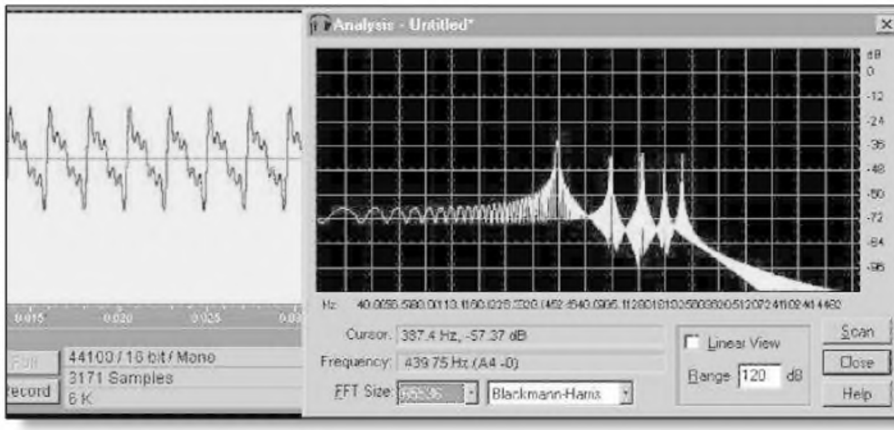

FIGURE 25: Frequency analysis and FFT panel. A-1625-25

Books for the Tube Audio Beginner

Being a beginner is exciting, but can also be frustrating when you can't find answers to your questions. In this article the author suggests some books—both from the golden age and from the present—that beginners may find interesting. **By Larry Lisle**

The first set of books I recommend is titled *Basic Electronics* by Van Valkenburgh, Nooger, and Neville. These theory books are written in a clear style with outstanding illustrations. They are the best books I've seen in taking the reader from practically zero electronic knowledge through some pretty sophisticated concepts, as painlessly as possible.

The books were originally written for the Navy and used as training manuals during World War II. After the war they were published by Rider, and can be found as five separate volumes. Vol. I (on power supplies) and vol. II (on tubes and amplifiers) are of the most interest. They were also published with all five volumes combined in one book. A sixth volume on solid-state was added later.

ABOUT THE AUTHOR:
Larry Lisle is a teacher in Rockford, IL. He has been writing articles, generally of a technical nature, since 1968. His hobbies include building and restoring audio and radio equipment, amateur radio, and coaching basketball.

TEACHERS

John Rider, who was a publisher of electronic books and manuals, and was also a writer of distinction, wrote one of

PHOTO 1: Here are a number of excellent books that introduce tube theory. They were very popular in the golden age of tubes and are easy to find.

the best books on how tubes actually work, *Inside the Vacuum Tube.* He takes the material that textbooks often compressed into a few pages and patiently explains it in ways that everyone can understand. Rider was a superb teacher and wrote for the reader who was trying to learn electronics from a book—many other classic texts were intended to be used in a classroom with a professor present. There's a real difference!

PHOTO 2: Abraham Marcus wrote books on radio repair as well as basic theory. He shows how to repair circuits by first explaining how they work. Marcus was an excellent teacher and worked hard to make the material understandable. *The Radio Amateur's Handbook* has lots of information for the enthusiast.

Rider was always trying little tricks to make learning easier for the student. For instance, in this book graphs are repeated as needed, so you don't need to keep flipping back and forth between the illustration and the text. He also uses lots of illustrations. I recommend his books very highly.

John T. Frye was a radio and television repairman who wrote many articles from the late '40s into the '70s. His *Basic Radio Course* is still a great reference for anyone studying the electronic circuitry of the tube era.

Another writer who was primarily a teacher was Abraham Marcus. His *Elements of Radio* is an excellent introduction to vacuum tubes. He also wrote an outstanding series of radio repair books. It may seem a little strange to suggest repair books to beginners who haven't built anything yet, but many of us old-timers in the tube circuit business learned a lot of what we know from Abraham Marcus.

Before you can repair a circuit you must understand how it works. Marcus was a great teacher who presented the lessons with the attitude of "you can learn this!" All of the Marcus repair books approach the same material from different angles, so you won't be wasting your money if you buy more than one of his titles, including *Elements of Radio* (mentioned earlier). They are usually very low-priced, simply because they were so popular and so many were in circulation.

The Radio Amateur's Handbook, published by the American Radio Relay League, is a valuable source of information on basic theory, audio amplifier design, construction practices, and tube data. The most useful editions for tube audio are from the '50s.

CONSTRUCTION

Books on theory are only part of what a beginner needs. There were many excellent books written during the golden age of tubes that showed the beginner how to actually build projects. Among the best were those by Alfred Morgan, including *The Boy's First* (and *Second*, *Third*, and *Fourth*) *Book of Radio and Electronics*. This series takes a reader from building crystal sets in the *First Book,* through building battery-powered amplifiers, to putting together a nice little push-pull amplifier with negative feedback in the *Third Book*.

The *Fourth* is all solid state with simple radios and amplifiers. The illustrations showing exactly how to wire and assemble the projects are especially helpful. Unfortunately, the books are rare and tend to be rather expensive if you can find them.

PHOTO 3: Alfred Morgan's *Radio and Electronics* series helped many beginners get started. Crystal sets are good first projects and *Radios that Work for Free* will show you how. Simple kits are available from Antique Electronic Supply.

There are, however, many other project books, such as those published by *Popular Science*, *Popular Mechanics*, and *Radio-Craft*, that show wiring and construction details, so keep an eye out for them. Some of these have been reprinted by Lindsay Publications, Inc., PO Box 538, Bradley, IL 60915, 815-935-5353, along with others for short-wave enthusiasts and other electronics buffs.

SAFETY NOTE

Some of the circuits in project books written during the '40s and '50s were of the AC-DC variety. That is, they didn't have a transformer and used the line voltage directly. This can be dangerous if you're not familiar with them, so always use an isolation transformer between the wall socket and the circuit (available from Antique Electronic Supply, 6221 South Maple Ave., Tempe, AZ 85283, 480-820-5411).

If you've never built anything electronic, I recommend a crystal set as a first project. A crystal set is a very simple radio that uses an antenna, but doesn't require batteries or other sources of power. It's easy to build, fun, and functional. Modern books about crystal sets include *Radios That Work For Free,* by K.E. Edwards. The book and crystal set kits are in the Antique Electronic Supply catalog.

Kits for simple tube amplifiers can be found at the same source. They work on batteries, which is strongly recommended for a beginner, from the standpoint of safety. The amplifiers are designed for headphones, but you can add a transformer and speaker for quiet listening. They're great for gaining experience, and you can use them with a crystal set, CD player, and so on.

Finally, there are a number of very useful new titles being published today that are available from Old Colony Sound Laboratory. *The Joy of Audio Electronics* by Charles Hansen isn't about tubes, but it is intended for the beginner. The author encourages the reader to get started in audio electronics and does his best to help. *A Beginner's Guide to Tube Audio Design,* by Bruce Rozenblit, is a little more advanced than some of the other books I've recommended, but it helps the beginner take the next step from the circuits of the golden age to those of the present (and delves into the basic math of electronic design).

Valve Amplifiers, by Morgan Jones, is an outstanding text that covers basic theory, as well as modern design practices. There's something here for readers of every level. The chapter on construction practices should be mandatory reading for anyone beginning tube amp construction.

There are many other excellent books I haven't mentioned in this short article, so if you see a title that looks interesting, don't hesitate. It's useful to have more than one book on a subject such as tube electronics, even if they're written at the same level. Looking at a concept from a different viewpoint often makes it more understandable.

You can obtain the out-of-print books by inter-library loan, from used book dealers on the Internet and otherwise, and at electronic flea markets, usually at very reasonable prices. Modern audio texts are available from Old Colony Sound Laboratory, PO Box 876, Peterborough, NH 03458, 888-924-9465.

So start your library today—and, just as important, start building! ❖

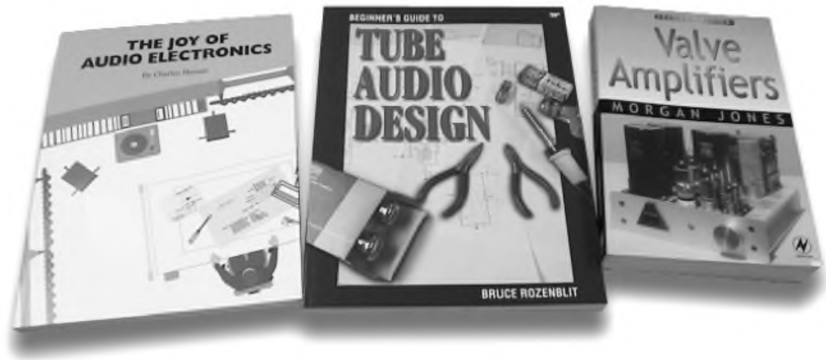

PHOTO 4: Here are some recent books on audio. These will take you into the world of modern circuits and math.

Mapletree Octal 6

from page 57

waveform. The duration between HF peaks is about 18µs, or 55kHz. There was no evidence of HF peaking at this frequency during the frequency-response measurements, so I repeated them again with the same result.

The THD+N performance of the Octal 6 did not vary noticeably with either frequency or load. The distortion level was essentially the same at 2V RMS output from 20Hz–20kHz and with loads from 10k–100k.

The output voltage at 1% distortion (clipping) is 12.5V RMS in CF mode and 17.7V RMS for SRPP. The Octal 6 never really goes into hard "brick wall" clipping. The peaks of the waveform are softly compressed with increasing output voltage. At 2V RMS output, 20Hz–20kHz, the THD+N spans only 0.10%–0.15% in either mode.

Maximum output voltage is over 35V RMS in CF mode and over 26V RMS in SRPP, at all frequencies from 20Hz–20kHz. Distortion is only 3.2% for maximum SRPP output, but over 10% for CF mode maximum output. I reduced the CF mode output to 16V RMS, the maximum output of the SRPP mode, but distortion was still above 7%. The SRPP mode thus delivers lower distortion across the entire output voltage range of the amplifier. The only exception I found was the 20Hz distortion with a low 10k load (0.46% SRPP versus 0.20% CF), but this is well below the recommended 50k load.

In a reversal of the above results, the SRPP mode produced higher intermodulation distortion (IMD) than the CF mode. The CCIF IMD (19 + 20kHz) at 12V p-p into 10k (the input impedance of my IM Distortion Tester) was 0.063% CF and 0.11% SRPP. Multi-tone IMD (9kHz + 10.05kHz + 20kHz) was 0.029% CF and 0.052% SRPP.

The spectrum of a 50Hz sine wave at 2V RMS into 47k, CF mode, is shown in *Fig. 5*, from 0–1.3kHz. The THD+N measured 0.14%, with the dominant second harmonic at −58dB. Notice the series of odd 60Hz power-line harmonics at 180Hz, 300Hz, 420Hz, and so on. This is consistent with the power-line spikes noted in the residual distortion and hum and noise waveforms, and is produced by the full-wave rectification.

It has been my experience that a fast recovery diode will sometimes adversely interact with parasitic reactances in the amplifier to produce higher rectification spikes than if a standard recovery diode had been used. This is probably academic in the case of the Octal 6, since the THD+N is low for a tube amplifier design.

A repeat of the above data for the SRPP mode is shown in *Fig. 6*. THD+N here is 0.21%, with the second harmonic at −54dB.

SQUARE WAVE TESTS

I viewed the response of the Octal 6 to three square-wave test frequencies on an analog scope using a 47k load. The response at 40Hz showed a moderate and acceptable amount of tilt, with less tilt in CF mode than SRPP. The 1kHz square wave was just about perfect. The 10kHz square wave showed moderate leading-edge rounding, with no hint of peaking or the 55kHz ringing on the SRPP waveform in *Fig. 4*. I can't really explain the presence of that HF signal in both SRPP channels.

Manufacturer's Response:

I welcome the opportunity to respond to the technical and listening reviews of the MAD Octal 6 Preamplifier. I am also appreciative of the effort required to evaluate the technical, aesthetic, and aural properties of a new audio component with no prior exposure to it. The reviewers have done an excellent job.

First, I should point out that the Octal 6 was first developed as a kit with construction methods and components chosen to maximize the performance/cost ratio. The assembled unit, which is the subject of the review, employs essentially the same circuitry but has slightly different styling including a solid oak base and an enclosure for the separate heater and plate power transformers (these permit switching to 230VAC/50Hz operation if required).

I personally wire and test each unit on a customer-by-customer basis. Thus, a customer can specify if outputs are to be wired for bi-amplification or for recording. In the kit version, there are instructions for wiring each option. The kit also offers the choice of side or rear panel mounting of the input/output RCA jacks.

As an engineer, I am pleased that Charles Hansen's performance measurements largely verify the published specs based on my own tests. I am unable to test for IM distortion so it is reassuring that the figures are of the same order as those for THD. The primarily second-order content of the harmonic distortion is not surprising given the tube characteristics.

I measured lower residual noise levels for the two active modes than Charles found. This may have something to so with the measurement conditions. I disconnected all inputs and measured the output voltage at full volume. I have noticed that noise increases somewhat when inputs of any kind are connected. I will be investigating the power supply switching transients. Perhaps these can be minimized by the use of small capacitors across the diodes or the transformer secondary.

Turning to the listening tests carried out by Nancy and Duncan MacArthur, I was disappointed that the system into which the Octal 6 was inserted was not well suited to the use of such a preamplifier. As they point out, with a 2V output from the DAC and a power amp sensitivity of 0.185V, severe attenuation of the signal is required somewhere along the line. They chose to make use of the level control on the Manley Stingray power amplifier. It would have been preferable to audition the Octal 6 with a power amplifier such as the AE-25 Super Amp reviewed in Glass Audio *5/00 with a sensitivity in the 1V to 2V range.*

I have found that with many amplifiers, the low-pass filter action produced by a level control set to a low volume causes enough reduction of high frequency response to change the aural impression independent of the nature of the input signal. Thus, it may have been preferable to change the volume control setting on the preamp rather than the power amp when comparing the passive and active modes so as to eliminate this possibility. Nevertheless, I think the MacArthurs did a thorough job in presenting the subtle differences between the two active modes.

I, too, find that on casual listening, taking into account the differences in gain, all three modes sound quite similar, with differences showing up only after repeated listening to a range of program material. Their ratings of the aural characteristics naturally reflect sonic preferences or expectations that may not always be consistent with the sonic signature of the Octal 6. To help me interpret their findings, it would have been useful to publish the ratings of the passive mode along with the two active modes. It doesn't seem reasonable that passive mode rated ten in every category.

Lastly, I should comment on the aesthetic issues raised by the MacArthurs. Since many customers have been enthusiastic about the styling, I can only conclude that, as one might expect, aesthetics are subjective. The "look" is certainly not one of mass production, since each unit is indeed prepared much as a "home-brew" component would be. It could be argued that there is an intangible quality inherent in a "hands-on" approach that may be appreciated by some but not by others. ❖

Llcyd Peppard, Ph.D.
Owner
Mapletree Audio Design

New Chips on the Block

Wolfson WM8740 Audio DAC

By Charles Hansen

The WM8740 is a high-performance stereo DAC designed for audio applications such as CD, DVD, home theater systems, set-top boxes, and digital TV. The WM8740 supports data input word lengths from 16 to 24 bits and sampling rates up to 192kHz. It consists of a serial interface port, digital interpolation filter, multi-bit sigma delta modulator, and stereo DAC in a small 28-pin SSOP package. The WM8740 also includes a digitally controllable mute and attenuator function on each channel.

The internal digital filter has two selectable rolloff characteristics. A sharp or slow rolloff can be selected depending on application requirements. Additionally, in the 8fs mode, the internal digital filter can be bypassed and the WM8740 used with an external DSP-based digital filter or HDCD decoder.

The WM8740 supports two connection schemes for audio DAC control. The SPI-compatible serial control port provides access to a wide range of features including on-chip mute, attenuation, and phase reversal. A hardware-controllable interface—in which two WM8740s can be connected in parallel to provide higher performance differential outputs without the need for external components—is also available.

SPECIFICATIONS

SNR 117dB
THD+N –105dB for stereo A weighting
Power +3.5V DC single supply

APPLICATIONS

CD, DVD audio
Home theater systems
Professional audio systems

Wolfson Microelectronics Ltd., Lutton Court, Bernard Terrace, Edinburgh, EH8 9NX UK, +44 (0) 131 667 9386, FAX: +44 (0) 131 667 5176, or 508-771-4346, sales@wolfson.co.uk, www.wolfson.co.uk.❖

BOGUS BOGEN

Unfortunately, the wrong schematic accompanied my letter about the Bogen HO-125 (*aX* 1/01, p. 97). The schematic shown is actually a Bogen HO-50. I can see how the confusion arose, as the two units are very similar.

Referring to the correct schematic (*Fig. 1*), the HO-125 does indeed operate at 840 plate volts. Other differences in the HO-125 include parallel 6SN7 halves as drivers for increased drive current, and another 6SN7 (both halves parallel) used as a bias regulator.

I am aware that 807s are rated at 750V maximum; however, I have seen pairs used in amateur radio transmitters as class C RF amplifiers and class AB modulators at 800-plus volts without problems. The ICAS rating for class-C operation is only 600V.

Apparently, push-pull 807s can output 120W or more. The RCA transmitting tube manual TT-5 shows a modulator circuit with a pair operating class B with 750V and 120W output. In the HO-125, according to my service manual, the output transformer is 4.8k load (plate to plate), which is lower than the tube manual recommends.

Eric Barbour replied in a letter that the HO-125 was a "notorious tube killer." I can neither confirm nor deny this, since I have no actual experience with the H0-125. I can confirm, however, that 807s made in the early 60s were much more reliable than those made in later years. Early ones can be found both NOS and used at hamfests.

Mr. Barbour also called my attention to the fact that the H0-125 had only high-impedence line outputs, and no voice-coil outputs.

I still believe that with a suitable high-quality output transformer, the HO-125 would make an excellent hi-fi amplifier. Hammond's PT-1650R or Dyna's A-440 (if you could find one) would be good choices. I also doubt that this amplifier could output 120W continuous, but in a typical home listening situation, there is a large differential between average and peak levels. Reliability (as well as sound quality) is greatly enhanced when an amplifier is operated well below clipping levels, allowing head room for peaks.

One thing that perplexes me about this unit is its power-supply design. The circuit is a choke input type, with only a small amount of filter capacitance (2µF) following the choke. While it is true that a choke input circuit is less demanding on the rectifier, it seems unrealistic that a single 5R4GY rectifier could support this amplifier.

Two 0.5µF capacitors (C9 and C10) are series-connected across the choke, and are referred to as "fixed trimmers" in the parts list. Perhaps someone could enlighten me as to their function.

Al Forbes
Charlotte, N.C.

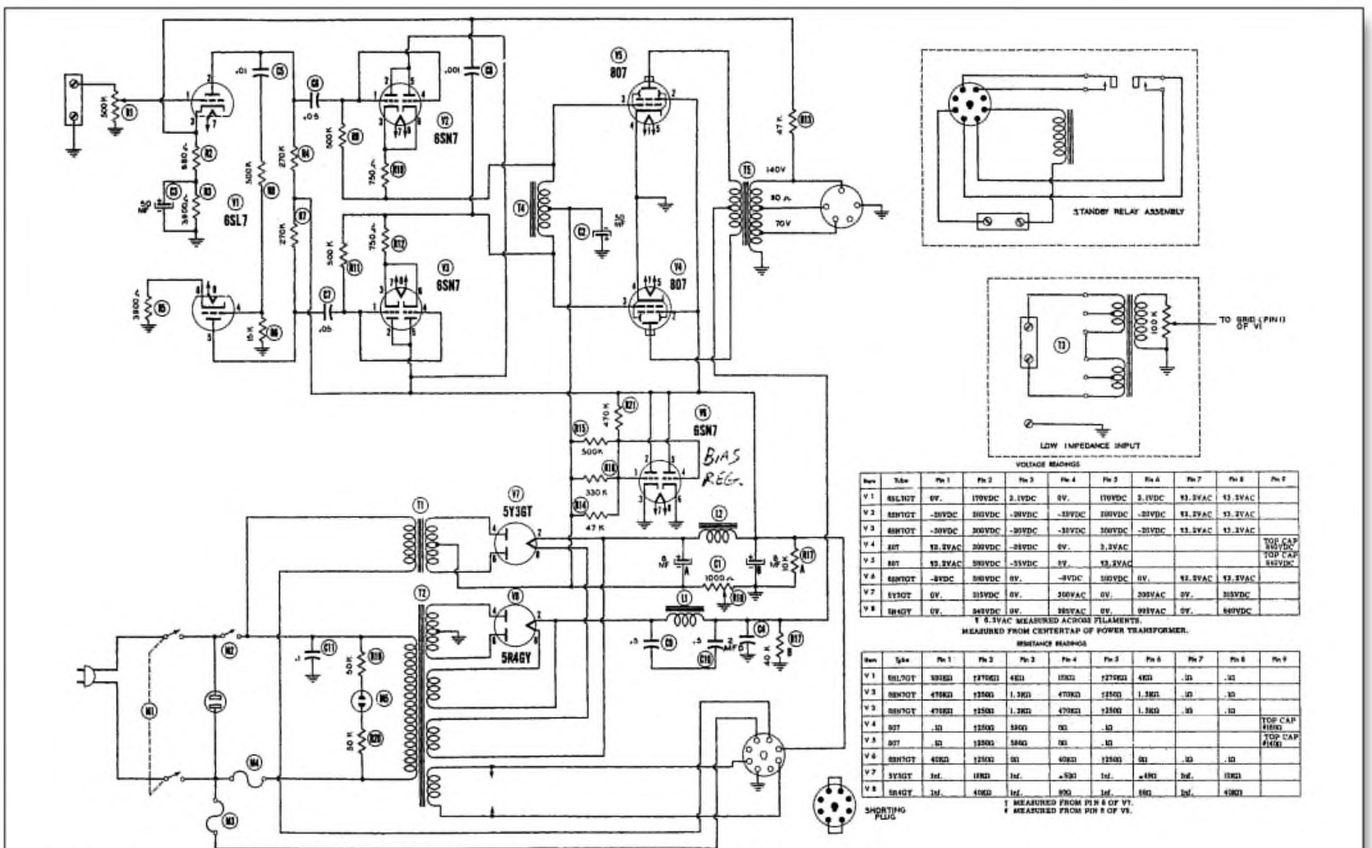

FIGURE 1: Bogen HO-125 amp.

PREAMP UPDATE

In Part 1 of my preamp article ("A Phono Pre-Preamplifier for the CD Era," *aX* 1/01, p. 24), the component placement and PC board patterns (Fig. 2) are distorted. It looks as though the patterns were enlarged for ease of presentation. However, in so doing, the traces were "pulled apart" and the boards would not, therefore, work properly. I assume that this was a problem with manipulating the electronic storage-supplied document. They are shown correctly in *Fig. 2*.

On the schematic (Fig. 1) included in the article, Q1 and Q3 are shown as MPS 6523 transistors, while Q2 and Q4 are shown as MPS 6521 devices. Those were the Pspice transistor models used in CAD analysis of the circuit's performance. Those transistors will work fine in this application, but the devices listed in the parts list (Table 1), i.e., LS 352 and LS 312, respectively, are the preferred transistors and are the ones on which the component placement illustration was predicated.

Also in the parts list, resistors R13 and R14 are listed on the line with R8 as 2k21Ω resistors, but should have been listed on the line with R6 and R7, that is, as 6.2Ω resistors. Finally, capacitors C7 and C8 should be listed on the line with C9 and C10, since they are also 1.0µF units rather than 220nF units. The schematic correctly shows these resistors and capacitors.

Norm Thagard
Tallahassee, Fla.

KUDOS

Let me congratulate you on your new magazine. I was tempted to call it the new old magazine because it shows a good deal of its heritage back to *The Audio Amateur*. I was really prepared to dislike the new format because I am not a real fan of solid-state sound equipment. I don't like to use it or work on it.

I was afraid the publication would be unbalanced and I would need to wade through articles that hold little interest for me. I was wrong. I think there is enough tube and speaker material to keep me happy and enough solid-state material to keep me at least a little modern. I enjoyed the entire magazine. Bravo!

Rob Lewis
Chandler, Ariz.

Just a short note to let you know how pleased I am with the first issue of *audioXpress*. At a time when the general "dumbing-down" of the population (at least in Australia) has meant that electronics magazines are now little more than "picture books," *audioXpress* stands out as a beacon, and the magazine represents all that was once great in the world of electronics publishing. I hope to remain a subscriber for a long time to come.

Terry Robinson
Woodend, Vic
Australia

Your decision to combine the *SB* magazine is a good one. I was ready to expand interest in the next related direction, and I think you did the natural thing...some diversity, if you will. I'm looking forward to finding a great tube design and building it in the future at a substantial savings—I hope with the help of your magazine.

Good luck and much success to you and the staff that makes it happen!

Marty Pawlowski
Byfield, Mass.

TRANSFORMER MEASUREMENTS

I wish to extend my thanks to Neal Haight for his letter (*GA* 5/00, p. 57). I found out that the transformer available has two primary windings showing each one's input impedance to be 450 for 8Ω and 225 for 4Ω for each of the two secondaries.

Is a pair of 8042/4652s appropriate in this situation? Which types of output circuits would be applicable for stereo? Is a B+ of 250 or less acceptable?

I'm considering as a phase splitter driver, the circuit of Ari Polisois (*GA* 1/00, p. 32)—in my case, the 6CU5. The 2A transformer I also have is a 6V center tap and 115V "secondary." Perchance the positive "leg" can be coupled to the 250 DC should circuits that require DC be on the plates.

R.L. Summer
Washington, D.C.

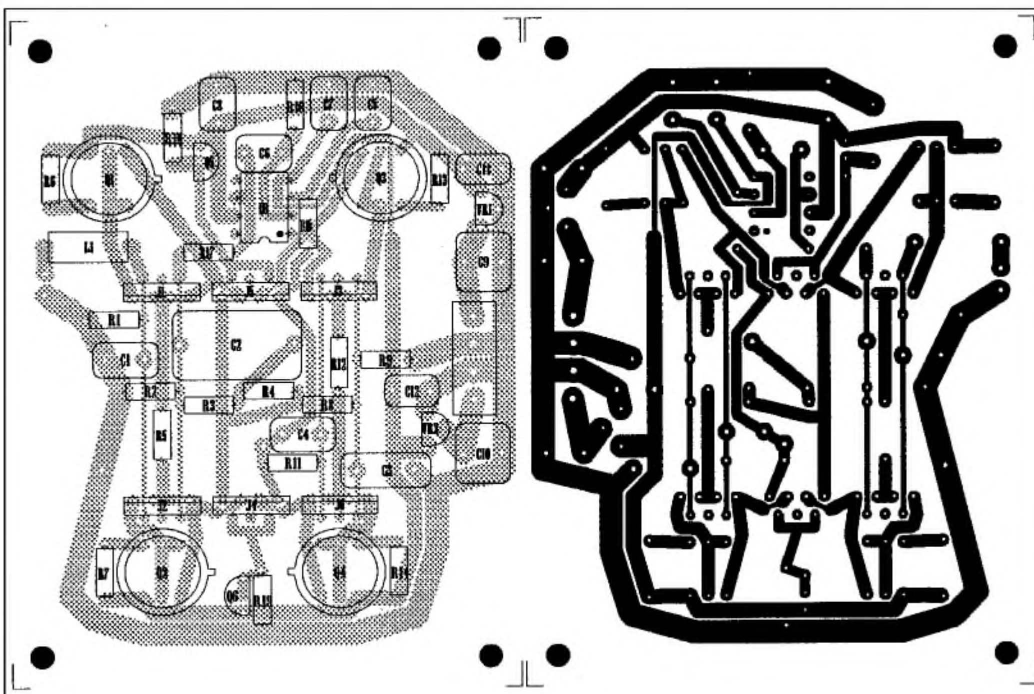

FIGURE 2: Preamp parts placement and PC board patterns.

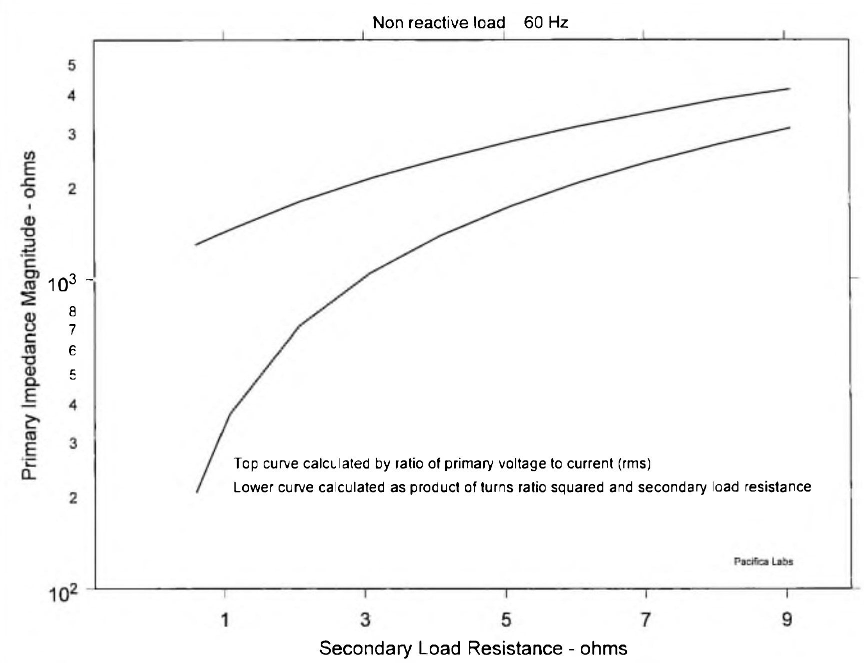

FIGURE 3: Calculated vs measured impedance magnitude.

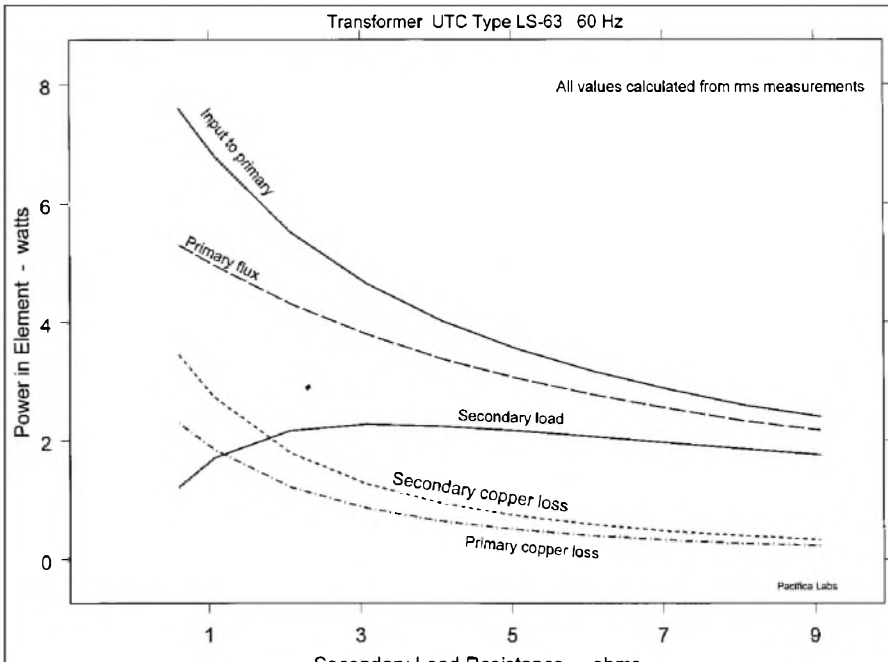

FIGURE 4: Transformer-coupled power elements.

Neal Haight responds:

It sounds to me as though your transformer's primary impedance is too low for tube use. The 225/450Ω primaries, versus the 4/8Ω secondaries, point to a transformer that was designed to be driven by transistors. Many transistorized amplifiers produced back in the 1960s used transformers to drive speakers. Electrolytic capacitors were used for this purpose later on.

The dual primaries connected to two transistors in a push-pull arrangement, while the secondaries, which were connected in series, provided the 4 and 8Ω outputs. Offhand, this is the only purpose I can think of for this type of transformer. Due to its low impedance, this transformer would not work well in any type of tube circuit.

I have no technical information for the 8042/4652, and, unfortunately, can't comment on it.

Like Mr. Summer, I am also impressed with the Ari Polisois phase splitter. I wouldn't recommend any tube other than the EL84 for that one because of that tube's

300V maximum plate and screen grid rating. The 6CU5, with its 200V maximum plate and screen grid rating, would literally fry. An upgrade would be a 7189 or 7189A.

The letter in *GA* 5/00 on transformer impedance measurement (p. 56), while a simple approach to the problem, is unfortunately incorrect. Two errors exist in the described method. One is that the so-called 120V mains in the US are seldom at 120V. This value can easily have a ±10% variation, and in some locations may vary between 125 to less than 100V. This is easy to correct, however, by measuring the voltage across the transformer primary during the test.

The second error is not as easily corrected. Open circuit voltage measurements of a power transformer will not provide a voltage ratio value that will be useful in the calculation method given in the letter. To obtain a practical value for the primary impedance of a transformer, you should use a load of approximately the "expected" value. The reason is that the current in the transformer has a significant effect on the reactance of the device.

09-24-1994 14:25:51 CODE OHMSTEP2 60Hz

(Turns ratio)^2 = 338.1 Primary = 390 ohms Secondary = 1.72 ohms

	Pv	Pi	Sv	Si	Sr	Pz	Wp	Ws	Ws/Wp	N*Sr	Pz/Sr
1	99.1	76.6	0.857	1.415	0.70	1294	7.589	1.212	0.160	205	2137
2	98.9	68.7	1.355	1.258	1.17	1440	6.799	1.705	0.251	364	1337
3	99.1	55.7	2.118	1.020	2.17	1777	5.521	2.161	0.391	702	856
4	99.1	46.9	2.646	0.858	3.17	2112	4.653	2.270	0.488	1043	685
5	99.1	40.5	3.021	0.740	4.17	2443	4.018	2.236	0.556	1380	598
6	99.2	35.7	3.314	0.652	5.17	2778	3.540	2.161	0.610	1719	547
7	99.2	31.9	3.542	0.582	6.18	3111	3.162	2.061	0.652	2058	511
8	99.2	28.9	3.721	0.526	7.17	3436	2.862	1.957	0.684	2392	486
9	99.2	26.1	3.870	0.479	8.18	3799	2.592	1.854	0.715	2732	470
10	99.1	24.2	3.992	0.440	9.19	4092	2.402	1.756	0.731	3067	451

Transfer Analysis (Ratio is flux transfer efficiency [copper losses removed]

	SRC	PCL	SCL	Pflux	Sflux	Ratio
1	0.605	2.288	3.444	5.301	1.212	0.6528
2	1.077	1.841	2.722	4.958	1.705	0.7625
3	2.077	1.211	1.789	4.309	2.161	0.8575
4	3.084	0.859	1.266	3.793	2.270	0.8984
5	4.083	0.641	0.942	3.376	2.236	0.9183
6	5.083	0.497	0.731	3.043	2.161	0.9345
7	6.086	0.396	0.583	2.766	2.061	0.9442
8	7.074	0.325	0.476	2.537	1.957	0.9496
9	8.079	0.266	0.395	2.326	1.854	0.9600
10	9.073	0.229	0.333	2.173	1.756	0.9545

TABLE 1: Transformer-related values.

Also, you must use the voltage across the primary terminals in the calculation when a load is placed across the secondary. If an incandescent lamp is used for safety reasons, the voltage drop across the lamp must be accounted for. As the current increases in the primary due to transferring power into the secondary load, the voltage drop across the incandescent lamp must be subtracted from the supply voltage.

I derived *Figs. 3* and *4* from a test in which an output transformer (UTC type LS-63 with an open circuit voltage ratio of 18.4) was energized across the primary by approximately 100V AC at 60Hz. The secondary was loaded by a variable, nonreactive resistance in the range of 10Ω to 1Ω. I recorded voltage across the primary, current in the primary, the load resistance, and the voltage across the load for each specific resistance value, and calculated the value of power dissipated in the other transmission elements. These elements are primary and secondary copper loss, as well as the efficiency of the flux transfer, and are plotted in *Fig. 3*.

Figure 4 shows the derived primary impedance magnitude for the range of load resistance used. A second plot shows the impedance calculated by the suggested method using the product of the load resistance and the squared value of the open circuit voltage ratio. The two values are obviously not the same.

Figure 3 shows how the secondary power varies with load for a constant voltage across the primary. While not usually considered an important parameter in audio frequency transformers—except for very high power transmission—low frequency energy transfer through a transformer exhibits the same "reflected power" characteristic—due to an impedance mismatch across transfer elements—as occurs with high frequency components. The point of maximum power transfer to a load, is, by definition, the impedance of the energy source (secondary to load).

For the specific conditions of these tests, this secondary impedance value is approximately 3.1Ω. The reflected impedance to the primary, for this condition, is approximately 2100Ω. These are the values that would normally be quoted in the transformer specification for these specific conditions (data item No. 4 in *Table 1*).

Two additional comments may be applicable when considering this type of test. Using 60Hz as the measuring frequency for an audio transformer, whose range can be 20 to 20000Hz, while producing a value for 60Hz, may be misleading when you consider the full audio range.

Second, the values being calculated are impedance magnitudes, not impedance values. Impedance is a complex term; i.e., it has a reactive component, which in this type of test is not considered, and is in fact carefully avoided by using a nonreactive load. The reactance of the load can greatly affect the impedance transfer characteristics of the transformer.

Don Jenkins
Tarzana, CA

Neal A. Haight responds:

I thank Don Jenkins for taking the time to follow up and respond to the method that I prescribed for determining output transformer primary impedance. In his testing procedure, he connected a secondary load. While he gives good reasons for doing so, I recommend not doing this, for added protection against overheating and possibly shorting out the primary windings or the voice coil windings. So please do not connect a secondary load during this test...other than your AC voltmeter.

While Don makes some very valid points why the procedure shouldn't work, surprisingly, it still does work. I tested a single-ended transformer with the impedance 5kΩ:8Ω printed on the body. The primary-to-secondary ratio in this case is 25-to-1. When I applied a (confirmed) 120V to the primary, the secondary provided 4.8V—verifying the ratio, and most of all the impedance, upon further calculation.

In defense of Don Jenkins, it is best to go by the manufacturer's specifications, if available. This is also the best way to go if you plan to devote care and big bucks to an amplifier project. On the other hand, if you're like me and don't have the big bucks for test equipment, other than a pocket-sized volt-ohmmeter, my testing procedure will quickly enable you to put your box of salvaged output transformers to use.

CURRENT ALTERNATIVE

I read with interest Joseph Norwood Still's article entitled "A 20W $260 Amplifier" (*GA* 5/00). Some of the virtues I like in the design are the reasonable price, easy setup (no bias adjustments or balance pots), and the use of readily available parts.

There is one aspect of the design that I believe could be improved upon. Placing the 1Ω bias monitor resistor in series with the plate could expose hobbyists to a shock hazard. Although readers are likely to be careful when measuring the bias current with a high voltage present, a slip of the hand could place hobbyists in direct contact with the 390V supply. A safer alternative would be to place the current sense resistor in series with the cathode, which sits at a much safer 30V.

To make matching the output tubes easier, you could place a 1Ω, ¼W, 1% resistor in series with both output tube cathodes (*Fig. 5*). Simply measure differentially across the two output tube cathodes. If the currents through the two tubes are balanced, one will read 0V.

The absolute value of the resistors placed in series with the cathodes is not very critical as long as the value is not so high as to affect the bias current. To obtain the most accurate results, the resistors should be closely matched. I am suggesting a ¼W value because such a resistor can act as a fuse if one of the output tubes shorts. You can calculate the total output stage bias current by measuring the voltage between either cathode and ground.

An alternative to adding resistors to measure bias and balance is to eliminate the resistor completely. If the tubes are well matched, they should not need to be selected for *acceptable* balance. If absolute balance is important, then you might consider balance adjustment capability as part of the design. That would detract from the simplicity and ease of use designed into the amplifier, which I believe is one of its prime virtues.

Jim Eding
San Jose, Calif.

Joseph Norwood Still responds:

Thank you for the very complimentary comments on my 20W, $260 stereo amplifier. Your comments regarding an alternate means of monitoring the currents of each 6L6 are well-received. If I build another amplifier for publication, I will use a separate cathode self-bias resistor for eac power output tube with a 1Ω current sampling resistor in series with the cathode resistor. Again, I thank you for your advice on an alternate means of monitoring the plate current of the 6L6GCs.

I commend Joseph Norwood Still for designing a fine, inexpensive amplifier, but I think he will find 400V on capacitors C1, C2, and C3 in the power supply when the unit is turned on, until the tubes warm up. Was R4 supposed to be across C3 to lower the voltage? C2 and C3 are rated for only 350V. I recommend using 100µF 450V capacitors for all three capacitors in the power supply. In the amplifier, you should change C1 to 1µF 630V for the same reason. I have found Mouser Electronics, www.mouser.com, to be an excellent source for parts.

The diagram uses the same symbol for connections and crossovers. This makes it look as though pin 5 is connected to pin 8 on the 6L6, and makes the wiring of the secondary of the output transformer puzzling.

Donald Passantino
Elmhurst, N.Y.

Joseph Norwood Still responds:

Thank you very much for your comments concerning my 20W amplifier and for the informative critique. You are correct: grid 1 of the 6L6 should not be tied to the suppressor grid and the cathode. In fact, the suppressor grid and the cathode connection are shown extended to the glass envelope. This is incorrect, since they are tied internally to the glass envelope.

R4 is a header resistor for the electrolytic capacitors. The capacitor voltage ratings of C2 and C3 could be increased to 450V, as you suggested, to ensure longer life and reliability. Capacitor C1, with its 250V rating, should be safe, as it receives its initial surge voltage through a 1MΩ resistor.

Again, I thank you for your constructive comments, which were indeed appreciated ❖

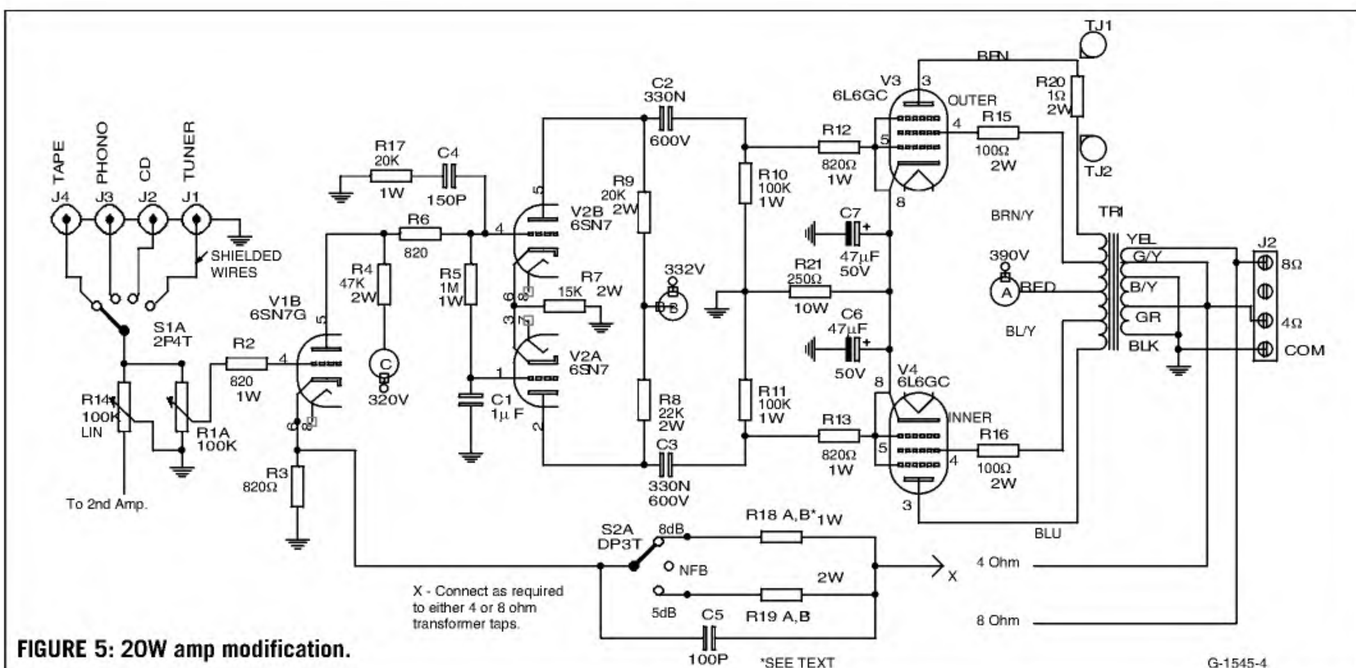

FIGURE 5: 20W amp modification.

New Chips on the Block

Chips Gone from the Block

By Charles Hansen

Here we are in the midst of the greatest information age in recorded history, and we can't seem to hang on to data only 15 years old. I have undertaken a number of fruitless searches for data sheets of recently discontinued integrated circuits: most recently for the Rohm BA1404 FM stereo modulator and the Toshiba TC9147P tuner controller*.

I searched the data sheet/applications section of both manufacturers' web sites, which did not even list these chips in their archive sections. I searched other sites such as Chip Directory, Gray's Web Pages, Data Bookshelf, NTE, Part Net, and SGS-Thomson. I also did a part number search in Google and Alta Vista. Finally, I contacted the manufacturers' tech support sites, all to no avail.

There is a lot of information on the web for upgrading the Ramsey FM-10 transmitter (it uses the BA1404 chip), but no data sheet for the chip itself. There was one dead link to a .gif file, so I know the information is out there somewhere. John Ramsey has sold 10k of FM-10 and FM-25 transmitters, but he does not have the BA1404 data sheet itself—the unit was designed in Europe.

It's not just Rohm and Toshiba. If it weren't for my old data catalogs, I would not have information on obsolete ICs from just about every chip maker. Yet there are still a large number of these discontinued chips successfully in service, and otherwise perfectly good equipment needing a replacement chip to put it back in service.

There used to be a company called Sunset Silicon that continued production of chips that the semiconductor manufacturers were going to discontinue. They would buy the masks and production rights, and ship to even small orders. Alas, they are gone as well.

When my company was closing what was the old Bendix Red Bank Division in Eatontown, N.J., they threw out boxes of data sheets on the Bendix hard-glass vacuum tubes they produced for the government and IBM in the 1950s—data that cannot be retrieved anywhere. The paper just took up too much space in the data archive area, so it had to go to make room for newer information. Much of this newer information then disappeared in the great clean-up prior to the facility closing in 1998.

This is a call to preserve our recent technical history. You can get a service or operating manual for nearly every piece of vacuum-tube equipment ever made, but if it's solid-state the odds drop alarmingly. If a proprietary IC was used in the design, the data for that device is most likely lost forever.

Somewhere on the internet there should be a place for this information. Semiconductor manufacturers could convert their old data or application sheets to .pdf files before they discard them, and post them on this as-yet-to-be-determined site.

* The Toshiba tech people managed to find a copy of the TC9147P tuner controller data sheet and sent it to me by Fed-Ex, at no cost! ❖

Motorola DSP56362: 24-bit Multichannel Audio Decoder DSP

By Charles Hansen

The DSP56362 is a high-performance DSP optimized for cost-sensitive consumer audio applications. A general-purpose DSP56362 is available as well as a multimode, multichannel audio decoder for consumer applications such as audio/video (A/V) receivers, surround-sound decoders, digital versatile disk (DVD) players, digital TV, and other audio applications (applicable licenses are required). The DSP56362 supports all of the popular multichannel audio decoding formats, including Dolby Digital Surround (AC-3), Moving Picture Experts Group Standard 2 (MPEG2), and Digital Theater Systems (DTS), in a single device with sufficient MIPS resources for customer-defined post-processing features such as bass management, 3D virtual surround, Lucasfilm THX5.1, soundfield processing, and advanced equalization.

The DSP56362 is a member of the 56300 Motorola Symphony™ DSP family and utilizes the single-instruction-per-clock-cycle DSP56300 core, while retaining code compatibility with the DSP56000 core family. The DSP56362 contains audio-specific peripherals and an on-board software surround decoder, and is offered in 100MHz/MIPS and 120MHz/MIPS versions at a nominal 3.3V.

The new Symphony® Surround Module (SSM) application centers around the DSP56362 and several components from AKM Semiconductor, which jointly developed a complete, low-cost digital audio reference design with Motorola, for use in development of new DSP decoder systems.

DSP56362 Features:

Multimode, multichannel decoder software functionality
Dolby Digital and Pro Logic
MPEG2 5.1
DTS

Digital Audio Post-Processing Capabilities:

Bass management
3D Virtual surround sound
Lucasfilm THX5.1
Soundfield processing
Equalization

Digital Signal Processing Core:

120 million instructions per second (MIPS) with a 120MHz clock at a nominal 3.3V
Object code compatible with the DSP56000 core with highly parallel instruction set
Data Arithmetic Logic Unit (Data ALU)
Program Control Unit (PCU)
Direct Memory Access (DMA)
Software programmable PLL-based frequency synthesizer for the core clock
Hardware debugging support: On-Chip Emulation (OnCE) module, Joint Test Action Group (JTAG), Test Access Port (TAP), and Address Trace mode.

On-Chip Memories:

Modified Harvard architecture allows simultaneous access to program and data memories. Program ROMs that may be factory programmed with data/program provided by the application developer.

3K × 24-bit program RAM
30K × 24-bit program ROM
5.5K × 24-bit X-data RAM
6K × 24-bit X-data ROM
5.5K × 24-bit Y-data RAM
6K × 24-bit Y-data ROM
192 × 24-bit bootstrap ROM

Off-Chip Memory Expansion:

Memory expansion up to 4-256K × 24-bit word memory for P, X, and Y memory when using SRAM.
Memory expansion up to 4-16M × 24-bit word memory for P, X, and Y memory when using DRAM.
Twenty-four data pin external memory expansion port (for high-speed external memory access allowing for a large number of external accesses per sample).
Chip Select Logic for glueless interface to SRAMs.
On-chip DRAM controller for glueless interface to DRAMs.

Peripheral And Support Circuits:

Enhanced Serial Audio Interface (ESAI) includes:
Six serial data lines, four selectable as receive or transmit and two transmit only.
Master or slave capability.
I2S, Sony, AC97, and other audio protocol implementations.
Asynchronous and synchronous operation.

Serial Host Interface (SHI) Features:

SPI and I2C protocols
Ten-word receive FIFO
Support for 8, 16, and 24-bit words
Byte-wide parallel Host Interface (HDI08) with DMA support features one serial transmitter capable of supporting S/PDIF, IEC958, IEC1937, CP-340, and AES/EBU digital audio formats.
Triple Timer module
On-chip peripheral registers memory mapped in data memory space.
Miscellaneous peripherals SHI, PLL, DAX, ESAI, GPIO

Reduced Power Dissipation:

Very low power (3.3V) CMOS design
Wait and Stop low-power standby modes
Fully-static logic, operation frequency down to 0Hz (DC)
Optimized power management circuitry

Package:

144-pin plastic Thin Quad Flat Pack (TQFP) surface-mount package

Contact Motorola Semiconductor Products Sector (SPS), www.mot-sps.com, and AKM Semiconductor, www.akm.com. ❖

Audio Aids

Sources For Carbide Router Bits

Like other readers, I prefer manually routing simple PCBs to the time and mess involved with etching. Two good sources for small carbide router bits are:

- American Science and Surplus, 3605 Howard St., Skokie, IL 60076, (847) 982-0870
- BW Trading Company, PO Box 692, Newark, OH 43058-0692, (740) 344-2772

Both companies frequently offer carbide bit assortments in the sizes and styles needed, for a very reasonable price. ❖

Doug Burkett
Eaton, Ohio

Classifieds

VENDORS

Home-theater poweramp kits
www.borbelyaudio.com

The LP track carries both stereo channels vs. alternating pits on the CD. The LP remains superior with "tight coupling for audio," $15; using an "OTL amplifier circuit," $35; driving efficient "Bass Horn" plans, $50. Visa, MC. ***STEEN AUDIO LAB***, PO Box 2185, Vancouver, BC V6B 3V7, CANADA.

FOR SALE

Model 109 audio noise and arbitrary waveform generator only $289. TDL Technology, Inc., www.zianet.com/tdl.

LinearX LMS System w/ ver 3.7 s/w $800 OBO and LinearX RTAjr with ver 1.14 s/w $600 OBO. Both in mint condition and used in non-industrial applications. All serious offers will be considered at jkorn1@pacbell.net.

WANTED

TUBE HIFI, HORN SPEAKERS, and HIGH END. Altec, Marantz, JBL, Mcintosh, Electrovoice, Fisher, Scott, Dynaco, Leak, Eico, Audio Research, CJ, Quad, Bedini, etc. Sonny, 405-737-3312, Fax: 405-737-3355, sonnysound@aol.com.

Yard Sale

FOR SALE

Brand new 845 tube mono amps with outboard Western Electric bias meter, $1,500 pair—with full warranty. Call Philip at (805) 501-5096 or e-mail trueanalog@aol.com.

ACI DV12 woofers (2). Like-new condition, rubber surround, rugged, dual voice coil, Fs 17Hz, Q_{TS}

0.44, V_{AS} 380, $75 each; dbx model NX40 noise-reduction system, $25. (828) 862-5567, or jfbundy@citcom.net.

Accuton C2-92 driver, $125 each; CLIO with PC extras, $850; Focal 11K7511 woofers, $225 pair; Focal 10L6411 woofers, $480 pair; MTM Raven R1.0 with Accutons and Sub cabs, $900 pair; Jordan Pyramids, $525 pair. Burt, mailboxb@att.net or (914) 961-4108 http://home.att.net/~mailboxb/sale.html

SEAS P17REX COAX. From Madisound, features SEAS 6 ½″ midbass, 25TFFN tweeter, and Leap designed crossover. Ideal for compact no-compromise center channel. Currently mounted in a 7″ cube with great results! List price of all components with leap crossover is $122. Sell for $45. Jerry, (262) 784-4726.

16NHT 1259 12″ woofers, $75 each; 24 Focal 5KO 13L 5.25″ bass-mid range spks, $40 each; two 80″ line source oak tweeter panels with 24 1″ Audax tweeters per panel (similar to McIntosh), $500 for both. Joe, (617) 254-0697 or jgemmato@earthlink.net.

Two power amps: for sale. Adcom GFA 555mk II, which has been turned on only twice and has been stored cool and dry; and Sonic Frontiers SFS 40 tube amp with little use and new matched Tesla tubes. Jim, (610) 378-1725 after 4:00 pm, or e-mail jamesg@excelonline.com.

Pair Plitron PAT-3025SE single ended output transformers (2.5Ω–4Ω), $200; pair Wakefield 476 K heatsinks (5 × 5 × 6), $80; pair Focal 7V014 DBL, $65; Celestion G-12 Alnico Blue guitar speaker, $120 (cost $270). All barely used. Rick, (406) 721-9463.

WANTED

EICO HF-32 mono tube amplifier, any condition—have one, need a second for stereo. As parts, need its power and output transformers—will build the rest. John Agugliaro, (845) 947-2748, e-mail: JAGUGL4546@aol.com.

Schematic with component values and identifications of solid state devices for A/D/S Model 10 Acoustic Dimension Synthesizer. Will pay for copying, handling, and postage. E-mail: jroesmer@libcom.com.

Schematics for 6BQ5/EL84-based single-ended amplifiers. Will pay any reasonable costs. Michael B. Wilcox, P.O. Box 69, Altamont, TN 37301, (931) 692-2762/2861.

WEBSITE RESOURCES

http://www.plitron.com

http://www.partsexpress.com

Adire
http://www.adireaudio.com

All Electronics
www.allelectronics.com

Audio Consulting
http://www.audio-consulting.ch

Divergent Technologies
http://www.divertech.com

Gasoline Alley
www.gasolinealleyllc.com

Marchand Electronics
www.marchandelec.com

McFeelys
www.mcfeelys.com

The Tube Store
www.thetubestore.com

Solen, Inc.
http://www.solen.ca

Vintage Hi-Fi
www.tubes.it

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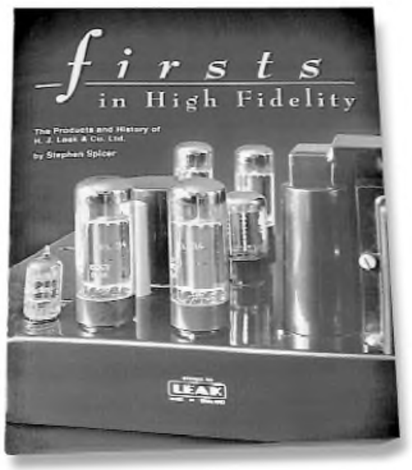
AD INDEX

Book Review

Firsts in High Fidelity—The Products and History of H. J. Leak & Co. Ltd.

Reviewed by Philip Taylor



Firsts in High Fidelity—The Products and History of H. J. Leak & Co. Ltd. Available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 603-924-9464, FAX 603-924-9467, e-mail custserv@audioXpress.com, $24.95, BKA58.

Walk around most vintage wireless meets and audio jumbles in Britain and you're almost certain to find Leak equipment, mostly from the late 1950s and early 1960s. Among hi-fi collectors and those wanting cost-effective valve sound for everyday use, Harold Leak's products are known to be reliable, well-made, and relatively easy to fix. Apart from transformers, there are few custom components and mostly standard circuitry.

Stephen Spicer's new book traces the history of H. J. Leak from his early days in the booming radio industry of the mid-1920s to the founding of Leak and its eventual sale to Rank Industries in 1969.

Early chapters detail Leak's start as a "Wireless Doctor" in 1926 with a school friend. This business was a sideline, but was successful enough to enable him to leave his employment in the wine trade. H. J. Leak and Co. was established in 1934, with Leak having gained prior experience in sound equipment engineering of the day by working for Gaumont British. By the end of WWII, Leak had 14 staff members designing and making custom audio equipment. Some Leak designs were published around 1937 by Norman Partridge in his booklets, *Partridge P.A. Manual* and *Partridge Amplifier Circuits*.

Leak surrounded himself with able personnel, and Stephen Spicer was able to interview a number of them, along with Leak family members. This chapter gives an insight into Leak's methods and ideas that led to the success of the company in the "golden age" of British hi-fi manufacturing in the 1950s and 1960s.

Leak products—amplifiers, tuners, and preamplifiers—are written up, with many photos and under-chassis views. The sandwich speaker and the moving coil pickup are illustrated, and mention is also made of some products that did not make it. Production dates and original prices of all Leak's equipment from 1945 to 1969 are included. The Leak-Rank era is illustrated with Leak-designed equipment and an overview, ending when Rank stopped British manufacturing in favor of Rotel-made Leak badged amplifiers and tuners. The sandwich speakers continued to be British-made until Rank discontinued Leak products in 1979.

There are 30 pages of schematics, covering all valved equipment from 1945 onwards, plus the solid-state Stereo 30, Stereo 70, and the Stereofetic tuner. The inclusion of these original Leak diagrams makes the book a real working reference.

There are some genuine nuggets—one in particular being a harmonic distortion chart for nine different amplifiers, at two power levels. There are some practical hints about restoration, and an extensive bibliography brings awareness of the amount of research needed to compile this extensive history.

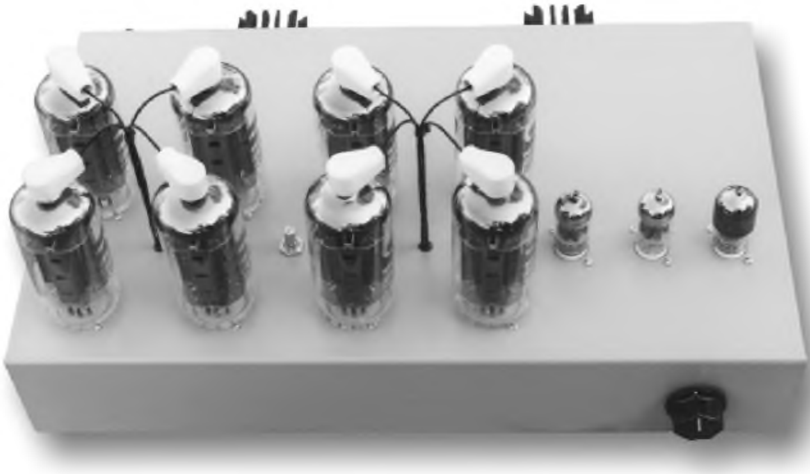
Some updates to the suppliers chapter are in order:

- Sussex Surplus of Horsham rarely supplies valves now.
- RS Components (formerly Radiospares) supplies only trade and industrial customers in the UK. Their Electromail division works from the same address and catalog, and will supply anyone on a cash-with-order and credit card basis, with no minimum order.
- Majestic Transformers no longer does re-winds.
- The Bulgin Bakelite connectors used on much Leak equipment are no longer made. It is still possible to find old stock at audio jumbles and elsewhere.

I can recommend Stephen Spicer's book to everyone wishing to know about Harold Leak and his company, to hi-fi restorers, and to those who want to hear a good tune on valved equipment that doesn't cost as much as a small car. I have only one slight moan—no schematic of the variable selectivity AM tuner. It turns out it was a bought-in design, with only a few hundred made in the early 1950s, and was expensive. FM radio transmissions started in the UK in 1955 and the VS-Tuner was discontinued, so I'm still looking for the circuit. ❖

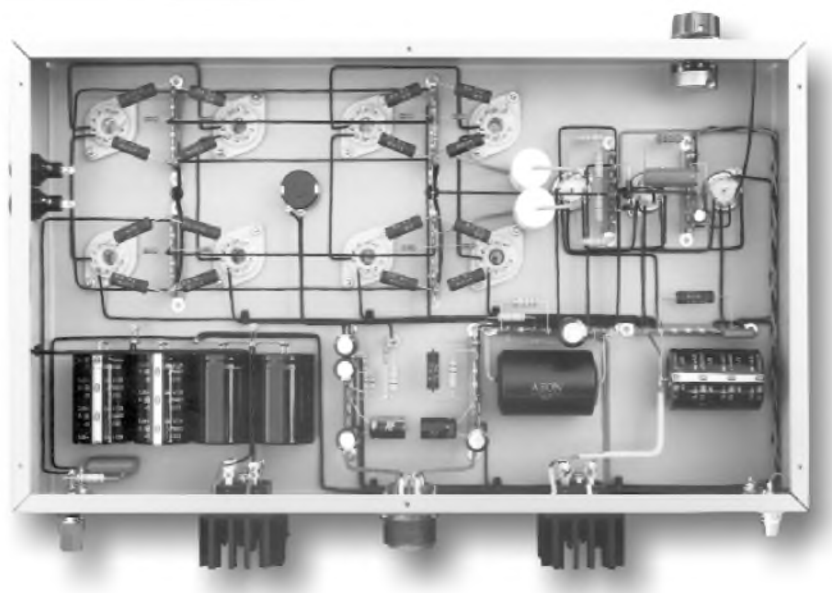
Glass Shard

Building Rozenblit's 80W OTL Amp



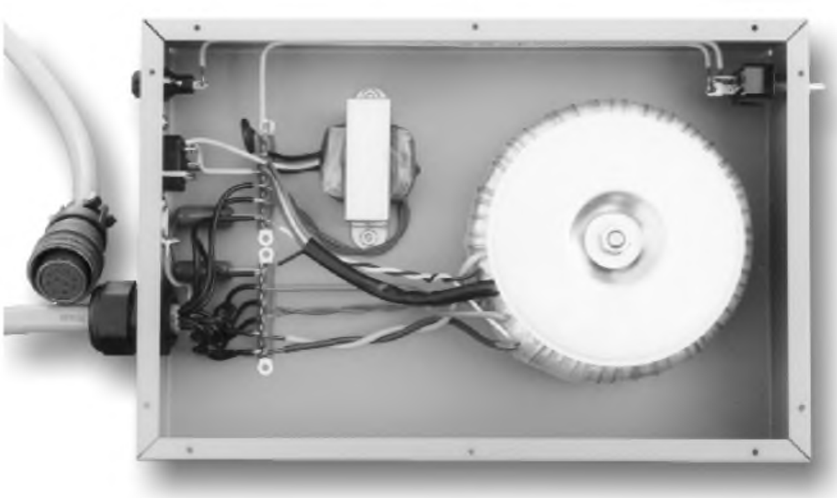
◀ **PHOTO 1: 80W version of Rozenblit's OTL.**

▼ **PHOTO 2: Inside view.**



Here are photos of my recently completed version of Bruce Rozenblit's 80W OTL from his book *Audio Reality* (available from Old Colony Sound Lab, (888) 924-9465, www.audioXpress.com). I purchased the toroid power transformer from Mr. Rozenblit. The layout is my own, as I have no idea what his layout looks like. I modified a few parts, such as the Amphenol mil spec connectors for the power supply umbilical and the addition of a volume control. As you would expect, the sound of the amp is quite good. ❖

Jerry Young
Spring Hill, Fla.



◀ **PHOTO 3: Power supply.**

Test Tracks

I play electric bass and drums, and I focus my listening on the type of rock and folk music that I like to play (or would like to be able to play!).

1. Steely Dan, *Two Against Nature*, Giant 9 24719-2, CD. "Negative Girl" starts with cymbals over deep electric bass, good for quick A-B comparisons. The percussion is especially well recorded, and the bass is very clean. This entire album is very well crafted and precise, studio recording at its best. Can you hear the snare drum offbeat roils on "Gaslighting Abbie"?

2. Bela Fleck and the Flecktones, *Greatest Hits of the 20th Century*, Warner Bros. 9 47301-2 CD. "Flight of the Cosmic Hippo" is a well-known deep-bass test. This group features virtuoso musicianship, especially leader Bela Fleck, who plays synthesizer/banjo, and Victor Wooten, who is the premier electric bass player in the business. *Outbound*, their latest album (Columbia CK 61418, HDCD), is well on its way to becoming a new reference for me even though I don't have an HDCD player.

3. Mouth Music, *Mouth Music*, Rykodisc RCD 10196 CD. Hard to describe—electronic Gaelic music?—this album combines good vocal quality, traditional Scottish songs, very sophisticated percussion, and pinpoint imaging. "Chi Mi Na Morbheanna," "Bratach Bana," and "Seinn Oi" are all recommended.

4. Suzanne Vega, *99.9F*, A&M 31454 0005 2 CD. Female vocals at their best in a very aggressive studio mix: "If You Were In My Movie," "In Liverpool," "As Girls Go."

5. Mary Chapin Carpenter, *Stones In The Road*, Columbia CK 64327 CD. More good female vocals in a calmer setting: "John Doe No. 24," "Why Walk When You Can Fly." Unlike the other albums listed here, the production on this one is a bit uneven, with some tracks much better than others.

6. Modest Mussorgsky, *Pictures At An Exhibition*, transcribed and performed by Jean Gillou, Dorian DOR-90117 CD. Another well-known bass test. By the middle of the second track you will know whether your system has real bass or just produces general rumbling. There is not any rumbling in the recording—they are all identifiable notes. In addition to the loud and spectacular parts, there is also some very low bass at low volume levels, even harder for most systems to reproduce well. I also find tracks 8 and 11 to be generally useful.

7. Igor Stravinsky, *Le Sacre du Printemps*, Antal Dorati and the Detroit Symphony Orchestra, London 400 084-2 CD. I also play tympani and other percussion in a local play-for-fun orchestra, and I know what orchestral instruments sound like. In general, orchestral recordings are disappointing, especially compared to good studio work for popular music. While there are many fine string quartet recordings, massed orchestral strings often have a fuzz or static sound layered over them. Also, recorded tympani and bass drums almost never sound anything like real life. This disk is pretty good. The instruments sound right, and the imaging is quite accurate, as well.

I don't offer any jazz recordings, since I don't know anything about jazz. I look forward to hearing more about good jazz and orchestral recordings from other readers! ❖

Bruce Bender
Wilder, Vt.