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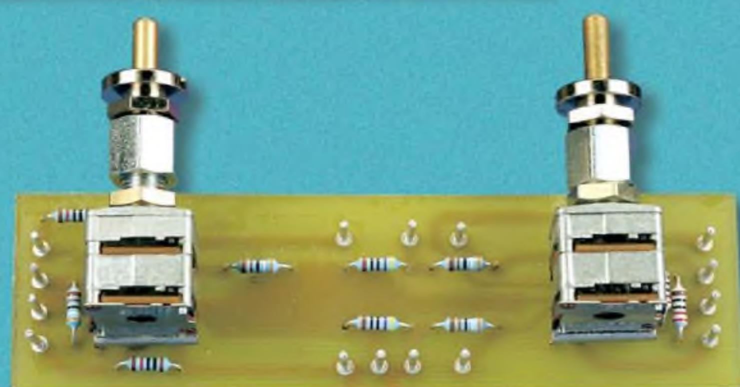

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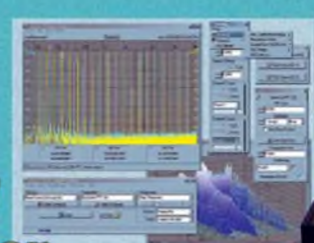

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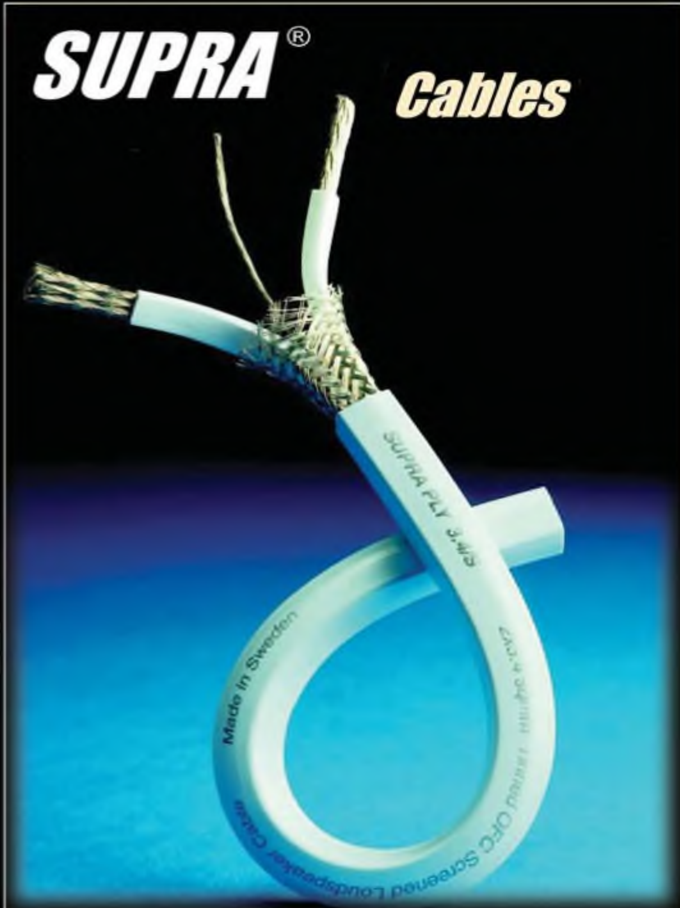
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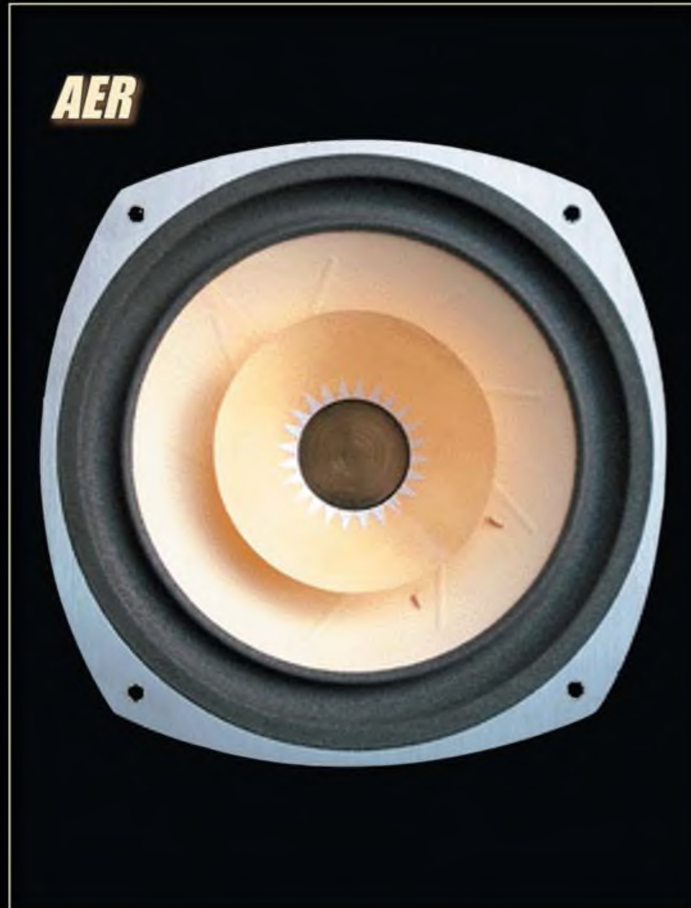
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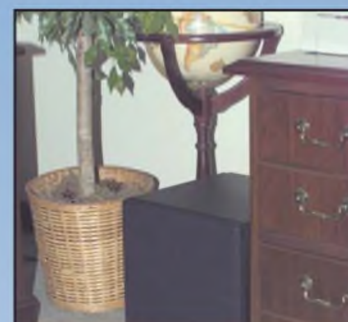

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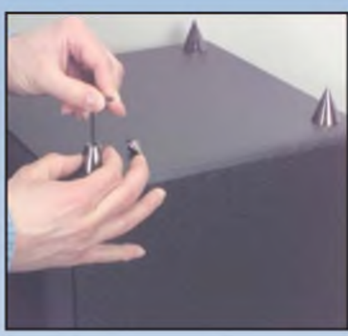

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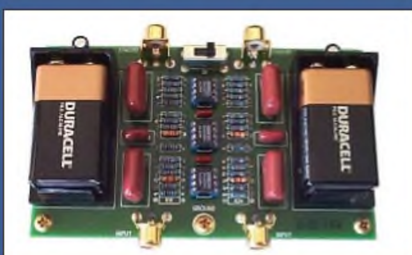

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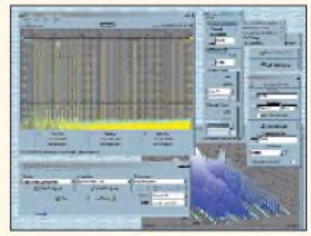

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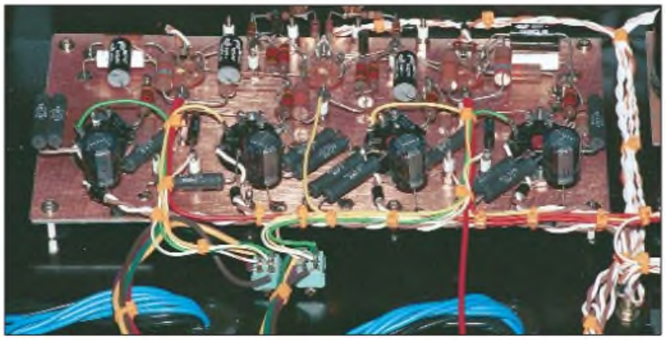

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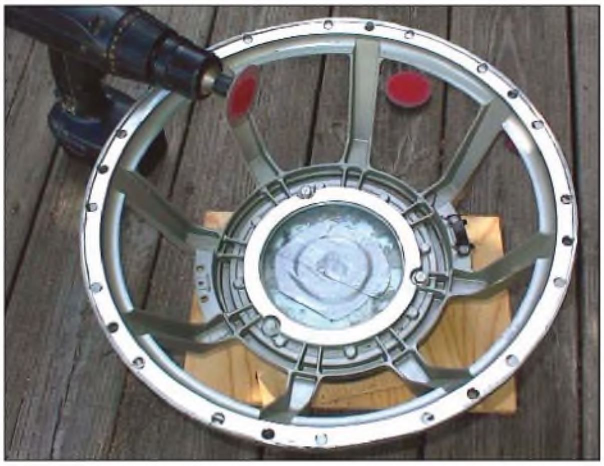

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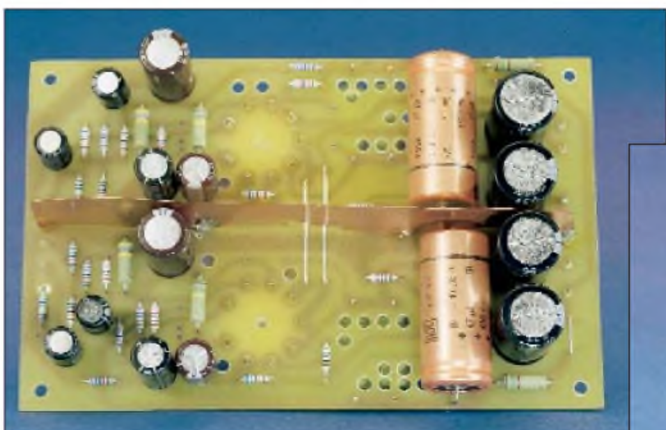

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.

JOHN STUART MILL

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Editorial

On Turning 32

Realizing that this enterprise will have just recently filled out its thirty-second year of existence, it seems appropriate to pause for a moment or two to reflect on that fact, and on what may be the reasons why we are still here. If I am honest, I must say that in 1969 I really had no dreams of being a publisher—or even an editor. No imagining of a magazine empire danced in my head at night, nor a wish that I might make a lot of money.

Only two passions lay behind the decision to scrounge for articles from the European magazines, look for mailing lists of audiophiles, and take money out of savings for mailing a prospectus to almost 100,000 names offering a new magazine about the audio craft.

The first passion was music itself. Many people from all manner of backgrounds have written attempting to describe the deep human wells which make music a universal human appetite. I have yet to read one which to me encompasses all of it. From plainchant to hip hop, the love of music animates something in nearly every human on the globe.

For as long as I have been listening to music—live and reproduced—I find my appetite for it only grows greater and more persistent as the years pass. The sense of adventure in hearing something new, or some new performance of something I may have heard dozens of times, never pales. A broadcast of something I do not recognize sends me scurrying for paper and pencil to note the composer's and the work's name—or possibly the conductor and performers.

Undoubtedly this passion occupies a place in my affections which was and remains the prime motive for any magazine or book I have undertaken to publish as a project of this company.

The second motive is simply technical craftsmanship in the service of our human delight in making a functional device. What are the elements in this strange propensity with which many of us seem to be afflicted?

Curiosity is first. The lure of a circuit diagram is a promise that if we can gather the parts, design the layout, craft the chassis, and put the whole thing together successfully, and then listen to it—we experience an absolutely unique human sensation for which I have never found a proper name. A large part of humanity is pleased to be mildly amused at this "do-it-yourself" affliction. A luxury only those who have never experienced it can indulge.

Aesthetic satisfaction is a high-falutin term for a pleasure which may be part of the craft of electronic construction. Some are more skilled than others in the command of a visual imagination which goes to design a project which turns out to be beautiful. There is a self-control and skill that is not widely dispersed to humans, which is a needed ingredient for such a result. Some crafted articles are more beautiful than others. To the craftsman, of course, the project result is beautiful. In general, however, the favored few manage a magic, which nearly everyone recognizes. Building a device with its own integral grace—even if only industrial grade—has a joy and satisfaction which makes the long effort worthwhile.

Materials intimacy is, although almost never acknowledged, a vital ingredient in building a project. To the devotee, the physical characteristics of the parts which make up the device have their own individual appeal, wordless and not quite ever acknowledged though it may be. A smoothly potted transformer; a heavy, authoritative choke; stainless steel screws; and hex nuts with captured lock washers are all somehow satisfying in ways that have little to do with their function. Elegant and colorful resistor or glass diode bodies are more like jewelry than mere components having a task to do in the circuit's operation.

The hardware's variety, including connectors, intricate switches, or potentiometers with satisfyingly smooth action, and heatsinks, or tube shields, or dozens of other pieces of the puzzle which go together in orderly arrangement, each has its unique satisfaction. All of these have a place in the whole, following the designer's map of functionality. They are all deeply satisfying to the builder, working patiently through the crafting of something new to go into the system.

The best, and most heart-stopping, moment comes when the project is completed. Tests have been done, parameters measured, adjustments completed. Then we hesitate, and do the checkout yet again. It is time for triumph—or disaster, shouts or smoke.

At first, of course, it sounds wonderful. The best so far. In time, reality sets in and we can admit to ourselves that it may only be different. Sometimes it is closer to what we hear live, sometimes not. This does not diminish the pleasures of the journey. Some destinations fulfill promise, some do not.

These two passions, and these alone, have been my engine of commitment to the Audio Amateur enterprise. Whatever I have done was always to forward the spread of those two satisfactions. Nothing else. Size of circulation or income or reputation have never been more than ancillary to the primary purpose of providing the means of helping people achieve better and more satisfying reproduction of music.—**E.T.D.** ❖

The Zen Amp Variations
Part 2: The Penultimate Zen's Current Source

Determining how much power to apply to our simple Zen amp.

By Nelson Pass

Welcome back to the Zen Amp Variations. This is Part 2 of many parts in which we explore some of the ways to make a very simple audio amplifier. In this and Parts 3 and 4, we will embellish upon the original Zen amplifier circuit, improving the performance and producing the Penultimate Zen amp.

INTRO

You may recall that the Zen amp is a single MOSFET transistor operated in what is known as common source mode, in which the input signal is fed to the Gate pin, the Source pin is grounded, and we take the output signal off the Drain. In order to get this arrangement to work, we must provide this transistor with a current source, which is just what its name implies. It is a source of current that provides the power for the gain device. In the case of the Zen amp, the current source acts as a mediator between the positive voltage supply and the gain transistor, feeding the right amount of current into the circuit to provide the optimum conditions for the device.

You can make the current source as simple as a resistor (or a light bulb), but it is generally advantageous to make it out of something more complex, because using a resistor results in about 8% or so efficiency, and since this is a power circuit, wasting this much power becomes costly and is socially incorrect. Just ask anyone who has built the Son of Zen, and they'll tell you that burning 600W to get 50W output is a bit over the edge.

In this article we will recap the operation of the original current source for the Zen, introduce an alternative current source, and then pull a trick out of the hat for a new source of current with a negative and occasionally imaginary source impedance.

THE ORIGINAL SOURCE OF CURRENT

Figure 1 shows a version of the original Zen amp. It is not exactly identical to the original circuit; I played around with it to give continuity to this article's progress, but it is close enough.

The actual gain stage portion of the circuit is Q1 and the components attached to it on the lower half of the circuit. Q2 and the components in the upper half of the circuit make up the current source. If you wish to improve your understanding of the lower portion of the circuit, I refer you to the original project articles, "The Zen Amplifier," *Audio Amateur* 2/94 and 3/94, on www.passdiy.com.

This type of current source is known as a constant-current source. Ideally, a constant-current source will deliver an exact, unwavering amount of current out of a connection (in this case the Drain of Q2), regardless of what is at-

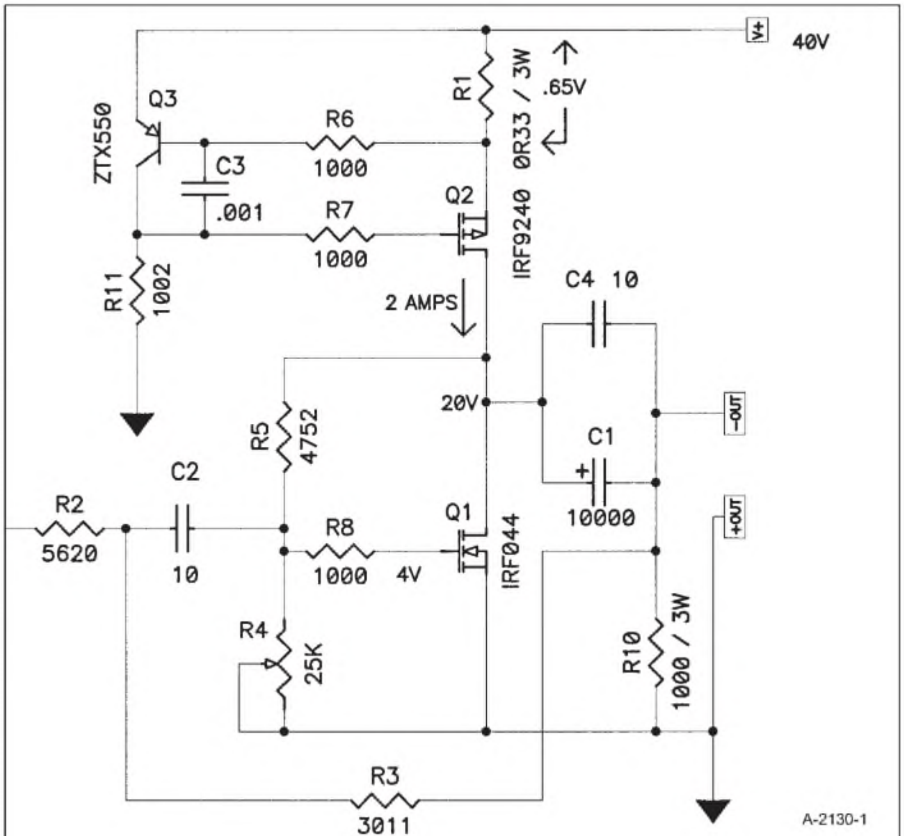

FIGURE 1: The original Zen constant-current source. Resistor values are three-digit and multiplier at .25W unless noted; capacitors are in microfarads unless noted. Electrolytics are 50V.


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tached to the connection or what the voltage conditions at that node might be. If you leave this point unattached to some circuitry, an ideal constant-current source will emit a small lightning bolt that will travel until it connects to something. The Zen amp constant-current source is not so ideal, but perhaps that is just as well.

The way this current source works is fairly simple. P-type MOSFET Q2 will conduct current when its Gate pin becomes negatively charged relative to its Source pin. An easy way to make that happen is to use R11 to pull the Gate voltage toward ground, and with the Source pin looking at about +40V, the MOSFET will begin conducting most enthusiastically.

To temper the enthusiasm of this conduction, we install the remaining parts of the current source: R1, Q3, R6, C3, and R7. When current begins flowing through the MOSFET, voltage is developed across .33Ω resistor R1, which will equal .33V for each amp of current flow. Q3, a PNP transistor, is set up so that it sees this voltage across R1. Bipolar transistors have a characteristic by which they will begin to conduct when their base to emitter voltage approaches .66V or so; and when the voltage across R1 reaches about .66V, Q3 begins to conduct.

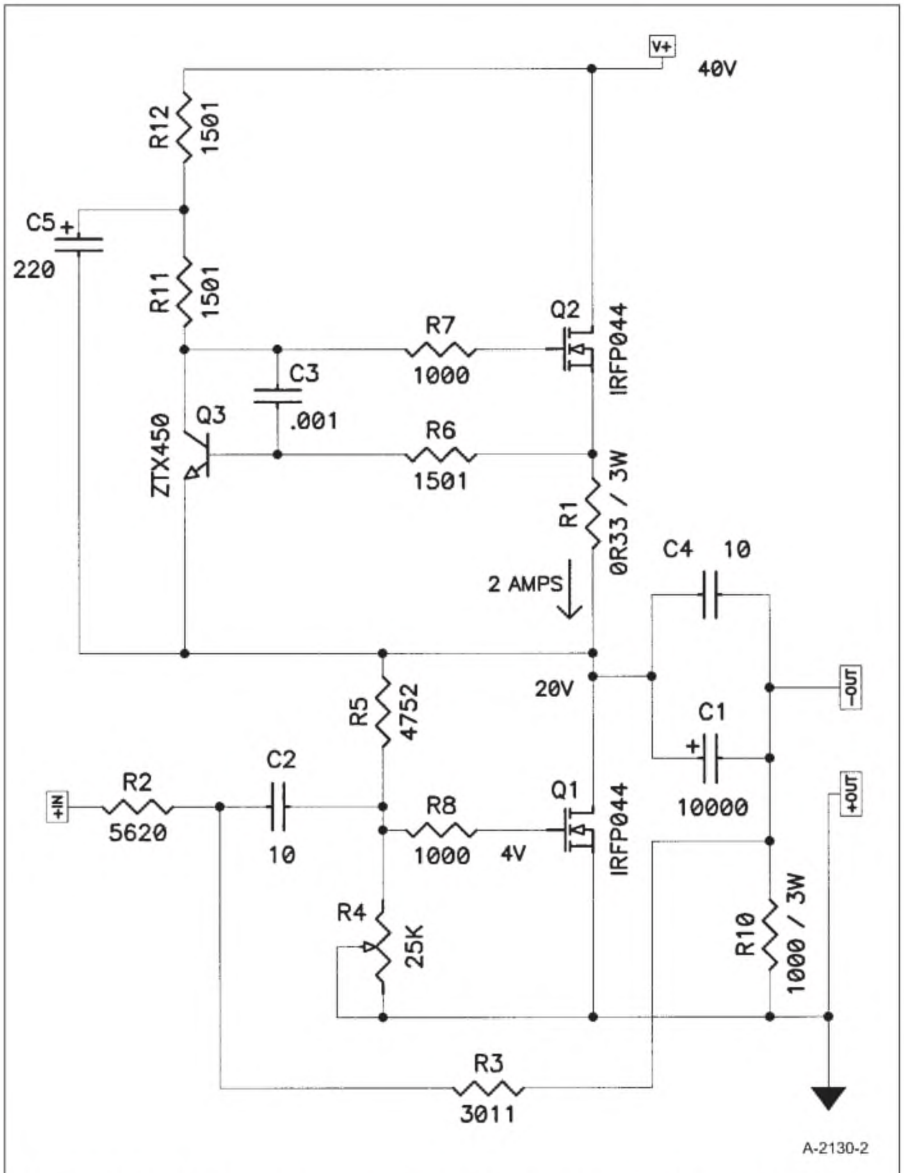


FIGURE 2: Modified constant-current source. Resistor values are three-digit and multiplier at .25W unless noted; capacitors are in microfarads unless noted. Electrolytics are 50V.

By comparison, MOSFETs begin conducting between 2 and 5V, and are quite variable between types and parts, which is why we use a bipolar device here. But I digress.

When Q3 begins to conduct, it limits the Gate voltage of Q2, and this forms a little feedback mechanism, which limits the voltage across R1 at about .66V. R6, R7, and C3 are there to help make this arrangement stable and reliable, and we could possibly omit these, but we won't.

If the voltage across R1 is a stable and constant .66V, then the current through both R1 and Q2 is a stable and constant 2A, and so we have a 2A stable and constant-current source. For Drain voltages reasonably within the values of the power supply provided, the current will be fairly constant and good enough for this amplifier.

AN ALTERNATIVE APPROACH

There is more than one way to skin a current source, and *Fig. 2* shows an alternative circuit that performs the same function but with an N-channel MOSFET instead of a P-channel type. In this circuit, the corresponding parts retain the same reference numbers as *Fig. 1*, with Q2 becoming an N-channel part, Q3 becoming an NPN part, and R11 changing value from 10kΩ to 1.5kΩ.

The function of this circuit is identical to *Fig. 1*, except that it has been turned upside down. The Gate of Q2 now must go positive relative to the Source pin to achieve conduction, and so on. To keep the current through R11 fairly constant—as it was with the previous version—we have added R12 and C5, which "bootstrap" the AC voltage at the junction of R11 and R12 so that the AC voltage variation seen across R11 is low.

The performance of this circuit is virtually identical to that of *Fig. 1*, and is documented in *Figs. 3–5*. One advantage to the circuit of *Fig. 2* is that it uses the same device as the gain transistor, both being the cheaper and more easily obtained N-channel types. The more crucial advantage to this innovation is that you can further develop it into a current variable current source, as in the Pass Labs Aleph amplifiers.

Before we do that, let's make note of

the circuit performance in *Fig. 2. Figure 3* shows the harmonic distortion curve versus output power into an 8Ω load, which rises smoothly from about .1% at .1W, to about 2% at 10W. This is mostly second-order harmonic distortion until you get above 10W, and by about 18W RMS you are clipping the output with peaks at 36W. This is more power than we got out of previous editions of the Zen amp by virtue of greater power supply voltage at 40V. Into 4Ω, the amplifier clips at about 9W RMS.

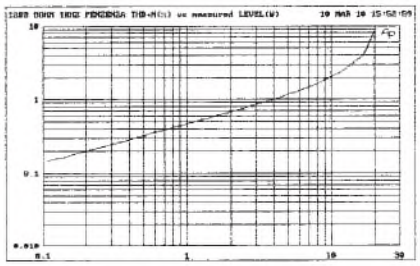


FIGURE 3: Constant-current source distortion versus power. A-2130-3

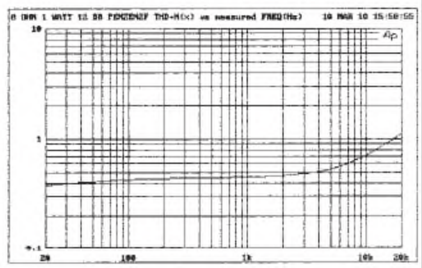


FIGURE 4: Constant-current source distortion versus frequency. A-2130-4

Figure 4 shows distortion versus frequency into the same 8Ω at 1W levels. Note the increase at high frequencies with a knee in the curve at about 6kHz. The increase at high frequencies is due largely to nonlinearities in the input capacitance of power MOSFETs and is an unwelcome feature that you must live with in this device.

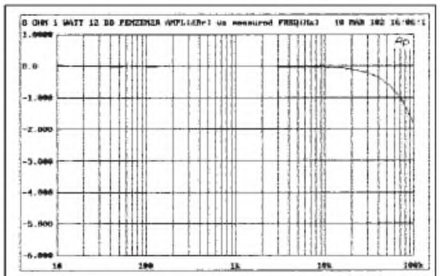


FIGURE 5: Constant-current source fequency response. A-2130-5

You can minimize this effect by minimizing the impedance of the feedback loop in the amplifier, resistors R2 and R3. Unfortunately, this also minimizes the input impedance of the amplifier, making it more difficult to drive. I will be addressing this problem in Part 4 and other parts of this series.

Figure 5 shows the frequency response curve of this amplifier, which is absolutely flat to 10Hz and down 2dB at 100kHz. The damping factor of this amplifier is about eight against 8Ω, giving it a 1Ω output impedance, typical of tube performance.

THE ALEPH CURRENT SOURCE: THE ALTERED ALTERNATIVE

In 1991 I authored U.S. patent # 5,710,522, which detailed the operation of an active current source that was constant for DC characteristics but variable at AC frequencies and that could track an arbitrary percentage of output current to a load, relieving the active single-ended Class-A gain device of some of the work. As a practical matter for audio, the best figure for that percentage is about 50%, which allows for pure Class-A operation to twice the current, boosting efficiency to an ideal limit of 50% over the 25% expected from a constant-current source.

Instead of the infinite impedance represented by an ideal constant-current source, this new variation has a negative impedance equal to twice the load impedance. If the load is 8Ω, the current source will have a −16Ω impedance, and the active portion of the amplifier will think it is driving 16Ω. The reactive portion of the loudspeaker is similarly mirrored in this manner so that it accurately produces a negative ghost of the load. Think of Sigourney Weaver with the mechanized loader in *Aliens*.

This circuit, implemented in the Pass Labs Aleph series of amplifiers, has become known as the Aleph current source. It is fairly painless to modify the current source of *Fig. 2* into this circuit, and the result is displayed in *Fig. 6*.

New components are introduced: R13 through 15 and C6. R13 and 14 are two 3W .47Ω metal film resistors paralleled to form a 6W .235Ω resistor. They see the output current to the load, and sensing voltage appears across them which is proportional to this output current. R15 and C6 communicate this to the feedback junction at the base of controlling transistor Q3, and the current through Q2 varies to support this output current.

The percentage of current contributed by Q2 is a function of the values of R13, 14, and 15 along with the other values of the current source, and in this circuit are set for best-sounding performance, which is close to 50%. As a practical matter, you would adjust R15 to vary this value.

As with the other Zen projects, there is considerable leeway in the selection of values, and if you find that you wish to vary the values of R1, 13, and 14, for example, you will end up adjusting the value of R15 to return to the 50% figure. Or maybe not; perhaps some other percentage will sound better to you.

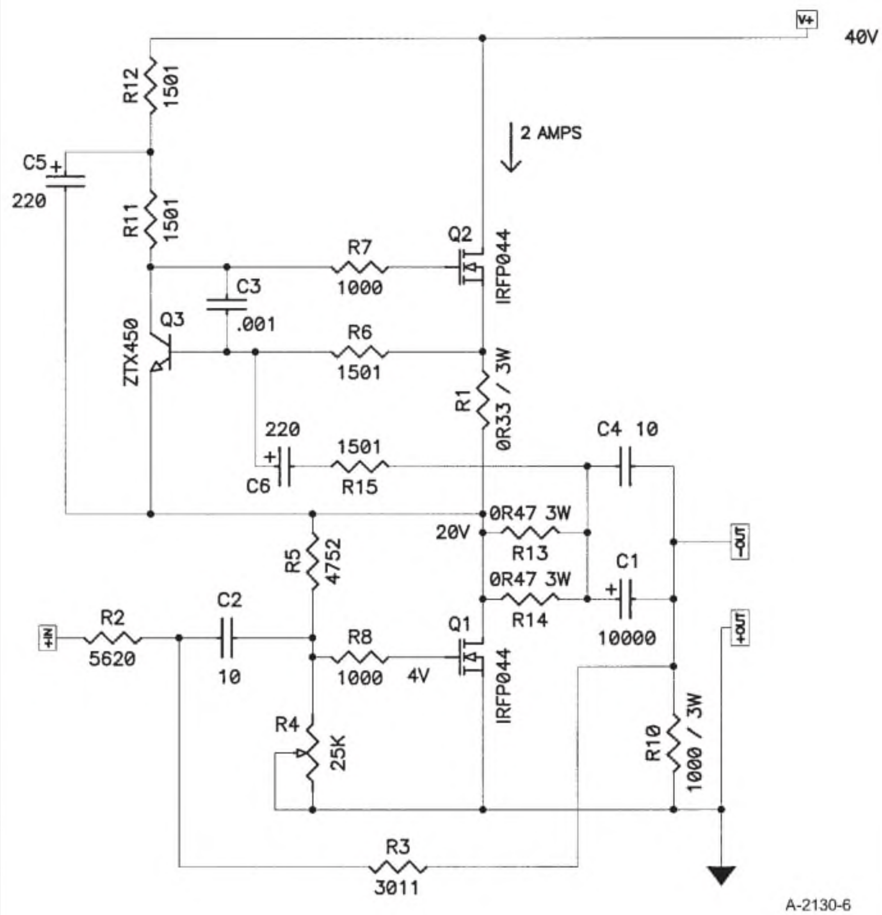


FIGURE 6: Aleph current source. Resistor values are three-digit and multiplier at .25W unless noted; capacitors are in microfarads unless noted. Electrolytics are 50V.

In any case, to achieve this 50%, you would need to adjust R15 so that the AC current through R1 is one half the current going through the parallel R13 and 14. In the case of the values shown, that means that when the amplifier is driving a load, the AC voltage across R1 is 70% of the voltage across the combination of R13 and 14. You can use an AC voltmeter at 60Hz to check this figure if you need to confirm proper operation of the circuit. Decreasing R15 increases the percentage, and vice versa. If you want a switchable variable/constant-current source, install a switch to break the R15 circuit.

Figure 7 shows a comparison of the distortion versus power of the circuits of *Fig. 2* and *Fig. 6*. It's a pretty straightforward result: the distortion drops by a factor of about ten.

Figure 8 shows the same comparison at 4Ω. Again, a factor of ten reduction until about 9W, where the older circuit clips and the newer circuit goes on to hit 1% distortion at 35W.

Figure 9 shows the distortion versus frequency for the two circuits. Alas, the dramatic improvement doesn't hold up as well at high frequencies, which I will be addressing in Part 4. The damping factor of the newer circuit is approximately 40, for an output impedance of about .2Ω, five times better than the older circuit.

OUTRO

Well, there you have it, the first major improvement of the original Zen amp in eight years. It sounds about as good as it looks. The bottom end is a lot tighter, there is greater clarity in the midband, and there is more power and dynamic range. You can operate this one balanced for about 70W into 8Ω with even lower distortion. If you have

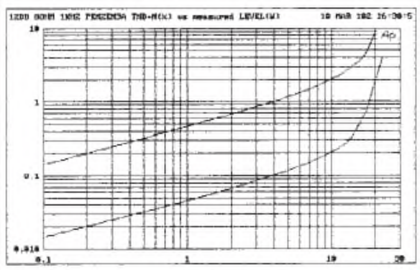


FIGURE 7: Constant versus variable current source distortion versus power into 8Ω. A-2130-7

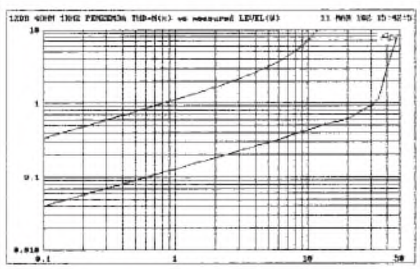


FIGURE 8: Constant versus variable current source distortion versus power into 4Ω. A-2130-8

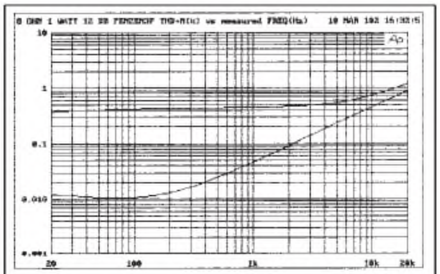
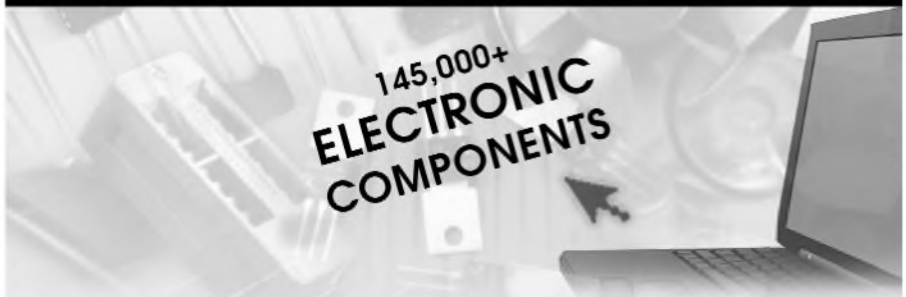


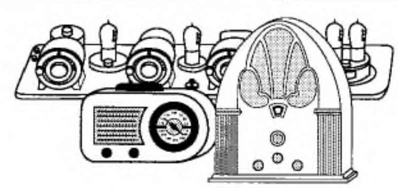
FIGURE 9: Constant versus variable current source distortion versus frequency. A-2130-9

already built the Zen, you can easily adapt it to the newer circuit, and it will perform well with the original 32V supply (although with somewhat less power).

In Part 3 we will build an active power-supply regulator for the Zen, and in Part 4 we will raise the input impedance to 47kΩ and clean up some of the high-frequency distortion. The result is fairly spectacular with minimal additional complexity and cost. And finally, the Penultimate Zen will get a PC board, and www.passdiy.com will offer a partial kit.

The Aleph current source is still covered by U.S. patent # 5,710,522, but as a matter of policy we do not concern ourselves with DIY efforts. All others can send checks to: nelson@passlabs.com. ❖





A Marantz 8B Replica

This article was originally published in the January 2000 issue of Japan's premier high-end audio magazine, *MJ Audio Technology*. **By Satoru Kobayashi**

I reproduced the Marantz 8B replica amplifier (*Photo 1*), which was a well-known amplifier about three decades ago. Mr. Uesugi originated the Marantz 8B-replica amplifier project in Japan in 1968, using Matsushita 6CA7 tubes and LUX OY-36 transformers. I wanted to build and replicate the same one using the Plitron toroidal transformer and a Svetlana-made vacuum tube, both of which are well known as state-of-the-art technology components in these days. I hoped to achieve a better result and better performance than that of the old replica I built in the past.

CIRCUIT

First of all, I referred to the original article by Mr. Uesugi in the January 1968 issue of *Radio Gijutsu* magazine[1] and the original circuit diagram of Marantz 8B[2]. What I learned from these were that the Marantz 8B consists of a basic resistor-loaded voltage driver by a triode-connected 6BH6, a long-tail phase splitter by 6CG7/6FQ7, and a pair of push-pull EL34s for the amplifier stage. In this configuration, several phase-compensating capacitors and resistors are installed to optimize the frequency response.

Mr. Uesugi's circuit provides:

- a 5pF feedback capacitor between the plate and grid of 6BH6
- a parallel-connected 4.7kΩ and 0.002µF over 178Ω of the cathode resistor of 6BH6
- a 33pF capacitor grounded at the plate of the upper 6CG7 of the phase-splitter stage
- a pair of 1.5pF feedback capacitors, cross-coupled between the EL34 plates and 6CG7 grids
- a parallel-connected capacitor over a NFB resistor

These components are supposed to work effectively in this particular old model, although the state-of-the-art technology toroidal transformer might not need these optional parts, because the transformer features a wider frequency and a better phase response than the conventional E-I cored transformer. Thus, I decided not to implement these parts at the beginning. However, if the performance characteristics after the measurement were not good enough, then I might fix these optional parts into the circuit, depending upon the result.

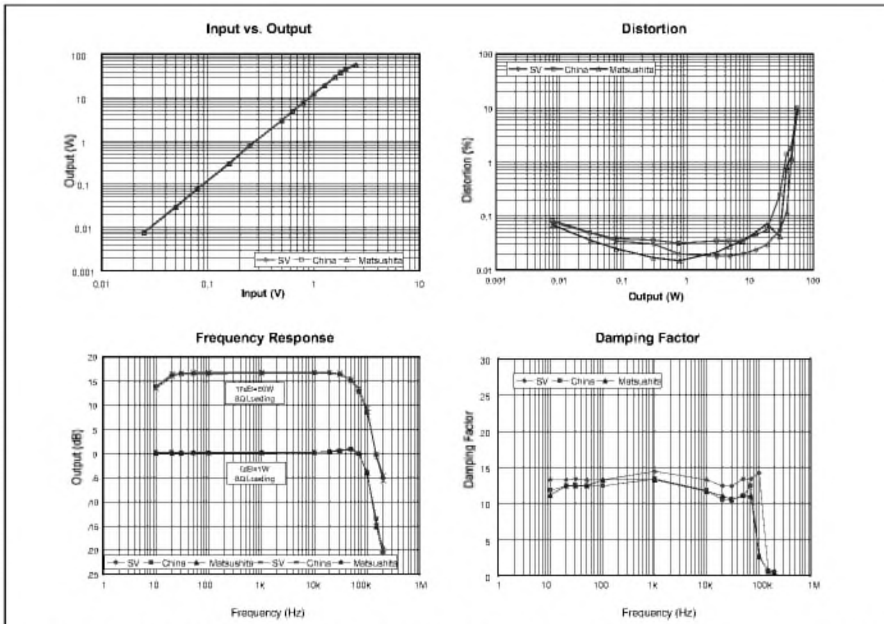


FIGURE 1: Characteristics of original replica amplifier by Uesugi. G-1817-1

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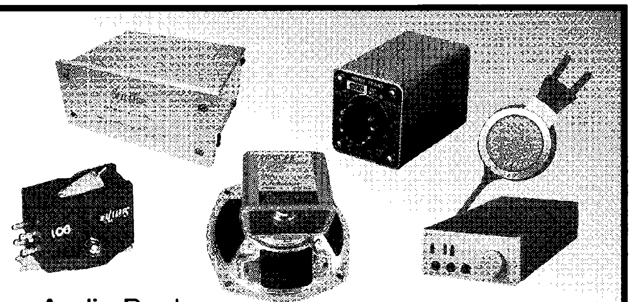

■Cartridge

Model	Price (US$)
Denon DL-102 (MONO)	150
Denon DL-103 (STEREO)	200
Denon DL-103R (STEREO)	250
Denon DL-103 PRO (STEREO)	350
Shelter Model 501 II (CROWN JEWEL REFERENCE)	650
Shelter Model 901 (CROWN JEWEL SE)	1,300

Postage (Air Economy)

Area	Countries
Area I $18	China, Korea, Hong Kong, Taiwan
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Area III $27	North America, Oceania, Europe
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Attractive Tube Amps Vol. 1&2	(Isamu Asano)	30 each
Top-Sounding Vintage Power Tubes	(Stereo Sound)	30
Output Trans of The World	(Stereo Sound)	30
The Joy of Vintage Tube Amps	(Tadaatsu Atarashi)	30
MJ Selected 300B Amps	(MJ)	30
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SE Amps by Transmitting Tubes	(Kouichi Shishido)	50
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20TH CENTURY OF AUDIO	(Stereo Sound)	30
The Remembrance of Sound Post	(Susumu Sakuma)	30
Tube Amp Craft Guide	(MJ)	30
FOSTEX CRAFT HAND BOOK	(FOSTEX)	20

■MC STEP UP TRANS

Model	Specifications			Price (US$)	Postage**
	Pri.Imp(Ω)	Sec.Imp(kΩ)	Response		
Shelter Model 411	3~15	47	20Hz~50kHz	980	Area I $25
Jensen JE-34K-DX	3	47	20Hz~20kHz	550	Area II $30
Peerless 4722	38	50	20Hz~20kHz	300	Area III $40 Area IV $50

**Air Economy

■STAX

Model	Price(US $)
OMEGA II System(SR-007+SRM-007t)	Ask
SRS-5050 System W MK II	Ask
SRS-4040 Signature System II	Ask
SRS-3030 Classic System II	Ask
SRS-2020 Basic System II	Ask
SR-001 MK2(S-001 MK II +SRM-001)	Ask

■Speaker

Model	Specifications					Price* (US $)	Postage** (US $)			
	D (cm)	Ω	Response	db	w		I	II	III	IV
Fostex FE208Σ	20	8	45Hz~20kHz	96.5	100	296	62	74	120	156
Fostex FE168Σ	16	8	60Hz~20kHz	94	80	236	42	50	73	98

*Price is for a pair **Air Economy

■TANGO TRANS (ISO) 〔40models are available now〕

Model	Specifications				Price (US $)	Postage** (US $)			
	W	Pri.Imp(kΩ)	Freq Response	Application		I	II	III	IV
XE-20S (SE OPT)	20	2.5, 3.5, 5	20Hz~90kHz	300B,50,2A3	396	47	56	84	113
U-808 (SE OPT)	25	2, 2.5, 3.5, 5	20Hz~65kHz	6L6,50,2A3	242	42	50	73	98
XE-60-5 (PP OPT)	60	5	4Hz~80kHz	300B,KT-88,EL34	620	62	74	115	156
FX-40-5 (PP OPT)	40	5	4Hz~80kHz	2A3,EL34,6L6	320	47	56	84	113
FC-30-3.5S (SE OPT) 〔XE-60-3.5S〕	30	3.5	20Hz~100kHz	300B,50,PX-25	620	62	74	115	156
FC-30-10S (SE OPT) 〔XE-60-10SNF〕	30	10	30Hz~50kHz	211,845	620	62	74	115	156
X-10SF 〔X-10S〕	40	10W/SG Tap	20Hz~55kHz	211,845	1160	90	110	180	251
NC-14 (Interstage)	—	[1+1 : 1+1]5	25Hz~40kHz	[30mA] 6V6 (T)	264	30	40	50	70
NC-16 (Interstage)	—	[1+1 : 2+2]7	25Hz~20kHz	[15mA] 6SN7	264	30	40	50	70
NC-20F (NC-20) (Interstage)	—	[1 : 1]5	18Hz~80kHz	[30mA] 6V6 (T)	640	42	50	73	98

Price is for a Pair

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■TAMURA TRANS (All models are available)

F-7002 (Permalloy)	10	3.5	15Hz~50kHz	300B,50	740	60	70	110	145
F-7003 (Permalloy)	10	5	15Hz~50kHz	300B,50	760	60	70	110	145
F-2013	40	10	20Hz~50kHz	211,242	730	70	84	133	181
F-5002 (Amorphous)	8	3	10Hz~100kHz	300B,2A3	1276	65	80	120	160

Price is for a Pair

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Figure 1 shows the characteristics of the original replica amplifier Marantz 8B designed by Mr. Uesugi and built by myself. I overhauled the amplifier a few years ago by replacing all capacitors with new ones, and now it works fine. As described previously, the frequency compensation was applied sufficiently, so the well-balanced performance characteristics such as frequency response curve and distortion characteristic curve are observed working nicely with the well-known output transformer of the LUX OY-36 series. I believe this was due to Mr. Uesugi's excellent design skill.

PHOTO 1: The completed unit (rear view).

TABLE 1
COMPARISON BETWEEN MARANTZ 8B AND REPLICA

ITEM	MARANTZ 8B	REPLICA
Subsonic filter	Yes	N/A
1st stage tube	6BH6 triode connected	1/2-12AU7
Circuit type	Resistor loading	
Load resistor	220kΩ	51kΩ
Plate voltage	90V	128V
Plate to grid compensation	3.9pF	N/A
Phase splitter	Long tail	
Tube	6CG7/6FQ7	6FQ7
Load resistor-upper side	15kΩ	18kΩ
Load resistor-lower side	18kΩ	20kΩ
AC balancing resistor	5kΩ VR	N/A
Common cathode resistor	13kΩ	
Grid pin compensation upper	1.5pF	N/A
Grid pin compensation lower	18kΩ, 1.5pF	N/A
Plate pin compensation lower	33pF	N/A
Plate voltage (upper/lower)	304/340V	312/316V
Final stage grid bias	Fixed	Self
Cathode resistor	6.8Ω	667Ω
Grid bias circuit	Yes	N/A
Grid bias voltage	−36V	
Plate voltage	404V	465V
Final tube	6CA7/EL34	
Triode-UL switch	N/A	Yes
Power supply	Voltage doubler	
Flywheel capacitor	40µF × 2	330µF × 2
Circuit type	Choke input	
Capacitor value	40µF	200µF
Voltage driver power supply	Resistor	Zener diode
NFB	Yes	Switchable
NFB components	Resistors, capacitors	Resistors
NFB resistor value	1620Ω	6.8kΩ/2.4kΩ
Transformer tap for NFB	Yes	8Ω
Output transformer	E-I conventional	Toroidal
Secondary impedance	?	4.4kΩ
Miscellaneous	Bias checking meter	N/A

For my own designed replica, I modified the circuit a little bit with a PTP terminal board, which allows for easy wiring and maintenance. I modified the original circuit by adding a self-bias circuit for the final stage and removing the AC balancing potentiometer as the part of plate loading resistor, which eliminates the need for DC and AC adjustment. Thus, the circuit was re-born out of the original Marantz 8B and Mr. Uesugi's replica. *Table 1* shows the comparison between the original and my version.

BIAS CIRCUIT FOR THE FINAL STAGE

I chose the same setting for the final stage as that of the Marantz 8B. The operating condition of Marantz 8B is 1) approximately −35~36V of grid bias, 2) 450V DC plate voltage, and 3) approximately 50mA of idling current.

In the conversion from the fixed bias to the self-bias you must keep the same idle current of 50mA, so the cathode resistor value is 700Ω (=35V ÷ 50mA). The cathode resistor will consume 1.8W (=36V × 50mA). Most amplifiers with a self-bias circuit use a large size of wirewound resistor such as 20W or so.

I chose three 2kΩ, 3W metal-oxide film resistors, connected in parallel, forming a 667Ω, 9W resistor equivalent, because I was concerned about installing a large part over a small space on the PTP terminal board.

DEFINING LOAD IMPEDANCE

I chose a Plitron toroidal transformer, PAT-4004, with a primary impedance of 2.4kΩ under 5Ω of the secondary output impedance. Since the winding ratio of this transformer is 23.48:1, the primary impedance would be about 4.4kΩ (=8 × 23.48^2) with an 8Ω output loading. This meets the requirement of the loading condition for an EL34/6CA7 push-pull amplifier, which is in the range between 3.5kΩ and 5kΩ.

TRIODE VERSUS ULTRALINEAR CONNECTION

To sample the sound difference between triode and UL connection, I added a switch to change between these two modes. I inserted a 500Ω resistor into the screen grid circuit of the final tubes.

Mr. Barbour, formerly at Svetlana, suggested its value via e-mail. It might be safer to protect the screen grid from the overcurrent drive of EL34, than the 100Ω usually seen in past examples. "The value of 500Ω will not degrade the sound," Mr. Barbour said.

The 500Ω resistor links to the 2-pole, 2-position switch to change the connection. The switch provides a locking position to avoid switching under operation, and was placed in the middle between a couple of output transformers over the chassis.

POWER SUPPLY

The original Marantz 8B uses a full-wave voltage doubler. To meet this setup, I found a Tango power transformer, MS-330D, which supplies 330mA DC after rectifying and fits perfectly into this amplifier. To make it better, the AC winding of MS-330D for the plate supply provides several 10V taps so that I could adjust B+ DC supply without a regulated power supply. I chose 175V AC out of the taps between 10V and 185V to get 450–470V DC after rectifying.

The rectifier consists of a couple of 1000V PIV-series-connected Schottky diodes, so that the operating margin has enough room. The ripple filtering was formed by using a choke input circuit with Tango CH-5-300D (5H, 300mA) with a couple of 100µF electrolytic capacitors, generating 460V DC with a low ripple voltage as a result.

DEFINING B+ VOLTAGE OF THE PHASE SPLITTER

The original Marantz 8B uses a 408V DC power supply at the long-tail phase-splitter stage. To meet this, I used a cascaded zener diode of 3W, 47V and got approximately 413V DC (=460-47V). The first stage supply voltage resulted in approximately 250V DC with a series-connected 3W 150V out of the 413V DC supply point.

PHASE SPLITTER

I have copied the resistor value from the original. The difference is the elimination of the AC balancing potentiometer. So I did a simulation with TubeCAD simulator, resulting in 18kΩ, 20kΩ for the plate resistor values, so there will be no need to adjust AC balance. I chose the 6FQ7, as in the original.

The first stage valve is a 12AU7, which was in my parts box. For some reason, it is difficult to find the 6BH6 in Japan these days.

After choosing the valves, I did another simulation using the TubeCAD software simulator. *Table 2* shows the result. The total gain became 100 (40dB), suggesting that the circuit might drive the final stage sufficiently without any NFB.

NFB FREQUENCY COMPENSATION

The Marantz 8B features a number of unique feedback components to obtain a flat frequency response curve. In general, the NFB resistor boosts the high-frequency response above 100kHz and the low-frequency below 50Hz. A compensation capacitor connected parallel to the NFB resistor suppresses its peaky response to make it flat over the entire frequency range between 10Hz and 20kHz.

Under these circumstances, I thought that even if I used NFB resistors and capacitors to get a flat and wider frequency response, such extra components would work inadequately

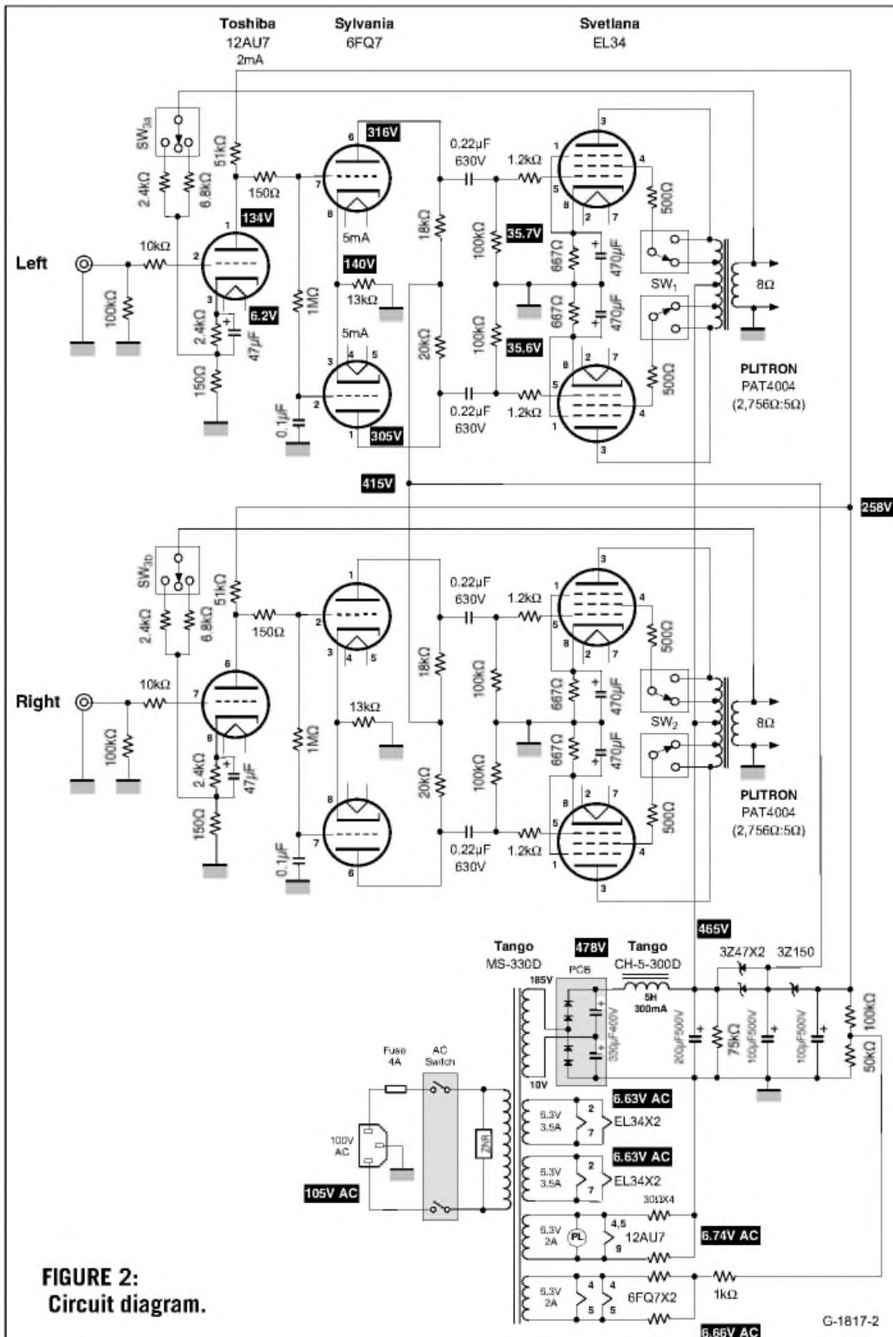


FIGURE 2: Circuit diagram.

with the state-of-the-art transformer and valves. I was extremely confident I would get a wide and flat frequency response with the Plitron toroidal transformers, because I was familiar with the characteristics of these new components through my past designs.

Besides, I expected that with this new component the wider frequency response would vary with NFB or non NFB circuit more than ever before. Thus I implemented a 2-pole, 3-position switch to select NFB levels of none (0), 6dB, and 12dB.

DRIVING FILAMENT

The filament of each valve is powered by AC and wired individually from the transformer's 6.3V AC windings so that the filament voltage to each of the valves is equal. To minimize a hum, the voltage driver circuit provides a 30Ω hum-balancing resistor. *Figure 2* is the full circuit diagram.

PARTS AND COLLECTION

I obtained a matched pair of Svetlana EL34s and a couple of Plitron transformers via the dealership of Tec-Sol Inc. in Hamamatsu, Japan. The Plitron toroidal transformer, designed by Mr. van der Veen in Holland, features a wider frequency response and a better

TABLE 2
SIMULATION RESULT

PARAMETER	INPUT STAGE	PHASE SPLITTER
Circuit	Resistor loading	Long tail
Tube	12AU7	6FQ7
Plate voltage	250V	400V
Input resistor	10kΩ	150Ω
Plate resistor	51kΩ	18kΩ/20kΩ
Cathode resistor	2kΩ	13kΩ
Output load impedance	330kΩ	100kΩ
Coupling capacitor	0.33µF	0.1µF
Plate current	2mA	10mA
Gain (dB)	14.48 (23.3dB)	6.91 (16.8dB)
Plate-cathode voltage	144V	180V
Grid voltage against the ground	0V	124V
Power supply rejection ratio	–17.8dB	–0.24dB
Maximum saturated input voltage	6.1V	5.74V
Maximum output voltage	–93/+93.5V	–42/+42V
Grid bias voltage	–6.46V	–6.08V
Input impedance	268kΩ	311kΩ
Output impedance	6.56kΩ	5.12kΩ
Lower cut-off frequency (–3dB)	1.43Hz	15.1Hz
Upper cut-off frequency (–3dB)	733kHz	>1MHz
Effective cathode resistor	3.24kΩ	—
Cathode bypass capacitor	4µF	—
Plate power consumption	288mW	900mW
Plate resistor power consumption	204mW	450mW
Total gain (100) = input stage (14.48)	× phase splitter (6.91)	
	40dB = 23.3dB	+ 16.68dB

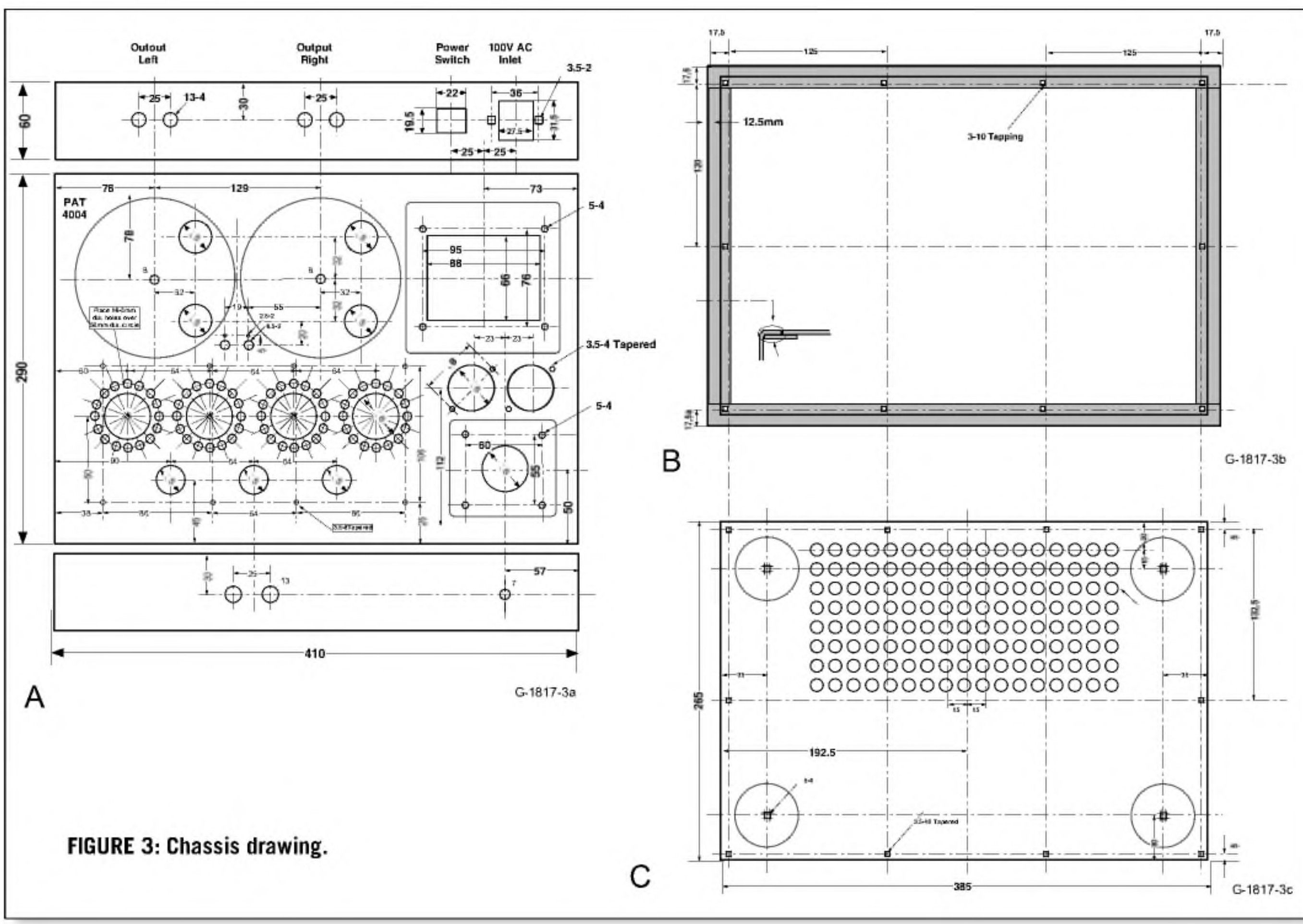


FIGURE 3: Chassis drawing.

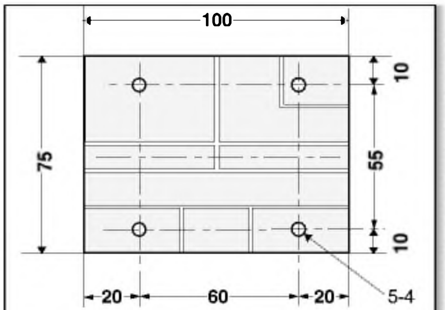


FIGURE 4: Power supply PCB drawing.
G-1817-4

phase characteristic than E-I cored transformers and provides output power of 70W. It is easily installed since it requires only three holes.

I chose a Tango power transformer, whose dimensions are compatible with the toroidal transformer diameter, making an easy allocation of these components on the chassis along with the Tango choke. The case (410 × 390 × 60mm) is made out of 1.6mm-thick steel, painted black so that all components are matched over the chassis, and custom-made by San-Ei Musen in Akihabara (*Fig. 3*).

The voltage driver valves, 12AU7 (Toshiba) and 6FQ7 (Sylvania), are from my parts box, saved for a long time. Other components such as resistors, capacitors, and semiconductors are from the dealership in Akihabara (Tokyo) and junk shops in San Jose, Calif. (*Table 3*).

ASSEMBLY

The geometry of the case and all components allocation was defined with ClarisDraw software on a Power Macintosh G3/267MHz. This saved a lot of time to fix the custom chassis design and parts placement. The power transformer and output transformers lined up nicely, since their diameters are mostly equal and their colors matched.

I did most of the internal wiring of electronic components on a self-made PCB board and a "PTP terminal board" from International Audio Group in Texas. This saved a lot of time and sim-

TABLE 3
PARTS LIST

AMPLIFIER COMPONENT	DESCRIPTION	QUANTITY
Final tube	EL34 matched pair (Svetlana)	2
Input stage	12AU7 (Toshiba)	1
Phase splitter	6FQ7 (Sylvania)	2
Output transformer	PAT-4004 Plitron	2
Resistor	100Ω ½W	2
Resistor	150Ω ½W	4
Resistor	500Ω 1W	4
Resistor	1.2kΩ 1W (can use 1kΩ)	4
Resistor	2kΩ 3W	12
Resistor	2.4kΩ ½W	4
Resistor	6.8kΩ ½W	2
Resistor	10kΩ ½W	2
Resistor	13kΩ 2W	4
Resistor	18kΩ 2W	2
Resistor	20kΩ 2W	2
Resistor	100kΩ ½W	2
Resistor	100kΩ 1W	4
Resistor	1MΩ ½W	2
Capacitor	0.1µF 400V ASC	2
Capacitor	0.22µF 630V Solen	4
Capacitor	220µF 16V Matsushita	2
Capacitor	470µF 63V Nichion	4
Switch	2-pole 3-position C&K Components	1
Switch	2-pole 2-position with Neutral lock ALCO	2
PTP terminal board	IAG made 250 × 120mm	1
POWER SUPPLY		
Power transformer	MS-330D Tango	1
Choke coil	CH-5-300D Tango	1
Diode	1000V IA RG4C Shin-Dengen or equivalent	4
Zener diode	3W 47V, 3Z47 Toshiba	2
Zener diode	3W 150V, 3Z150 Toshiba	1
Resistor	30Ω 2W Cement type resistor	4
Resistor	1kΩ 1W	1
Resistor	50kΩ 1W	1
Resistor	75kΩ 5W	1
Resistor	100kΩ 1W	1
Electrolytic capacitor	330µF 400V Nichicon	2
Electrolytic capacitor	100µF + 100µF 500V ELNA Cerafine	2
Pin terminals	Insulated	
Epoxy copper-clad board	100 × 75 × 1.6mm	1
CHASSIS		
Custom-made chassis	410 × 290 × 60mm 1.6mm San-Ei Musen, Akihabara	1
Socket	8 pin (QQQ made) for EL34	4
Socket	9 pin (QQQ made) for 12AU7, 6FQ7	3
RCA pin jack	Black, red Supertron	2
Power switch	125V 10A C&K components	1
AC power entry module	Mini fuse implemented (250V 4A)	1
Speaker terminal	Black, red San Ei Musen	2
Pilot lamp	6V LED with resistor SATO PARTS	1
Brass spacer	13mm 3mm	8
Brass spacer	30mm 3mm	2
Brass spacer	15mm 4mm	4
Metal feet 3mm volt/nut	IAG made	4
Hookup wire	Teflon insulated	
Pin terminals	Teflon insulated	

plifies maintenance. Because it eliminates a number of internal hook-up wires, it looks nice, too.

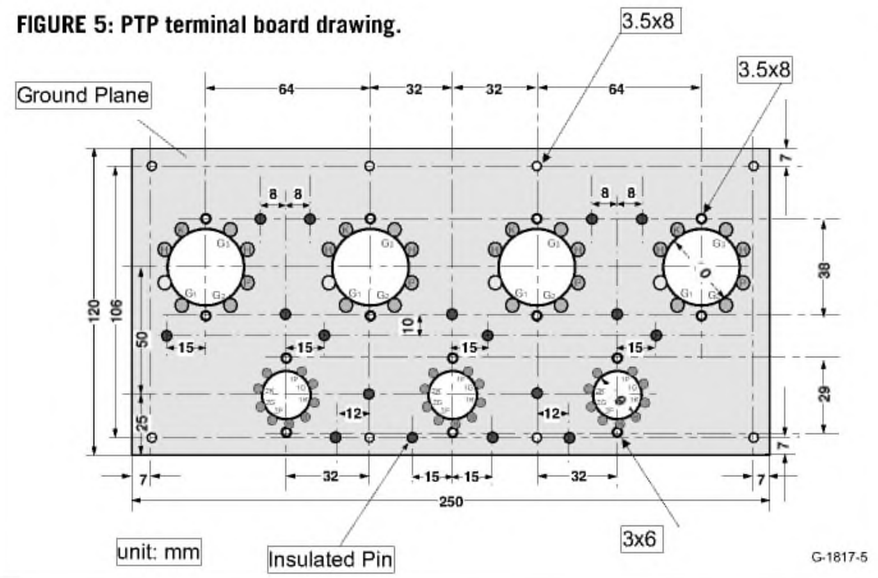
The power supply uses the voltage doubler, assembled on a 1.6mm-thick copper-clad epoxy board (10 × 7.5cm). The board is grounded via a brass spacer (15mm height) from the negative terminal on the PCB, using the screwholes of the choke coil. This scheme enables a sort of star-grounding to eliminate hum. Be sure to scrape away the paint at the contact point where the brass spacer meets the chassis to ensure solid grounding.

I made the PCB trace with a utility knife by peeling off isolating strips, after sliding a soldering iron over them for several seconds. Also, I mounted Teflon®-insulated pins on a PCB to make wiring simple and easy. I flux-coated the PCB to protect it, avoiding erosion. *Figure 4* shows a trace drawing.

The first stage and phase-splitter circuit are integrated on a PTP terminal board (*Figs. 5* and *6*), which offers a rigid structure on a 3.2mm-thick copper-clad epoxy board and a firm star-connect grounding. You can order the custom-designed pin alignment for the board from IAG by e-mail (hiag@n-link.com) by attaching a PowerPoint drawing file. I received my order in Tokyo via Express mail within ten days.

The PTP board is mounted 1.3cm underneath the top plate. At the bottom lid, I affixed IAG-supplied metal feet, which are aluminum-milled and nickel-plated. Use sticky black fabric cloth to cover the table while you work, to protect the finish.

FIGURE 5: PTP terminal board drawing.



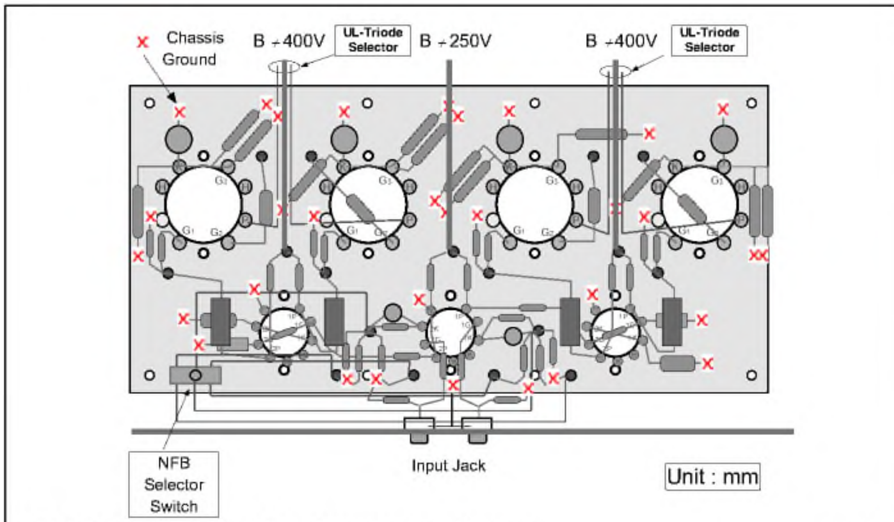


FIGURE 6: PTP terminal board component alignment.

ASSEMBLY SEQUENCE AND WIRING

1. Installation of major components (*Photo 2*).
2. AC line wiring. Since the AC power entry module and the AC switch are on the back panel, close to the power transformer, the wire length is only 5cm or so.
3. Filament wiring (*Photo 3*). Each valve

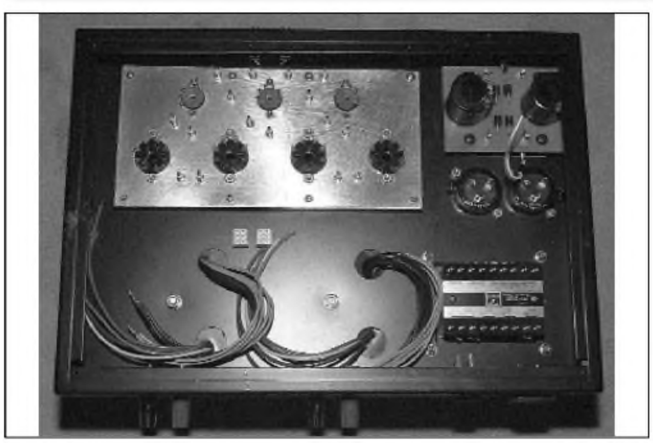


PHOTO 2: Installing major components.

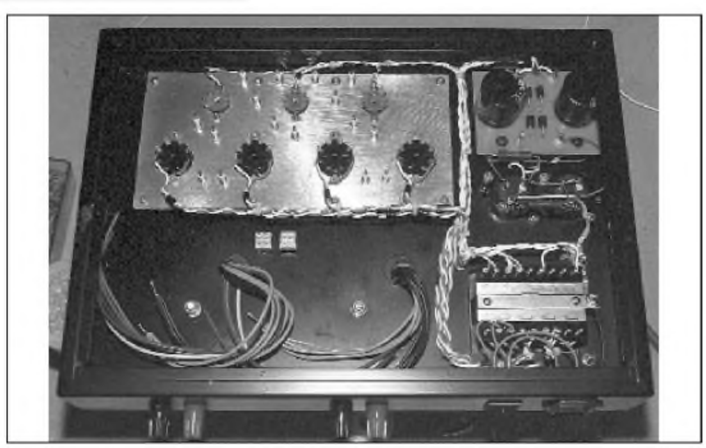


PHOTO 3: Wiring of filament line.

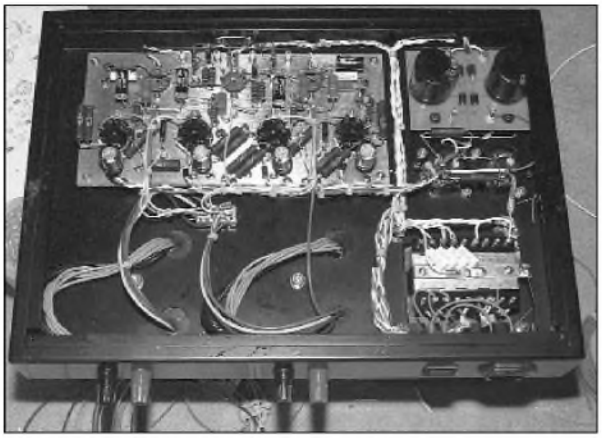


PHOTO 4: Completion of PTP terminal board.

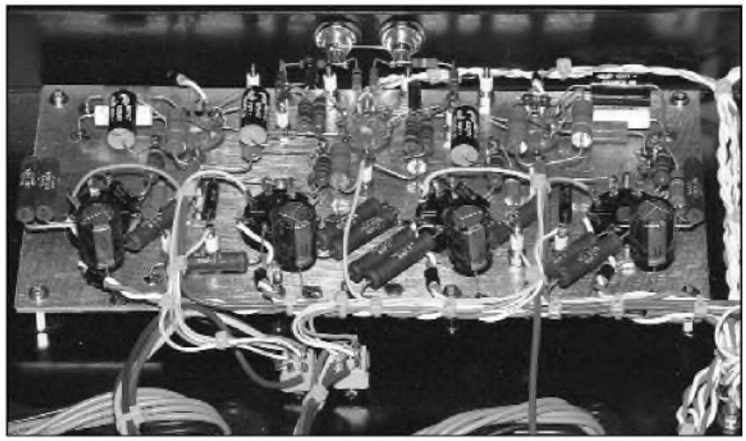


PHOTO 5: PTP board close-up.

used an individual hook-up wire from 6.3V AC windings of the power transformer using 0.50mm^2 thermally durable hook-up wire for the drivers and 0.75mm^2 Teflon-insulated hook-up wire for the output tubes.

4. Installation of the components on the PTP terminal board (*Photo 4*). Wire from the first stage to the phase splitter along the signal flow from the input to the output stage. I used a scheme that puts either resistor or capacitor between the valve-socket pins and the PTP board pins, minimizing a number of terminal pins on the board. Bend the component lead-wires so that these components fit into these pins in the space on the board. Thus I intentionally used a larger resistor than the one required by calculation. This provides a thicker lead wire than the smaller one, to hold it in the air between pins over the board (*Fig. 6*). *Photo 5* is a close-up of the PTP board.

   Due to this wiring scheme, I was able to eliminate any shield wire for the signal line. Also due to the synergy effect, I did not use any shield wire between the RCA jack at the front panel and the grid pin of the first stage.
5. The DC power line wiring is Teflon-insulated (*Fig. 7*).

NFB SELECTOR

Originally, at the design phase, I did not intend to include this selector switch, but I changed my mind while measuring the amplifier characteristics. Since the voltage driver has enough gain, the input sensitivity became quite high. I hit upon the idea of implementing the NFB selector switch, placed on the PTP board with a 30mm long spacer, and soldered directly to the PTP board. The switch itself is 2-pole, 3-position with a neutral stop in the middle to turn off NFB.

The switch knob goes through a vent in the bottom lid plate. The switch works well to let you sample a different flavor of sound depending on NFB level, and also works like a volume control to change the level.

ADJUSTMENT

Prior to final adjustment, double-check for wiring errors. Basically the amplifier does not need any adjustment by design, though I suggest you check the voltage at the major nodes with a digital multimeter to confirm the correct setup.

First, turn the power switch on before plugging the valves into their sockets. Check out B+ power-supply voltage—which may be 450V DC or so. Second, check the filament voltage, which should be 6.3 ±0.5V AC.

Turn off the power switch, then plug in all the valves. After several minutes warm up time, check the cathode voltage of the final tubes; it should be 36V or so with the well-matched Svetlana tubes. This also brings the well-balanced idling current from matched tubes to within a few mA difference between them. That is all for the checkup. The voltages of major nodes are shown in *Fig. 2*.

MEASUREMENT—*FIGS. 8, 9*

There is very little difference between the UL and triode connections in performance characteristics such as input versus output, frequency response curve, and so on. However, the triode connection shows a better linearity and a poorer sensitivity of approximately 2dB. The output clipping level is approximately 18W with an input voltage level of 0.9V through 1.9V.

DISTORTION

Both UL and triode modes show no more than 1% distortion up to 10W within the most convenient output range. In the range between 10W and 18W under triode connection, the distortion was 4–9%, which is somewhat high, although this might be OK in practical use. The UL connection generates up to 35W output with –12dB NFB, even with a self-bias circuit.

FREQUENCY RESPONSE

The frequency response shows a gradual decay above 20kHz at the output level of 1W under both UL and triode connections without NFB. Triode connection shows a better frequency response than UL connection. The high-end cut-off frequency response at –3dB level is 150kHz without NFB.

Even at 10W output, response extends up to 90kHz. The 12dB NFB extends the high-end cut-off frequency to 200kHz, although the peak level was up only 1dB.

DAMPING FACTOR

Damping factor was determined by on-off method at 2V-output level at 8Ω loading. The calculation comes from three switchable cases: non-NFB, –6dB, and –12dB NFB.

WAVEFORMS—*FIG. 10*

Waveforms came from the 8Ω loading driven by a 1kHz, 100Hz, and 10kHz square wave. Waveforms show no overshoot, resulting from the typical performance of the toroidal transformer. Once I turned NFB on, a minor ringing resulted, indicating the extension of high-end cut-off frequency. It is obvious that Plitron toroidal transformers do not show any overshoot at either 10kHz or 1kHz, even without NFB. Overall, the waveform is a good shape.

LISTENING IMPRESSION

I used a couple of JBL S3100 speakers

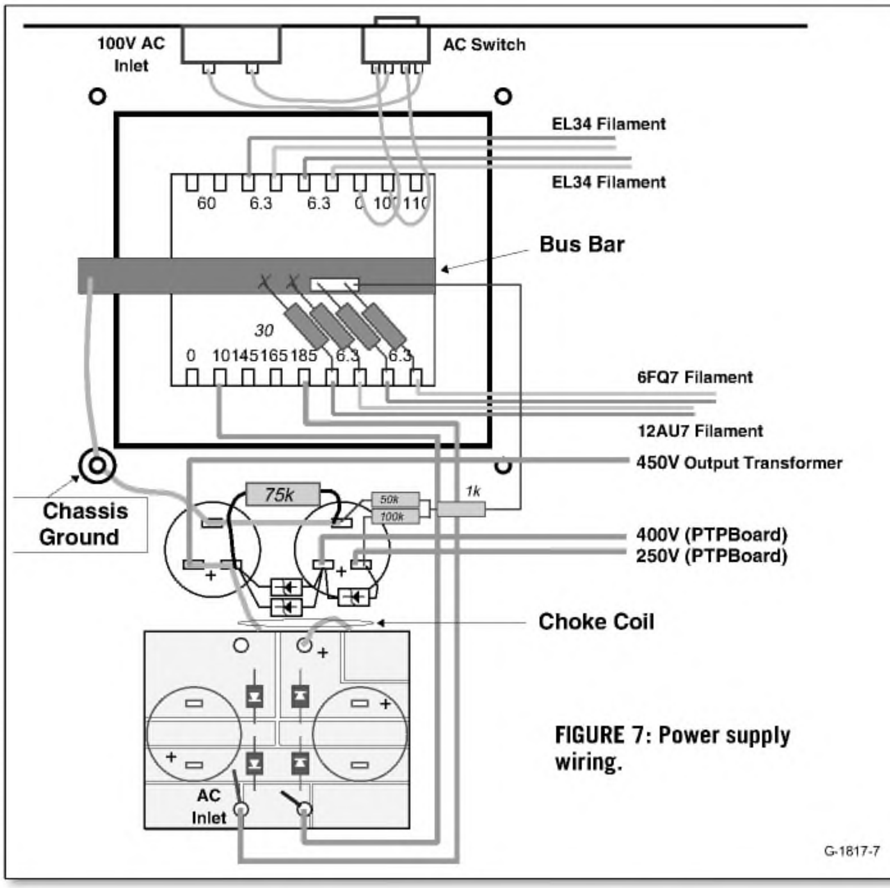


FIGURE 7: Power supply wiring.

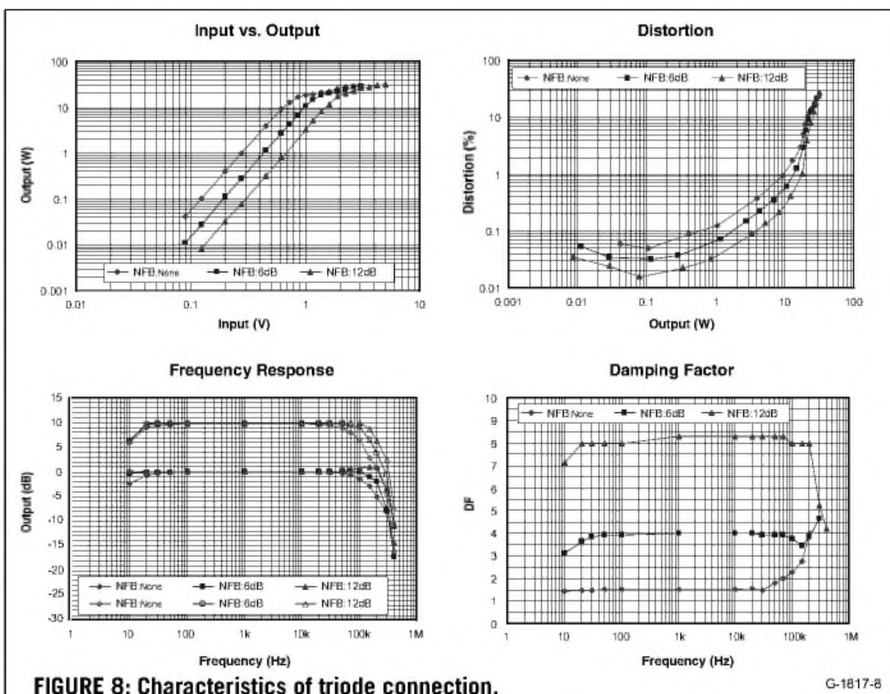


FIGURE 8: Characteristics of triode connection.

with a selector switch between this amplifier and a homebrew 300B amplifier to compare the sound difference.

The first impression is quite clear; e.g., it showed a high clarity of the sound. I was particularly impressed that the sound from this amplifier has a clearer, more vivid, and stronger sound than that of the 300B amplifier, as long as I play soft rock, jazz vocals, and so on.

Changing the source from soft rock to classical music with an orchestra, I noted an obvious difference. Trumpets and similar musical instruments bring warmer tone than that of the 300B amplifier. Damping on the low tone is stronger than ever. You might say that the sound of the 300B amplifier seems weaker than this replica amplifier.

However, this sound difference can be monitored only when I switch between this and the 300B amplifiers. The difference is extremely small.

SUMMARY

I was able to expand the high-end cut-off frequency from 90kHz to 200kHz, by adding only 12dB NFB while keeping the peak boost as small as possible. Thus I do not think this unit requires the extra components of the original Marantz 8B.

I could reach the goal of wider power bandwidth of 200kHz with the simplest NFB technique, using state-of-the-art components. I believe this technology would work even in the era of digital audio. I dare say that this replica amplifier exceeds the overall performance of the original Marantz 8B. ❖

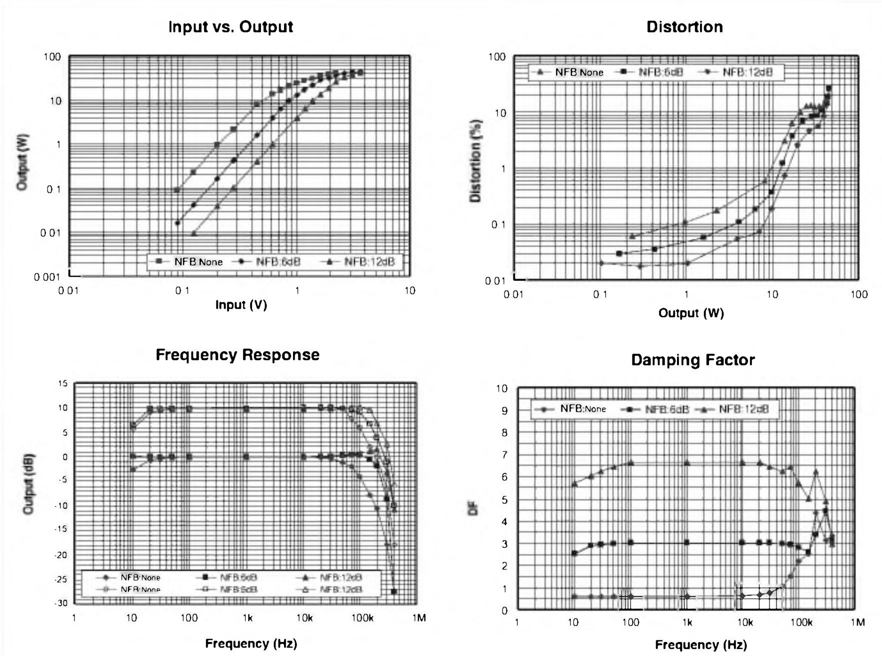


FIGURE 9: Characteristics of ultralinear connection. G-1817-9

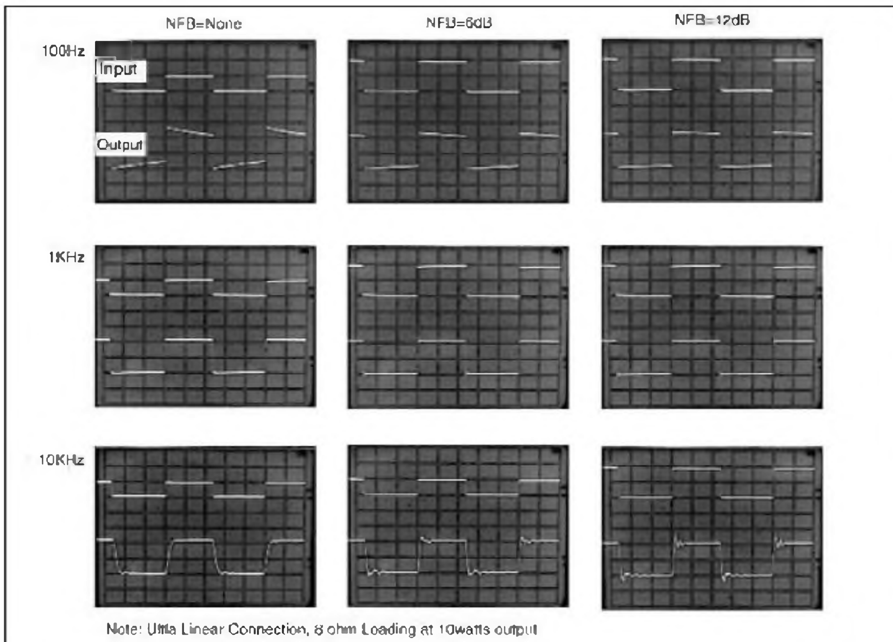


FIGURE 10: Output waveforms. G-1817-10

SOURCES

Tec-sol Inc.
Hamamatsu-shi, Wada-cho 514
Shizuoka 435-0016 Japan
053-468-1201
FAX: 053-468-1202
URL: http://www.plitron.com/
URL: http://www.tec-sol.com/
URL: http://www.svetlana.com/
Vacuum tubes, Plitron toroidal transformer

Hirata Denki Seisakusho—Closed as of Oct. 2000
Arakawa-ku, Nishi-Ogu 1-24-7
Tokyo, Japan
03-3800-2251
Power transformer, choke coil

San Ei Musen—Closed as of Aug. 2000
Chiyoda-ku, Soto-Kanda 1-15-16
Radio Kaikan Bldg 4F Tokyo 101 Japan
03-3251-7985
FAX: 03-3251-2343
Chassis

International Audio Group Inc.
PO Box 10096
Killeen, TX 76547-0096
Phone/FAX: (254) 699-8702
Hiag@n-link.com
PTP terminal board, metal feet

San-Ei Musen (Akihabara, Tokyo)—Closed as of Aug. 2000
Suzu-sho (Akihabara, Tokyo)
Akizuki-Denshi (Akihabara, Tokyo)

HALTEK (Mountainview, CA)—Closed as of April 2000
Resistors, capacitors, semiconductors, miscellaneous

MEASUREMENT EQUIPMENT

Audio Analyzer HP-334A
Audio Oscillator Kenwood AG-204D
AC Volt Meter HP-403B
ATT HP-3467A
Dummy Load Homebrew 8Ω 50W, 2ch.
Digital Multimeter Fluke 8020A
Oscilloscope HP-1746A

REFERENCES

1. "Building 6CA7 Stereo Push-Pull Amplifier," Y. Uesugi, *Radio Gijutsu*, Jan., 1968.

2. Marantz 8B original circuit diagram. Parts Connexion Catalog.

ACKNOWLEDGMENT

To Mr. Eric Barbour at Svetlana for his useful suggestion via e-mail, which helped me to complete this project successfully.

A DVD Rescue

Lights, cameras, action! Here's a simple modification to DVD players that's guaranteed to rate two thumbs up. **By Darcy E. Staggs**

PHOTO 1: Example DVD player including improvement resources.

In a brief letter to *audioXpress*[1], I introduced the benefits of improving the signal path capacitors for the audio and video in my TV set, and of damping mechanical components in a DVD player. It turns out my report was woefully premature, since continued tests showed I had barely scratched the surface. Following my own advice, I soldiered on, ultimately applying both techniques to each video component to the limit of my skills. Here's the entire story.

First, a confession: I began this project with a very lukewarm attitude, because TV has always appeared so lackluster. Step by step my enthusiasm grew, as the picture quality kept making very unexpected incremental gains. I discovered that a thorough approach to this modification process elevated video imagery to spectacular picture quality—from my viewing position ten feet away—and I am eager to get the word out.

I spent hours agonizing over how best to breathe life into my normally dry text, and how to motivate readers to duplicate these rewarding efforts. I wish to share the thrilling graphic improvements I uncovered because these efforts have exposed a paradoxical casualty of the keen price competition in the video marketplace—picture quality.

The intriguing processes you read about here will almost certainly never be exploited commercially, due to their unsuitability for mass production. Yet the experimenter—with very little effort beyond normal caution with electrical circuits—can achieve stunning audio/video from consumer electronics, which will make nearly everything else look broken.

If you are among those who believe that an electronic circuit is completely defined by its schematic, stop here. If you recognize an electronic instrument as a complex assembly of specialized physical components—through which electric currents are forced to pass—which react in many, many ways often ignored, then I am pleased to share the following drama with you.

THE SETTING

When newer DVD players hit the market, I located a superseded Sony DVP-S300 on sale (*Photo 1*, shown with items used for modification), brought it home, and cabled it into my 1996, 27″, 525-line Zenith SY2768S. I had high expectations, and, sure enough, the demo DVDs made all other program sources pale by comparison. The specs are impressive; not only for the video, but for the 96kHz, 24-bit audio as well. And it played CDs.

Cinema is arguably the greatest continuing art form on our planet, and to be able to collect and enjoy it on the superb DVD format is, for many, a unique and pleasurable experience—we have nearly a century's worth of films from which to choose.

THE PLOT

As time went on, however, numerous annoying clues made me realize that the anticipated video nirvana was not yet at hand at my house. Not only that, but the audio tracks had hashy highs, and the CD playback was noticeably tame.

At this point, my experiences with thoroughly hot-rodding a CD player came to mind—after all, a DVD player is basically a very advanced CD player. What could I expect from tweaking a DVD player?

Cutting momentarily to the climax, you, too, can achieve video nirvana with little more than a pair of scissors, some tools, and a roll of 1″ wide, 0.005″ thick, adhesive-backed lead foil tape from McMaster-Carr Supply Company (mcmaster.com), stock number 76315 A14, plus a few hours of your time. As a bonus, the audio CD performance of your DVD player will more than likely deliver much of what high-end stereo is all about.

So, settle into your comfortable chair, and continue reviewing your theater program before curtain time.

THE ANTAGONISTS

The fiercest, most pervasive nemesis of DVD playback—as with CD—is vibration . . . from your loudspeakers, the motors within the player, its transformers, and, fascinatingly, the electronic devices on the PC boards. Even the system timing crystals are guilty here—after all, the whole principle of a crystal is piezoelectric mechanical vibration. Worse, your video monitor is also a sad victim. These evil forces aren't immor-

tal—you can, in fact, oppose them and be rewarded with cinema-like video presentations.

PROLOGUE

I begin by assuming that you wish to extract every colorful raster's worth of image from your equipment. For that purpose I use high-quality coaxial video cable—RG-59 (solid core center conductor) with gold RCA connectors; Sorbothane style or thick felt feet for the player, and a TV monitor which is completely doctored up.

The success of obtaining all the video quality available from your monitor can't be overemphasized, since the typical consumer TV set undergoes a remarkable transformation when thoroughly enhanced. The improvement in detail and color accuracy is striking even on broadcasts (from 35 miles away, in my case), so you might as well complete it prior to tackling the DVD player, and be entertained while you work.

EDITOR'S NOTE: Please make sure your equipment has been *off* long enough so that all capacitors have been fully discharged. This means that all units have been unplugged from the wall for at least three hours prior to work on the chassis.—Ed.

ACT I—THE MONITOR

To begin, complete the capacitor upgrades described in reference 1. Electrolytic capacitors need all the help they can get[2]. The work is easy, but if you aren't confident around PC boards carrying high voltages, get help from someone with suitable experience. With those absolutely vital modifications on board, you are ready for the rest, with my set serving as an example.

There are only two large ICs in my Zenith—one is the microcontroller that "operates" the set; the other is the video processor, an impressive example of large-scale integration. I put two layers of lead foil accurately on top of both these large ICs, and a few strips of lead on all large, resonant aluminum heatsinks, to calm them down also. The two crystals received two layers of the same foil.

The video output board, plugged onto the back of the picture tube, rings like glass if tapped. No wonder—part of it *is* glass. I applied two layers of lead along a couple of its edges, remaining well clear of component leads or conductor traces.

I very carefully put foil on the three plastic high-voltage power transistors on the video output board, which control the three color guns. I wrapped the high-voltage-supply filter capacitor serving these transistors, here a 10.0μF 350V electrolytic, with two layers of lead tape. This cap provides current surges to all three guns, causing it to vibrate. I also coated the film caps added in parallel to the electrolytics, and similarly treated the existing high-voltage ceramic caps, for better sharpness and color.

I have collected four resources to keep the monitor in top shape. Besides the service literature, there is a degaussing coil—Electronix Express (elexp.com) no. 01DGC. Another is an old Heathkit test-pattern generator, IG-5240, used to adjust convergence, and finally the home theater test/setup DVD, "Video Essentials" (800.com, amazon.com, etc.).

Image sharpness and color purity are ultimately dependent on doing a patient job of degaussing, purity adjustment, static/dynamic convergence, and focus. With a little practice this becomes a straightforward job, simply by following the sequence in your manual, and represents normal periodic maintenance. These adjustments were done very quickly during manufacture, but you can improve upon them at home with more care. Leave these settings until last, since your set's color response will normalize, thanks to your improvements.

ACT II—DVD ON THE OPERATING TABLE

As the curtain rises and the stage lights come up, you are at your table, turning a DVD player upsidedown, as I did. The underside is an undulating terrain of plastic or stamped metal, with an occasional screw or plastic fitting breaking the monotony. Using lead tape, coat the bottom as conveniently as possible—100% is unnecessary—leaving holes, screw threads, plastic fittings, and so on, untouched. Burnishing the tape down maximizes its effectiveness.

Next take off the top cover (six

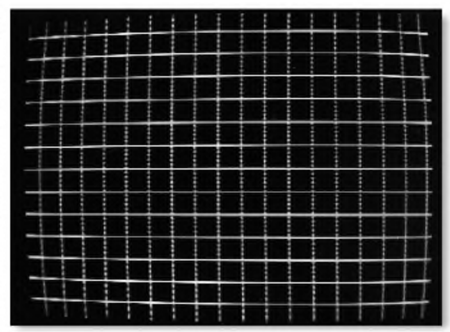


FIGURE 1: Crosshatch pattern on modified Zenith TV.

THE DRAMA CONTINUES

Some months after completing the work documented in this article, I had a couple over one evening, and volunteered to demonstrate my video system before the after-dinner conversation began in earnest. They agreed, so I let the TV warm up on a broadcast channel while gathering up the best demo DVDs.

My guests' first reaction was to stare in utter fascination at the TV image. Next, they both fastened their gaze on me with huge question marks hovering over their heads. Worldly-wise raconteur and master of quick comebacks that I am, I reached down into my vast English repertoire and came up with the unforgettable remark, "I've been working on the TV set." Pondering this fascinating explanation, all eyes again returned to the TV. The picture quality they viewed during the next 15 minutes was followed days later by an e-mail request for what I had done, since the young man's brother pursued electronics as a hobby.

Since then, I have completed another extremely beneficial modification. It seems that six wires of unknown pedigree, each 25″ long, reached from the main PC board to the high-voltage board on the CRT neck, three of which are the red, green, and blue signal wires. I replaced all six wires with wires of identical gauge and length, but which were silver-plated, Teflon®-insulated copper.

Now it's painfully apparent that some broadcast TV images are even better than many DVDs. Color purity, edge sharpness, and resolution of small details greatly improved. A new antenna on the chimney was easy enough to accomplish. Given the age of my TV set, I think it would be far wiser to fiddle with a modern system with component video or RGB input. However, playing full-data DVDs results in a near-cinema display. When HDTV and plasma monitors break the price barrier, I'll be there, but meanwhile things look very, very good.—DS screws, in my case). Run tape on both the cover and portions of the chassis across the area where they meet. On the inside top of the cover, apply lead foil or any other damping material you prefer and for which space is available, to quiet this lively structure. Lead tape over the damping pad works exceptionally well.

ACT III—CUTTING DEEPER

Delving deeper, I removed the RF shields from above and below the main circuit board and carefully coated them with foil. By removing more screws, I was able to lift out a large, box-like metal shield, which took me all the way down to the transport assembly. This shield also received a lead jacket. I even taped up the plastic beam that holds the upper spindle for the disk. Your player may be similar—keep exploring. Most players are assembled by robots now, which means improved mechanical simplicity and accessibility for experimenters.

The final, independently tested modification to the DVD player was to cover all large integrated circuits with two layers of foil, and medium-sized ones with a single layer. I skipped the smallest ICs due to their tiny size. The image quality gained in sharpness, as did color purity, sense of linearity, and detail. So did the soundtrack!

Damping the heatsinks, filter capacitors, and power transformer frame, for example, improves a DVD player. Just be careful of electrical short circuits. Everything I damped improved the image to some degree, and it all—repeat, all—adds up.

INTRIGUE—TAMING THE LIGHT FANTASTIC

This next operation may sound like stage magic, but it works. From your local hobby shop, obtain a small bottle of flat camouflage paint, and brush a coat on the foil you added—it can reflect stray laser light. For the same reason, I painted a broad swath on the inside of the transport cover over the path of the read head—stray light is scattered by the disk, so it must be absorbed.

ACT IV—THE CLIMAX

Lights, fanfare: Reassemble, install, press play! If your experience is anything like mine, the following improvements will be spectacularly evident:

1. Sharpness—*The Fifth Element* (Columbia) is reference quality. The annoying shadows along contrasty vertical edges have essentially vanished, and previously unseen tiny details greet your surprised eyes. Try a black-and-white film, such as *Casablanca* (MGM), where the disturbing rainbow artifacts disappeared from certain scenes, and cut-glass pendants on the table lamps in Rick's Café Américain look sharp and jewel-like. Resolution improved to almost single-raster level, which was one of the real quarries of this hunt, now essentially over.

2. Color purity—jaw dropping. My hat's off to the people who capture such subtlety of hue, richness, and delicacy, and get it accurately onto a DVD. *Roxanne* (Columbia) is a cinematic delight, for its play of color, pastels, lush outdoor scenes, and thoughtful details. *The Fifth Element* is a great example of bold, lavish hues. Broad expanses of color contain no noisy blemishes—just pure, smooth texture, vividly evident on the demo tracks of "Video Essentials."

3. Contrast—Somehow, the pictures look more three-dimensional. The artwork in the animated *Titan AE* (20th-Century Fox) even looks 3D, as do black-and-white films. This was quite unexpected, but certainly complements the cinema sensation.

4. Soundtrack—clean, detailed, effortless, dynamic. The 94kHz, 24-bit audio does wonders for movie sound. Too bad I don't have 5.1 channels! All grit has disappeared from the treble, and the spectrum is balanced and clear. An aural transformation.

EPILOGUE

Verifying the efforts presented here, investigator Michael Danbury has collaborated by repeating them in a totally different DVD player. He even paralleled its electrolytic output capacitors with film caps, all with significant improvements. The eerie feeling persists—perhaps there is more that can be done!

(Curtain. House lights. The audience files out, conversing in lowered voices. Phrases such as "globalization," "bean counters," and "my computer monitor" are overheard.)

REFLECTIONS

As we stand by the busy nighttime street waiting for a taxi, a few parting remarks are offered as this evening's performance echoes in our minds. Who will actually see the stunning images available on their collections of DVDs? Most of us will buy consumer-grade video, take it home, and plug it in. But this doesn't necessarily let the prodigious genie out of the bottle. You now see how I have done so, and with unexpected success. I think you, too, will be surprised when you see what wonders await. It can be said with certainty—DVD will be around for a very long time. ❖

REFERENCES

1. "Audio Aid: Improving DVD & Video Performance," Jan '01 *audioXpress*, p. 112.

2. Curl, John, and Walt Jung, "A Real-Time Signal Test for Capacitor Quality," *The Audio Amateur* 4/85, pp. 22–24.

A Flared Port Study

Flared vs. straight ports? This author has already done all the testing to help you determine which is better. **By Jim Moriyasu**

While doing some low-frequency tests with a vented loudspeaker, I noticed some odd "chuffing" sounds coming from the port. Since this was a test cabinet with a removable front panel, I decided it would be easy to compare the existing straight port to a flared port. A study by Vance Dickason[1] indicated that flared ports are much better than straight ports in dealing with vent turbulence.

TEST SETUP

As you can see in *Photo 1*, the ports are attached to the inside of a removable front panel that connects to the enclosure with toggle clamps. The frame holding the panel has a ⁷⁄₁₆″ neoprene gasket. The Peerless 1858 8″ woofer is tuned to 23Hz in the 1.4ft^3 enclosure.

Because the 3″ diameter ports are so long—17″ for the straight port and 17¾″ for the flared port—they must be mounted so they face outwards. The flared port needs to be longer than the straight port because, for a given vented box tuning, a larger port diameter requires a longer port and the flare increases port diameter.

The flared port is made by Precision Sound Products and is marketed as the Precision Port™. According to Steve Gahm, president and owner of Precision Sound Products, patent No. 5623132 covers the Precision Port, which is available in 3″ and 4″ inside diameter versions. As you can see in *Photo 2*, the Precision Port is a modular system. Each kit includes one 12″ tube, one inside flare, one outside flare, and two connecting rings. The outside flare has a mounting flange that is textured for better cosmetics and has pre-drilled holes for mounting. *Photo 3* shows the assembled port from the front, while *Photo 4* shows the rear view.

The 3″ diameter kit has a 5¼″ diameter inside flare and a 6¼″ diameter outside flare. To mount the port, all you need to do is cut a 5¼″ hole with a jig-

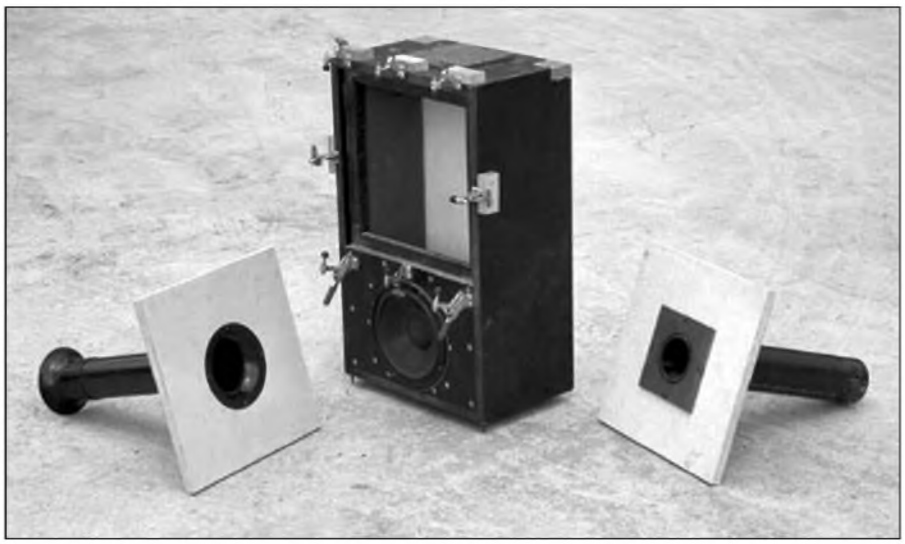


PHOTO 1: Test box with Peerless 1858 8″ woofer, flared port, and straight port on removable panels.

PHOTO 2: Precision Port flared port kit, which includes one 12″ tube (trimmed for shorter port), one inside flare, one outside flare (on the right), and two connecting rings.

saw or fly-cutter on a drill press. To ensure an airtight seal, I made a ½″ wide, 1⁄32″ thick neoprene-cork gasket. I bought my Precision Port kit from Meniscus for $12. ABS plastic pipe is commonly available in the plumbing section of your local hardware store and is used for drainpipes.

I made low-frequency sound pressure level (SPL) measurements with Loudspeaker Measurement System (LMS), which is offered by LinearX. I placed the loudspeaker system on the ground more than 25′ from my workshop, and put the LMS microphone on the ground to do ground-plane measurements. The microphone was at either a meter or half-meter from the loudspeaker. The half-meter distance was necessary when measuring at 1V or less because of noise problems.

MEASUREMENTS

Before doing SPL measurements, I checked the box tuning to make sure the ports were tuned to the same frequency. I also wanted to see how the ports performed with higher power levels. Using a VI box from LinearX, I made impedance measurements at 1.00, 2.83, 5.66, and 11.32V.

Figure 1 shows the impedance curves for the flared port and the straight port at 2.83V. The lowest value at the "valley" is 3.44Ω for the flared port and 3.65Ω for the straight port with both more or less at 23.0Hz, which indicates both ports are tuning the enclosure to the same frequency.

In *Fig. 2*, with 5.66V, the effects of port compression become evident as the maximum impedance at 15Hz declines and becomes more peaked. The minimum impedance for the flared port rises somewhat, but the minimum impedance for the straight port rises even more. At 11.32V, as seen in *Fig. 3*, the maximum impedance of the 18Hz peak drops further as port compression becomes severe. Minimum impedance for the flared port increases to 3.76Ω, but minimum impedance for the straight port jumps to 4.36Ω.

So, the flared port performed better than the straight port in this comparison, because its minimum impedance didn't increase as much. Both ports performed similarly below 20Hz, however, because the level of their impedance peaks below 20Hz were about the same.

Figure 4 shows SPL measurements of the loudspeaker with the flared port or the straight port at 1.00 and 11.32V levels from 0.5m. At both voltage levels, the upper trace is the SPL measurement of the loudspeaker with the flared port.

You can plainly see with the higher level that the flared port has about 2dB more output below 40Hz than the straight port. At the 1.00V level the output gain is less than a decibel. You can see this more readily in *Fig. 5*, which shows the results of subtracting the SPL curve for the flared port from the straight port.

SUBJECTIVE TESTS

The SPL measurements don't tell the whole story, it seems. At some point, as the frequency declines and wavelengths become longer, or as output is increased, all ports start producing turbulent airflow that is quite audible. It's that "chuffing" sound I spoke of earlier. Are flared ports better at minimizing "chuffing"?

Table 1 shows the approximate frequency where "chuffing" becomes noticeable relative to power. It is interest-

TABLE 1

VOLTS	PRECISION PORT™	STRAIGHT PORT
2.83	no chuffing	26.7Hz
5.66	no chuffing	30.4Hz
11.32	23.6Hz	36.3Hz

PHOTO 3: Assembled Precision Port with outside flare closest.

PHOTO 4: Assembled Precision Port with inside flare closest.

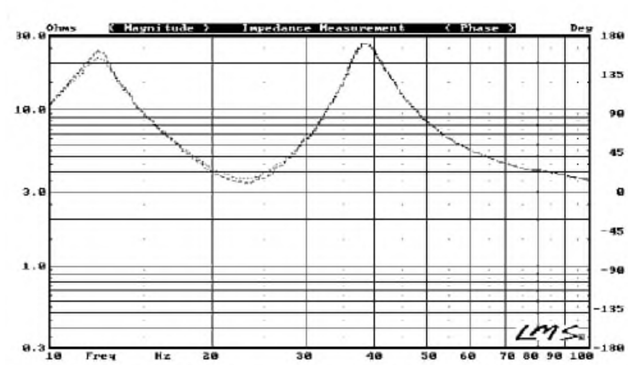


FIGURE 1: Impedance comparison of loudspeaker with flared port (lower trace) versus straight port at 2.83V. B-2084-1

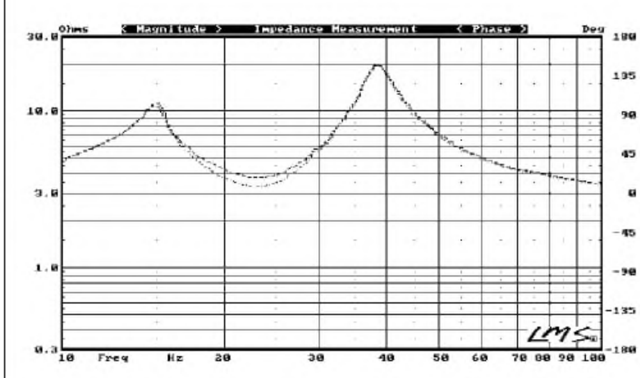


FIGURE 2: Impedance comparison of loudspeaker with flared port (lower trace) versus straight port at 5.66V. B-2084-2

ing to note that the chuffing appears to come from the end that is not attached to the baffle. The chuffing sound seems to come from inside the box when the port is in the cabinet. When it is outside, the chuffing definitely comes from the end that is not attached to the baffle.

The flared port performs very well in this subjective test. Even at 11.32V and 23.6Hz, the chuffing isn't as loud as the straight port at 36.3Hz.

I also held a panel in front of the ports and alongside them to see how close you could get before causing chuffing. It is thought that, in the case of a small or narrow cabinet, placement of a port too close to a wall could lead to chuffing. Some authors suggest a port diameter separation from any surface.

I conducted this test with 11.32V at 25Hz. With the port installed outside of the cabinet, I was able to hold a panel 2″ from the end of the flared port or 1″ alongside it without causing chuffing. I tried the same test with the straight port, but it chuffed so much that it really doesn't make a difference how close you hold a panel.

Another discovery that really amazed me is the forcefulness by which a straight port expels air, compared to a flared port. With 11.32V at 25Hz, you can easily feel the air buffeting your hand from more than 2′ away with a straight port. If you wanted, you could blow-dry your hair with the straight port!

With a flared port, you must move your hand to within 3″ to feel the slight

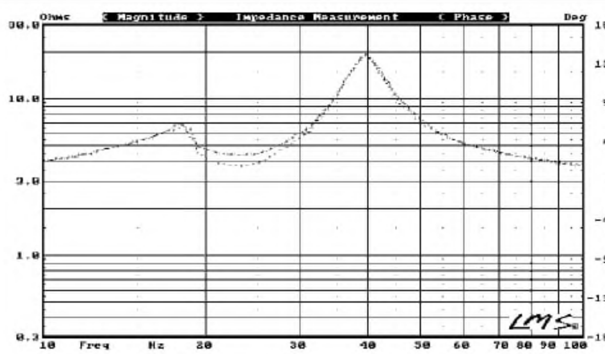


FIGURE 3: Impedance comparison of loudspeaker with flared port (lower trace) versus straight port at 11.32V.

B-2084-3

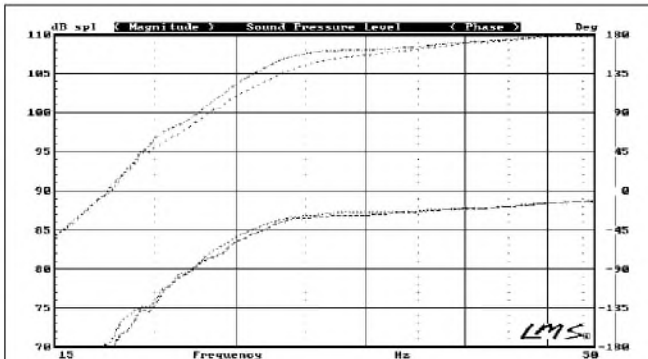


FIGURE 4: Sound pressure level (SPL) comparison of 17¾″ flared port against 17″ straight port at 1.00V and 11.32V. Flared port SPL is the top trace at both power levels.

B-2084-4

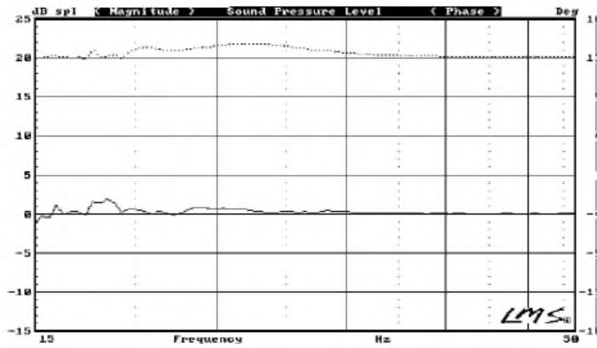


FIGURE 5: Difference curve of 17¾″ flared port over 17″ straight port at 1.00V (0dB) and 11.32V (curve raised to 20dB).

B-2084-5

FIGURE 6: Sound pressure level (SPL) comparison of 9½″ flared port against 8½″ straight port at 1.00V and 11.32V. Flared port SPL is the top trace at both power levels.

B-2084-6

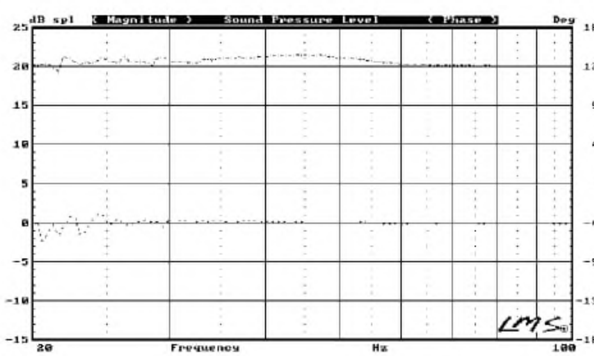


FIGURE 7: Difference curve of 9½″ flared port over 8½″ straight port at 1.00V (0dB) and 11.32V (curve raised to 20dB).

B-2084-7

FIGURE 8: Velocity of air in 17″ port (top curve) and 8½″ port at 1.0V in meters/second.

B-2084-8

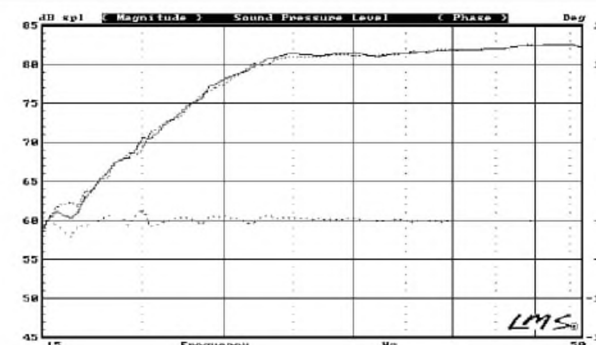


FIGURE 9: Sound pressure level (SPL) comparison of 17¾″ flared port against 17″ straight port at 0.50V. Flared port SPL is the top trace. Difference curve of flared port over straight port raised to 60dB.

B-2084-9

movement of the air! This probably explains and confirms the previous test that shows you can place a flared port within 1–2″ of an enclosure's surface.

So, apparently, the air velocity at the end of a flared port slows dramatically compared to a straight port. This fact no doubt accounts for the much-diminished turbulence, the consequent lack of "chuffing," and 1–2dB higher output of a flared port at higher power levels. Why? Well, a 3″ port has a cross-sectional area of approximately 7 in^2, while the area of a 5¼″ port is over 21 in^2. So, I guess, with the cross-sectional area tripling because of the flare, air velocity drops dramatically.

DISCUSSIONS WITH VANCE DICKASON

Further review of the SPL measurements led me to realize a discrepancy in the low-level, 1V measurements. My measurements still showed the flared port to have more output than the straight port, while Vance Dickason's study showed little or no difference at the 1V level. Curious about this, I rang up Vance to discuss this puzzling situation. After some thought, he suggested that the difference might be caused by the difference in port length, because his study used a vent that was about half the length of the ones I used in this study.

So, I then conducted the same tests with an 8″ Vifa M22WR in a 0.90ft^3 enclosure tuned to 41Hz. The vent lengths were 8½″ and 9½″ for the straight and the flared port, respectively. Again, the flared port must be 1″ longer than the straight port to achieve the same box/vent tuning frequency.

As you can see in *Fig. 6*, which compares the two ports at 1V and 11.32V, the flared port has 1–2dB more output at 11.32V than the straight port. At 1V, there is little difference. *Figure 7* is the difference curve, which shows a very modest advantage for the flared port at the 1V level. These results are exactly what Dickason's study reports.

Vance was glad to hear that the test

confirmed his hypothesis. He then suggested I check the vent velocities. Since I don't have a mini anemometer to insert into the vent, I modeled both woofers with Loudspeaker Enclosure Analysis Program (LEAP) from LinearX.

Figure 8 compares the velocity of the air in the ports at 1.00V; it shows the peak velocity in the longer vent to be about 1.4m/s versus 0.9m/s for the shorter vent. Subjective testing also confirmed these results. You must hold your hand much closer to the front of the shorter vent to feel the air blowing than you do with the longer vent.

So, I went back to the longer vents and measured their SPL at 0.5V. The results are shown in *Fig. 9*. The flared port trace is the top line, while the difference curve has been raised to the 60dB level. The flared port has very little or no advantage between 30–40Hz and has maybe less than 0.25dB more output than the straight port below 30Hz.

Thus, it appears that flared ports have a distinct advantage over straight versions when port velocities are higher than 1m/s. Typically, this would be when implemented in a subwoofer or any loudspeaker meant to be played at 1V or higher. This suggests they should be used in nearly all situations, because 1V is equivalent to ¼W for an 8Ω load. That amount of power with the Vifa M22WR, an 8Ω driver, produces just 67dB at 3m.

CONCLUSION

Flared ports are so much better than straight ports that I can't imagine why anyone would bother to use the latter or purchase a loudspeaker with one. I'll never use a straight port again. For audiophiles looking for a modification or enhancement that really produces results, replacing a straight port with a flared one would certainly be worth the effort and modest expense.

And the Precision Port is easier to install than ABS drainpipe, which requires a hole-saw or a fly cutter on a drill press. Installing a Precision Port, by comparison, is a snap, because all you need is a jigsaw. However, if you want to flush-mount the Precision Port for a cleaner look, you will still need to use a fly-cutter and a router with a rabbetting bit.

Finally, in my opinion, the Precision Port is impressive to look at compared to a lowly piece of drainpipe. After all, why have an ugly hole on that beautiful loudspeaker when you can have a curvaceous (even sexy) flared port? My e-mail address is moriyasuj001@hawaii.rr.com. ❖

SOURCES

Meniscus Audio Group, Inc.
(616) 534-9121
www.meniscusaudio.com

LinearX
(503) 612-9565
www.linearx.com

Precision Sound Products
(815) 599-0662
www.psp-inc.com

REFERENCES

1. Dickason, Vance, *The Loudspeaker Design Cookbook*, Sixth Edition, 2000, Audio Amateur Press, p. 69

Don't Toss It—Recone It

No need to blow your cool if you blow your driver. This quick fix can restore your speaker and your peace of mind. **By Bill Fitzmaurice**

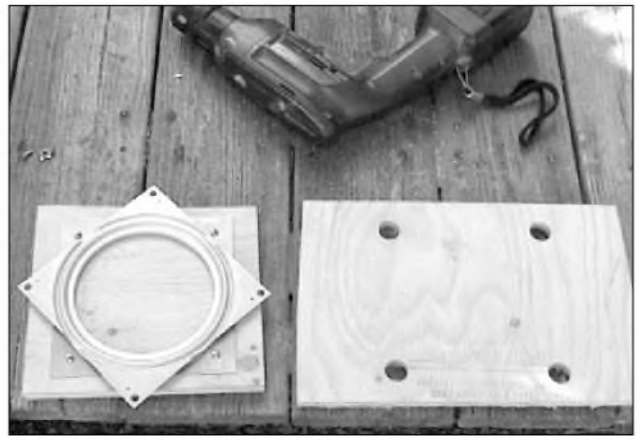


PHOTO 1: Materials for a speaker reconing turntable.

It was only a slight popping sound, but I knew it wasn't good. I had been testing my latest speaker when it occurred. First the "pop," and then silence.

Somehow I had managed to blow the voice coil on my trusty old EVM-15B™. I opened the box, hoping to find a disconnected wire. Not finding one, I used my VOM (volt-ohm-milliammeter) to check for continuity across the driver terminals. No reading.

No doubt about it—the driver was shot. What next, I thought, toss it and order a new one? Well, what about a recone?

I went on-line and typed in "recone," not knowing what to expect. What I found out is that you can usually have a blown driver reconed at less than half the price of a new one, and you can save even more by doing it yourself. The process is fast and easy.

KIT FIX

You can obtain recone kits, in many cases, from the driver manufacturer, although my experience with Electro-Voice is that manufacturers shy away from dealing with DIY'ers. No problem. Just contact Wolf Recone Centers (WVS), either by phone or e-mail.

In most cases they can provide you with an OEM recone kit, and if not, they will piece together the required components from your driver's pertinent measurements. They can also provide you with a step-by-step video of the recone process, and if you are still in doubt, you can even have them do the recone for you, though the shipping charges back and forth will erase much of the savings over buying a new driver.

Before ordering your kit, take a trip to your local hardware store and pick up a "lazy susan" bearing, using it along with a couple of pieces of plywood to produce a small (6″ × 6″ or so) turntable (*Photo 1*). The hardest part of the recone process is applying the adhesive to the driver frame; the turntable will allow you to easily rotate the driver as you apply adhesive for a fast and professional job. Place the driver on the turntable, magnet down. You may need to make spacer rings to allow the driver to sit flat if the magnet assembly has an extended pole piece.

Next, remove the old cone, slicing the surround and spider and cutting the leads with a utility knife. Don't destroy the cone, because you may need to get stock numbers from it. As soon as you have removed the cone, cover the voice-coil gap in the pole piece with tape, to prevent it from accumulating any debris.

Then use a putty knife or chisel to first remove the gasket around the frame and then clear as much of the old adhesive as possible from the frame and the back plate where the spider was attached. Make a mental note of where the adhesive bead attaching the spider to the back plate runs so that you can duplicate it later. Finish the cleanup job with a small rotary sanding disc and portable electric drill (*Photo 2*). With the old cone in hand you can now contact WVS to order the correct replacement.

There are two options on the kit. You may order the cone, voice coil, and spider as separate components, or you may order a "Wolf Kit," which has them pre-assembled, along with a center dome and gasket. In most cases with common drivers the Wolf Kit is a better idea; the separate components are a must only when no off-the-shelf Wolf Kit exists. For my EVM I got the Wolf Kit (*Photo 3*); the rest of the instructions here refer to its installation.

PREPARATION WORK

When you get your recone kit, remove the tape from the back plate and trial-fit the cone assembly into the driver frame to check for a proper match. If all is right, you need to make a spacer to align the voice coil. You can order Mylar spacers when you order the recone kit, or you can use rolled paper, as I did. I have paper in my home office in a variety of weights, and found that for my application 75 lb card stock worked perfectly.

To make the spacer, cut a strip of paper about 2″ wide, and long enough so that when you roll it and place it inside the voice coil the two ends don't quite meet, leaving a gap of about 1⁄16″ (*Photo 4*). Place the cone onto the frame again, with the spacer in place. The

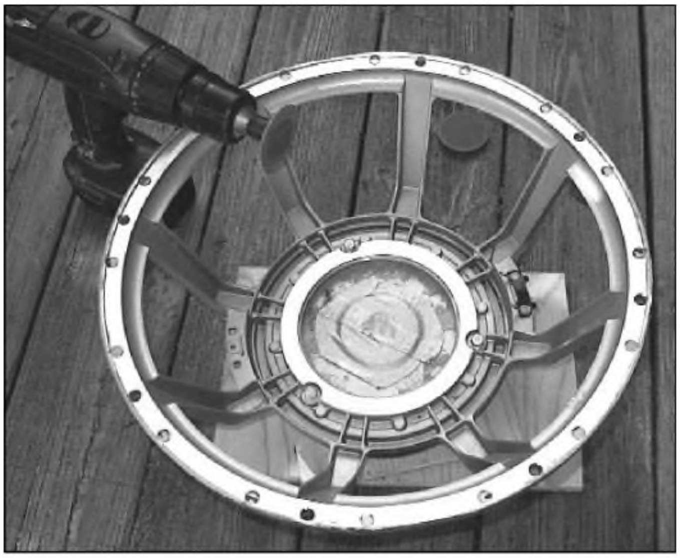


PHOTO 2: Driver frame sanded clean; note tape covering the voice-coil gap.

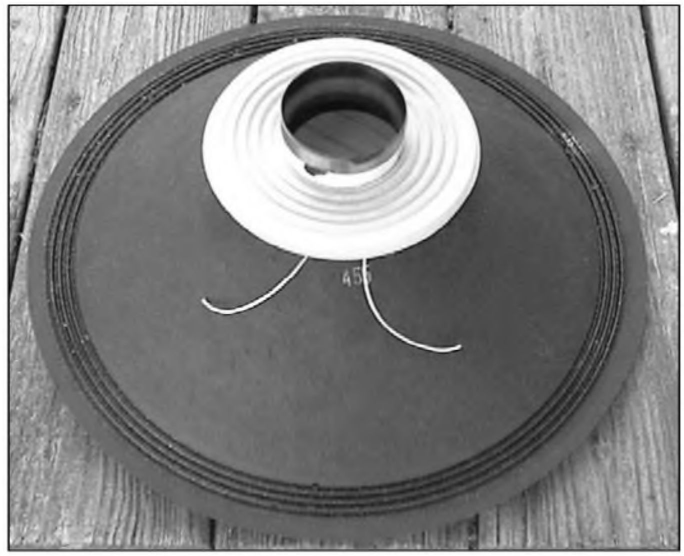


PHOTO 3: Wolf recone kit.

spacer should produce just enough friction between the voice coil and the pole piece so that when you lift the cone a bit from the frame it won't fall back. If the cone floats freely, the spacer is too thin; if you can't get the spacer between the coil and pole piece, it's too thick.

ADHESIVE WORK

Once you have the correctly sized spacer, you're ready to glue everything up. The adhesive of choice is super-glue (cyanoacrylate), with which you will also need a spray bottle of cure accelerator (*Photo 5*). You can order these with the recone kit, or find them at a local hobby supply store. Super-glue is a semi-anaerobic adhesive, in that it cures very quickly when parts are tightly clamped together, but fairly slowly when left exposed to air. When hit with accelerator, however, it sets up virtually instantly, which works well for our purposes.

When all the preparation is com-

PHOTO 4: Voice-coil spacer inside the voice coil.

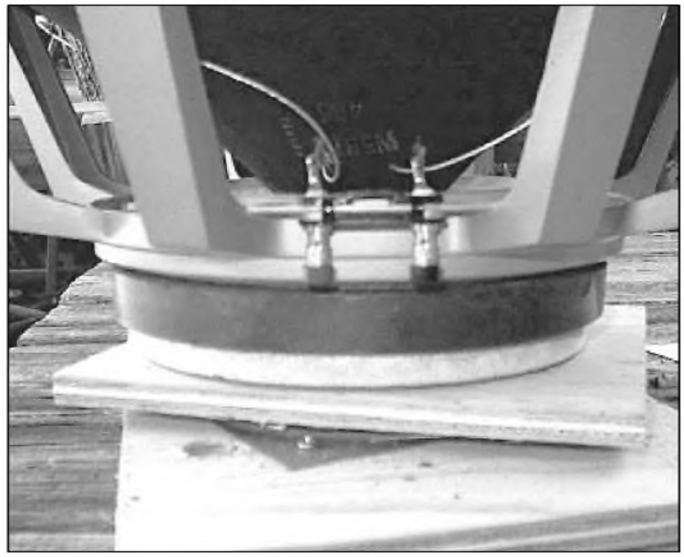


PHOTO 6: Voice-coil leads properly positioned near terminals.

PHOTO 5: Super-glue and spray accelerator.

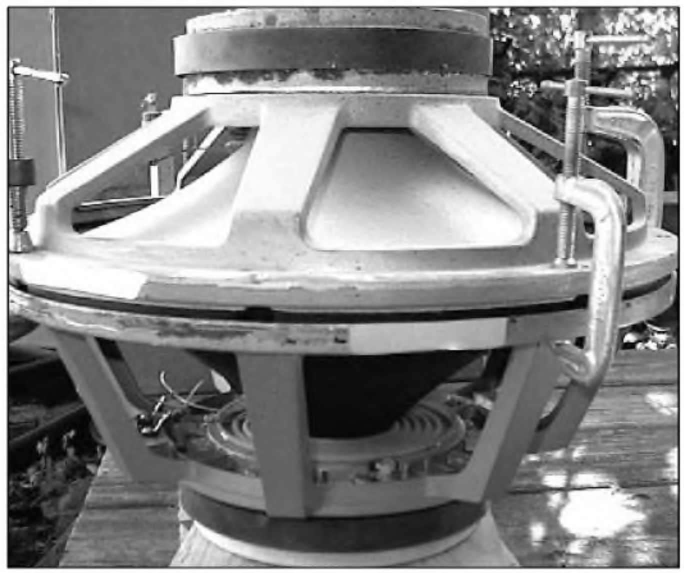


PHOTO 7: Using a second driver and clamps to put even pressure on gasket.

plete, set the cone in place with the spacer, being sure that the coil leads are placed opposite the terminals in the frame (*Photo 6*). Pull the cone upward about ¼″, so that the nozzle of the adhesive bottle will fit between the spider and the back plate. Use the turntable to rotate the driver as you place a bead of adhesive around the back plate in the same location as the original; it may take a couple of turns to get a good continuous bead. Push the cone back into place.

The spider is made of a porous cloth; you should see the super-glue seep through it. Because it is still exposed to air, the super-glue won't set for at least a minute. Using a pencil eraser (not your fingers), tamp down lightly on the spider anywhere that the super-glue has not seeped through. When you're sure that the glue bead is ready, hit it with a few pumps of the accelerator, rotating the driver as you spray. The adhesive will be cured in a second or two.

Now comes the cone surround. Insert the tip of the adhesive bottle between the surround and the frame. Rotate the turntable, squeezing glue onto the frame as you go. To be sure of applying enough adhesive, rotate the frame at least twice. Tamp the surround into place with the pencil eraser. Then quickly apply more adhesive on the face of the surround, after which the gasket is set in place, being sure to align the holes in it with the frame.

Because you can't get accelerant where you need it to effect a cure, you now must clamp the gasket down, using some sort of device to spread pressure evenly. You can make a circular wooden caul for this purpose, or, as I did, use another driver of the same size to spread the pressure of the clamps (*Photo 7*). Allow two minutes for the glue to set and then remove the clamp arrangement.

Remove the voice-coil spacer. Gently

SOURCE

WVS—Wolf Recone Centers
1707 N. Harrison St.
Fort Wayne, IN 46808
219-424-5463
FAX: 219-422-4133
E-mail: team@www.recone.com
Website: www.recone.com

PHOTO 8: The finished reconed driver.

push the cone in and out to be sure that there is no rubbing of the voice coil. If you did it correctly, the cone will move in and out without coil rub; if not, you'll need to get some super-glue remover and try again.

Now place the dome on the driver, rotating the frame to verify that it is centered. Use a finger to hold the dome lightly in place while you tack it to the cone with three or four drops of super-glue, spraying with accelerator. With the dome now securely in place, again rotate the driver, taking two or three turns, while laying a good bead of glue around the dome/cone joint. When the bead looks good, hit it with accelerator and you're almost done.

The last step is to cut the voice-coil leads to length and solder them to the terminals. Make the leads long enough so that they won't be stretched tight in long excursions, but short enough that they can't short out to either the driver frame or each other. You're done (*Photo 8*). Even a beginner can easily finish the job in less than an hour.

VIDEO INSTRUCTION

Now, about the WVS video. If you're using a Wolf Kit you can probably get along without it. However, if you have to piece together all your parts from scratch, things become a bit more complicated, though not more difficult. There is also the matter of replacing rotted foam surrounds and HF driver diaphragms, both of which are also covered in the video. For those cases you'd be better off to have it than not.

The video is not exactly of the highest standards, being obviously self-produced on a camcorder; you can hear delivery trucks in the background, for instance. But aesthetics aren't what you're paying for, so they matter little. What you do get is a "looking over the shoulder" view of WVS owner Tom Colvin actually doing recones, surrounds, and diaphragm replacements, with such clarity that after one viewing I was able to do my recone job perfectly the first time. If I had the time or inclination, I might even take up reconing as a business, and if you should want to do so, WVS will help you with that, too.

So remember, if you hear a little "pop," followed by silence, from your speaker, don't toss that driver! Save that frame, and more than a few bucks, and recone it. ❖

Diffraction Insights

Loudspeaker builders who choose to ignore cabinet diffraction do so at their own risk. This author offers some solutions to address the diffraction problem. **By Mithat Konar**

If you stare at the mathematics describing wave propagation long enough, you will begin to see that whenever there is a change in the environment through which a wave travels, part of its energy will be transferred into the new environment, and part will be reflected back. The change in environment doesn't need to be a change in material; the same thing will happen if there is, for example, a change from a hemispherical to a spherical radiation space. Furthermore, the reflection may—and in the case of cabinet diffraction does—take on the appearance of a reradiation of the wave at the boundary.

So, the scenario in a typical loudspeaker system is this: a wave coming off the loudspeaker diaphragm propagates initially into what is effectively a hemispherical space—a hemisphere defined by the baffle and the rest of the world (*Fig. 1*). Eventually, the expanding wave encounters the edges of the baffle, after which the wave sees effectively a spherical space (*Fig. 2*). Since a change occurs in the environment seen by the wave—from hemispherical to spherical space—diffractive reradiation occurs at the baffle edges (*Fig. 3*).

HOW BAD CAN IT BE?

Now that you have some idea of what diffraction is, you can begin to look at its influence on loudspeaker performance. At very low frequencies, the wavelengths of the sound are long enough that the cabinet itself is essentially "invisible," so you can safely ignore the influence of cabinet diffraction. At higher frequencies, this is not the case. To understand this a bit better, consider the following.

Imagine a very tiny island at sea, a land mass so small that the size of the surrounding natural waves are much larger than the island itself. As the waves pass by, the island effectively does nothing to change the propagation of the waves—it is just too small to make any effect on them. This is analogous to a speaker cabinet at low frequencies—it is as though the cabinet weren't there at all.

Once in a while a small paddleboat passes by, producing tiny waves whose lengths are much smaller than the island. As these lap up against shore, they are reflected back to sea or eaten up by the sand on the beaches. In this case, the propagation of the waves is very much affected by the presence of the island. This is analogous to a cabinet at higher frequencies.

So a simple model of diffraction based on the above is that at very low frequencies sound waves radiate into a spherical space, but at higher frequencies, they behave as though they were radiating into a hemispherical one. Thus wavelengths significantly smaller than the baffle dimensions are reinforced by 6dB compared to those much larger than the baffle. In between, there will be a gradual transition. This phenomenon is often called the "diffraction step."

MOVIN' ON UP

Unfortunately, the model I've outlined is overly simple to accurately describe a loudspeaker cabinet's diffractive be-

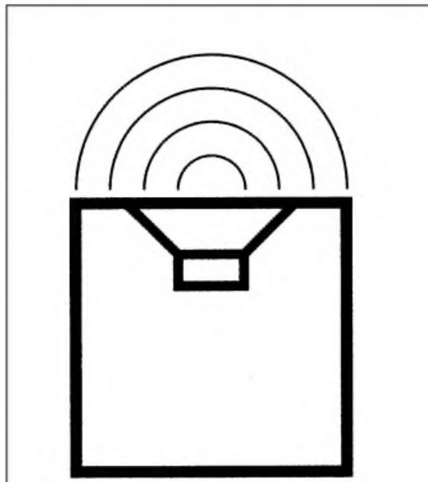


FIGURE 1: Hemispherical radiation off baffle.

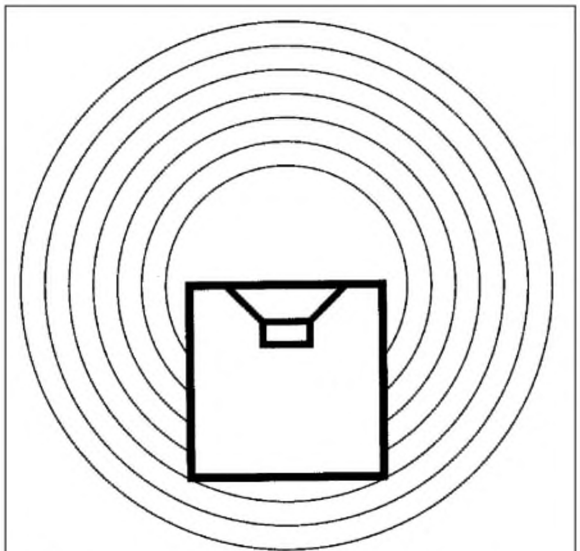


FIGURE 2: Spherical radiation beyond cabinet boundaries.

ABOUT THE AUTHOR

Mithat Konar is the founder of Biro Technology, a manufacturer of high-performance audio equipment (http://www.birotechnology.com/), and he is secretary of the Upper Midwest section of the Audio Engineering Society. He spends a lot of his almost nonexistent spare time playing guitar and saxophone, ruminating on music, and sometimes recording it.

havior. While the overall trend of a 6dB rise at mid and high frequencies is certainly valid, superimposed on this are significant response perturbations caused by constructive and destructive interference from the energy reradiated from the baffle edges. What's worse, analytic solutions for this scenario are possible only for some very simple and generally useless baffle geometries. So what is a poor builder to do?

It so happens that John Vanderkooy of the University of Waterloo has developed an efficient way of modeling just the kind of diffraction phenomena that occur at mid and high frequencies in loudspeaker systems.[1] Agreement between model predictions and actual results are on the order of 1dB or better for axial measurements.[2] I have implemented a modified version of Vanderkooy's model, and have been working with it for several years.

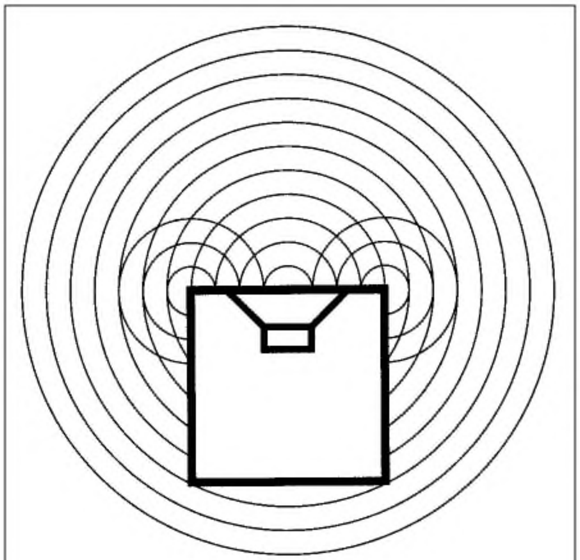


FIGURE 3: Diffractive reradiation.

It would be wonderful indeed if all interested speaker builders could use their computers to investigate the effects of cabinet diffraction. However, until I get around to developing a releasable version of the program, you must be content with my sharing with you some demonstrative examples I have come up with. But don't be disappointed. There is a lot to be learned this way.

First I will show you what happens when two drivers are placed right in the middle of a 13.25″ square baffle. While this configuration isn't exactly typical of what a speaker designer might do, it does show how much the effects of cabinet diffraction can impact the response of the system.

THE SQUARE BAFFLE

Figures 4 and *5* show, respectively, the predicted responses of an idealized 1″ tweeter and a 6.7″ woofer on the square baffle. By idealized I mean that the tweeter is a perfect second-order highpass with a resonant frequency of 700Hz and a Q of 0.45, and the frequency response of the woofer is flat from DC to beyond audibility. Furthermore, I have modeled the radiation patterns of both drivers as a simple piston of appropriate diameter. Because of the limitations inherent in Vanderkooy's model, the results should be consid-

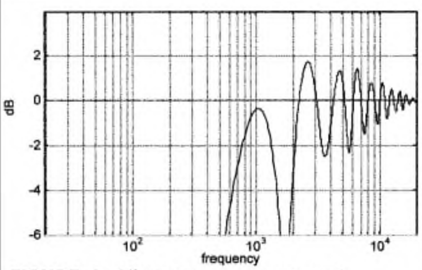


FIGURE 4: 1″ tweeter on center of square baffle.

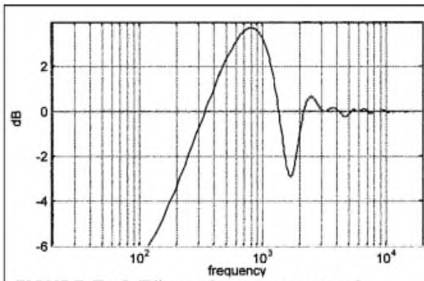


FIGURE 5: 6.7″ woofer on center of square baffle.

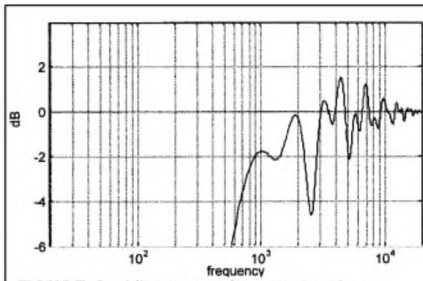


FIGURE 6: 1″ tweeter in center of rectangular baffle.

ered valid for frequencies above approximately 100Hz in all the following figures.

As you can see, the impact on the drivers' responses is not subtle. In the case of the tweeter, periodic ±2dB ripples occur in the passband, and what registers as a 6dB narrowband dip close to 2kHz will easily manifest itself in a system design with even a fourth-order crossover at 3kHz. The periodicity of the ripples is the result of the comb-filtering effects of the symmetrical diffraction. In other words, the diffractive reflections are not distributed very widely in time (and therefore in frequency).

In the woofer's response, you can see the expected 6dB diffraction step. However, superimposed on this is a broadish 4dB peak at 800Hz and a 3dB dip at 1.5kHz. As you will see in the examples that follow, this midrange peak will prove to be quite persistent. (Inasmuch as a peak of this sort will subjectively impart a "boxy" coloration, it looks as though it might be a good idea to question whether the "boxy" sound of box speakers is actually caused by resonances in the box material and internal standing waves, as is commonly assumed, or by these kinds of diffractive effects.)

RECTANGULAR BAFFLE

The next examples use a baffle with the same surface area as the square one, but this time with a 2.17:1 aspect ratio; i.e., it measures 19.5″ × 9″. *Figure 6* shows the response for the 1″ tweeter in the center of this baffle, and *Fig. 7* does the same for the woofer. Clearly, moving from a square to a rectangular baffle has helped to smooth things up—even though the latter has a larger perimeter.

The severity and periodicity of the tweeter's ripples are much reduced with the rectangular baffle (although a nasty 4dB dip has shown up around 2.5kHz). The broad midrange woofer peak has improved by about a decibel, which shouldn't be too surprising. Instead of a fourfold repetition of the same diffractive behavior in the case of the square baffle, the rectangular version results in a twofold repetition of two different diffractive behaviors. In other words, the effects of the diffraction are more distributed in space and time.

TYPICAL MOUNTING

Next, I'll show you what happens when the drivers are placed in locations typical of where a builder might actually put them. *Figures 8* and *9* show predicted responses for the 1″ tweeter and 6.5″ woofer when placed on the vertical axis of the baffle. The center of the tweeter is 3.5″ down from the top, and the woofer is 5.5″ further down.

The tweeter's response appears to have become worse compared to the same driver in the center of the baffle. The overall "rippliness" is no better, and a 3.5dB valley spanning 2–3kHz has developed. While this may not be the result you expected, it is an example of the somewhat unpredictable nature of diffraction.

The new position has not significantly affected the woofer's response. One reason is that in the previous example the woofer was already close to the present "typical" location. Another is that the woofer's response starting in the midrange begins to become directional, meaning that less energy overall is radiated along the plane of the baffle, so there is less energy to be diffracted.

MOVING THE TWEETER

Intuitively, it would seem that horizontally offsetting the tweeter a little should improve diffractive effects by breaking up the vertical symmetry. To this end, many designers recommend a horizontal offset of approximately 0.75″. *Figure 10* shows the response of the 1″ tweeter in the "typical" position of the previous example, but with a 0.75″ offset.

The idea seems to have worked. The response ripples are smoothed out a bit, and the valley spanning 2 to 3kHz has been replaced by a less-offensive dip at 2kHz. However, this "customary" offset is really only the beginning. For any tweeter and baffle combination, it stands to reason that there are a few ideal locations for the driver.

Finding these optimal locations can be a costly and time-consuming affair if you must build dozens of cabinets to do it. But it's a breeze if you have an accurate computer model to tell you what to expect. *Figure 11* shows the response of the 1″ tweeter in one such optimal location found by iterative search. In this case, the models indicated the optimal location for the tweeter—constrained by the need to leave enough room on the baffle for the woofer—was 3.875″ from the top with a horizontal offset of 1.5″.

All the diffractive effects are now well distributed in time, with the result that from 1kHz up there are no significant peaks or dips. This is a vast improvement over the previous examples. You might notice that there is a 0.75dB rise in overall level that takes place around 4kHz, but you can easily deal with this kind of aberration in the crossover.

MOVING THE WOOFER

The examples so far have shown that the driver most sensitive to its location on a baffle is the tweeter. However, the

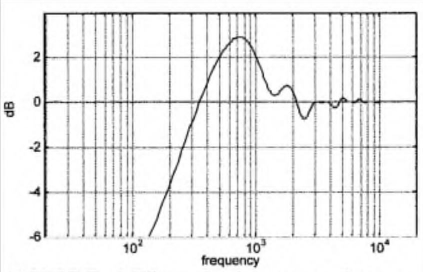


FIGURE 7: 6.7″ woofer in center of rectangular baffle.

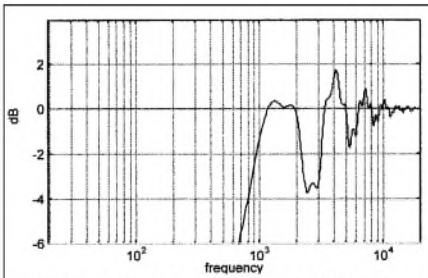


FIGURE 8: 1″ tweeter in "typical" position on rectangular baffle.

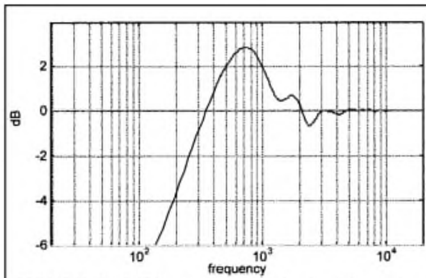


FIGURE 9: 6.7″ woofer in "typical" position on rectangular baffle.

woofer does not remain unaffected. Therefore, it makes some sense to see whether you can smooth the woofer's response by tweaking its location on the baffle. *Figure 12* shows the results of this optimization.

The improvement over *Figs. 7* and *9* is essentially negligible. Two factors combine to give this result. First, at upper midrange and higher frequencies, the driver becomes directive as previously discussed. Second, at lower frequencies, the wavelengths are long enough that the kind of position adjustments you can make are significantly smaller than they need to be. In other words, you could do better only by positioning the woofer outside the boundaries of the baffle! Neat trick if you can manage it.

So the good news here is that you don't need to worry too much about the woofer's location—provided its diameter is comparable to the baffle dimensions. The bad news is that you still must deal with the response modifications that the diffraction introduces.

WHAT YOU CAN DO

In this example, I minimized the impact of diffraction on the tweeter's smoothness by locating it on the baffle where the effects are distributed over time and frequency. However, the final locations I determined are not transferrable to other baffles and or drivers. For optimum results, you need to examine each baffle/driver combination individually to find the optimal driver locations. As a general rule, however, by offsetting the tweeter by 0.75″ or more, you can suppress some of the worst effects of cabinet diffraction at upper midrange and high frequencies.

While offsetting drivers on a rectangular baffle is likely the easiest and least costly means of accomplishing the goal of minimizing the effects of diffraction, it certainly isn't the only way

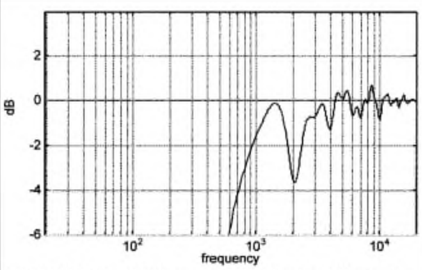


FIGURE 10: 1″ tweeter in "typical" position with 0.75″ offset.

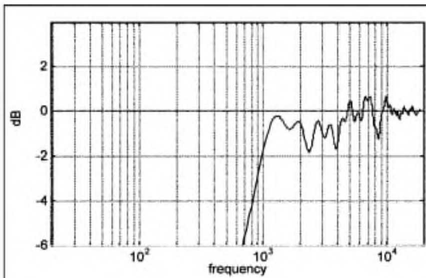


FIGURE 11: 1″ tweeter in optimized position.

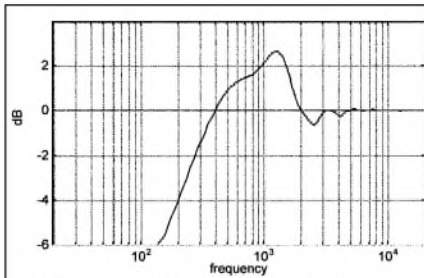


FIGURE 12: 6.7″ woofer in optimized position.

to go about it. Often, loudspeaker manufacturers will radius or chamfer the edges of the baffle in an attempt to ease the transition from hemispherical to spherical space. Doing this is another way of distributing the effects of diffraction in time and space. The extreme implementation of this concept results in a spherical cabinet.

For this approach to be successful, the size of the radius or chamfer must be comparable to the wavelength of the sound over which the diffraction effects are strongest. Using my example, you can see that you need to control effects that show up at 4–5kHz and lower. Inasmuch as a 4.5kHz wave has a length of 3″, the ¾″ radii and chamfers that you frequently see in commercial designs are totally useless in this respect. Some cabinets do use radii in the 3″–4″ range with success; however, such a construction is very expensive for the commercial manufacturer and quite difficult for the home builder.

Another successful approach, used in my own company's products as well as those by B&W and others, is placing the tweeter, the driver most sensitive to the effects of cabinet diffraction, outside the cabinet altogether. For this approach to work, the tweeter's magnetic assembly must be small enough to allow placing the diaphragm very close to the cabinet wall. Furthermore, the driver's housing in all dimensions (including the front-to-back) must itself be free from undesirable diffraction effects.

This generally limits the choice of drivers to those with very compact magnetic systems and housing shapes that generally don't exist in off-the-shelf products. In addition, this approach doesn't eliminate the effects of diffraction. Rather, it changes the nature of the diffraction so that with additional understanding and tools, you can more effectively manage it. Thus, this approach is again fraught with difficulty for the home builder.

IF A TREE FALLS IN THE FOREST . . .

Two questions I've conveniently skirted until now are the following: can you even hear the artifacts introduced by cabinet diffraction at mid and high frequencies, and does minimizing those artifacts in the axial response actually help? To answer this, I can only offer my own subjective experience and the feedback of my trusted customers and listening jurors.

I will state without reservation that the response ripples introduced by cabinet diffraction do introduce audible artifacts, and when they are minimized as this article suggests, system resolution, imaging, and "transparency" are significantly improved, while "grain" and fatigue are noticeably reduced. That's a pretty good deal in my world. ❖

REFERENCES

1. J. Vanderkooy, "A Simple Theory of Cabinet Edge Diffraction," *JAES*, Dec. 1991.

2. S. Rasmussen and K.B. Rasmussen, "On Loudspeaker Cabinet Diffraction," *JAES*, March 1994.

A Valve Preamplifier, Part 2

Using only the best components to stuff all the boards that go into the design, you then proceed to assemble and wire the valve preamplifier.

By G. Haas

It is important that you observe the suggestions given in the text and the components lists, and avoid using inferior-quality "equivalent" types. All printed circuit boards supplied by Experience Electronics for this project are made using fiberglass-reinforced epoxy with a 70µm copper lamination. All resistors in the components list are 1% tolerance, 0.7W metal film types, unless otherwise indicated, with a lead spacing of 10mm. The 2W and 4.5W resistors are metal oxide types with a tolerance of 5%, and have lead spacings of 15mm and 25mm, respectively. The gold-plated cinch sockets, as well as the potentiometers, should, of course, be very high quality.

Before I discuss the construction of the circuit boards, a few words about the enclosure are in order. Even the best electronics—no matter how clever the design may be—are of no use if not housed in a suitable enclosure. With valve amplifiers in particular—given their high working voltages—electrical safety is a primary consideration! A metal enclosure, connected to the protective ground lead, provides both safety and screening. If the enclosure also has an attractive appearance, there will be nothing to disturb your listening pleasure.

All circuit modules are housed in an aluminum enclosure. The advantage of aluminum is that it is non-magnetic, so that it avoids magnetic distortions. In addition, it has very good design properties.

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The enclosure used here has rivetless joints, and the surface is polished and covered by bright nickel plating. Nickel has a warm tone, in contrast to the bluish tone of chrome, and this optically reinforces the "flair" of a valve amplifier.

In order to avoid having an excessive number of screws visible on the top surface of the enclosure, a mounting plate is used to fit all of the electronics. This plate is then screwed to the chassis using eight bright-nickel-plated Phillips screws. In this way, you can produce a visually attractive piece of equipment.

BOARD STUFFING

Now you can stuff the boards. Start with the relay board, as shown in *Fig. 9*.

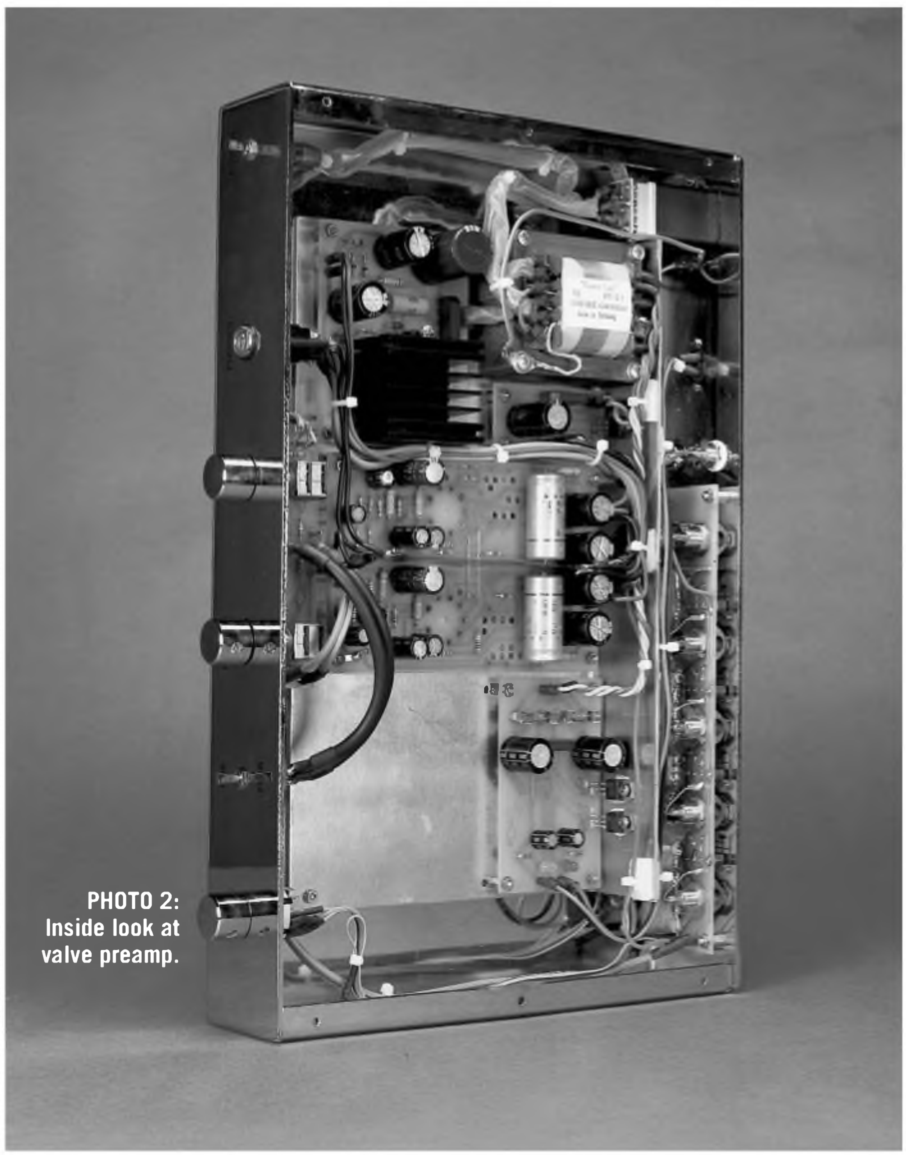

PHOTO 2: Inside look at valve preamp.

The cinch sockets are screwed to the board. You must first tin the tracks at the points of contact, to assure good connections. Apply a drop of solder to each fastening nut to secure it. This guarantees long-term, reliable ground connections. If there is contact resistance in the ground connection at this location, interference is an unavoidable consequence.

The photos show the construction of the circuit board and the placement of the components. Solder strips of 0.15mm-thick copper foil between the sockets. These provide screening on both sides

TABLE 1
COMPONENTS LIST

INPUT BOARD
Resistors:
R1–R8 = 100kΩ
Rx = see text
Re1–Re8 = SIL reed relay, 12V, 1 make contact

Miscellaneous:
K1–K9 = cinch sockets, chassis mount
Two cinch sockets, chassis mount, for recorder outputs
S1 = rotary switch, 1 pole, 4 contacts, break before make
Solder pins
Copper foil

TABLE 2
COMPONENTS LIST

AMPLIFIER BOARD (ONE CHANNEL)
Resistors:
R1 = 680kΩ
R2 = 1kΩ8
R3 = 10kΩ
R4 = 33kΩ
R5 = 2kΩ2
R6 = 150kΩ, 2W
R7 = 470kΩ, 2W
R8 = 2kΩ7
R9 = 8kΩ2
R10 = 680kΩ
R11 = 150Ω
R12 = 270Ω
R13 = 27kΩ
R14 = 150Ω, 2W
R15, R16 = 6kΩ8, 4.5W

Capacitors:
C1 = 1μF 63V, 5mm raster
C2 = 10μF 400V, 5mm raster
C3 = 47μF 40V, 5mm raster
C4 = 2μF2 400V, 5mm raster
C5 = 220μF 40V, 5mm raster
C6 = 47μF 450V, size 18.5 × 41mm
C7, C8 = 22μF 400V, 7.5mm raster
C9 = see text

Miscellaneous:
D1 = see text
V1 (Rö1) = ECL86
1 ceramic "Noval" (9-pin) socket, PCB mount
Copper foil
Solder pins

TABLE 3
COMPONENTS LIST

PROTECTION CIRCUIT
Resistors:
R1, R2, R9, R12 = 10kΩ
R3, R5, R6 = 33kΩ
R4 = 1kΩ
R7, R11, R13 = 100kΩ
R8 = 10Ω
R10 = 390kΩ

Capacitors:
C1 = 0.22μF MKT, raster 7.5mm
C2 = 0.33μF MKT, raster 7.5mm
C3 = 10nF ceramic, raster 5mm
C4, C5 = 1μF 63V, raster 5mm
C6 = 220μF 40V, raster 5mm
C7 = 47μF 40V, raster 5mm

Semiconductors:
D1, D3 = 1N4007
D2, D6 = zener diode 12V, 1.3W
D4, D7 = 1N4148
D5 = BAT43
T1 = BC546B
T2 = BD139-16
IC1 = 555
IC2 = 4023

Miscellaneous:
1 off 8-way DIL IC socket, gold-plated contacts
1 off 14-way DIL IC socket, gold-plated contacts
Solder pins

TABLE 4
COMPONENTS LIST

VOLUME CONTROL
Resistors:
R1, R1′ = 100Ω
R2, R2′, R4, R4′ = 3kΩ9
R3, R3′ = 470Ω
R5, R5′ = 10kΩ
P1, P2 = 10kΩ stereo potentiometer, linear, tracking
2 holders for potentiometers

Miscellaneous:
S1 = rocker switch, 2 poles, 3 contacts
Solder pins

and between the two channels.

In this regard, in many modern pieces of equipment you will find pairs of cinch sockets in plastic modules that can be screwed to the rear panel of the enclosure, with pins that can be soldered directly to the circuit board. These may be less costly for the manufacturer, and they are easy to mount, but using such modules here will impair the channel separation of the entire device, due to the close spacing of the sockets and the lack of screening.

Since soldering the nuts requires a lot of heat, this should be done first, before any other components are fitted. After this, you can insert the resistors, diodes, relays, and solder posts.

The main circuit board with the amplifier (*Fig. 10*) is laid out so you can use it for other applications by adding or omitting components. Use a wire bridge in place of zener diode D1, which is marked with a star on the cir-

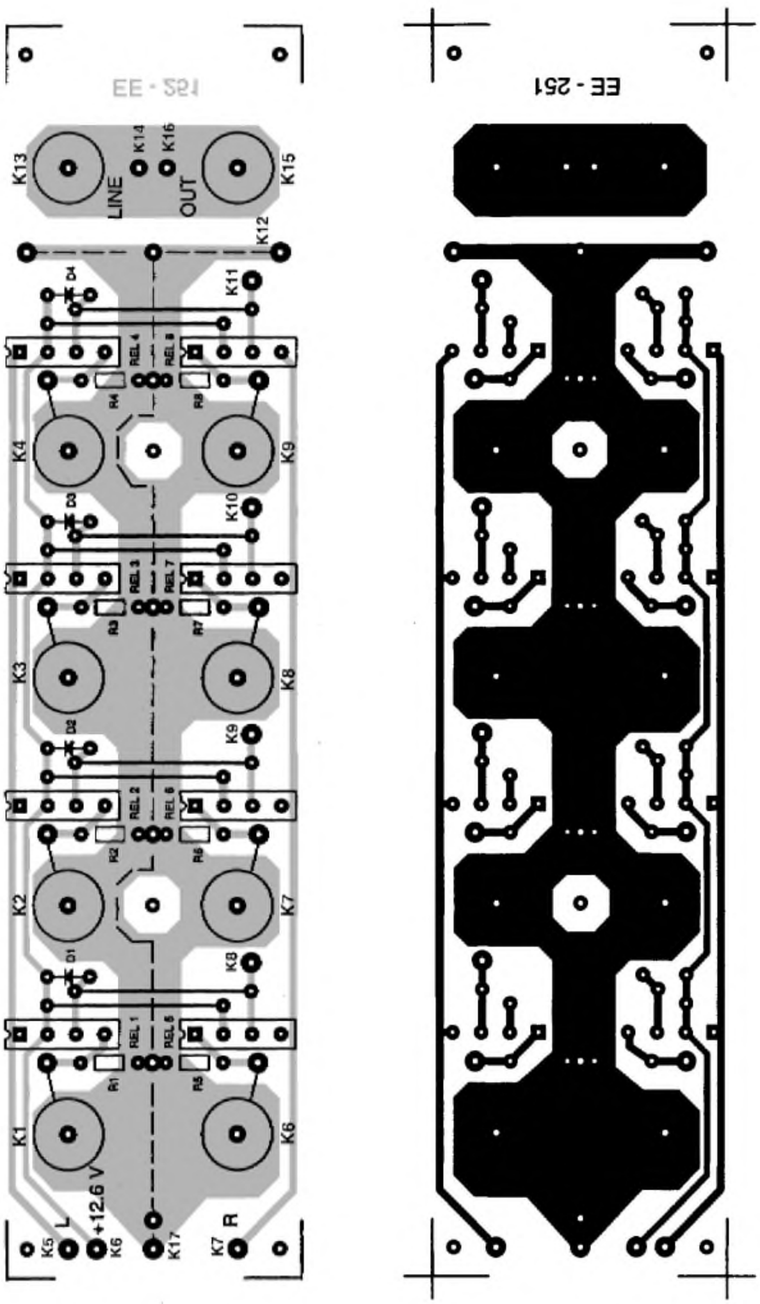

FIGURE 9: Relay board: printed circuit board track layout, component layout and photograph. (Shown at 75%)

G-1829-9

PHOTO 3: Relay board.

cuit board. This diode is only used as an option for other applications in which the amplifier circuit is operated from a higher supply voltage. The good qualities of a stabilized power supply are retained by the low impedance of the zener diode. A high-valued series resistor would only degrade the quality of the supply voltage.

You should pay attention to certain

TABLE 5
COMPONENTS LIST

HIGH VOLTAGE SUPPLY (*FIG. 16*)
Resistors:
R1 = 10kΩ, 2W
R2 = 100Ω, 4.5W
R3, R4 = 1kΩ
R5 = 6Ω8, 2W
R6 = 150kΩ, 2W

Capacitors:
C1 = 100µF 400V, raster 10mm
C2, C3 = 22µF 400V, raster 7.5mm

Semiconductors:
T1 = BUZ92
T2 = BC546
D1, D2, D3 = zener diode 110V, 1.3W
D4 = zener diode 18V, 1.3W
Gl1 = B500C1500, rectangular case (500V PIV, 1.5A peak)

Miscellaneous:
Heatsink for T1: SK 68/50 (Fischer, Dau Components)
Isolation and mounting material for T1 and heatsink
Solder pins

TABLE 6
COMPONENTS LIST

HEADPHONES/LINE SWITCH
K1 = stereo headphones socket, PCB mount
Re1, Re2 = relay, 1 changeover contact, 12V coil (Omron G6E)
R1, R2 = 680kΩ

TABLE 7
COMPONENTS LIST

LOW VOLTAGE SUPPLY (*FIG. 17*)
Capacitors:
C1 = 2200µF 40V, raster 7.5mm
C2 = 100µF 40V, raster 5mm
C3, C4 = 100nF, ceramic, raster 5mm

Semiconductors:
Gl1 = B80C1500, rectangular case (80V PIV, 1.5A peak)
D1 = 1N4148
IC1 = 7812

Miscellaneous:
Isolation material for IC1
Si1, Si2 = fuse, 0.2A, slow, with PCB mount holder
Solder pins

details when inserting the components in the circuit board. In order to make the valves protrude nicely from the chassis—so that they are readily visible—mount the valve sockets on the solder side of the circuit board. This also ensures good heat dissipation. The socket pin assignments are shown in *Fig. 11*.

Power resistors R15 and R16, which become quite warm, are also mounted on the solder side, separated from the board by a certain amount, in order to improve their heat dissipation. It also doesn't hurt to make a series of ventilation holes in the circuit board in unused areas. This avoids the creation of heat pockets. Mount all other compo-

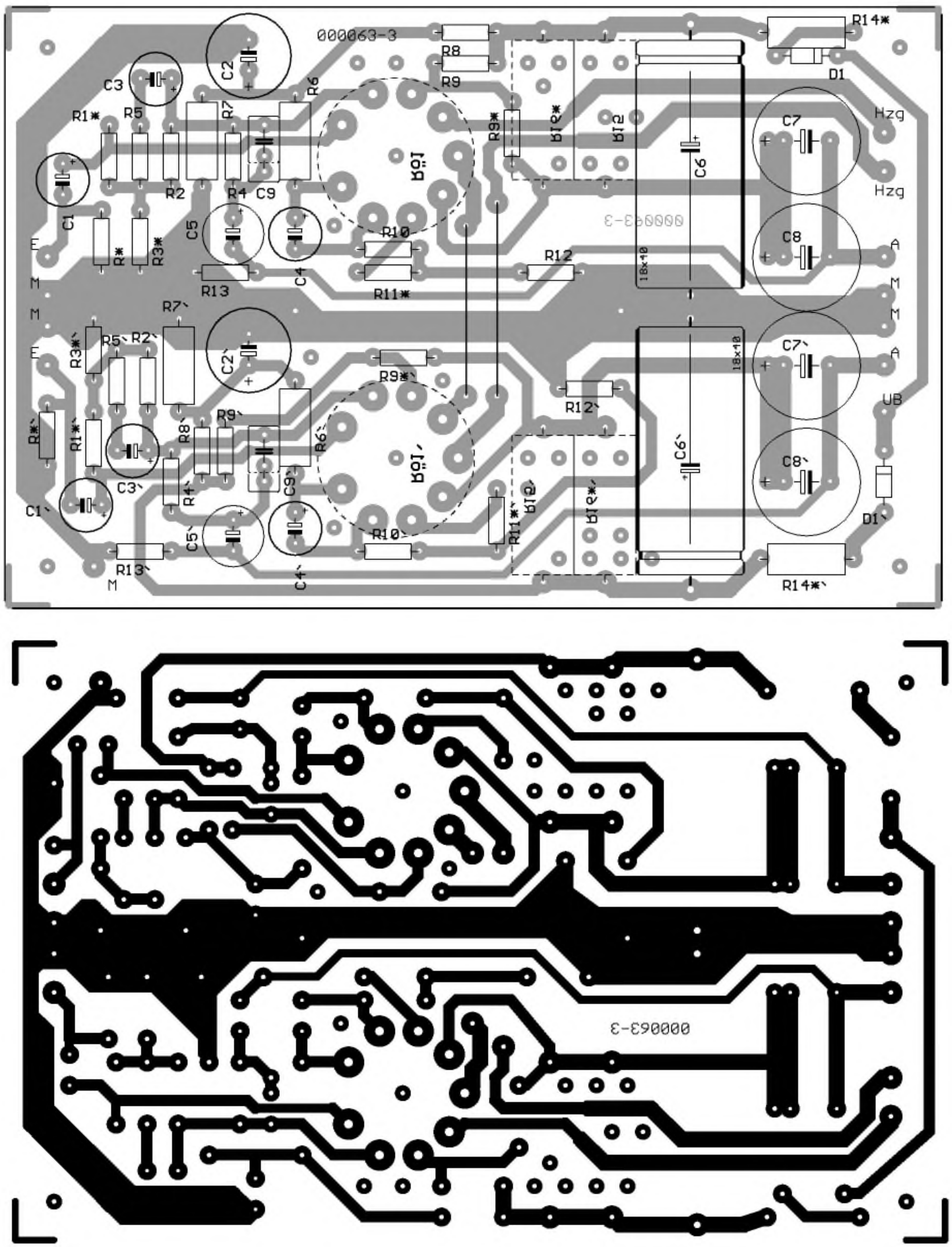

FIGURE 10: The amplifier board is laid out symmetrically, with the two valves in the middle.

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nents on the component side, as usual, and mount the board to the chassis using suitable spacers.

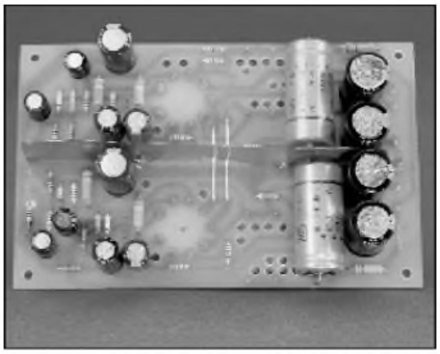

PHOTO 4: Amp circuit board, component side.

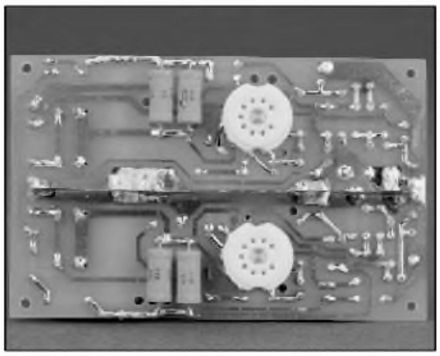

PHOTO 5: Amp circuit board, solder side.

There is not a lot to say about fitting the components to the protection circuit board (*Fig. 12*). You may use sock-

PHOTO 6: ECL86 tubes.

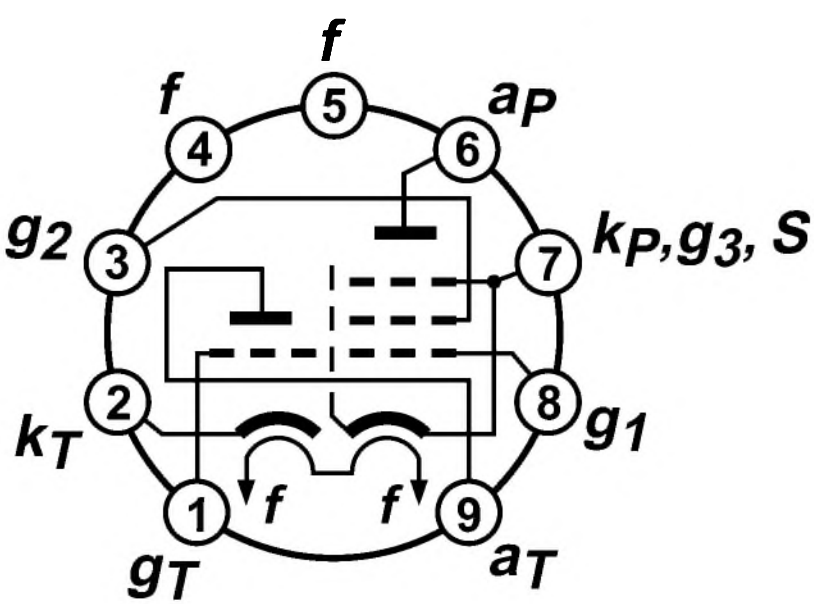

FIGURE 11: Pin assignments of the ECL86, viewing the pins from the bottom—k_P pentode cathode, S internal shield, g_T triode grid, g_2 pentode screen grid, g_1 pentode signal grid, g_3 pentode suppressor grid, f filament, k_T triode cathode, a_P pentode anode, a_T triode anode.

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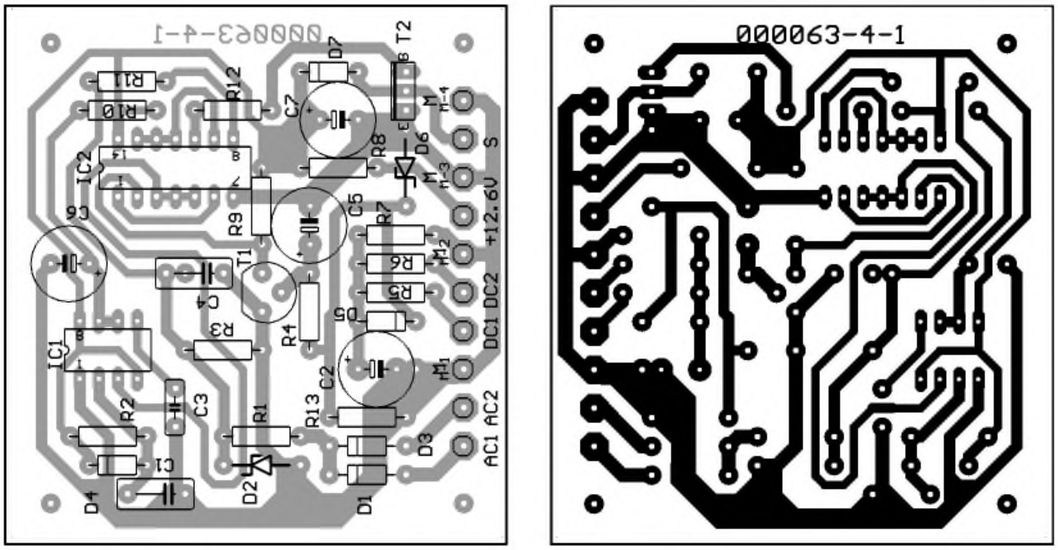

FIGURE 12: Printed circuit boards for the protective circuitry.

G-1829-12

ets for the DIL ICs. If you pay attention (as always) to the correct locations and polarizations of all components, everything will be OK.

MORE BOARD WORK

Next comes the volume control board (*Fig. 13*). *Photo 8* shows how it is built. There is nothing difficult about mounting the components, and you can solidly attach the board to the front panel via the potentiometers, using suitable hardware.

Mount the output relays and the headphone socket on the small circuit board shown in *Fig. 14* and *Photo 9*. The socket is also used to fix the board to the chassis, so no additional fitting hardware is needed. You must pay careful attention to where the signal from the amplifier comes from and where it goes afterwards. The headphone socket is protected only if the wiring is done properly.

In addition, make sure that the relays are correctly polarized, as shown on the component overlay. The selected relays are fully enclosed, which protects the contacts from contamination. In addition, the contacts are gold-plated and rated for 250V operation. No circuit board is available for the optional symmetric power supply. However, perhaps you can use one from other projects.

Now that you have finished building all the circuit boards, you can fit them into the enclosure. If you use the original enclosure, as described previously, all of the necessary holes are pre-drilled, so all you need to do is to fit the hard-

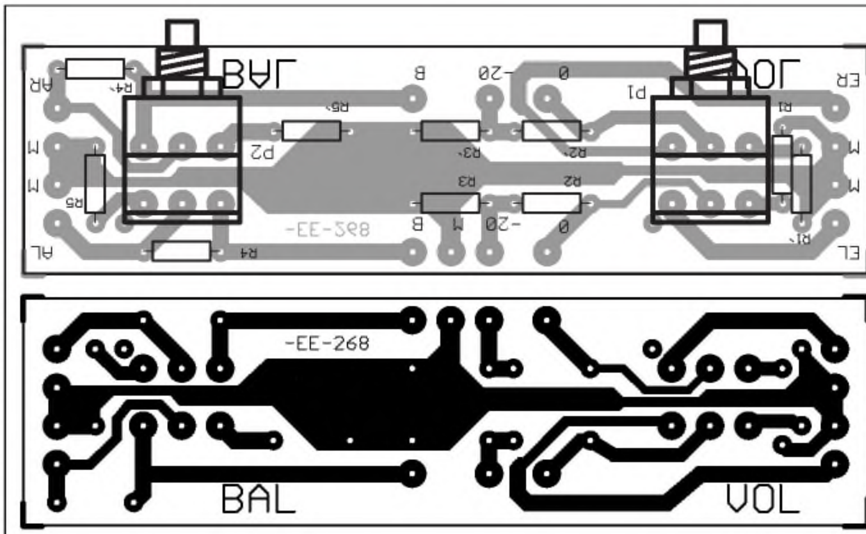

FIGURE 13: The small printed circuit board for the volume and balance controls.

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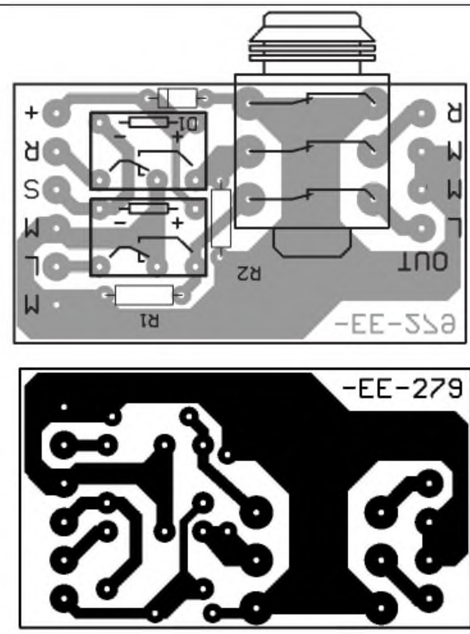

FIGURE 14: The small printed circuit board for the output relays and headphone socket.

G-1829-14

ware for the individual parts (including the mains cable input socket, the power switch, the potentiometers, and the recorder sockets). If you prepare your own enclosure, take care to arrange the individual items in a favorable manner and ensure that the circuit boards are solidly attached. Don't fit the input module right next to the mains transformer, because this will introduce mains hum into the audio signals.

INTERNAL WIRING

Installing the wiring that interconnects the various circuit boards is probably the most complicated task of all. Be sure to use the wiring diagram in *Fig. 15*. You should photocopy this and mark each connection on the copied diagram once you have made the connection. This manner of working has proven to be the most foolproof.

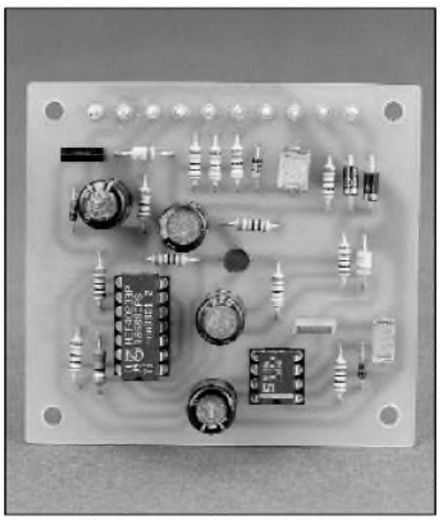

PHOTO 7: Protection circuit board.

All leads that carry audio signals must be made using good-quality screened cable. You may use ordinary light-duty multi-strand wire for the remaining leads, which consist of wiring for the signal source selector switch, the protection circuit, and the power-supply. The wire should have a diameter of 1mm and good insulation. Place the signal wiring and the other wiring so that the two types are separated as far as possible from each other, and fix the cables to the chassis using clips. Connect the recorder output sockets to the common busses for the other circuit boards using wire bridges.

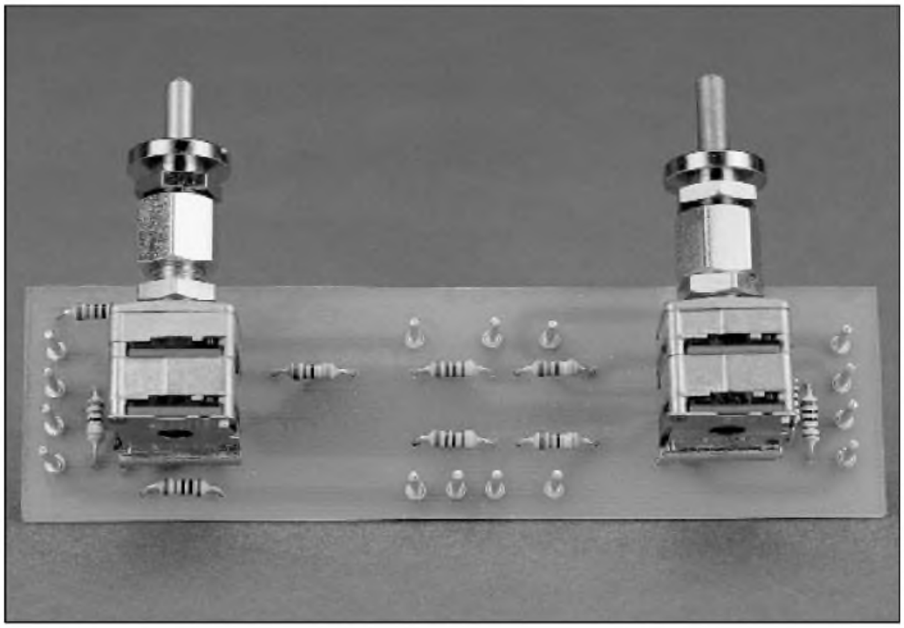

PHOTO 8: Volume control board.

THE MOMENT OF TRUTH

When you have finished the wiring and checked and rechecked everything, you are ready to try the first functional test. First, break the connection between the mains transformer and the high-voltage circuit board (and insulate it against ac-

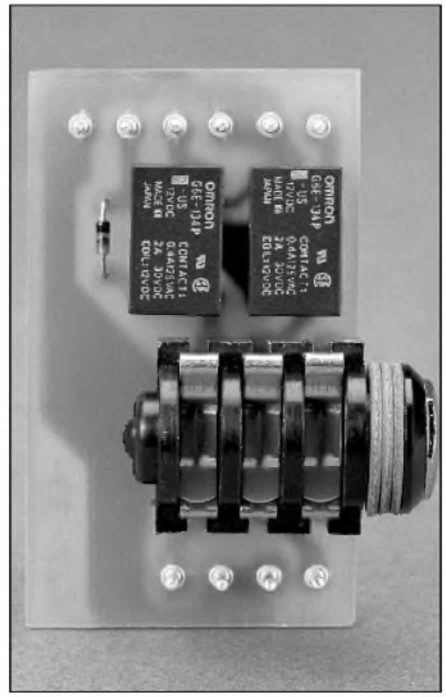

PHOTO 9: PCB for output relays and headphone socket.

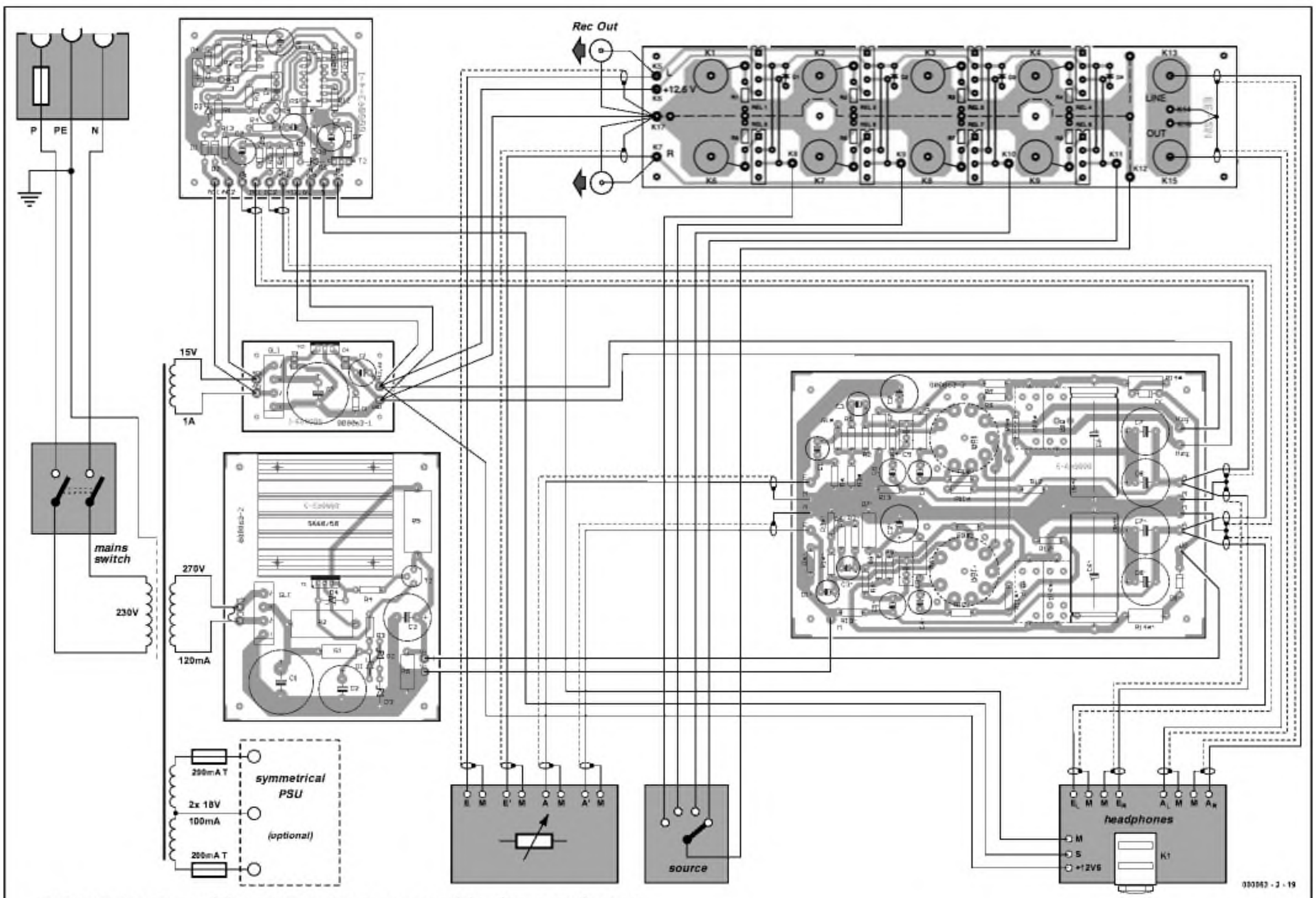

FIGURE 15: The wiring of the valve preamplifier is complicated.

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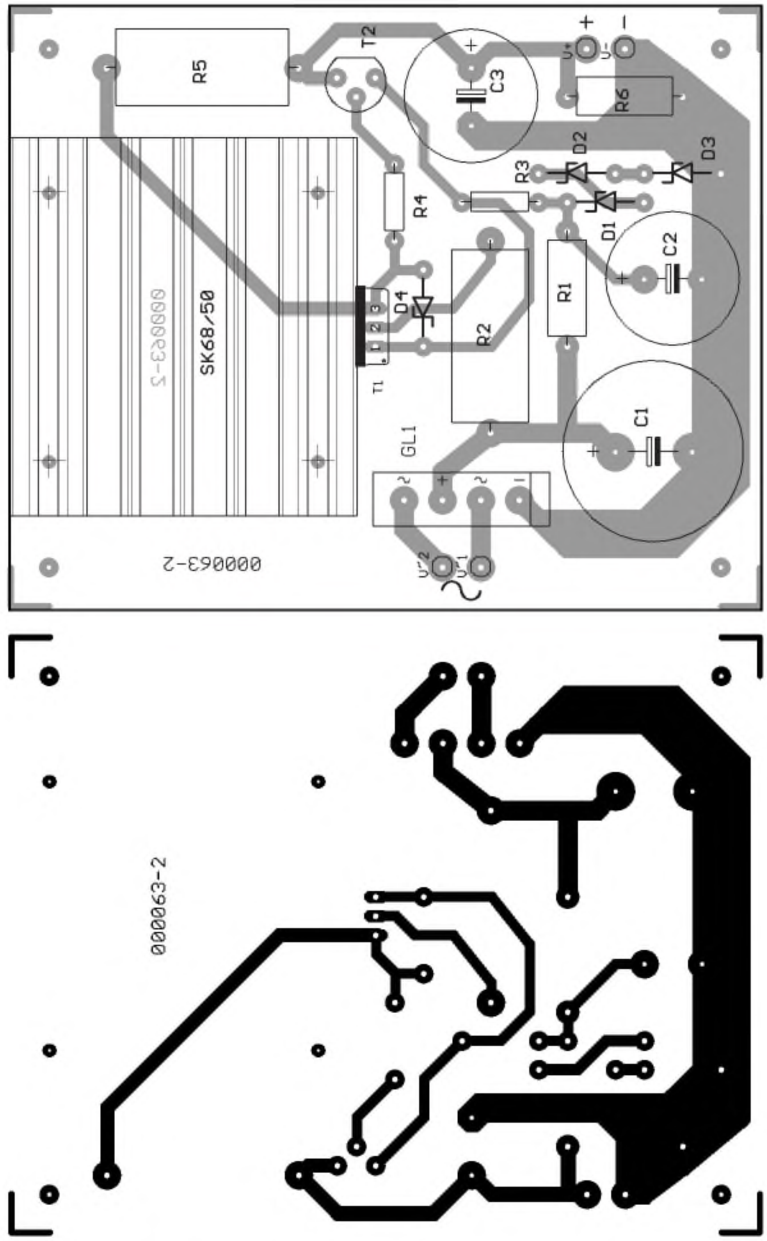

FIGURE 16: The high-voltage power supply.

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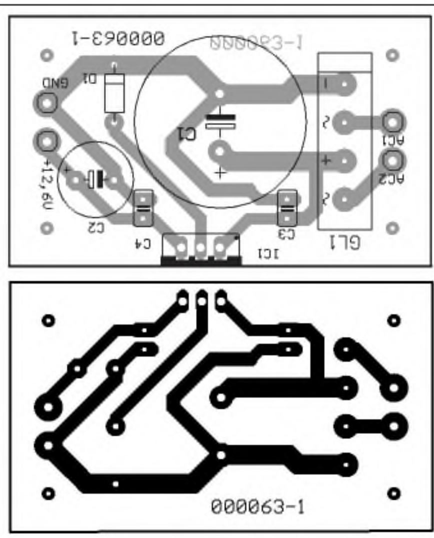
FIGURE 17: The low-voltage power supply.



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MEASURED PERFORMANCE

We made our own measurements of the performance of the valve preamplifier in the Elektor laboratories. The "raw" numbers are listed in the table. All measurements were made after a four-hour warm-up interval, with an effective input voltage of 1V and an effective output voltage of 2V. The outputs were terminated in 10kΩ, and free inputs were terminated in 600Ω. The balance potentiometer was in the middle position.

Note that we made the measurements without any screening of the valves, so RF disturbances from the measurement environment (Elektor lab with PC-based measurement equipment) may have influenced the measured results. If RF interference sources are present in the vicinity of the preamplifier, screens should be provided for the valves.

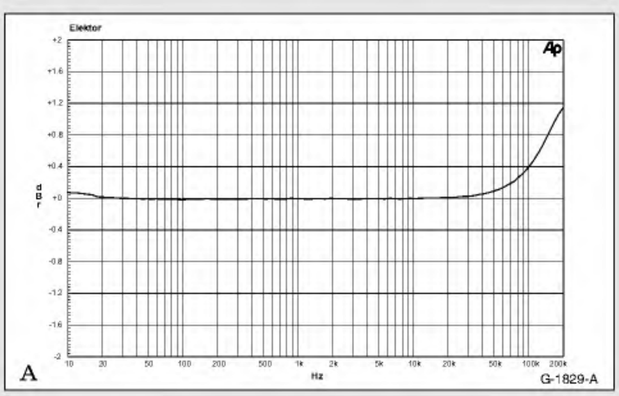
The five measured performance curves (*Figs. A–E*) show the following:

A) Amplitude response
On the strongly enlarged scale up to 200kHz (the upper performance limit of our audio generator), you can see a slight rise in the amplitude response. Within the "interesting" part of the audio frequency band, the curve is dead straight!

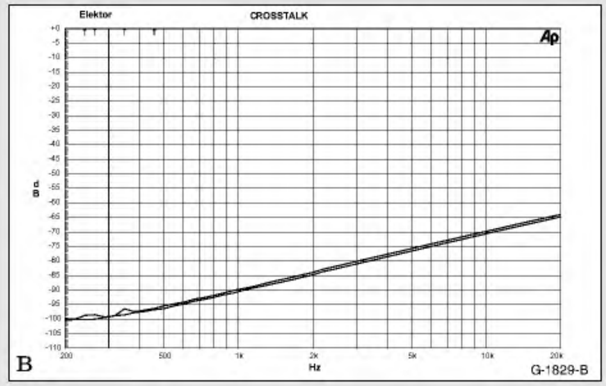
B) Channel separation
The curves are self-explanatory; the two channels track each other very nicely. These curves start at 200Hz, in order to eliminate the effect of power-supply ripple on the measurements.

C) Frequency spectrum
You can see the effect of power-supply ripple in the frequency spectrum. The spectrum of this ripple reaches to around 800Hz. This explains the relatively large difference between the A-weighted and linear measurements (100dBA and 87dB, respectively).

SIGNAL TO NOISE RATIO	A-WEIGHTED	100dBA
	linear, 22Hz–22kHz	87dB
THD plus noise	1kHz, BW = 80kHz	<0.05%
	1kHz, BW = 22Hz–80kHz	<0.01%
	1kHz, BW = 400Hz–22kHz	<0.05%
IMD	(50Hz:7kHz = 4:1)	<0.02%
DIM	(3.15kHz square wave + 15kHz sinusoid)	0.003%
Channel separation	1kHz	84dB
	20kHz	63dB
Crosstalk	1kHz	<–115dB
	20kHz	<–93dB
Input impedance	Minimum volume	6.4kΩ
	Maximum volume	2.1kΩ
Output impedance		<200Ω
Amplification factor		2.54
Bandwidth		3.5Hz–500kHz
Balance adjustment range		+3 to –4.7dB
Attenuator		18dB
THD	Output voltage = 50 V_{eff}	0.1%



A
G-1829-A



B
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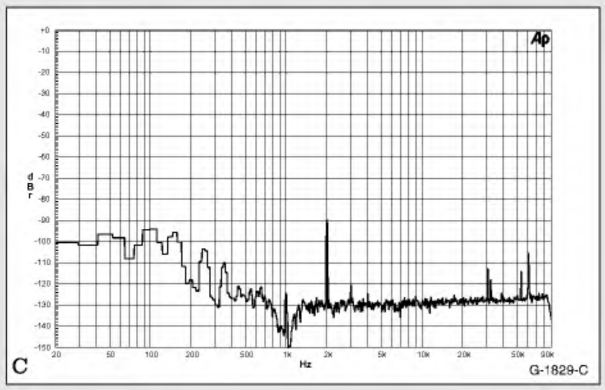
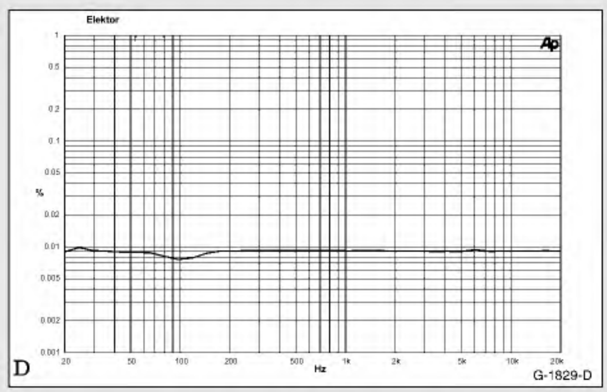
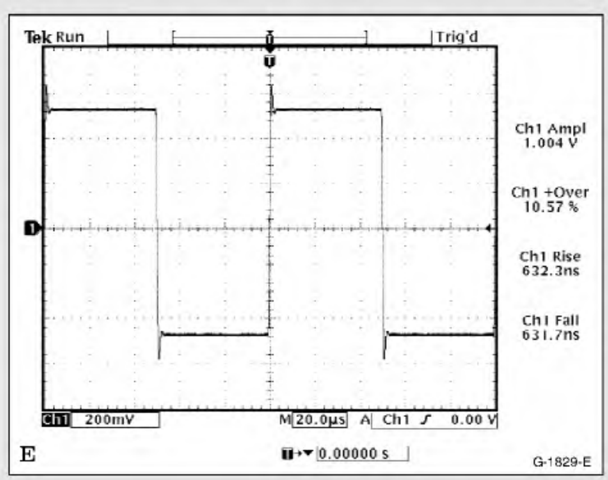


C
G-1829-C



D
G-1829-D



E
G-1829-E

cidental contact). The filament supply must measure 12.6V as soon as it is switched on, and it must be short-circuit proof. This voltage may vary by up to ±5%.

The valve filaments must glow visibly after around one to two minutes. Later on, the filaments will be the "pilot light" for the preamplifier. However, if you want an additional, more distinct power-on indicator, you can simply connect an LED in series with a resistor and 1N4007 diode to the 15V winding of the mains transformer. You can also check the operation of the relays right away.

After this initial test, you can activate the high voltage. It should reach its nominal value shortly after being switched on. If nothing smokes, check the expected values noted on the schematic diagram. After this, use a sine-wave generator and oscilloscope to check the audio paths and functions.

Once you have successfully completed this test, close up the enclosure, connect the preamplifier to signal sources and a final amplifier, and switch everything on. A valve cathode must warm up for two to five minutes before it can emit enough electrons for the valve to be operational. However, you will need to wait around ten to twenty minutes for the valve to be thoroughly warmed up before you will hear the "right" sound. After this, you can relax and enjoy what you hear! ❖

With the 1kHz tone (1 V_{eff}), essentially only the second harmonic is visible, reaching up to –90dB. You can see the influence of interference signals, which in this case come from some old monitors located near the amplifier, with the unscreened valves as peaks in the high-frequency region of the spectrum (30kHz and 60kHz).

D) THD plus noise
We obtained this curve with a bandwidth of 22Hz to 80kHz. The interference signal components come from the effects of power-supply ripple, and probably also from induced signals radiated by the transformer.

E) Step response
A square-wave signal at the input (10kHz, 1V) produces a small overshoot (around 10%) at the output.

Kits, PCBs, and parts are available from
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Software Review

Praxis from Liberty Instruments

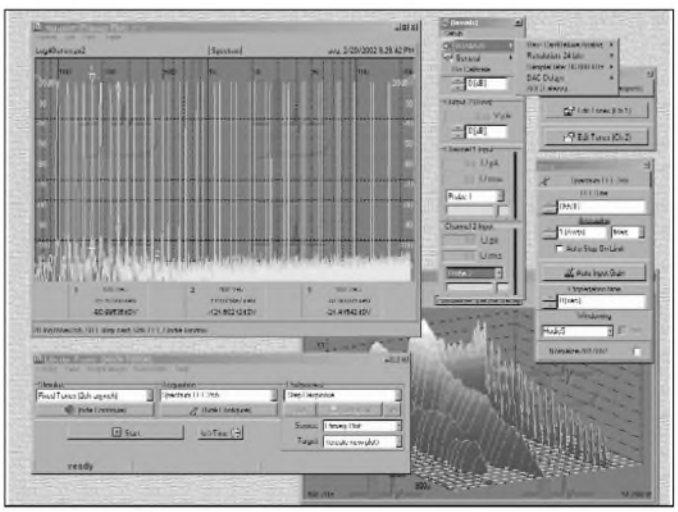
By Vance Dickason

Reprinted from Industry News and Developments, Voice Coil *6/02.*

Liberty Instruments was enjoying success with their IMP and LAUD analyzers until their main supplier of a duplex soundcard that would work with the LAUD analyzer discontinued the product and left the company without a hardware platform. I am pleased to announce that Liberty Instruments is back with a new analyzer, with some significant improvements over their previous products.

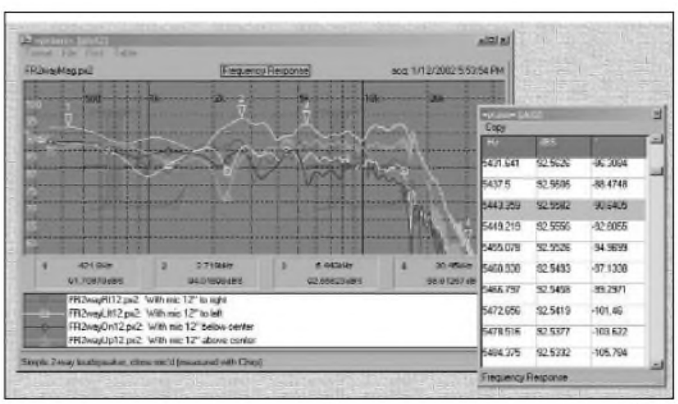
Praxis is a software-based one- and two-channel audio frequency measurement system that operates in a Windows 32-bit operating system (Windows 98SE, WindowsME, Windows2000, WindowsXP). Interface for the computer with microphones and voltage measurement is via the USB-connected AudPod device that provides dual microphone inputs, differential probe inputs, and a calibration source. Sample rates are available to 96kHz, 16-bit and 24-bit. The soundcard must be full duplex, such as the Soundblaster Live (48kHz sample rate) or the CardDeluxe (24bit/96kHz), but the current selection of duplex cards on the market is fairly plentiful. Since the software relies on the host computer CPU for horsepower, Liberty recommends a processor speed of 500MHz and higher.

Praxis has a wide variety of computer generated stimuli to draw from, including logarithmic and linear chirp (swept sine wave), stepped sine wave (gated or continuous at user selectable frequencies or tone burst), MLS, pink filtered MLS, white noise and pink noise, im-



S-dicka-1

FIGURE 1: Praxis multiple menu display.



S-dicka-2

FIGURE 2: Praxis frequency response display.

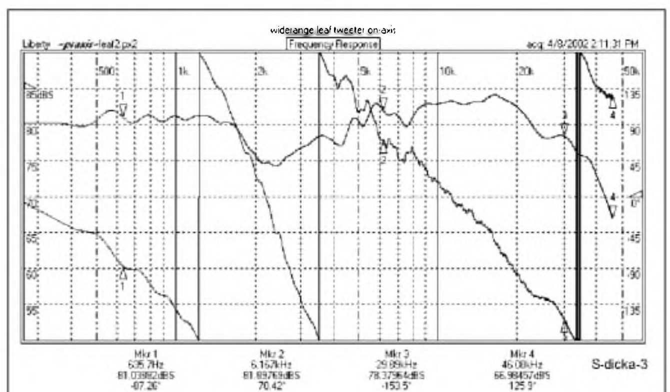
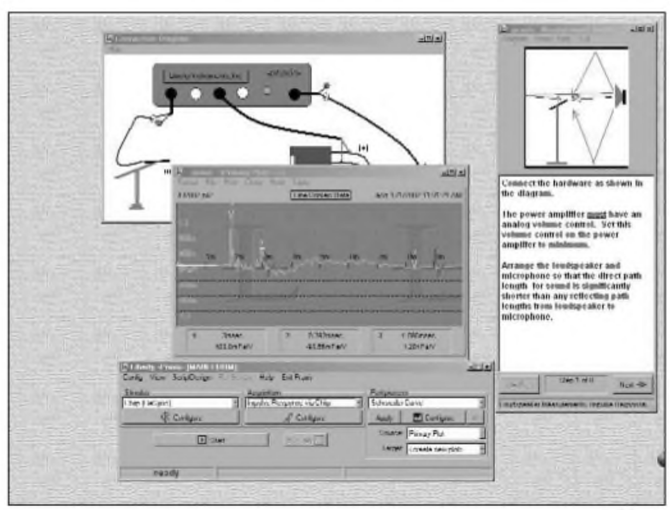
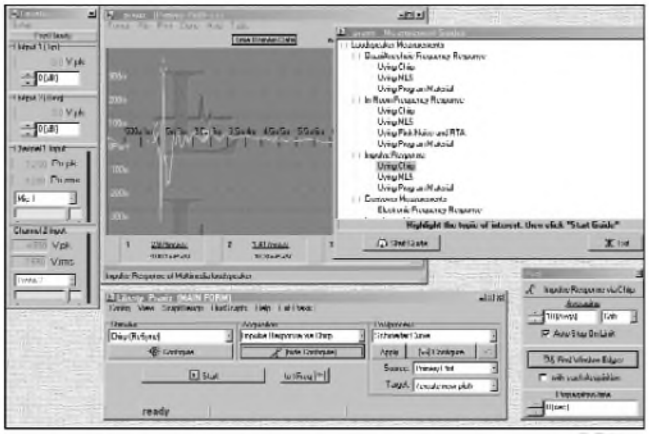


FIGURE 3: Example of Praxis frequency response and phase printout.



S-dicka-4

FIGURE 4: Praxis time domain menu with graphic help screen.



S-dicka-5

FIGURE 5: Praxis impulse response screen showing measurement guide menu.

pulse, live program material, or WAV file. The software provides the following stimulus modes: synchronous (used mostly for burst signals), asynchronous (useful for harmonic distortion or intermodulation distortion measurements), and resynchronous (good for SPL and impedance measurements).

Measurements Praxis supports include time domain (single- or dual-channel), frequency response, impulse response, impedance, distortion (harmonic or multitone), and spectral contamination. Using post processing on these various methods results in FFT/IFFT, step response, smoothing, filtering, math curve conversions (add, subtract, multiply, and divide curves), Schroeder and STI curves, Cumulative Spectral Decay curves (waterfall plots), Thiele/Small parameter measurement, Hilbert transform (phase curve derivation), and group delay.

You can open any number of windows when operating Praxis as you can see on the example screen in *Fig. 1. Figure 2* shows an example of some on- and off-axis measurements made with Praxis, while *Fig. 3* displays an example of a printed frequency response curve. One of the great features of this software for new and inexperienced users is the on-screen measurement "guides" that provide text and graphic step-by-step instruction through some of the measurement processes (*Figs. 4* and *5*). Scripts generated with the Praxis software can also be used to automate procedures for quality control measurements.

Praxis is priced at $980 for the full system, which includes the Praxis software, AudPod USB interface, and calibrated microphone, or $850 without the mike. For a limited time, former users of the LAUD system will receive a $120 discount toward the purchase of the system. You can obtain the Praxis manual and a free download of the software (version 1.16) at the Liberty Instruments website at www.libinst.com. The free download can perform many of the functions, including T/S parameter measurement and RTA monitoring, although the AudPod is necessary to really make use of the instrument. For more, contact Liberty Instruments, PO Box 1454, West Chester, OH 45071, phone/FAX (513) 755-0252, e-mail for sales and pricing is carolst@one.net, and for technical questions is libinst@one.net. ❖

Four ILP Amplifier Modules

By Charles Hansen and John and Sandra Schubel

PHOTO 1: ILP amplifier modules.

ILP Direct LTD, Spong Lane, Elmsted Ashford Kent, UK TN25 5JU, +44 (0) 1233 750481, FAX +44 (0) 1233 750578. Available from audioXpress (www.audioXpress.com). ILP Power Amp Modules with Integral Power Supplies: HY2001, 30W, $4\Omega/8\Omega$, $34.99; HY2003, 60W, $4\Omega/8\Omega$, $52.99; HY2005, 120W, $4\Omega/8\Omega$, $104.99; HY2007, 240W, 4Ω only, $181.99. Dimensions: HY2001, 76mm W × 30mm H × 100mm D, 200g; HY2003, 120mm W × 30mm H × 115mm D, 360g; HY2005, 120mm W × 75mm H × 115mm D, 950g; HY2007, 190mm W × 80mm H × 120mm D, 1.5kg.

This series of monoblock audio power amplifiers with integral heatsinks is available with or without integral power supplies. *Photo 1* shows the four modules that were provided for this review. The toroidal mains transformer requirements are shown in *Table 1*. The data sheets recommend that you use a quick-blow fuse in series with the speaker output, and appropriate sizes are listed for each selected speaker impedance. Appropriate Plitron toroidal power transformer part numbers are listed on the *audioXpress* website.

Although the amplifiers do not need fan cooling or supplemental heatsinking, they must be mounted to allow a vertical flow of air through the heatsink fins. They have thermal protection as well as anti-thump power-on circuitry. They are designed to provide modular power for a variety of audio amplification needs, especially applicable for powered speakers, including subwoofers. Equivalent versions without the integral power supply are also available as HY2000, HY2002, HY2004, HY2006, respectively.

INSIDE THE AMPLIFIERS

Each module consists of a black-anodized heatsink with a fully encapsulated double-sided PTH circuit board, which contains the power-supply rectifier and reservoir filter capacitors. All other electronic components are inside the 20mm deep encapsulated section of the module. Cruciform-shaped slots on each side of the heatsink accommodate the hex nuts used for mounting the heatsinks, and the mounting hardware is provided.

The PC boards have plated-through holes that can accommodate 22-gage wire to make the transformer and input/output connections. An internal 100V filter capacitor (presumably a film type) is provided at the line-level audio input. Additional pads provide access to the V+ and V– DC supply voltages. The input, output, and power-supply center-tap use a common ground.

You can use the three smaller bipolar transistor modules with either 4Ω or 8Ω speakers. A double-sided plated-through pair of PC traces labeled "link" presets the module for 8Ω use. If you want to select the 4Ω mode, you must open the "link" circuit.

The data sheet suggests a spot face cutter or 3–5mm drill for this operation. If you use the drill to remove the plated-through hole, you can later restore the link circuit with a bare wire jumper. The HY2007 MOSFET amplifier is rated only for 4Ω speakers, and does not have the "link" traces.

The 30W HY2001 module (*Photo 2*) uses a 1.5A DC rectifier bridge and a pair of 2200μF 35V filter caps. The 60W HY2003 module (*Photo 3*) uses a 4A DC rectifier bridge and a pair of 4700μF 50V filter caps. The 120W HY2005 module (*Photo 4*) also uses a 4A DC rectifier bridge, while its two 4700μF filter caps are rated for 63V. The 240W HY2007 module (*Photo 5*) uses four discrete 6A diodes and four 4700μF 63V filter caps.

TOPOLOGY

A schematic was not furnished with any of the amplifiers. There is very little technical detail on the data sheets, other than the audio specifications.

The HY2007 is described as a lateral MOSFET design, so I assume the other modules have bipolar output devices, although the 30W rating of the HY2001 makes it possible that it is an audio power IC. The encapsulation makes circuit analysis (as well as repair) impossible. You can easily replace the aluminum capacitors and rectifiers, however.

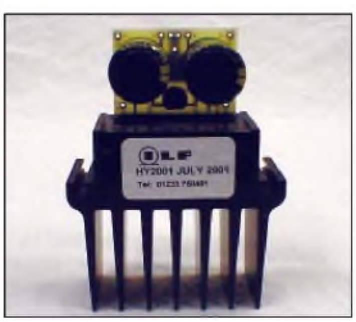


PHOTO 2: HY2001 amplifier.

PHOTO 3: HY2003 amplifier.

PHOTO 4: HY2005 amplifier.

MEASUREMENTS—TEST NOTES

Because the amplifier modules were new, I did not break the link required for 4Ω operation, and all test data is for 8Ω, except the HY2007, which has only 4Ω capability. I used a single toroidal power transformer (40-0-40V, 4.1A) for all the testing, and dropped the secondary voltage to the level specified in the data sheets for the three smaller modules with a Variac™ in the AC primary. I mounted each module on an aluminum chassis test fixture, with hole patterns for each individual module.

HY2001 MEASUREMENTS

I operated the HY2001 at 2W into 8Ω for 1 hour. The initial 0.03% THD reading dropped to 0.018% at the end of this run-in period. There is no noise at all during turn-on or shut-down. The module was quiet, and with my ear against the speaker, I detected no hum and only a very low-level 120Hz buzz. Output hum and noise (input shorted) measured 0.5mV pp, with -0.22V DC offset (more on this later).

The HY2001 does not invert polarity. Input impedance is 100k at 1kHz. The gain at 2.83V RMS output into an 8Ω load is 30.3dB. The output impedance at 1kHz is 0.23Ω, increasing slightly to 0.26Ω at 20kHz.

The frequency response (*Fig. 1*) was within ±1dB from 35Hz–120kHz, at an output of 2.83V RMS at 1kHz into 8Ω. The −3dB points were a rather high 18Hz–235kHz. When I connected a load of 8Ω paralleled with a 2μF cap (a test of compatibility with electrostatic speakers), the HF −3dB gain point dropped all the way to 18.7kHz. There was no evidence of ringing or HF peaking with any load.

The IHF load, which simulates a loudspeaker impedance peak at 50Hz, produced an insignificant 0.2dB higher response compared with the 8Ω resistive load alone. The HY2001 will be insensitive to variations in speaker impedance with frequency.

THD+N versus frequency is shown in *Fig. 2* for the loads indicated at the right side of the graph. During distortion testing, I engaged the test set 80kHz low-pass filter to limit the out-of-band noise. The distortion residual waveform for 5W into 8Ω at 1kHz showed 120Hz ripple overlaid with noise. THD+N at this test point is 0.018%.

When I attempted to run THD+N

(to page 62)

TABLE 1
SPECIFICATION FOR 120V POWER TRANSFORMER SELECTION

MODULE	V sec V RMS	V Supply	I sec A RMS	QB Fuse
HY-2001 (30W, 4Ω)	16-0-16	±20V DC	1.6A	2A
HY-2001 (30W, 8Ω)	20-0-20	±26V DC	1.2A	1.5A
HY-2003 (60W, 4Ω)	20-0-20	±28V DC	2.3A	2.5A
HY-2003 (60W, 8Ω)	28-0-28	±36V DC	1.6A	2A
HY-2005 (120W, 4Ω)	32-0-32	±38V DC	3.3A	3.5A
HY-2005 (120W, 8Ω)	40-0-40	±50V DC	2.3A	2.5A
HY-2007 (240W, 4Ω)	40-0-40	±50V DC	4.6A	5A

TABLE 2
HY2001 MEASURED PERFORMANCE

HY2001 PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Power Output:	40Ω rms (sic) max—4Ω or 8Ω	5W 8Ω steady-state (see text)
Frequency Response:	15Hz–50kHz (–3dB)	18Hz–235kHz ±3dB
Total Harmonic Distortion:	0.005%— 1kHz	0.015%—10W 8Ω
Signal to Noise Ratio (DIN):	100dB	
Slew Rate (typical):	10V/μS	
Rise Time:	5μS	
Input Sensitivity:	500mV rms	
Gain:		30.3dB 8Ω
Input Impedance:	100kΩ	100k
Damping Factor (8Ω, 100Hz)	>400	
Output Impedance:	N/S	0.23Ω 1kHz 0.26Ω 20kHz
Max DC Rails, 8Ω:	±30V DC max	

TABLE 3
HY2003 MEASURED PERFORMANCE

HY2003 PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Power Output:	75W RMS (sic) max—4Ω or 8Ω	58W 8Ω 3%THD
Frequency Response:	15Hz–50kHz (–3dB)	23Hz–500kHz ±3dB
Total Harmonic Distortion:	0.005%—1kHz	0.05%—10W 8Ω
IMD—CCIF (19 + 20kHz):	N/S	0.013% CCIF
MIM (9 + 10.05 + 20kHz):		0.009% MIM
Signal to Noise Ratio (DIN):	100dB	
Slew Rate (typical):	15V/μS	
Rise Time:	5μS	
Input Sensitivity:	500mV RMS	
Gain:		30.0dB 8Ω
Input Impedance:	100kΩ	100k
Damping Factor (8Ω, 100Hz):	>400	
Output Impedance:	N/S	0.20Ω 1kHz 0.24Ω 20kHz
Max DC Rails, 8Ω:	±40V DC max	

TABLE 4
HY2005 MEASURED PERFORMANCE

HY2005 PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Power Output:	150W RMS (sic) max—4Ω or 8Ω	58W 8Ω 3%THD
Frequency Response:	15Hz–50kHz (–3dB)	13Hz–110kHz ±3dB
Total Harmonic Distortion:	0.005%—1kHz	0.03%—10W 8Ω
IMD—CCIF (19 + 20kHz):	N/S	0.02% CCIF
MIM (9 + 10.05 + 20kHz):		0.013% MIM
Signal to Noise Ratio (DIN):		100dB
Slew Rate (typical):	20V/μS	
Rise Time:	5μS	
Input Sensitivity:	500mV RMS	
Gain:		32.6dB 8Ω
Input Impedance:	100kΩ	100k
Damping Factor (8Ω, 100Hz):	>400	
Output Impedance:	N/S	0.13Ω1kHz 0.16Ω 20kHz
Max DC Rails, 8Ω:	±55V DC max	

TABLE 5
HY2007 MEASURED PERFORMANCE

HY2007 PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Power Output:	300W RMS (sic) max—4Ω	180W 4Ω 3%THD
Frequency Response:	15Hz–50kHz (–3dB)	15Hz–85kHz ±3dB
Total Harmonic Distortion:	0.005%—1kHz	0.11%—10W 4Ω
IMD - CCIF (19 + 20kHz):	N/S	0.025% CCIF
MIM (9 + 10.05 + 20kHz):		0.011% MIM
Signal to Noise Ratio (DIN):		100dB
Slew Rate (typical):	50V/μS	
Rise Time:	5μS	
Input Sensitivity:	500mV RMS	
Gain:		36.3dB 4Ω
Input Impedance:	100kΩ	100k
Damping Factor (8Ω, 100Hz):	>400	
Output Impedance:	N/S	0.08Ω 1kHz 0.19Ω 20kHz
Max DC Rails, 8Ω:	±60V DC max	

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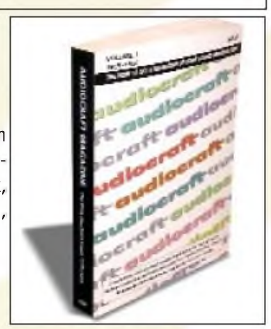

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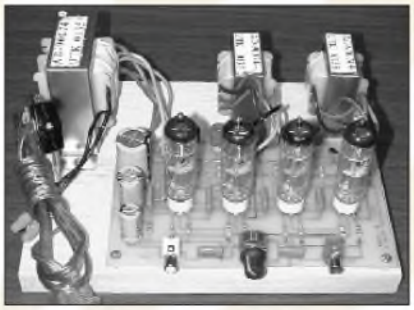

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(from page 59)
versus output power, I ran into a problem. Above 5W load, the DC offset began to increase with time and temperature. Initially –0.22V, it turned positive and rose toward the +DC rail until it reached +6V DC. After ten minutes at 5W AC, the output dropped to 3W and V offset dropped to +2.8V DC, while the upper half-cycles collapsed from +14V pk to +10V pk and THD increased to 10%. The heatsink temperature only reached 49°C.

I monitored the signal at the Link PC trace of the HY2001 module, assuming it was located in the feedback loop. Initially it measured 14mV RMS, and out-of-phase with the 190mV input signal. As the output DC offset increased in the positive direction, the 14mV AC signal took on an increasingly negative DC offset voltage. I don't know the topology of this module, and don't know where in a non-inverting feedback amplifier circuit the signal could be out of phase with the input and output.

As I increased the power toward the rated 30W, the sequence happened again, only faster. At near full-power the output clipped very hard before folding back. At no time did the recommended 1.5A speaker protection fuse blow. Perhaps this problem was due to leakage in some internal component, but I wouldn't use these modules without adding some form of DC offset protection.

At this point I terminated testing on the HY2001. What limited THD versus Power data I did obtain is shown in *Fig. 3*.

I normally measure the 1kHz product of the test amplifier by reproducing a combined 19kHz + 20kHz CCIF intermodulation distortion (IMD) signal at 12V pp into 8Ω. Then I repeat the test with a 9kHz + 10.05kHz + 20kHz multitone IMD (MIM) signal. However, the HY2001 would not maintain a stable steady-state signal long enough for the spectrum analyzer to capture the data.

A 2.5V p-p square wave into 8Ω at 40Hz showed considerable tilt (*Fig. 4*). This corresponds to the early rolloff in bass response measured in *Fig. 1*. The 1kHz and 10kHz square waves (not shown) were almost perfect. The leading edge of the 10kHz square wave was

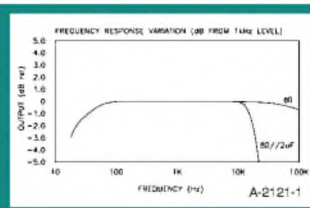


FIGURE 1: HY2001 frequency response.

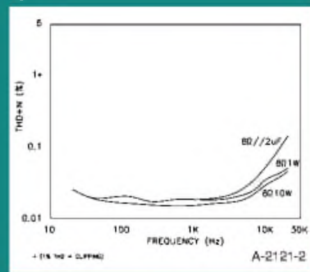


FIGURE 2: HY2001 THD+N versus frequency.

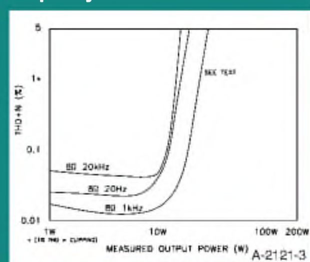


FIGURE 3: HY2001 THD+N versus output power.

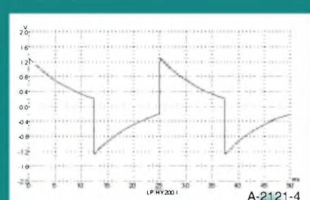


FIGURE 4: HY2001 square-wave response, 40Hz 2.5V pp 8Ω.

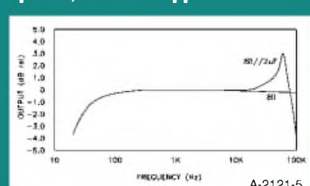


FIGURE 5: HY2003 frequency response.

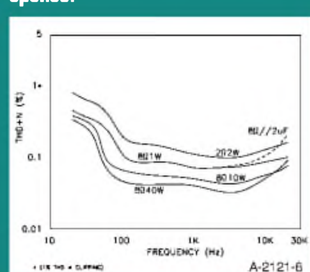


FIGURE 6: HY2003 THD+N versus frequency.

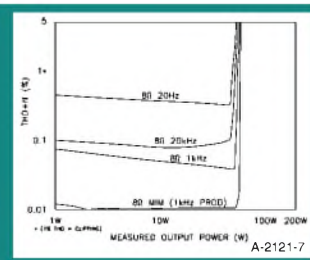


FIGURE 7: HY2003 THD+N versus output power.

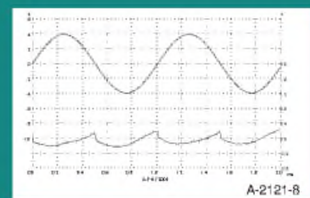


FIGURE 8: HY2003 residual distortion.

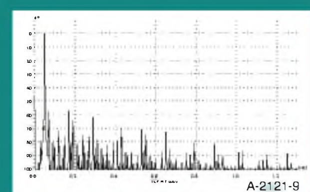


FIGURE 9: HY2003 spectrum of 50Hz sine wave.

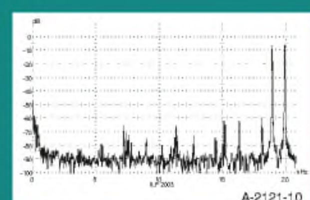


FIGURE 10: HY2003 spectrum of 19kHz + 20kHz intermodulation signal.

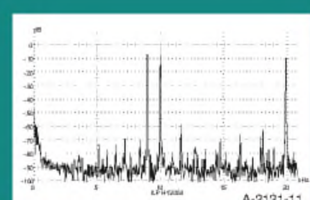


FIGURE 11: HY2003 spectrum of 9kHz + 10.05kHz + 20kHz intermodulation signal.

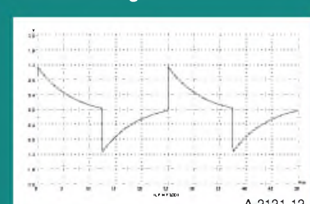


FIGURE 12: HY2003 square-wave response, 40Hz 2.5V pp 8Ω.

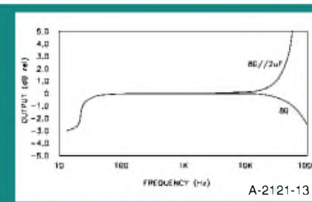


FIGURE 13: HY2005 frequency response.

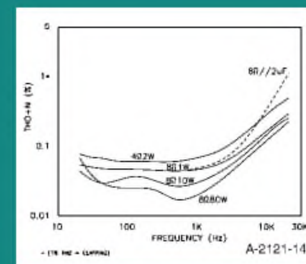


FIGURE 14: HY2005 THD+N versus frequency.

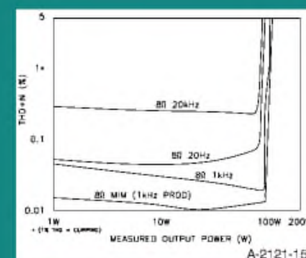


FIGURE 15: HY2005 THD+N versus output power.

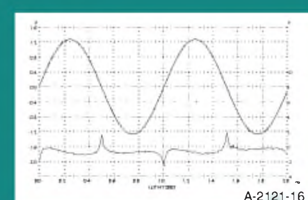


FIGURE 16: HY2005 residual distortion.

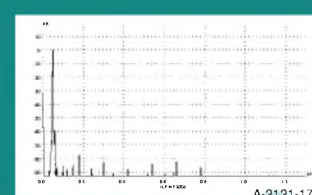


FIGURE 17: HY2005 spectrum of 50Hz sine wave.

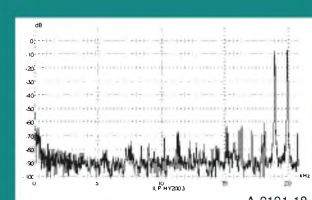


FIGURE 18: HY2005 spectrum of 19kHz + 20kHz intermodulation signal.

only slightly rounded. When I connected 2µF in parallel with the 8Ω load the amplifier was unfazed.

HY2003 MEASUREMENTS

I operated the HY2003 at 10W into 8Ω for 1 hour. Its initial 0.05% THD remained unchanged at the end of this run-in period. There is no noise at all during turn-on or shut-down. The module was quiet, and with my ear against the speaker, I detected only a very low-level 60Hz hum. Output hum and noise (input shorted) measured 7mV pp, with +2mV DC offset.

The HY2003 does not invert polarity. The signal at the "Link" point of the PC board was 1mV with a 128mV input signal, and in phase with the input. The input impedance is 100k at 1kHz. The gain at 2.83V RMS output into an 8Ω load is

CRITIQUE

By John and Sandra Schubel

We received the ILP amplifiers individually wrapped in bubble wrap and packed each in its own cardboard box. Technical specifications followed separately in the mail. As noted in his review, Charles Hansen mounted two of the units—the ILP HY2003 and the ILP HY2005—on a chassis for the listening test.

SETUP

We used an HHB CDR-800 (reviewed in *Audio Electronics* 2/00) to play selections from *Hi-Fi News and Record Review's* CD Test Disk III, as well as selections from our private collection. The CDR-800 was directly connected to the ILP amplifiers using a passive gain control. Because only one sample of each amplifier was provided for review, we used the 120W unit for the right channel and the 60W unit for the left. Each amplifier drove a NHT SL-2 subwoofer connected to a NHT Model 1.3 via the subwoofer's internal crossover network.

Because the amplifiers as provided were set for 8Ω speakers, and the NHT speaker combination presents approximately 8Ω nominal, 3.7Ω minimum to the amplifier, we repeated some listening tests using the 8Ω rated Model 1.3 alone. We used an NAD Stereo Power Amplifier 214 as a test reference, and because louder is often perceived as better, we used a Realistic sound level meter and a 1000Hz test signal to ensure that the sound level from each amplifier was the same, at least at 1000Hz.

Both ILP amplifiers produced a faint hum audible only when we listened within inches of the speakers in a quiet room. This was true both with and without inputs connected. When, for the reviewer's convenience, the distance between the passive gain control and the amplifier was increased by substituting a 12′ RCA cable set manufactured by Monster Cable, the hum and buzz increased to levels easily audible from the middle of the room.

Making sure that all cables were separated and—to the extent possible—perpendicular to each other, we moved the amplifier away from the wall to minimize any potential for hum pickup from house wiring, with no effect on the hum and buzz. Grounding the cable center conductors to the shield also had no effect. We then tried the same 12′ cable configuration with the NAD amplifier, and there was no audible hum or buzz. Rather than pursue the problem further, we decided to use 3′ cables for subjective testing.

ABOUT THE AUTHORS

Sandra Schubel has studied Voice and Piano, and has performed with various little theater companies in New Jersey. She also has an MBA in Marketing and International Business, and heads a management consulting firm. John Schubel is a telecommunications consultant with degrees in Electrical and Electronic Engineering. He has been a long-time audio enthusiast, dabbling in audio projects as career and family permit. On Sunday mornings, John and Sandra can be found singing in their church choir.

Our first reaction to the ILP amplifiers was that the performers were distant, coming from behind the speakers. Cymbals sounded fuzzy, complex sounds were muddied, and bass was solid but lacked crispness. We then burned in the amplifiers for about 24 hours, and the sound quality improved noticeably. The sound was now much cleaner, but still did not give the listener a sense of intimacy with the performance.

We listened to the ILP amplifiers for another two weeks, and found them easy to listen to, even with large orchestral pieces. We remark upon this because by comparison, our Sony DA555ES sounds better on first encounter, but becomes tiring to listen to after about the fifth playing of the overture to Leonard Bernstein's "Candide." The bottom line is that the ILP amplifiers as tested perform respectably, especially considering their low cost.

TESTING

We sampled selections from "Hi-Fi News & Record Review's CD Test Disk III" to gain further insight into our perceptions. The Jerusalem/Parry track provided a musically complex test for the amplifiers. The sections of the chorus could be clearly distinguished and placed in the sound field. The accompaniment did not fare as well: the horns, violins, and percussion seemed distant. You had to strain to hear the harp.

Our impression was that the performance sounded farther away from the listener. In the beginning, where men have a prominent part, their sound was not full and rounded, and resonances of the hall were absent. When the sopranos soared later in the piece, the sound remained subdued and the highlights of their voices were missing.

Similarly, the horns were muted and the bowing of the violins not crisp. The highlights of the cymbals were also missing. In general, the sound of the chorus as reproduced by the ILP amplifiers was not particularly pleasing, and the performance placed the listener back under the balcony.

The Vivaldi Trumpet Concerto provided a revealing test. On the surface, the trumpets were well separated and you could clearly place each trumpet on the soundstage. The violins, however, were not as sweet, and the harpsichord disappeared from perception.

It was only by comparison with the NAD amplifier that we realized how dramatic was their absence. Even the silence as the last echoes die down had a different quality. This track provided perhaps the most revealing information about the amplifier's ability to recreate the soundstage.

Peter and the Wolf provided another interesting point of comparison. Again the ILP amplifiers placed the listener farther back from the performance, with a corresponding lack of detail. What was surprising was that the bass, while not as tight as with the NAD amplifier, gave the listener the perception of resonances below those revealed by the NAD! The triangles were not as crisp and some of the highlights of the bassoon were more subdued. The individual instruments did not stand out with clarity.

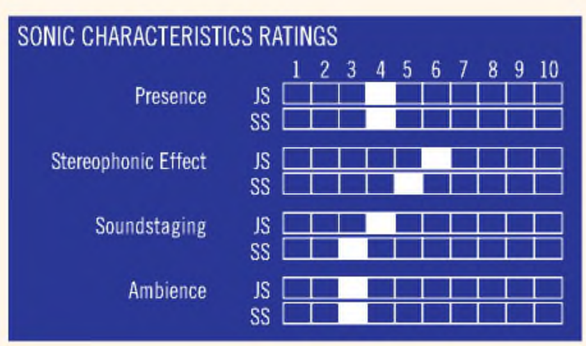
The overall sound is pleasing, but again we are sitting further back in the concert hall. We listened to this phenomenon many times without definitive explanation. The subjective review of this amplifier was deliberately done without access to the results of the technical analysis. Overall, Peter and the Wolf was still a very satisfying performance on the ILPs.

The Corkhill percussion selections provided a dramatic demonstration of the ILP's reproduction of the soundstage. When experienced with the NAD amplifiers we seemed close to the drums, as if the setting were a small club. When reproduced through the ILP amplifiers, the sense of setting was absent. The drums sounded distant, and lacked the percussive quality and snap heard on the NAD system.

A personal favorite listening selection is Leonard Bernstein's overture to "Candide," recorded by the Los Angeles Philharmonic, Leonard Bernstein conducting (Deutsche Grammophon 427 042-2). Again, the ILT amplifiers were quite listenable, but lacked the smoothness of sound and the detail experienced with the NAD amplifier. Instruments tended to blend together. The triangles lacked clarity and sounded distant, as experienced in the Vivaldi Trumpet Concerto. The individual instruments were not well defined. By comparison, the NAD accurately separated the sound of the various instruments.

Much of any listening test is subjective, and the listeners' preferences may in some cases make the ILP amplifiers preferable. Sandra, for example, has never come to appreciate the harpsichord and, while listening to the Vivaldi Trumpet Concerto, was gratified that the harpsichord was "not as annoying as it should be." If you prefer to sit further back in the concert hall, this amplifier works well.

John, by contrast, doesn't mind sitting in the fifth row, looking up at the soundboard of the piano. If this is your preference, the ILP amplifiers are probably not for you.



30dB. The output impedance at 1kHz is 0.20Ω, and 0.24Ω at 20kHz.

The frequency response (*Fig. 5*) was within ±1dB from 43Hz–425kHz, at an output of 2.83V RMS at 1kHz into 8Ω. The −3dB points were even higher than those of the HY2001, at 23Hz and 500kHz. When I connected a load of 8Ω paralleled with a 2μF cap, the HF −3dB gain point dropped to 94kHz, and reached a HF peak at 60kHz, which was 3.1dB above the 1kHz response. At this point the 2A FB speaker protection fuse opened.

The IHF load, which simulates a loudspeaker impedance peak at 50Hz, produced an insignificant 0.13dB higher response compared with the 8Ω resistive load alone. The HY2003 will be insensitive to speaker impedance variations.

THD+N versus frequency is shown in *Fig. 6* for the loads indicated on the graph. During distortion testing, I engaged the test set 80kHz low-pass filter to limit the out-of-band noise.

Figure 7 shows THD+N versus output power into 8Ω for various frequencies. At about 50W, the 20Hz sine wave had some scoop-out just after the peaks of each half cycle, and the 20kHz sine wave showed some trailing peak HF oscillation. The amplifier didn't quite reach its rated 60W, and clipped very hard and blew the speaker protection fuse when driven above 3% distortion. The heatsink was never too hot to touch.

I also plotted the 1kHz product of the multi-tone intermodulation (MIM) 9kHz + 10.05kHz + 20kHz test signal versus output power. This gives a better indication of the amplifier's nonlinear response, because it is a closer approximation to music than a sine wave.

The distortion residual waveform for 10W into 8Ω at 1kHz is shown in *Fig. 8*. 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 consists of obvious crossover distortion, overlaid with noise. Expanding the time scale showed a significant kick in the residual signal at the peak of each 60Hz half-cycle as well. THD+N at this test point is 0.05%.

The spectrum of a 50Hz sine wave at 10W into 8Ω is shown in *Fig. 9*, from zero to 1.3kHz. The 2nd, 3rd, 4th, and 5th harmonics measure −85dB, −76dB, −97dB, and −90dB, respectively. However, significant (−58dB) power-supply artifacts are present at 60Hz and 120Hz, along with the odd harmonics of 60Hz. Additional non-harmonic signals (70Hz, 170Hz, 290Hz, and so on) are present throughout the spectrum. The THD+N measures 0.24%, with the true 50Hz harmonics contributing only 0.017%.

Figure 10 shows the amplifier output spectrum reproducing a combined 19kHz + 20kHz CCIF intermodulation distortion (IMD) signal at 12V pp into 8Ω. The 1kHz IMD product is 0.013%. Repeating the test with a multi-tone IMD signal (9kHz + 10.05kHz + 20kHz, shown in *Fig. 11*) resulted in a 1kHz product of 0.009%. (I plotted MIM distortion versus output in *Fig. 7*.)

The 2.5V p-p square wave into 8Ω at 40Hz showed considerable tilt as you can see in *Fig. 12*, returning to the zero line at the end of each half-cycle. This is due to the early low frequency rolloff designed into the amplifier (*Fig. 5*). The 1kHz and 10kHz square waves (not shown) were almost perfect. When I connected 2μF in parallel with the 8Ω load, there was significant ringing in the 1kHz and 10kHz square waves at about 60kHz, which correlates with the peaking seen in the frequency-response test.

HY2005 MEASUREMENTS

I operated the HY2005 at 10W into 8Ω for 1 hour. The THD throughout this run-in period remained at 0.026%. There is no noise at all during turn-on or shut-down, and a

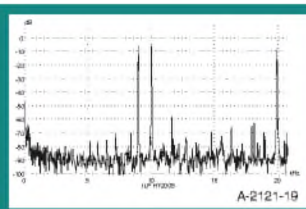


FIGURE 19: HY2005 spectrum of 9kHz + 10.05kHz + 20kHz intermodulation signal.

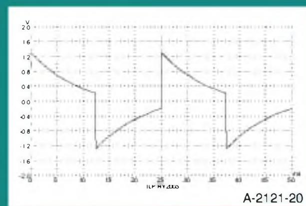


FIGURE 20: HY2005 square-wave response, 40Hz 2.5V pp 8Ω.

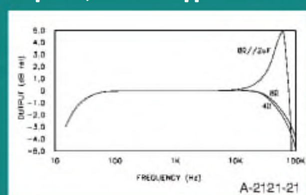


FIGURE 21: HY2007 frequency response.

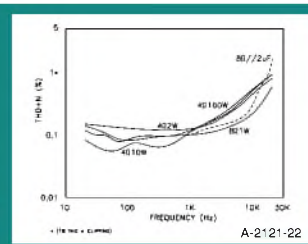


FIGURE 22: HY2007 THD+N versus frequency.

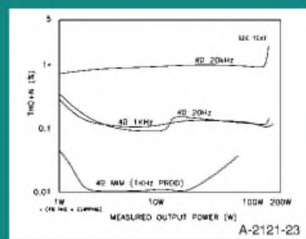


FIGURE 23: HY2007 THD+N versus output power.

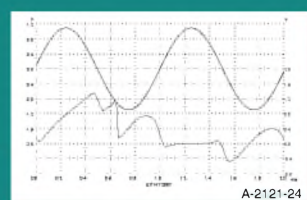


FIGURE 24: HY2007 residual distortion.

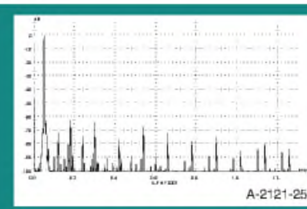


FIGURE 25: HY2007 spectrum of 50Hz sine wave.

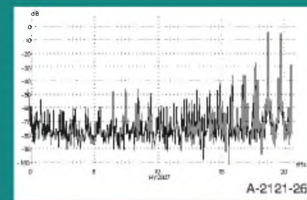


FIGURE 26: HY2007 spectrum of 19kHz + 20kHz intermodulation signal.

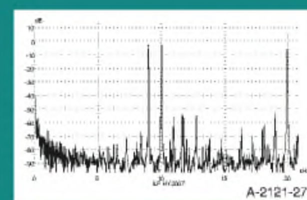


FIGURE 27: HY2007 spectrum of 9kHz + 10.05kHz + 20kHz intermodulation signal.

PHOTO 5: HY2007 amplifier.

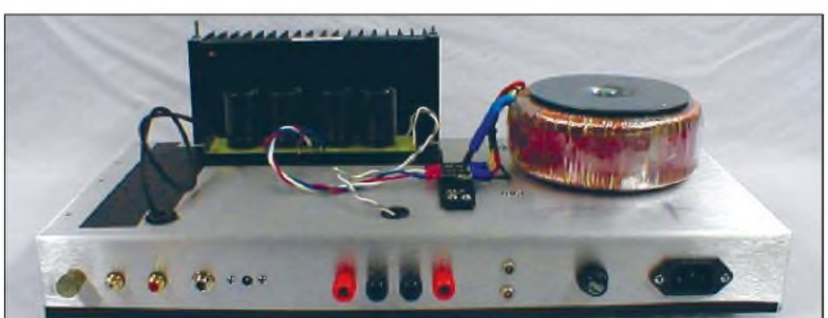


PHOTO 6: ILP amplifier module test fixture.

relay in the potted block operates at about half the AC line voltage. The module was quiet, and with my ear to the speaker, I detected only a low-level 60Hz hum. Output hum and noise (input shorted) measured 0.6mV pp, with +4mV DC offset.

The HY2005 does not invert polarity. The signal at the "Link" point of the PC board measured 73mV with a 200mV input signal, and it was in phase with the input. The input impedance is 100k at 1kHz. The gain at 2.83V RMS output into an 8Ω load is 32.6dB. The output impedance at 1kHz is 0.13Ω, increasing to 0.16Ω at 20kHz.

The frequency response (*Fig. 13*) was within ±1dB from 24Hz–57kHz, at an output of 2.83V RMS at 1kHz into 8Ω. The −3dB points were 13Hz and 110kHz. When I connected a load of 8Ω paralleled with a 2µF cap, the HF gain continued to increase with frequency, with the speaker protection fuse opening with celerity at 70kHz.

The IHF load, which simulates a loudspeaker impedance peak at 50Hz, produced an insignificant 0.12dB higher response compared with the 8Ω resistive load alone. The HY2005 will be insensitive to speaker impedance variations.

THD+N versus frequency is shown in *Fig. 14* for the loads indicated on the graph. During distortion testing, I engaged the test set 80kHz low-pass filter to limit the out of band noise.

Figure 15 shows THD+N versus output power for various frequencies. As with the other ILP modules, this amplifier also didn't reach its power rating, in this case 120W. At 3% THD+N, it produced 103W at 1kHz, 94W at 20Hz, and 88W at 20kHz. The 20kHz sine wave showed some HF oscillation just after each peak of the waveform. Its clipping was a bit softer than with the HY2001 and HY2003. The heatsink was never too hot to touch.

I also plotted the 1kHz product of the multi-tone intermodulation (MIM) 9kHz + 10.05kHz + 20kHz test signal versus output power. This gives a better indication of the amplifier's nonlinear response, since it is a closer approximation to music than a sine wave.

The distortion residual waveform for 10W into 8Ω at 1kHz is shown in *Fig. 16*. 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 again consists of obvious crossover distortion, overlaid with noise. THD+N at this test point is 0.03%.

The spectrum of a 50Hz sine wave at 10W into 8Ω is shown in *Fig. 17*, from zero to 1.3kHz. The 2nd, 3rd, 4th, and 5th harmonics were very similar to the HY2003, at −85dB, −85dB, −97dB, and −91dB, respectively. Again, a significant −58dB power-supply peak is evident at 60Hz, with a lower peak at 180Hz. However, the broad spectrum of higher frequencies seen in the HY2003 are absent. The THD+N at this load measures 0.03%, with the true 50Hz harmonics contributing only 0.0085%.

Figure 18 shows the amplifier output spectrum reproducing a combined 19kHz + 20kHz CCIF intermodulation distortion (IMD) signal at 12V pp into 8Ω. The 1kHz IMD product is 0.02%. Repeating the test with a multi-tone IMD signal (9kHz + 10.05kHz + 20kHz) resulted in a 1kHz product (*Fig. 19*) of 0.013%.

The 2.5V p-p square wave into 8Ω at 40Hz showed significant tilt (*Fig. 20*). The 1kHz square wave (not shown) was almost perfect. The 10kHz square wave showed some peaking at its leading edges. When I connected 2µF in parallel with the 8Ω load, there was significant ringing in the 1kHz and 10kHz square waves at about 70kHz, which correlates with the peaking seen in the frequency-response test.

HY2007 MEASUREMENTS

I operated the HY2007 at 10W into 4Ω for 1 hour. The initial 0.09% THD reading dropped to 0.075% at the end of this run-in period. As with the HY2005, a relay in the potted block operates at about half the AC line voltage. There is no noise at all during turn-on or shutdown. With my ear against the speaker, I could hear low-level hum and a 120Hz buzz. Output hum and noise (input shorted) measured 5.2mV pp, with +3.2mV DC offset.

The HY2007 does not invert polarity. Input impedance is 100k at 1kHz. The gain at 2.83V RMS output into a 4Ω load is a bit high at 36.3dB. The output impedance at 1kHz is a low 0.08Ω, increasing to 0.19Ω at 20kHz.

The frequency response (*Fig. 21*) was within ±1dB from 26Hz–45kHz, at an output of 2.83V RMS at 1kHz into 4Ω. The −3dB points were a rather high 15Hz, to 85kHz at the top end. When I connected a load of 8Ω paralleled with a 2µF cap, the HF −3dB gain point jumped to 83kHz with HF peaking at about 60kHz.

The IHF load, which simulates a loudspeaker impedance peak at 50Hz, produced an insignificant 0.05dB higher response compared with the 8Ω resistive load alone. The HY2007 will be insensitive to variations in speaker impedance with frequency.

THD+N versus frequency is shown in *Fig. 22* for the loads indicated on the graph. During distortion testing, I engaged the test set 80kHz low-pass filter to limit the out-of-band noise. There was no evidence of HF oscillation during the testing.

Figure 23 shows THD+N versus output power for various frequencies. My toroidal power transformer was rated at 333VA rather than the 368VA specified on the data sheet, so I could not achieve the rated 240W into 4Ω. The heatsink was never hot to the touch, even at the 160W 2/3 power point. Ultimately, I was able to drive 180W into 4Ω at 1kHz, and about 150W at 20Hz and 20kHz before the AC line fuse protecting the transformer primary blew. There is no "Link" point on the PC board, because the HY2007 is rated for only 4Ω loads.

I also plotted the 1kHz product of the multi-tone intermodulation (MIM) 9kHz + 10.05kHz + 20kHz test signal versus output power out to 67V pp. This gives a better indication of the amplifier's nonlinear response, because it is a closer approximation to music than a sine wave.

The distortion residual waveform for 10W into 4Ω at 1kHz is shown in *Fig. 24*. 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 some crossover distortion with higher levels of power line harmonics. THD+N at this test point is 0.11%.

The spectrum of a 50Hz sine wave at 10W into 8Ω is shown in *Fig. 25*, from zero to 1.3kHz. The 2nd, 3rd, 4th, and 5th harmonics measure a low −91dB, −92dB, −94dB, and −94dB, respectively. However, significant power-supply artifacts are present at 60Hz and its harmonics. The THD+N measures 0.21%, with the true 50Hz harmonics contributing only 0.005%.

Figure 26 shows the amplifier output spectrum reproducing a combined 19kHz + 20kHz CCIF intermodulation distortion (IMD) signal at 12V pp into 8Ω. The 1kHz IMD product is 0.025%. Repeating the test with a multi-tone IMD signal (9kHz + 10.05kHz + 20kHz, shown in *Fig. 27*) resulted in a 1kHz product of 0.011%. (I plotted MIM distortion versus output in *Fig. 23*.)

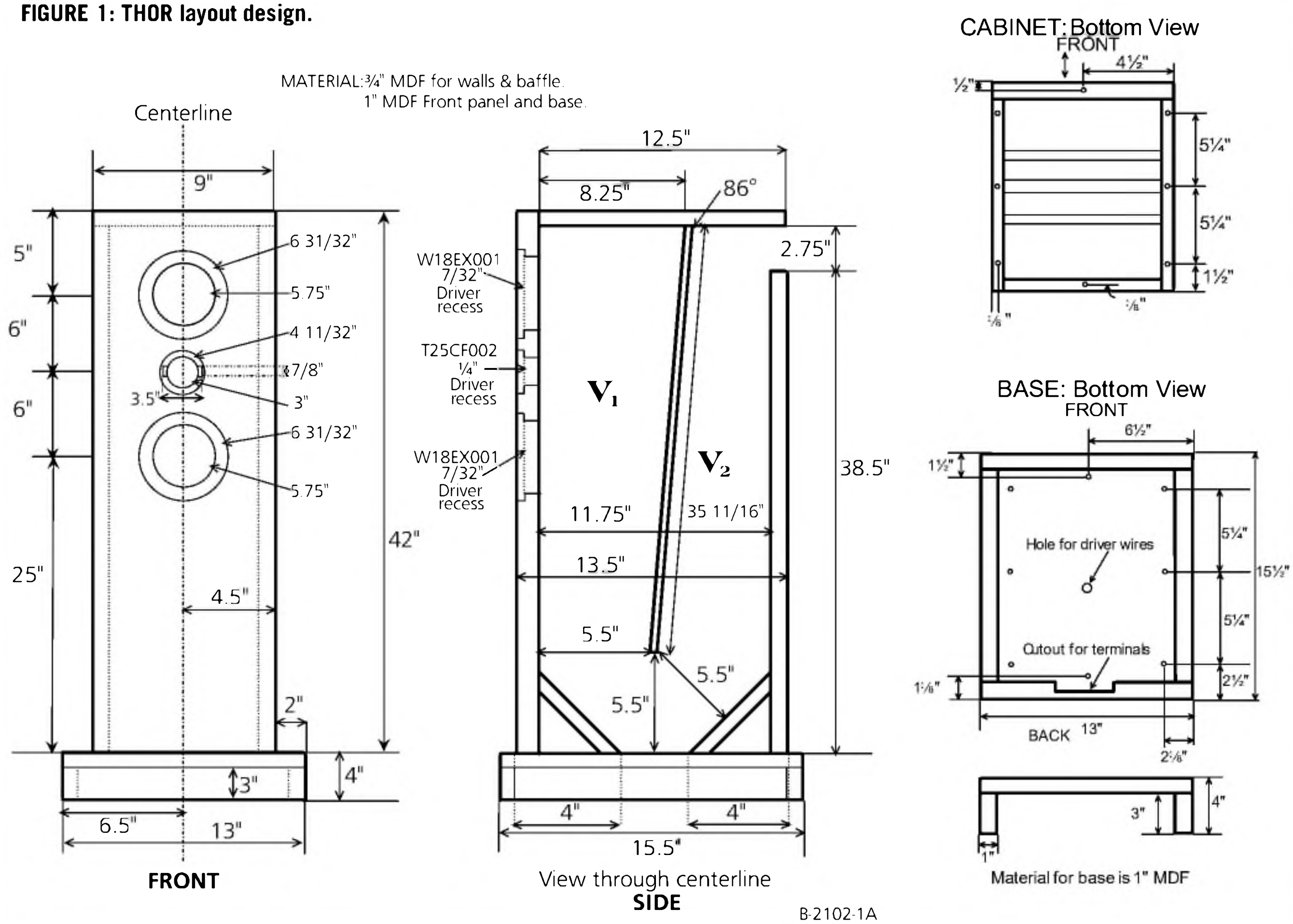
The 2.5V p-p square wave into 8Ω at 40Hz showed the same tilt you see in *Fig. 20* for the HY2005. The 1kHz square wave (not shown) was almost perfect. The leading edge of the 10kHz square wave had a very small peak. When I connected 2µF in parallel with the 8Ω load, I saw underdamped ringing, which reflects the 60kHz peaking measured in the frequency-response test.

LISTENING TEST LIMITATIONS

In order to send the amplifiers on for listening tests, I had to pick two of the four modules and connect them for stereo operation. The HY2001 was malfunctioning, and the HY2007 was a MOSFET design. None of them have the extended bass response required for subwoofer amplification, so I did not think that was a viable option for the HY2007. In fact, these amplifier modules have four noticeably different audio personalities based on my measurements.

Therefore, I mounted the HY2003 and HY2005 BJT amplifiers on my test fixture. I designed a 100k T-section attenuator (17k4 series arms with a 287k shunt resistor) for the input of the HY2005 to bring its gain down to that of the HY2003. Further, I had to operate both modules at the 28-0-28V AC secondary voltage limit of the HY2003. I ran some additional tests to be certain the HY2005 was operating properly at this reduced voltage. ❖

FIGURE 1: THOR layout design.



CORRECTION

For the benefit of our readers without X-ray vision, we reprint the layout designs and cutting guide (*Figs. 1* and *2*) for the THOR transmission line (May '02). We regret that these figures published with the article contained lines too faint to be readily detected. —Eds.

VINYL REVIEW

Thank you for publishing a turntable review—Music Hall MMF-2.1 by Charles Hansen in the August 2001 issue. Mr. Hansen writes that he was asked to take as many objective measurements as he could; however, unlike his excellent reviews of tube amplifiers with many useful measurements, he seems able to take only a few of the Goldring cartridge, and none of the turntable itself. Mr. Hansen does state that no flutter or wow was visibly obvious with his strobe, but he does not state whether this was performed under a dynamic load condition or merely with the platter spinning. Which was it?

Speed variations under dynamic load are audibly annoying, especially with a large dynamic range program such as occurs with sudden loud transients in classical music, and so on, though less annoying with some other types of music.

Other useful measurements are of vibration reaching the platter; platter and mat internal damping; and resonances in the tonearm, if Mr. Hansen is able to take such.

I am hoping you will have the opportunity to review the new Music Hall MMF-7, with its adjustable motor position/drive belt tension, surely of interest to vinyl-loving DIY enthusiasts!

Chris Logan
Sydney, Australia

Charles Hansen responds:

Mr. Logan is correct in that I do not have the test equipment to test the mechanical aspects of the turntable (wow and flutter, vibration,

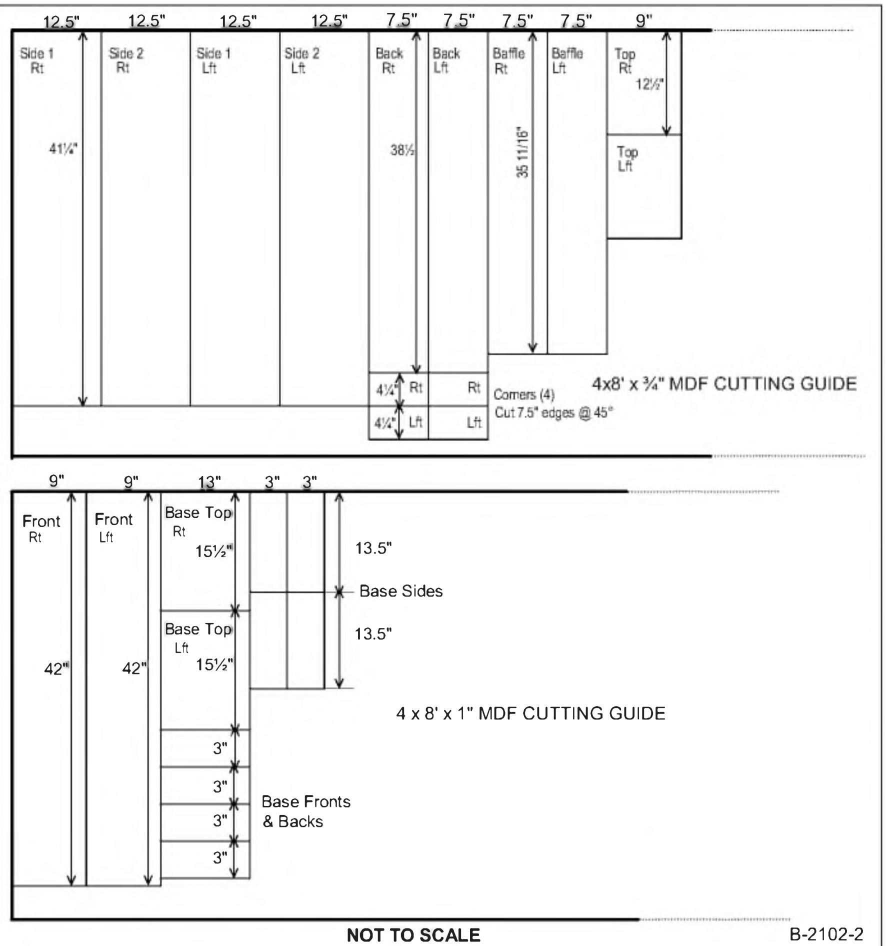


FIGURE 2: THOR cutting guide.

and so on). I did test for visible flutter using the LED strobe with an LP playing, however.

VINTAGE UNITS

I read with interest the article on the two-way open box system ("The Infinite Box Concept," Jan. 2002, p. 8). I think you could do better for less—both in labor and parts—along with better frequency response.

All you have to do is dig up a copy of the esteemed late Gilbert A. Briggs's book, *LOUDSPEAKERS.* His Wharfedale Wireless Speaker Works in Idle, Bradford, Yorkshire, UK in 1958 (5th Ed) published this. I partway followed his instructions found on p. 181 for his "Sand filled Baffle for Home Construction" that uses a twelve inch, coupled to a ten inch in parallel and a 1″ HF unit separated by a 4µF capacitor.

These are mounted on a ¾″-thick baffle backed by another panel the same thickness. The two panels are separated by strips of 1″ × 1″ fir. These strips not only go around the baffle edges, but around the openings for the drivers. This is so that the void between the panels can be filled with mason's sand, the whole being painted a matte black before the speakers are mounted to the back of the front panel.

The weight of each speaker system is considerable at this time. The whole thing from top to bottom should be angled backwards somewhat, to sort of time-adjust the driver. I believe others copied this to make a similar configuration.

The speakers I built were 38″ wide and about 34″ in height, with a prop behind or attached at both ends. They are best placed angled, their inner ends near the wall, the opposite ends about nine to ten inches out from the wall. I veneered the units on each outer third of each panel, and on the center I used a decent cover sold at the time by Allied Radio for the grille cloth.

I remember that not only were they easy to build, but they were also a delight and cost next to nothing. I know this would not be the case today, but neither would that backless box.

I gave my set away to a friend in 1998. These had Wharfedale 12″ drivers coupled with an English WE (Whitley Engineering) 10″ model that had 4, 8, and 16Ω taps; they lasted over 40 years. The sound was very satisfying and easy on the ears, never irritating. That is far more than you can say for a lot of other speakers. It would be near impossible to duplicate, even with the prints. One of the biggest mistakes I have ever made was to give these things away.

A few issues ago you ran an article on a Westinghouse console radio ("Recreating a 1948 Console Amp," Aug. 2001 *aX*, p. 54). Were you aware that Telefunken had some of their larger all-wave radios put in Canadian-built consoles?

I have a beautiful model, a Marquis in a Canadian, French Provincial walnut cabinet. That would be hard to beat. At least that is my thought. It has never gone out of alignment on any bands, nor has any tube failed in over 35 years. Though it does not get played an awful lot, it has always responded.

Harry P. Wood
Cudahy, Wis.

PLUG IT IN

Regarding Charles Hansen's February 2002 review of the VacuTrace (p. 56), I noticed the complaints about the Tektronix 7603 oscilloscope and 7B53A time-base plug-in in the review. You must be aware that the versatile 7603 mainframe can accept a vertical amplifier plug-in (such as the 7A26) in the timebase slot. The trick is to use two vertical amp plug-ins–one in the mainframe's horizontal slot, another in a vertical slot. This would give you a true X-Y display, with 1-2-5 decade and variable attenuators on both the X and Y axis.

Scott Newell
Fort Smith, Ark.

Charles Hansen responds:

Mr. Newell is absolutely correct. The instruction manuals for my 7A16 and 7A18 vertical plug-ins mention that they can be used in any position in the mainframe, and they do indeed produce the results he describes. Thanks for the helpful tip.

FREEWARE

Your August 2001 issue notes the availability of Circuit Creator ("Audio News," p. 6). Many other programs of this type have demonstration versions that lack utility because the main functions are disabled. Many beginners and students use these free demo versions, because they cannot pay for a complete program of this type. For those who fall into one of these two groups, I suggest you get an evaluation copy of CSiEDA 4.0.

The program may be the most generous free version of all. It allows a maximum of 50 components, with a maximum of 250 pins, including schematics, pcb, simulation, gerber, and even visualization in 3D. I believe it just about covers all the needs of any beginner. It is available at www.csieda.com, or at www.capcad.fr.

To download this whole program requires a lot of time through the Internet. Although the program I obtained was from a French magazine, and it can be downloaded in French, the installation is also available in English.

The company also offers a professional program to make schematics and pcb called Target 3001. This version has a maximum of 400 pins and at around $50, may be the cheapest on the market.

Practically any beginner may access this program if he needs to make some project larger than usual. It is available at www.ibfriedrich.com. A freeware version, with a maximum of 100 pins, is also available.

In simulation programs, the most generous version is perhaps that of SIMetrix: offering up to 120 nodes; it may also be the easiest-to-use program. It is available at www.newburytech.com.

Another free program of this type is SuperSpice. I'm not familiar with its features, but I do know that it is free and available at www.anasoft.co.uk.

I subscribed to *Audio Amateur* before *audioXpress.* As other readers have already said, the quantity of articles dedicated to valves seems a little excessive, but this probably only reflects the preferences of the readers. A web page that could be of interest to those fond of valves and to those who prefer the transistors is peufeu.free.fr/audio.

I would like to see you publish some projects relating to balanced preamplifiers, and active speakers. I believe that they are the only road toward the authentic hi-fi.

I have looked for projects of this type on the Internet, but I have only been able to find a balanced preamplifier that, in fact, is a copy of a Nelson Pass design. It is a little expensive, so I would prefer a design with operational amps. I have not been able to find any design of active speakers.

Finally, I recently have bought myself an ECR meter. This is not a very well-known measuring instrument, but it is of great utility for repairs and to judge the quality of capacitors. This interests a lot of those in the hi-fi field. I acquired it at www.monitortest.net. The version is a kit and very low cost. It will probably interest other readers.

Juan Raúl Couto Domínguez
La Coruña, Spain

EQUALIZING DRIVER UNIT OUTPUTS

I'm currently assessing bass and midrange drivers for use in a new speaker system. The most suitable bass unit I've found has a sensitivity of 88dB, and the midrange has a sensitivity of 90dB. No problem, I thought, I'll equalize the drivers using a series resistor for the midrange unit. I can calculate this easily.

But when I looked through my pile of technical literature I couldn't find any formula for calculating this easily. So here goes:

1. Most drivers have an impedance of 8Ω, and their sensitivity is measured using 2.83V input, at a distance of 1m. For meaningful comparisons, you need to check that your drivers are measured in the same way as each other!
2. The problem is that the sensitivity measurement is a logarithmic measure, whereas you need to use a linear relationship to calculate the ratio of padding resistor to speaker impedance. So you first must convert the sensitivity measurements to a linear scale.
3. For the first driver, $90 = 20 \log (P/P_0)$,

(to page 72)

Classifieds

VENDORS

ALL-FET phono, DAC, line, power amps **www.borbelyaudio.com**

The **BassZilla** has landed! Finally, a high-efficiency and affordable loudspeaker kit by Dick Olsher. Kit plans at: **http://www.blackdahlia.com.**

Low density CONCRETE CABINET loudspeakers
www.faradaysound.co.uk
Self-build instructions available

Check our growing line of audio test instruments: data sheets, user guides, software. **TDL Technology, Inc., www.zianet.com/tdl/ppage2.htm.**

Cabinets for North Creek Okara 2 or other design using Vifa P13WH. EBS alignment, –3dB @ 56Hz. Cabinets for ACI SVC10. Both customizable, starting at $175. 505-280-8829. E-mail: kennethmartinez@earthlink.net.

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.

Principles of Power, tube audio books, kits, FAQ **www.londonpower.com**.

AD INDEX

Yard Sale

FOR SALE

Digital only CardD made by Digital Audio Labs. S/PDIF 16-bit input/output card for PC. New. Superior audio quality. List $395. Make offer. Doug Pomeroy 718-855-2650, e-mail pomeroyaudio@att.net.

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To find out how you can order an ad for the classified pages, see our website at www.audioXpress.com.

WANTED

2 Dynaudio D-76 3″ dome midranges. Philip Souza 530-823-8232 (days), or psouza@sbcglobal.net.

Heathkit Model IM-12 manual. Also any 1960s and 1970s catalog. *The Fisher Handbook*, any copy. RCA, Sylvania, and GE tube manuals. High Fidelity's Test Reports—any edition. Shure V-15 Type III phono cartridge and test record—no need to be stylii. Catalogs too. Pekka J. Poutiainen, PL 136, FIN – 90101 OULO, FINLAND.

Used/rebuilt PC28 board for Dynaco Stereo 400 power amp, julian@ibd.ab.ca.

Hobbyist seeking pre-recorded, 2-track reel-to-reel tapes—7.5 or 15 ips. Can be 1950s/1960s commercial product or old studio dubs. Andy Pennella, 203-329-7498 or ajpennella@att.net.

Bedini 25/25, Heath UA1 or UA2 tube amps, Cotter MC transformer, AR Holographic series speakers, Quad(uk) amp/speakers, Pamphonic tube amp, old tube guitar amps, and horn speakers. Sonny, 850-314-0321, e-mail: sonnysound@aol.com.

Focal Egg II kits, enclosures, or complete speakers. 651-733-6875 or rlseverance@mmm.com.

WEBSITE RESOURCES

CONTRIBUTORS WANTED!

AUDIO AIDS— Whether your thing is solid or hollow state. Crossovers, power supplies, or transformers. You have experience and know-how others could use. Why not share that information with *audioXpress* readers? We encourage short (several hundred words) tips and hints for other electronic DIYers and will pay a modest stipend for your efforts. Contribute to the wealth of information in this industry by sending your "Audio Aid," and any accompanying photographs or artwork, to:

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FAX 603-924-9467 E-mail editorial@audioXpress.com

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or 4.5 = log (P/0.0002)—this will give you the driver sensitivity in microbars. (The reference sensitivity P_o = 0.0002 microbars). $10^{4.5}$ = P/0.0002, or P = $10^{4.5} \times 0.0002$. You can easily find this using the POWER function in EXCEL: P = 6.3 microbars.

4. For the second driver, 2 P = $10^{4.4} \times 0.0002$; i.e., P = 5.0 microbars. Note how the comparatively insignificant difference on the logarithmic scale changes to a very significant difference on the linear scale!
5. In order to equalize the outputs, you must reduce the midrange output by (6.3 – 5.0 microbars). So you need to pad down the voltage to the midrange in the ratio 1.3:5.0.
6. Assuming an 8Ω impedance, this gives a padding resistor of 8 × (1.3/5.0) = 2Ω.

It's easy to simplify the calculations a bit and set them up as a function in EXCEL using the sensitivities as input parameters. Mind you, no doubt most of us use a potentiometer for this task, vary the value until the driver outputs are equal, and take a reading of the potentiometer setting!

Bill Petrie
Tenerife, Spain

VOTES OF CONFIDENCE

I like the new *audioXpress*, as my interests span the audio range. Please limit reviews to products of interest to the project builder. I can find reviews of commercial audio products elsewhere—e.g., *Stereophile* at lower per-page cost, and, I might add, lower interest quotient.

I am sure that I am like many other project builders (in all fields) who are interested in many project articles. Even though we may never build most of the items described, we nevertheless enjoy reading about them for insights into how the designer approached problems, tips that may solve our own dilemmas, and so on. I think articles that give general information are probably the best.

Please keep up the good work.

Andrew Tomlinson
Palo Alto, Calif.

CLASSIC CIRCUITRY CONTINUED

In response to the "Generic" Amp letter in Xpress Mail (January 2002, p. 74), the center-tapped connection used on the amp's output sections has absolutely nothing to do with balancing output to the speakers to reduce noise by way of common-mode techniques.

The real reason it has been used in this circuit is to provide the amp with a center-channel output. If you were to think about that kind of a connection on an ordinarily connected pair of output transformers, then it is easy to see you would get the difference (L – R) signal. Wouldn't sound too good. The connection as shown in the Pilot 264 schematic provides a (L + R) signal. That's what is needed.

Interference in speaker cables is most unlikely, because they are usually at very low impedance. The technique is used to advantage primarily in low-level circuits of higher impedance.

John L. Stewart
Ontario, Canada

HELP WANTED

In the Oct. 2001 issue ("Xpress Mail," p. 80), Bob Cleary asks about problems of drifting DC offset related to the zener reference circuit in the Hybrid Amp. Could it be that R15's value is too high, only allowing less than 3.5mA of current through the zener? From past experience, I thought that a ½W zener should have at least 20mA going through it to function.

If any of your readers have successfully built this amp, could they share their thoughts/ideas on any circuit enhancements/changes that they've done to make it work. And also, I'd like to know their opinion as to whether they believe that it sounds as good as claimed by the author, in comparison to other amplifiers (all solid state) they may own. ❖

Joe Wdowiak
flyfishonly1@hotmail.com

Readers with information on this topic are encouraged to respond directly to the letter writer at the address provided. –Eds