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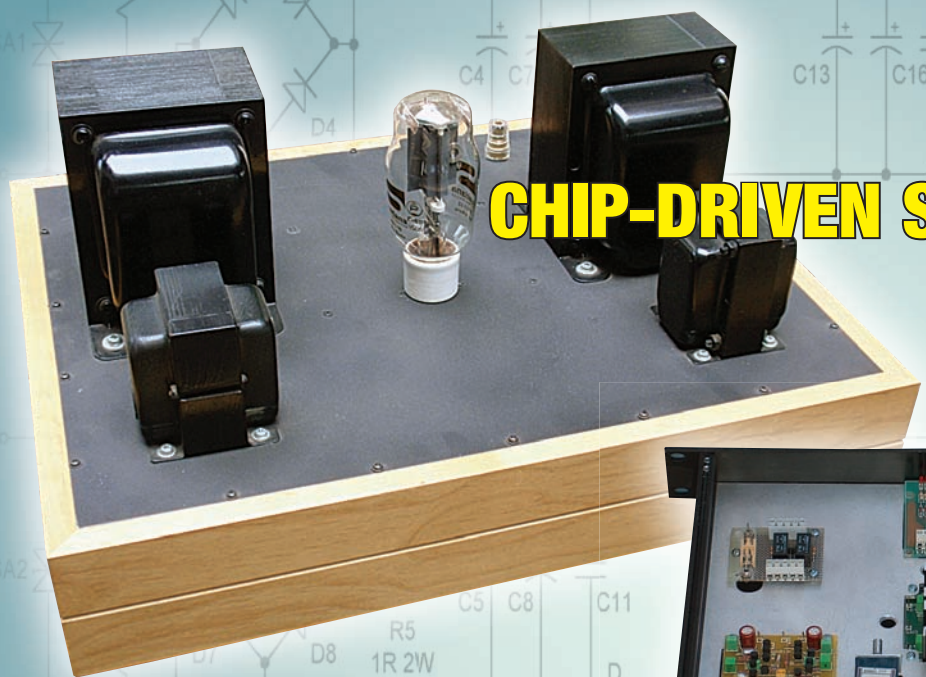

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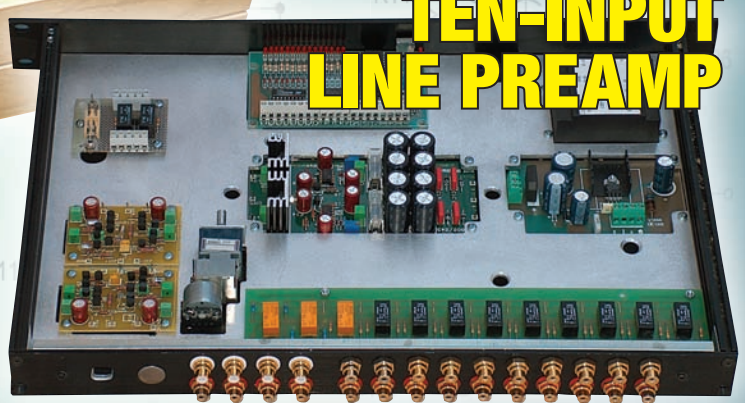

REVIEW: BENCHMARK'S NEW DAC

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Capable of withstanding direct exposure to all types of weather, the AC.17, from **Terra Loudspeakers**, is a three driver two-way system featuring dual 6.5″ cast frame woofers and a 1.1″ titanium dome tweeter. The AC.17 includes unique technical features such as Terra's Anodized Ceramic Aluminum Diaphragms, which make for very light, stiff, and responsive weather-proof cones, as well as a magnetic fluid centering system which eliminates the typical woofer spider. Every component of the AC.17 has been designed to remain rust and corrosion free. Multiple connector variations are available, as well as various color choices. For further information go to www.terraspeakers.com or 207-725-1005.

The Groove Master Phono Preamp is the newest product from **ColinElectronics**, and features vinyl playback with supreme transparency of tone and space, and 103dB dynamic range. To learn more, please visit http://colinelectronics.com.

The **Consumer Electronics Association** has released *The MRAV Guide*, from its DigitalTips.org website, which allows homeowners to take full advantage of music and video collections in all areas of their homes through multi-room audio/video. The guide follows a day in the life of the Johnson family using their MRAV system to bring the family together, and includes a printable checklist with step-by-step directions on how to speak with builders, contractors, and others involved in the process.

The SW-T2.1 1800 is a touch speaker system from **Genius**, which features an embedded equalizer so users can choose between all types of music. Also included are a line-in jack or headphone jack, a 6" subwoofer, and an easy-to-use touch screen. For more, go to www.geniusnetusa.com/buy.php.

The SchmartSolder is a new tool from **SchmartBoard** that makes it simple to hand solder through-hole components. SchmartSolder are loops of solder produced in just the right diameter to fit over the component leads, which, once the solder is flush to the circuit board and touched by the soldering iron, melts and flows into the hole, making flawless, neatly soldered components. To see more, go to www.schmartboard.com/index.asp?page=resources_howto.

Parts Express, one of the top suppliers of audio, video, and speaker components, introduces its new line of **Dayton Audio** Cat5e/Cat6, coax and speaker cable. Perfectly suited for all types of applications, these installer grade products include the Cat 6, with 550MHz bandwidth, copper clad steel 18AWG center conductor, and range from 100 to 1000′ lengths. For more information, visit www.parts-express.com.

Alcons' LR14/90 pro-ribbon line array is a passive two-way ultra compact module. It features a very compact design and maximum SPL of 130dB+ per unit, and has 800W peak power handling and crystal clear response. Frequencies below 1kHz are taken care of by two 6.5″ neodymium mid-bass drivers, and the total system weight is 33 lbs. For more information, please go to www.alconsaudio.com and www.theribbonnetwork.net.

By Paul J. Stamler

Tri-Way Low Voltage Supply, Pt. 1

You can use this Tri-Way—a three-output regulated supply—for a variety of applications.

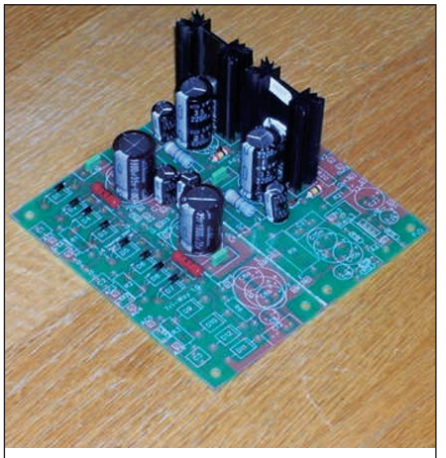

PHOTO 1: The Tri-Way board, with most of the parts installed for use in a compressor.

This is the second in a series of multiple-purpose power supply boards which began with the Gamp[1]. That board was intended primarily for low-power tubed projects, and used passive filtering (although it could be adapted to add regulation).

This board was designed for a different class of projects: solid-state designs using regulated ± supplies to feed amplifying circuits (discrete or integrated), plus a third positive supply which could feed phantom-powered microphones, logic circuits, or relays. Partway through the design process, I realized that with a few extra jumpers the same board could be used to feed regulated DC to the filaments of tubed projects, while still retaining that extra supply for phantom, logic, and so on.

Digging further, I designed the board so that Supplies 1 and 2 could be fed by a center-tapped transformer, by a single transformer with separate secondary windings, or by two separate transformers. This opened the way toward using the board with wall-wart transformers, useful for meeting UL specifications[2]. Using separate windings or transformers, in turn, provides the ability to float Supply 2 (or 3), elevating the filament above ground to avoid exceeding a tube's heater-to-cathode voltage limits.

JOB DESCRIPTION

As always, I began by listing what I expected the design to do:

1. Provide up to ±30V DC, regulated, from Supplies 1 and 2, at current levels up to 1.2A, for solid-state preamplifier and signal-processing circuits.
2. Provide separate regulated voltages up to 30V DC from Supplies 1 and 2 for use with tube filaments.
3. Allow Supply 2 to be floated up to +75V DC above ground.
4. Provide up to +48V DC regulated voltage from Supply 3, at up to 0.5A, for microphone phantom-power supplies, logic circuits, and relays.

In the course of doing those jobs, the board should:

1. Filter out ripple.
2. Filter out line garbage.
3. Minimize diode-switching noise.
4. Provide properly-regulated output voltages when line voltage dips down to 10% below nominal, and avoid fried components when the line voltage runs 10% above nominal.
5. Minimize component stress under adverse conditions, including hot working environments (Missouri summers are legendary).
6. Be usable with one, two, or three external transformers (wall-warts) if needed.
7. Allow off-board regulators for high-current designs.

I added one refinement during the design process: I made the board in two halves, with a cut line. This means you can use it as a single 5 × 5″ board, with raw supplies and regulators together in the cabinet, or separate the raw supply section and mount it in an external power pack, useful for keeping hum-radiating transformers away from audio circuits. I designed the two halves so that each would fit vertically into a 2U rack chassis.

DECISIONS

Once I'd settled on the board's requirements, I had two more major decisions: what level of regulation would the board provide, and what sort of passive filtering would precede the regulators?

After a few decades, I've decided that for solid-state audio circuits (particularly IC op amps), the level of regulation provided by adjustable IC regulators such as the LM317/337 is the minimum for good sound. In my experience, designs which use three-terminal fixed voltage regulators (78xx/79xx) don't sound as good as designs with LM317/337 regulation or better. Walter Jung has published tests comparing the technical performance of fixed and adjustable IC regulators, along with fancier circuits, and in virtually every respect the adjust-

able devices outclassed the fixed-voltage chips[3].

So the regulators would be LM317/337 chips; for phantom power, the TL783 chip (which has the same pinout as an LM317) would work fine. What about higher performance? For those who want tighter regulation, lower dropout, and other improvements, LT1085/1033 devices are pin-compatible with LM317/337s; they'll work fine on this board, too. The best regulation comes from op amp-based circuits like the ones designed by Michael Sulzer[4] and Walter Jung[5]; I like to put regulators such as those onto the audio boards rather than the supply card[6]. The Tri-Way card's Supplies 1 and 2 make good pre-regulators for this type of design.

What about passive filtering before the regulators? I chose to use two-stage RC filters rather than single capacitors. A properly-designed two-stage filter lets through less ripple than a single stage with the same total capacitance, and using a smaller input capacitor decreases the strain on the rectifiers at turn-on.

Two-stage filters are also more versatile for home constructors. Manufacturers can specify exactly what voltage a transformer should supply; home builders don't get to do that, but must use off-the-shelf stock values. A two-stage filter lets you design for whatever transformers are actually available, then drop extra voltage in the resistor (rather than stressing the regulator).

A two-stage filter made from smaller caps can also be more effective in filtering out switching noise and line junk, as smaller electrolytic capacitors remain capacitative out to a higher frequency. If you're separating the raw supply from the regulated, using an external box for the raw section, the second stage of filtering (which is on the regulator half of the board) will filter out any additional garbage picked up by the umbilicus that connects the two boxes.

As I've mentioned elsewhere, I'm a fan of Ben Duncan's ideas about designing anti-resonant power supplies to keep diode and line-borne noise away from audio circuits[7]. To those ends, I used fast-switching diodes and snubbers to minimize noise generation, bypassed filter capacitors with smaller electrolytics and film caps, and placed a small resistor between the rectifiers and the first filter cap. Jumping ahead a bit, I followed Duncan's suggestion and used 1N4148s as regulator safety diodes, along with current-limiting resistors; these are less likely to conduct high-frequency crud around the regulators. I also left room for adding surge absorbers across the transformer secondaries, which can help long-term reliability in this age of not-quite-stable wall current.

Finally, I made room for the grounding system I often prefer to pure star-ground: the grounded end of the regulator circuit(s) goes to the load ground, which, in turn, sends a separate wire to main system ground. Most of the time I find this provides a quieter supply.

JUMPING JACKS

Take a look at **Fig. 1**. Perhaps the most salient feature of this design is that it has

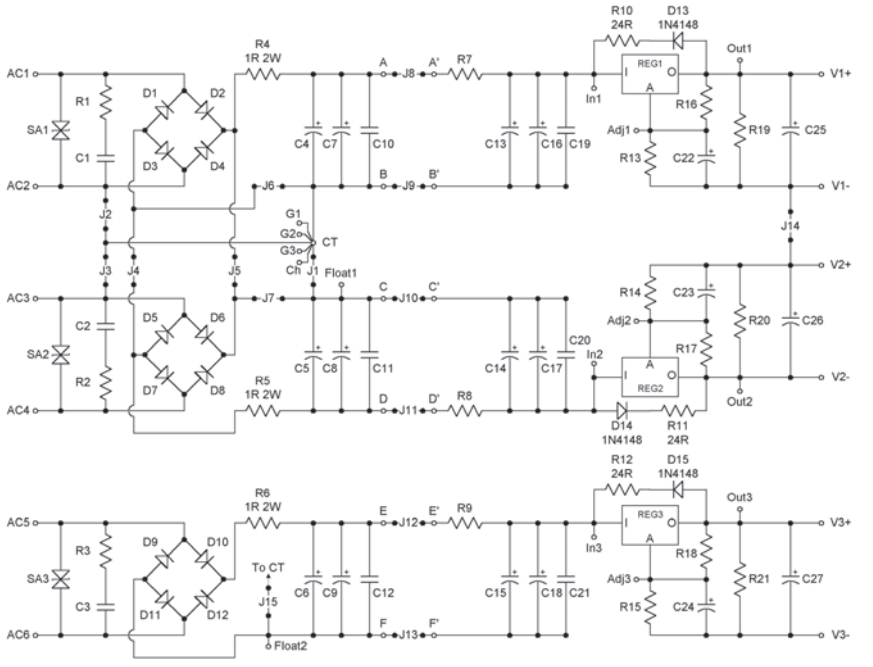

FIGURE 1: Tri-Way supply, basic schematic.

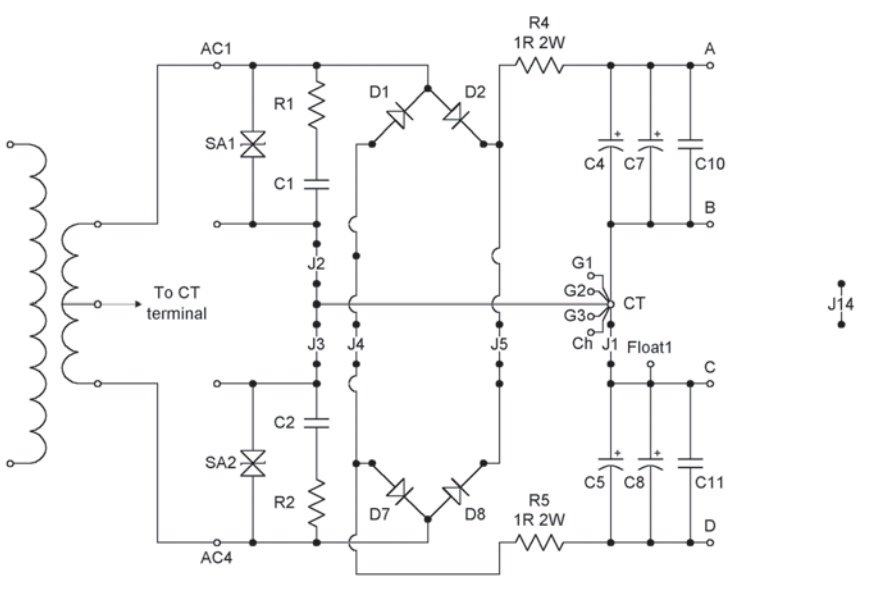

FIGURE 2A: Supplies 1 and 2 input section hookup with center-tapped transformer. Note use of J14.

a *lot* of jumpers.

Let's take the easiest ones first: J8-13 connect the input half of the board (snubbers, rectifiers, first filter section) to the output half (second filter section, regulators). When the board is used in one piece, these jumpers are simply short lengths of 18ga wire. If you split the board, however, and put the transformers and input section in a separate power pack, the jumpers become wires in an umbilical cable, which connects the boxes.

Jumpers 1-7 select the transformer type and circuit configuration of Supplies 1 and 2. Take a look at **Table 1**, which offers several possibilities:

1. ± voltage, center-tapped transformer (**Fig. 2A**). In this hookup, you'd connect the transformer's center-tap to the CT terminal on the board (duh). This produces a full-wave bridge rectifier while permitting separate snubbers for the two halves of the transformer. J1 connects the two supplies' ground sections together, producing the main power supply ground point, while J14 ties the two regulators' ground points together, to feed the load ground.
2. ± voltage, separate windings or transformers (**Fig. 2B**). This is the configuration I recommend when the transformers are AC wall-warts. Again, J1 and J14 tie the plus and minus halves together.

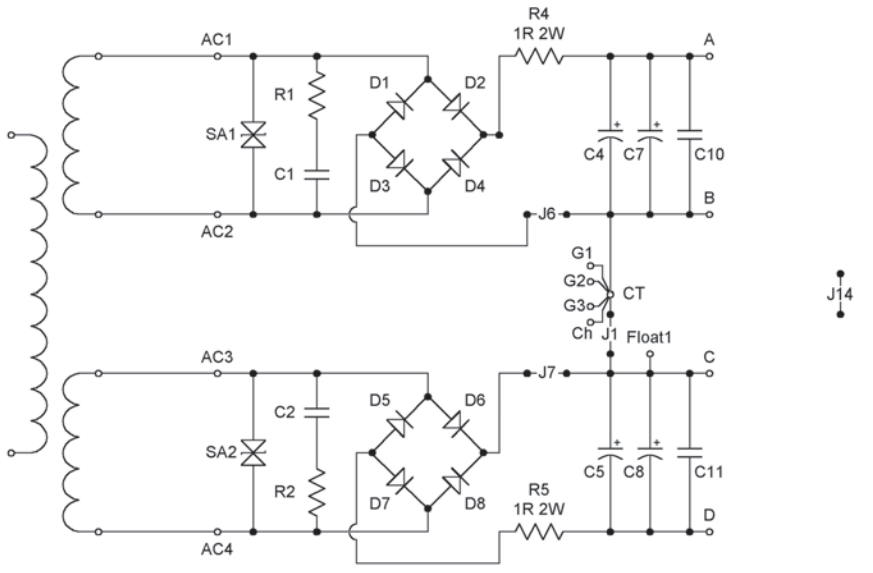

FIGURE 2B: Supplies 1 and 2 input section hookup with separate transformer secondary windings; these connections are also used with separate transformers. Note use of J14.

3. Separate supplies for filament voltages, also with separate windings or transformers (**Fig. 2C**). In this case, leave out J14; to float Supply 2, leave out J1 and connect the Float1 terminal to the floating terminal on the plate supply (there's one on the Gamp board, or you can make one with a couple of resistors—see the Gamp article for details). Note that Supply 2 uses a negative regulator; in a filament supply this shouldn't make any difference.

Table 1 Jumper Configurations

Configuration	Jumpers	Diodes	Transformer connections	Output ground
Supplies 1-2, CT transformer (Figure 2a)	J1-5, J14	D1-2, 7-8	AC1, AC4, CT	V1- or V2+ to load ground, load ground to G1, G2, or G3
Supplies 1-2, transformer w. sep. windings or 2 transformers (Figure 2b)	J1, J6-7, J14	D1-8	AC1-AC2, AC3-AC4	V1- or V2+ to load ground, load ground to G1, G2 or G3
Supply 1 grounded, Supply 2 floating; sep. windings or transformers (Figure 2c)	J6-7	D1-8	AC1-AC2, AC3-AC4	V1- to its load ground, load ground to G1, G2 or G3; V2+ to its load "ground", load "ground" to Float1
Supply 3 grounded	J15	D9-12	AC5-AC6	V3- to load ground, load ground to G1, G2 or G3
Supply 3 Floating		D9-12	AC5-AC6	V3- to load "ground", load "Ground to Float2

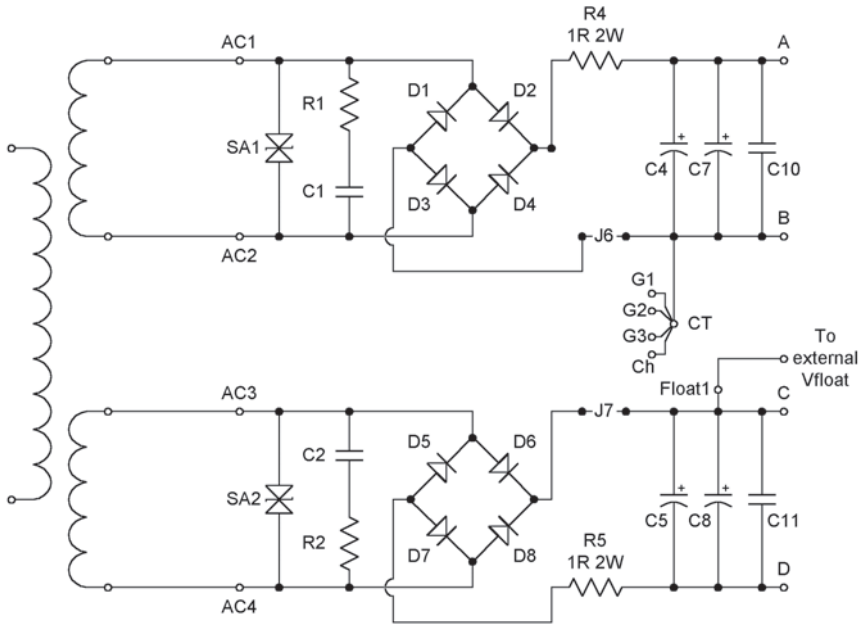

FIGURE 2C: Supplies 1 and 2 input section hookup with Supply 2 floating. Note that this requires separate transformer secondary windings or separate transformers, and that J1 and J14 are omitted. The external float voltage can be up to +75V DC.

Supply 3's ground is normally tied to the main power supply ground via J15; however, you could choose to float that one as well for another filament supply by omitting J15 and tying Float2 to a float point. **Figure 3** shows suggested wiring for the transformer primary/primaries.

FIRST STAGE

Calculate snubber values using Hagerman's article[8]. C1-3 should be stacked film caps (Panasonic V series or equivalent); I've left room on the board for values up to 0.82µF/50V, or 0.1µF/100V. Use 2W carbon comp resistors if possible, but 3W metal oxide resistors will work in a pinch. The surge absorbers are standard Panasonic ZNR or equivalent; choose a device with an AC clamping voltage 30% higher than the transformer's nominal secondary voltage.

As specified in the parts list, I chose diodes with fast turn-off times to minimize noise generation. On the board, I left space for either 1A or 3A diodes; if you use 1A diodes, the holes may be a bit loose. Use spacers to lift the diodes about 3/8″ above the board surface to allow air to circulate around them, and to prevent their heat from cooking the board.

R4-6 are anti-resonant resistors, as suggested by Ben Duncan. He uses multiple paralleled 1Ω resistors; I use just one. These should be 2W, metal oxide or carbon composition.

The capacitor tests I did a few years ago persuaded me that multiple smaller capacitors are more effective than single large ones for filtering out high-frequency junk, either line- or diode-sourced[9]. The first version of this board had six caps for each section—three big electrolytics, a 100µF electrolytic, a film cap, and a ceramic disc. That's a pile o' capacitors, buddy.

Sanity prevailed, however, and I trimmed the count to one or two big electrolytics, one small electrolytic, and a film cap. What sort of capacitors?

For the big electrolytics in the first stage, I made space on the board for three possibilities. In each section there's room for a 16mm (diameter) wire-lead cap, a 22mm snap-in cap, or two 12.5mm wire-lead caps. (Probably the latter will provide the best performance for a given value of capacitance.) These

packages are all bog-standard in diameter and lead-spacing, and available from Panasonic, Nichicon, and others. For wire-lead caps, I prefer the Panasonic NHG series, which have the least inductance for a given capacitance (and thus remain capacitative to a higher frequency). The small electrolytic is 8mm in diameter, typically 100µF/50V for Supplies 1 and 2, and again I like the Panasonic NHG series.

What about the film caps? They're metallized polypropylene with a maximum size of 16.5 × 7.2mm, and lead spacing 10mm. I added pads for 5mm and 7.5mm lead spacing for Wima or Epcos stacked-film capacitors. I'd typically use a Panasonic 0.1µF/50V ECQ-P series or 0.1µF/160V Epcos MKP series cap.

RESISTANCE IS NOT FUTILE

In many ways the dropping resistors between the two filter stages are the heart of this design. They drop the raw voltage to a level that's appropriate for the regulator inputs, absorb some of the voltage drop of a too-high transformer (taking the heat instead of the regulators), and provide the R for RC filtering. I'll talk about calculating the resistor values when I do some worked examples in Pt. 2.

There's room on the board for sand-cast 5W resistors; for lower-power circuits, you can use 2W carbon comp or metal oxide resistors, but figure out the worst-case dissipation on the case before choosing, and be conservative.

IF IT MOVES, REGULATE IT

The default regulators are LM317T for the two positive supplies and LM337T for the negative. These are TO-220 packages, readily available, and mount on Wakefield 637-series heatsinks. (The heatsinks come in four heights, for different power levels.) These handle low-voltage, low-current loads just fine—but they're not the only possibilities.

What if you need higher voltages—for example, +48V from Supply 3 for phantom-powered microphones? In that case, you can use a Texas Instruments TL783CKC regulator, which is pin-compatible with the LM317T and uses similar design procedures, although with slightly different numbers.

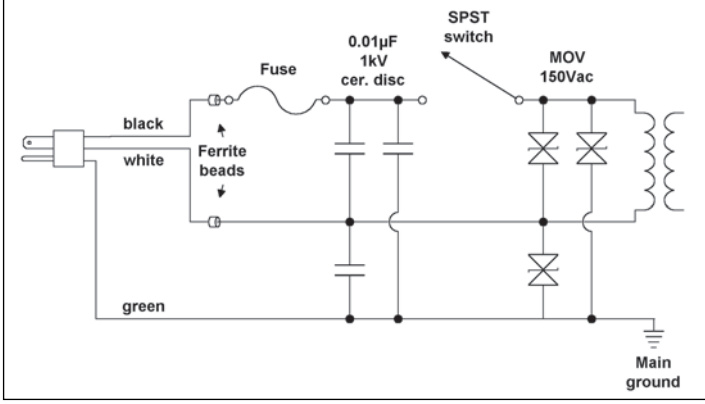

FIGURE 3: Suggested wiring of primary circuit, size fuse as needed. In a 240V system, use a 300V MOV.

Higher-current circuits such as tube filaments may draw more current than the TO-220 package can safely dissipate. Happily, you can still get LM317K and LM337K devices; these are TO-3 packages and (when mounted off-board on suitable heatsinks) can dissipate more power. I've put pads on the boards for connecting to off-board regulators.

Finally, LT1085/1033 regulators, designed for higher currents, also provide somewhat better regulation and line rejection; choose these if you want the board to provide higher audio performance. They cost a lot more, though. (There's probably no advantage to higher-end regulators if you're using the Tri-Way as a pre-regulator).

My regulator circuits are pretty much textbook designs, with a couple of exceptions. The application notes for regulators suggest that, when you use them with large output capacitors, you should connect a protection diode between the

Table 2 General parts list

Item	Value	Mfg.	Part #	Quan.
C1-3	* Polyprop. stacked	Epcos/Wima MKP		3
C4-6	* 22mm D electrolytic	Panasonic TSHA		3
	-or-			
	* 16mm D electrolytic	Panasonic NHG		3
	-or-			
(C4a-6b)	* 2x12.5mm electrolytic	Panasonic NHG		6
C7-8, 16-17	100µF 50V electrolytic	Panasonic NHG	P5571-ND	4
C9, 18	same, or 33µF 100V el.	Panasonic NHG	P5595-ND	2
C10-12, 19-21	.1/50 polypr. film	Panasonic ECQ-P	P3925-ND	4
	-or-			
	.1/160 polypr. stacked	Epcos MKP	495-1375-ND	4
C13-15	* 16mm D electrolytic	Panasonic NHG		3
	-or-			
C13a-15b	* 2x12.5mm electrolytic	Panasonic NHG		6
C22-23	100µF 50V electrolytic	Panasonic NHG	P5571-ND	2
C24	same, or 33µF 100V el.	Panasonic NHG	P5595-ND	1
C25-6	1000µF 50V electrolytic	Panasonic NHG	P5575-ND	2
C27	same, or 470µF 63V el.	Panasonic NHG	P5585-ND	1
D1-8	100V 3A	Diodes Inc.	FR302DICT-ND	8
	-or-			
	100V 1A	Diodes Inc.	UF1002DICT-ND	8
D9-12	same, or 400V 1A	Diodes Inc.	UF1004DICT-ND	4
D13-15	1N4148	Fairchild	1N4148FS-ND	3
HS1-3	634-10[20,30]ABP(TO220)	Wakefield	345-1028[29,30]-ND	3
	-or-			
	680-125A (TO3)	Wakefield	345-1051	1-3
R1-3	* 3W MO	Panasonic	PxxW-3BK-ND	3
R4-6	1Ω 2W MO	Panasonic	P1.0W-2BK-ND	3
R7-9	* 5W WW	Yageo	xxW-5-ND	3
R10-12	24Ω 1/4W CF	Yageo	24QBK-ND (pkg 5)	3
R13-15	* 1/4W MF	Yageo	***XBK-ND (pkg 5)	3
R16-18	* 1/4W MF	Yageo	***XBK-ND (pkg 5)	3
R19-20	* 2W MO	Panasonic	P***W-2BK-ND	2
R21	* 3W MO	Panasonic	P***W-3BK-ND	1
REG1	LM317T	Fairchild	LM317TFS-ND	1
	-or-			
	LM317K	STMicroelect.	Mouser 511-LM317K	1
REG2	LM337T	Fairchild	LM337TFS-ND	1
	-or-			
	LM337K	STMicroelect.	Mouser 511-LM337K	1
REG3	LM317T	Fairchild	LM317TFS-ND	1
	-or-			
	TL783CKC	TI	Mouser 595-TL783CKC	1

Notes:
1. An asterisk (*) denotes a part with a value selected by builder.
2. All parts are sourced from Digi-Key unless otherwise indicated.

input and output terminals (D13-15 in this circuit). This lets the output cap discharge through the diode while clamping output-input voltage, which keeps the regulator from frying.

Unfortunately, the 1N400x diodes recommended for this application have high capacitance, which can conduct high-frequency crud from the regulator's input to the output and thence into your audio circuits. Ben Duncan suggests using 1N4148 or 1N914 diodes instead; these have lower capacitance and thus provide a better barrier. They're lower-power devices, though; to keep them from frying themselves, you need a small series resistor to limit the current during discharge (R10-12).

There are also three load resistors (R19-21). I'll talk about those in the final worked example (most of the time, you won't need them). The regulator circuits include bypass capacitors (C22-24) between the adjust terminals and ground. These improve performance on LM317/337 regulators, and LM1085/1033s as well. Texas Instruments, however, recommends *not* using them on the TL783, so you should leave them off if you use that chip.

VARIATIONS ON A GROUND

These days, most audio designs use a strict star ground; everything (audio circuits, power supplies, chassis) is returned to a single grounding point. I've found, however, that there's one exception: the "ground" reference point on regulators should be connected to the load's ground point, with the latter connected to main system ground. This is easy to do on the Tri-Way board, since the regulator reference points have their own output connection points.

MECHANICAL CONSIDERATIONS

I've drilled all input and output pads with 0.042″ holes, which can hold a wide variety of standard terminals. In an experimental or hobbyist setup, it's often worthwhile using terminals rather than soldering wires directly to the pads; if you wind up desoldering and resoldering, terminals make it much less likely that you'll lift traces on the board. Even if you don't desolder, repeated flexing of wires as you troubleshoot a project can put mechanical stress on solder joints and pads.

The board has a line across it, between the diode/first filter section and the second filter section/regulator. If you like, you can saw the board in half and place the first piece in an external box while the regulator piece goes into the main chassis, or mount the two halves vertically in a 2U rack chassis.

I've placed holes, suitable for 4-40 screws, in the four corners of the board sections. While I've tried to keep these away from live circuit traces, it's still a good idea to avoid shorts by using nylon standoffs rather than metal. The holes can also be used for right-angle brackets; again, watch out for shorts!

Finally, I've designed the Tri-Way so that the cathodes (striped ends) of all diodes and the + terminals of all electrolytic capacitors are on the right, as you look at the component side. Nothing is idiot-proof (I've pulled my share of dumb moves), but this helps avoid assembly errors that cause loud bangs.

Next month, in Part 2, the author puts this board through its paces, showing three different designs using the Tri-Way.—Eds. *aX*

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For a look at component placement, as well as the foil side, of the Tri-Way PC boards, visit our website at www.audioXpress.com

Remote-Controlled Ten-Input Line Preamp

In need of more inputs? Try this line amp, courtesy of the author, Borbely Audio, and Velleman.

These days, with all the possible sources of sounds and music, a preamp with five or six inputs cannot handle them all. My Yamaha C-80 preamp had five line inputs and I badly needed more. As a first approach, I was considering building a passive preamp and selecting all inputs via rotary switch—a simple, cost-effective solution.

Further considerations led me to believe that the best choice was to place a buffer between sources and the final amplifier. As I got to that point, I believed I needed to add a remote-controlled circuit to select all inputs and to adjust the volume control.

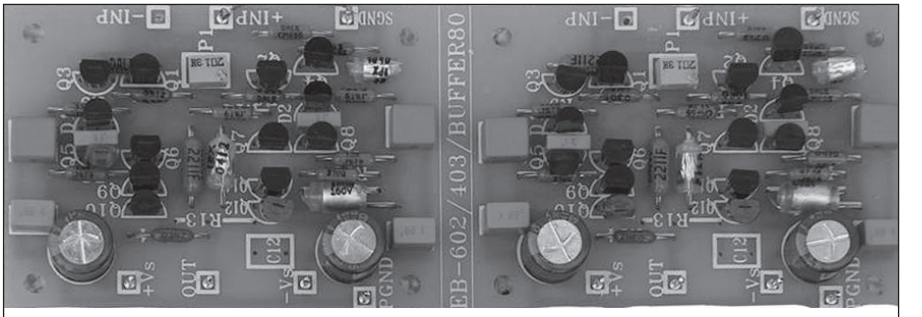

PHOTO 1: The Super Buffer circuit.

CIRCUIT SELECTION

In the May 2005 issue of *audioXpress*, I found the answer to my first point: an all FET Super Buffer and related power supply kits by Erno Borbely (www.borbelyaudio.com), while from other sources I have found an infrared transmitter/receiver Velleman kit.

The all FET Super Buffer is a simple circuit to build (**Photo 1**) with excellent specifications and great performance

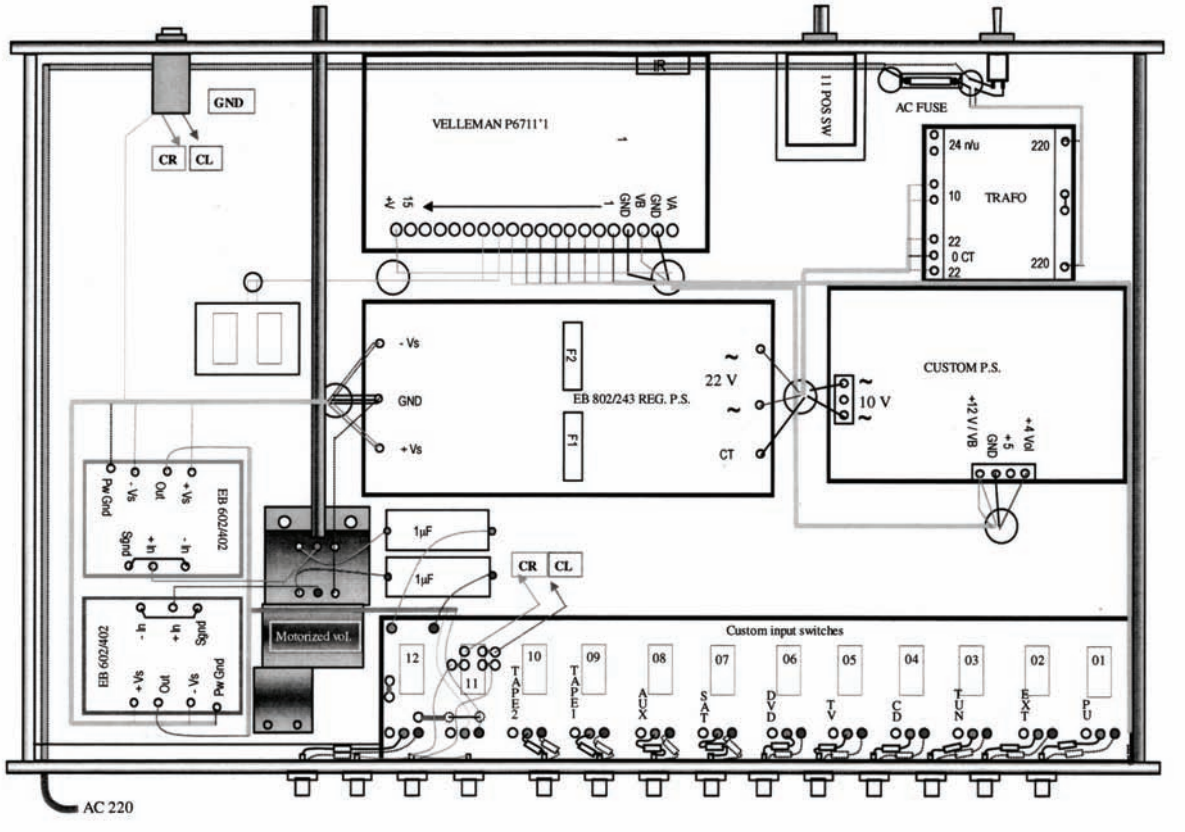

FIGURE 1: The final chassis layout.

such as:

Gain (choice of)	0 or 6dB (2×)
Freq response	> 1MHz (with unity gain)
Input impedance	1MΩ
Output impedance	< 50Ω
Min load impedance	1kΩ
Distortion (6dB gain)	unmeasurable (0.00XX)
Power requirements	±24V 25mA

The Erno Borbely regulated power supply kit was even easier to build and required a single adjustment to set the correct output voltage.

Both circuits worked fine with the first attempt. The Velleman IR transmitter/receiver (**Photos 2** and **3**) also came in the kit. Their assembly was very neat and did not require any adjustments.

With all these basic functional units on hand, I needed to draw a schematic to put them together. The final circuit (**Fig. 2**) includes the option of selecting all inputs via relays which you could activate either manually (rotary switch) or from a remote control.

I could not find any ready-made input switches, so I designed my own, including the printed circuit (both a joy and a pain for a hobbyist like me). The actual version (**Photo 4**) has the advantage of being very compact, thus minimizing the wire lengths from the RCA input connectors to the actual relay switching point and so being very close to the volume control and super buffer input points (**Photo 5**).

PHOTO 2: Velleman's remote control.

OPERATION

Each input selection relay performs a double function: to connect selected inputs to the bus, when energized; or short-circuit

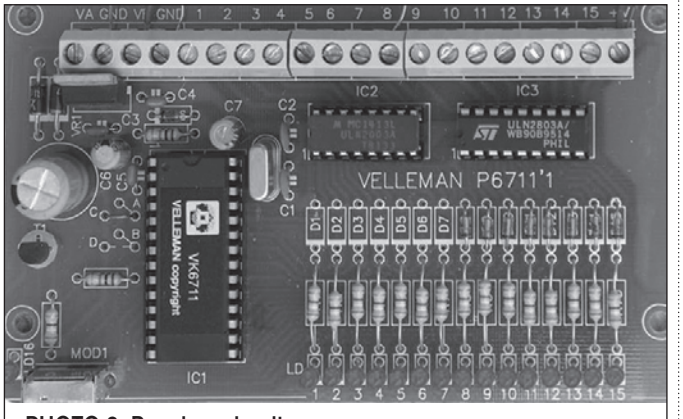

PHOTO 3: Receiver circuit.

to ground unused inputs via 1Ω resistors when in a normal position. This arrangement prevents stray capacitance to pick up signals from unselected input and mix them with the active one.

Due to the limitation of the Velleman receiver, you can remotely control only eight inputs (relays Ry 3 through 10); you must manually activate via ro-

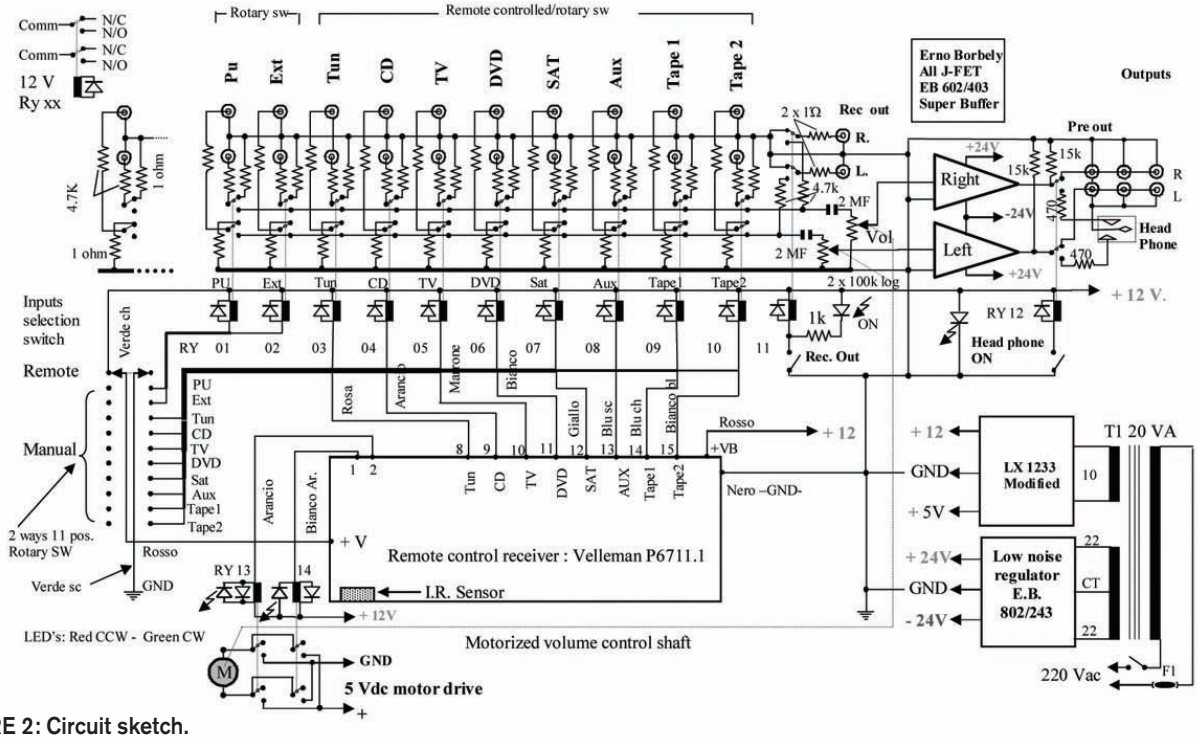

FIGURE 2: Circuit sketch.

tary switch the other two inputs (relays Ry 1 and 2).

Each relay, when energized, sends the stereo signal to a bus which feeds the super buffer through two high-quality 2MF Infinity capacitors and a motorized stereo Alps volume control.

You can adjust the latter via remote control by means of channel 1 or 2 outputs; these outputs energize momentarily either Ry 13 or Ry 14 to provide a clockwise (CW) or counterclockwise (CCW) rotation. I have built a small handmade wired board to provide this simple function.

Super Buffer output feeds three RCA connectors, via Ry 12 n/o contacts, which can be hooked up to an equal number of final amplifiers if required; this thanks to its very low output impedance of the Super Buffer.

Ry 12 connects Super Buffer output to the headphone when the related switch is "on"; outputs to power amp RCA connectors are disconnected and you can use a high impedance (> 600Ω) headphone instead.

Similarly, the *rec out* RCA connec-

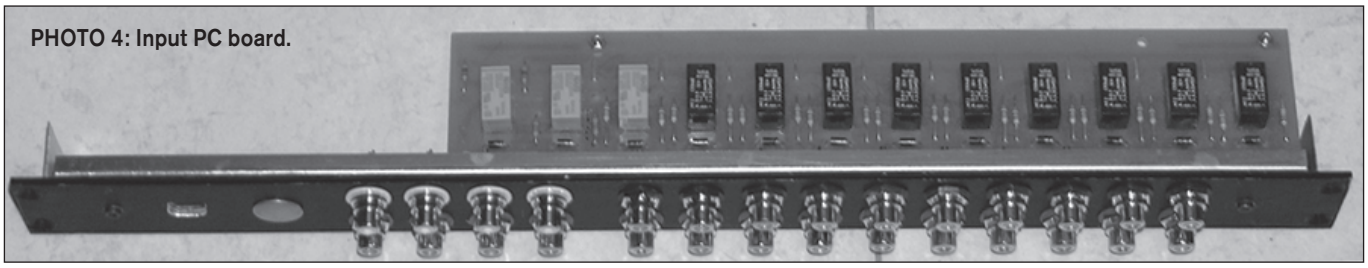
PHOTO 4: Input PC board.

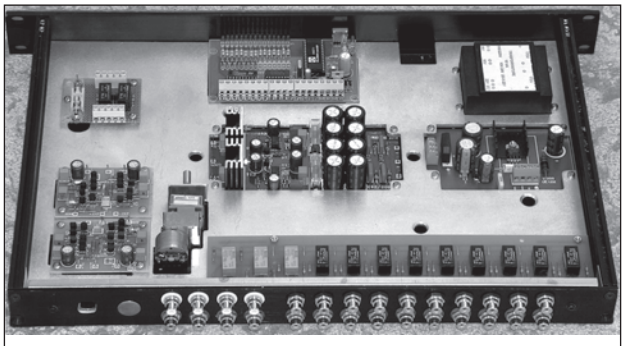
PHOTO 5: Supporting panel layout.

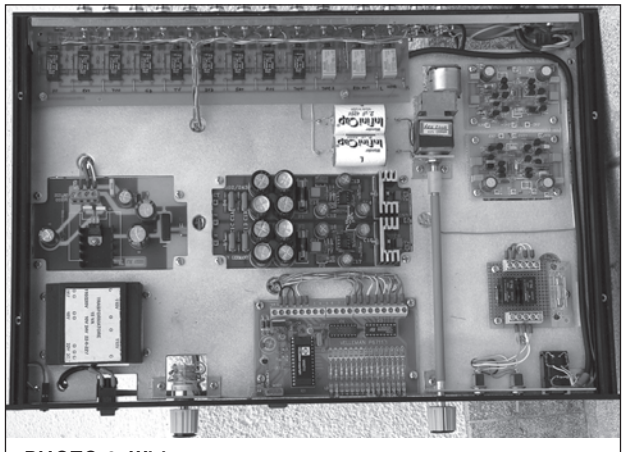
PHOTO 6: Wiring.

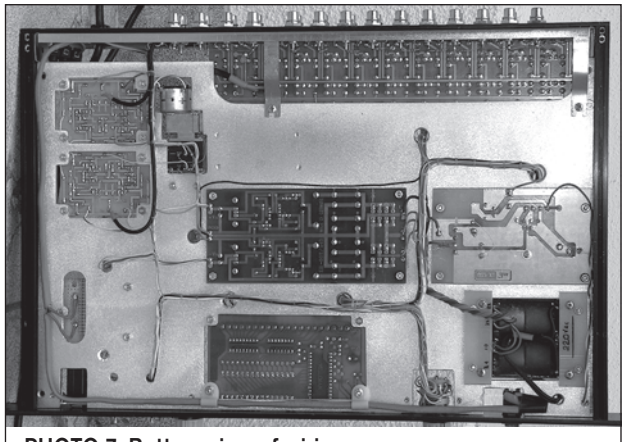
PHOTO 7: Bottom view of wiring.

tors become live when you activate the "*rec out*" switch. The Velleman IR receiver, to function, requires +12V, which is supplied by a "nuova elettronica" modified kit. This power supply also provides +5V to drive the volume control motor.

When you activate an IR receiver output (say, to select tuner, "Tun" 1 input via Ry3), the level at pin 8 goes down and Ry3 is energized.

Similarly, when you move manual switch to the Tun position (for example), it takes away +12 to the +V input to disable Velleman relay drivers and ground pin 8 to energize Ry3 to perform its required function. For some reason, however, Velleman relay drivers are not de-energized when +V is removed, and the remote control continues—unexpectedly—to function. I believe this is due to a +12 voltage internal return path (but perhaps a Velleman engineer could provide an answer and a fix to this inconvenience). However, by pressing the "Clear" key on the remote control, I can achieve the expected result.

ASSEMBLY

The major task was the physical location of all sub-assemblies in a 4cm (a bit more than 1½″) height chassis. This, in fact, required extensive work to provide a cutaway to properly fit all parts in the supporting panel (**Photo 5**). I recommend using a higher profile chassis.

After subassembly, I began wiring with the help of a sketch prepared under **Fig. 2**. **Photos 6** and **7** highlight this last phase of my work, while **Photo 8** shows a front view of it.

As I powered up, everything worked properly, with the exception of what I described previously in relation to the +12 Velleman receiver. I now have ten inputs to play with, and I definitely resolved my input availability problem. The Super Buffer performs very well.

I have hooked up my Suisse Revox CD directly to the power amp and compared its sound when fed through the Super Buffer. Apart from the 6dB gain (2×), I could not detect any difference; Super Buffer does nothing to the sound, it is not there—and this, to me, is a clear sign that its performance is top notch.

Tests with other sources, however, did not manifest the same behavior; the presence of Super Buffer made a significant improvement to the sound, which appeared more natural, smooth, and dynamic. Probably the optimization of various input/output impedances performed by the Super Buffers is what makes the difference. *aX*

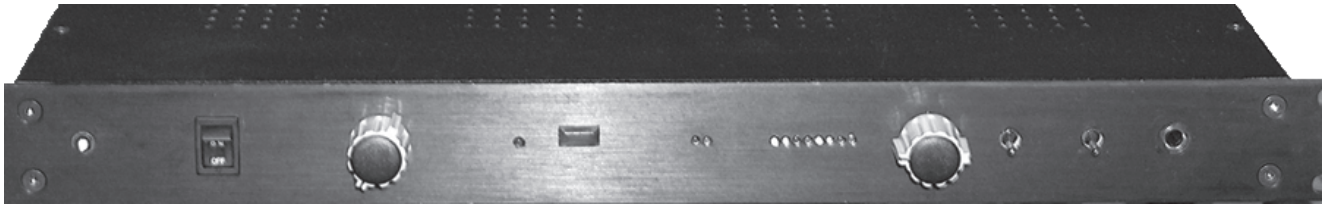
PHOTO 8: Front view.

A Chip-Driven SE Monoblock

Two years ago, I obtained some 810 transmitting tubes that I thought I'd like to use as output tubes. Jack Elliano of Electra-Print and I discussed driving power tubes in A2 mode, with Jack suggesting the need to supply power to the tube's grid to get a lot of power out. So, he came up with the idea of using the TDA2030 loaded with a stepup transformer. He sent me the first one to try. My first experiment was driving an 810 tube at 1050V, 150mA bias. I got 55W RMS out of this tube with the Electra-Print 5.5K 300mA output transformer.

Jack continued to improve his step-up transformer (the current model is A2LD7). He was kind enough to send me his updated ones. Last year, I built another pair of monoblock power amps using the Svetlana SV811-10. Because the SV811-10 is a very tough tube and I wanted to get maximum power out of it, I set the bias point at 770V 116mA. That was about 90W of plate dissipation and I got 25-27W RMS output power per channel. I lent these

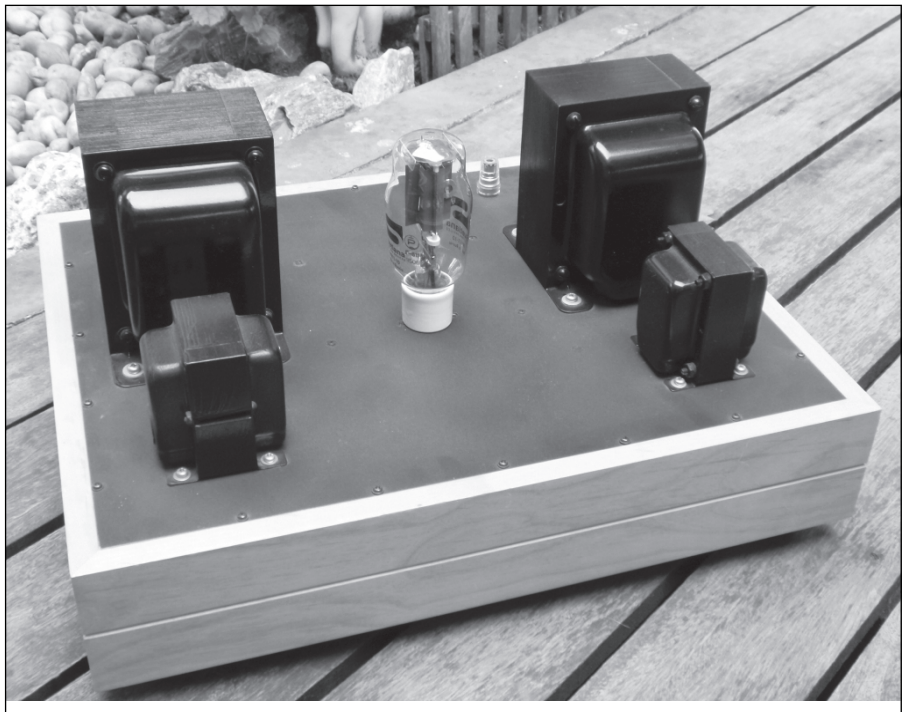

PHOTO 1: The SV811-10 monoblock power amp.

monoblocks to my friend, who owns a coffee shop, for about a year. They were turned on 12 hours a day, 7 days a week, playing all kinds of music for the customers throughout the year without any failure. The SV811-10 is a truly great tube!

I also recommend the 211 and GM70 tubes, which are easy to drive due to higher gain than other low mu triodes. You can expect to get output of 40W RMS or more from these. However, you won't be able to drive the 845 because this tube needs much more voltage drive than the 211 or GM70.

You can find an A2 article written by Jack Elliano at www.electra-print.com/techblog.php?blogid=2, as well as the sample schematic (**Fig. 1**), which is shown here with LM1875 instead of TDA2030. I chose LM1875, which I preferred over TDA2030, because it needs a minimum gain of 15, while TDA2030 needs 20. You may use SV572-10 if conservative rating is a concern. *aX*

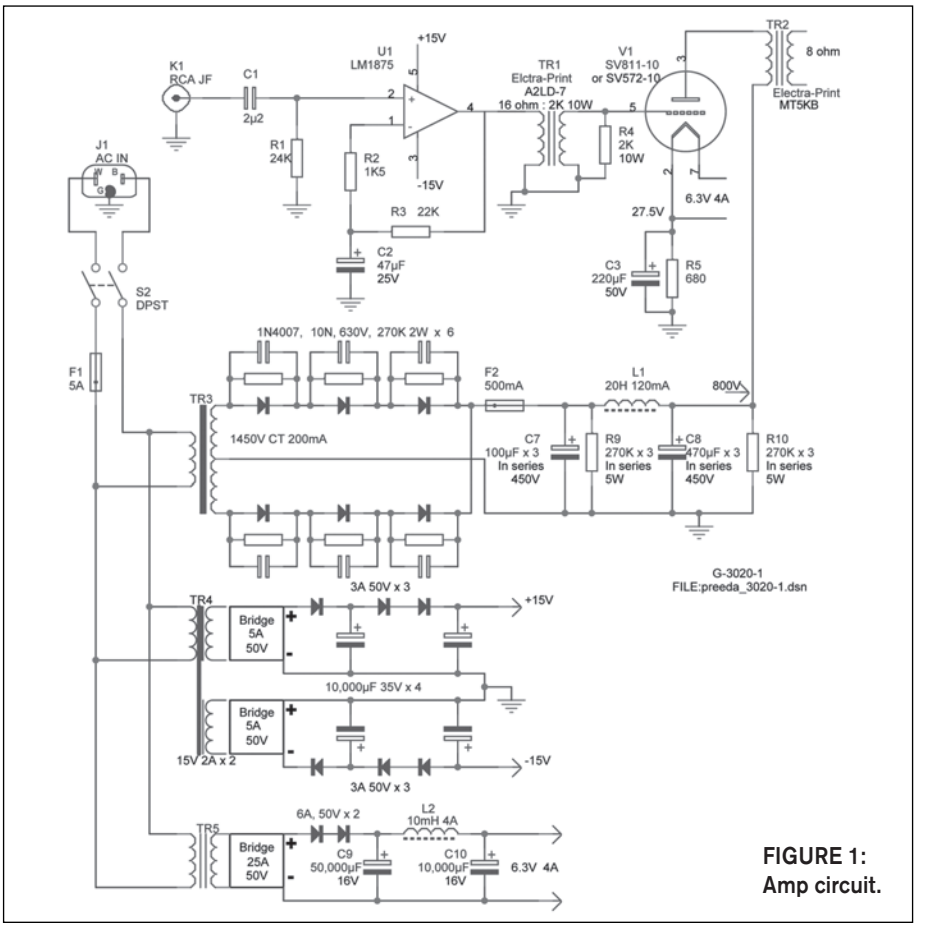

FIGURE 1:
Amp circuit.

Benchmark DAC1 USB

By Gary Galo, Regular Contributor

PHOTO 1: Front view of the Benchmark DAC1 USB. The nicely machined aluminum front panel is also available with a silver finish. The rack mount is optional.

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List Price: $1275.00

Benchmark Media Systems has developed an excellent reputation in the professional audio world for their microphone preamplifiers and digital conversion products. Unlike many "American" companies, Benchmark not only designs, but actually manufactures all of their products in the US, in the upstate New York city of Syracuse. Their DAC1 digital-to-analog converter has been available for at least four years, and has undergone some design refinements during that time. The DAC1 USB is the latest version of that product. (DAC1 Pre is the latest version).

FEATURES

The DAC1 USB is a two-channel, 192kHz, 24-bit DAC which has been designed with Benchmark's AdvancedUSB Audio Technology and UltraLock™ clock system. The headphone jacks are driven by Benchmark's HPA2™ headphone amplifier (the HPA2 is also used in Benchmark's H1 headphone amplifier).

The DAC1 USB has four digital inputs—S/PDIF via a BNC connector, a standard three-pin Neutrik gold-pin XLR connector for AES/EBU, and Toslink optical. What makes the DAC1 USB unique among outboard D/A converters is the inclusion of a USB connection. You can connect the DAC1 USB to any computer with a USB jack, allowing the user to bypass the mediocre sound cards found in most computers, using the DAC1 USB as the interface between the computer and monitoring system.

A three-position, spring-loaded front panel toggle switch, combined with three vertically mounted blue LEDs, is used for input switching. The switch simply toggles—up or down—through the various inputs. LED 1 indicates S/PDIF, 2 is AES/EBU, 3 is Toslink, and the USB input is indicated by *both* LEDs 1 and 3. As supplied, the DAC1 USB defaults to the USB input at power-up, but you can change the default input to any input by re-setting a jumper on the PC board. The input switching method is simple and well thought out.

The DAC1 USB has no power switch—the unit is intended to be left on at all times, or connected to a switched power line filter. If no digital input is detected for 15 seconds, the unit goes into standby and mutes the audio. The three blue LEDs are also used to indicate input status and errors. For example, 16 slow flashes indicate that no digital signal is present. An invalid sample is indicated by 64 rapid flashes.

When used with the three standard digital inputs, the DAC1 USB will operate at any input sampling frequency from below 32kHz to above 192kHz, at any word length up to 24-bits. The USB input functions at sampling rates up to 96kHz. You can connect variable-pitch digital sources directly to the Benchmark DAC without any external sample rate conversion. The DAC1 USB supports the Red Book de-emphasis function, at sampling frequencies up to 96kHz, making the unit truly backwards compatible.

The DAC1 USB has unbalanced RCA and balanced XLR line outputs. You can operate these either at a fixed, "calibrated" level or switch them to "variable," with output level set by the front-panel volume control. The "calibrated" levels are adjustable with 10-turn trimmers. In the variable position the DAC1 USB functions as a preamp line stage, allowing you to connect it directly to a power amplifier with the front-panel pot controlling volume. Users who need multiple line inputs may also consider the DAC1-PRE, which has an extra analog line input.

ADVANCED USB TECHNOLOGY

Benchmark devotes three pages of the instruction manual to discussion of their "Advanced USB Audio Technology." The DAC1 USB doesn't require any drivers in order to operate with Windows Vista/XP/2000 or Macintosh OS X. Digital data is transported from the computer source to the DAC1 USB in a bit-transparent fashion, without any data modification, and without any need to re-configure software.

Previous native USB audio devices have been unable to stream audio at any resolution higher than 48kHz/16-bit. Benchmark's USB technology allows native USB audio at resolutions up to 96kHz/24-bit. The DAC1 USB will automatically operate with any sampling rate and bit depth being fed to it via the USB port, without reconfiguring drivers (there aren't any!) and software. Benchmark also buffers the USB datastream to

prevent interruptions, which usually rear their ugly heads in the form of clicks and pops.

Benchmark notes that Windows 2000 and XP have a digital mixer known as a "Kmixer" (kernel audio mixer), which passes all audio before it reaches native USB devices. Kmixer contains a poor-quality sample rate converter, but Benchmark's USB technology allows Kmixer to default to a transparent mode, avoiding the sample rate conversion. Vista's sample rate converter is of excellent quality, and automatically defaults to the highest sample rate supported by the connected device. Nonetheless, you can defeat Vista's sample rate converter, if desired.

With 16-bit datastreams, the Windows volume control is bit-transparent only when set to full volume. But, Benchmark discovered that the Windows volume control is distortion free when streaming 24-bit datastreams. The DAC1 USB forces Kmixer to stream 24-bits so the Windows volume control doesn't degrade the audio quality. Benchmark also found that the Mac OS X volume control has poor quality sample rate conversion, and the system volume control degrades the audio unless it's set to maximum. OS X is capable of bit-transparent audio if the sample rate is manually set to match that of the audio being played, and the system volume control is set to 100% (use the volume control on the DAC1 USB). For more information on PC audio, go to www.benchmarkmedia.com/wiki.

JITTER CONCERNS

Benchmark is extremely concerned with the effects of clock jitter on audio performance. Better D/A converters, sample rate converters, and outboard jitter suppressors have generally employed two-stage phase-locked loops for improved jitter suppression, including my reference sample rate converter, a Parts Connection Assemblage D2D-1 (discontinued).

Benchmark claims that their UltraLock clock system exceeds the jitter performance of conventional two-stage PLL circuits.

The UltraLock system isolates the conversion clock from the digital audio interface clock, so jitter on any digital input can't have any measurable effect on conversion clock jitter. With this system, interface jitter will not degrade the conversion process. The DAC1 USB manual has an interesting and informative discussion of jitter, including two problems that are rarely discussed. One is that jitter causes phase modulation of the audio signal, producing unwanted sidebands on either side of every discrete tone in the audio signal. These sidebands are unrelated to the audio signal and are easy to hear and measure. Benchmark also says that jitter can severely degrade the digital anti-aliasing filters in an oversampling converter. Though seldom considered, they note that this effect is also easily measurable. The UltraLock system puts clock jitter below the measurement limit of the Audio Precision System 2.

DESIGN DETAILS

The DAC1 USB is not supplied with a schematic, but a look inside reveals most of the design details. Digital inputs are routed to an AK Semiconductor AKM4114 Digital Audio Interface, which includes the input receiver. The AKM4114 has eight digital inputs, and includes a low-jitter analog phase-locked loop with a lock range of 32kHz to 192kHz. The S/PDIF and AES/EBU inputs are each coupled with their own pulse transformer.

I don't see any evidence of input switching near the digital inputs—it appears that each input, including the Toslink and USB (after its own interfacing), is fed directly to one of the eight inputs on the AKM4114. This is very sensible, because the internal input switching of the AKM4114 probably results in lower clock jitter than external input switching. This capability is a compelling reason to use the AKM4114 for multiple digital inputs.

The I^2S output of the AKM4114 is fed to an Analog Devices AD1896 High Performance Asynchronous Sample Rate Converter. The AD1896 is the current flagship asynchronous converter chip from Analog Devices and boasts THD+N of -120dB and DNR (with filter) of -139dB. The AD1896 can convert any input sample frequency to any other, up to 192kHz. In the Benchmark DAC1 USB, the AD1896 is set up to convert all inputs to a sampling frequency of 110kHz, which Benchmark has found

to be optimum.

The 110kHz output of the AD1896 is fed to Analog Devices' AD1853 DAC, a 24-bit stereo, multi-bit, Delta/Sigma converter chip operating at sampling frequencies up to 192kHz. De-emphasis, if needed, is performed in the AD1853. The DAC chip's differential current outputs are fed to 5532 dual op amps, which may seem an odd choice for current-to-voltage conversion these days. The DAC1 pre uses an LM4562 op amp throughout.

With a noise level of 5nV/√Hz, the 5532's low noise level and low cost still makes it attractive to some designers. Yet in other regards, particularly slew rate and bandwidth, there are many other chips that would seem more suitable for high-performance designs in 2008. The Burr-Brown/TI OPA627 has even lower noise than the 5532—4.5nV/√Hz, a slew rate of 55V/µS, and a gain-bandwidth product of 16MHz. By comparison, the 5532 has a slew rate of 9V/µS and a gain-bandwidth product of 10MHz.

Analog Devices recommends the OP275 for I/V conversion and differential-to-single-ended conversion on page 15 of the AD1853 datasheet. Benchmark seems to be using a topology similar to that recommended by AD. Another 5532 (one half per channel), rather than an OP275, also appears to be used for the differential-to-single-ended conversion (doubling as an output buffer). The analog filter in the AD datasheet is a Gaussian type with a -3dB point of 75kHz. I really don't understand why Benchmark chose a device as dated as the 5532 for use with a state-of-the-art DAC such as the AD1853.

(Note from the Manufacturer: The 5532s were chosen because, unlike most substitutes, they do not exhibit a rise in the THD with frequency above 20kHz. This prevents intermodulation distortion caused by ultrasonic audio content.)

The unbalanced outputs are driven by National Semiconductor LM4562 dual op amps, with a 5532 used as unity gain buffers for the two channels. Similarly, the balanced outputs are also driven with LM4562s (one per channel, in this case), with a 5532 again used as the DC servo. Only one 5532 is used for both balanced output channels. I suspect that, for each channel, half of a 5532 is used as a servo amp around the first half of an LM4562, which feeds the second half of the LM4562 configured as inverting amplifier to form the other leg of the balanced output.

The LM4562 was designed as a high-performance audio device and boasts THD of 0.00003% into 600Ω loads. Slew rate is 20V/µs, gain-bandwidth product is 55MHz, and noise is exceptionally low at 2.7nV/√Hz. I would think that the LM4562 would also have been an excellent choice in the other places where Benchmark used the 5532.

The two headphone jacks appear to be driven by the same amplifier. The only difference is that one of the jacks can be configured to cut off the line outputs on the rear panel, while the other always leaves them on. The headphone amp has a pair of 5532 op amps and two Burr-Brown/TI BUF634 buffer amplifiers.

The BUF634 is an outstanding device capable of 250mA of output current, albeit with considerably more heatsinking than it's given in the DAC1 USB. Benchmark has used the five-lead DDPAK surface-mount package, with a small piece of ground-isolated PC land providing some additional heatsinking beyond the devices' own metal tab. For each channel, half of a 5532 provides the gain, buffered by a BUF634, with the other half of the 5532 operating as a DC servo amp.

(Note from Manufacturer: The PCB is designed as a heatsink. This is extremely effective, and can be shorted all day without overheating.)

Benchmark notes that the headphone amplifier design has an output impedance of 0Ω—the BUF634 is connected directly to the headset without the usual series resistors. Resistors will degrade the performance of any headphone amplifier; if the headphone does not have a flat impedance curve, the series resistor will change the headphone's frequency response. The amplifier will drive headphones with impedances as low as 30Ω, maintaining less than 0.00003% THD+N.

Fixed, three-terminal 7818A and 7918A regulators are used for the ±18V analog supplies (see Chuck Hansen's comments on supply voltages in the mea-

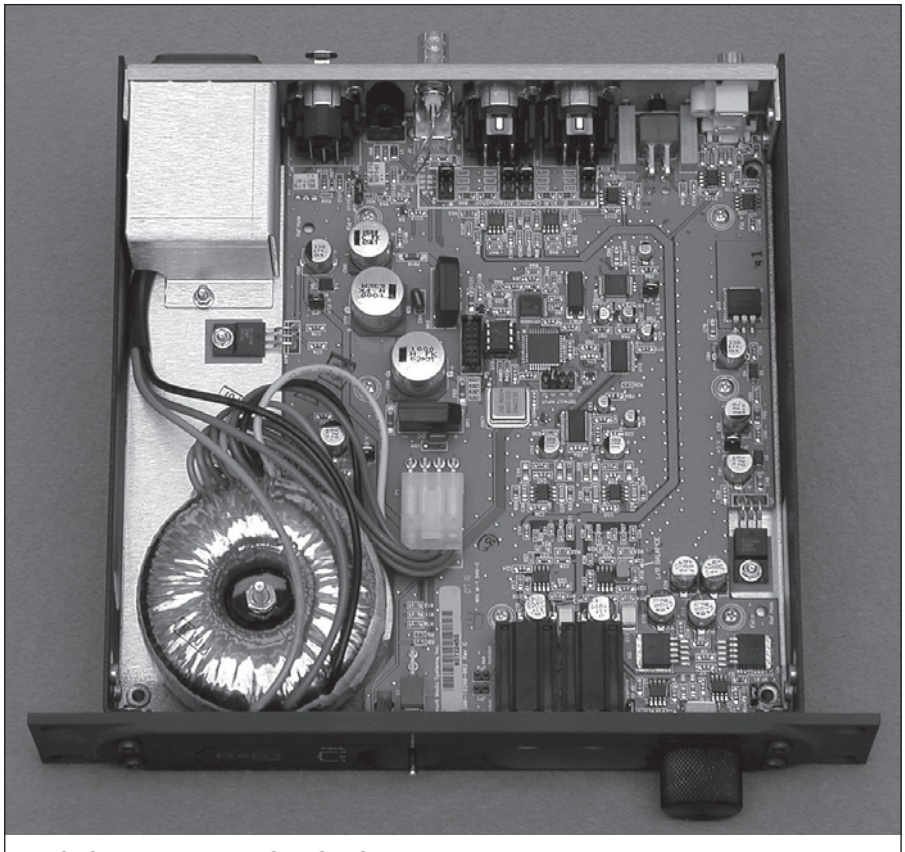

PHOTO 2: Inside the DAC1 USB. State-of-the-art digital chips are used, including Analog Devices AD1896 Sample Rate Converter and AD1853 D/A converter. Construction quality is excellent.

surements portion of this review). Two digital regulators are also used, a 7805 for the 5V supply and an LM1085 for the 3.3V supply. Separate rectifier bridges are used for the analog and digital supplies, fed by separate windings on the toroidal power transformer. Most resistors and capacitors are surface-mount "chip" type, except for the electrolytics. The DAC1 USB doesn't contain any passive parts normally thought of as "audiophile" grade.

PERFORMANCE

Chuck Hansen's accompanying measurements show the DAC1 USB to be beyond reproach. Indeed, its measured performance taxes most available test equipment, short of an Audio Precision System 2. The System 2 is used in-house by Benchmark, and the manual contains their own set of measurements performed on the Audio Precision instrument.

Given the data collected by Chuck, there's no reason to doubt the outstanding measurements published by the manufacturer (Benchmark also includes jitter tests with their measurements). The printed manual supplied with the DAC1 USB is also available on Benchmark's website. I suggest downloading this version of the manual, because all illustrations and photos are in color.

I have been using the DAC1 USB in my office playback system at The Crane School of Music, SUNY Potsdam, for about two months. During the time I've had the DAC1 USB connected to my office system, I've heard dozens of my own Crane recordings on this DAC (most made with Schoepps MSTC-5 or MSTC-6 ORTF stereo microphones, a Millennia HV3B microphone preamplifier, and Tascam DV-RW1000HD digital recorders). All I can say is that I simply hear much more in my recording than before—more detail, improved low-level resolution, better dynamics, and a bigger soundstage with more precise localization.

I also installed the Benchmark DAC in my home audio system for a week, where I compared it to my recently purchased NAD M55 multi-format player used as a standalone device. The M55 also served as the transport for the DAC1 USB. I also compared the DAC1 USB to the Monarchy M24 (reviewed in Oct. 2007 with measurements by Chuck Hansen in June 2007).

The DAC1 USB is the most transparent DAC I've auditioned. CDs and DVDs are reproduced with pristine clarity. The sonic picture is detailed and spacious. Soundstaging is incredibly precise, with a large amount of hall ambience in the rear of the stage. On the Munch recording of Dukas' *The Sorcerer's Apprentice*, the violins playing harmonics at the beginning were reproduced with a delicacy that I've not heard previously on this recording.

The cymbals in Fritz Reiner's *Pictures at an Exhibition* are crisp and inner detail in *tutti* passages is exceptional. The Mercury recording of Respighi's *The Birds* is clean, detailed, and transparent. Strings articulation is excellent in both recordings. Precision is a word that continually comes to mind in listening to the DAC1 USB.

The Benchmark DAC was definitely designed with accuracy in mind, rather than euphony. In the Mercury recording of Respighi's *The Birds*, the NAD M55 as a standalone player sounds slightly warmer and sweeter, while the Benchmark sounds more analytical and more transparent. In the Classic Records DVD of Rachmaninoff's *Symphonic Dances*, the DAC1 USB sounds more transparent and more detailed, and the M55 again warmer and sweeter.

In the Reiner *Pictures*, the DAC1 USB is leaner, with the sonic picture reproduced with exceptional precision. This is especially true in the passage for the glissandi strings in "Gnomus" where the Benchmark's localization of each string section is more precisely focused than I've ever heard. Again, the M55 sounds warmer, with perhaps a bit more weight in the bottom octave.

In general, the comparisons I've made between the Benchmark DAC and the NAD M55 as a standalone also apply to the Benchmark versus the Monarchy M24. The Monarchy M24 was designed with a bit of euphonic coloration rather than absolute accuracy. The NAD M55 as a standalone player leans more toward the Monarchy M24 than the Benchmark.

I have only two criticisms of the DAC1 USB. Tonally, the Benchmark DAC is neutral except for a tendency toward brightness. Normally, the Benchmark DAC is never fatiguing, but the bright-

ness doesn't always flatter already bright recordings such as Dorati's Mercury CD of Schoenberg's *Five Pieces for Orchestra*. The violin section in the Classic Records DVD of Stokowski's Everest recording of "Wotan's Farewell" from Wagner's *Die Walküre* is also brighter than I'd like, but the inner detail is also exceptional. It doesn't help that the Houston Symphony's strings were well below world-class standards in 1960.

Violin sections in orchestral recordings are not as sweet as the real thing. Given that the digital end of this DAC is practically beyond reproach, I can't help asking whether the DAC1 USB would sound even better if Benchmark would replace the 5532 op amps with another device. I think that the National Semiconductor LM4562 used for the analog output stages would be an improvement over the 5532 in other locations as well. That said, I would never have guessed, based on listening, that there were any 5532 op amps in this device had I not known better.

I've never heard a product with 5532 op amps sound this good. Could the Texas Instruments 5532 op amps be audibly superior to the types made by NJR and other Far East sources, and used in so many mid-fi products? Or, is the digital end *so* good that the use of 5532s isn't that detrimental? Still, I wonder what the Benchmark DAC would sound like with LM4562s used throughout.

Connecting the DAC1 USB to a computer couldn't be simpler. Plug it into a USB port and Windows will recognize it within seconds. Sonically, the difference between the Benchmark DAC and my computer sound card is astonishing. If you do a lot of digital editing, the ability to really hear what's on your editor will be most welcome. I have permanently installed the DAC1 USB with my computer editing system at work.

JITTER ELIMINATION?

I still use my Parts Connection D2D-1 Sample Rate Converter between the NAD M55 and Monarchy M24 DAC. Despite the dual phase-locked loop around the input receiver, and excellent jitter suppression properties of the built-in asynchronous sample rate converter, the D2D-1 has never eliminated sonic differences between various transports. My NAD M55 is the best-sounding transport (and the best-sounding stand-alone player) that I've owned. As a transport feeding the D2D-1, it is noticeably cleaner and more transparent than my older Onkyo DV-SP800 multi-format player, which, in turn, is better than the comparatively foggy Marantz PMD-340 CD player. As a transport, the Marantz sounds quite unrefined compared to the Onkyo and, especially, the NAD. Jitter is the only reasonable explanation for differences in the sound of various transports, and as good as the D2D-1 is, it must not be eliminating jitter, just reducing it.

Enter the DAC1 USB. My listening evaluations would indicate that Benchmark's claims for the jitter performance of the DAC1 USB are justified. I spent one evening comparing the NAD M55 and the Marantz PMD 340, with the NAD connected to the Benchmark's S/PDIF coax input (with D.H. Labs D-75 digital interconnect fitted with Canare 75Ω BNC connectors) and the Marantz connected to the AES/EBU input (D.H. Labs D-110 digital interconnect and Neutrik gold-pin XLR connectors). This made it easy to go back and forth between the two transports by simply switching the DAC1 USB's input selec-

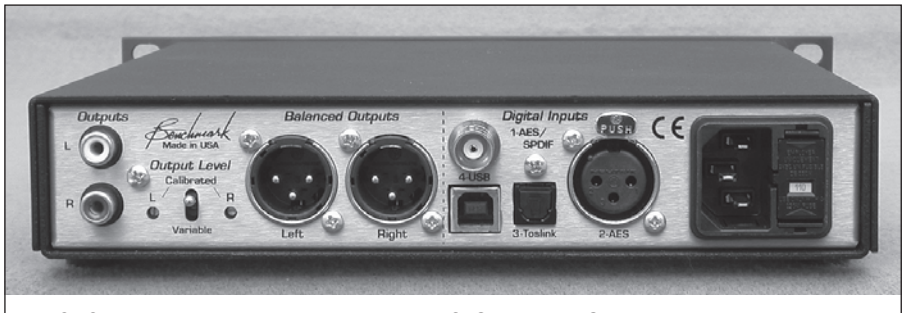

PHOTO 3: Rear view of the Benchmark DAC. S/PDIF, AES/EBU, and Toslink optical inputs are provided, along with a USB connection. Balanced and unbalanced analog outputs are also included.

tor and moving the reference CD to the other transport.

For the first time since I began evaluating digital hardware, I can honestly say that I found it difficult to tell the difference between two transports. The old, relatively unrefined Marantz sounded excellent connected to the DAC1 USB—spacious, detailed, and refined. There were times when I thought that the NAD M55 sounds slightly better—perhaps a shade better defined with the stereo image a bit more precise. Occasionally, I thought the Marantz sounded a bit mushy and the NAD crisper, the Marantz a little edgy in the treble and the NAD smoother. Then, I'd run another comparison and not be so sure of my conclusions.

The bottom line is that the Benchmark DAC reduces jitter to vanishingly low levels—measurably and audibly—to the point where even a transport that would normally not qualify as "audiophile" sounds excellent. If there are differences between my NAD and Marantz transports, those differences are very small, indeed. This is most impressive.

It almost goes without saying that any outboard jitter suppressor connected between your transport and the DAC1 USB is superfluous. I inserted my D2D-1 between the NAD M55 and the DAC1 USB to see whether this had any effect on the sound. I left the D2D-1 in the "Transparent" mode, so it was functioning strictly as a jitter suppressor, without any sample rate conversion. By simply swapping digital cables, it was easy to remove and re-connect the D2D-1.

The results were similar to comparison of transports, except this time I really was convinced that there was an audible difference, even though it was very subtle. The D2D-1 degraded the sound very slightly. It caused a slight loss of focus and detail and put a very subtle veil over the sound. It also softened the treble slightly. Benchmark has shown that it's possible to design a DAC so that external jitter suppression devices are no longer necessary or desirable.

TABLE 1 Reference Recordings

CDs

Dukas: *The Sorcerer's Apprentice.* Boston Symphony Orchestra conducted by Charles Munch. RCA Victor Living Stereo 68978-2.

Ravel: *Alborada del Gracioso.* L'Orchestre de la Suisse Romande conducted by Ernest Ansermet. London 433 717-2.

Rimsky-Korsakov: *Scheherezade,* Op. 35. Chicago Symphony Orchestra conducted by Fritz Reiner. RCA Victor Living Stereo 68568-2 (UV22-Encoded Limited Edition Gold CD version of 68168-2).

Mussorgsky/Ravel: *Pictures at an Exhibition,* especially track 2, "Gnomus." Chicago Symphony Orchestra conducted by Fritz Reiner. RCA Victor Living Stereo 68571-1 (UV22-Encoded Limited Edition Gold CD version of 61958-2).

Schoenberg: *Five Pieces for Orchestra,* Op. 16. London Symphony Orchestra conducted by Antal Dorati. Mercury Living Presence 432 006-2.

Respighi: *The Birds.* London Symphony Orchestra conducted by Antal Dorati. Mercury 432 007-2.

Wagner: *Der Ring des Nibelungen,* especially "Siegfried's Death and Funeral March" from Götterdämmerung (CD 4, Tr. 10-11), and the "Forging Scene" from *Siegfried* (CD 2, Tr. 3-5). Birgit Nilsson, Wolfgang Windgassen, et al. Vienna Philharmonic Orchestra conducted by Georg Solti. Decca 455 555-2.

DVDs

Rachmaninoff: *Symphonic Dances,* Op. 45. Dallas Symphony Orchestra conducted by Donald Johanos. Classic Records DAD 1004. (96kHz, 24-bit PCM transfer of 30-ips analog tape, or possibly a 15-ips copy, engineered by David Hancock).

Wagner: *Die Walküre*–Wotan's Farewell and Magic Fire Music. Houston Symphony Orchestra conducted by Leopold Stokowski. Classic Records HDAD 2029. (192kHz/24-bit DVD-Audio transfer of original Everest 35mm 3-track tape).

CONCLUSIONS

The Benchmark DAC1 USB is an impressive achievement—an exceptionally refined DAC that offers remarkable performance at an extremely reasonable price. Benchmark's UltraLock circuitry sets new standards for jitter suppression, and the USB interface provides a significant improvement over monitoring via computer sound cards. Benchmark offers a 30-day, risk-free trial period. If you're in the market for an outboard DAC, you can't afford not to try one. *aX*

Manufacturer comments made by

Michio Aida
Advertising Director
Benchmark Media Systems, Inc.

Measurements for the Benchmark DAC1 USB

By Chuck Hansen

SETUP

I used a 75Ω coax cable between my Alesis ML-9600 Masterlink recorder digital output terminated with a 75Ω BNC female to male RCA adapter, and the Benchmark DAC1 USB AES/SPDIF 75Ω BNC Input 1. While the DAC1 comes with an RCA female to 75Ω BNC male adapter to use with a conventional digital audio cable that is terminated on both ends with RCA plugs, I prefer to treat the digital AES/SPDIF signal as an RF signal (which it is) and use the 75Ω BNC terminated cable with my BNC to RCA adapter. I applaud Benchmark for providing the 75Ω BNC digital connector on the DAC1, and I would like to see it become standard in lieu of the inferior RCA jack.

CAUTION: Never use a 50Ω BNC plug on the DAC1 digital 75Ω BNC input jack. The center pin of a 50Ω BNC plug is slightly larger in diameter than that of a 75Ω BNC plug, and you could permanently expand the 75Ω female input pin enough that it will no longer make good contact with a 75Ω BNC plug center pin.

In order to provide a suitable warm-up before testing, I played over one hour of 16-bit and 24-bit audio test tracks. The DAC1 USB easily locked onto 16-bit 44.1kHz digital signals, as well as 48kHz, 88.2kHz, and 96kHz 16-bit to 24-bit signals (I don't have any 192kHz digital test data). There is no indication of the received data bit depth or sample rate on the front panel. If there is an error in acquiring digital data, the input LEDs on the front panel will flash to reveal various data error codes as listed in the very thorough instruction manual.

The DAC1 USB provides many internal jumper-selected settings, but I left them all as-received. I also left the rear panel Output Level switch in the calibrated (fixed output) position. I did not attempt to control the DAC1 USB by means of a computer connected to the USB port. The unit uses linear IC voltage regulators rather than a switching power supply, so the case temperature increased from 25° C to 36° C at the bottom of the chassis.

The DAC1 USB preserves normal polarity from all its analog outputs. The RCA jack analog outputs produced 2.004V RMS (L) and 2.007V RMS (R) at 1kHz, or an insignificant 0.03dB maximum above the Red Book Standard of 2V RMS. The balanced XLR outputs both produced 1.267V RMS (pin 2 hot) with the internal attenuation jumper left unchanged. The headphone output at the maximum volume control setting is 2.87V RMS at 1kHz, with about 0.4dB imbalance between the two headphone jacks at the unity gain setting. The volume control does not operate on the balanced or unbalanced RCA outputs in calibrated mode.

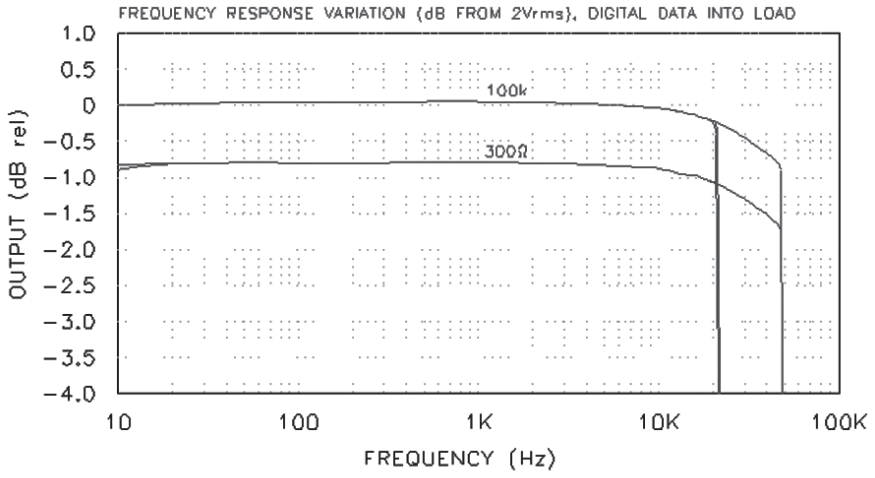

FIGURE 1: Frequency response.

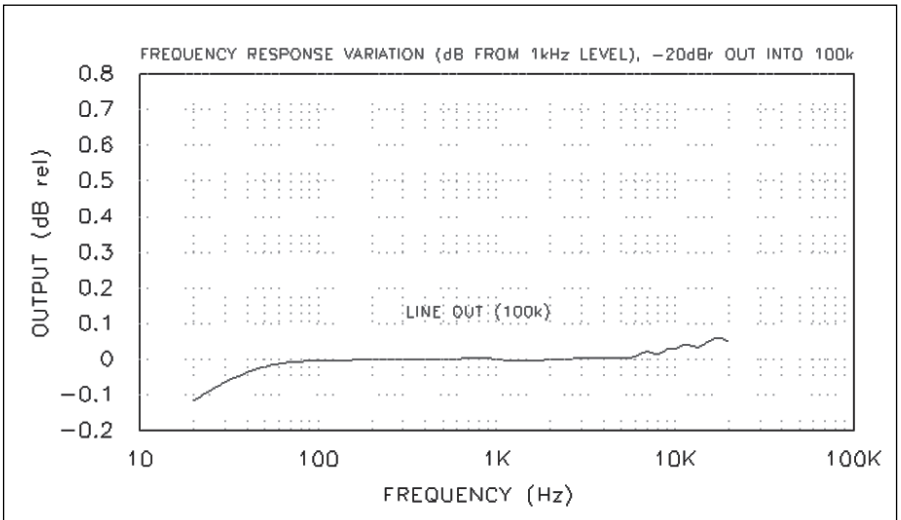

FIGURE 2: De-emphasis error.

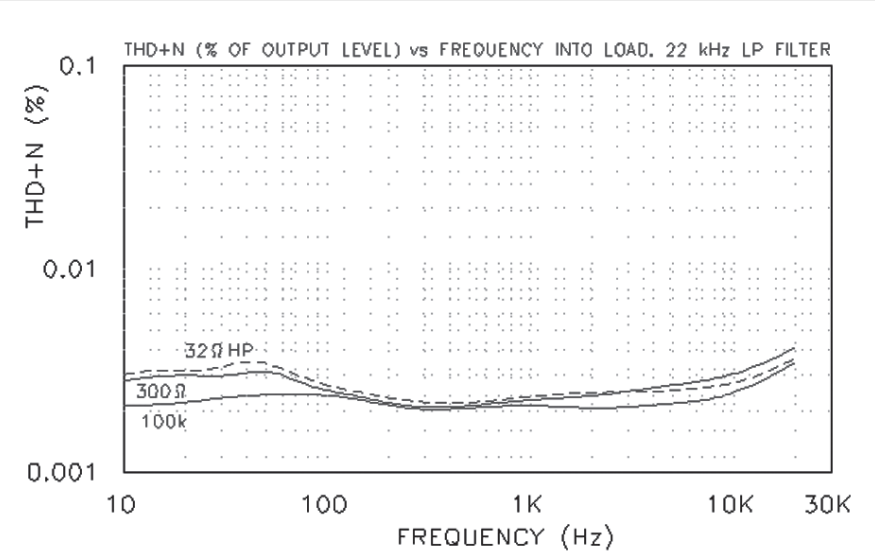

FIGURE 3: THD+N vs. frequency.

MEASUREMENTS

The output impedance of the RCA jack analog outputs measured 29Ω at 20Hz and 1kHz, increasing slightly to 31Ω at 20kHz. The balanced XLR outputs measured 133Ω over the audio band (this will vary with the attenuator jumper settings). The front panel headphone output measured less than 1Ω with the volume control set for unity gain.

Hum and noise measured a low -110dB relative to 2V RMS, and -116dB A-weighted. The analog outputs had a negligible DC offset of 3.4mV. Separation between the stereo channels was about -105dB at 10kHz.

I recorded the frequency response shown in **Fig. 1** from the RCA analog outputs into 100k and 300Ω loads, using 16-bit/44.1kHz Red Book data as well as 24/48kHz and 24/96kHz data. The DAC1 USB response drops off by -0.23dB at 20kHz (44.1kHz data), and -0.81dB at 43kHz (96kHz data). When I decreased the line stage load to a difficult 300Ω, the frequency response fell by -0.84dB at 1kHz, with a slight droop at 10Hz with 16/44.1kHz data.

Figure 2 shows the deviation from flat response when I fed pre-emphasized Red Book 16/44.1kHz digital data to the digital input. The de-emphasis correction is accurate ±0.09dB, 20Hz to 20kHz.

The THD+N versus frequency for the DAC1 USB RCA analog outputs using 16/44.1kHz digital data is shown in **Fig. 3**. I used the steep distortion test set 22kHz low-pass filter to remove out-of-band noise. Note that the results shown here are just about at the low limit of my distortion measurement capability. Decreasing the load from 100k to 300Ω showed only a slight increase in THD+N. The headphone output with a 32Ω load is shown as a dashed line. This is all-around excellent performance.

Figure 4 shows THD+N versus output voltage for the RCA output into 100k at 1kHz. The THD+N level increases in a straight line as the output level decreases, up to the 0.5% vertical axis limit in my graph.

The DAC1 USB output spectrum of a 50Hz sine wave at 0dBfs into 100k is shown in **Fig. 5**, from zero to 1.3kHz. The THD+N measured 0.0027%, with no visible harmonics above the -105dB noise floor of my spectrum analyzer. Raising the frequency to 1kHz and expanding the spectrum range to 20.5kHz shows a similar benign result (not shown). The distortion residual waveform for a

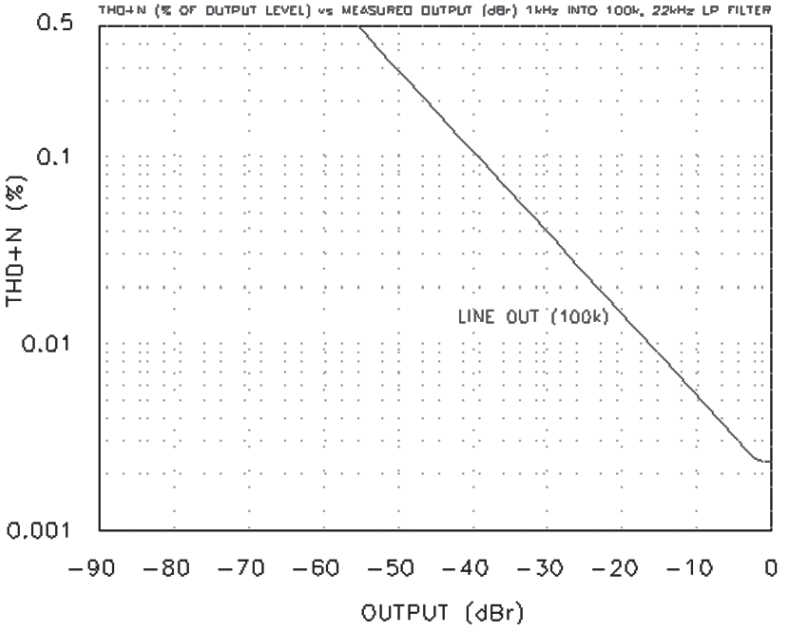

FIGURE 4: THD+N vs. output.

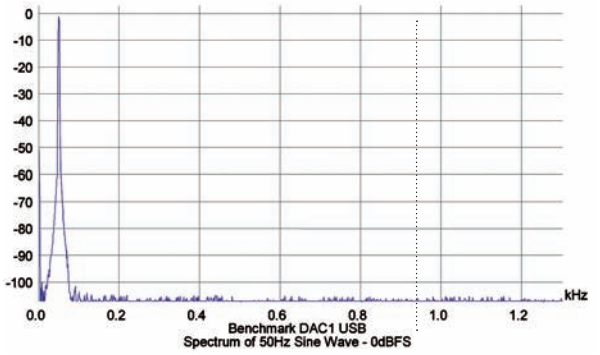

FIGURE 5: Spectrum of 50Hz sine wave.

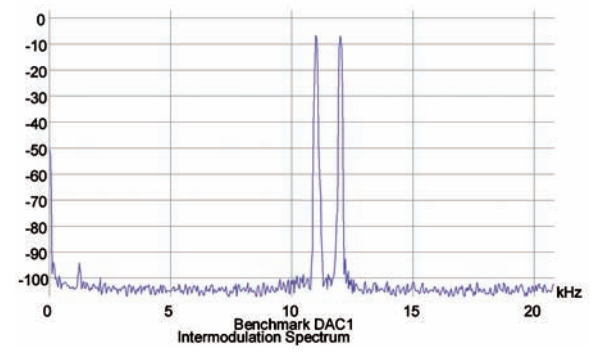

FIGURE 6: 11kHz + 12kHz intermodulation distortion.

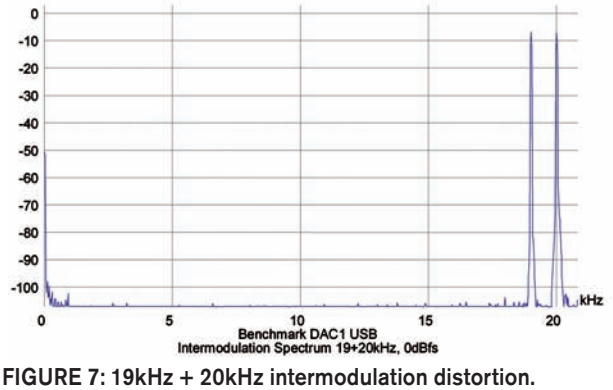

FIGURE 7: 19kHz + 20kHz intermodulation distortion.

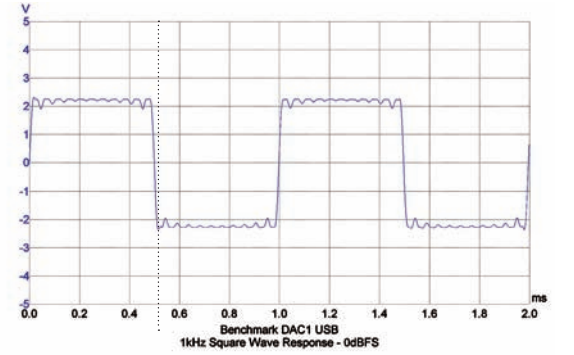

FIGURE 8: 1kHz square wave, 16/44.1 data.

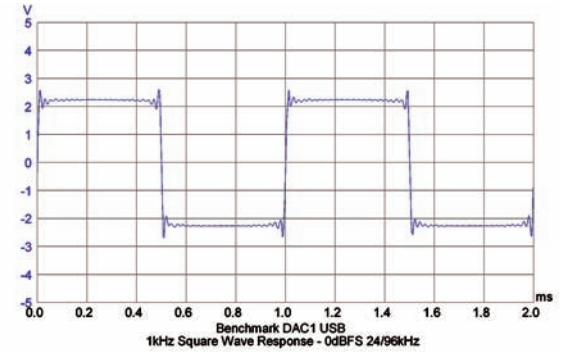

FIGURE 9: 1kHz square wave, 24/96 data.

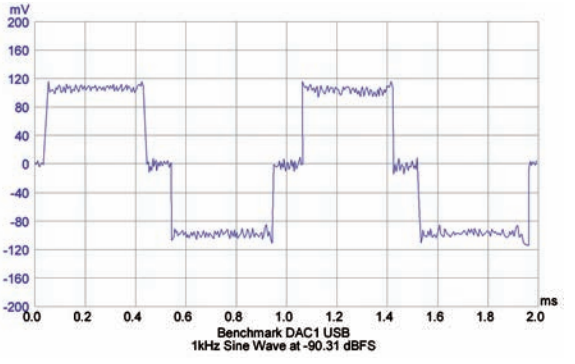

FIGURE 10: 1kHz sine wave at -90.3dBfs with 16/44.1 data.

0dBfs 1kHz sine wave, taken after the distortion test set notch filter, showed no discernible distortion harmonics or AC power line artifacts and just a low level of noise (also not shown).

Figure 6 shows the DAC1 USB output spectrum reproducing a combined 11kHz + 12kHz intermodulation distortion (IMD) signal at 0dBfs into 100kΩ. The 1kHz IMD product is -95dB (0.0018%). Repeating the test with a 19kHz + 20kHz CCIF IMD test signal (**Fig. 7**) resulted in a -102dBfs 1kHz product (0.0008%).

I applied a 1kHz square wave at 0dBfs, 16/44.1kHz digital data to the DAC1 USB and monitored the resulting scope trace from the DAC1 USB (**Fig. 8**). The Gibbs Phenomenon ringing is unremarkable except for the slight leading edge damped oscillation. Increasing the digital data to 24/96kHz (**Fig. 9**) showed a better representation of the square wave, with a bit higher peaking on the leading edges.

The reproduction of a 1kHz sine wave at -90.3dBfs with 16/44.1kHz data is shown in **Fig. 10**. The sine wave at this ±1 bit signal level appears as three distinct levels, just as it should. Increasing the data resolution to 24/96kHz at -90.3dBfs produces the fine sinusoidal shape shown in **Fig. 11**.

The DAC1 USB produced an excellent set of measurements. The instruction manual includes 15 pages of test graphs provided by an Audio Precision System 2 Cascade.

POWER SUPPLY CONCERN

I removed the cover to take a look at the PC board since the instructions to get internal access were in the instruction manual. I was a bit alarmed to see that the analog IC regulators are ±18V (7818A and 7918A). This is fine for the 5532 op amps (±20V maximum), but the LM4562 op amps and the BUF634s have an absolute maximum rating of ±18V. I didn't see any other local lower voltage PC board mounted regulator chips, nor a -5V regulator IC that might be teamed with the 7805A for a ±5V supply for

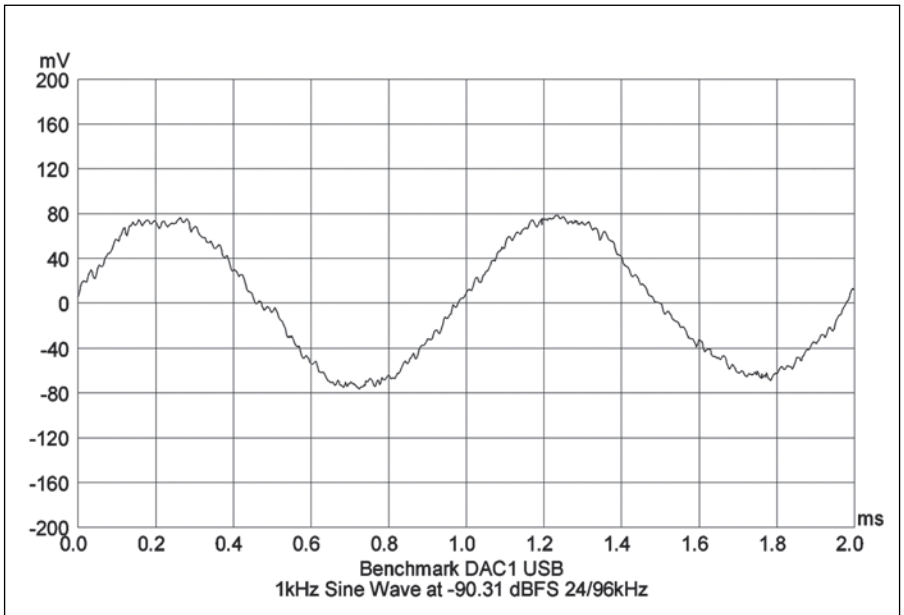

FIGURE 11: 1kHz sine wave at -90.3dBfs with 24/96 data.

the op amps and buffer ICs. This is not really possible, because the maximum headphone output is specified to be +21dBu (8.69V RMS) and the balanced outputs are specified to be +29dBu, or 21.8V RMS.

I did not attempt to measure the actual voltages at the op amp supply pins, but the ±18V regulators raise a red flag for me if those parts are actually used at ±18V (or even higher, since the tolerance on 78xxA and 79xxA regulators is 2% at 25° C, or 4% over the full temperature range). My absolute safe maximum for any analog IC is 90% of maximum rated, or ±16.2V in this case.

The reason I use 90% is because that's the maximum number that both MIL-HDBK-217 and the Telecordia (Bellcore) TR-332 "Reliability Prediction Procedure for Electronic Equipment" have in their MTBF calculations. I work pretty close with both documents and talk with the guys who have cognizance over the military handbook, and they say if you go above 90% for semiconductors, you are conducting experiments, not providing suitable power for long-term reliable operation of analog ICs and semiconductors. Calculated MTBF effectively becomes zero hours above 90% Vcc. Except for passive phono preamps that require lots of headroom to make up for the loss through the EQ stages, there is rarely any need for more than 2V RMS output to a power amp, so ±15V DC rails should be fine. *aX*

The Munich Hi-End Show 2008

A tribute to two-way analog stereo.

I'm not really a hi-fi show guy. They are fine if you want to get an idea of next month's crop in the hi-fi shops, but I want something different. I want to be able to compare—sound-wise—different systems, different components, different concepts. But, first, most shows are not set up for that. The rooms are often not ideal to judge the sound reproduction.

Second, you would need to make the effort to take your own CD, and unscrupulously force the people who run the demos to put on your music and let you handle the level control. There's bound to be some serious opposition to that. So, don't expect me to extol the virtues of the umpteenth "new and improved" component with features that defy physics. Still, the show had some pleasant surprises and made some progress to accommodate the needs of serious audiophiles. Two events got my attention in that respect.

PHOTO 1: German *Audio* magazine's ultimate source reference.

LISTENING TESTS

The first was a demo set up by the German journal *Audio* (**Photo 1**). They repeated some of the tests they described in their May issue. The experiment used two different systems—one very hi-end (and very expensive!), the other an "affordable" system. Amplification on the high-end system was by Charles Hansen's AYRE, and the conversion of electrical to acoustical power was done by large TAD speakers with "the world's only full beryllium tweeter." The "affordable" system had amplification by Meridian and speakers by KEF.

But the real interesting part was the choice of sources (hold on): LP, DVD-A, streaming audio from a hard disk, and directly from a 30+ year-old REVOX analog studio tape machine with a direct copy from the master tape. The DVD-A and the hard disk also had direct copies from the master tape, digitized at 24 bit, 192kHz by Keith Johnson using a Pacific Microsonics A/D converter.

For more on the Munich show, go to www.audioXpress.com.

The Rocky Mountain Audio Fest

Perfect Sound forever?

The fifth edition of the annual Rocky Mountain Audio Fest (RMAF) took place in The Mile High City last October.

Success can be overwhelming. What started in 2004 as an audio get-together organized by Alan Stiefel, Marjorie Baumert, and Ron Welborne, supported by volunteers from the Colorado Audio Society, has, within five years, expanded to what is perhaps the largest consumer hi-end audio show in North America. After the first few years, a dedicated organization was set up, headed by Stiefel and Baumert. Al is well known as the owner of Red Rock Audio in Denver, manufacturing a wide range of audio electronics and speakers at the very highest fidelity level.

"Let me be clear," Al said, "without many 'audiophools' behind the scene, we could not have pulled this off. We have a dedicated group from the Colorado Audio Society (including friends and family) who volunteer their time during the weekend starting on Thursday. We also have spouses of attendees who pitch in while their husbands attend the show. Because of the size the show has become and the extensive directory we publish, we hire part-time help just prior to the show to take some of the burden from us. There's a myriad of things to take care of for a show with over 160 dealer demo rooms in two adjacent hotels and a whole floor of vendors offering things from rare LPs and CDs to cables, connectors, speakers, and speaker system components."

A show with over 160 demo rooms is too large to cover in detail, so you need to make a choice of which to visit. This year my interest was more in speakers than in electronics. Having just bought a pair of Emerald Physics CS2s, which were a hit at RMAF 2007, I auditioned the new CS1 and several other systems.

Clayton Shaw of Emerald Physics introduced his flagship design, the CS1 (**Photo 1**), which sports four large woofers per side (double the CS2), and a better high-frequency waveguide and driver. This is an important improvement; while the CS2 is a fine speaker, the rather high crossover frequency requires the woofers to reproduce the lower midrange. In the CS1, not only is the crossover frequency lower, but double the number of woofers means that they have only half the excursion (and half the distortion). The CS1 bass reproduction was strong, fast, balanced, and uncolored, and belied any concerns about low-frequency reproduction from open baffles. This is a very high-end speaker system for a very reasonable price.

PHOTO 1: Emerald Physics' new flagship, the CS1 (company photo).

For continued coverage of the Rocky Mountain show, visit our website at www.audioXpress.com.

Burning Amplifier 2008

An Audio DIYer's Dream Come True!

The BAF event is organized by a group of people who meet regularly at what is perhaps the largest online DIY audio forum, www.diyaudio.com, with close to 100,000 registered members. On the forum, each has a nickname, and at the BAF, people were given stick-on labels with their nickname and real name. It is always a pleasure to meet someone in the flesh with whom you have had conversations online. You feel as though you know that person well, even if you haven't seen him before. The audio hobby allows you to (sometimes heatedly) discuss your opinions and experiences with other DIY audio fans, but you also exchange something of your personality and character in the process.

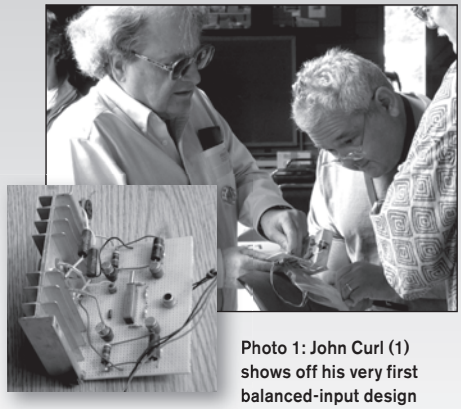

Photo 1: John Curl (1) shows off his very first balanced-input design (inset) to Scott Wurcer that landed him a job at Mark Levinson in the late '60s.

Almost 200 participants brought their projects to the show, ranging from small items such as a tube amp supply regulator (yours truly), to refrigerator-size (but much heavier) subwoofers with 12Hz cutoff frequency. This particular subwoofer, built by a member that goes by the nickname of DSP_Geek (Francois Souchay), was used to reproduce the songs of blue whales, in the very lowest audio octaves. People from the other rooms came running asking whether anyone else felt the earthquake. Projects on display varied from technically sophisticated DSP systems, to equally sophisticated CNC/CADCAM power amplifier constructions. Styling was straight no-nonsense tech to something that was called "steam punk." But even in this case, good design principles were in evidence, for instance, making the enclosure non-resonant and non-cubic.

The organizers chose a venue that allowed several activities to take place at once, which was an improvement over last year. There was one room for presentations and demos, while systems were displayed in another room. In a third large room people could hang out, have lunch, and attend the raffle and auction.

For more coverage of the BAF, visit our website at www.audioXpress.com.

ELEKTOR DEBUTS IN USA

Elektor is coming to North America. This well-respected electronics magazine, with editions in English (UK), Dutch, French, German, and Spanish, is now available through its distributor, Audio Amateur. Each project published in the magazine has been built, tested, and approved by the magazine's team of electronic engineers. To subscribe, contact Elektor, PO Box 876, Peterborough, NH 03458, 888-924-9465, **www.elektor.com/usa.**

PIPE DESIGN

I'm working through a design based on Cornelius Morton's article "A Hegeman Subwoofer," published in the December 2003 edition, and have two questions.

If I wanted to extend the design to six pipes from four, as originally published, should the resonant frequency of the additional pipes be placed at the square root of 2 multiple? This would give a set of frequencies at Fo, 1.4*Fo, 2*Fo, 2.8*Fo, 4*Fo, and 5.6*Fo. Or would it be better to use something different, such as the third root of 2 for a multiple, which would yield a set of frequencies at Fo, 1.26*Fo, 1.59*Fo, 2.00*Fo, 2.52*Fo, 3.17*Fo?

Would it be better to scale the area of the tunnel such that the pipe Qs occur over a narrower range, which might make tuning easier? I realize that this would complicate the construction of the cabinet.

Rob Weinstock
rweinstock@archstoneconsulting.com

Cornelius Morton responds:

Thanks for your interest. The original Hegeman model ones were a six tube design that was subsequently changed to a four tube configuration to accommodate ease of tuning the Q of the individual tubes. If you examine the typical free-air impedance curve for a subwoofer driver, the area from one octave below Fs to one octave above Fs is the area that requires control to properly damp the value of Q below one.

Considering this range, then, a four tube design requires that the individual Qs be 2.5 to 3, and for a six tube design the Qs should have a value between 4 to 5, the value of Q being inversely proportional to the spacing between the tube frequencies. The Q values required for the six tube design approach the maximum values that are obtained in a tube of this design leaving little room for adjustment; the Hegeman ones did not use lossy material to control the Q values. The required Q value may be decreased slightly by increasing the frequency range that the tubes control, usually at the high end.

For example, if the driver requires control from 17Hz to 57Hz, the range of tuning could be from 20Hz to 75Hz. But doing this comes at a price. Resonance control is accomplished by inserting controlled resistance into the electromechanical circuit of the driver. When this is extended past the resonance hump, the efficiency of the driver at frequencies beyond the hump is decreased.

Determining the frequency spacing for a six tube design. As an example, I will use the 20 to 75Hz range mentioned before and an Fs (free-air resonance) of 28Hz. Let

$20x^0$ = 20Hz and $20x^5$ = 75Hz. Then x^5 = 75/20 = 3.75 and x = $3.75^{(1/5)}$. Then x = 1.3026. The frequency values of the tubes are, starting with x^0, 20Hz, 26.05Hz, 33.94Hz, 44.2Hz, 57.58Hz, and 75Hz. Typical Qs will be 3.7, which will allow for some adjustment. The value of Fs is very close to the lower edge of my target range, which is 0.8 × 33.94Hz to 1.2 × 33.94Hz. This is easily adjusted by lowering low frequency from 20Hz to 19Hz, which sets the third tube to 32.24Hz and the last tube to 71.25Hz, or x may be re-evaluated using 19 and 75Hz.

Tube area. I have found that setting the total mouth area of the tubes to 80% of the effective cone area of the driver provides good results. The general effective cone area for a 10″ driver is around 60 in^2, which would result in a tube area of 10 in^2 for a six tube design and 15 in^2 for a four tube design.

General. The free-air resonance, Fs, has been used throughout this discussion. As the impedance of the tubes as seen by the driver is a resistance instead of a reactance as seen in most other designs, the resonant value does not change when the driver is mounted in the box, thus Fs is the proper design value. A small increase in damping efficiency may be obtained by adding another divider in the plenum. Use a divider similar to the one in the design but on the opposing wall, from a line just clearing the outer edge of the 58.8 tube up to the top edge of the plenum.

The original corner reflectors were made slightly narrow to increase the tube bandwidth—I found this to be unnecessary as the angled stops are sufficient. Increase the design width from 2.18″ to 3.888″. The tuned tubes also operate as a resistance that is an odd harmonic of the design frequency, phantom tubes.

For example, the 21.5Hz tube acts as a phantom tube at 64.5Hz, and the 30Hz tube shows a phantom tube at 90Hz. Both of these phantom tubes provide an additional bit of damping, dependent upon the Qs at which the parent tubes are set. At even harmonics, primarily the second, the tubes act as though the mouths have been closed, reducing the volume that the driver is working into. The 21.5Hz tube shuts off at 43Hz and the 30Hz tube shuts off at 60Hz. This increases the efficiency of the tubes operating near or at those shutdown frequencies. Because these effects depend on the Qs of the parent tubes, they result in interactions during tuning.

CORRECTION

There seems to be an error—probably trivial—in Table 1 of Dennis Colin's article, "Tweaking the Inverse RIAA Network" (August 2007). The correct value for the enhanced RIAA compensation at 20kHz should be -18.979dB, not -18.975dB as illustrated. I checked the math three different ways. All of the other values, save this one, would appear to be correct to 0.001dB.

Jack Walton
Short Hills, N.J.

Dennis Colin responds:

Mr. Walton, you are correct. While no one could hear 0.004dB, especially at 20kHz, I sincerely welcome the pointing out of any mathematical error. Plus, I needed the hand calculator exercise (using rectangular to polar conversions on my TI-30Xa calculator). My corrected result is 18.9786513512dB for enhanced RIAA equalization at 20kHz, re 1kHz. Using a 10-digit calculator, but storing

12 digits, I obtained the last two digits by subtracting 18.978 and then multiplying by 1000. So there's my result to one tenth of a nano-dB.

Thanks for your interest. By the way, you can view the completed Groove Master phono preamp on my website (colinelectronics.com).

SPEAKER SETBACK

I have wondered about speaker setback and its relative location in 3D space ever since I built a three-way system and separated the mid-tweeter assembly from the bass. I mounted my final D'Appolito MTM configuration in a very small, rigid box (the mids had an enclosed back), and the bass was a 90 liter acoustic suspension, also very rigid with internal hardwood/softwood rods bracing across all walls, with Butterworth three-way active crossovers.

I installed the bass units into the front wall, tight to each corner and on a fireplace mantel, the bass driver center about 55″ off the floor. The front faces of the bass units were about 8″ away from the wall, to use the "Allison effect." The MTM boxes were at ear height, mounted on a 1″ steel pipe spiked to the floor and braced to the sides with a gate-like arm to allow them to be moved away for other family reasons such as cleaning or enjoying a fire. The front-to-back location was defined with a rigid rod that locked to the mantel and the speaker.

By carefully listening to many disks to determine the spacing between the bass and MTM, I finally settled on 49″ away from the front wall, 42″ from the sides, with the bass to MTM distance about 55″ on the diagonal. I measured the output of each driver and balanced them with their crossovers, from several locations, and checked the phase at each crossover notch. I spaced the two mids according to the frequency at crossover, and, apart from a slight "hooded" quality from the mids (Phillips 2″ dome midrange), it has been the best system I have heard in my listening room, and certainly as good as any commercial system I've heard in friends' homes.

The amplification included homemade MOSFETs 36W/A with Darlington power supplies and Plitron toroids in a separate chassis, 6 amps/4 power supplies for the MTMs, and a hefty Nad 200W/channel for the bass. These were built before the advent of commercial subwoofers being added to music systems, when surround sound was analog!

I carefully analyzed the resulting sound using many music sources, with many critical listeners, and at no time did we sense that there were disconnects or anomalies. I reasoned that in live music, with several performers, the original sounds emanated from diverse locations, yet in a simple system they all are projected from one place.

One obvious artifact was heard in the original design when the front panel of the MTM box was at least twice as large: a blurring of the treble. I reduced it somewhat with carpet felt rings around the tweeter, but after I used the small, narrow front baffle, and adjusted the center distances according to the D'Appolito calculation, that disappeared.

I sadly built into the amps a fatal flaw—no fuses—and when our local power company dropped the neutral line onto a hot line, the spike blew out the power supplies, allowing the rails to see 120V AC, and poof, the voice coils on one channel were toast. I obtained replacement mids from one of your contributors (many thanks, Darcy), and I'm gradually re-designing the power supplies and amps.

David Mayfield
david.mayfield@rogers.com

DIY AMP

Looking at the amp (it looks like my winter audio project) presented by Joseph Norwood Still ("A $450 Triode/Ultralinear Integrated Stereo Amp," Sept.'08), I had three questions:

The first is with the schematic. The version shown is the UL—that's clear from the circuit description. The wire colors for the UL windings are not indicated on the output transformers. I'm sure they are included with the transformer docs, but it would be nice to have them in front of me as I build this amp.

Second, I'm curious about why you wouldn't want to include a switch to

configure triode/UL mode for a listening comparison. Is the answer risk due to an open screen grid?

Third, I don't see any wiring challenges to wiring in a single chassis, but if there is a need to isolate, you could easily fashion a pair of dividers. It would make the unit much easier to transport and reduce the number of connectors.

Jim Albanowski
Princeton, N.J.

Joseph Norwood Still responds:

You can construct the amplifier on a single aluminum (17 × 10 × 3) chassis, which is available from Mouser (800-346-6873), PN 546-1444-32. Detailed wiring and color-coding instructions are provided with the transformer. Be sure to ground the black wire and the black and white striped wire to the chassis. To ensure no oscillations occur, four separate switchers would be required for UL and triode stability. This complicates construction and can also lead to burnout of the 6 550s. *aX*

CONTRIBUTORS

Paul J. Stamler ("Tri-Way Low Voltage Supply, Pt. 1," p. 8) is a recording engineer/producer, musician, and technical writer; he also hosts a radio program, "No Time to Tarry Here," featuring traditional folk music and related stuff. He has delighted in 78s since he was a boy, when they were still being made.

Atto Rinaldo ("Remote-Controlled Ten-Input Line Preamp," p. 18) is an active retiree after 32 years at IBM, where he experienced various job responsibilities in the US and other parts of the world. He received a radio techniques diploma 45 years ago. Vacuum tubes were the technology used at that time, and he has loved them since. Transistors were just born; he had never heard about integrated circuits. Joining IBM in the late '60s, he has seen vacuum tube computers phasing out just about that time. When transistors and ICs took over, he said to himself that one day he would play with tubes again. He did.

Karin Preeda ("A Chip-Driven SE Monoblock," p. 24) is the chief test engineer for Celestica Thailand. His interests include building tube audio equipment, and playing bass or lead guitar in a band.

Gary Galo (Review: Benchmark DAC1 USB, p. 26) is Audio Engineer at The Crane School of Music, SUNY Potsdam, where he also teaches courses in music literature. A contributor to AAC since 1982, he has authored over 230 articles and reviews on audio technology, music, and recordings. He has been the Sound Recording Reviews Editor of the *ARSC Journal* (Association for Recorded Sound Collections) since 1995, was co-chair of the ARSC Technical Committee from 1996 to 2004, and has given numerous presentations at ARSC conferences (www.arsc-audio.org). Mr. Galo is also a frequent book reviewer for *Notes: Quarterly Journal of the Music Library Association*, has written for the *Newsletter of the Wilhelm Furtwängler Society of America*, and is the author of the "Loudspeaker" entry in *The Encyclopedia of Recorded Sound in the United States,* 1st edition.

Chuck Hansen (Measurements for the Benchmark DAC1 USB, p. 32) is an electrical engineer and holds five patents in his field of engineering. He began building vacuum-tube audio equipment in college. He plays jazz guitar and enjoys modifying guitar amplifiers and effects to reduce noise and distortion, as well as building and restoring audio test equipment. He enjoys sailing and has over 150 magazine articles to his credit.

Jan Didden (Show Reports, p. 39) built his first OTL amp with 807 tubes 35 years ago. He has built speakers, preamps, and tape recorders, but is most interested in power amps, especially using error correction as discussed by Hawksford. Many of his projects have been published by *audioXpress*. Now retired from a career with the Netherlands Airforce and NATO, he tries to complete all those half-finished projects accumulated for lack of time. He now also has the time to travel to interesting audio events and interview audio luminaries. His projects are documented on his Linear Audio website.

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