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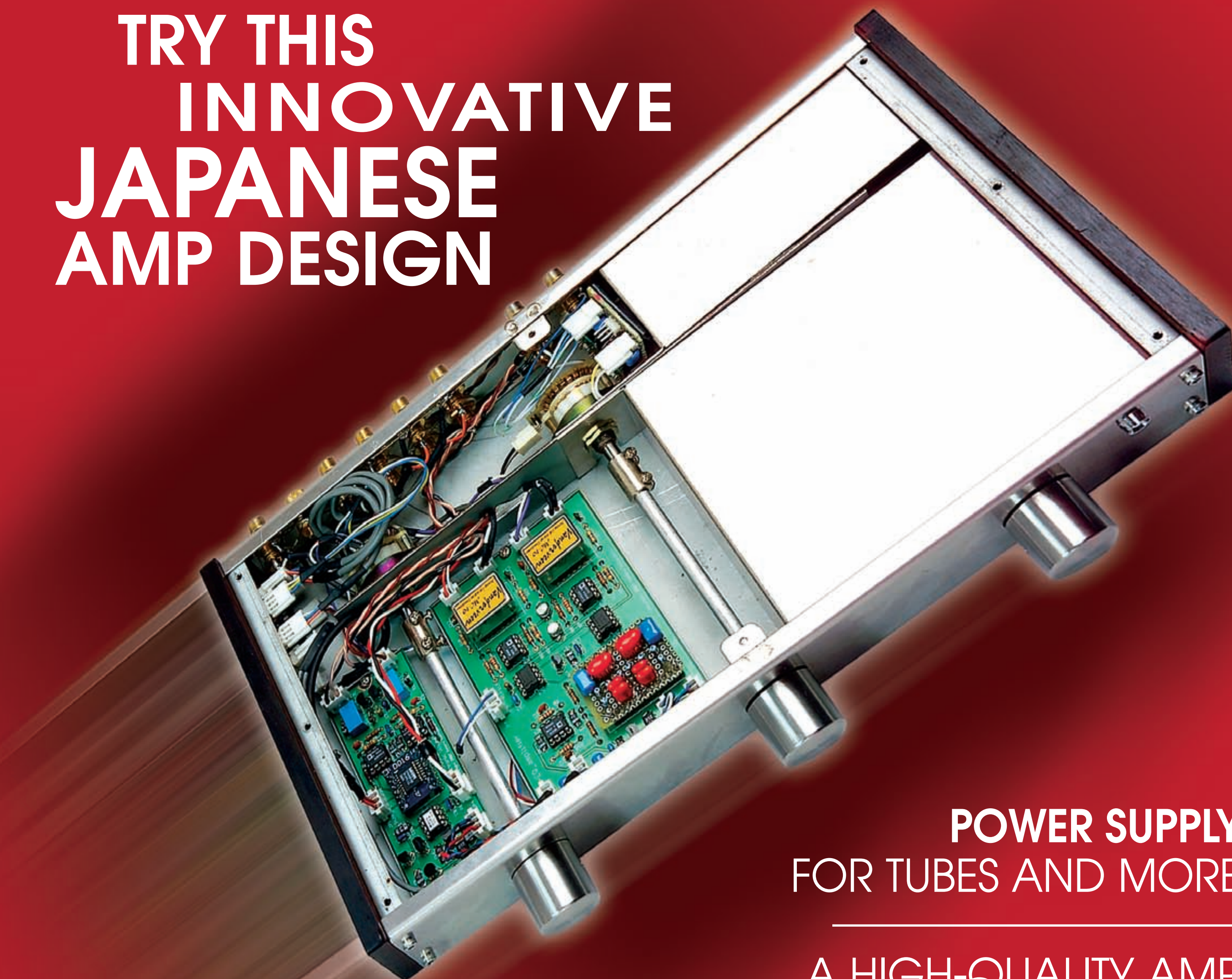

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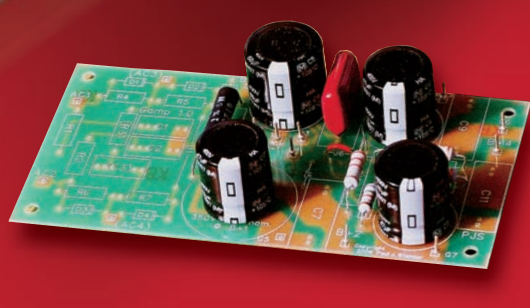

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# Music Inside the Brain

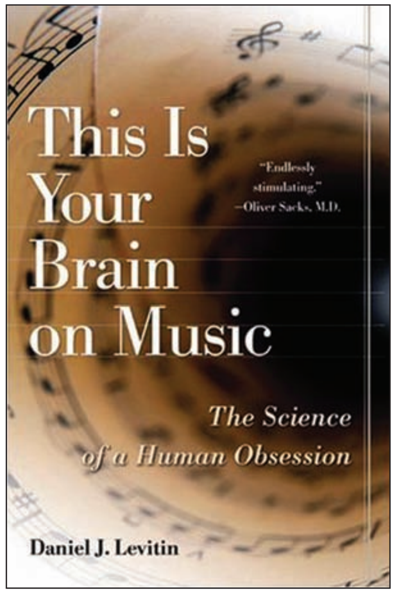
Oliver Sacks was my first introduction to the world of neuroscience in his book "The Man Who Mistook His Wife for a Hat." In it he relates the story of an elderly lady who was orphaned in Ireland when only five years of age, and subsequently emigrated to the United States. In her eighties she apparently suffered a small stroke in the middle of the night and awakened complaining that someone had left the radio on. She was hearing Irish tunes learned in her homeland, suddenly recovered clearly and completely as a result of this damage to her brain.

This raises the question of just how the human brain processes music and suggests that whatever music we have heard is retained within the brain. Ever since reading Sacks and his subsequent writing (plus seeing his remarkable movie), I have followed neuroscientific authors (see accompanying listing) in the hope that their research would shed light on how we process music, its history, and its relevance to our development.

There is excellent news for those of us who love music and are interested in how our brains process it. Daniel J. Levitin has written an outstanding introduction to understanding how we humans handle music listening based on the two disciplines of which he is obviously a master. (*This Is Your Brain on Music, The Science of a Human Obsession,* Daniel J. Levitin, Dutton, New York, 2006, $24.95. A paperback edition is promised for August 2007.)

The origin of the author's interest in music started with a $100 stereo system he bought with income from grass mowing when he was 11. His father, as a defense, bought headphones for the boy, which heightened his sonic experience. This led to his becoming a recording engineer and producer for ten years, working his way upward from no-name groups to some internationally famous ones.

Levitin obviously has the kind of curiosity and spirit of inquiry that leads him further into whatever problems he encounters. He wondered about why some musicians succeed and others do not. This led him to Stanford to ask questions about how




music is processed in the brain. He now occupies the Bell Chair of the Psychology of Electronic Communication and runs the Laboratory for Musical Perception at McGill University in Montreal, Canada. He has the ideal tools and experience to link what our brains do with music, and the world of music as performed and heard throughout humankind.

Levitin takes nothing for granted. He recognizes that the great majority of music listeners are not musicians. In order to understand how the brain processes music, he fully and simply explores music's basics. If you don't know the difference between pitch and timbre, you will after reading his chapter "What Is Music." This is a wonderful and comprehensive explanation of music's components. He dedicates a second chapter to loudness, rhythm, and harmony. The book is full of startling insights, including some theorizing about why music is so pervasive and what purpose it might serve in the evolution of homo sapiens. There is some speculation as to whether music might be the precursor of speech. Levitin takes a look at alternative views of others, especially Steven Pinker whose work on inherited grammars in newborns is widely accepted. When it comes to music, however, Pinker thinks it is nothing more than "cheesecake" as an adjunct to language.

An enormous amount of painstaking work has been done by a variety of scientists to discover how the brain works. Oliver Sacks has said that neuroscientists' work is like an auto mechanic attempting to understand the car by studying wrecked vehicles. Damage to brains has been the means of discovering what part of the brain does what. More recently, magnetic resonance imaging has revealed much more about sensory responses to all manner of stimulii in human and animal brains.

One of the primary discoveries of neuroscientists using fMRI tools is that music activates first one of the most primitive parts of the brain and makes a trace or link to the frontal lobe. It is likely that every song or melody that a person hears is stored in such a link. The brain's ability to recognize voices is another ability of the brain's amazing acuity.

Levitin takes on many more questions about music and brains. How do illusions affect our musical processing? How do we categorize music? How does music which arouses emotion affect the brain? The author goes on to a very insightful discussion of what is it that makes a musician. Is it innate skill, diligent practice, or what?

Then there is the basic question of why we humans like the music we like. I was pleased to discover that inside the womb we humans at a point of development begin to "listen" to music, both what mom is singing and whatever music is playing. Before our children were born my wife and I discussed at length what we would do about our sound system, which was playing music most of every day. We agreed that we would not curtail our music listening or the levels to "avoid disturbing the children."

Thus they grew up with music from their time in the womb until they left for college. Our three-year old son invariably raced to the speaker (mono) when WGBH played a short take from Handel's "Entrance of the Queen of Sheba" yelling, "My roosic, my roosic." Levitin notes that children respond more actively to music they were exposed to while in the womb than to other examples.

Music Instinct is a chapter dealing with why music is so pervasive and whether it has some relationship to our survival as a species. Any human characteristic as pronounced as our love for music is bound, say evolutionary experts, to fulfill some developmental purpose. There is still a sharp difference among experts on music's place in evolutionary development, however. Levitin agrees with Darwin's belief that music is vital to romance between humans, as well as between some other creatures such as birds and whales.

Levitin's book is a major achievement, an orderly introduction to the physiology undergirding our music processing apparatus. It may shed some light on such perennial controversies as meter readers versus subjectivists. It is a remarkable summary of what music is and its basic elements.

The author has an astonishing and enviable knowledge of music and musicians to illuminate his points of argument. He also has the rigor of the disciplined scientist, as well as a fine sense of humor and is a master of crisp, readable prose. The book is a rare gift to the music lover and we hope for many more from this pen.—E.T.D.

**Suggestions for further reading:**

Pinker, Steven, *How the Mind Works,* New York, Norton, 1997.

Wallin, Merker, and Brown, eds., *The Origins of Music,* MIT Press, 2001.

Damasio, Antonio, *The Feeling of What Happens,* Harcourt Brace, 1999.

By Satoru Kobayashi

# IC Preamp with Electronic Level and Selector

This Japanese preamp design features electronic analog switching. . . plus an MC step-up transformer.

It has been several years since I designed a simplified preamplifier using the DACT CT-100 and CT101 amplifier unit. I still would love to use DACT components such as a rotary switch and/or a volume to achieve a professional feel and smoothness when turning the knob. But the design I present here uses state-of-the-art semiconductor devices such as an electronic volume, electronic selector and so on, resulting in excellent presentation and an isolation characteristic as well as S/N ratio.

The sound is really pure and natural and brings a presence and dynamism. The amp is encapsulated into a compact chassis, including a RIAA equalizer and a couple of MC step-up transformers. The footprint of this unit equals that of a PC Notebook.

## DESIGN CRITERIA

1. Use an electronic volume. Although Burr-Brown's PGA2310 is well known, I chose a Wolfson electronics (UK-based) device, WM8816, in conjunction with the Tachyonix (Japan-based) 8-bit microcontroller chip (8616S06M) exclusively designed for WM8816.
2. Use an electronic selector, called an analog switch, available from sev-

eral IC manufacturers.

3. Make the chassis as compact as possible. Actually, this project is an upgrade of my project using DACT-made modules. I renovated the internal circuit, using my own circuit, with a RIAA equalizer.
4. Use the Van der Veen MC step-up transformer. Because the transformer is tiny (about the size of a couple of sugar cubes) it exhibits excellent performance.
5. Use DACT-made attenuator and a rotary switch, which provide a nice feeling when turning the knob.

## BASIC CIRCUIT DIAGRAM

The amp (**Fig. 1**) consists of three major blocks: RIAA equalizing amp, an electronic switch and electronic volume with a buffer amp, and a switching power supply with a ripple filter.

Because I had fallen in love with Van der Veen's toroidal transformers, I had been greatly anticipating his newly designed MC step-up transformer (MC-10, *please note this is not a toroidal!*) whose

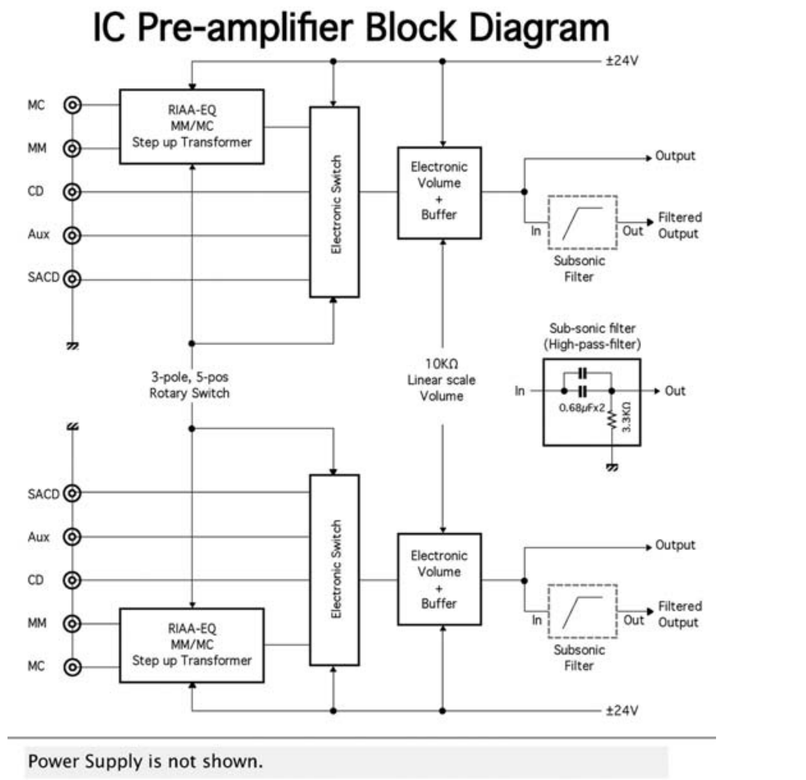



FIGURE 1: Circuit block diagram (power supply not shown).

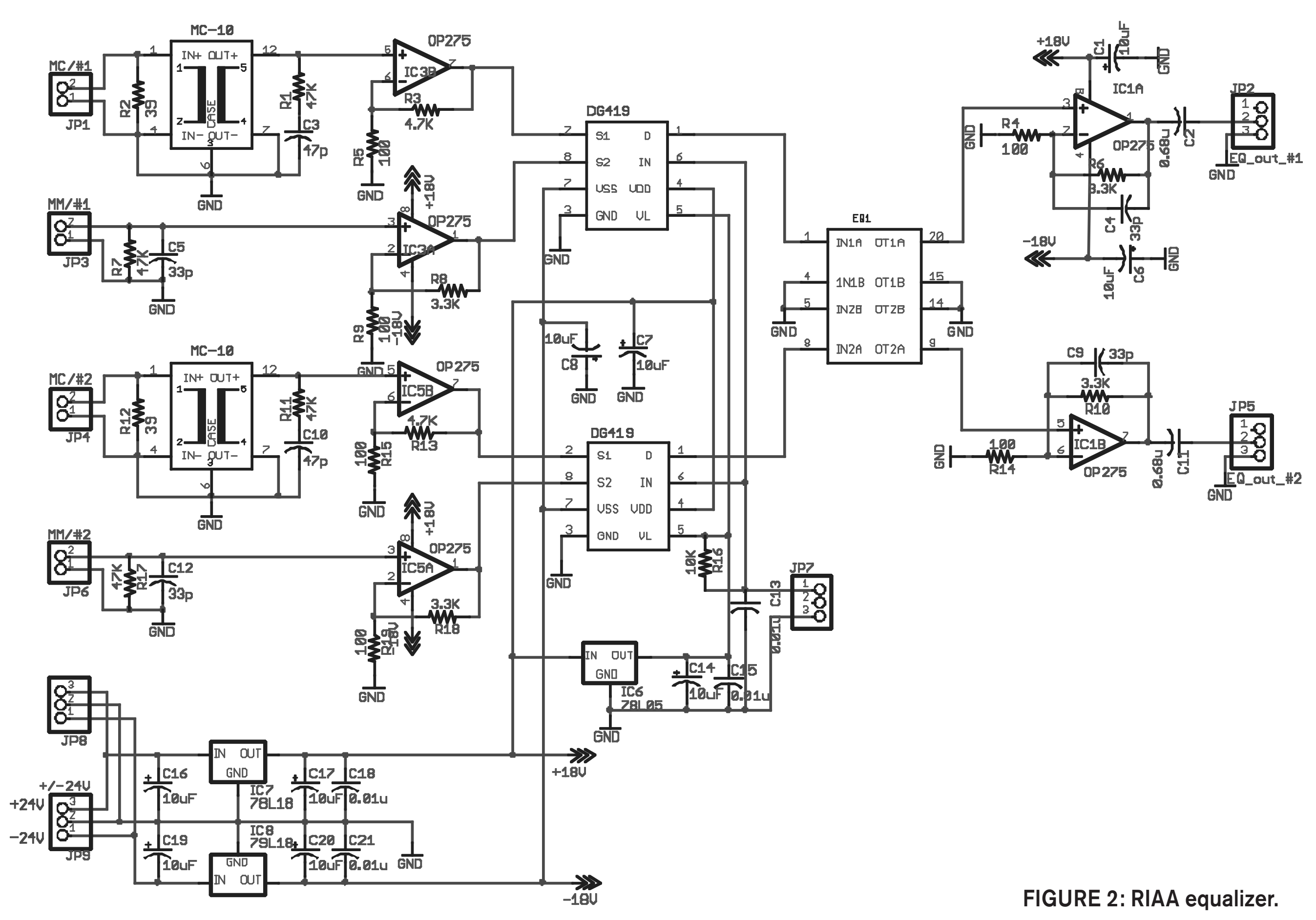



FIGURE 2: RIAA equalizer.

tiny, compact shape fit into a printed circuit board. Visit his website at www.mennovanderveen.nl/.

I also chose a CR-filter type equalizer rather than a NFB type equalizer. The key component is an IC chip, such as OP275 by Analog Devices, featuring an ultra-low-distortion characteristic. It might not need to use a NFB, lowering distortion as little as possible.

I calculated the CR filter components based on the traditional formula, prioritizing a capacitance value rather than a resistor value to minimize the deviation between the original RIAA curve and its actual value.

The circuit consists of a preamp and a post-amp by OP275. The former amplifies either MC or MM cartridge signal with 34X gain, the latter amplifies further with 48X gain, equalizing the CR-filter with a 20dB loss (1/10X), inserted between a couple of amplifiers.

The equalizing RIAA amp provides an electronic switch, an analog SPDT type that lets you choose either an MC or MM signal after pre-amplifying its signal with a 34X gain. The overall gain of RIAA equalizing buffer for MM and MC will be 116X(≈41dB, =34x1/10x34) and 1958X(≈66dB, =12x48x1/10x34).

Other signals such as CD, SACD, and so on will go to an electronic switch directly with a buffer amp and an electronic volume, providing approximately 2X gain.

After the selected signal the amp will finally go to a low-cut filter, with a 40Hz cutoff frequency, preventing a howling when you listen to a LP record with a LP player, working as a subsonic filter.

The final block, a ±24V DC power supply, drives these functional blocks securely, in conjunction with a ripple filter, which reserves a time-constant for a soft start.

## RIAA EQUALIZER

**Figures 2** and **2a** show the RIAA equalizer circuit and its RIAA CR-filter circuit. I mounted the RIAA filter over a small chunk of PCB so that I could easily fine-tune and replace components on the PCB, using a PCB pin and connector.

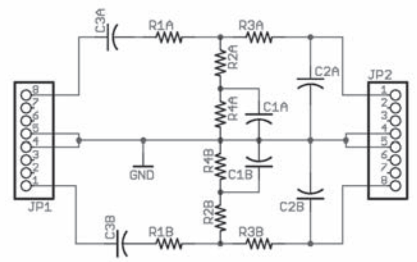



**FIGURE 2a: Equalizer filter.**

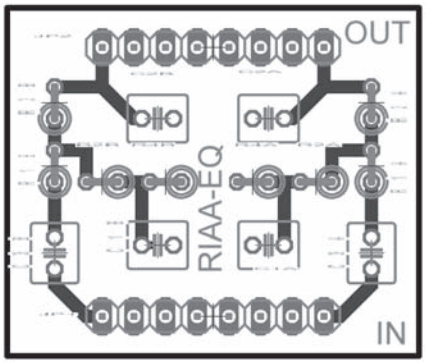



**FIGURE 2b: Equalizer filter board.**

**Fig. 2b** shows the completed etched PCB. I originally used a universal PCB for the experiment (**Photo 1**), but switched to the etched PCB.

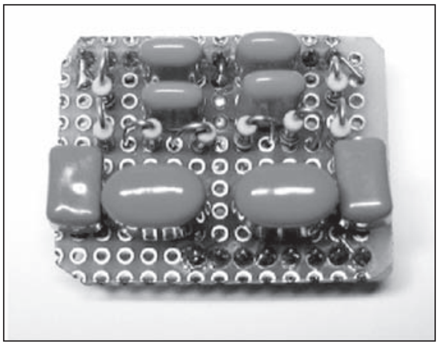


**PHOTO 1: Experimental board using a universal PCB.**

## ELECTRONIC SWITCH

Instead of a conventional mechanical switch, I have recently found an electronic switch that could improve crosstalk characteristics (**Figs. 3** and **3a**). Accuphase (Japan-based) produces excellent products, featuring nice isolation using a compact mechanical relay of either NEC or Takamizawa, for example. Those components need a lot of space and an extra electronic control circuit to choose a signal that I want. However, an electronic analog is made of an IC chip contained in a standard 300mil-wide DIP or a smaller IC package such as SOIC and so on, featuring a better isolation and a higher S/N ratio than a conventional switch. So it did not take long for me to decide to implement such a device into this new preamp.

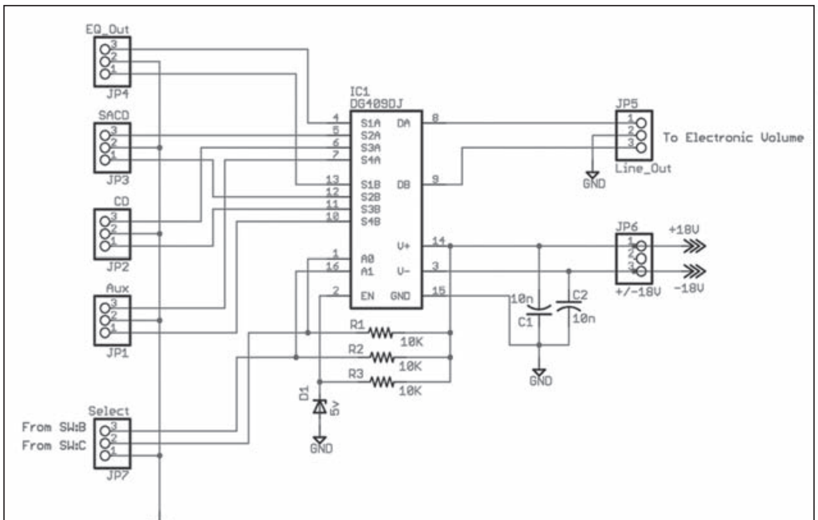



**FIGURE 3: Electronic switch.**

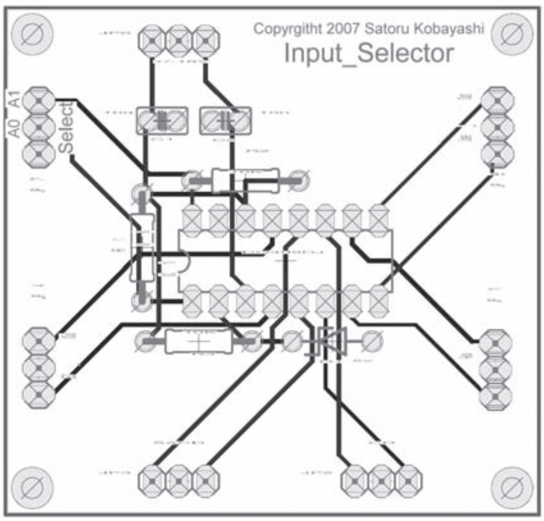



**FIGURE 3a: Electronic switch PCB.**

I first chose a SPDT switch selecting either an MC signal or MM signal after preamplified via a step-up transformer in case of the MC signal. I chose DG419 by Maxim in the US, whose compatible chips are available from several IC vendors such as TI, Vishay, Analog Devices, and so on.

Next I chose a dual 4-input, 1-output analog multiplexer, selecting one of four inputs (equalized LP signal, CD, SACD, or Aux signal) electronically. Through actual listening tests, I chose the Maxim DG409, which is compatible with the MAX4509, MAX309, MAX339, and so on.

Both electronic switches need a digital logic signal, a 0 or 1—i.e., below 0.8V or higher than 2.4V DC—to activate the switch or a multiplexer. So a conventional mechanical rotary switch works to apply such a logical input signal to these electronic switches. The 1 (high level) comes with a 10kΩ pullup resistor to a 5V power supply, and the 0 (low level) occurs by grounding a terminal. **Figure 3a** shows the PCB.

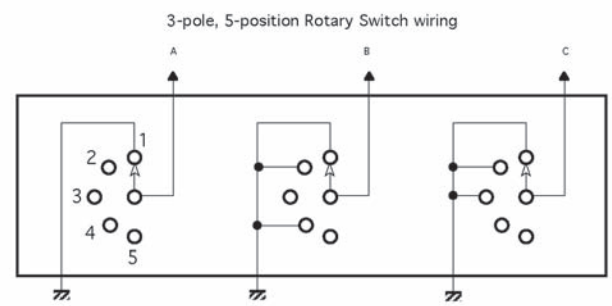
## MECHANICAL SWITCH

DACT CT-3 features a 4-pole, 5-position switch that produces a smooth feeling when you turn the knob, so I love this component as well. Out of the 4 poles, 3 poles need to generate a logical signal to those switches. The first one generates either a "0" or "1" logic level. The second and third poles generate a binary logic code such as "00," "01," "10," or "11" with a combination of wiring shown in the **Fig. 4** diagram.

## ELECTRONIC VOLUME

I learned that Marantz Japan introduced the high-end preamp SC-7S1 and SC-7S2, using Wolfson's electronic volume for the first time. This prompted me to use this device for my design because of its low-noise performance, high S/N ratio, and high crosstalk characteristic (**Photo 2**).

Fortunately, I found a dealership (www.tachyonix.co.jp/) locally in Japan to purchase this device with a customized microcontroller chip. I placed the order to get those, to taste a flavor of this device. The key component, WM8816 (www.wolfson.co.uk/), is available, and you can program its controller chip very easily using Microchip's 8-bit PIC technology




| Switch Pos. | Input Source |
|---|---|
| 1 | MC |
| 2 | MM |
| 3 | SACD |
| 4 | CD |
| 5 | Aux |

**FIGURE 4: Mechanical switch.**

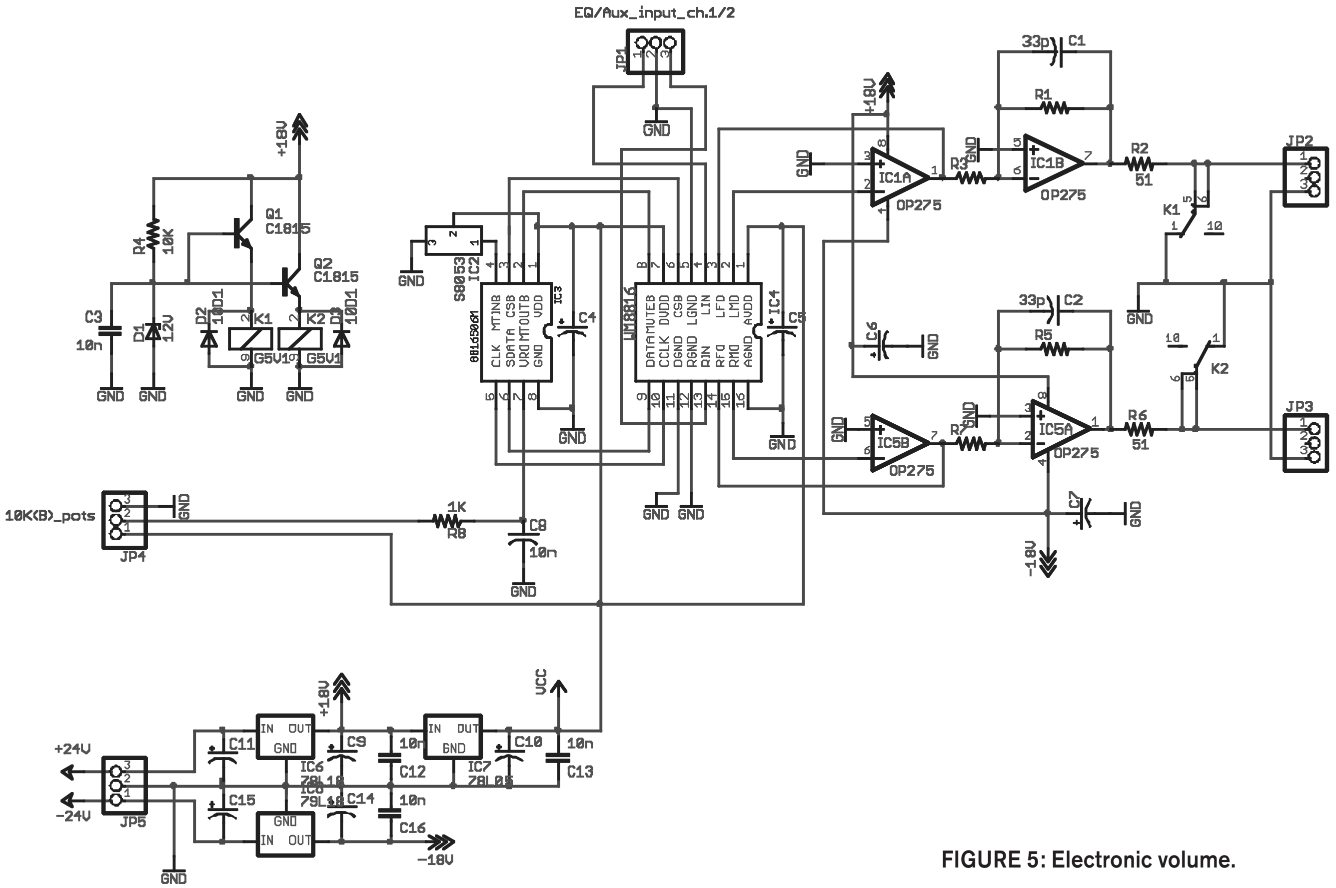


FIGURE 5: Electronic volume.

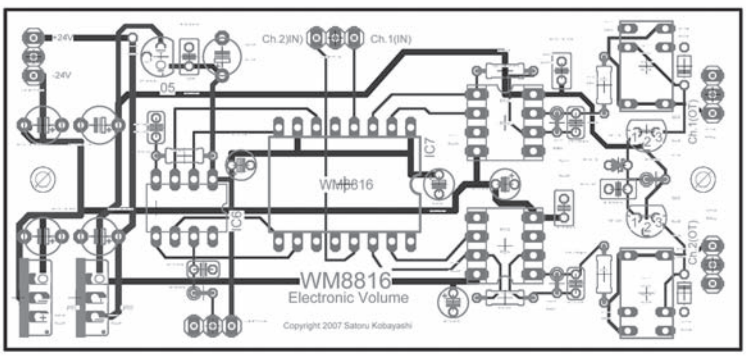


FIGURE 5a: Electronic Volume PCB.

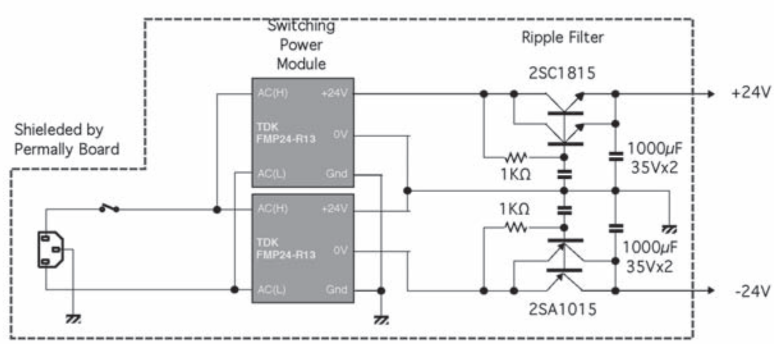


FIGURE 6: Power supply with a ripple filter.

(www.microchip.com/).

Prior to assembling the preamp, I performed an initial evaluation, using a standalone electronic volume, and putting the chip and the control chip as well as op amps onto a small universal PCB, enclosed in a neat case (**Figs. 5 and 5a**). The result from the listening test showed excellent presentation. This prompted me to upgrade my IC preamp instantly.

The buffer amplifier stage is also built in. Because the Wolfson WM8816 chip needs an extra inverting buffer by an op amp, generating the inverted output against the input signal, I included another inverting amplifier cascaded after the inverting amplifier.

## SUBSONIC FILTER

When listening to LP records with a RIAA equalizer, I needed to implement a subsonic CR passive filter to avoid howling at the low tone below 40Hz at the final stage of this electronic volume. I placed the subsonic filter circuit just next to the non-filtered output jacks, mounting the filter onto a small chunk of universal PCB.

## POWER SUPPLY

The original design used a ±24V switching power supply by TDK, but this time I added a ripple filter after the switching power supply in order to produce a slow-starting feature of the power supply with an approximate 0.6 second time delay. I built in a couple of permalloy boards to enhance a magnetic shield between the power supply section and the amp section and to improve S/N ratio (**Fig. 6**).

On the other hand, some audiophiles may wonder how switching power supply degrades the sound quality when such a power supply, installed inside the chassis, uses a 100kHz through 300kHz switching device over a PCB. I propose an alternative component using a PCB mount toroidal power transformer with a linear regulator chip of 78XX and 79XX series three terminal regulators (**Figs. 7** and **7a**).

## PARTS SELECTION

Japanese dishes, according to traditionalists, need a good chef, well-chosen raw materials such as fish, meat, vegetables, gradients and so on, and cooking must be as simple as possible. With this philosophy of simplicity in mind, I carefully chose the components to fit perfectly into this amplifier.

**• MC10 designed by Van der Veen**

The small size of the MC-10 (**Photo 3**) allowed me to integrate it into the chassis. This will fit in my Denon DL-103R cartridge.

**• Stainless-Steel Case**

The 320 × 200 × 55mm case (**Photo 4**) is shell-structured with the top cover and

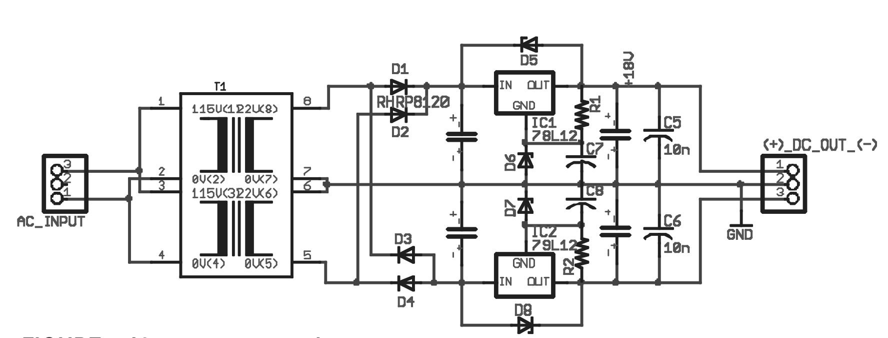



FIGURE 7: Linear power supply.

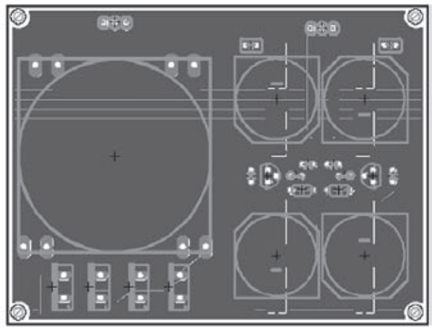


FIGURE 7a: Linear power supply PCB.

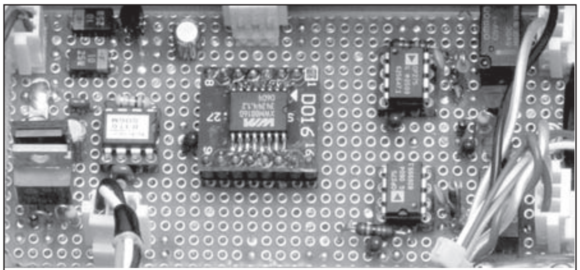


PHOTO 2: Switch components on PCB.

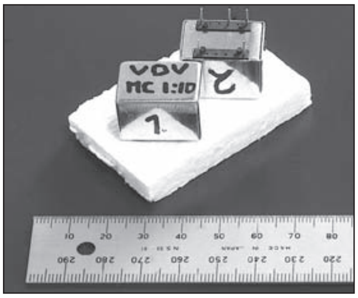


PHOTO 3: MC10 transformers.

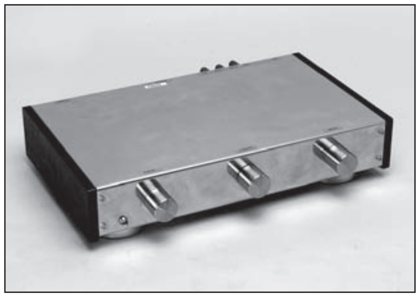


PHOTO 4: Stainless–steel case.

the internal case using a 1.5mm-thick stainless-steel plate. The major reason to use a top cover is to hide screws to fix the parts over the inner case. The top cover gives the amp a professional look and matches with stainless steel knobs by DACT on the front panel. The drawback is the added weight, which is over 2kg—heavier than a typical notebook PC.

The side walls are made of teak wood panels, 200W × 58H × 9mm thick. I polished the side panels with #400 through #600 sandpaper, drilled the inner sides, used three screws to attach onto the metal case, and polished again with walnut oil. As a result, the surface of the wood appeared smooth and professional-looking.

**• Linear Scale CT-2 DACT Mono Volume**

The Tachyonix control chip (8816S06M), which sends a program value of attenuation to WM8816 chip, needs a linear scale 10kΩ volume, instead of a log-scale volume, which has been used for the conventional analog audio equipment. Now I needed to modify the DACT-made CT-2 mono volume for this application. The DACT CT-2 volume uses chip resistors soldered over the PCB, where the chip resistors are placed peripherally to form a compact shape and its log scale curve. The value of those chip resistors is carefully calculated and selected to provide an accurate log scale curve.

But a linear scale volume for a digital volume chip does not need one at all. I pulled all chip resistors out of the PCB and put the new chip resistors of 470Ω evenly to make the attenuation curve a linear scale. Once modified, the new volume works with the control chip to generate a linear scale voltage to apply the control chip that converts an analog value to a digital code for WM8816 (**Photo 5**).

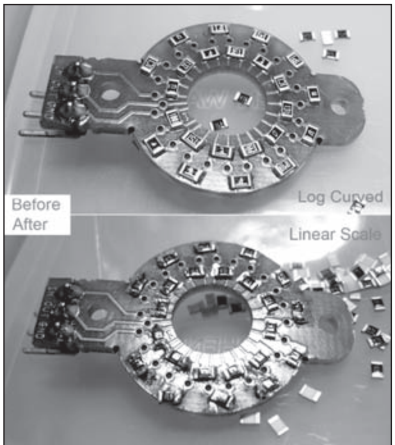



PHOTO 5: Linear scale CT-2 DACT mono volume.

## ASSEMBLY

Because this project is an upgrade of the previous one, there was a restriction in terms of assembly board size. I used a universal PCB that provides 100mil spacing through holes, allowing me to assemble components easily. The PCBs are 1) RIAA equalizing amplifier, 2) electronic volume with a buffer amplifier, 3) subsonic filter, 4) electronic switch, and 5) a ripple filter board, installed separately in the case. I implemented a bunch of PCB connectors so assembly and disassembly for the adjustment would be simple and easy.

After a while, I had a chance to try PCB CAD software, called EAGLE. So a few universal PCBs were updated with this software, to get a professional-made PCB.

## HOOKUP AND SIGNAL WIRES

Hookup wire was made by Sumitomo Electronic and featured a high melting point at 105° C. I also used Neglex brand signal wire by Mogami-wire in Japan (www.mogami-wire.co.jp/e/index.html) to link between the input and output terminals over the PCB and the terminals placed on the rear panel.

## ADJUSTMENT

While no adjustment is required, you should at least have a tester at hand to check wiring errors prior to power-up. Once you have found no errors, then disconnect the internal DC power connectors and pull all IC devices out of the amp modules. Then turn the AC power on.

Check the power module output pin level, which should be +24V or so. And check the voltage of all Vcc pins of the IC devices, such as ±18V or 5V, respectively.

After this, turn off the power and plug the connector cable back in, put all IC devices into the socket, and turn the power on again for standby.

## CHARACTERISTICS

*Electronic Volume Stage*

**• Input Versus Output Characteristics**

The electronic volume stage mounting Wolfson WM8816 chip and a couple of OP275 op amps showed a perfect linear curve; maximum output level showed approximately 12V RMS under ±18V regulated power supply (**Fig. 8**).

**• Distortion**

Distortion characteristics showed an extremely low value below 0.01%, even with 0.002% distortion of signal source, due to Analog Devices' OP275 chip (**Fig. 9**).

**• Frequency Response and Crosstalk**

Due to Wolfson's chip, the crosstalk showed an extremely low level below -100dB under EIAJ standard measurement conditions, bringing the silence in the listening room even at the maximum volume position of the preamp (**Figs. 10**

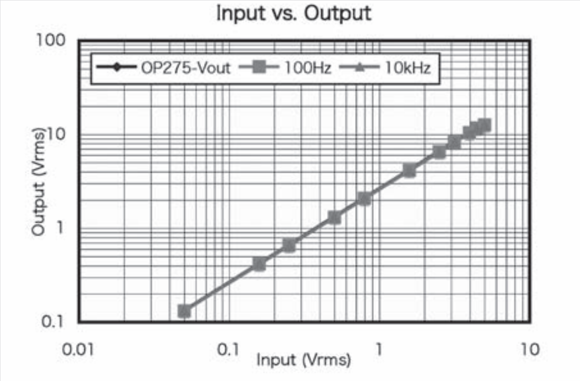



FIGURE 8: Input vs. output.

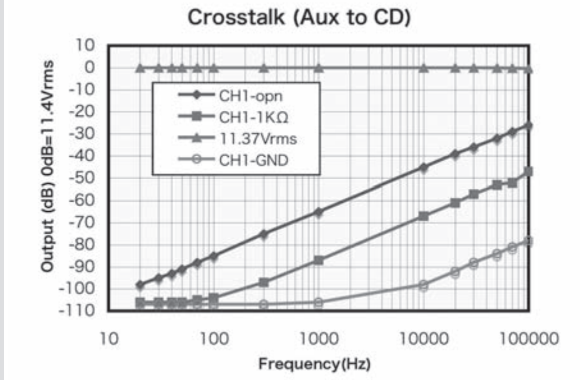



FIGURE 10: Frequency response & crosstalk (AUX->CD).

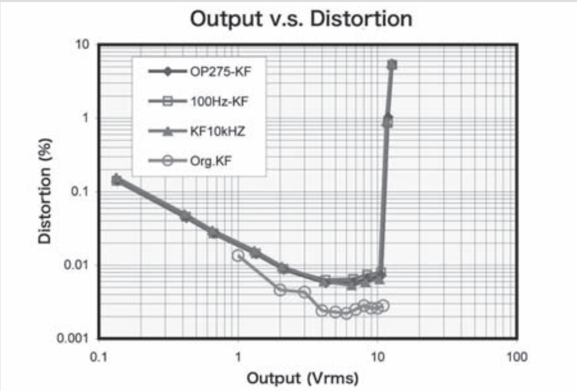



FIGURE 9: Output vs. distortion.

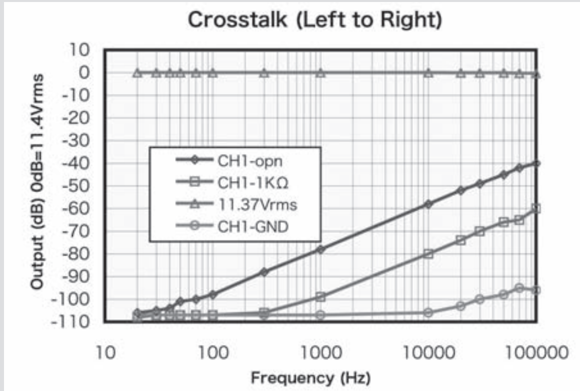



FIGURE 10a: Frequency response & crosstalk (Ch1->Ch2).

and **10a**).

**•Subsonic Filter**

At either MC or MM position when enjoying LP music, you might need a subsonic filter to avoid a howling caused by unwanted vibration via the room acoustic. Such a 40Hz low cut filter might work effectively, consisting of a basic passive CR filter (**Fig. 11**).

**•Electronic Volume Attenuation Curve**

The Tachyonix control chip works fine with the WM8816 chip, showing the linear stepped attenuation in log scale with an approximate 2.5dB per step. At the minimum position of the volume, the attenuation reaches to below -100dB, bringing the silence at the speaker system (**Fig. 12**).

***Equalizer Amplifier Stage***

**• Input Versus Output Characteristics**

With the RIAA equalizer and CR filter, the input versus output characteristics at 100Hz, 1kHz, and 10kHz showed a very clean linear curve in the range from the minimum value of the measurement system to the maximum output voltage at over 10V RMS (**Fig. 13**).

**• Distortion**

The minimum value shown is below 0.01%, which might be comparable to the manufacturer's preamp. At 100Hz, because of a hum, somehow the distortion value showed worse than at 1kHz and 10kHz (**Fig. 14**).

**• RIAA Curve Deviation**

In the range between 20Hz and 21kHz, I measured the RIAA equalizer devia-

### Parts List

| | |
|---|---|
| **RIAA Equalizer Amplifier** | |
| Components | Model, specification, manufacturer |
| MC step-up transformer | MC-10, Van der Veen (2) |
| op amplifier | OP275, Analog Devices (3) |
| Analog switch | DG419DJ, Maxim (2) |
| IC pin header | 2-pin, 3-pin Nihon Attyaku Tanshi (8) |
| IC socket | 8-pin (5) |
| resistor | 39Ω ¼W (2) |
| resistor | 100Ω ¼W (6) |
| resistor | 3.3kΩ ¼W (4) |
| resistor | 4.7kΩ ¼W (2) |
| resistor | 10kΩ ¼W |
| resistor | 47kΩ ¼W (4) |
| capacitor | 33pF ceramic (4) |
| capacitor | 47pF ceramic (2) |
| capacitor | 0.68μF film 5% (2) |
| capacitor | 0.01μF film (4) |
| capacitor | 10μF 25V tantalum (9) |
| linear regulator | 7818 or 78L18 |
| linear regulator | 7918 or 79L18 |
| capacitor | 1000μF 35V electrolytic (4) |
| Universal PCB | P-00185, Akizuki Denshi |
| PCB | custom made (OLIMEX) |
| **RIAA Equalizer filter** | |
| R1 | 33kΩ 1/6W~¼W (2) |
| R2 | 3.3kΩ 1/6W~¼W (2) |
| R3 | 130Ω 1/6W~¼W (2) |
| R4 | 100kΩ 1/6W~¼W (2) |
| C1 | 0.1μF film 5% (2) |
| C2 | 0.015μF × 2, film 5% (4) |
| C3 | 1.5μF film 5% (2) |
| IC pin header | 8-pin Nihon Attyaku Tanshi (2) |
| Universal PCB | P-00186 Akizuki Denshi |
| **Subsonic filter** | |
| resistor | 3.3kΩ ¼W (2) |
| capacitor | 0.68μF (4) |
| IC pin header | 3-pin Nihon Attyaku Tanshi (2) |
| Universal PCB | P-00184 Akizuki Denshi |
| **Input selector** | |
| Analog switch | DG409DJ, Maxim |
| IC socket | 16-pin |
| IC pin header | 3-pin Nihon Attyaku Tanshi (7) |
| resistor | 10kΩ ¼W (3) |
| capacitor | 0.01μF film (2) |
| Universal PCB | P-00186 Akizuki Denshi |
| Zener Diode | 5V 500mW, Hitachi |
| **Electronic Volume** | |
| Electronic Volume | WM8816S, Wolfson |
| controller | 8816S06M, Tachyonix |

### Parts List cont.

| | |
|---|---|
| op amplifier | OP275, Analog Devices (2) |
| Voltage decector | S-8053ALR Seiki Denshi |
| transistor | 2SC1815 Toshiba (2) |
| relay | G5V-1, 12V OMRON (2) |
| resistor | 51Ω ¼W (2) |
| resistor | 1kΩ ¼W |
| resistor | 2.47kΩ ¼W Dale (2) |
| resistor | 4.75kΩ ¼W Dale (2) |
| resistor | 10kΩ ¼W |
| capacitor | 33pF ceramic (2) |
| capacitor | 0.01μF (5) |
| capacitor | 10μF 35V electrolytic (9) |
| IC pin header | 3-pin Nihon Attyaku Tanshi (5) |
| zener diode | 12V 500mW, Hitachi |
| Silicon diode | 10D1 (2) |
| IC pin socket | 8-pin (3) |
| IC in-line pin | 8-pin (2) |
| **switching power supply** | |
| switching regulator | FMP24-R13, TDK (2) |
| transistor | 2SC1815 Toshiba (2) |
| transistor | 2SA1015, Toshiba (2) |
| resistor | 1kΩ ¼W (2) |
| capacitor | 1000μF 35V electrolytic (4) |
| **Linear power supply** | |
| toroidal transformer | 70015k, AMVECO International |
| diode | RHRP8120, Fairchild (4) |
| linear regulator | 7812 |
| linear regulator | 7912 |
| zener diode | 6.2VV 500mW, Hitachi (2) |
| capacitor | 1000μF, 50V (4) |
| capacitor | 0.1μF 50V (2) |
| resistor | 10kΩ ¼W (2) |
| IC pin header | 3-pin Nihon Attyaku Tanshi (2) |
| Universal PCB | P-00184 Akizuki Denshi |
| **Case and Peripherals** | |
| Stainless-steel case | custom designed 320 × 200 × 55mm |
| permalloy board | Ohtama Denki (2) |
| rotary switch | CT-3, DACT |
| volume | CT-1-10kΩ, DACT, modified to linear scale |
| sidewall board | Teak wood, 200 × 55 × 9mm (2) |
| metal feet | 50mm, 20mm height IAG made (4) |
| stainless-steel knob | 25mm, DACT (3) |
| power switch | ARN01, 25A 125V AC, OTAX |
| extension rod | 6mm, aluminum (2) |
| extension | 6mm, no brand (2) |
| RCA jack | white, red (14) |
| ground pin | gold plated |
| IEC inlet | 25V, fuse built-in |
| AC outlet | 250V, 10A |

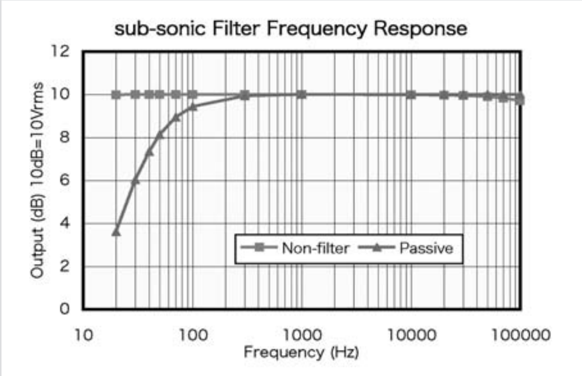



**FIGURE 11: Subsonic filter frequency response.**

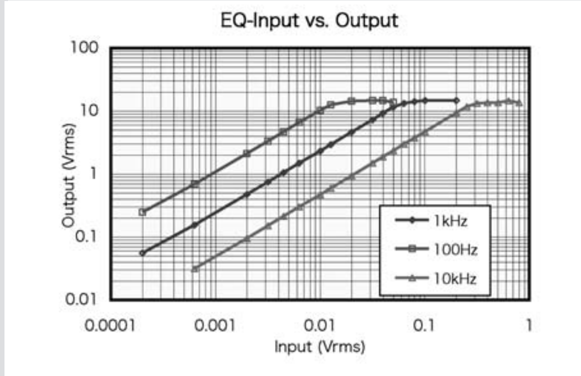



**FIGURE 13: RIAA equalizer input vs. output.**

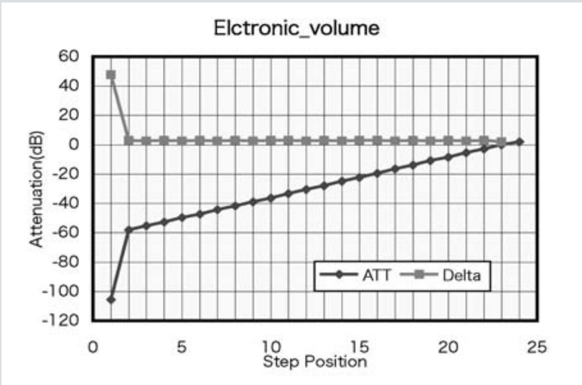



**FIGURE 12: Electronic volume attenuation curve.**

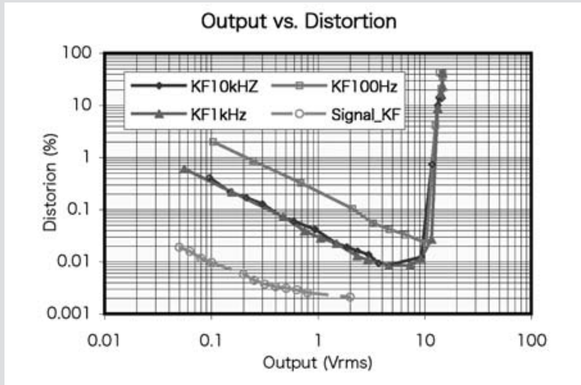



**FIGURE 14: RIAA equalizer output vs. distortion.**

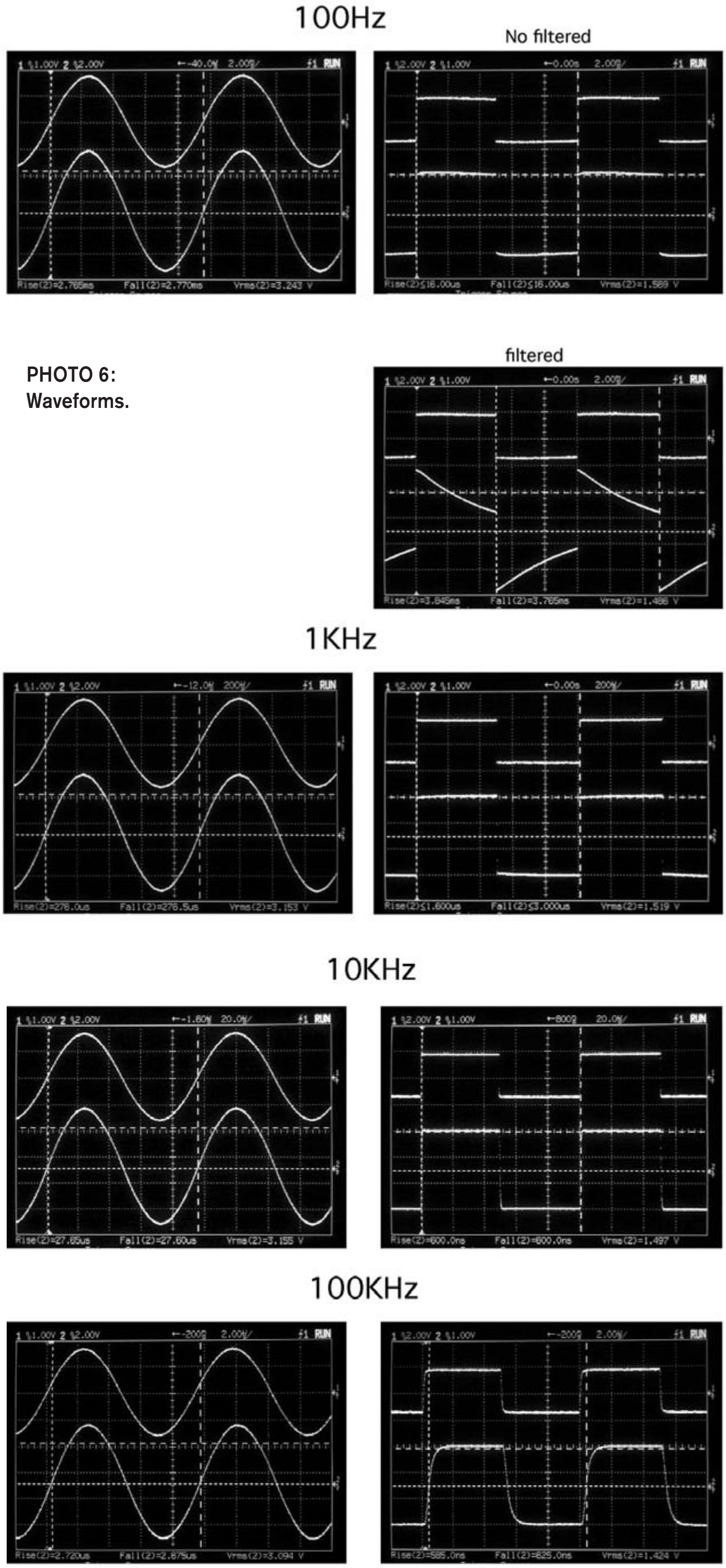



**PHOTO 6:** Waveforms.

tion by calculating the balance between the measured value and the theoretical value and plotted the values onto **Fig. 15**. The deviation calculated is extremely small below 0.1dB in the entire frequency range of RIAA equalizer (**Fig. 15**).

**• Waveforms**

Waveforms (**Photo 6**) at 100Hz, 1kHz, and 10kHz sine and square wave show a typical clean shape due to the lowest distortion environment. I observed neither visible distortion nor phase errors.

**• Residual Noise Voltage**

The measured residual output voltage level was extremely low as though it were a measurement instrument. Due to such an extreme low level of residual noise level, my B&W 802 speaker system does not generate any hum—even in CD mode at the front edge of speaker cone with the maximum volume level of both power amplifier (Luxman M-7f) and this preamp. The only sound I could hear was a noise surrounding me coming from outside my house, which is typical Japanese style.

Even in the MM input mode, the speaker produces just a clean noise other than a hum (100Hz or so). During this listening session, I could detect only the silence.

In the MC position, some improvement is needed, because of the magnetic shield structure; it produced some S/N ratio degradation, which corresponds to 10dB or so. The measured noise level was approximately 3× worse than at MM input. However, at the regular listening volume position on my system, which is about the mid-position of the volume (corresponding to 12 o'clock position), the noise was negligibly small (**Table 1**).

## LISTENING IMPRESSIONS

What was most noticeable when listening to this amp was the perfect silence even at maximum volume before starting the CD or LP players. Furthermore, no crosstalk was observed at all.

## RIAA EQUALIZER

This seemed very natural to me. I have tested several LPs, classical music, jazz, R&B, Japanese POPs, Japanese Enka,

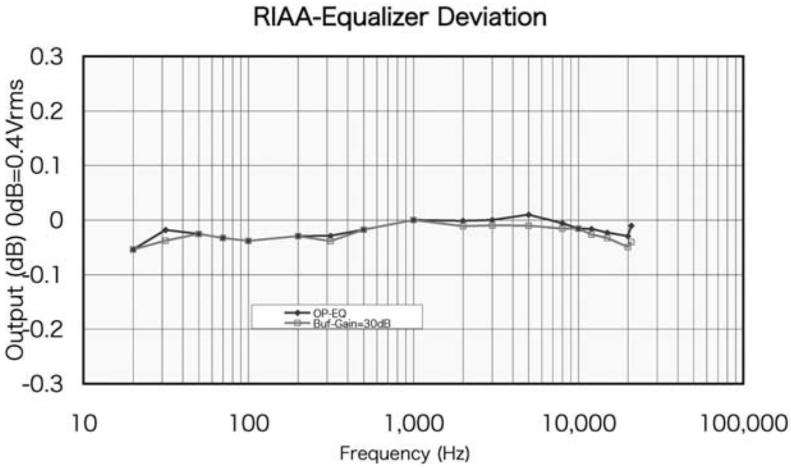



**FIGURE 15: RIAA equalizer deviation.**

**TABLE 1**

| A-weighed filter | AWT-30 | Nihon-Audio | | |
|---|---|---|---|---|
| Right Channel | | | | |
| Residual Noise | vol. min. | vol. mid. | vol.max. | Ω |
| Grounded | 25 | 33 | 136 | µV |
| A Filtered | 5 | 7 | 19 | µV |
| 1kΩ-Grounded | 42 | 43 | 135 | µV |
| A Filtered | 5 | 6 | 19 | µV |
| Residual Noise | vol. min. | vol. mid. | vol.max. | Ω |
| MM-Grounded | 43 | 55 | 400 | µV |
| A Filtered | 5 | 8 | 100 | µV |
| MM-1kΩ-Grounded | 44 | 55 | 410 | µV |
| A Filtered | 5 | 8 | 108 | µV |
| MC Grounded | 43 | 85 | 2050 | µV |
| A Filtered | 6 | 16 | 320 | µV |
| MC-1kΩ-Grounded | 43 | 77 | 1750 | µV |
| A Filtered | 5 | 11 | 285 | µV |

and so on. The sound of those was really natural and exhibited no stress and strong enhancement at all. As compared to a few CDs, which are identical to LP records, both showed the same presence as well.

## CD PRESENTATION

It surprised me that Wolfson's electronic volume chip produced an excellent presentation of CD music. I found this electronic volume chip with an external op amp produced a new era of the sound as compared to the valve line amp. Also, I could customize the sound by altering the external op amp with other favorites, such as the OPA627.

### Listening Test with the Valve Line Amp

I recently had a chance to test this with the valve line amplifier using a 12AX7 valve with 18dB NFB between its plate and a grid terminal. This valve was sorted out of over 20 tubes designed and built by a friend of mine.

The result was no significant sound difference; it was difficult to describe the difference. Somehow, because he loves the valve amplifier, he believed the sound of my amplifier gave a sort of solidness as compared to his 12AX7 line amp. aX

**Measurement Equipment**

1) Audio Analyzer HP-8903B
2) Audio Generator Kenwood AG-204D
3) Audio Generator HP-3311
4) Attenuator HP-3467A
5) Digital Multimeter Fluke 177
6) Oscilloscope HP-54600B

**CONTRIBUTORS**

**Edward T. Dell, Jr.**, ("Music Inside the Brain," p. 6) is owner and publisher of Audio Amateur, Inc., publishers of *audioXpress, Voice Coil,* and *Multi Media Manufacturer* magazines.

**Satoru Kobayashi** ("IC Preamp with Electronic Level and Selector," p. 8), from Tokyo, Japan, has been interested in audio and ham radio since he was in his teens. After majoring in EE in Tokyo, he joined the semiconductor industry, designing DRAM chips for a living, although he now works in the technical and marketing area of semiconductor packaging industry. He periodically writes on the subject of audio for a few different magazines.

**Paul J. Stamler** ("Adaptable Power for Tube Amps," p. 20) 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.

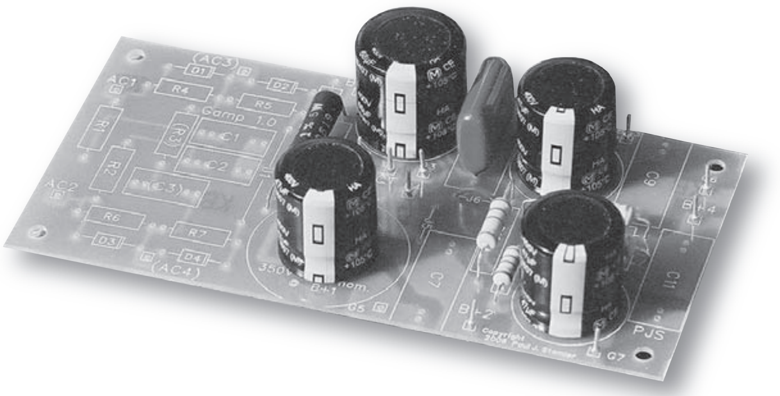
**Jack Walton** ("High End 120W MOSFET IC Driven Amp," p. 30) is an investment consultant and entrepreneur in Short Hills, NJ. His educational background is in chemistry, physics, and finance. He has had a lifelong interest in electronics, ham radio, photography, and the arts.

**Matthias Brennwald** ("MATAA: A Free Computer-Based Audio Analysis System," p. 36) lives in Switzerland.

**David J. Weinberg** (Review, HIFICRITIC, p. 48) is an engineering consultant and technical journalist on audio, video, and film technology. He provides audio and home theater engineering consultation and professional on-location digital audio recording services to companies, radio stations, and individuals. He brings to his work an MSEE, a First Class Radiotelephone license, and over 40 years of continued study and active involvement in the audio, video, and computer industries. He is Vice Chairman of the Audio Engineering Society's DC section, a manager in the Society of Motion Picture and Television Engineers' DC section, and membership officer of the Boston Audio Society (www.BostonAudioSociety.org).

# The Gamp: Adaptable Power for Tube Amps

You can configure this versatile power supply for use in tubed and a variety of other projects.




This project began with an amplifier. I was working up a small, tubed, Champ-class guitar amplifier, and the first step was inevitably to build its power supply, so I could begin experimenting with the audio circuits. Like all guitar amps in my experience, it would have several stages of filtering to power the various sections of the amplifier. It would not, however, use active regulation.

As I was working out the design, I realized that this type of power supply is applicable to a variety of projects. Many people are building tube doohickeys of one sort or another—microphone preamps, compressors, guitar amps, clones of classic tubed gear—and each one needs a power supply. Why reinvent the wheel every time, and design a board for it to boot?

I began designing the board using a new open-source layout program, Free PCB (www.freepcb.com); along the way I decided to make the board layout as flexible as possible, so it could be used for a broad range of projects. I wound up with a design and board that can be used for any application requiring a well-filtered power supply with a nominal voltage level up to 350V. Along the way, I found ways to make it sit up and do tricks, up to and including regulation.

## WANT LIST

As always, step one in the design was figuring out what I wanted it to do. First, I wanted the supply to be usable with either tubed rectifiers or solid-state. With solid-state diodes, I wanted the option of using RC snubbers as described by Jim Hagerman[1], as well as the anti-resonant series resistance suggested by Ben Duncan[2]. I wanted the diodes, if solid-state, to be usable either as a center-tapped full-wave rectifier or a full-wave bridge. I wanted users to have the option of LC filtering, and, for safety reasons, I wanted a bleed resistor to bring the voltage down quickly when the power gets turned off.

I also wanted to offer jumper-selected options for the filter circuits: a standard ladder, in which each stage's filter taps off the previous one, or a variant which allows two filters to fan out independently. Finally, I wanted users to have the option, in each stage after the first, of using a standard star ground or a different grounding scheme (again jumper-selected).

There was some pleasant serendipity. By using two series resistors for the bleeder, I could elevate a filament supply above ground, either for hum reduction (an AC filament is less likely to radiate into the adjoining cathode when it's biased above the cathode's DC level) or to avoid exceeding heater-to-cathode limits when a tube is being used as a cathode follower, SRPP, or similar circuit.

I found I could use the spaces for the two series bleed resistors for zener diodes instead, producing a simple regulator; by adding an external pass transistor (and heatsink) I could even implement a full emitter- or source-follower regulator.

I called my design the Gamp supply, named for the little guitar amp that was its first purpose. It's a general-purpose supply, but it's important to observe that 350V limit on the nominal voltage. When designing supplies, I assume that AC line voltage will vary ±10%. If the nominal voltage is 350V, then the voltage when the line is 10% over normal will be 385V. There are standard safety guidelines for the maximum voltage permitted between traces on a PC board for a given spacing; a handy online calculator is at www.desmith.com/NMdS/Electronics/TraceWidth.html.

When designing this board, I used a minimum gap of 0.1″ (100 mils) in most locations, which corresponds to just over 385V. If you try to run the boards hotter, you risk having the voltage arc over between traces, damaging the board and blowing either a fuse or your transformer. (There are a couple of conductors on the board that have closer spacing, but they are all ground traces and should have no voltage differential between them.) For those needing a higher voltage supply (to feed, say, a Dynaco-class power amp or a bigger guitar amp), that's on my to-do list.

## BASIC DESIGN

The power supply design is shown in **Fig. 1**. Note that most of the part designations are not filled in; parts selection is open-ended as long as you use components that fit the spacing of the board, which is set up for bog-standard sizes. The electrolytic capacitors C4, C8, and C10 can be as large as 270µF/450V (1˝ diameter, Panasonic TSHB series or equivalent). I left room for C6 to be larger (1.4˝); it can be up to 470µF/450V if you like. The whole TS series of caps has the same pin spacing, so you can also use smaller ones: 33µF/400V, for example, in the TSHA series.

The bypass capacitors (C5, C7, C9, and C11) are presumed to be polypropylenes, again in standard sizes; I included mounting holes for 0.1µF caps from Panasonic's ECQ-P series at either 400V or 630V.

For input and dropping resistors R8-11, I made room on the board for 5W sand-cast power resistors; it's also possible, of course, to use lower-wattage resistors if the design warrants. The bleeder string (R12-13) is designed to pull 1mA of current; note that R13 is specified as 75k, which will provide 75V at the junction of R12-R13 for filament elevation.

## INPUT SECTION

I'll talk first about hooking up the board with solid-state rectifiers. The most common connection for tube circuits is a full-wave center-tapped design (**Fig. 2a**). The transformer's plate leads (red) are connected to the AC1 and AC2 terminals on the board, while the transformer plate winding's center-tap (red-yellow) is connected, logically enough, to the terminal marked CT.

The diodes specified have ultra-fast turnoff times to minimize generation of radio-frequency crud that can leak into audio circuits. To ensure that they never see a reverse voltage higher than they can stand, each leg has two diodes in series, with parallel resistors (R4-R7) that ensure voltages across the diodes are equal. (Old trick from the *Radio*

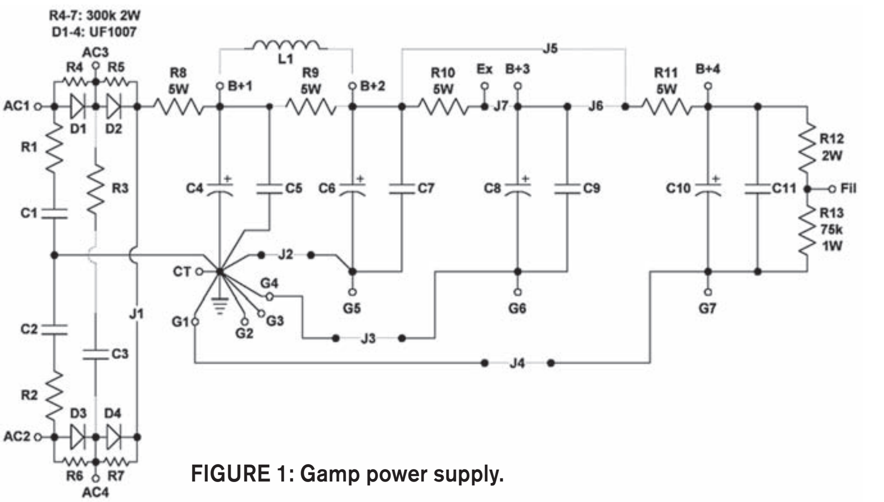


**FIGURE 1: Gamp power supply.**

*Amateur's Handbook*.)

For further protection against RF interference, I've included room on the board for Hagerman-style RC snubbers. At Jim's suggestion, I treat each half of the transformer secondary as a separate winding, so there are two snubbers, R1-C1 and R2-C2. (For information on how to calculate resistor and capacitor values in the snubbers, please see his article.) R8 is the input damping resistor á la Ben Duncan. In this full-wave center-tapped configuration, R3 and C3 aren't used.

What if you have a transformer without a center tap? In that case, you'd use a full-wave bridge rectifier (**Fig. 2b**). The transformer's plate leads now go to terminals AC3 and AC4, and the single snubber consists of R3-C3. R4-R7 are omitted, along with R1-R2 and C1-C2. Terminals AC1 and AC2 are connected to each other, and to terminal CT. I'd originally envisioned a heavy wire from AC1 to AC2, and another wire connecting the centerpoint of that wire to CT, but I realized you can make the connection by inserting jumpers in place of R1, R2, C1, and C2. (Nice, heavy wires, if you please.)

Tubed rectifiers are almost always used in full-wave center-tapped circuits; **Fig. 3a** shows the hookup for an indirectly heated tube (e.g., 6X4), while **Fig. 3b** shows the hookup for a directly heated tube (e.g., 5AR4). The tube's cathode connects right to terminal B+1; all of the components associated with the solid-state diodes, including R8, are omitted. Note that tubed rectifiers have limits on the size of the first filter capacitor; these restrictions aren't completely graven in stone (I'll discuss them when I do a worked example later), but you should keep them in mind when designing a tube-rectified supply.

## FILTERING: VARIATIONS ON AN RC THEME

First, the filter doesn't need to be RC. It's easy to substitute off-board chokes for any of the dropping resistors in the filter sections—I've shown one in **Fig. 1** between B+1 and B+2, but you could use chokes on B+3 and B+4 if you desire. In fact, you could leave out C4 and C5 entirely, and make the Gamp a choke-input supply. Nobody would complain.

I mentioned that the board has room for a really big capacitor at C6, to run an output stage perhaps, or to stabilize a screen voltage. Why there, in the B+2 stage, rather than the input?

As noted in the previous section, tubed rectifiers don't like to see large capacitors in the first filtering stage; it can shorten their lifetimes. Solid-state diodes are more forgiving, but they still undergo a lot of stress from inrush current at the moment of turn-on. The first filter capacitor acts like a dead short until it's charged, and because a bigger cap takes longer to charge, the stress on the diodes is prolonged. C6, on the other hand, is on the downstream side of R9 (or L1), which acts as a current limiter, minimizing stress on the diodes. (The turn-on stress is instead on R9, so it's worth using a 5W unit here even if its normal operating dissipation will be much lower.)

Next down the line are two mysterious-looking jumpers, J5 and J6. What are they about?

Look at **Fig. 4a**. That's the usual design of an RC-filtered power supply; each stage feeds the next. On this board, you'd choose that option by including J6 and omitting J5. But what if you'd prefer something like **Fig. 4b**? Perhaps you need two separately decoupled filters with the same DC voltage; you'd select that option by connecting J5 and omitting J6.

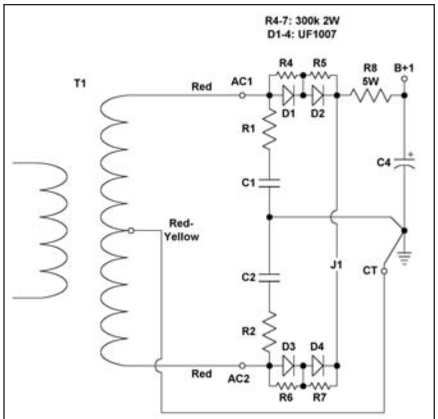



**FIGURE 2A: Full-wave rectifier, center-tapped transformer.**

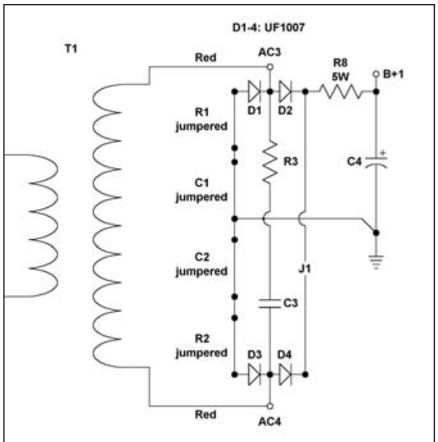



**FIGURE 2B: Full-wave bridge rectifier.**

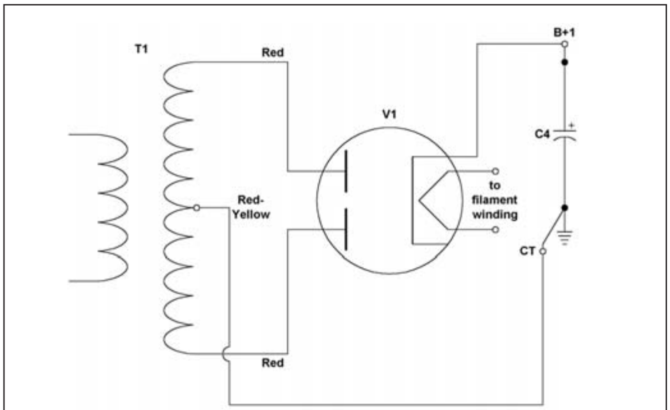



**FIGURE 3A: Tubed rectifier, indirectly heated.**

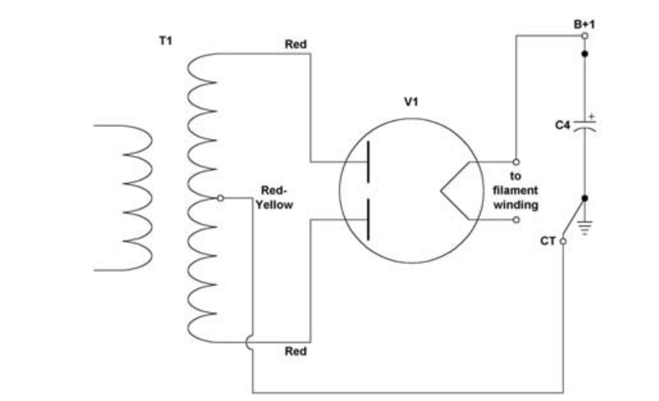



**FIGURE 3B: Tubed rectifier, directly heated.**

I designed the bleeder string, R12-R13, to draw 1mA from the supply at full voltage. R13 is specified at 75k; it dissipates 75mW, so its 1W rating provides a good safety margin. The value of R12 depends on how many volts it drops; I'll look at that when I work out an example.

## SEEING STARS

Most audio projects use a standard "star" ground; every power supply section, and every audio section, is grounded directly to the main grounding point, often (as here) the negative terminal of the first filter capacitor. The chassis is also connected to this point (this board has a terminal for that, pin ChG). To implement a classic star ground on the Gamp supply, use J2-4 and connect the audio stage grounds to the terminals G1-4.

Not everyone prefers to use this grounding scheme, though. It's sometimes quieter to use a different, non-star

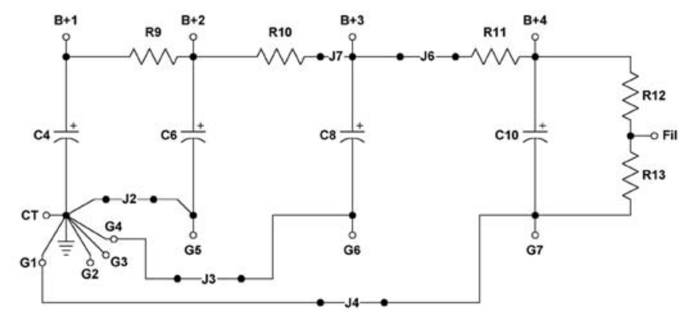



FIGURE 4A: Cascaded filter, bypass caps omitted for clarity.

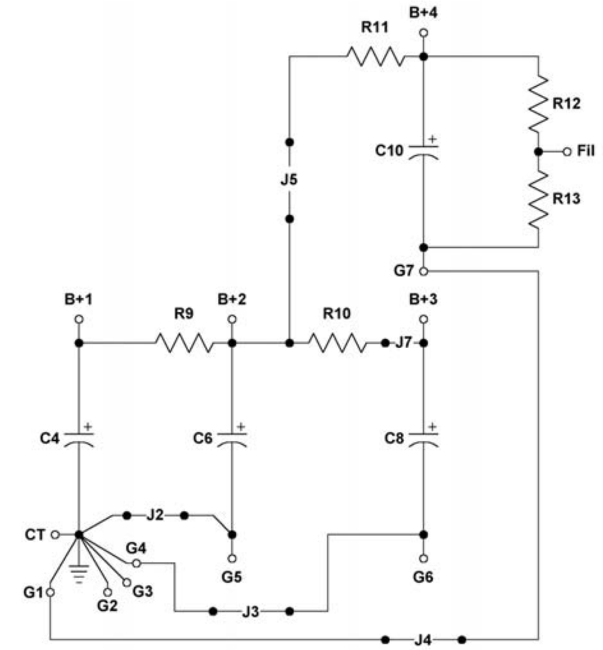



FIGURE 4B: Fanout type filter, bypass caps omitted for clarity.

hookup (**Fig. 5**). In this arrangement, the ground on the filter capacitors connects directly to the audio stage's ground point, which then sends a ground wire back to the main grounding point. It's easy to do this on the Gamp simply by omitting that stage's grounding jumper (J2, J3, or J4) and connecting the capacitor ground terminal (G5, G6, or G7) to the audio circuit's grounding point, wiring the latter to G2, G3, or G4. You can, of course, mix and match the ground arrangement on the various stages; I've found it useful on a guitar amp, for example, to use a star ground on B+2 and B+3, but the alternate hookup on B+4 (which feeds the audio input stage).

## GETTING FANCY

It's easy to connect the bleeder string to elevate a filament supply above ground. With an AC filament supply, simply hook the terminal marked Fil to the center tap of the transformer winding or, if it's not center-tapped, to the junction of a pair of matched 10k resistors hooked across the winding. (There isn't room for these on the board; use a small terminal strip.) A DC filament supply is equally easy; hook the 0V terminal of the supply to the Fil terminal instead of ground.

Hoisting above ground cuts hum radiation from an AC filament supply, but it also can keep tubes operating properly. Every tube has a maximum heater-cathode rating, and for long life and good performance it's a bad idea to exceed that rating. A cathode-follower circuit, or any of the totem-pole circuits popular in recent years, has a cathode that sits well above ground, sometimes as high as 150V. That's above the heater-cathode rating of most tubes, and I've seen 12AU7s start to arc under those conditions.

A 75V bias, used with a 150V cathode follower, brings the heater cathode voltage down to 75V, well below the 90V rating common to many tubes. It's low enough, however, that it won't cause problems for a voltage amplifier in the same circuit, which might have a cathode sitting at 4V.

It's possible to include a simple regulator. The holes for R12-13 are sized so you could place a pair of 5W zener diodes there; R11 now becomes the current-limiting resistor for the zener string. You can use this variation whether the B+4 section is wired for ladder or fanout (J6 or J5). Depending on the zeners' voltage, you could still use the Fil terminal to elevate a filament supply.

## FULL REGULATION

Once I designed the zener-regulated variant on the Gamp, I realized that only one more jumper was needed to implement a fully-regulated emitter- or source-follower supply. Look at **Fig. 6**; in this circuit J7 is omitted, R10 acts as a current-limiter and/or dropping resistor for the pass device, and R11 plus the two zeners supply the base or gate with a reference DC voltage. The capacitors at B+3 and B+4 now act as bypass caps for the regulated output and the reference voltage, respectively.

It's important to calculate dissipation on the pass device carefully (including what happens when the line voltage is 10% high) and use adequate heatsinking. The stopper resistor (R14) is also important, because it keeps the pass device from breaking into oscillation; you should connect the stopper (with short leads) directly to the base or gate termi-

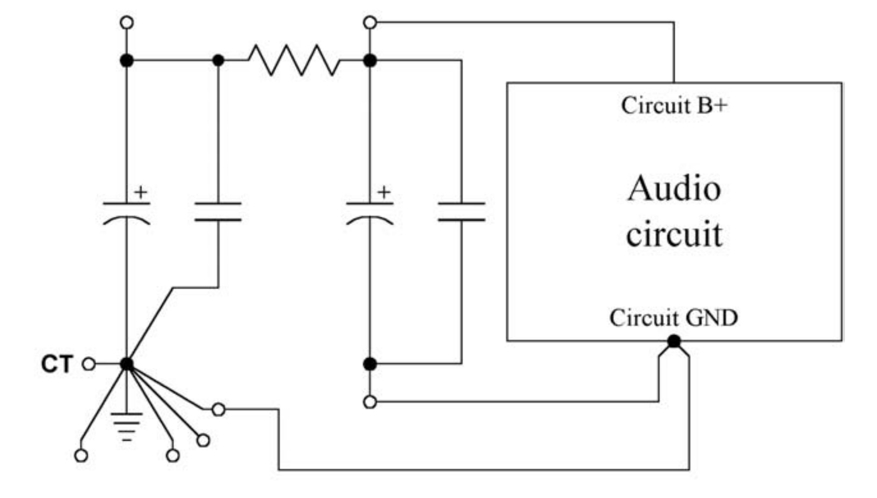



**FIGURE 5: Alternative ground hookup.**

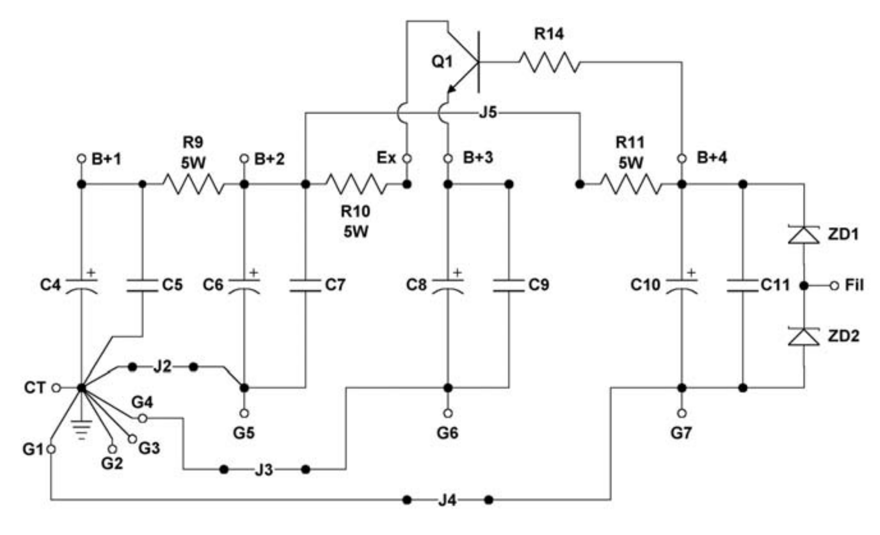



**FIGURE 6: Full emitter-follower regulation. J5 is used, J6 and J7 are omitted; ZD1 and ZD2 are substituted for R12 and R13. R11 serves as current-limiting resistor for the zener string; R10 as dropping and current-limiting resistor for the pass device. You should mount the pass device Q1 on a heatsink which is mounted off-board. R14 is a stopper, which you should connect directly to the pass device. Q1 could also be a MOSFET or Darlington pair.**

nal of the pass device.

If the pass device is bipolar rather than a MOSFET, the regulator will have a lower output impedance if you use a Darlington configuration.

## MECHANICAL CONSIDERATIONS

The four mounting holes in the board are sized for 4-40 screws. Because of the high voltages on the board when it's used for tubed circuits, I strongly recommend you mount the board on non-conductive standoffs.

All of the square input-output connecting pads on the board are drilled for 0.042″ pins; while it's certainly possible to solder wires directly into the holes, I don't recommend doing so for most home construction projects. Changes in hookup, desoldering, and resoldering are almost inevitable accompaniments to experimental designs, and these take their toll on printed circuit boards, leading to lifted traces and unreliable connections.

Making changes to pins is much less destructive; for several years now I've used Vectorbord K30C/M pins, which are easy to insert (even without a specialized tool) and hold up well. They project below the board; be sure to take that into account when choosing the length of your standoffs. Ideally, the bottom of the pins should be at least ½″ from the mounting surface. You can get 1000 of them for about $50 from Mouser Electronics, which ought to last a very long time. Be sure the cat doesn't knock over the can.

(Note: I discovered after writing the first draft of this article that the K30C/M pins seem to have been discontinued. Nuts! But there are still plenty of other 0.042″ pins available.)

Nicely flush components look neater on the board, but run hotter. I mount rectifier diodes, resistors, and zeners about ¼″ above the board to let some air circulate around them; this also keeps any heat they generate from toasting the board quite as badly. You can use plastic spacers on the leads, but I think those are necessary only when the leads are light gauge and liable to flex.

Note that while J2-7 are used to select options on the board, J1 is mandatory. That was the only trace I could not route on the bottom side of the board, and it seemed wasteful to turn this into a two-sided board for a single trace. Use 18-gauge wire, insulated, for this and the other jumpers; at the low currents normal for low-powered tubed circuits, heavier wire won't make a difference.

## OTHER USES

Just because this supply board was designed for tubes' B+ circuits, that doesn't mean it can't be employed elsewhere. You could, for example, use 75V capacitors with an appropriately sized transformer to produce a +48V phantom power supply, regulated or otherwise.

Let me remind you again that this board is designed only for nominal voltages (at B+1) of 350V or below, with a +10% permissible overvoltage. Anything higher, and you're out of bounds. 350V, of course, is plenty high to zap you good and proper; exercise appropriate caution as you would around any dangerous voltage. Unplug the supply and wait for the bleeder resistors to do their work before you get near the board, or any other energized parts in your project.

## PARTS AND CHOICES

**Table 1** is a generic parts list. The parts that never vary are shown in detail, while the others are given more generalized descriptions. The resistor values (except for R13) are left open; fill in the xxx designations in the part numbers. Most of the parts are available in the US from Digi-Key, but I've included the option of using 10nF 1kV ceramic disc capacitors in the snubbers. Digi-Key inexplicably doesn't carry these, but Allied Electronics does. They have a minimum purchase, though, so you'll want to order other items at the same time.

The capacitors listed are the maximum values that will fit onto the board; as you'll see in the worked example, you may choose not to use the biggest caps. Snubber parts on the generic list are left wide open; see Jim Hagerman's article for information on choosing them.

You're not, of course, limited to the brands I specified. Snap-in capacitors come in standard case sizes, and you could easily substitute Nichicons or another brand if you prefer them. The resistors' sizes are pretty standard, too. If you go with solid-state rectifiers, though, do stick with the ones I specified, or seek out similar fast-switching diodes.

## A WORKED EXAMPLE

Look at a typical application for this board. I chose a small guitar amp—in fact, the one that inspired the board's design. It uses an EL84 power pentode, single-ended, and two halves of a 12AX7 as an input amplifier and a driver. The power requirements are:

| | |
|---|---|
| Plate | 280V at 38.4mA |
| Screen | 262V at 4.25mA |
| Driver | 200V at 0.45mA |
| Input | 172V at 0.45mA |

For straight cascaded filtering 1>2>3>4, I'll use J6, and omit J5. The transformer's center-tapped, and I want to use a 6X4 rectifier tube, so I'll leave out all the rectifier diodes on the board, along with their snubbers and parallel resistors, and R8; the cathode of the rectifier tube will connect directly to B+1.

**TABLE 1 Basic Parts List**

| Item | value | mfg. | mfg. part # | sup. | sup part# |
|---|---|---|---|---|---|
| R1-3 | 3W metal oxide | Panasonic | | DK | PxxW-3BK-ND |
| R4-7 | 300k 2W metal oxide | Yageo | | DK | 300KW-2-ND |
| R8 | 5W maximum, wirewound | Yageo | | DK | xxW-5-ND |
| R9-11 | 5W maximum, wirewound | Yageo | | DK | xxW-5-ND |
| R12 | 2W metal oxide | Yageo | | DK | xxxW-2-ND |
| R13 | 75k 1W metal oxide | Yageo | | DK | 75KW-1-ND |
| C1-3 | varies | Epcos | MKP series | DK | |
| or | 0.01µF maximum 1kV c.D. | Vishay | 5HKS10 | AL | 926-3258 |
| C4, 8, 10 | 220µF maximum 450V el. | Panasonic | ECO-S2WB221CA | DK | P10163-ND |
| C6 | 470µF maximum 450V el. | Panasonic | ECO-S2WB471EA | DK | P10170-ND |
| C5, 7, 9, 11 | 0.01µF max. 630V pp. | Panasonic | ECQ-P6104JU | DK | P3521-ND |
| or | 0.01µF maximum 400V pp. | Panasonic | ECQ-P4104JU | DK | P3488-ND |
| D1-4 | 1000V 1A | Diodes Inc. | UF1007 | DK | UF1007DICT-ND |
| [D5, 6] | 5W zener | Microsemi | 1N53xx | DK | 1N53xxBMSTR-ND |

DK = Digi-Key, AL = Allied Electronics

I'll use star grounding for the screen and driver sections, but a "looped" ground (I don't have a better name for it) for the input section. That means I'll include J2 and J3 but omit J4; I'll run a wire instead from G7 to the input stage ground, and another from that ground to G1.

How much current will I draw from each supply terminal? Remember that the filter sections are cascaded, so each one not only supplies its audio stage, but also provides current for the next section. The last stage also supplies 1mA of bleeder current. The supply requirements now look like this:

$$B+1 = 280V;\ i_1 = 44.55mA$$
$$B+2 = 262V;\ i_2 = 6.15mA$$
$$B+3 = 200V;\ i_3 = 1.9mA$$
$$B+4 = 172V;\ i_4 = 1.45mA$$
$$\text{Fil float} = 75V;\ i_5 = 1.0mA$$

Ohm's law determines the resistor values:

$$R9 = (B+1 - B+2)/i_2 = 18V/6.15mA = 2{,}927\Omega$$
$$R10 = (B+2 - B+3)/i_3 = 62V/1.9mA = 32{,}632\Omega$$
$$R11 = (B+3 - B+4)/i_4 =$$

$28V/1.45mA = 19{,}310\Omega$
$R12 = (B{+}4 - \text{fil float})/i_5 =$
$97V/1.0mA = 97k$

How much power will they dissipate? Under steady-state conditions, using the nearest standard values and computing power dissipation using the formula $P_d = i^2 * R$:

$R9 = 3k; P_d = 0.113W$
$R10 = 33k; P_d = 0.119W$
$R11 = 20k; P_d = 0.042W$
$R12 = 100k; P_d = 0.1W$

That's pretty benign; you'd think the resistors could all be half-watt units. There's a pitfall, though: During turn-on, as C4 is charging, C6's voltage will lag behind it, so a significant voltage difference will appear across R9, and that will dissipate wattage. Resistors can stand a surge greater than their rated power for a short time, but not beyond, and for best reliability it's wise not to push them.

If this supply used solid-state rectifiers, I'd make all the resistors nice and hefty, just for insurance. With a slow-turn-on tubed rectifier, it's less of an issue, so I think 2W carbon comp or metal oxide resistors will be adequate for R10-12—besides which, 5W wirewounds are hard to find in values greater than 3k. Oops: In fact, Yageo doesn't make them bigger than 2k, so I'll need to find another brand for R9. Huntington will do (next page in the Digi-Key catalog).

OK, how about capacitors? The makers of tubed rectifiers tell you not to hang huge capacitors on them, because it shortens their lives drastically. Well, I can testify that a 68µF cap in the C4 position doesn't seem to have shortened the rectifier's life in the little Kalamazoo guitar amp that inspired this project; the 6X4 that was there when I bought the amp four years ago is still chugging along happily. That's not a scientific experiment, perhaps, but it's a good enough data point for the moment. I'll use a 68µF 450V cap for C4.

What will the ripple be? First, it's necessary to figure out what the equivalent load on the supply is, in ohms. B+1 = 280V, and it's supplying a total of 44.55mA; by Ohm's Law, the equivalent load is 280/0.04455, or 6285Ω.

There's a calculator at http://hyperphysics.phy-astr.gsu.edu/hbase/electronic/rectct.html#c3 that computes approximate ripple voltages. Plug in a $V_{pk}$ of 290V (an approximation, but it'll do) and an equivalent load of 6285Ω, and you get approximate ripple of 1.4V. Does that seem high? The original amp design used a 20µF capacitor on B+1, and that should produce a ripple voltage of about 5.5V. The amp didn't produce any audible hum; part of the reason was power-supply rejection in the output circuit, but the more significant reason was that the speaker, in its open-backed cabinet, has significantly puny response at 120Hz. Sometimes lo-fi lets you get away with murder.

What about subsequent stages? Say I made the caps 47µF apiece. The resistor-capacitor combinations form low-pass filters; the low-pass filter for B+2, for example, is formed by C6 and the parallel combination of R9 and R10. Its cutoff frequency is 1.23Hz. The ripple frequency is 120Hz. (I live in the US; in most of the world, the ripple frequency would be 100Hz.)

Because a passive RC filter approaches 6dB/octave attenuation, the ripple will be reduced at B+2 by a factor of about 1.23/120, or 0.0103x. If the ripple at B+1 is 1.4V, the ripple at B+2 will be about 14mV.

I'll spare you further arithmetic; here are the ripple voltages at the four terminals:

B+1: 1.4V
B+2: 14mV
B+3: 32µV
B+4: 51nV

Just for reference, that last number is about -144dBu. Cascaded RC filters are powerful tools.

Wait a minute, though. This is a guitar amp, not a hi-fi amp. What if I wanted a nice, saggy, badly regulated supply, like the gloriously dirty amps of old?

It's doable; Panasonic's NHG and EB series caps come in smaller capacitance values at appropriate voltages, and I could duplicate the little Kalamazoo's original 20-10-10µF supply cascade if I wanted to. These caps have radial leads 7.5mm apart instead of the 10mm pin spacing standard on TS-series caps and their equivalents, so I'd need to spread the leads a bit. For mechanical stability (guitar amps get slammed around), it'd probably be a good idea to hot-glue such caps in place.

For bypass caps, I'll use the smaller 0.1µF 400V Panasonic polypropylenes.

**Table 2** is a detailed parts list for the worked example. Note that several of the pieces aren't used, because I'm using the board with a tubed rectifier. The cost column is drawn from my most recent printed Digi-Key catalog; prices have been fairly stable lately (knock wood!), but to get more exact figures you should check their website.

The total cost for the supply board, including the rectifier tube and its socket, is about $28, which doesn't include the power transformer, nor mechanical parts (standoffs, screws, terminal pins, and the board itself). **Photo 1** shows the board stuffed and wired for the above example. For now, I decided not to bypass filter capacitors after the first stage.

**TABLE 2 Parts List, Worked Example**

| Item | value | mfg. | mfg. part # | sup. | sup part. # | cost |
|---|---|---|---|---|---|---|
| R1-8 | not used | | | | | |
| R9 | 3.0k 5W wirewound | Huntington | | DK | ALSR5F-3.0K-ND | $1.30 |
| R10 | 33k 2W metal oxide | Yageo | | DK | 33KW-2-ND | $0.18 |
| R11 | 20k 2W metal oxide | Yageo | | DK | 20KW-2-ND | $0.18 |
| R12 | 100k 2W metal oxide | Yageo | | DK | 100kW-2-ND | $0.18 |
| R13 | 75k 1W metal oxide | Yageo | | DK | 75KW-1-ND | $0.15 |
| C1-3 | not used | | | | | |
| C4 | 68µF 400V electrolytic | Panasonic | ECO-S2GA680CA | DK | P6839-ND | $3.45 |
| C6, 8, 10 | 47µF 400V elec. | Panasonic | ECO-S2GA470BA | DK | P7520-ND | $10.35 |
| C5, 7, 9, 11 | 100nF maximum 400V pp. | Panasonic | ECQ-P4104JU | DK | P3488-ND | $3.28 |
| D1-6 | not used | | | | | |
| V1 | 6X4 rectifier tube | | | TE | 6X4 | $7.45 |
| VS1 | 7-pin tube socket | | | TE | 7pinchpls | $0.79 |

**total cost: $27.31**

DK: Digi-Key, TE: Triode Electronics

## ENVOI

What began as a simple design for a particular project evolved into something more flexible. It's still simple in concept, but the details have been left as loose as possible; you can configure the board in many different ways, and I'm sure there are folks who will make it sit up and do tricks I haven't thought of.

Good power supplies are the backbone of good audio. This board will, I hope, provide the backbone for many good projects. Enjoy! *aX*

**PHOTO 1: The Gamp power supply board.**

## SHOULD YOU MAKE ONE?

For many years I made my own circuit boards, but I've sworn off. The chemicals involved are toxic, and what do you do when they're no longer useful?

A professor of mine once postulated a series of laws having to do with environmental issues, and one of them was, "Everything has to go somewhere." The chemicals industrial circuit board makers use are just as toxic, but they have disposal resources not available to home users, with government inspectors looking over their shoulders to make sure the chemicals are disposed of as safely as possible. I hope.

If you still want to make your own boards, the foil side is shown in **Fig. 7**, and the component side in **Fig. 8**. For best results, you should probably download these drawings from audioxpress.com rather than try to reproduce them from the magazine, to avoid the vagaries of copying a printed magazine.

You could also buy them from me. I've done a run of the boards, with silkscreening and solder masking, produced and drilled by Futurelec. They did a good job. I'm selling the boards for $24 apiece plus shipping and handling (with a small discount if you buy more than one). No, I don't expect to get rich. For information about ordering, please e-mail me: pstamler@pobox.com. ■

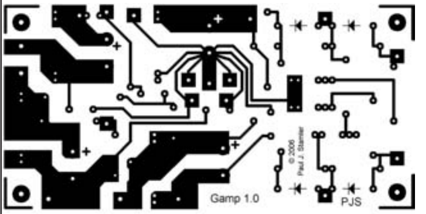



**FIGURE 7: Foil side.**

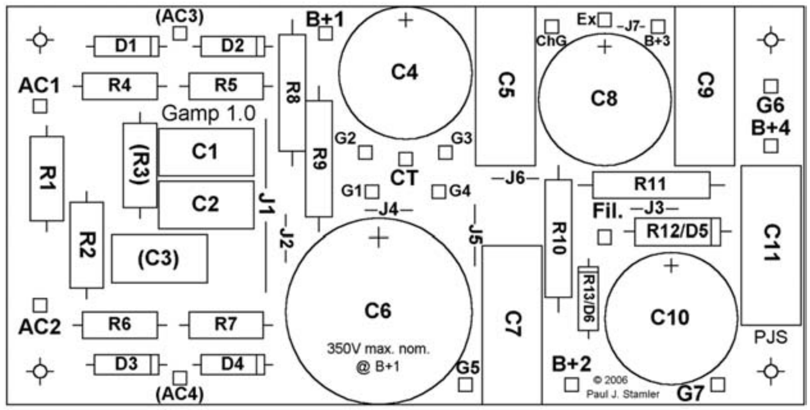



**FIGURE 8: Component side**


**REFERENCES**

1. Jim Hagerman, "Calculating Optimum Snubbers," 1/98 *Audio Electronics*, p. 26.
2. Ben Duncan, "A State-of-the-Art Preamp: AMP 02," *Hi-Fi News & Record Review 34:11 (November 1989)*, p. 45.


**SOURCES**

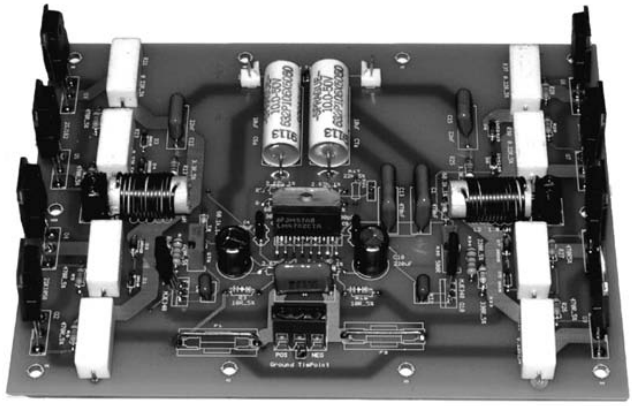
Allied Electronics, Inc.
7410 Pebble Drive
Fort Worth, TX 76118
1-866-433-5722
www.alliedelec.com

Digi-Key Corp.
701 Brooks Ave. South
Thief River Falls, MN 56701
800-344-4539
www.digikey.com

Triode Electronics
5633 W. Irving Park Rd.
Chicago, IL 60634
733-871-7459
http://store.triodestore.com/

By Jack Walton

# High End 120W **MOSFET** IC Driven Amp



National Semiconductor's LM4702 meets up with Renesas lateral MOSFETs.

National Semiconductor has been fast at work designing analog products for the high-end audiophile market. As fortune would have it, these products such as the LM4780 Power Amplifier Chip and the LM4702 Driver Chip are easy to implement into high-quality designs for the audio DIYer.

National recently introduced the LM4702, a high-voltage driver chip that interfaces bipolar or MOSFET output devices. The LM4702 provides two *small signal, input stages* and *voltage amplification stages (VAS)*. All you need to do is design a suitable bias system and output stage.

Not to be outdone in making the job easy for us amateurs, National provides a comprehensive design outline, guide to printed circuit board construction, ideas for the power supply, bill of materials, and suggested layout. In an application note dated May 2006[1], the company described an amplifier using high-quality construction techniques and off-the-shelf passive components. The result is an amplifier with a vanishingly low level of distortion, 0.0006%.

The amplifier described here comes close to achieving the performance stats described by National, but this *DIYer* has neither a Faraday shielded room nor Audio Precision distortion analyzer with which to squeeze the last iota out of performance. At 1kHz, 10W into 8Ω THD+N is around 0.004%. You can probably assemble the amplifier for $150, with the bulk of the expense going towards a suitable power transformer and chassis in which to house the device.

## PICKING UP WHERE NATIONAL LEAVES OFF

The LM4702 is limited by its ability to sink or source a maximum current of 10mA from source to sink pins. As a result, output bipolar devices must use a Darlington configuration for the kinds of output power which now seem *de rigueur*. Nevertheless, National hints in their literature at the possibility of using MOSFET output devices, suggesting that those with low turn-on voltages might be appropriate in such an application.

Picking up on this idea, I designed a very high-quality amp that uses lateral MOSFETs from Renesas as the output devices. Lateral MOSFETs enter into their linear phase at a lower bias voltage than vertical MOSFETs and are well-suited for the task.

## THE LM4702 DRIVER

The LM4702 driver (**Fig. 1**) provides two sets of input and VAS stages. As with the Overture™ series LM3886 and LM4780, the device also provides a mute function. The chip requires external compensation; for the circuit in this article I use 30pF silver mica capacitors, as did Brasfield in his application note.

National leaves the method of biasing the output devices to the end user. In this case I have employed a VBE multiplier with a PNP device, the MPSA56. The schematic (one channel) of the completed driver for the amplifier is shown in **Fig. 2**.

In many respects the input and VAS stages of the amplifier do not differ from the high-performance unit described in National's application note. The lateral MOSFETs require a few tricks to keep from oscillating, however. In this case I placed 330pF ceramic capacitors on each of the n-channel devices from gate to source and compensated the feedback loop by bypassing the feedback resistor with a 10pF capacitor, C14 in series

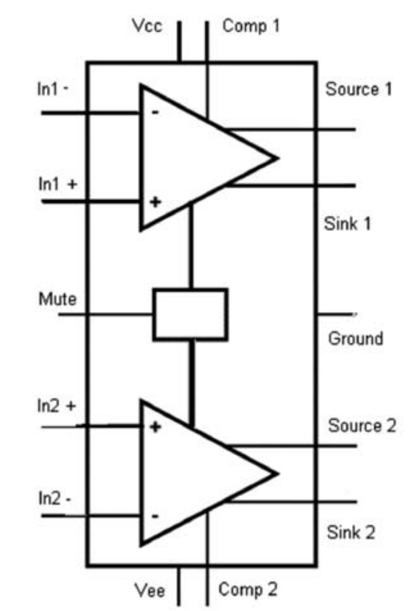



**FIGURE 1: The LM4702.**

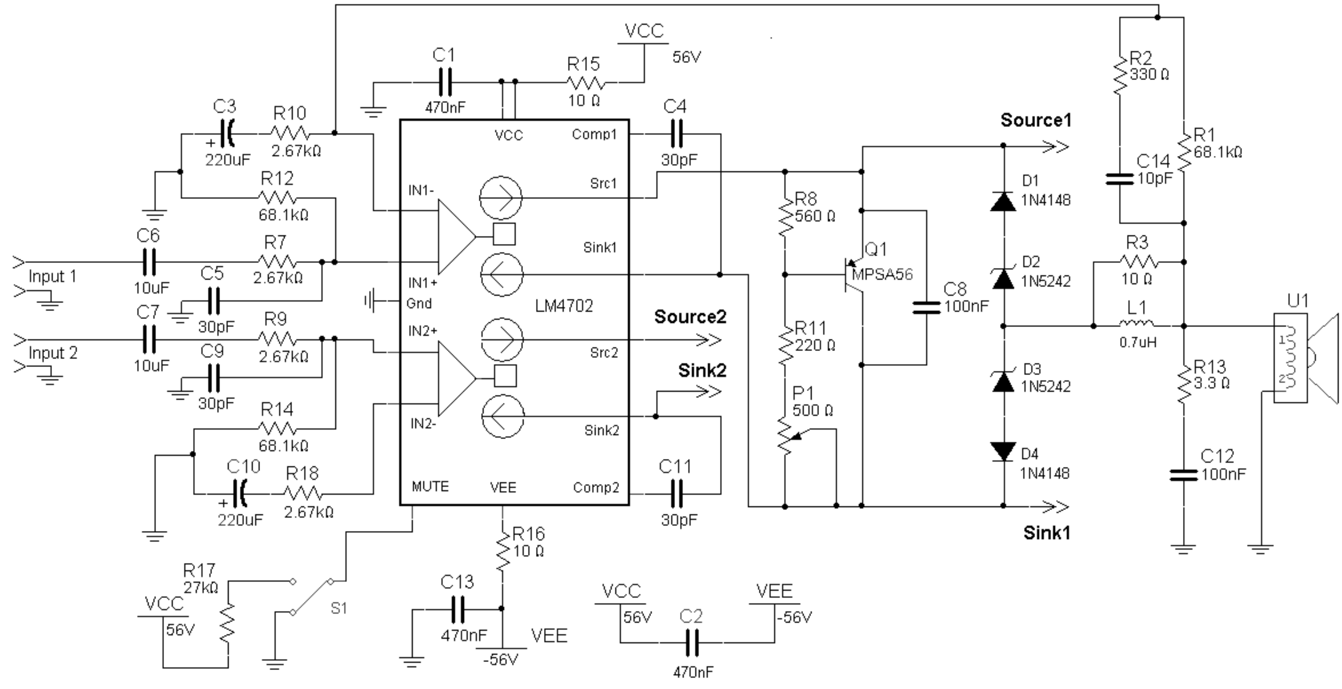



**FIGURE 2: The schematic of the completed driver (one channel).**

with a 330Ω resistor across the feedback resistor (R1). In one iteration of testing the amplifier, I placed 47pF and 22pF mica caps from drain to gate on the 2SK1058 and 2SJ162 output transistors, but found the prior setup to provide the best balance between power bandwidth and distortion.

The LM4702's supply lines are decoupled with 10Ω resistors (R1, R24) to tweak the performance a bit. The author of the application note pointed out that you could omit capacitors C6 and C7 for critical listening input. If you do this, however, a servo might be necessary to prevent DC from appearing on the output. Local power pin decoupling is important with the LM4702.

I used 470nF polypropylene capacitors for C1, C2, and C13 because I had these on hand. I placed capacitor C2 across the power input pins of the printed circuit board. Diodes D1-D4 protect the devices from gate-source breakdown. For simplicity sake I have shown a SPST switch used to control the LM4702's mute pin. One to two milliamps should flow into this pin to bring the amp out of mute. The output stage schematic is shown in **Fig. 3**.

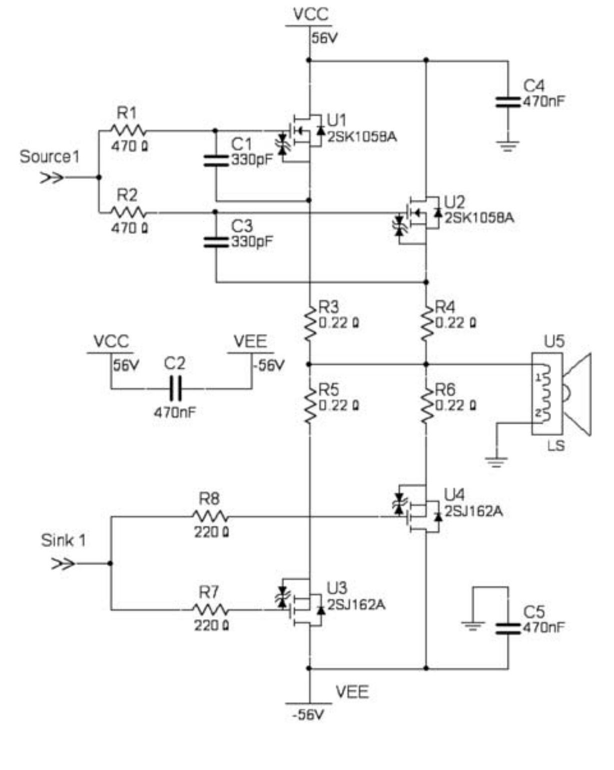



FIGURE 3: The output stage.

## POWER SUPPLY

The power supply used for the amplifier is fairly straightforward, employing a 400VA transformer, On-Semi MUR860 diodes, and 39,000µF capacitance for each rail. Both AC inputs to the power transformer are switched and fused. In my amplifier I filtered the AC input with a Corcom 10VK1 filter. This takes up little space and will handle the current demands without difficulty. The filter keeps electromagnetic and radio interference from propagating into the power supply and amplifier circuitry.

I use General Electric CL-30 inrush current limiters to protect the diodes. Resistors R2 and R4 discharge capacitors C5, C6 in a period of several minutes. Given the large capacitors used, it will take a few minutes to fully discharge C5 and C6 by 99%. You can use the auxiliary regulated voltages to drive speaker protection circuitry, a DC servo, or crossover if you think this is necessary.

A typical diode snubber circuit is illustrated in **Fig. 4** for those concerned about diode switching noise. The values of R and C in the snubber depend upon the secondary leakage inductance of transformer T1. A discussion of the values and the necessity of diode snubbers is contained in "Calculating Optimum Snubbers," which you can find on Jim Hagerman's website www.hagtech.com/pdf/snubber.pdf and/or in *The Audio Amateur*[2].

**Note:** As described, I used 39,000µF 80V DC capacitors in the LM4702 power supply. Each of these capacitors is tested prior to shipment and may contain a residual charge. This could cause a nasty little shock if you are not careful in handling the devices.

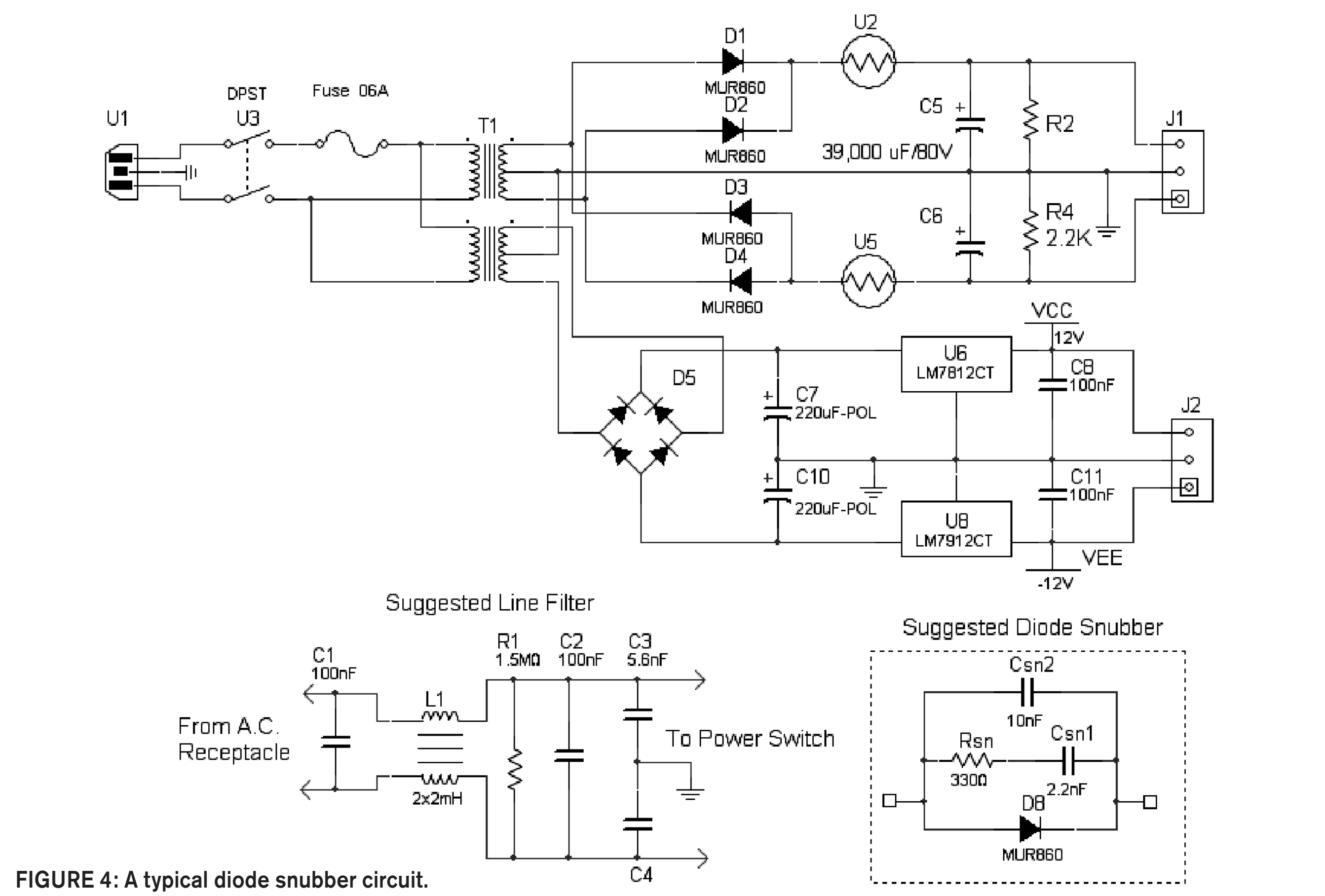



FIGURE 4: A typical diode snubber circuit.

## PRINTED CIRCUIT AND LAYOUT

I originally designed the printed circuit board incorporating both driver and output devices and used this design for the testing and performance data found at the end of this article. For ease of use, however, this article includes a printed circuit board design that separates the driver and output boards. This will allow much higher power designs, the use of lateral or vertical MOSFETs, or bipolar Darlington devices if you so choose.

National recommends that you keep the layout as symmetrical as possible, and my performance tests over the course of several months convinced me that this discipline will improve the amplifier's performance by several thousands of a percent of THD. No ground plane is shown, and it is National's recommendation not to include one in your design if you want the best performance.

To simplify the connection of the power supply leads, I ran a #12 stranded wire from the front to back of the printed circuit board. Otherwise, the power, output, and speaker ground traces at 200mils seem to be adequate for the job. **Figures 5-7** show the bottom, silk screen, and top printed circuit board

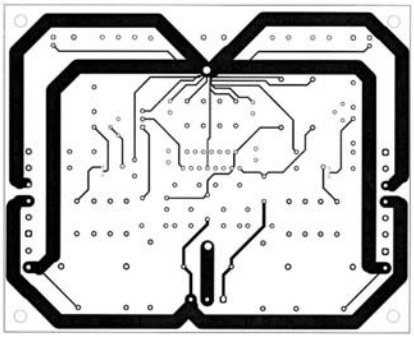


FIGURE 5: LM4702 driver board bottom.

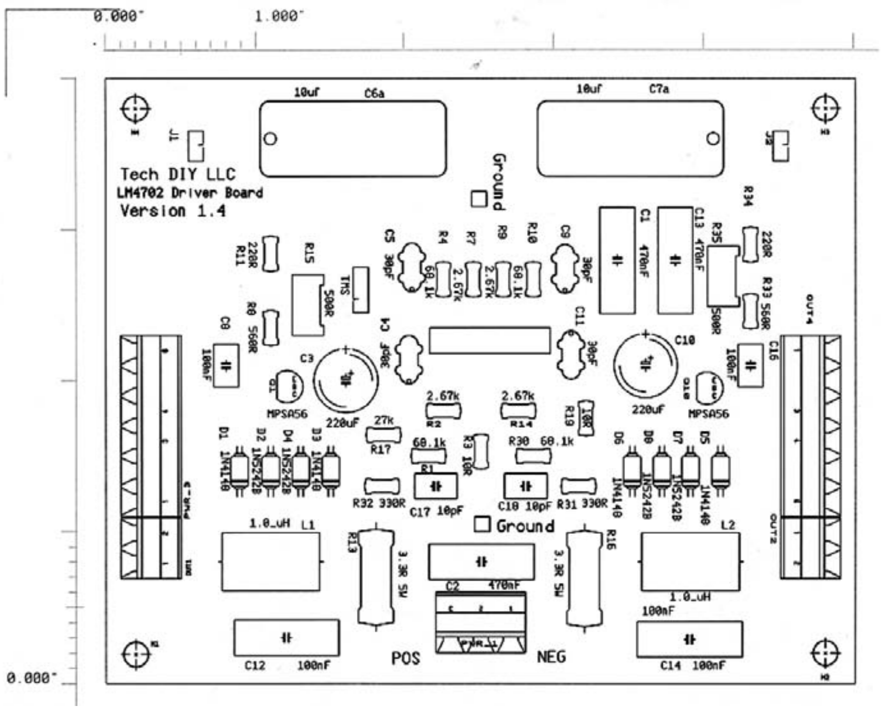



FIGURE 6: LM4702 driver board silkscreen.

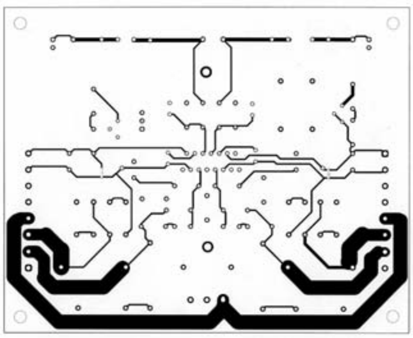


FIGURE 7: LM4702 driver board copper top.

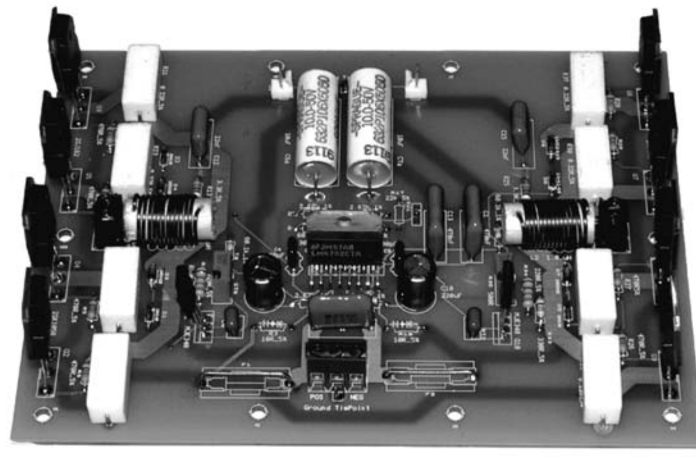


**PHOTO 1: The completed amplifier module used for testing.**

**PARTS LIST**

(Part Numbers relate to the schematics, but the quantities for those parts with an asterisk represent the quantities necessary for two channels)

| **Driver Board** | | |
|---|---|---|
| **Part Number** | **Description** | **Quantity** |
| C3, C10 | 220μF Electrolytic | 4 |
| C1, C2, C13 | 470nF 250V DC Polypropylene | 2 |
| C4, C5, C9, C11 | 30pF Silver Mica | 4 |
| C6, C7 | 10μF Polycarbonate or Polypropylene | 2 |
| C8* | 100nF Polypropylene | 2 |
| C14 | 10pF ceramic | 2 |
| D2, D3 | 1N5242B | 4 |
| D1, D4 | 1N4148 | 4 |
| L1* | 0.7μH (15 turns #16, 0.500 i.d.) | 2 |
| Q1* | MPSA56 PNP | 2 |
| R1*, R12, R14 | 68.1k 1% 0.5W metal film | 6 |
| R2* | 330Ω 0.5W metal film | 2 |
| R3* | 10Ω 5% 25W | 2 |
| R7, R9, R10, R18 | 2.67K 1% 0.5W Metal Film | 4 |
| R8* | 560Ω 5% 0.5W metal film | 2 |
| R11* | 220Ω 5% 0.5W Metal Film | 2 |
| R15, R16 | 10Ω 5% 0.5W metal film | 2 |
| R13* | 3.3Ω 5% 5W | 2 |
| R17 | 27k 5% 0.5W Metal Film (value depends upon power supply – see text) | 1 |
| P1* | 500Ω linear Bourns 3296Y or equivalent potentiometer | 2 |
| S1 | SPDT Switch for mute function | 1 |
| U1 | LM4702A | 1 |
| **Output Stage** | | |
| C1*, C3* | 330pF Ceramic 100V | 4 |
| C2*, C4*, C5* | 470nF Polypropylene | 6 |
| Q1*, Q2* | 2SK1058 Lateral MOSFET | 4 |
| Q3*, Q4* | 2SJ162 Lateral MOSFET | 4 |
| R1*, R2* | 470Ω 5% 0.5W | 4 |
| R7, R8* | 220Ω 5% 0.5W | 4 |
| R3*, R4*, R5*, R6* | 0.22Ω, 5W | 8 |
| **Power Supply** | | |
| C1, C2 † | 100nF 250V AC Polypropylene | 2 |
| C3, C4 † | 5.6nF 250V AC Polypropylene | 2 |
| C5, C6 | 39,000μF 80V DC | 2 |
| C7, C10 | 220μF 25V DC Electrolytic | 2 |
| C8, C11 | 100nF Polyester | 2 |
| D1, D2, D3, D4 | MUR860 | 4 |
| D5 | 50V 1A Bridge | 1 |
| L1 † | 2 × 500μH Bifilar Wound Choke | 1 |
| T1 | 44VCT, 12VCT 300VA Transformer | 1 |
| U2, U5 | CL-30 InRush Current Limiter | 2 |
| U6 | LM7812 12V Positive Regulator | 1 |
| U8 | LM7912 12V Negative Regulator | 1 |
| R1 † | 1.5MΩ | 1 |
| † Components are contained in Corcom 10VK1 EMI Filter | | |
| Rsn, Csn | See Text | |

images in respective order (although reduced to save space). The Gerber files are available for free download at www.audio-Xpress.com.

## COMPLETED AMPLIFIER MODULE

**Photo 1** shows the completed amplifier module used for the testing. There is sufficient space given between the transistors to distribute the heat across the sink.

To the greatest extent possible, I have maintained symmetry between the two channels of the amplifier. You could certainly consider different part orientations, particularly if you want to place the output devices on the rear of an amplifier chassis. This layout, however, allows for the placement of two large heatsinks on the right- and left-hand sides of the chassis.

I strongly recommend using a TO-220 heatsink on the LM4702. I have found that as the chip warms up, the harmonic distortion of the amplifier will tend to increase as the internal temperature of the amplifier rises. Under certain circumstances the LM4702 will thermally cycle if it isn't heatsinked.

## SETUP AND OPERATION

The amplifier is pretty easy to set up. (A Variac is helpful, however.) Short the input connectors to ground and bring up the line voltage. With a voltmeter, measure the voltage across the 0.22Ω source resistors. Adjust the potentiometers for 22mV of voltage for each device. This equates to 100mA of bias current. I let the amplifier cook for another 15 minutes and check the bias to make sure it hasn't wandered.

## PERFORMANCE

The LM4702 driven MOSFET amplifier performs quite well. While I was unable to attain the remarkable 0.0006% that National demonstrated in application note 1490, the lateral MOSFET doesn't seem to get into quite as much trouble at high power levels. For comparison purposes, at 10W output power the lateral MOSFET amplifier exhibited THD of 0.004%, while the bipolar Darlington unit from National displayed 0.001%.

The test setup consisted of both Tek-

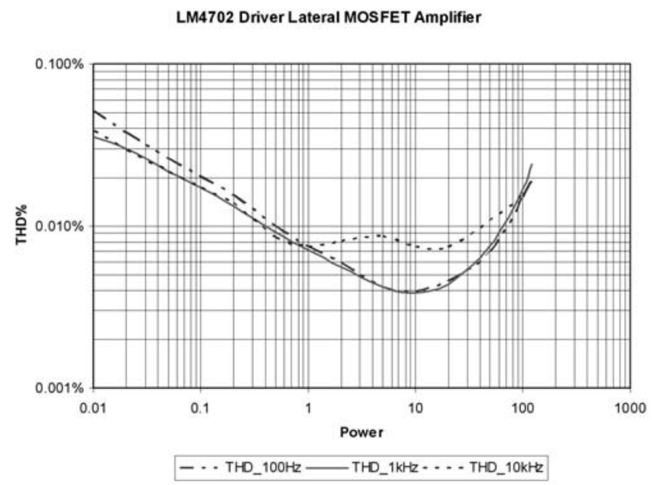



**FIGURE 8: Plot of total harmonic distortion vs. power at 100Hz, 1kHz, and 10kHz.**

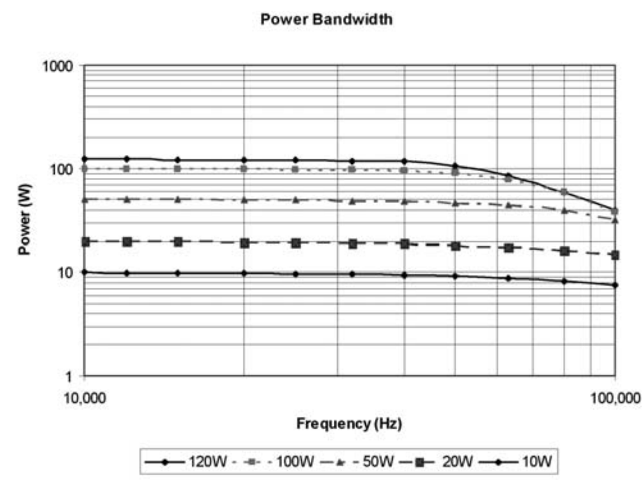



**FIGURE 9: Power bandwidth from 10kHz to 100kHz.**

tronix SG505 and AA501 and Boonton 1120 Distortion Analyzers, a Hewlett-Packard HP 3577 Network Analyzer, Tektronix TDS3012B oscilloscope, and an 8Ω resistive load. The maximum power that I was able to attain from the amplifier as shown was a continuous 150W, at which level it expressed its unhappiness with me.

The plot of total harmonic distortion versus power at 100Hz, 1kHz, and 10kHz is shown in **Fig. 8**. Power bandwidth, from 10kHz to 100kHz, is illustrated in **Fig. 9**.

## PARTS LIST

Most of these components are available from the usual sources including Digi-key, Mouser, Allied, Jameco, and so on. You can use the Hammond 182P22 225VA rated transformer (available from Digi-key and Mouser) as long as you don't use the amp to drive servo motors or some industrial load. A heavier duty imported transformer is available from Antek, in North Arlington, N.J. Their AN3240 is rated 300 V.A. for continuous industrial service in servo use. (Antek is an ISO-9001 rated company serving the CNC industry.) Their AN3240 is rated 300VA for continuous industrial service in servo use. You can purchase Antek's transformers from their website, www.toroid-transformer.com.

It is important that the line capacitors C1, C2, C3, and C4 be rated for AC operation. High quality 250V AC polypropylene X and Y capacitors should be used in these positions unless you use the Corcom 10KV1 EMI Filter. aX

## REFERENCES

1. Brasfield, Mark, "LM4702 Amplifier, National Semiconductor Corporation Application Note 1490," National Semiconductor Corporation, Santa Clara CA, 2006.

2. Hagerman, Jim, "Calculating Optimum Snubbers," *The Audio Amateur* 3/94.

By Matthias S. Brennwald

# MATAA: A Free Computer-Based Audio Analysis System

Discover this useful software tool, which acts as a versatile audio analyzer.

I needed an audio analyzer to test my loudspeakers and amplifiers. However, the systems available on the market are either not flexible enough for my needs, don't work with my preferred computer platform, or cost more than what I am willing to spend. I therefore started a new DIY project "building" my own audio analysis system using my computer's soundcard. After a while, the project grew bigger and I realized that I produced a very versatile and powerful audio analyzer, which I call "Mat's Audio Analyzer" (MATAA).

The operating mode of MATAA is the same as with many other computer-based audio analysis systems. Apart from that, however, MATAA is a little different from most of these systems in many ways. For instance:

1. MATAA does not rely on proprietary hardware (such as an expansion card or a "switch box") to handle the sound input/output and the electrical connections to the device under test (DUT). In contrast, you can use whatever soundcard, cables, switches, plugs, amplifiers, and so on that you think are appropriate for your needs.
2. MATAA does not rely on a specific computer platform, because MATAA runs from within MATLAB or GNU Octave. These "number-crunching" programs run on virtually all current platforms (keep reading if you are not familiar with MATLAB or Octave).
3. MATAA is free software released under the GNU General Public License[1] (GPL). You are free to study the source code, adapt it to your needs, and release your improved version under the GPL. You can download MATAA from www.audioroot.net/mataa.html.

MATAA is essentially a collection of MATLAB/Octave programs, which provide the building blocks for analysis procedures to test all kinds of audio devices. Furthermore, MATAA contains ready-made scripts using these building blocks to conduct typical analyses and tests (e.g., to measure the impulse response and the frequency response of a loudspeaker). Alternatively, you can design specific test routines to efficiently analyze virtually anything that can be analyzed with audio test signals.

## HOW MATAA WORKS

Similar to most other computer-based audio analysis systems, MATAA feeds a test signal to the DUT and simultaneously records its response signal, which is then analyzed as desired (see **Fig. 1**, which I will explain in more detail). One notable strength of MATAA is that you can use MATLAB/Octave to generate *any* kind of test signal and then feed it to the DUT. Together with the wide range of MATLAB/Octave tools for data processing and analysis, this allows carrying out virtually *any* analysis you can think of.

MATAA does not have a whiz-bang graphical user interface. While this may seem a bit anachronistic for today's computer software, the lack of a graphical user interface is one of the reasons MATAA is so flexible. Have you ever experienced a program that couldn't do what you wanted because the corresponding button was missing, although the functionality would have been built into the software? Not with MATAA.

## MATAA SOUNDCARD REQUIREMENTS

In principle, any soundcard that allows simultaneous sound input and output ("full duplex sound") is suitable for use with MATAA. However, the quality of the soundcard will, of course, crucially determine the quality of the measured data. For instance, the maximum sampling rate determines upper frequency limit; the signal/noise ratio and the sampling bit depth determine the dynamic range; and, depending on the intended type of analysis, the number of input and output channels available may also

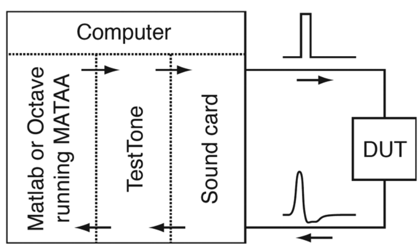



**FIGURE 1: Block diagram illustrating the operating mode of MATAA. MATAA sends and receives the test signal data to and from TestTone, which handles the sound output and input with the soundcard. The soundcard simultaneously feeds the test signal to the device under test (DUT) and records its response.**

be important (stereo soundcards are suitable for the vast majority of possible applications).

## MATLAB AND OCTAVE

MATLAB and Octave are "number-crunching" programs that provide an all-in-one environment for analysis, processing, and graphing of data. Both MATLAB and Octave run on various computer platforms and operating systems. They provide a powerful programming language, which, compared to many others, is easy to learn and understand. Programs written for MATLAB are (largely) compatible with Octave (and vice versa). While MATLAB and Octave are very similar, they differ in one important point:

MATLAB is a commercial product by The MathWorks company[2], whereas Octave is free software released by the GNU project[3].

One shortcoming of both MATLAB and Octave with respect to MATAA is their limited capabilities for sound input and output. While both MATLAB and Octave provide commands to play and record sound, these commands do not work on all operating systems. Furthermore, simultaneous sound input and output, a prerequisite for MATAA, is not well implemented.

Therefore, I wrote a short program called TestTone that handles the sound input and output for MATAA externally from MATLAB/Octave (**Fig. 1**). MATAA talks to TestTone from within MATLAB/Octave, so the MATAA user does not need to care about TestTone, which is based on PortAudio[4], a free cross-platform library for sound input/output. It is therefore straightforward to enable sound input and output for MATAA on all computer platforms supported by PortAudio.

At the time of this writing, I have compiled TestTone for MacOS X. Also, Sang Shu (sangshu@hotmail.com) has compiled TestTone for Windows (thanks, Shu!). MATAA therefore supports sound input and output on these platforms, but other platforms (e.g., Linux) are not yet supported. If someone agrees to compile TestTone for Linux or any other computer platform and provide it to the public through the MATAA website (with your nametag on it), you are more than welcome to do so!

## USING MATAA

In this section, I will show how to operate MATAA and how MATAA works in real-world applications. I ran these examples using Octave, but the operation and results would have been exactly the same with MATLAB. I used an Apple PowerBook G4 laptop computer with its generic built-in soundcard, which supports various sampling rates in the range of 32–96kHz with a sample depth of up to 24 bits. By feeding sine signals of varying frequencies from an analog signal generator to the soundcard input and analyzing the digitized result, I found that this soundcard has an efficient anti-aliasing filter, whose cutoff frequency is automatically adjusted to the sampling rate selected.

## RC HIGH-PASS FILTER

In this first example, I use MATAA to test a simple RC high-pass filter with $R$ = 4.79kΩ and $C$ = 220nF. While not rocket science, this example serves as a good introduction to how MATAA works. **Figure 2** shows the RC filter and how it is connected to the soundcard.

To start, I demonstrate how to feed a square-wave signal to a simple RC filter and measure its output signal. Using the MATAA command mataa_signal_generator, I produced a 0.1s long square-wave signal with a frequency of 1kHz and a sampling rate of 96kHz by typing the following command at the MATLAB/Octave prompt:

```
s = mataa_signal_generator('square',96000,0.1,1000);
```

The samples of the square-wave signal are now stored in the MATLAB/Octave variable **s**, but the signal has not yet been fed to the soundcard or the RC filter.

The mataa_measure_signal_response command feeds the square-wave signal to the input of the RC filter and simultaneously records the response signal at the output of the filter:

```
res = mataa_measure_signal_response(s,96000);
```

The samples of the response signal are now stored in the variable **res**. **Figure 2** shows one cycle of the response signal (this figure was produced using the mataa_plot_signal command, but you could just as well use the standard MATLAB/Octave **plot** command).

Apart from the slight high-frequency ringing, the shape of the signal looks exactly as expected for the RC high-pass filter. The high-frequency ringing is due to the steep anti-aliasing filter of my soundcard (I will discuss how to deal with signal distortions due to the anti-aliasing filter or other artifacts introduced by the sound hardware). In summary, typing only three commands was enough to produce a test signal, feed it to the RC filter, record the response signal, and plot the result.

In the next step, I demonstrate how to measure the impulse response of the filter and how to determine the transfer function of the filter in the frequency domain. By typing the following three commands, MATAA measures the impulse response using a white noise signal and calculates the frequency response (magnitude and phase) of the RC filter:

```
w = mataa_signal_generator('white',
96000,0.3);
h = mataa_measure_IR(w,96000);
```

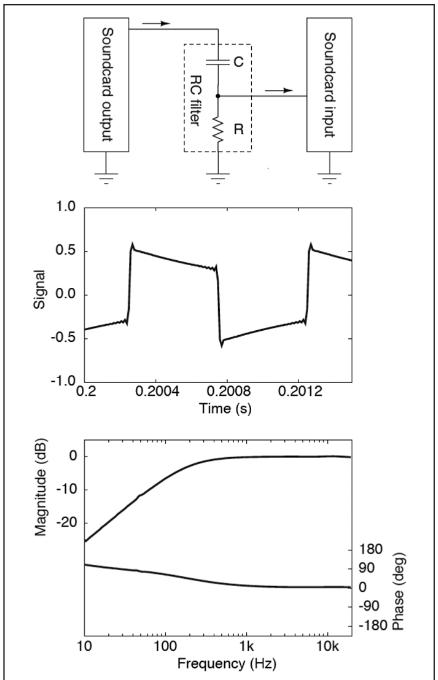



**FIGURE 2: RC filter analysis. Top: setup of the RC filter and sound hardware. Center: squarewave response of the RC filter. Bottom: frequency response of the RC filter (magnitude and phase corrected for excess phase).**

```
[mag,phase,f] = mataa_IR_to_FR(h,96000);
```

On the first line, a white-noise signal **w** with a sampling rate of 96kHz and a length of 0.3s is generated. On the second line, this white-noise signal is used to measure the impulse response of the RC filter using the MATAA command mataa_measure_IR. This command first feeds the signal in **w** to the RC filter and records the response of the RC filter (analogous to the prior square-wave test). The response signal is then deconvolved from the original signal (**w**), which results in the impulse response **h** of the RC filter.

On the third line, mataa_IR_to_FR is used to transform the time-domain impulse response to the frequency response using the Fourier transform (magnitude mag and phase phase as function of frequency f). The resulting frequency response (**Fig. 2**) corresponds to the expected first-order high-pass transfer function of the RC filter.

Note that the computer needs some time to process and feed the test signal data to the sound output. Also, while not the case in this example, the signal may be further delayed by the DUT itself or due to long signal travel times in the measurement setup (e.g., the time needed for a signal to travel from a loudspeaker to a microphone). If the signal recording stops immediately after the last sample of the test signal has been output from the audio output, the recoding of the DUT's output signal will therefore be truncated. By default, MATAA therefore extends the recording of the test signal by 0.1s before and after the test-signal output to avoid cutting off the recording (different delay values may be specified).

In the frequency domain, this delay in the DUT output signal corresponds to a phase shift that increases linearly with frequency. In the RC-filter example, the true phase shift is virtually zero at frequencies much higher than the cutoff frequency. You can therefore compute the excess phase due to the signal delay from the measured phase and the expected (zero) phase.

By specifying the frequency range where phase is expected to be zero (e.g., 5–20kHz), the following command removes the excess phase, leaving only the phase response of the RC filter (the so-called minimum phase, **Fig. 2**):

```
phase = mataa_phase_remove_trend
(phase,f,5000,20000);
```

As an alternative, MATAA also allows using the Hilbert transform to determine the minimum phase from the magnitude of the frequency response.

## LOUDSPEAKER IMPEDANCE

**Figure 3** shows the measurement setup to measure the electrical impedance of a loudspeaker[5] (or, in fact, anything else). The sound output is connected to a series combination of the loudspeaker and a resistor, which acts as an impedance reference. The value of the reference resistor should be of the same order as the impedance of the loudspeaker.

In contrast to the setup in **Fig. 2**, both stereo channels of the sound input are used. One channel records the signal voltage across the loudspeaker (DUT channel), while the other records the signal voltage at the sound output, which will be used as a reference (REF channel). By default, MATAA allocates the left channel to the DUT and the right channel to REF.

The purpose of using the second channel (REF) is to make the analysis immune to distortions of the original test signal by the sound output electronics. Instead of comparing the DUT signal to the original test signal generated by MATAA (as in the RC-filter example), the DUT signal is compared to the REF signal, i.e., to the true signal applied to the DUT.

Many soundcards are not designed to deliver undistorted output when directly driving loudspeakers or similar low-impedance loads. The REF signal may therefore be distorted with respect to the original test signal generated by MATAA. Furthermore, as previously illustrated, the anti-aliasing filter of the sound input may also distort the test signal. You can sidestep this distortion problem by comparing the DUT signal to the REF signal rather than the original MATAA signal, because the soundcard distortion affects both the DUT and REF signals in the same way.

The reference resistor R and the loudspeaker with its impedance Z constitute a voltage divider between the soundcard output and ground (**Fig. 3**). $U_A$ is the voltage between point A and ground (i.e., the voltage across the loudspeaker). $U_B$ is the voltage between point B and ground (i.e., the voltage across the loudspeaker and the reference resistor). The ratio of these two voltages is $U_B/U_A = (R+Z)/Z$. Solving this equation for the loudspeaker impedance gives:

$$Z = R\frac{U_A}{U_B - U_A}$$

For a sinusoidal test signal, $U_A$ and $U_B$ reflect the amplitudes of the sine signals at points A and B. Note that the sine signals may show a phase shift due to the complex nature of the loudspeaker impedance. Mathematically, this is expressed with complex values $U_A$ and $U_B$. For non-sinusoidal signals, you can compute $U_A$ and $U_B$ as a function of frequency using the Fourier transform of the signals measured at points A and B. Inserting these frequency-dependent values in the previous equation then yields the loudspeaker impedance Z as a function of frequency[5].

This procedure to determine the frequency-dependent impedance of a loudspeaker (or any other DUT) is implemented in the MATAA tool mataa_measure_impedance. With a reference resistor R = 8.6Ω (that's what came out of my parts box), the following command measures the loudspeaker impedance (magnitude mag and phase phase) in the frequency range from 10–10000Hz:

```
[mag,phase,f] = mataa_measure_impedance(10,10000,8.6);
```

By default, this command uses a sine-sweep test signal of appropriate length, selects a suitable sampling rate, and smoothes the result in 1/48-octave bands (other values may be specified). The smoothing attenuates the noise that will be picked up by the loudspeaker, because it also acts as a microphone. **Figure 3**, generated with the mataa_plot_impedance command, shows the result for a Fostex FE108Σ full-range driver operated under free-air conditions.

## ACOUSTIC LOUDSPEAKER ANALYSES

In this section, I demonstrate how to measure the impulse response of a loudspeaker, and how to calculate the step response, the anechoic frequency response, and a cumulative spectral decay diagram ("waterfall plot") of the loudspeaker. I recommend the book by Joe D'Appolito[5] for a thorough and well-written introduction to the methods and concepts involved in these analyses (and everything else you ever wanted to know about loudspeaker testing).

**Figure 4** shows the setup to measure the impulse response of a loudspeaker (the same Fostex FE108Σ as before). Both the microphone (a Behringer ECM8000) and the loudspeaker were mounted 90cm above the floor, with a distance of 100cm in between each other. Note that the loudspeaker is driven through a power amplifier to buffer the soundcard output signal. In contrast to the impedance analysis, calibration of the analysis using the REF channel is therefore not mandatory. Hence, for the sake of simplicity, I decided not to use the REF channel, although MATAA does allow calibrating the impulse-response measurement using the REF channel.

Acoustic analyses can be prone to environmental noise. I therefore used a maximum length sequence (MLS) test signal to achieve a good signal/noise ratio for the impulse-response measurement. Other signals such as pink or white noise are suitable, too.

The following commands first pro-

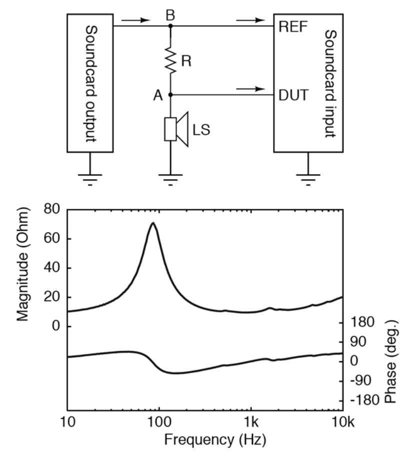



**FIGURE 3: Loudspeaker impedance analysis. Top: measurement setup. Bottom: result of impedance measurement for a Fostex FE108Σ driver in free air.**

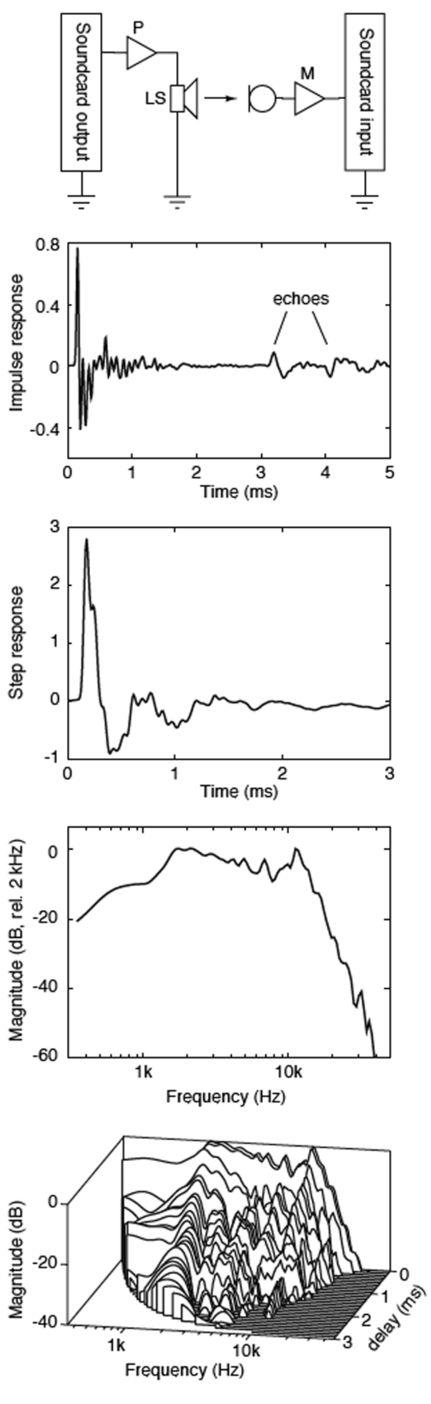



**FIGURE 4: Acoustic loudspeaker analysis. From top to bottom: measurement setup (P: power amplifier, M: microphone amplifier, LS: Fostex FE108∑ loudspeaker in free air), impulse response (with room echoes), step response (with echoes truncated, note the different time scale), anechoic frequency response, and cumulative spectral decay diagram ("waterfall plot").**

duce an MLS test signal (**s**, with a length of $2^{14}$ - 1 = 16383 samples), which is then used to measure the impulse response **h** of the loudspeaker using a sampling rate of 96kHz (analogous to the RC-filter example):

```
s = mataa_signal_generator('MLS',96000,0,14);
h = mataa_measure_IR(s,96000);
```

Note that the impulse response in **h** will show a delay due to the travel time of the test signal from the loudspeaker to the microphone (about 2.92ms for a distance of 100cm between the driver and the microphone).

The following two commands remove this delay and shorten the impulse response to a length of 5ms:

```
t0 = mataa_guess_IR_start(h,96000);
h = mataa_signal_crop(h,96000,t0,t0+0.005);
```

On the first line, the start time (**t0**) of the impulse response starts is determined automatically. On the second line, the impulse response is cropped to the range between **t0** and **t0**+5 ms.

**Figure 4** shows the resulting impulse response, which is followed by echoes from the floor and the walls of my room. The first echo occurs at about 3.1ms, which corresponds to the time delay of the echo from the floor with respect to the direct sound from the loudspeaker. You must remove the echoes to determine the anechoic frequency response.

The simplest method to do this is to shorten the impulse response to the time range that is free of room echoes (i.e., 0–3ms):

```
h = mataa_signal_crop(h,96000,0,0.003);
```

Alternatively, MATAA also provides a tool to multiply the impulse response by various types of window functions in order to attenuate the echoes while retaining the anechoic part of the signal. For the sake of simplicity, however, I won't discuss signal windowing here.

The impulse response is now free of room echoes, but still needs to be corrected for the frequency response of the microphone. This is done as follows:

```
h = mataa_microphone_correct_IR('Behringer_ECM8000',h,96000);
```

This command reads the frequency response of the microphone from the specified ASCII file and uses this data to correct the impulse response.

After this correction, the echo-free impulse-response can now serve as the basis for various analyses, such as those shown in **Fig. 4**:

- Step response: `hs = mataa_IR_to_SR(h,96000);`
- Frequency response (anechoic) smoothed to 1/24 octave bands (magnitude and phase): `[mag,phase,f] = mataa_IR_to_FR(h,96000,1/24);`
- Cumulative spectral decay diagram (waterfall plot) with 30 lines covering the spectral decay of the impulse response during the full anechoic range (3 ms): `T = linspace(0,0.003,30); [mag,f,T] = mataa_IR_to_CSD(h,96000,T,1/24);`

## AMPLIFIER DISTORTION

**Figure 5** shows the measurement setup to analyze the harmonic distortion of an amplifier. The amplifier output is connected to a load resistor $R_L$ = 7.6Ω and a potentiometer $R_G$ = 4.7kΩ to attenuate the output signal of the amplifier (otherwise, the sound input would be overloaded due to the gain of the amplifier). Again, I chose these values because that's what came out of my box.

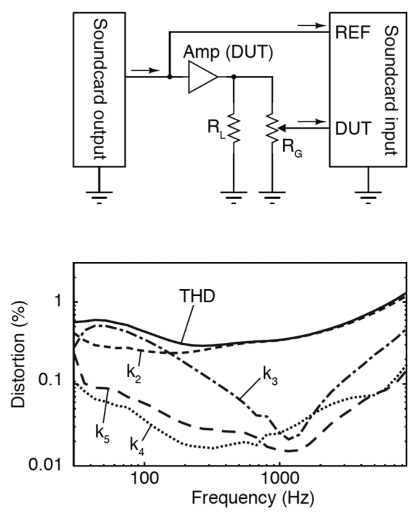



**FIGURE 5: Distortion analysis. Top: measurement setup (RL: load resistor, RG: potentiometer to attenuate and adjust the signal level for the sound input). Bottom: result of distortion measurement for my Quad II tube amplifier.**

MATAA provides the `mataa_measure_HD(f1,T,fs,N)` program to measure the harmonic distortion products and total harmonic distortion (THD) at a given base frequency. This command feeds a sine signal to the DUT (**f1** is the frequency of the sine, **T** is its length, **fs** is the sampling rate, **N** is the number of harmonics to be included in the analysis). The program then determines the Fourier spectra of the resulting DUT and REF signals at the soundcard input. These spectra are then normalized such that the amplitude of the base frequency (**f1**) is $k_1$ = 100%.

To correct for possible distortion artifacts introduced by the soundcard, the REF spectrum is subtracted from the DUT spectrum, yielding in the "true" distortion spectrum of the DUT. Finally, the program returns the amplitudes $k_2, k_3, \ldots, k_N$ of the harmonics and the THD, which is defined here as the geometric sum of the harmonics:

$$\mathrm{THD} = \sqrt{k_2^2 + k_3^2 + \ldots + k_N^2}$$

(note that other definitions exist).

In the current example, I demonstrate how to measure THD and the first four harmonics ($k_2, k_3, \ldots, k_5$) as a function of frequency. To this end, the distortion analysis is repeated at 32 different frequencies in the range of 30–9600Hz. I accomplished this by putting the following commands into a MATLAB/Octave script file:

```
Nf = 32; Nk = 5; fs=96000;
f = logspace(log10(30),log10(9600),Nf); thd
= repmat(NaN,1,Nf);
k = repmat(NaN,Nk,Nf); for i=1:Nf
[thd(i),k(:,i)] = mataa_measure_
HD(f(i),0.1,96000,Nk); end
```

On the first line, the number of frequencies at which distortion is to be analyzed (**Nf**) is set to 32, the number of harmonics to be analyzed (**Nk**) is set to 5, and the sampling rate (**fs**) is set to 96kHz. On the second line, a list (**f**) of the frequency values within the range of 30–9600Hz is produced (the frequeny values are distributed logarithmically). On the third line, the variables to hold the THD data (**thd**) and the distortion harmonics (**k**) are initialized with empty values **(NaN).** The last three lines constitute a loop in which the THD and distortion harmonics are analyzed repeatedly at the increasing frequencies in **f**.

I used this MATLAB/Octave script to analyze the distortion of my Quad II tube amplifier. I set the output power to 13.9W (at higher power, clipping becomes visible on the scope). The result of the distortion analysis is shown in **Fig. 5**. Note that the resulting THD is nicely in line with the specified[6] THD value of ~0.2% at 700Hz and 12W output.

## CLOSING REMARKS

The versatility of MATAA and the lack of a graphical user interface may (wrongly) suggest that operation is cumbersome and less streamlined than with other computer-based audio analyzers. In contrast, these peculiarities are one of the reasons I prefer MATAA over other audio analysis systems. On the one hand, you can have MATAA conduct virtually any analysis you can think of, while on the other hand, you can save the necessary steps for a specific analysis to a MATLAB/Octave script file so that you can repeat the analysis by simply running this script. In addition, MATAA already includes several scripts that automatically conduct some typical analyses.

MATAA therefore offers "plug-and-play" for these applications and allows new users to become familiar with MATAA. Finally, from my own experience, I have the impression that MATAA forces the user to think first about how to conduct the analysis best. This ultimately leads to a better understanding of the analysis and therefore improves the quality of the results and their interpretation.

I would like to stress that you shouldn't abuse your soundcard. Shortening the soundcard output, for example, usually does not destroy anything, but too much voltage at the soundcard input will. While a few volts won't do any harm, I have learned the hard way that a capacitor with a maximum voltage of 100V is not suitable to protect the soundcard from a 400V B+ voltage in a tube amplifier (I tried to analyze the ripple voltage of the B+). The best way to protect your soundcard is to think twice before applying a signal to it.

Also, inserting a voltage limiter just before the soundcard input may be a good idea. For instance, grounding the soundcard input through two series-connected 5V zener diodes with reversed polarity will limit the signal to ±5V (but the zeners will also blow up if too much voltage is applied for too long). Finally, if you are paranoid, you can minimize the risk of destroying the entire motherboard of your computer by using an external soundcard connected to your computer via the USB or FireWire port, for instance.

MATAA works fine for me, but I am sure other users will find a few rough spots that I am not yet aware of. Also, there are applications that are not (yet) covered by MATAA. Instead of trying to think of such applications myself and writing suitable programs, I decided to let other users request, suggest, or even write new code to improve and expand on the functionality of MATAA. I hope this results in a broad and ever-growing range of applications that are supported by MATAA "out of the box." So get in touch with me and other MATAA users through the MATAA homepage www.audioroot.net/mataa.html if you need (or want) MATAA to do something you don't know how to accomplish, or if you've written code that expands the functionality of MATAA. *aX*

## REFERENCES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2. The Mathworks, Inc. www.mathworks.com.
3. Free Software Foundation (FSF). www.gnu.org.
4. PortAudio. PortAudio—an open-source cross-platform audio API. www.portaudio.com.
5. Joseph D'Appolito, *Testing Loudspeakers*, Old Colony Sound Lab, 888-924-9465, custserv@audioXpress.com.
6. *Instruction Book—QUAD 22 Control Unit and QUAD II Power Amplifier.* Quad, 1964.

## CORRECTION CORRECTION

The schematic of my "Low-Noise Measurement Preamp" was published with my stated corrections in the June issue (p. 50). However, this contains a new error: C3 is now shown backwards (compare with the original schematic in 4/07, p. 26). While it might appear that the negative end of C3 would connect to the junction of R4 and R5 (since these come from the −9V supply), the Q1 source terminals are at enough positive bias so that C3 (but not C4) should have its *positive* terminal connected to R4/R5, as in the original schematic.

Also, in the original article (April 2007), in Fig. 1A (p. 26), the switch in the bottom right corner is S5 (not S3). And, in Table 3 (p. 28), the maximum right voltage should be ±100mV DC.

*Dennis Colin*
*Gilmanton I.W., NH*

I believe there may be a drafting error in Dennis Colin's article "A Low-Noise Measurement Preamp," April '07, pp. 26-33. The gain stated for the first composite stage is 40dB. With R8 at 1.5kΩ as shown, even without any negative feedback, I calculate a gain of about 42, or about 32.5dB. Feedback as shown will only make this smaller. Perhaps the resistor is actually 1.5MΩ.

Also, once the stage has sufficient loop gain, the JFET gate-source capacitance will be bootstrapped by the series feedback and greatly reduced in the midband—to what extent depends on how closely the sources follow the gates. Thus the input capacitance could be dominated by the drain-gate Cs (their contribution, absent Miller multiplication as the author points out, about 11pF based on the datasheet) and a small contribution from traces and the 1N4148 input clamping diodes. I noticed as well that the number stated in the text of 33pF doesn't match the marker pen inscription on the box in Photo 4 of 72pF.

*Brad Wood*
*Canoga Park, Calif.*

**Dennis Colin responds:**

You're absolutely right: R8 should be 51.1k. The 33pF input C is correct; the 72pF box marking was from an earlier design not having feedback to the JFET sources. The feedback bootstrapping thus decreased CIN by 39pF. I measured the 1N4148 diodes to be 6pF each at 0V DC.

I'd like to mention something about the LSK389B JFET: Out of six samples, four had a 1kHz noise density between 0.98 and 1.15 nv/√Hz, but the other two were 2.8 and 3.0nV/√Hz. The datasheet says 0.9 typical, 1.9 maximum at 2mA, 10V. In the preamp, ID is 3.8mA each, which should lower the noise somewhat.

The JFET I used by itself measured about 0.97nV/√Hz, 1-20kHz averaged, at 3.8mA. The two sections paralleled would then be 0.686. RMS adding the 0.406nV from the 10Ω resistor (R4) would result in 0.80nV/√Hz. The overall preamp input noise voltage density measured 0.972nV/√Hz, 1-20kHz averaged, 1.7dB higher than calculated. This is probably from the noise contributions of the drain resistor and other components, or Murphy's Noise Law: Measured noise is always greater than calculations!

Regarding the 2 of 6 out-of-spec JFETs, and considering that Linear Systems' ad (p. 53 of the April '07 issue) claims "1nV Low Noise Dual JFET," I would request units meeting that claim. Thank you for noticing the error.

## FLAT PANEL SPEAKER

I was very interested in Daisuke Koya's article on the flat panel speaker (Nov. '06, p. 34). He said he obtained your SoundPad exciter from a Target store. Can you give more details so I can purchase one as well before ordering more from Si-5 technologies? I am using the exciter to convert a violin to a speaker for violin music.

*yy shum*
*shumyy@yahoo.com*

**Daisuke Koya responds:**

Thank you for your interest in my article. Upon checking product availability at Target.com, I found that the SoundPad is only available from their online store. I'm unsure about converting a violin to a distributed-mode loudspeaker, because the density of wood is not consistent as synthetic materials, and its shape is complex. Experimentation is encouraged, however, and perhaps with trial-and-error exciter placement, you can find an optimal exciter location.

## CABLES

Having followed the cable discussion now for years, with no proof to my ears and brain or to any ears/brains used in any blind tests that cables are audibly influencing the sound, I take issue with your cable review by Mr. Milford in your March issue ("PNF Audio Cables," p. 54).

One more waste of space like this in a magazine that takes itself seriously—and I am gone. . . gone to cancel my subscription.

*Peter M. Moritz*
*paw2004@xplornet.com*

## MORE TRANSFORMER MAGIC

Neal Haight wrote an interesting letter in the April issue of *audioXpress* (p. 54), in which he suggested a different output transformer for my little amp, "Mighty Mouse," (April '06). He retracted the math I used to calculate the proper turns ratio and suggested an alternative hookup of the Hammond P-T1609.

I had some concern because Neal used the transformer in a manner different than the manufacturer intended. Of course, I also used the original output transformer differently, but I was concerned that in Neal's hookup, most of the transformer primary remained unused. The transformers were also wound differently, my transformer being of the superior toroidal construction.

On the other hand, Neal's transformer probably had thinner laminations of superior materials. On paper, the specs of the two transformers appeared similar, so there was nothing for me to do but spend the long green and order a pair of Hammonds. I have done so, and can report that these transformers are superior, but at two to three times the cost.

The most striking difference is in the highs. The Hammond transformers have improved detail and depth, while the original Avel-Lindberg transformers have greater bass extension and authority. Lateral imaging is slightly worse in the Hammonds, but that might be due to the clip leads I used to insert them into the circuit.

The personality of the amps has changed: While the original transformers made the amp sound slightly dark, the Hammonds make the amp sound a little bright. Overall, the change is positive, and for a slightly dull-sounding system, definitely worth the cost, but for a bright-sounding system, the Avel-Lindberg transformers may be preferable.

I thank Neal for his interest, and welcome other readers' comments. You may contact me at musitronix@yahoo.com.

*Bob McIntyre*
*Toledo, OH*

## ARTICLE WANTED

I'd like to see an article on a power regenerator, which is a device that reproduces a low-wattage 60Hz voltage that is independent of the AC line. Such examples include the now famous PS brand power regenerator with its rather sophisticated variable ±60Hz sine wave or variation thereof. A more recent edition is Lloyd Walker's power regenerator for his turntable. There are a few others meant for low-wattage components. Power amps need not apply because they have their own rather beefy power supplies.

Recently in an Audio Asylum post, someone used a Walker turntable power regenerator on his CD player, DAC and/or transport and reported a night and day improvement! I believe this instrument is simply a power amplifier with a built-in 60Hz oscillator and an output transformer to boost the voltage back to 120V. It doesn't need to have all the bells and whistles that the $1000 units have. Perhaps your OMP/MF MOSFET amp or a subwoofer "plate" could be the basis of the unit. I'd like to keep the cost around $300.

*John Kwinn*
*jkwinn@aol.com*

[In the May '07 issue, we misspelled the name of one of our authors. We regret the error, and extend our apologies to **Bill Elliott**--Eds.] ***aX***

# Hi-Fi Critic(al)?


There's a new bi-monthly magazine out of London that claims to be the hi-fi version of the Seven Samurai—out to protect us, for a fee, from poorly designed equipment and bad information.

**Hi-Fi Critic** (www.HiFiCritic.com) is the creation of long-time freelance audio journalists Martin Colloms (publisher) and Paul Messenger (editor). This magazine, which claims to provide "about 40,000 words of original editorial content per issue," takes no advertising, and costs US subscribers $147/six-issues/year (discounts for multi-year sign-up) airmailed to their homes. The print edition is complemented by website addenda.

Contributors on equipment include Chris Binns, Chris Breunig, Martin Colloms, Alvin Gold, Paul Messenger, and Malcolm Steward. There also are music reviews of classical, pop, and jazz recordings.

From the website: "Although British audio journalists have built a fine reputation for high quality critical assessment of hi-fi equipment, current reviews seem to have become increasingly 'soft' and lacking in serious critical insight. . . Relentlessly positive 'rubber stamp' reviews simply devalue the long term credibility of the whole reviewing process. . . We've therefore decided to take a stand, and use our experience and critical faculties to restore credibility to the reviewing process, while also addressing the serious issues facing hi-fi today."

Their focus is "classic stereo systems: preamplifiers, power amplifiers, tuners, LP vinyl and CD players, turntables, tone arms, and cartridges, plus phono equalizers. Integrated amplifiers and designs with still more functionality will be tested. Loudspeakers will figure strongly in the reporting, not forgetting audio cables, line conditioners, equipment supports and the like. . . [Lab test results will emphasize] power distortion, load matching and input matching, plus investigative work to try and improve the correlation between lab data and sound quality."

Colloms will use his subjective sound quality rating scheme. "Some 25 years ago I used an IEC-based scale from 0 to 10, when zero represented nil fidelity, 10 was assigned to essentially perfect reproduction, and given a sufficiently large and representative group of product, a score of 5 represented the group average for sound quality. Such results were self limiting and self sufficient. Maintain-

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ing a historic standard based on such results proved difficult as equipment continued to improve and the resulting scores edged upwards inexorably towards 8 and 9."

He opened up the top end, with the concept that a doubling of the score relative to a reference piece of equipment should represent a perception of twice the sound quality. The top score reached to date is 120, by "the Krell Evolution ONE and TWO pre-power amplifiers, separately and in combination." More detail about the ratings scheme is at www.HiFiCritic.com/colloms/ratings.aspx. His amplifier ratings archive does not include a McIntosh tube amp.

## THE WEBSITE

- The Archive section provides access to 27 articles in *.pdf format, almost all written by Colloms, either solo or with a co-author, during the 1970s-1990s.
- The Music Room: "Renowned recording engineer Tony Faulkner will supervise the release of non-compressed music excerpts for download. These will illustrate aspects of recording and production techniques and the effect on sound quality, or relate to topics to be published in forthcoming issues of *Hi-Fi Critic*. The download should then be formatted to a standard CD-R for replay on your audio system."
- The Audio Scene provides reports on audio shows covered by their writers.

## ISSUE 1

Malcolm Steward explains how to get CD-quality downloads, Chris Bryant looks at the perceived degradation of CD sound quality over its quarter-century life, Tony Faulkner wonders whether recording practices are getting better or worse, Martin Colloms provides a limitedly scientific analysis of cable performance characteristics, plus reviews of speakers, CD players, amps, and recordings.

Steward presents a fair appraisal of downloading. For higher-quality downloads, he recommends FLAC (Free Lossless Audio Codec: http://FLAC.SourceForge.net), which "doesn't trouble itself with . . . any DRM" and is being used on an increasing number of music sites as an alternative to MP3, although it generates substantially larger files. For FLAC-encoded music, he points to www.AllofMP3.com, a commercial Russian site with "a vast range of recordings." A FLAC codec is available for Windows Media Player (the latest stable version is 0.71.0946 from www.illiminable.com/ogg/downloads.html). Open-source players such as Cog are available at http://CogOSX.SourceForge.net. Steward also suggests that FLAC is a good choice to back up your CDs, because it "provides an exact duplicate of the original," and points to websites offering software to convert FLAC files to regular CD format.

Chris Bryant declares that "now that the very best of [today's CD players] can produce sounds as good as vinyl, we can finally enjoy our silver disc collection. It's therefore exasperating, if not totally unexpected, to discover that many of the CDs themselves are mediocre. The surprise comes when you discover that the older discs in your collection are often the ones you enjoy the most. . . The best of the 1980s CDs sound far more like vinyl than those made later. . . The most disliked CDs tended to be digital re-masters of earlier recordings." He is not categorically against digital audio technology.

Bryant admits that technically, today's equipment performs better than earlier designs, and finds that new CD players sound "better and cleaner." Bryant believes that extensive editing, which has become much easier and more powerful than when you had to cut-and-splice tape, "is the biggest reason why many—if not most—modern classical CDs sound anemic and un-involving. They have been hacked to shreds in a workstation in pursuit of some mummy-knows-best horizontal and vertical perfection. . . Many productions now are made using Pro Tools multi-track as a matter of course, and I believe this is a further reason why modern CDs sound less convincing than earlier ones. The more processing. . . the greater the scope for the intervening technology to dull the musical effect and energy." Bryant blames technology for the degradation of newer versus older CD sound quality.

Martin Colloms looks into the physics and sonic effects of cables. "My personal experience with hundreds of different cable types has shown that the judicious selection of cables is crucial in achieving a finely tuned and well-balanced system." He acknowledges equipment's susceptibility to EMI (electromagnetic interference), claiming the effects in current equipment are generally subtle. He views every cable as an antenna that can carry spurious signals into equipment, with the magnitude of the effect dependent "more on the details of a cable's construction than its absolute cost."

Colloms acknowledges the interaction of amp-cables-speakers with his mention of the Grodinsky/Cornwell three-decade-old patent for an integrated system, but only addresses the EMS (electromagnetic compatibility) factor, and in his exploration of the potential problem makes judgments aurally. Among cable construction characteristics, he discusses insulation, physical strength, vibration damping, metals, supercooling during manufacture, and others. He claims that he "(and others) have little trouble in differentiating between grades of solder used for interconnect cable connections." Colloms warns readers of possible fraudulent manufacturers and overcharging, and clearly believes cable selection is an important factor in system sound quality.

In Colloms' review of the Krell EVO-202 two-box preamp, which includes extensive detail on the circuit design, he notes that the short interconnect cable between the power supply and the processor keeps them close, raising the concern of induced hum, ameliorated by the "shielding plate [to keep] induced hum out of the preamp above." He does not raise the same concern over the separately packaged phono preamp stages that are connected via "single-ended unbalanced" RCA plugs. His descriptions use terms that require study: "overall sound has a remarkably vibrant energy"; "explosive bass has a lean muscularity"; "transient edges were sharpened"; "sound is moderately upbeat, with quite good rhythm and good timing, showing decent synchronicity over the whole frequency range." He credits an immediate improvement in "midrange clarity, marginally improved timing and greater soundstage depth" when he inadvertently left a magazine on top of the preamp.

Messenger's selection of his favorite speakers in several price categories is the most rational review in the magazine. What's missing is a description of speaker placement and listening room acoustics, plus the speakers' dispersion characteristics. At least he is not averse to different design philosophies, including both direct radiator and horn-loaded designs.

Alvin Gold's analysis of three CD players that he calls mid-priced (about $1700 srp) yields the only source of substantial negativity: "The three players here are not necessarily better than older, less-celebrated designs. There was really nothing that these players did on test that couldn't have been replicated a long time ago, perhaps at much lower prices and from less-revered names than these."

Ken Kessler's tribute to Alastair Robertson-Aikman (ARA), known for SME (originally the Scale Model Equipment Company Ltd.) products, was respectfully laudatory, providing ARA's and SME's histories, how important music was to ARA, and his efforts to provide products and knowledge, especially the need for a "no-compromise room" that could bring

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"superior sound" to his "fellow music lovers." ARA's passing "is the loss of one of very few remaining audio pioneers from the Golden Age still active in the business a half-century after gramophones turned into record decks, and needles became styli." Kessler's respect for, and friendship with, ARA is quite clear in this touching homage.

Christopher Breunig's reviews of recent and re-issued-historic classical recordings are thoughtful, anecdotal, informative, and not all positive. He compares these with other recordings, explaining why he prefers other performances, and reports sound quality and recording anomalies, for example noting that "the tympanist is as suddenly to the fore in the development section of (i) as the tenor's solo in (iv) dips into submersion:" and "sound quality is notably variable, with some over-separation in the balancing. Less excusable is the precipitate editing of ambience at movement endings."

Nigel Finn's reviews of rock, pop, and other nice music are equally useful.

## ISSUE 2 (FROM THE TABLE OF CONTENTS ON THE WEBSITE)

"Martin Colloms continues his magnum opus on cables, examining directionality, bi-wiring, and reviewing 16 different speaker cables."

Paul Messenger tries out exotic speaker cables.

A variety of classical and other music recordings reviewed.

## ASSESSMENT

So far there is a high proportion of tube equipment reviewed.

A telling comment is from Colloms' "Subjective Sounds" (issue 1): "As the rest of [my hi-fi] system downstream from the sources is incrementally improved, so the advantages of analogue seem to become that much more obvious." He praises his upgrade to the Magnum Dynalab 106T FM tuner as providing "a firmer, more solid soundstage and a quite obviously much sweeter, more delicate and transparent top end than [his previous tuner]."

The equipment reviews provide history to each model's place in its manufacturer's line, and delve deep into design and construction characteristics and engineering choices, even to identifying the OEM of subsystems such as CD transports and the Chinese IAG factory that, according to Colloms, now makes Quad speakers.

Colloms' speaker reviews go into each manufacturer's design theory without expressing his preference, discussing materials, dispersion (in some reviews), the designer's goals, and so on. His extensive description of the Quad 2805/2905 designs is a tutorial on electrostatic speaker science, although I wonder what must be going on in the speaker to justify his claim that turning off the speakers' lighted logos "removed a certain colouration."

I agree that we have yet to quantify our hearing to the point of being able to completely correlate measured performance with subjective perceptions, but am challenged, beyond my ability, to accept many of the claims made regarding cause and effect. For example, it is well known that as more time is spent listening to a system, or attending concerts in a specific concert hall, our hearing acclimates to its sonic/acoustic idiosyncrasies, accepting some characteristics and revealing others. It often leads to the sound becoming normal/preferred, generating a new psychoacoustic reference.

I would give *Hi-Fi Critic* more credibility if they hadn't praised most of the reviewed equipment so highly, including only ever-so-slightly negative comments to apparently rationalize a differentiation among price classes. Substantial negativity was found only in the four reviews of the Arcam Solo, and most of it was aimed at Britain's DAB (digital audio broadcasting, also known as Eureka 147) system.

Messenger, Colloms, et al. aren't Luddites. They report, without deprecation, the use of computer analysis and high-tech materials in speaker designs. They accept the evolution to digital technology, although they seem to long for the sound of analog. My biggest problem is with their claim that it's better, instead of expressing it as their sonic preference.

*Hi-Fi Critic* is a UK magazine that seems aimed at the British audiophile sitting by the fire in his study sipping an expensive red wine, deciding what to buy based on high-enough price and mellifluous audiophile verbosity. *Hi-Fi Critic* is informative, but will find it difficult to succeed with no advertising and such a high subscription price. *aX*

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# Ad Index



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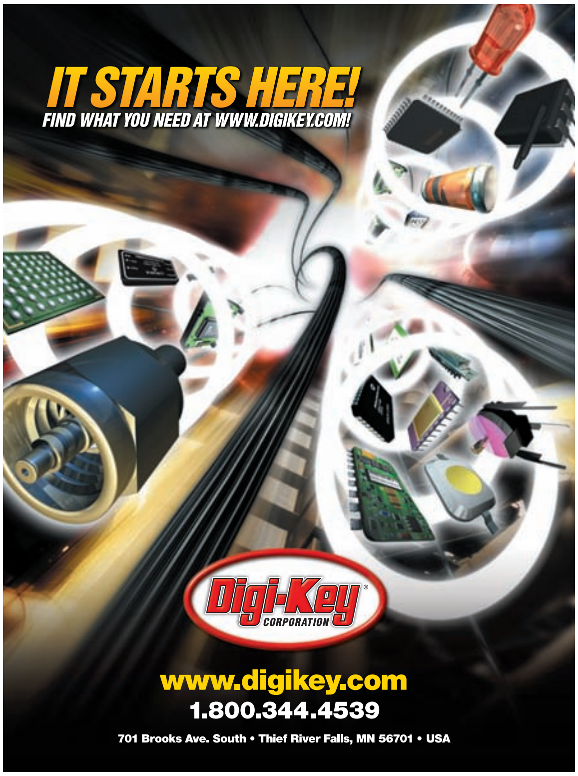
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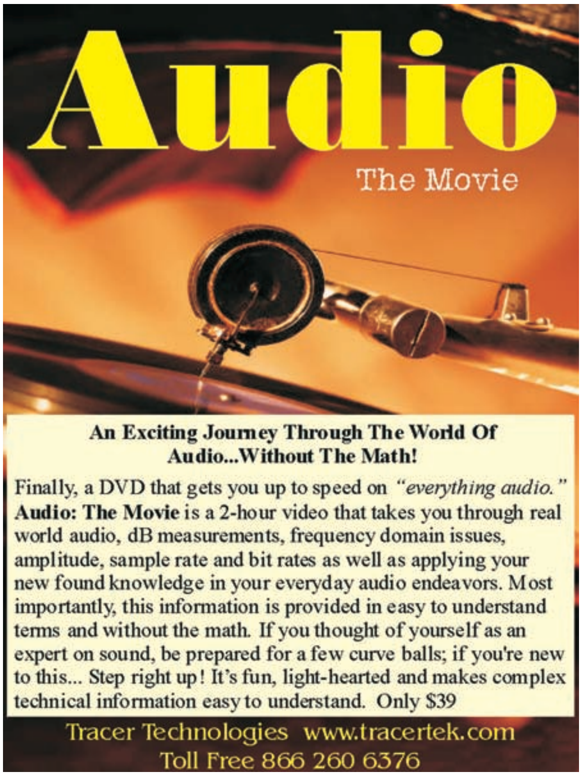
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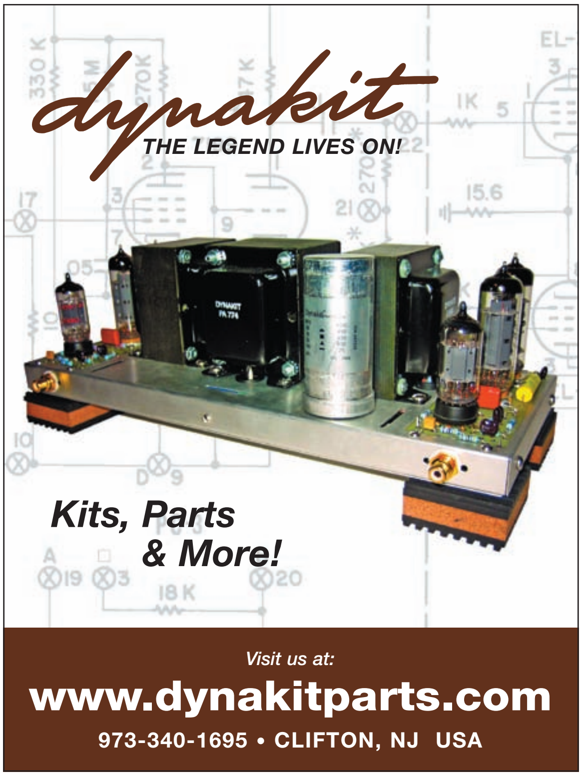


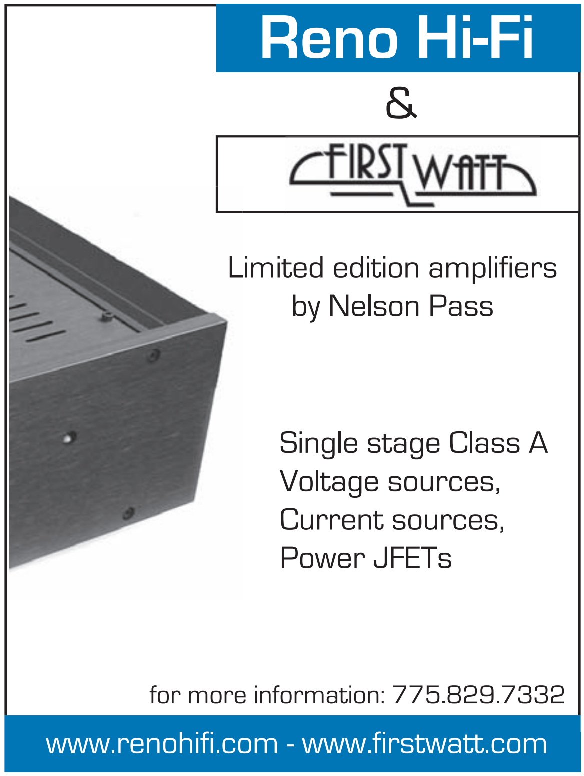



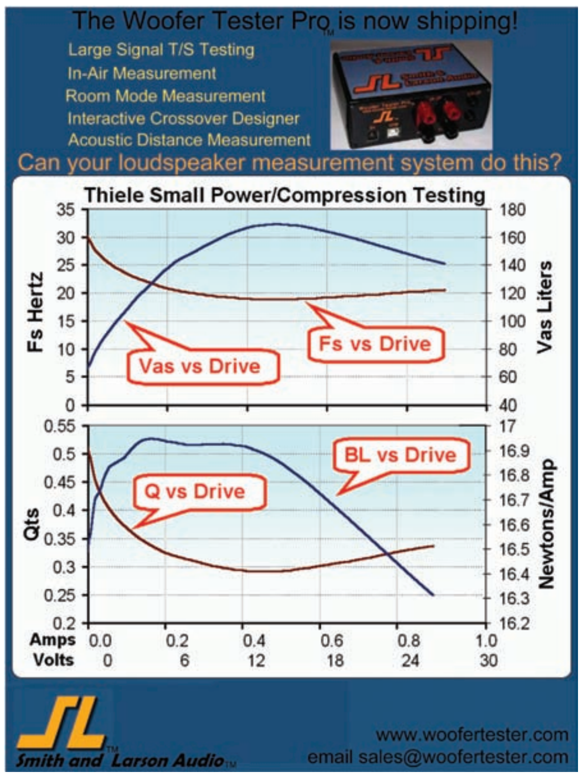

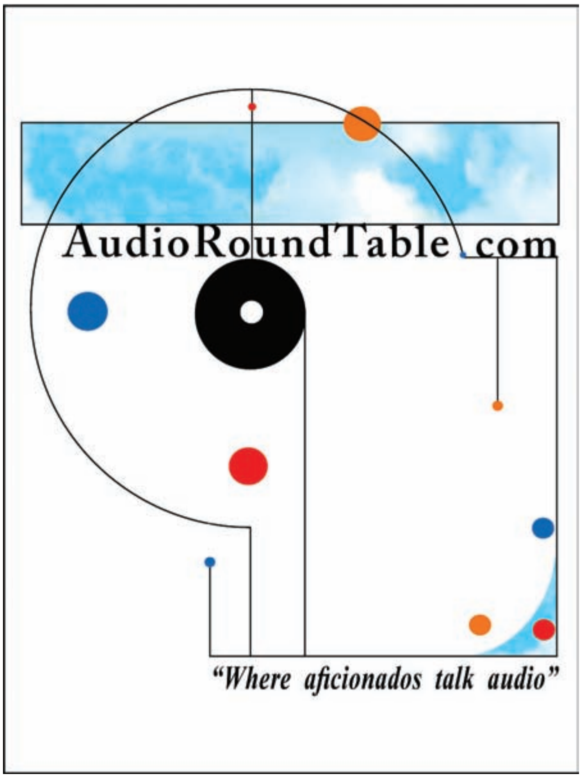
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