Back

A Balanced Nonoversampling Retro DAC

This high-end DAC design is a perfect blend of both contemporary digital technology and old vacuum tube and transformer techniques.

By Yury Gelgor

PHOTO 1: The completed DAC.

Why should audiophiles pay attention to digital analog converters (DACs)? Well, in my case, when cheap hi-fi VCRs sound better than three times more expensive CD/DVD/SACD players, it is a good time to take note. And this was after I had already made a pretty good amplifier with speakers and could still hear a difference.

This article shows how to construct a DAC that is an upgrade of a previous one I have posted on my website www.geocities.com/yury_g. However, this article is not just about the upgrade; it is about the whole RetroDAC (**Photo 1**). This DAC is not a cheap upgrade of a CD player, but a pretty expensive unit that stays below my reasonable limit just because I did not use silver-wired transformers, exotic capacitors, and other fancy components—with one exception.

WHY BALANCED?

The 1950s–1970s were an era of audio reproduction when a very high standard was set in the studio and nobody knew about good audiophile interconnect cables. Simple twisted wires could deliver that quality. How so?

The answer is simple: 600Ω balanced transformer outputs. This schema has many advantages such as ground loop breaking, reduced hum, and natural low output impedance.

The first disadvantage of such a solution is the demand for a high-quality output transformer, which comes with a high price, of course. Some famous microchip producers make ICs of "transformer-like" balanced line transmitters/receivers; however, the quality of transformer schema remains beyond compare.

Another problem with the transformer is DC offset, which is simple to avoid by using a push-pull amplifier that reduces even distortions. You need a phase splitter to feed push-pull schema. I think the best way to build this phase splitter is to use additional DAC chips in each channel to produce two signals with 180° phase shift. I will explain this presently.

WHY NONOVERSAMPLING?

Let me remind you first what sampling is. Studio recording devices measure the voltage of audio signals 44,100 times per second and record those numbers on CD. On the listening side, a CD player reads recorded numbers and sends them to the DAC for conversion into voltage. Those numbers that appear 44,100 times per second are samples, and only they represent true values of the signal in some moments of time. No true information is available between the moments when the samples were taken.

Oversampling is an attempt to calculate those signal values using a sine approach to shape. It is probably good to measure single signal distortions; however, it is not correct at all. Nobody can calculate music, which is obviously not a sine signal.

Another reason for not using oversampling: the higher the oversampling, the higher the jitter level. It means that two times oversampling DAC has two times higher jitter than a nonoversampling one with the same CD/DVD mechanism (transport).

THE DIGITAL PART: KEEP IT SIMPLE!

A transport processor reads audio information from CD/DVD and sends it as a bit stream to digital output. The protocol (the way of bit stream coding) is called S/PDIF (Sony/Philips Digital Interface Format). Digital output of the transport could be organized in one of two ways: coaxial or optical. Optical is called Toslink (Toshiba optical link). I prefer coaxial connections between the transport and DAC.

As a true hi-end lover, I am sure that the simpler the schema the better. I also think that the best resistor is no resistor at all, the best capacitor is no capacitor, and so on. So you know that when I use a part in this DAC it is really necessary and essential.

Therefore, the first task of the digital part is to receive this S/PDIF signal from the transport, synchronize to it, recognize the audio format, and convert it to signals the DAC chip can "understand." IC1 takes care of all that. It recognizes regular CD 44.1kHz format, practically all other professional formats, and most DVD-audio formats.

Figure 1 shows the simplified digital part of the DAC. IC1 produces three important signals:

- SCK-clock signal—For a regular CD it has a frequency of 44.1kHz * 64 = 2822.4kHz
- SDATA—Data for recording to input register of the DAC and the proportional output signal.
- FSYNC—High/low level here shows that data on SDATA output is for the left/right channel, respectively.

IC3 and IC4 are precision twin DAC chips. Each has the following inputs: D+, D– (data inputs for no inverted and inverted shoulders), L+, L– (latches for recording of data into register), and CLK (input clock).

It takes one inverter IC2B to separate information for the right channel. It also takes another inverter IC2A to get inverted data for both. This is it, no more parts. The very smart receiver IC1 makes everything that's needed to bring necessary signals to the set of precision DACs IC3 and IC4 and only two inverters to separate the bit stream for the left and right channels and organize completely balanced schema.

DIGITAL PART 2: UNDER FIRE

Since I put the previous version of DAC on my website, I have received several e-mails from folks who criticized the digital part of my schema. Two subjects under fire were phase shift and code for inversed shoulder:

1. Phase Shift. At the recording, both channels' samples are made at the same time. CS8414 has serial outputs: the first goes out the left channel, then the right channel with 32 clock-beatings delay (signal FSYNK), which is T=1/(2 * Fsmp) = 1/(2 * 44100)= 11.34 microseconds, where Fsmp is the sample frequency.

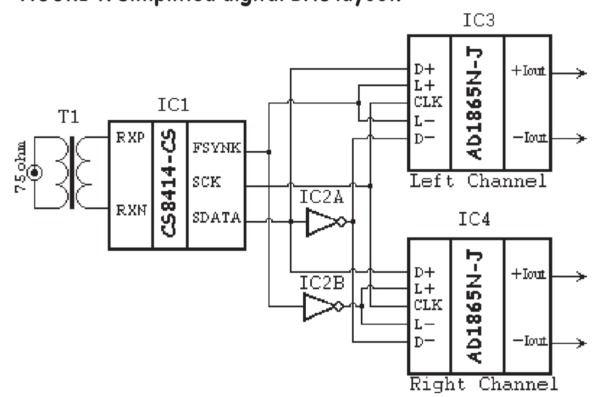
It produces the phase shift between the channels = 360 * F * T: 360 * 1000 * 0.00001134 = 4.08° at 1kHz and 360 * 20000 * 0.00001134 = 81.7° at 20kHz.

It is definitely not significant at 1kHz and hard to say whether it is possible to hear such phase shift at 20kHz. Maybe somewhere in the middle? When using headphones? I do not think so, but I believe that 32-bit shift register, which needed to correct this problem, can bring more digital noise to the power supply that caused more jitter. So I think the effect of using such a register will be negative.

2. Inversed code versus complementary code for inversed shoulder. Forming data for an inversed shoulder DAC strictly with the inverter is not correct: to get a correct 16-bit complementary code you must add one after the inversion of code; otherwise, you always have one bit misbalanced. It is the same linear signal as in "positive" shoulder, but 180° inverted and one bit shifted to the negative side. For example, when no signal is applied, the code for "positive shoulder" is 0000,0000,0000,0000 and for "negative shoulder" is 1111,1111,1111,1111. This is exactly one bit misbalanced: just add one to the negative shoulder code and you get "all zeroes." There is any signal or no signal at all.

The maximum output current for the DAC AD1865 is 1mA, which obviously happens when the signal is maximal. So this misbalance is 1mA/8192 bit =

FIGURE 1: Simplified digital DAC layout.



0.00012mA. About a dozen chips schema is needed to correct this problem, and, of course, the twin triode has much bigger misbalance, so it's best to forget about this "one bit problem." For me, the dozen additional IC is the same reason to be afraid of jitter.

ANALOG PART: EVEN SIMPLER!

Figure 2 shows one channel schema of my DAC. The twin DAC is "pumping" the balanced signal into R3, called "I/V resistor," and T2 ("I/V transformer") primary where I/V conversion is made. T2 has a ratio of 1:6.4 + 6.4. That signal goes to the twin triode balanced schema, which is also "pumping" the primary of T4. The secondary of T3 is the 600Ω output of the DAC. This schema is very simple, but some parts of it deserve further comment.

First, consider the value of the I/V resistor. There are so many different opinions about it! Some folks reduce this value to 10Ω, others increase it to 1k. So what? Some say that Burr-Brown DACs work better with smaller values and 25–40 is fine. But any attempt to use values over 100Ω causes sound similar to signal clipping. Some also say Analog Devices DACs are more tolerant to high values of this resistor, which could be up to 600Ω.

So I changed the value of this resistor to listen to the difference. Obviously, the higher its value, the bigger the output signal. And to my ear, values less than 100Ω sound worse: drums, explosions, all high energy short sounds play better within 200–400Ω rather than 25-50. The low value of that resistor suppresses dynamics of sound.

However, the higher the value of this resistor, the higher the inductance of the T2 primary must be: for 25Ω it is fine at 0.5H, for 250Ω it is 5H. So the primary inductance of the transformer sets the limit of this resistor.

Now consider the high-frequency response of transformer T2. Every transformer has high-frequency (HF) resonance (**Fig. 3**, F2). The higher the frequency of this resonance, the better this transformer is for normal amplifiers, but things are a little bit different for the DAC.

Figure 4 shows a signal waveform at the I/V resistor. Instead of a smooth wave of signal, one step follows another. This is the way it works: the DAC chip converts incoming code into analog and keeps its value until the next one. Each code (sample) follows another with a frequency of 44.1kHz, and the output signal includes a large amount of this frequency. And yes, your guess is right: it harms the signal, and the I/V transformer must have HF resonance beyond that.

ANALOG PART 2: RLC FILTER VS. TRANSFORMER

An important role of the transformer is to be a filter for reducing the sample frequency level. A regular RLC filter just changes impedance and reflects incoming energy back to the source (DAC output). In turn, the source again reflects the rest of the signal to the filter. The signal becomes reflected many times, losing quality and sharpness exactly like a TV picture without a properly adjusted antenna.

Look again at **Fig. 3**. A correctly made transformer has a flat response at 20kHz (F1), HF resonance (F2) about 28–33kHz, and 44.1kHz is supposed to be on high-frequency rolloff (F3). A transformer such as this mostly turns that energy into heat and consumes it, which makes the signal's shape much clearer (**Fig. 5**).

POWER SUPPLY

Design of the power supply (**Fig. 6**) is as important as signal audio or the digital part. After many experiments, I think that if I need several voltages, it is better to use a separate transformer for each one, not just separate secondary windings. Furthermore, the best transformer is no transformer at all when I need about 150 DC voltage. That is why I use +170V and –170V one-wave rectifiers for tube output. Note: This is dangerous, so you should take some precautions to avoid electrocution.

If you have no high voltage experience or are not sure about the result, just use a 1:1 ratio transformer for tube plates. Under no circumstances should you use

FIGURE 2: Analog portion of DAC.

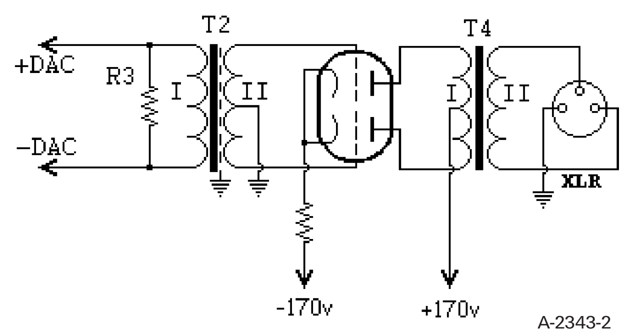


FIGURE 3: Typical frequency response of transformer.

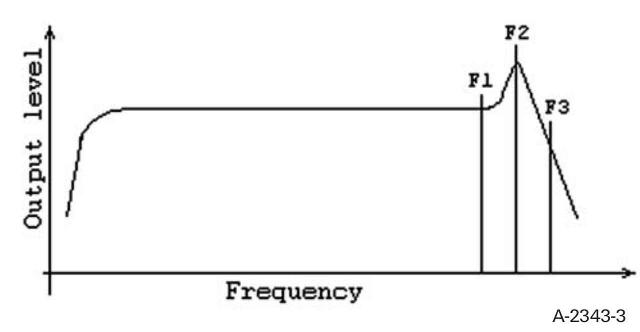


FIGURE 4: Waveform and samples.

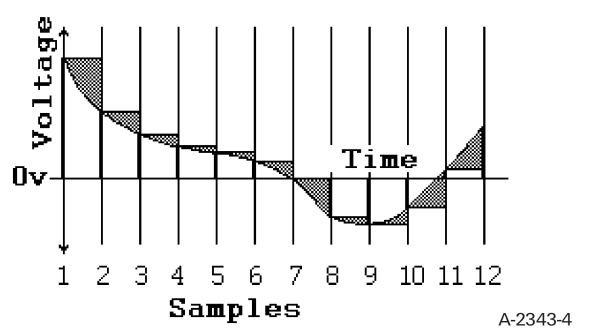
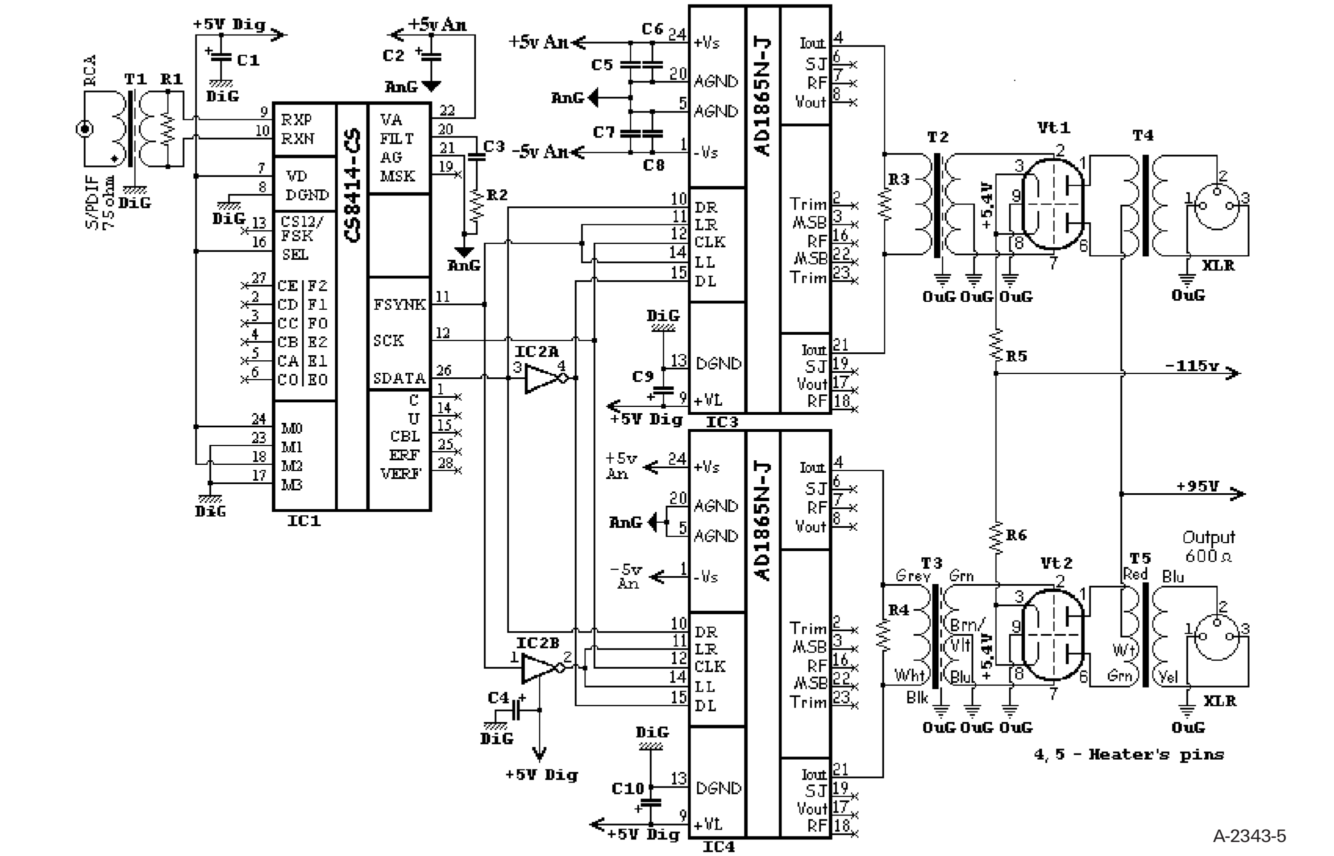


FIGURE 5: Signal (digital and analog).



this schema with no transformer, unless you're absolutely sure of your abilities and knowledge in this area. I take no responsibility for any accident resulting from this. You do this at your own risk.

A schema with "Alert" neon lamp shows when polarity is wrong, and "Polarity" switch allows you to change polarity. It is a good idea to use a special AC socket for this kind of schema. Working with such schema is simple:

1. Do not connect any interconnect cables for the first plug-in. Do not touch any metal part of the DAC.
2. Make sure the "Alert" lamp is not on. Use "Polarity" switch when it is on.
3. Turn the power off and connect all necessary cables.

The other sources of the power supply are pretty simple. Using ferrite shield beads for tube heaters is something that you may not see too often. They are used to reduce the chance of high-frequency parasitic oscillation.

I used three different ground circuits: Digital Ground (DiG), Analog Ground (AnG), and Output Ground (OuG). All of them were made absolutely indeon the bottom side of this frame to make it solid. So, mechanical design is also very simple!

You probably already found the BNC connector near the S/PDIF RCA input. It is not connected and is for future use only.

It is very important to make a good ground. See **Photo 4**. Output ground (OuG) is made of 8 gauge (about 3.5mm) solid copper wire. A ground star (where all grounds come together and connect to the metal chassis) is shown in the white circle.

TUNE-UP PROCEDURE

1. Remove IC1-IC4 from the sockets. Do not connect any interconnect cables. Remove all tubes.
2. Plug in. Make sure the "Alert" light is off. Flip "Polarity" switch to another position when "Alert" light is on.
3. Use potentiometer R119 to set output voltage of tube heater regulator to 6.3V on pins 4 and 5 of tube sockets Vt1 and Vt2.
4. Check out voltages on capacitors C111, C113, and C116. They must be +5V, +5V, and –5V, respectively.
5. Check out all necessary voltages and grounds on each chip according to schema. (For example, pins 8, 17, 21, and 23 must be 0V and 7, 16, 18, 22, and 24 must be +5V in socket of IC1. The chip is not inserted yet!)
6. Unplug unit, put in Vt101. Plug in. Make sure you have about –170V on pins 3 and 8 and +170V on pins 1 and 6 of tube sockets Vt1, Vt2.
7. Unplug unit, put in tubes Vt1 and Vt2. Plug in. Make sure you have about +5.4V on pins 3 and 8 and +93V on pins 1 and 6 of tube Vt1, Vt2.
8. Unplug Unit. Insert IC1–IC4. Do not forget to use Electro Static Discharge (ESD) bracelet.

PHOTO 3: IC1 on SOIC-to-DIP adapter.

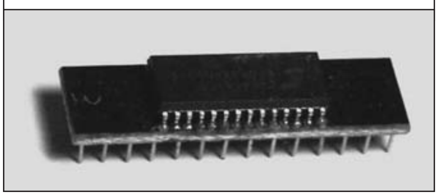


PHOTO 4: Output Ground (OuG).

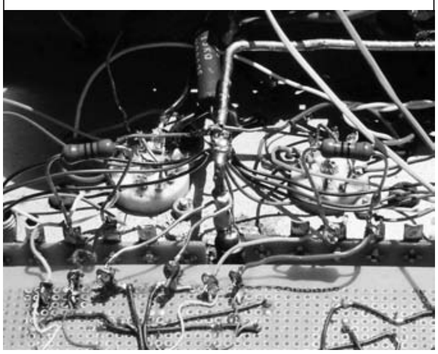
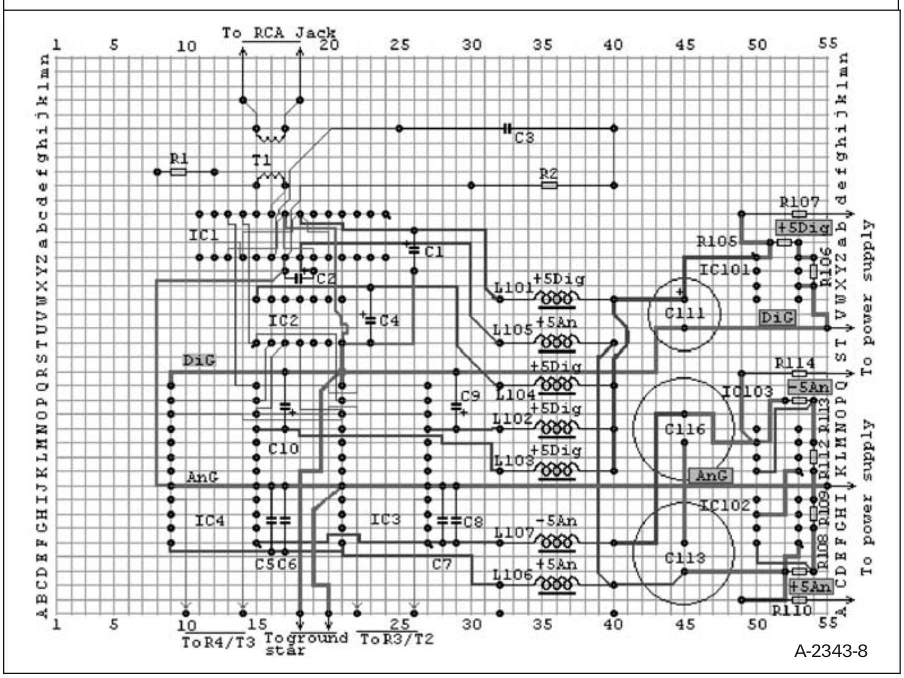


FIGURE 8: Digital board layout.



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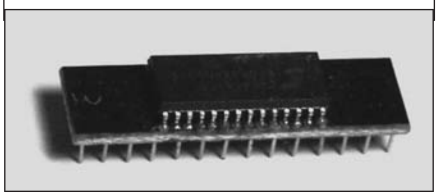

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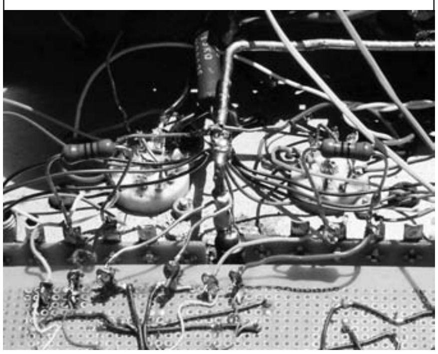

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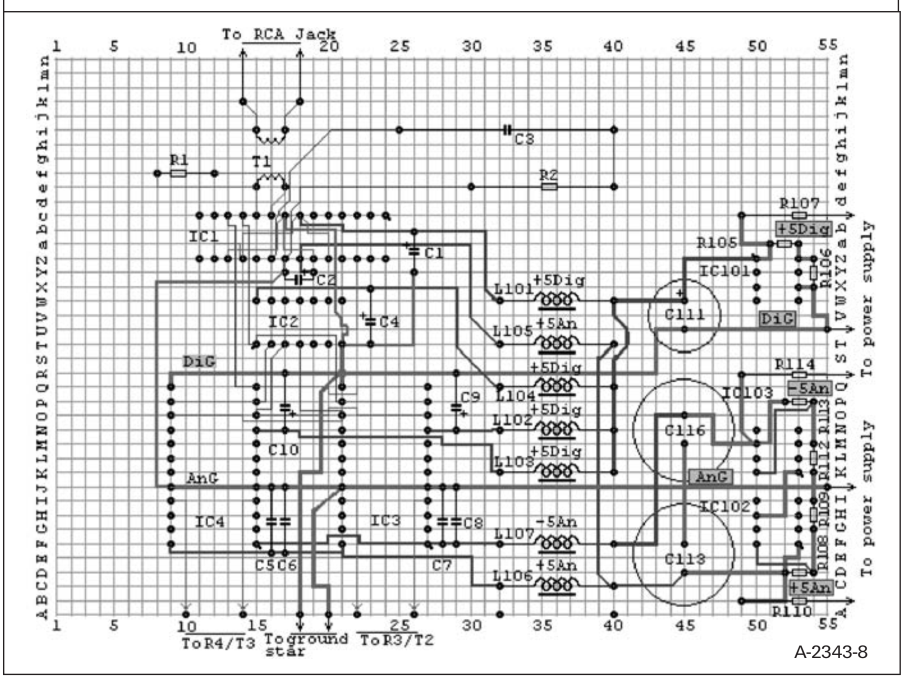

9. Plug in interconnect cables, plug unit in.
10. Enjoy! Do not forget to program your CD/DVD player according to its user manual. Many of them have S/PDIF coax output initially turned off.

UPGRADE

I did not change the schema of the DAC during the upgrade with one exception: I added Black Gate capacitors C102 and C107. All other changes are:

1. Recommended for CS8414 Scott SPDF transformer, I replaced with Audio Note TRANS-280. This is the only transformer with silver wiring.
2. I replaced Sowter 8347 T2 and T3 transformers with custom-made Sowter 9762.
3. I replaced resistors R1, R2, R108, R109, R112, R113 with Audio Note Tantalum.

TABLE 1: Parts List

COMPONENT	VALUE	MANUFACTURER	SUPPLIER
BR101	5A	Any diode bridge 5A or more	
C1, C2, C4, C9, C10	100µF 10V	Black Gate PK	Parts ConneXion
C3	68nF 100V	Multicap RTX	Parts ConneXion
C5, C7	22µF 6V3	Black Gate NX	Parts ConneXion
C6, C8, C101, C106	4n7 500V	Silver Mica	www.newark.com
C102, C107	68µF 350V	Black Gate NH	Parts ConneXion
C103-C105, C108-C110	150µF 350V	Black Gate VK	Parts ConneXion
C111	220µF 16V	Black Gate FK	Parts ConneXion
C112, C114, C115, C117, C118	1000µF 25V	Black Gate Std	Parts ConneXion
C113, C116	1000µF 25V	Black Gate NX	Parts ConneXion
C119	100µF 16V	Any	
C120	1µF 16V	Any	
C121	23000µF 16V	Mallory	www.newark.com
D101	NTE174	NTE (see note)	www.mouser.com
D102, D103	HFA08PB60	International Rectifier	www.newark.com
D104, D105, D106, D107	NTE174	NTE	www.mouser.com
F101, F104, F105, F106	0A25 250V	Any slow fuse	
F102	1A 250V	Any slow fuse	
F103, F107	0A5 250V	Any slow fuse	
L101-L107	47µH	Any RF coil	
LA101	Red Neon Assy	272-712	
LA102	Green Neon Assy	272-708	
R1	110Ω 0.5W	Audio Note Tantalum	Parts ConneXion
R2	470Ω 0.5W	Audio Note Tantalum	Parts ConneXion
R3, R4	220Ω 1W	Audio Note Tantalum	Parts ConneXion
R5, R6	22k 12W	Mills MRA12	Parts ConneXion
R101, R103	2k2 12W	Mills MRA12	Parts ConneXion
R102, R104	1k 12W	Mills MRA12	Parts ConneXion
R105, R106	1k5 0.75W	Caddock MK132	Parts ConneXion
R107	100Ω 12W	Mills MRA12	Parts ConneXion
R108, R109, R112, R113	1.5k 0.5W	Audio Note Tantalum	Parts ConneXion
R110, R111, R114, R115	75Ω 12W	Mills MRA12	Parts ConneXion
R116, R117	220Ω 2W	Any	
R118	120Ω 0.5W	Any	
R119	500Ω 1W	Any potentiometer	
IC1	CS8414-CS	Cirrus Logic	www.newark.com
IC2	74HC04	Any	www.newark.com
IC3, IC4	AD1865N-J	Analog Devices	www.newark.com
IC101-IC103	TL431	Any	
IC104	LM338T	Any	
SB101-SB103	250Ω at 100MHz	Fair-Rite	www.newark.com
T1	TRANS-280	Audio Note S/PDIF	Parts ConneXion
T2, T3	Sowter 9762	Sowter	www.sowter.co.uk
T4, T5	Sowter 8650	Sowter	www.sowter.co.uk
T101	185C10	Hammond	www.newark.com
T102, T103	185C24	Hammond	www.newark.com
T104	185C16	Hammond	www.newark.com
Vt1, Vt2	6H30Pi	Russia	www.tubestore.com
Vt101 5Y3			www.tubestore.com

RetroDAC Versus Audio Note DAC 5.0

The main differences between RetroDAC and Audio Note 5.0 are:

1. I used two DACs to produce differential signal with no grounded primary tap. With this tap, schema is not completely differential, sounds worse (to my ear), and looks more like two Audio Note DACs, working in push-pull mode.
2. I did not use filters, described in the Audio Note patent (US # 5,420,585). Balanced schema allows you to diminish filter to the same I/V transformer, which is obviously better in my opinion.

So which DAC sounds better? I do not know. I never listened to DAC 5.0. It is not a competition, but just another parallel universe.

HOW DOES IT SOUND?

If you expect to discover powerful bass, which is going to blow you away, or super highs—this DAC is not for you. However, if you enjoy big classic orchestra or chorus, "smart rock" such as Pink Floyd or Genesis, or big jazz bands—in short, any music performed with good musicians and many instruments—you will be surprised. RetroDAC easily reproduces the most complex music. I hear many new sounds, instruments, and voices in records I know perfectly well.

RetroDAC also has an outstanding dynamic of sound. It reproduces well loud and faint sounds at the same time

ACKNOWLEDGMENT

I want to express my appreciation to Mr. Brian Sowter; who was very cooperative during about a month-long exchange of opinions over e-mails, regarding construction and parameters of a custom Sowter 9762 I/U transformer (Sowter Audio Transformers, England, www.sowter.co.uk). The result of those conversations exceeded my expectations.

with minimum interruptions. The same feature gave another interesting result: a depth of sound picture. In many cases, especially in movies, you may judge how far the sound is from you.

Even my wife admitted that now she can listen to the music for a long time without tiring. RetroDAC brings the mood of music or movies. Horror or joy, happiness or sadness—no matter what.

And do not forget, please! You must use a really good amplifier and almost perfect speakers to get full advantage from RetroDAC. Those two components of your audio system are most important. Sound cannot be significantly improved just by using any DAC if you use average hi-fi. **aX**

Yury Gelgor *was born in the Ukraine and received his bachelor's degree in industrial engineering. He is an experienced industrial electronic technician, electronic engineer, and computer programmer. He moved to the U.S. in 1994. Both a hi-fi and tube-sound lover, his dream is to find a job that matches his hobby.*

PHOTO 5: Top view. Digital board cover removed.

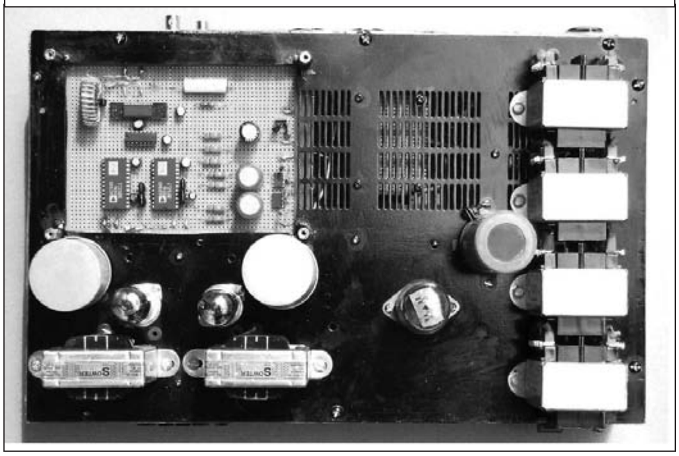

PHOTO 6: Bottom view.

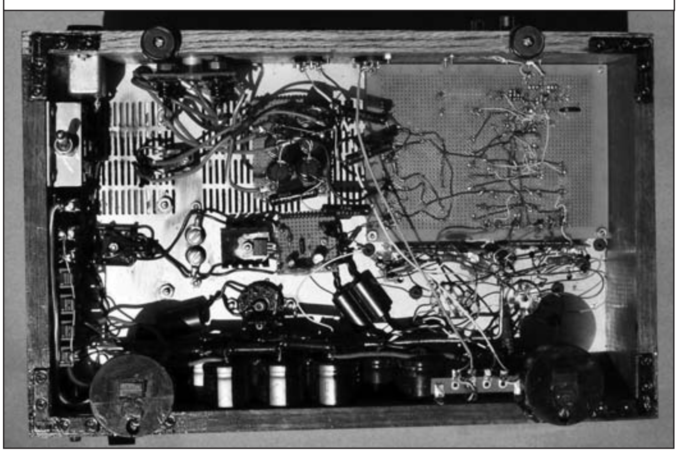

PHOTO 7: Rear view.

LAST MINUTE UPGRADE

It has now been some time since I've written "A Balanced Nonoversampling Retro DAC," and I could not stay off my DAC and just listen, so I made another upgrade I want to share with you.

First, I completely removed negative power supply for tube bias. It means the following parts were completely removed: F104, D101, C106, C107, C108, C109, C110, R103, and R104. Resistors R5 and R6 are now connected not to 115V power supply, but directly to Output Ground (OuG). New values of R5 and R6 are 2.2k 1W and they are tantalum Audio Note resistors now.

Second, I replaced capacitors C1, C2, C4, C9, C10 with Black Gates NX 100µF 6.3V and put jumpers instead of coils L101–L107.

Third, C102 is removed, C103 and C104 are Black Gate 220µF 160V and C105 is 68µF 350V now.

All that not only made the schema more simple, but significantly improved clarity and musicality of the DAC. ■