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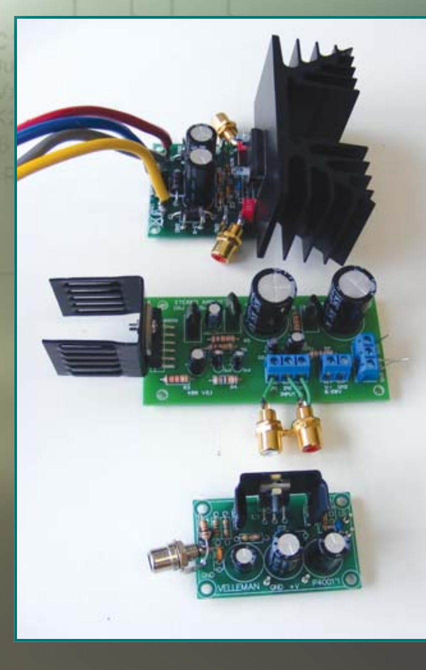

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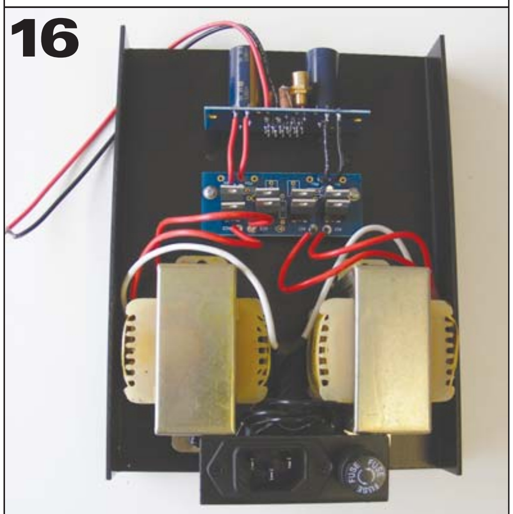

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A Push-Pull 7591 Power Amp

Another example of Thai interest in tubes to meet the needs of audio enthusiasts.

PHOTO 1: Finished power amp.

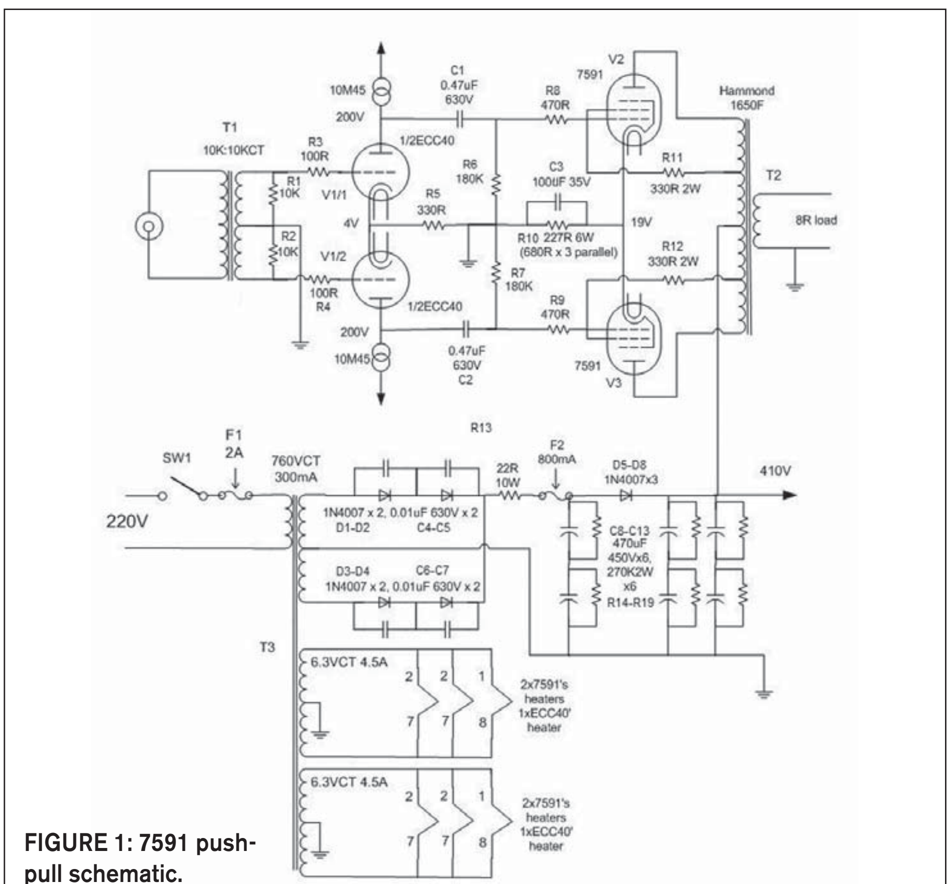

FIGURE 1: 7591 push-pull schematic.

I have never owned the 7591 power tube because I couldn't find NOS ones for a reasonable price. I believe most NOS 7591s are replacements for vintage units, such as Fisher, Scott, McIntosh, and countless brands I never heard of. Although 6CM5 is the 7591 in a different base, vintage collectors prefer not to change the tube sockets because it reduces resale value of the unit.

Fortunately, JJ Electronic and Electro-Harmonix (www.jj-electronic.sk, www.ehx.com) started making 7591s in recent years. That eased the need for NOS 7591s. Although the retail price is a little higher than other new production tubes, such as 6L6GC or EL34, it's still a good price ($16-$18 for new production vs $50 for NOS). However, I didn't own any vintage units, so I decided to build a new one (**Photo 1**).

NO FEEDBACK DESIGN

Several traditional designs of the 7591 amplifier are based on the Williamson or Mullard design, which has plenty of gain in driver stages, and about 20dB of global negative feedback to reduce this gain and widen the frequency response and lower output impedance. My design went the opposite way by using the 7591 without feedback and had moderate gain in driver stage. The no feedback design made it easy to hear different sound from different brands of tube. I still hope that someday I can find NOS samples to compare to the current production.

DRIVER TUBE

I designed the power section based on the Tung-Sol 7591 datasheet. The suggested bias point for push-pull ultralinear with cathode resistor bias was 420V at the plate, and -21V cathode and current was 88mA (for two tubes). The power output was 26W. With -21V bias, I didn't need a high mu tube such as 12AX7 or 6SL7, but a triode with a mu of 20-40 would work just fine.

6SN7 seemed to fit the bill for this project, but because I have built several amps with the 6SN7, I decided to use another tube. I dug into my private stock and found an ECC40 that I had bought from a local shop in Thailand ten years ago, but I couldn't do anything with because there was no socket available. That applied to other rimlock tubes such as the ECC40, EL41, and EF40 sold by online retailers but without their sockets.

Until recently, I noticed the new production of rimlock socket available on eBay as well as from some online tube retailers. I guess the demand for using these tubes has increased lately. So, using ECC40 is now possible for DIYers. Philips is perhaps the most common brand of ECC40, but I see almost every European brand, such as Valvo, Siemens, Telefunken, and Mullard, for the ECC40.

THE CIRCUIT

Figure 1 shows the amplifier schematic and power supply. The design is quite simple. The IXYS 10M45 current regulator IC is used as a constant-current source (CCS) load for ECC40. You may use Supertex DN2540 as the CCS. I chose 200V, 6mA as the bias point of the ECC40. The voltage bias at the cathode of ECC40 was 4V, which is more than enough for driving small power tubes such as 7591.

I recommend adjusting the 10M45 before you put it to the real circuit. You can use the schematic in **Fig. 2** for the adjustment. The power amp's power supply is used for the adjustment process. A stopper resistor on the 10M45's anode is recommended to prevent oscillation.

Before turning on the power supply, adjust pot 5kW to maximum value for the lowest current. Then turn on the power supply and check out the voltage reading across the 30K resistor. Adjust the 5kW pot until you read 180V across the resistor. That means you adjust the 10M45 to pass the current of 6mA.

In the original 7591 datasheet, 1MΩ resistor was the maximum value of

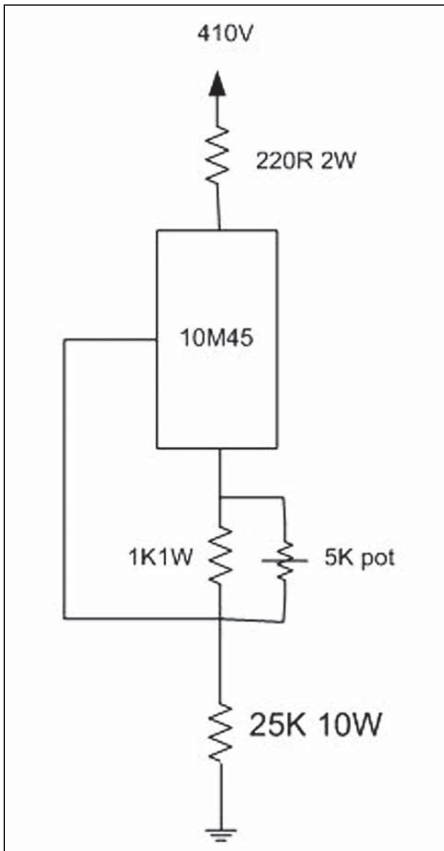

FIGURE 2: A 10M45 adjustment.

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Dimensions W x H x D:
W: 137mm (5.4")
H: 218mm (8.6")
D: 81mm (3.2")
Cut-Out W x H:
W: 108mm (4.25")
H: 190mm (7.5")
AC Voltage: 115V / 230V

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the grid resistor for cathode bias. This may cause instability for new production tubes. Many tube dealers recommend reducing the value to 150-200kΩ to avoid the problem. I chose 180kΩ, which was more than enough for the 10kΩ plate resistance of ECC40. I used 0.47µF coupling cap because I have many of them. You can use 0.22µF without losing low-frequency response.

The total 7591's plate and screen grid dissipation is about 16W, which is well below 22W (19W for plate and 3W for screen grid). So, the power tubes should have full life service. The output transformer I chose was Hammond 1650F. With 7.8kΩ primary impedance, I got 20W RMS with good frequency response from 20Hz to 20kHz. Not bad for a non-NFB power amp. I may get more power with lower primary impedance, but I don't have other output transformers to try.

You can use the female XLR connector for a balanced input connection. Just solder two wires from the input transformer's primary to pin 2 and 3 of the connector. Then solder pin 1 to the chassis ground.

If you don't have an input transformer for the phase inverter, you can use a balance driver IC such as the Burr-Brown DRV134. Or you may use a unity-gain stable op amp for the phase inversion job. **Figure 3** shows a simple phase inverter using Analog Devices OP275.

CONSTRUCTION

I built this power amp for bedroom listening, so I needed to squeeze everything into a small chassis. To do that, I cut an aluminum plate and drilled holes for sockets, output transformers, and the power transformer. First, I assembled the amplifier circuit. Then, I mounted the output transformers and connected the primary wires to the output tubes. I also mounted the power transformer during this step because I needed to connect filament wire to the tubes before stacking the power supply board on top (**Photo 2**).

I assembled the HT power supply on a blank fiberglass board with holes drilled for filter capacitors and solder terminal. I mounted the power supply board on another aluminum plate, to be installed on the top of the amplifier circuit later. If you do it the same way I did, you need to make sure that you have enough clearance between the amplifier circuit and the aluminum plate. **Photo 3** shows how the amplifier circuit and power supply aluminum plate mount together.

I also cut another fiberglass board and drilled holes for solder terminals and soldered on 10M45s, anode stopper, and an adjustment resistor. This way, I was able to set the current of the 10M45s before installing the board on the aluminum plate. I placed the 10M45's board next to the power supply board to keep the wiring short (**Photo 4**).

I tested the amplifier in "nude" assembly before placing it in the aluminum chassis. Finally, I placed the aluminum chassis in a wood frame, which I had input-transformer-mounted behind the front panel. I did that because the input transformer was very sensitive to hum that's caused by the power transformer.

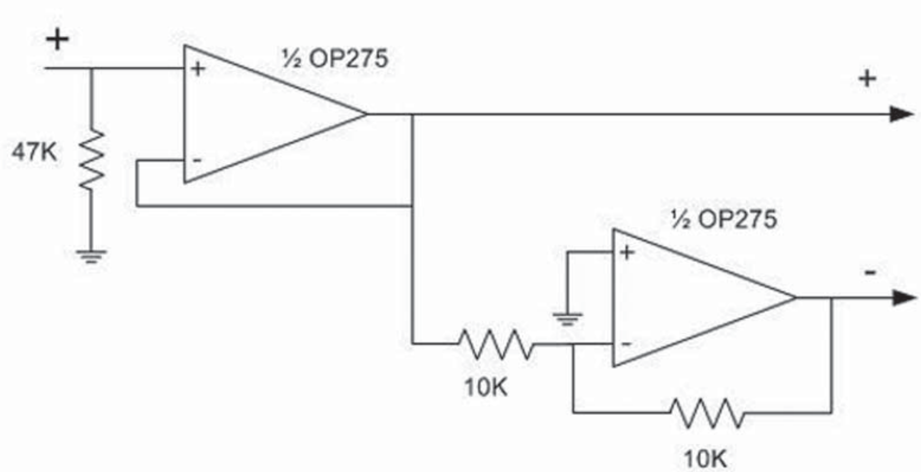


FIGURE 3: Simple phase inverter.

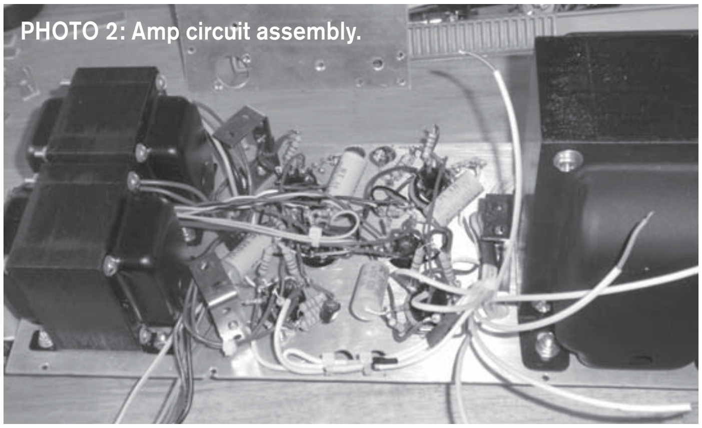


PHOTO 2: Amp circuit assembly.

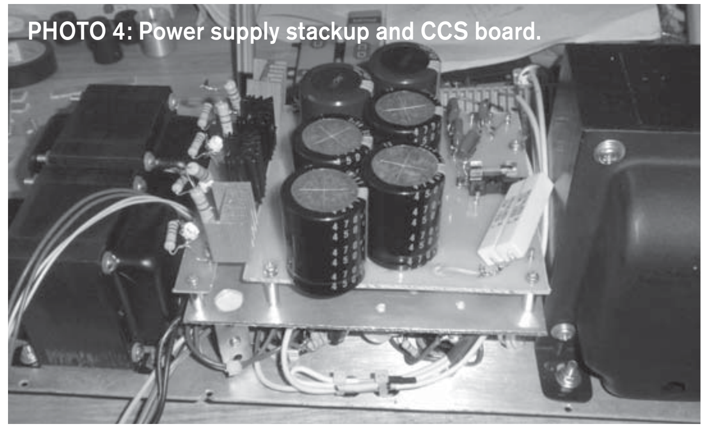


PHOTO 4: Power supply stackup and CCS board.

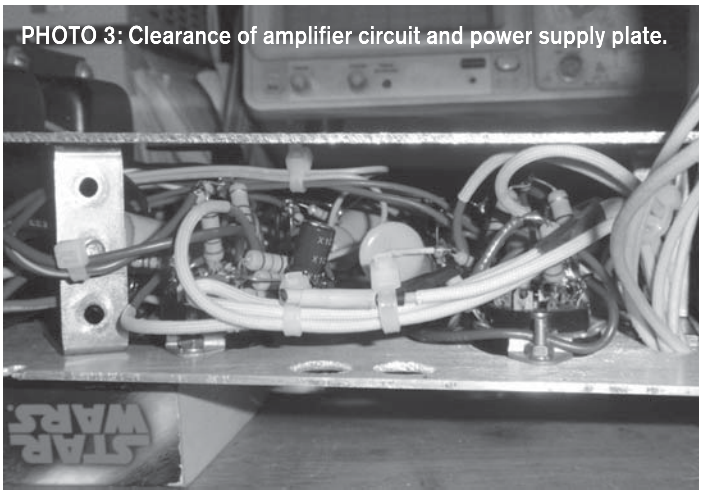


PHOTO 3: Clearance of amplifier circuit and power supply plate.

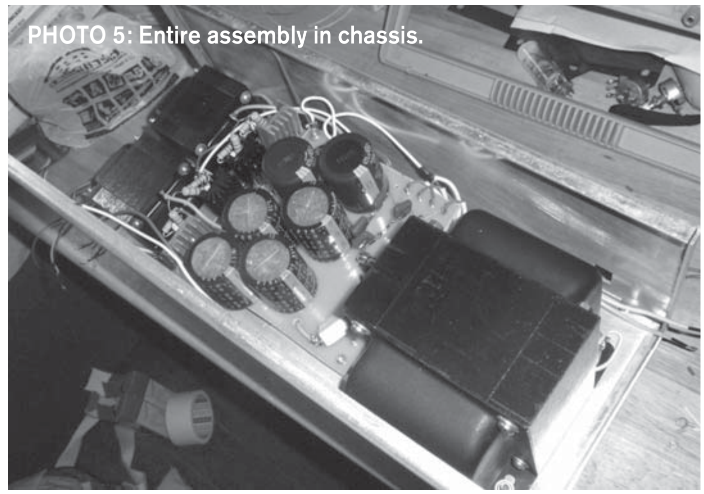


PHOTO 5: Entire assembly in chassis.

Photo 5 shows how everything fits in the small chassis.

SOUND QUALITY

I used the Fostex FE166 driver in a double back load horn cabinet for the audition. The double horn is basically Fostex's recommended horn that's stacked with the inverse configuration of its own on top. It gives deeper bass and can play louder than the original Fostex's recommended horn. I used the Denon DCD1400 CD player and a DIY WE407 preamp with 8:1 step down output transformer.

This power amp has an interesting sound quality, with good bass reproduction—not super tight or deep, but just the right amount. It also gives a good balance of bass, mid, and treble. With 96dB/W, it has plenty of headroom for many kinds of music. It also does a good job with regular two-way loudspeakers with 90dB/W sensitivity.

This power amp surely is not a classic one. It's for anybody who wants to try the 7591 tube but doesn't want to stick with a vintage circuit. It's easy to build and the sound is very pleasant to listen to. *aX*

PARTS LIST

Amplifier (1 channel)

R1, R2	10kΩ ½W
R3, R4	100Ω ½W
R5, R11, R12	330Ω 2W
R6, R7	180kΩ ½W
R8, R9	470Ω ½W
R10	680Ω 2W × 3
C1, C2	0.47μF 630V
C3	100μF 35V
V1	ECC40
V2, V3	7591
T1	line level input transformer 10K:10KCT
T2	7.8K:8Ω output transformer (Hammond 1650F)

Power supply (for both channels)

R14-R19	270kΩ 2W
C8-C13	479μF 450V
D1-D8	1N4007
T3	Power transformer 220V primary 760VCT 300mA, 6.3V 4.5A × 2 seondary

By Claude Goeuriot

A Power Sine Generator for Turntables

Take control of your turntable with this variable generator.

PHOTO 1: A power sinus generator for a turntable synchronous AC motor.

I like old Thorens turntables, and I recently started to renovate a venerable TD 160 whose platter is belt driven by a synchronous AC motor. When I measured the TD 160, even with a brand new belt, the speed of this record spinner was 0.77% lower than expected, giving 3126Hz for the 3150Hz signal of the test record. Not a big deal, but my hunt for perfection led me to design a variable 50/60Hz sine generator for more precise speed (**Photo 1**).

The main advantages of the Power Sine Generator are:

- Stable frequency (quartz generated)
- Possibility of changing the frequency of the output signal by 0.4% step
- Pure output signal (compared to the mains AC signal in my flat)
- Electronic 33/45 speed switch
- 115V/230V AC output
- Optional 50Hz/60Hz output (by downloading the appropriate program to the processor).

In this article, the signal frequency is 50Hz, but a software version is also available for 60Hz signal.

WARNING

This project is connected to 230V AC mains supply (alternatively, 115V AC mains supply) and is potentially lethal. Furthermore, it generates lethal output voltages (again, 115V AC/230V AC). As a result of this, please observe the following:

- **Do NOT build this project unless you are completely familiar with main wiring practices.**
- **The circuit MUST be built into a fully enclosed case connected to ground or in an isolating plastic enclosure.**
- **Do NOT touch anything inside the case when the circuit is powered (even if turned off).**

DESIGN

Today's technology will help generate a signal that will vary around 50 (60) Hz with the use of a microprocessor. **Figure 1** shows the block diagram of the PSG-1: a microprocessor running at 20MHz drives a digital analog converter to generate a 50Hz low voltage signal. Each period is generated with 200 intermediate steps. The signal then goes through a low-pass filter that eliminates the staircase effect of the quantification.

The final stage is an audio power amplifier driving a step-up transformer for an output signal of 115 or 230V AC. A nine-position switch allows for a ±1.6% variation of the frequency around the nominal value. Different voltages are necessary for the PSG-2: ±30V for the amplifier, ±12V for the low-pass filter, and +5V for the microprocessor.

Figure 2 shows the interconnections between the different boards. The transformers are the only components that are not mounted on the boards. The first transformer provides the 2 × 20V AC (30W) necessary for the power supply of the amplifier board and the second one is used as a step-up transformer to generate the 115/230V AC output signal.

OPERATION

The PSG-2 is built around three different printed circuit boards (**Photo 2**):

- The front face PCB that houses the switches and the associated electronics
- The microprocessor and low pass filter PCB (**Photo 3**)(with additional ±12V daughter board)
- The audio power amplifier.

The front face PCB is a very simple board (**Fig. 3**), parallel to the front face, that houses the 12 (nine used)-position rotary switch for speed variation and the 33/45 switch. The electronic circuit is simple: a TTL 74LS147 or equivalent converts the signal from the rotary switch into a four-digit value that will be acquired by the microprocessor.

The microprocessor (**Fig. 4**) is an old PIC 16F84 that I used because I had it in stock. Microchip producers now offer more functionality in the same package, but the old 16F84 does the job. The 16F84 is clocked by 20MHz quartz and drives an 8-bit digital analog converter based on R-2R topology. Connector P1 links the microprocessor to the front face PCB, and connector P2 allows you to connect the PICKIT2 debugger and memory loader.

FIGURE 1: PSG-2 block diagram.

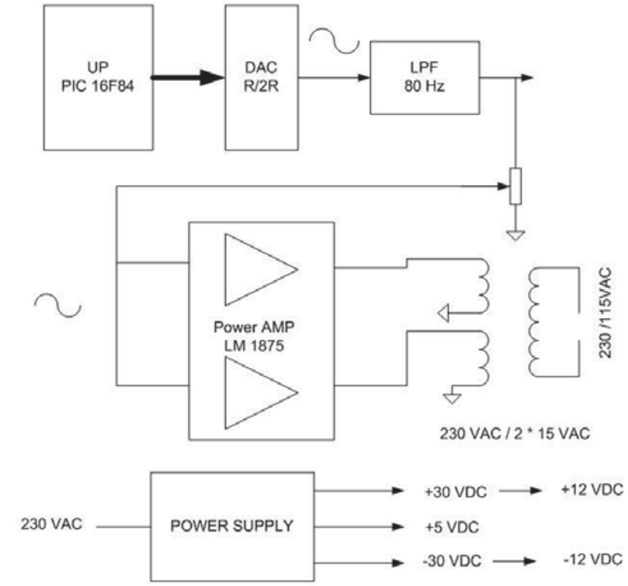
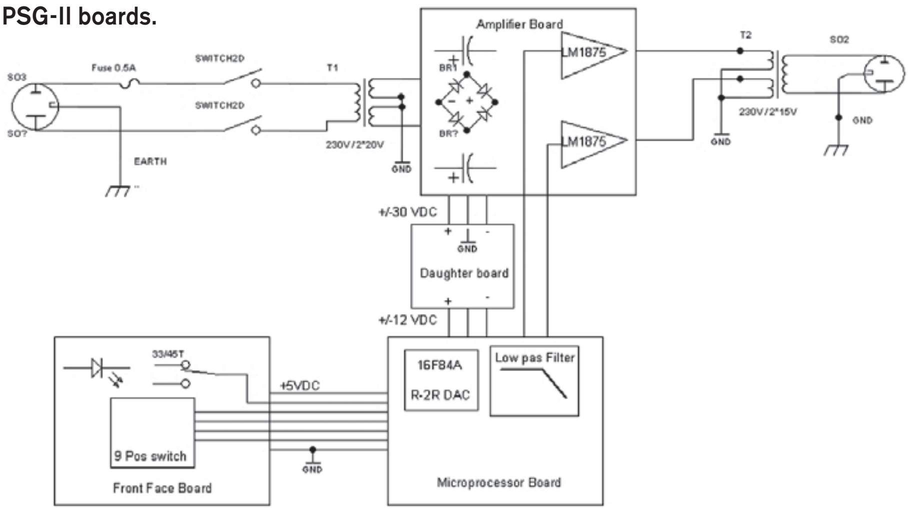


FIGURE 2: Interconnection between PSG-II boards.



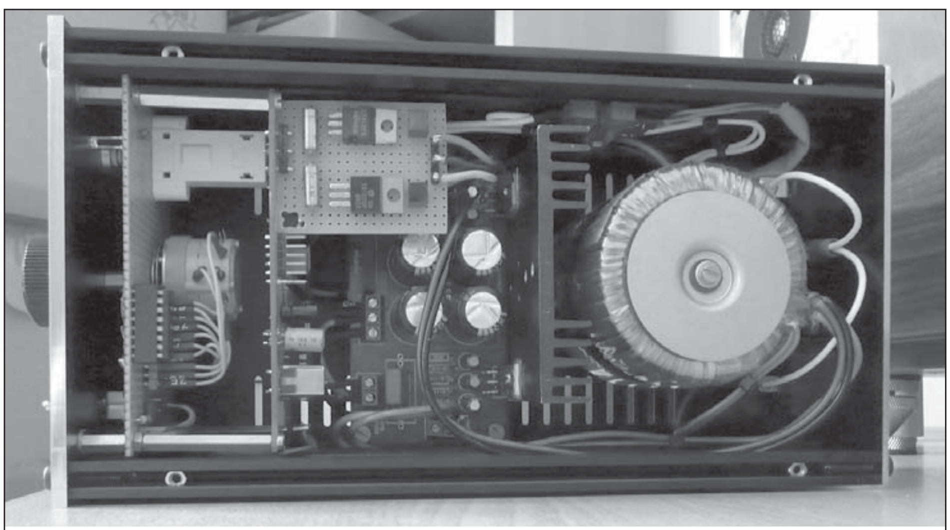


PHOTO 2: Physical arrangement of the different blocks in the cabinet.

The software includes a very simple loop whose duration is based on the 33/45 switch. The duration of the loop, basically 100µs for 33T speed, is modified in relation to the position of the rotary switch. During these 100µs, the microprocessor incrementally reads the value of the signal to be generated in a look-up table. After 200 cycles (200 × 100µS = 20mS or 50Hz) the counter resets and a new 50Hz period starts.

The low-pass filter is a 12dB/octave

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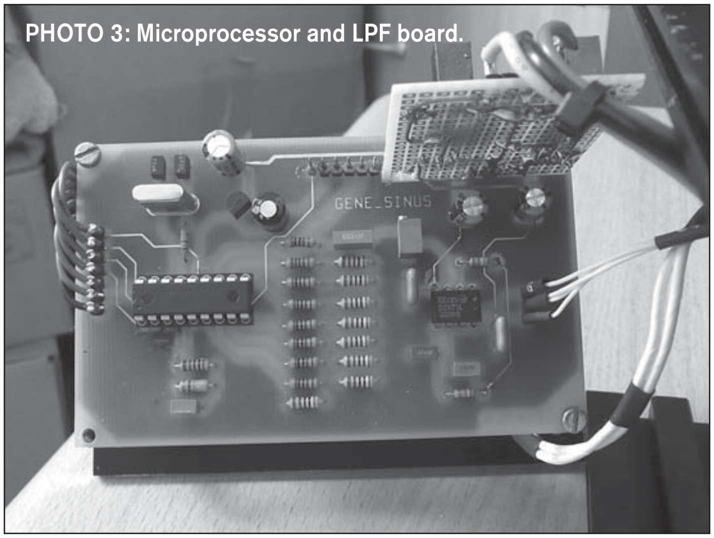

PHOTO 3: Microprocessor and LPF board.

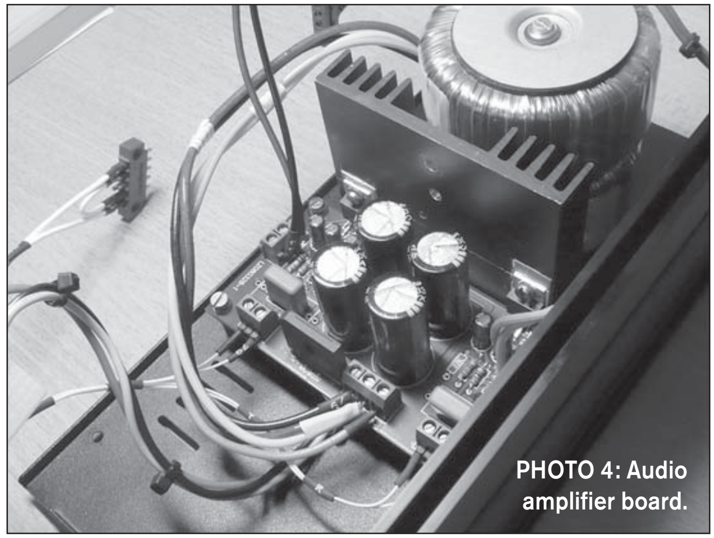

PHOTO 4: Audio amplifier board.

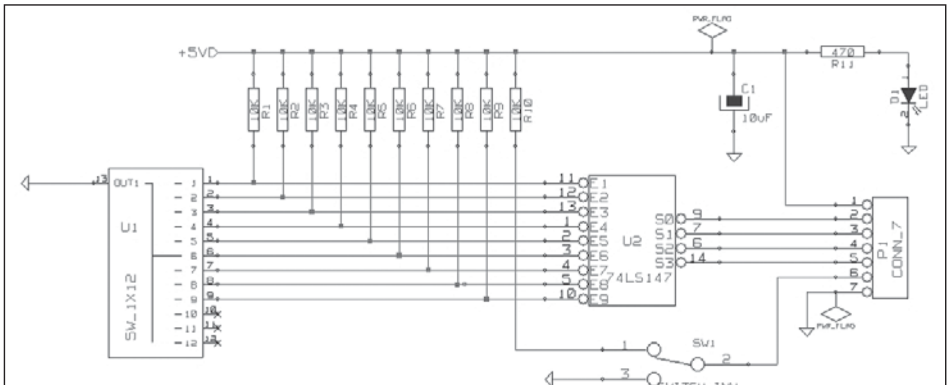

FIGURE 3: Front face board schematic.

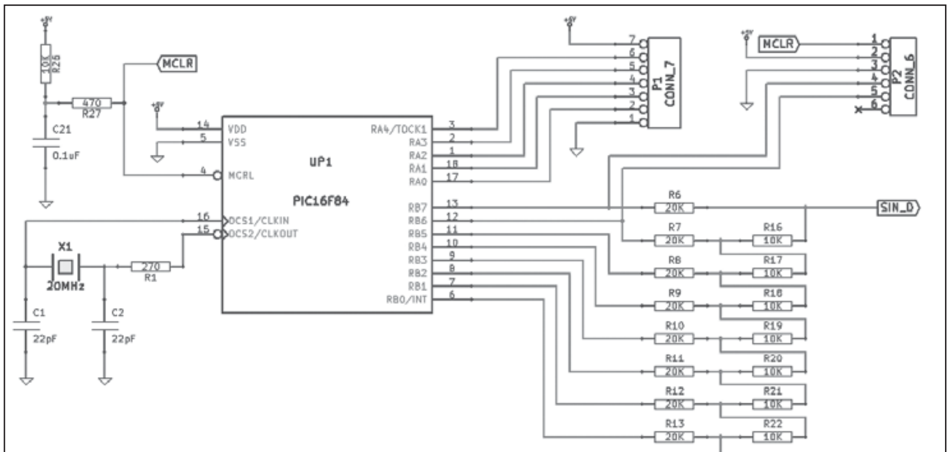

FIGURE 4: Microprocessor schematic.

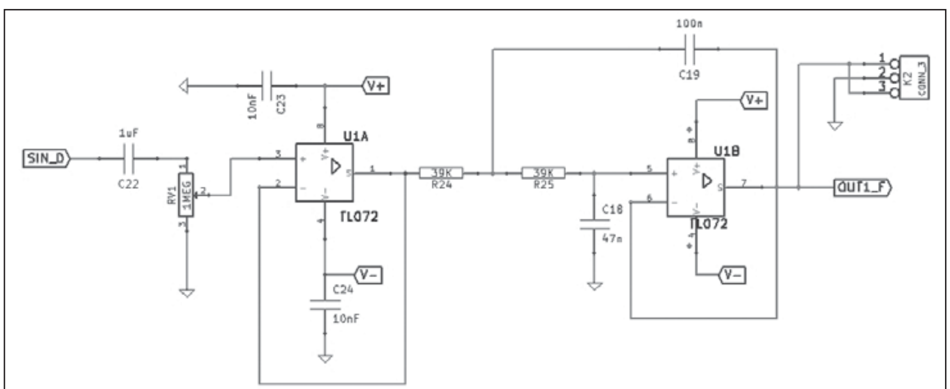

FIGURE 5: Low-pass filter.

cell built around a double op amp (**Fig. 5**). The potentiometer allows for fine-tuning of the output voltage. The first op amp is just a buffer and the second op amp is used for a standard Sallen-key filter structure. Cutoff frequency is around 70Hz (because the output frequency will be 67.5Hz for 45T operations) and can be adjusted by replacing C18 and C19. For 60Hz operation, I change the value of C18 and C19 to 33nF and 68nF, respectively.

The PCB, which is sandwiched with the front face PCB, is a double-sided PCB that has been designed using KICAD, a great CAD package available for free. A nice feature of this package is that it allows you to see a 3D picture of the PCB (**Fig. 6**).

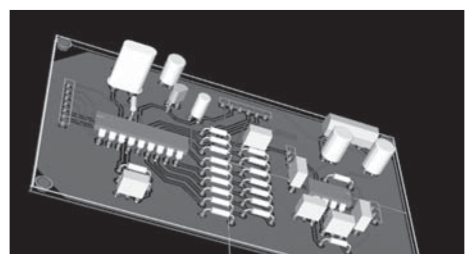

FIGURE 6: Microprocessor and low-pass filter (LPF) PCB.

Note: My initial plan was to use an audio amplifier board that would generate the ±12V DC, but, unfortunately, it didn't fit the case I wanted to use. So I bought another amplifier board online and was forced to build a daughterboard on a prototype board to generate the ±12V DC from the ±30V DC of the amplifier board, using standard TO220 7812 and 7912 regulators (**Fig. 7**). Note that the +5V DC regulator is mounted on the Microprocessor Board.

The amplifier board (**Photo 4**) is a kit that uses LM 1875 devices and is available on eBay for about $22 plus shipping. With two channels available, I chose to drive each side of the output transformer (2 × 15V AC/230V AC) by one amplifier channel. I use a spare heatsink to cool the two LM 1875s. The board includes a diode rectifier and large power supply capacitors and should be connected to a 230/115V AC/2 × 20V AC transformer. Building the kit was straightforward and worked the first time I powered it.

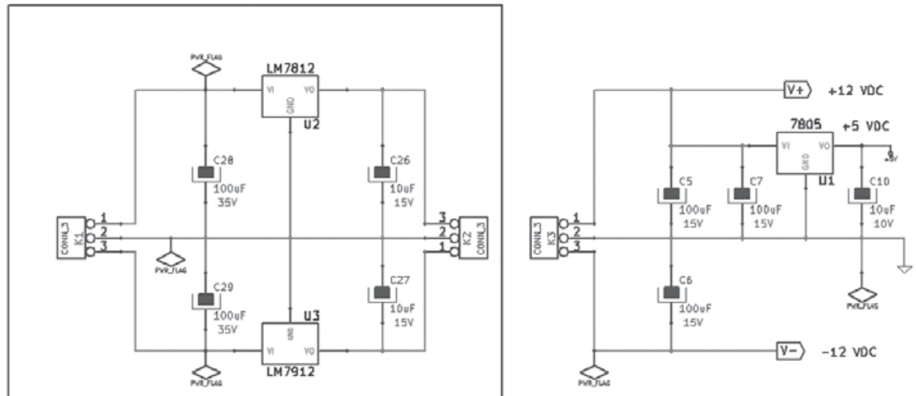

FIGURE 7: ±12V DC daughter-board, +5V DC supply.

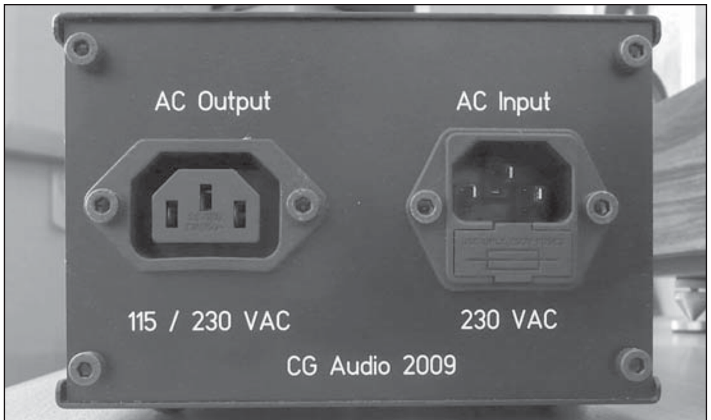

PHOTO 5: Rear plate.

PHOTO 6: Front face.

HOUSING

The housing of the PSG-II is a 104W × 80H × 230mm D (internal) box available in France at www.audiophonics.fr. The box is pretty well stuffed with the transformer and the heatsink, leaving just enough room for the three PCBs at the front of the box. Front and rear plates have been designed with Front Designer software and have been manufactured by Schaeffer. It gives a professional and retro look to the PSG-II (**Photos 5** and **6**).

The unit is nearly finished. After the traditional verification (Do it twice, remember you're playing with lethal voltage), you can power up the PSG II and verify that you have an output AC signal. With the unit powered, adjust the output AC voltage using the potentiometer RV1 located on the Microprocessor board.

TESTS & MEASUREMENTS

I performed tests using either a dummy 4.7kΩ load or a real turntable. The instruments used for the tests included:

- PC running under Windows XP

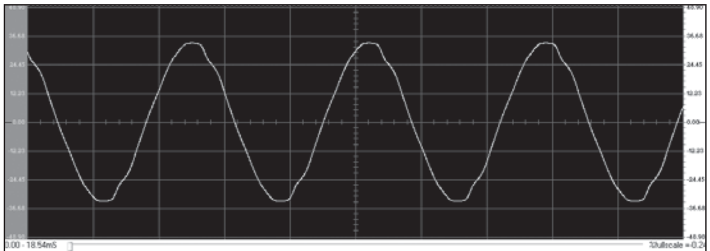

FIGURE 8: AC main signal.

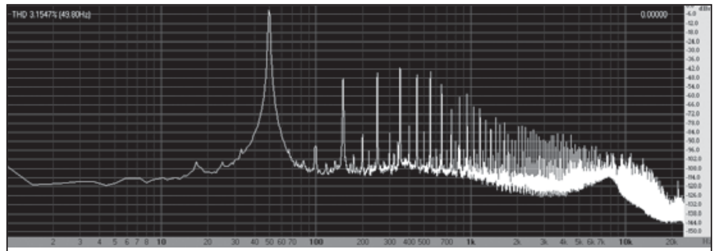

FIGURE 10: AC main spectrum.

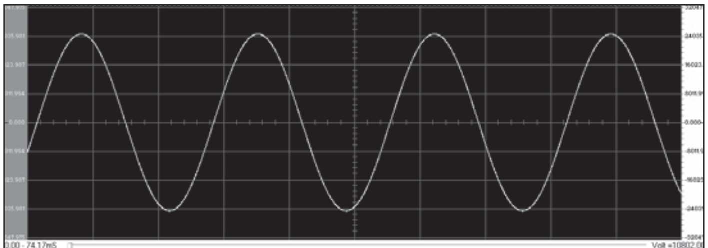

FIGURE 9: PSG-II output signal.

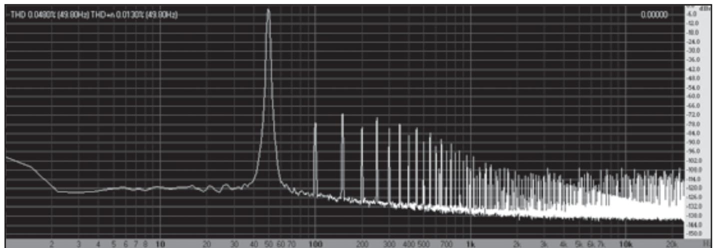

FIGURE 11: PSG-II output spectrum.

PHOTO 7: Hi-fi setup used during test and measurement.

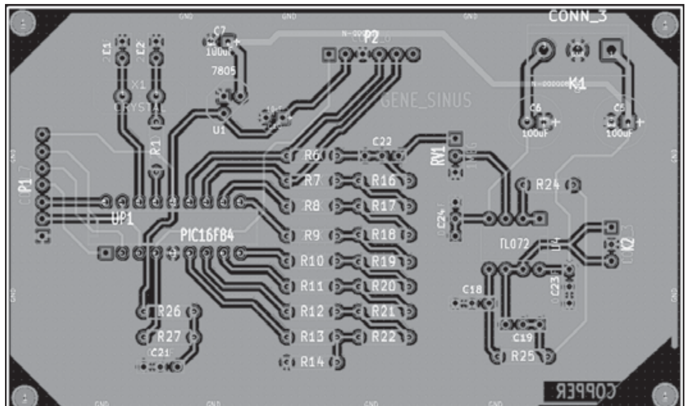

FIGURE 12: Microprocessor + LPF PCB.

- External sound card E-MU 0404 USB
- Virtual Analyzer Software
- HI-FI News test record & the Ultimate Analogue test LP (with 3150Hz signal)
- Frequency counter

Obviously, you also need some hi-fi gear (**Photo 7**): a turntable (Thorens TD 160, fully restored, with a DIY plinth), a power amplifier (Denon PM 520A), and some loudspeakers (Mustang, a DIY loudspeaker, see Sept. '10 *aX*).

The AC main signal (**Fig. 8**) is highly distorted compared to the PSG-II output signal (**Fig. 9**). And the spectrum of the signals confirms what you see on the virtual oscilloscope. Total harmonic distortion is much higher for the AC main signal, with several odd harmonics at –36dB compared to the fundamental signal (**Fig. 9**). With the PSG-II, all harmonics are 66dB under the 50Hz signal (**Fig. 11**).

Table 1 summarizes the performances in speed accuracy obtained with the PSG-II:

- With a 3150Hz signal generated by the test record, the turntable powered from the AC main delivers a 3126Hz signal with a 0.76% error.
- The PSG-II is able, on position N°2, to achieve a 0.06% error, with a signal delivered at 3148Hz.

TABLE 1: 33T SPEED ACCURACY

PSG-II Switch position	Frequency (HZ) 3150 frequency	error %	1000 frequency	error %
-4	3073	2.44%	976	2.39%
-3	3085	2.06%	980	2.00%
-2	3098	1.65%	984	1.61%
-1	3110	1.27%	988	1.21%
0	3122	0.89%	992	0.82%
1	3135	0.48%	996	0.43%
2	3148	0.06%	999	0.06%
3	3160	0.32%	1004	0.37%
4	3173	0.73%	1008	0.80%
AC	3126	0.76%	992	0.77%

TABLE 2: 45T SPEED ACCURACY

PSG-II Switch position	Frequency (HZ) 452.5 frequency	error %	1350 frequency	error %
1	4242	0.25%	1347	0.21%
2	4265	0.29%	1355	0.34%
AC	4200	1.237%	1333	1.259%

TABLE 3: FRONT FACE PARTS LIST

Reference	Value	Comment
C1	10μF	electrolytic, 15V
D1	LED	3mm blue
P1	CONN_7	single in line
R1-10	10k	¼W 5%
R11	470	¼W 5%
SW1	SWITCH_INV	toggle switch on-off
U1	SW_1X12	12 position switch, locked to 9 positions
U2	74LS147	DIP TTL LS or equivalent (TTL, HC)

TABLE 4: MICROPROCESSOR PARTS LIST

Reference	Value	Comment
C1, C2	22pF	ceramic 50V 2.54mm
C21	0.1μF	ceramic 50V 5.08mm
P1	CONN_7	SIL
P2	CONN-6	SIL
R1	270	¼W 1%
R6-14	20k	¼W 1%
R16-26	10k	¼W 1%
R27	470	¼W 1%
UP1	PIC16F84	DIL 18
X1	20MHz	quartz HC 18

TABLE 5: LOW-PASS FILTER PARTS LIST

Reference	Value	Comment
C18	47n	
C19	100n	
C22	1μF	
C23	10nF	ceramic 50V
C24	10nF	ceramic 50V
K2	CONN_3	terminal block, 3 pins, 5.08mm
R24, 25	39k	¼W 1%
RV1	1MEG	
U1	TL072	DIP 8

TABLE 6: POWER SUPPLY PARTS LIST

Reference	Value	Comment
C5-7	100μF	electrolytic, 15V
C10	10μF	electrolytic, 10V
C26, C27	10μF	electrolytic, 15V
C28, C29	100μF	electrolytic, 35V
K1-3	CONN_3	terminal block, 3 pins, 5.08mm
U1	78L05	T092
U2	LM7812	T0220
U3	LM7912	T0220

This result is a little bit lucky because the difference between two positions of the rotary switch is 0.4%. So depending on the turntable, the maximum error can reach half of this 0.4% figure. You can improve this with software, but at the expense of a more limited range of speed variation.

Switching to 45T, but using the same test record, the best accuracy is achieved on position N°2, with 0.25% error, compared to 1.24% error with AC main.

CONCLUSION

Do I hear the difference between the turntable powered by mains or PSG-II? I should say definitely Yes, but I'm not unbiased because I've built this equipment and I really want it to make a difference! In addition, I have never considered my ears as a reference and I'm over 50. But now, I am sure that the turntable spins at the right speed and I suppose that wow and flutter have also been improved by the use of the PSG-II (I have no way to accurately measure it).

Further improvements to the PSG-II might include an LCD display and ± push buttons instead of the rather old-fashioned rotary switch to vary the output frequency.

ACKNOWLEDGMENTS

This article was deeply inspired by the document: "TTPSU—Power supply for turntables with AC motors" available at http://www.norre.dk aX

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- KICAD: http://kicad.sourceforge.net
- Virtual Analyzer: http://www.sillanumsoft.org/
- Microchip Mplab: http://www.microchip.com
- LM1875: http://www.national.com/mpf/LM/LM1875.html#Overview
- Denon: www.denon.com

By Ed Simon

Power Amp Kits for Beginners

Tiny amps for computers, MP3s, and iPods.

Here are some simple amplifier kits you might consider, along with a small loudspeaker idea.

A question that often arises is how to improve the sound from computers, iPods, or even portable CD players. Personal music players are designed to drive headphones, but it is often nicer to share music over small loudspeakers. You can use small self-amplified computer speakers, but these are often limited in capability. This type of project is often the first one many folks attempt. It can start out very basic and there really is no limit to how involved it can become.

I decided to try out some of the easier kits (with the help of some fellows of various kit-building abilities): two Velleman amplifier kits K4001 7W mono at $19.95 and K4003 2 × 30W stereo at $33.95, one kitsrus kit from Carl's Electronics 2 × 10W stereo at $19.95, a single channel card from AmpsLab Lm60 rated at 60W into 8Ω for $95, and a pair of gainclones from AudioSector—the chip is rated by its maker at 56W for $59. Shipping was extra for all of the kits. None of them came with a power supply, case, or, in some cases, not even the heatsinks.

VELLEMAN KITS

The first kit I tested was the Velleman K4001, which is a single amplifier that runs on 18V or so. I asked Tim to assemble this kit. Tim has lots of experience soldering connectors but has never put together a kit. He took it home and spent a small part of an evening assembling it. He noted that it was difficult to read the resistor markings because they were so small and was concerned he mixed up the feedback resistors. The kit uses ¼W carbon film resistors, which I do not consider small. I noted the kit instructions say that it comes with the parts on a tape typical of automated assembly with all the parts arranged by part number.

The finished unit has six pins for the input, output, and power connections. I soldered an RCA jack directly to the input and used clip leads to a bench power supply and a loudspeaker. I plugged in my portable CD player and got a little bit of sound out of the amplifier. It seemed to have no gain. A quick check verified what experience told me—the 470Ω and 4.7Ω feedback resistors were mixed up. I used solder wick to remove the solder and swap the resistors. The PC board held up well to being reheated and the throughhole size was more than adequate.

The resistors were marked correctly but they were a bit more difficult than usual to read because the painted code lines were not uniform or even well spaced. Of course, a good lesson here is if you are ever designing a kit, use completely different values for parts (i.e., 470Ω and 6.8Ω would be harder to mix up).

After the swap the amplifier performed well except for the bad habit of oscillating when the input was left unused. I put a 10K ¼W resistor across the input to cure the problem.

This amplifier, including the heatsink is quite small, so it would be useful in a small loudspeaker to be used as a self-powered unit. A 12 to 18V DC wall-wart style power supply providing at least 250mA could power it. Although the peak current draw could hit about 2A, this unit has an output capacitor that is the same size as the onboard power supply filter, so a supply with greater current may not offer significant improvement. The power seemed to be adequate for background level music.

The Velleman K4003 stereo amplifier assembled by Bryan, who does not solder as well as Tim, was his first try at a kit. This kit was a bit different, requiring a 24V center-tapped transformer of at least 50VA for an onboard power supply. This unit is rated at 2 × 30W into 4Ω. Because the power supply had positive and negative rails, an output coupling capacitor is not required. Even though the output signal current must flow through the

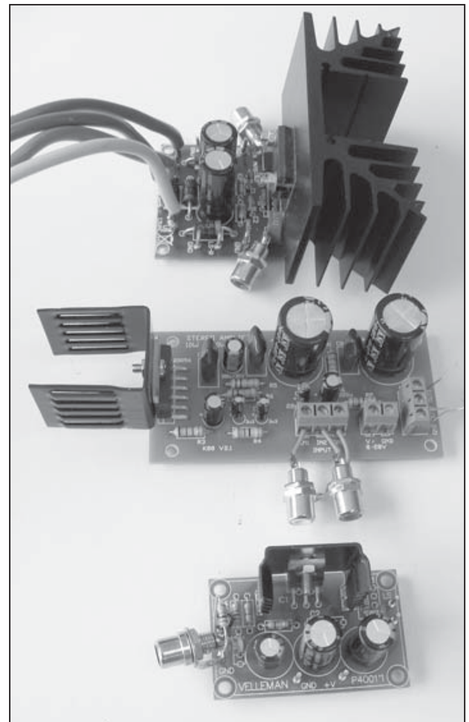

PHOTO 1: Velleman and kitsrus kits.

filter capacitors, the feedback loop compares the output voltage to ground so any capacitor effects now have feedback correction. This results in a noticeably tighter bass response. The volume is now loud enough for a reasonable listening volume with a small bookshelf loudspeaker, but certainly not a party system.

The inputs and outputs are handled by pins soldered into the PC card. There are ground pins by each input and output. I used a 300 VA, 2 × 12V AC toroidal transformer for the power transformer. Certainly more than the 50VA or so that is really required. Overall, this is a nice start for a small stereo amplifier. In addition to the transformer, a case, volume control, input and output connectors are required. The heatsink was the largest provided by any of the kits.

FIRST-TIME KIT

The Carl's Electronics' kitsrus kit #88 requires a DC supply of 8 to 42V, uses output coupling capacitors, and has larger values than the Velleman K4001, so the bass is a little bit better controlled, but not as good as the K4003. When used with a higher supply voltage, it also has a bit more oomph. This kit is more complete in that screw terminals are provided for the input and output. The downside is that they provide only one input ground terminal and just one for the outputs. The Vellemans used a TDA2003 amplifier chip for the K4001 and a TDA2616 for the K4003; this amplifier used a TDA2009.

I gave this kit to Gersh, who had a few concerns about how things fit but no real problems assembling it even though he had never done anything like it and only had rudimentary soldering skills. This was probably the easiest kit to assemble. The terminals make it easier to connect leads, although I prefer solder connections for reliability. I would expect a newbie to tin the leads of the wires going into the terminals to make it neater, not knowing that the solder will continue to flow under the screw pressure and eventually loosen up or slip out.

This kit would be fine in an old cigar box or fruitcake tin for a first project amplifier. You would need a wall-wart style power supply of about 1A at 24V, and some connectors. I would not use a volume control, but rather the volume control on the personal music player.

HIGH-QUALITY KIT

Next up was the gainclone amplifier kit. The term gainclone originated from a well-regarded stereo power amplifier made by 47 Labs called the Gaincard. It used a very simple circuit with short circuit paths and the National Semiconductor LM3875 IC as the active component. This inspired many to make similar projects.

I found a kit on the web from AudioSector that was offered in two versions, one with standard resistors and a second with the bulk metal Vishay resistors and other better parts. I picked the one with the standard parts. One of the parts provided was a miniature metal film resistor. I had previously made distortion measurements on this type of resistor. It has more thermal distortion than larger resistors, but I assumed it was selected because it has more pleasing-to-the-ear even-order harmonic distortion than most resistors. I changed my critical resistors to the Vishay Dale RN65C 1% equivalents, which were much larger, so I needed to mount them on the circuit board standing up. In my measurements of resistor distortion, these rated among the very best. I expect my version to be more representative of the deluxe version of the kit.

This kit came with nicer circuit boards (four total) than all the others, gold-plated double-sided and not run-of-the-mill fiberglass. All the others seemed to be G10 fiberglass single-sided copper. This kit did not come with a heatsink. So I punched out a heatsink that would also hold the power supply card, the amplifier card with the IC, two leftover 18V AC 50VA leftover transformers, a fuse holder, and an IEC input connector. These were designed to mount on the back of my test loudspeakers to make them powered speakers.

I needed to refer to the website for directions in assembling the four circuit cards to make two power supplies and two amplifiers. The power supply cards are designed to support either a center-tapped power transformer or a dual winding version. I assembled them with all of the provided low switching noise rectifiers to use the two winding version because I was actually using two transformers. The only problem was that the holes for input, power, and output were too small to take my standard 16 gauge MTW wire. I had to clip off a few strands of the conductors to get the wire to fit.

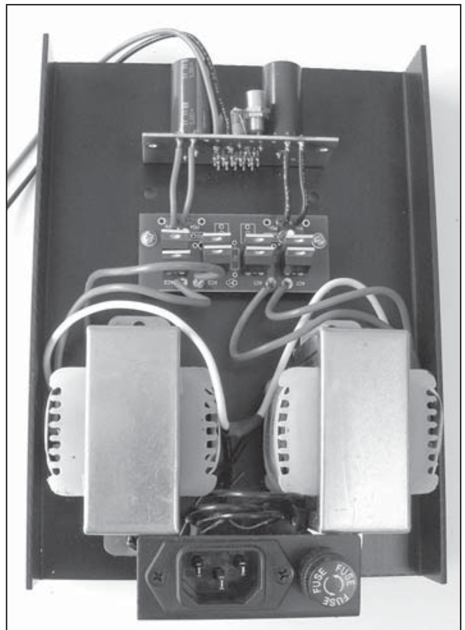

PHOTO 2: Gainclone fully assembled.

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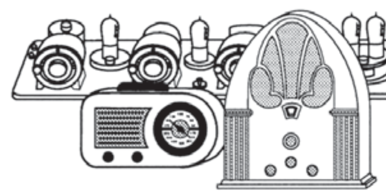

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These amplifiers made a nice stereo pair and met my expectations for a reasonably high-quality amplifier. I would not rate this an easy-to-assemble kit because of the options offered. A beginner might be able to get it to work, but might require some guidance.

Again, with this kit more parts are needed to make a complete amplifier. Include a heatsink to the list of parts. You also may wish to use a stepped attenuator instead of a standard plastic film volume control. Don't even think of using a cheap carbon volume control.

Dave assembled the final kit, which was from AmpsLab Lm60. Dave has previously built an amplifier on a breadboard and is skilled at soldering. I also punched a heatsink for this kit. The kit contained a CD with instructions, and all discrete parts using depletion mode power FETs for the output devices and was only a single channel. It did not include any power supply parts.

A 30-0-30V transformer (60V center tapped) with at least 10,000µF capacitors rated at 50V was recommended. I really had to search my junkbox, but I found a nice hefty transformer that unloaded was 62V and center-tapped. I had some 10,000 at 63V capacitors and a nice 15A 200V diode bridge. I cobbled these into the power supply. I had to solder wires to the PC card to make connections. Like the gainclone cards, the holes were not large enough for my stock 16 gauge wire.

This kit, unlike all of the others, required adjustments. After hooking up the power supply, I set up my voltmeter and light bulb box to adjust the output voltage trim and FET bias. A quick check showed the output was at –40V. The light bulb in series with the AC line did not light up, which means at least I probably did not fry anything.

A quick check showed that the PC board ground traces at the input were also at –40V. A bit of investigating showed that jumper #2 was missing. It should have been installed below one of the output power resistors to continue the ground circuit from the output side of the board to the input. Making sure the power supply capacitors were discharged, I soldered in the missing jumper on the bottom of the PC card. When I asked Dave about it, he mentioned he could not find where J2 was located and assumed the PC layout had changed and it was no longer needed.

After reassembling everything, I tried again. It worked. The output voltage trimmer started at 30mV or so and trimmed to just a few with ease. The output bias trimmer was a ten-turn type, so when setting it to minimum value as per the directions, you really don't know when you are there, so give it at least ten turns to be sure. The final adjustment of the bias was fairly easy, and not very far from the no bias end of adjustment.

This amplifier card is clearly intended to be a project amplifier. A nice front panel, a decent chassis, and other parts are in order. You might consider this for a multi-channel amplifier. Sound quality was quite pleasant.

MEASUREMENTS

Figure 1 shows the distortion + noise of the amplifiers (using one channel of the stereo amplifiers) into an 8Ω resistor versus the input drive level. You could easily misinterpret these curves. All of the amplifiers start out showing what appears to be 1% or more distortion. This is usually just noise, and is why the distortion appears to drop as the level increases.

What is clear is that the gainclone and the Lm60 had superior performance. What was not clear is whether

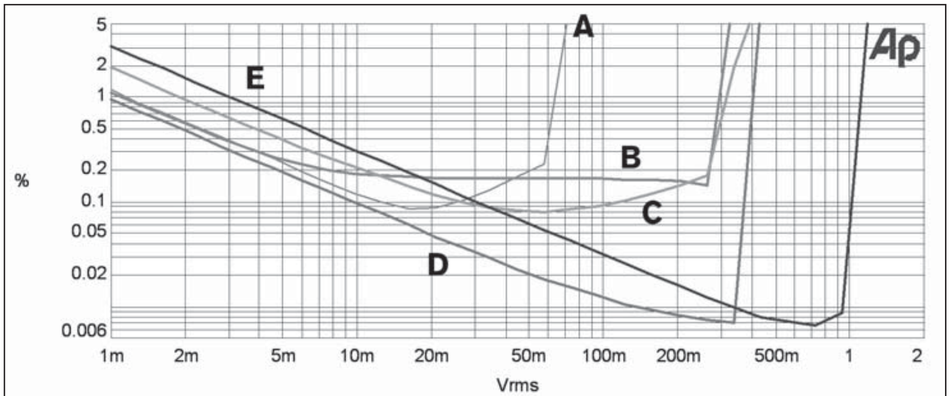

FIGURE 1: Distortion vs. input level. A = Velleman K4001 18V 8Ω, B = K88 amplifier 18V 8Ω, C = Velleman K4001 24VCT AC 8Ω, D = gainclone, E = Ampslab Lm60.

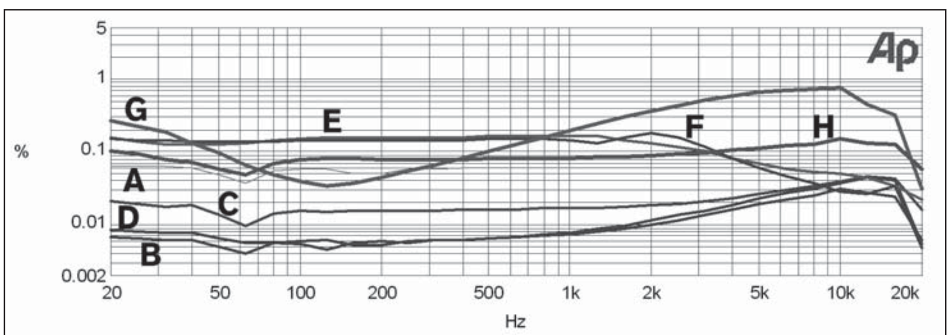

FIGURE 2: Distortion vs. frequency. A = Ampslab Lm60 50mV, B = Ampslab Lm60 500mV, C = gainclone 50mV, D = gainclone 250mV, E = K88 50mV, F = K88 250mV, G = Velleman K4001 50mV, H = Velleman K4003 50mV.

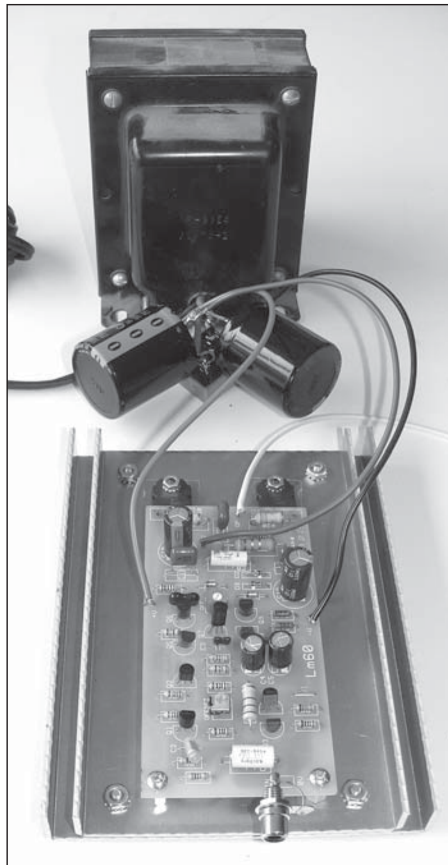

PHOTO 3: Lm60 mounted on heatsink.

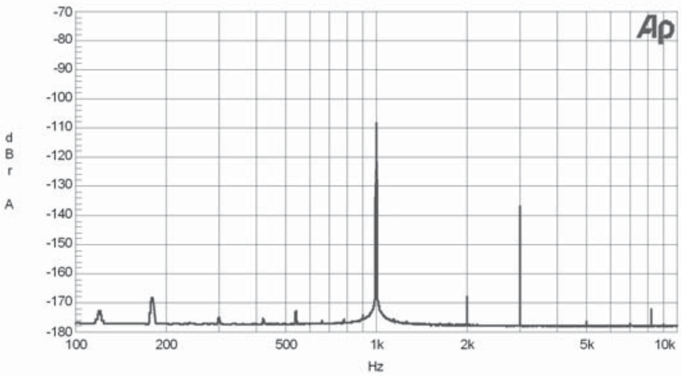

FIGURE 3: Distortion of the resistor type supplied with the basic gainclone kit. 1kHz test signal is at 0 shown here suppressed to view distortion better.

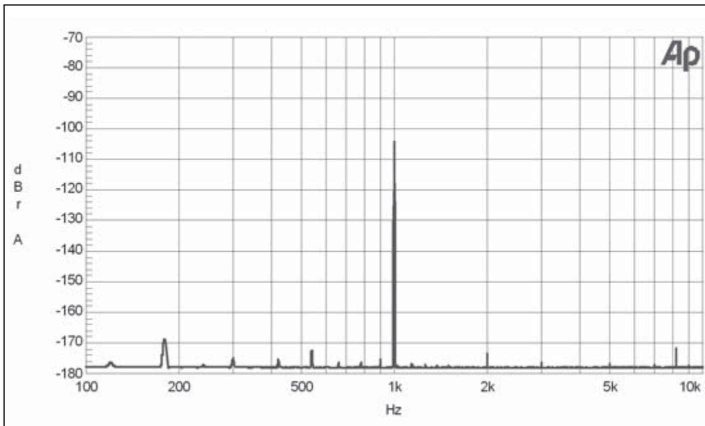

FIGURE 4: Distortion of Vishay Dale RN65C I used in the gainclone.

the Lm60 is noisy at lower levels or just showing low-level crossover distortion.

Figure 2 shows the distortion versus frequency. I ran two different levels for the three simple, similar kits, excluding the two Vellemans.

As you can see, the Lm60 at 50mV (A, at .07%) is straight. I assume that is because the noise is dominant. At 500mV it had the lowest measured distortion by almost literally a hair! So I would not use this amplifier with high sensitivity loudspeakers.

The gainclone at 50mV also showed the effect of noise but less than the Lm60. At 250mV (it clips with less input than the Lm60) it had similar very good results. The other three kits showed similar performance except the high frequency distortion was less for the kitsrus K88, probably due to the output network RC network loading.

COMPANION BOXES

There was one very happy discovery in the process of testing these small amplifier kits. I decided to pick up a small bookshelf loudspeaker kit typical of what could be used with these kits. I was shopping for some other needed parts at Madisound (www.madisound.com) and asked their recommendations on a small loudspeaker kit of high quality and typical of a first project. They were enthusiastic about the Zaph Audio SR71, but were out of the Madisound MD14 enclosure's parts. I ordered the kit with upgraded capacitors but no enclosure, because I have a reasonably complete woodshop.

The kit arrived with preassembled crossovers on a standard PC card. You have the choice of biamplifying these speakers, but I used jumpers to couple the inputs, because I wanted to evaluate amplifiers.

I looked at my wood stock and found it lacking in 1″ thick stock to make nice boxes. When I went shopping for a few small sticks, there was nothing outstanding at the local specialty shop. At the Home Depot I came across some nice-looking thick pieces of wood being sold as premade stair treads! Glued up from smaller pieces and covered with veneer, these were available in pine or oak, and just the right size for the boxes needed. I chose the pine, which I could easily stain and cost less money.

I had no noteworthy problems building the boxes from the plans provided with the kits. Lurking in my paint cabinet was a very old can of mahogany oil stain. That and a few coats of lacquer from a spray can finished the boxes.

The recommended burn in of 100 hours or so was just as easy. I hooked them up to the shop's solar-powered radio (I am frugal) and let them play for a month or so. Immediately noticeable was the amazing amount of low-frequency energy coming out of such a small box. A small loudspeaker I can truly recommend for beginner to advanced. I strongly suggest going with the MD-14 enclosure. I spent too much time just looking for the wood for such a simple project.

PHOTO 4: Zaph Audio SR71 in stair tread case with grille removed.

So if you want a really simple upgrade to your computer or personal music player, try any of the small amplifiers on a decent loudspeaker, but if you want stunning results, try one of the more powerful kits on a top-notch small speaker. *aX*

SOURCES

http://www.parts-express.com/home.cfm (Velleman kits)
http://www.electronickits.com/ (Carl's Electronic kits)
http://kitsrus.com/
http://www.audiosector.com/
http://ampslab.com/
http://www.vellemanusa.com/
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The Split Load Phase Inverter Revisited

The split load phase inverter (SLPI) has been the matter of several discussions in the Xpress mail section of *audioXpress*: Testing Impedance by C. Paul in the April '09 issue and Impedance Dispute by R. Mallory in the August '09 issue are two examples. Both instances cite the *Radiotron Designers Handbook* 4th edition (*RDH*), which is a superb reference. The following discussion will be involved with the feedback aspects of the SLPI and how they provide the unique properties of the circuit.

But first, a quick review of gain calculation for a triode and the feedback calculations.

Low frequency gain, Ao, equals -u*Rl/(rp+Rl) where u = gm*rp and Rl is the plate load resistance plus any unbypassed resistance in the plate current path. The negative sign indicates that the plate output is inverted with respect to the grid input.

Feedback gain, A', equals Ao/(1-B*Ao) where the feedback factor, B, equals the ratio of feedback voltage to the output voltage of the amplifier and is negative for negative feedback. For the balanced SLPI one half of the total output is fed back so B is –0.50.

Figure 1 illustrates the typical SLPI and is labeled to conform to the illustration on page 329 of the *RDH*, figure 7.25. Assuming one section of a 6SN7GTB the parameters are

rp = 7700Ω
gm = 2600 S
u = 20

Setting Rk = Rl = 10kΩ and B+ = 300V DC and Rl'= Rl + Rk, a load line may be drawn on the plate characteristics from 15mA Ip to 300V Eb and used to select a bias voltage, in this case –6V DC resulting in an Ip of 7mA. Then Ao = u(Rk+Rl)/(rp + Rl +Rk) = 14.44. Now for the feedback part. Referring to **Fig. 1**, the feedback is developed by Rk, since Rk = Rl then the feedback factor, B, equals -0.5, so A' = Ao/(1+ B*Ao) or 14.44/(1 + 7.22) = 1.7567. The gain is split between Rk and Rl so that Ak = 0.87835 as does Ap. Note that Ek = 70V DC, Ep = 230V DC, and E bias = 64V DC. These are the conditions for the standard balanced SLPI, which will be called case 1.

For case 2, Rl will be loaded so that the value of Rl = 7kΩ, then B = Rk/(Rk+Rl) = 10k/17k = -0.58824. Ao = 20*17k/(7.7k + 17k) = 13.7652. A' = 13.7652/(1+ 8.0914) = 1.514. Ak = 0.89035 and Ap = 0.62365; note that Ak has barely changed while Ap has lost 0.2547 or 29% of its gain while the change Ip = Ik is very small. Looks like rp has become rather large when looking at the plate circuit.

Case 3 will have Rk loaded to 7kΩ. Then B = -0.411765. Ao = 13.7652 (same as case 2). A' = 13.7652/(1 + 5.66803) = 2.06436. Ak = 0.85003 and Ap = 1.21433. In this case Ap has increased by 0.3359, 38%, while Ak has changed by very little and Ip has increased by 38%. This indicates a low value of rp when looking at the plate.

Case 4 is the combination of 2 and 3, both Rk and Rl are loaded to a value of 7kΩ. As the loads are balanced B = -0.50, Ao = 20* 14k/(7.7k + 14k) = 11.05991. A' = 11.05991/(1 + 5.52995) = 1.69372. Ak = Ap = 0.84686, Ip has increased from case 1 by 37% while the gain has only changed by 0.03154, 3.6%, indicating a low rp for the cathode and plate outputs, 972Ω. Using the initial values in equation 31 on page 330 of the *RDH* a value of 843Ω for rp is obtained—a decent correlation.

DETERMINING RP

Traditionally rp is determined by maintaining a fixed bias voltage and varying the plate voltage by a small percent and measuring the resulting change in Ip; then rp equals the change in Eb divided by the change in Ip. In this case the change in Rl or Rk will cause a change in gain and a change in Ip. Assuming a constant input signal of 1V then the gain equals the output voltage. Comparing these changes to the values of case 1, the reference will provide the information needed to determine rp as was done for case 4. For case 2 then the plate voltage change is 0.2547V, the plate current is 0.62356/7000Ω = 8.908 * 10^{-5}A. The plate current of case 1 is 8.7835*10^{-5}, the difference is then 1.245*10^{-6} and rp = .2547/1.245*10^{-6} = 204,578Ω. Using the values of Rk and u of case 2 in equation 30 on page 330 of the *RDH*, a value of 190,000Ω is obtained. Closer agreement would be obtained for smaller variations of Rl or Rk, however the effects of asymmetrical loading of SLPI would not be as evident.

Equation 34A on page 330 of the *RDH* seems to be a bit confusing as to why and what is means. Taking the original equation, Rp*Rl/(rp + Rl*(u+2)) which may be simplified, using the 6SN7GTB parameters, by;

1. u + 2 = 22 then gm'= 22/7700Ω = 2857*10^{-6} and gm'*rp may be subbed for u + 2.
2. That allows the rp's to cancel then the equation becomes Rl/ (1 + gm'Rl) which looks like the equation for rp for a cathode follower (rp'= rp/(1 + u) with Rl subbed for rp.
3. Evaluating the equation of 2 above then Ro = 10000/(1 + 2857*10^{-6}*10000) = 338Ω.

As the grid input, u, and the cathode circuitry of the SLPI are the source of both Ip and Ik then the circuit acts like a cathode follower with a bunch of plate baggage.

CONCLUSIONS

1. The SLPI is a feedback amplifier where both the cathode load and the plate load affect the feedback parameters.
2. Loading effects cannot be measured

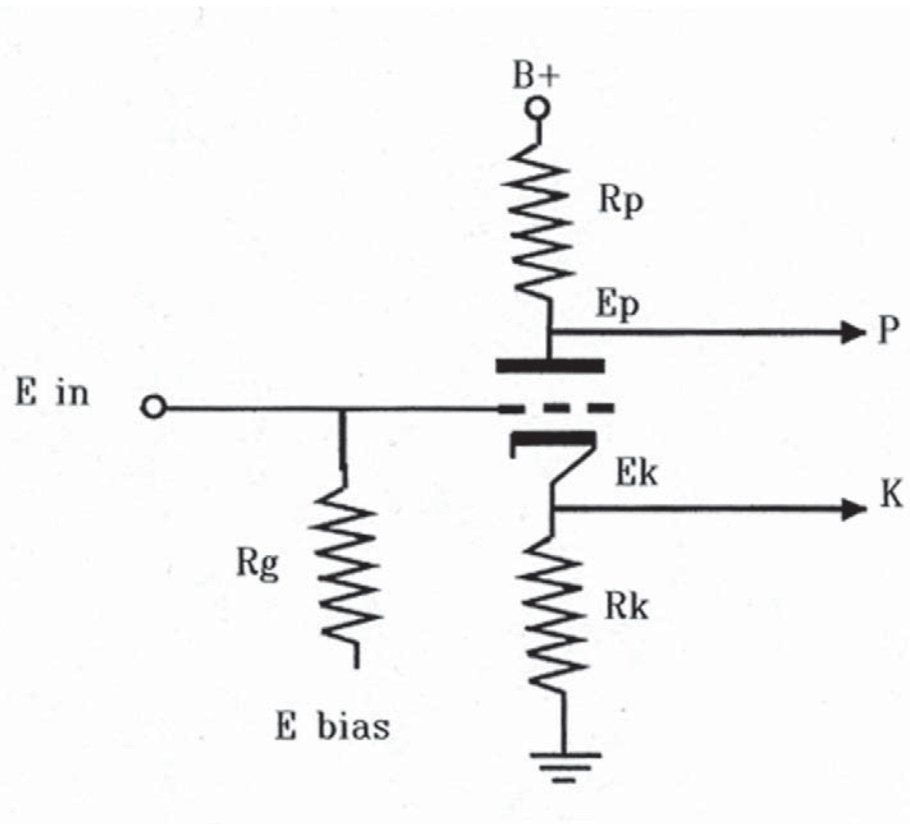

FIGURE 1: Typical split load phase inverter.

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accurately by loading only the cathode or plate circuits, see 1 above.
3. Cathode and plate loads must be equal whether resistive and or reactive.
4. One percent or better resistors should be used as Rk and Rl as well as following input resistors.

A link to 6SN7GTB datasheets is given below; of note is the Average Transfer Characteristics showing rp, gm, and u as a function of plate current. Due to the rapid increase in rp and the decrease in gm below an Ip of 5mA, I recommend that loads and bias combinations causing an Ip less than 5mA not be used. The *RDH* recommends that loads not exceed 2 times rp.

http://tubedata.milbert.com/sheets/137/6/6SN7GTA.pdf *aX*

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CONTRIBUTORS

Karin Preeda ("A Push-Pull 7591 Power Amp," p. 6) is currently working at Celestica Thailand as a chief engineer of product and test development department. He enjoys building tube amplifiers and playing bass guitar.

Claude Goeuriot ("A Power Sine Generator for Turntables," p. 10) resides in France.

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Cornelius Morton ("The Split Load Phase Inverter Revisited," p. 20) worked in the military electronics field for 42 years, primarily with surveillance radar systems. He became interested in audio around 1958 and has been enjoying the audio field ever since.

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PREAMP CORRECTIONS

I just received the August issue of *audioXpress* and I really enjoyed reading it. Thanks for the good and interesting publication!

There is a small glitch I would like to point out: In the preamp article ("The Metz Preamp") on page 10 the headphone amplifier circuit in the lower left corner, has a few mistakes:

1. The part numbers of the output devices do not correspond to the part list
2. The symbols of the output devices are actually not correct—the upper device is a n-channel device, where the little arrow should be pointing inwards, not outwards. The same with the lower device; it's a p-channel, so the arrow should be pointing outwards. The gate/drain/source connections appear to be correct, however.

Another small point: given the part numbers of the output devices, it appears that these are in a DPak package, which is an SMD package. Given the power dissipation of 0.6W at idle (15V × 0.04A), I suggest using IPaks or TO220 with a little heatsink to improve reliability.

Alfred Hesener
hesenera@netscape.net

Reinhard Metz responds:

Thank you for your interest in the preamp article. You are correct on all counts—there was a change in the headphone amp FET selection and the parts list is correct—the change was not picked up on the schematic. The N and P channel symbols are indeed reversed in the schematic, while the Q1/Q2 numbers do correctly correspond to the parts list. Q1 is the N channel and Q2 is the P channel.

Concerning the idle dissipation, I agree it may be a reliability improvement to use TO220 versions of Q1 and Q2, although I have found on my prototypes that the ground plane does a decent job of spreading the heat. Also, if a builder uses a signal generator and oscilloscope or distortion analyzer to set bias, it is likely that a lower current than 40mA will achieve a satisfactory bias point, thereby also decreasing the dissipation somewhat.

BOOK REVIEW

I would like to comment on the review of my book *Current-Driving of Loudspeakers*, published in the July issue (p. 18). While it was for the most part rather accurate and relevant in what it discussed, the most important findings of the book, dealing with the flaws of voltage drive, were not really considered. Also, there appeared some notions and interpretations in the review that deserve amendment. Some excerpts from the review:

"*According to conventional wisdom, there are two very good reasons why voltage drive is used universally: loudspeaker resonance and damping.*"

Here, I have to wonder about the words "two" and "very good." The damping of a loudspeaker's resonance is a single well-defined task to be performed; and once done in the frequency domain, the time domain will automatically settle in place since the time behavior of a linear system is always only a reflection of its frequency behavior, as determined by the Fourier transform relationship.

The back-EMF can really be called back-EMF only near the resonant frequency where this voltage acts approximately in phase with the applied signal, thus reducing the flow of current on voltage drive and hence affecting the damping. However, it can be shown by basic modeling and measurement that when frequency rises from the resonance area, the EMF soon turns perpendicular to the current and at the same time decreases in magnitude, falling below the resistive voltage somewhere between 100 and 200Hz. Thus, throughout the whole mid-frequency region, the EMF, which is now a perpendicular EMF, does not damp or control anything but acts merely as an uncontrolled interference source, playing havoc with the crucial voltage-to-current conversion.

So, electrical damping is nonexistent beyond an octave or two from the resonance; and in the resonance region it can be substituted in all aspects by mechanical damping with the same outcome. Thus, there is not any valid reason, let alone very good ones, to perform against the clear directive of the governing law $F = Bli$, especially in the middle and treble regions. It is also fully feasible to use some amount of passive electrical damping for bass and increase the impedance level for other frequencies, where the benefits of current-drive mostly appear.

"*In other words, implementation of these ideas requires a system approach, as can be done effectively with powered loudspeaker products.*"

In principle, there is dualism between the two modes of operation, and current-drive does not necessitate system approach more than voltage drive does. As I have demonstrated, passive speakers with flat-response amplifiers can also be made to work. With dedicated amplification for each driver, the source impedance can be kept more ideal, but just the same kind of restrictions apply when one is striving for ideal voltage drive.

"*Most of his objections to voltage drive stem from the back-EMF...*"

Here, we come to the main omission of the review. The objections to voltage drive are only halfway due to the motional EMF. The effects of the inductance EMF are at least as grave, but this major issue was not even given a mention. (The voltage across any inductance, in this case that of the voice coil, is also an EMF, as it is induced by a fluctuating magnetic field.) The most important chapter of the book is #4, where the flaws of voltage drive are demonstrated; and without an idea of these phenomena, readers are not able to make any reasoned judgments on the subject. Trying to reproduce some of these effects would have been more useful and revealing than just focusing all attention on the bass region damping, as the loudspeakers could not be modified.

"*What we call current drive is, in reality, a voltage source in which the voltage is automatically and instantaneously adjusted up or down, to maintain the desired current.*"

So it can be said if we are using a usual voltage mode output stage. It is also possible to employ a current mode output stage, in which case we are adjusting only current and can also achieve higher output impedances.

"*It seems that the reverse current generated by spurious cone movement "distorted" the voltage drive net current in such a way as to correct some of the distortion.*"

This is due to the partial velocity feedback effect that occurs within an octave or so from the resonant point but is ineffective at other frequencies, where the EMF current is smaller and perpendicular to the applied signal.

About voltage clipping and output transistor damages

For a given nominal impedance level, current-drive asks somewhat more voltage but correspondingly less current than voltage drive. To compensate this shift, it may be appropriate to use 4Ω drivers.

As for damages to the output transistors, I have numerous times forgotten to connect the load when playing or testing current amps, and never had such faults. Proper protection diodes to the supply rails are usually sufficient.

In current-drive, one doesn't need to beware of shorting the speaker leads either; and driver protection also becomes easier and better controlled because it is possible to use power transistors as the protective devices.

The pulse tests

(Im)pulse responses, in general, do not render useful information as to the qualities of the driving modes since the impulse response of any linear system contains only the same information as the frequency response but in a different form. As I have tried to emphasize in the book, the frequency behavior and time behavior are not separate things but one and the same thing only viewed differently. Therefore, if we have a current-drive speaker that exhibits the same frequency response (including phase) as some voltage drive speaker, their time responses (for a given input) are also inevitably identical, as dictated by the Fourier theorem; so in this regard one cannot be better than the other.

When interpreting oscillations, you should always be mindful of what frequencies are dealt with. In Fig. 7 in the review (KLH pulse), you can observe periodicity of some 0.5ms, which corresponds to the 2kHz response peak appearing in Figs. 5 and 6. In Fig. 8 (SEAS pulse), there occurs quite pure 5kHz oscillation that stems directly from the sharp cone break-up peak seen in Figs. 3 and 4. In Fig. 9, the oscillations are above 10kHz, the corresponding response peak being left beyond the scale of Figs. 1 and 2.

In either case, the oscillations reflect the high-frequency prominences occurring in the respective frequency plots but are unrelated to the Q-value or any bass damping properties of the systems.

The step tests

The same principles also hold for the step responses shown in Figs. 10 and 11; the time domain properties are a consequence of the frequency domain properties rather than the technology used. In Fig. 11, you can discern a period of about 30ms, which is attributable to the 35Hz peaking found in Fig. 4. The current-drive amp cannot be responsible for any damping here; it is the driver-enclosure mechanical resistance that retards the movement but is only somewhat insufficient.

The acoustic phase

In a 2nd-order high-pass system, like a woofer in a sealed enclosure, the phase shift approaches zero at high frequencies, rises to 90° at the resonant frequency, and levels off to 180° below the resonance. In Fig. 12, the excess phase lead in addition to that must be due to the mike's phase response, when approaching or passing its lower frequency limit. Anyway, the result also shows that low frequencies are reproduced *in advance of* higher ones, at least in terms of phase, and not after as is commonly thought.

The bass distortion

Concerning Fig. 13, lowering the mechanical Q is indeed key to control this type of distortion; but it is also noteworthy that the driver was mounted in a very large enclosure (I was told 95 ltr) and without proper stuffing, so the system is operating at the mercy of the driver's own nonlinear spring force and damping force. Both of these can be linearized considerably by a small enough cabinet and effective use of damping material (and driver optimization).

As a general note, to get a proper picture of the subject, we must be able to widen the perspective outside the bass response, that can always be tailored according to one's preferences, and look squarely at the manifold interference and nonlinearity mechanisms, that reach even to the 10 percent range and corrupt the flow of current in the middle and treble regions.

There has not happened any major development in loudspeaker technology in our lifetimes, and even the numbers of enthusiasts have undergone a downward trend. Wouldn't it be a high time to do something about this; the more so when there are very definite and concrete benefits to be gained!

Esa Meriläinen
www.current-drive.info
aX

JBL LSR6325 Biamplified Studio Monitor

By Tom Nousaine

JBL Professional
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- 1″ tweeter with titanium composite high-frequency transducer
- 5.25″ woofer with tempered paper cone, cast aluminum basket, and butyl rubber surround
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- Low-high frequency crossover: 2.3kHz 4th-order electroacoustic Linkwitz-Riley
- LSR Linear Spatial Reference technology ensures neutral off-axis response
- User-selectable boundary compensation
- Front panel volume and power controls.

Dimensions: 6.8″ W × 10.6″ H × 9.5″ D
Weight: 7.7kg (17 lb)

As far back as the late 60s, I was making my own speakers, first as kits and then later designing my own. That all lasted up until the mid-90s when I tested the Paradigm Active 20 and found that it performed far better than any speakers that I had built myself. In thinking about how that could have happened, I concluded that there were a couple reasons.

First, because I was an amateur, I was left with designing speakers with drivers around printed specifications, and the drivers I bought sometimes didn't closely match the specifications, so I was left with passive component crossover and equalization. So while my own speakers were often better than commercial products, they didn't match the performance of these active speakers. Said another way, the Paradigm Active 20 had drivers that were delivered to them or built in-house to given parameters that were part of the original design.

Second, performance could be optimized by active crossover and equalization that was much more precise than passive components (which meant that gain was actually possible—think about it: passive components can only "cut"), and the speaker could be optimally powered, meaning that the amount of power could be exactly that required—neither too much or too little for each driver. Over the years I've tested quite a few active monitors, but only a few of them were as good as the Paradigm Active 20, which was discontinued over a decade ago.

FEATURES

The JBL 6325p is one that is arguably even slightly better than that speaker. So why aren't they all this good? Well, like all products, many are built to cost, cosmetic, or other limiting factors. Some don't have good dynamic characteristics at low frequencies, and some can't be optimized with user controls (I tested one with a series of tweeter level controls that just couldn't be set optimally—in other words, at default the tweeter output wasn't perfectly aligned with the woofer and no tweeter level control setting would produce a flat response. Why? I just don't know). Finally, many are just too expensive.

Most active speakers are professionally oriented. The JBL certainly is sold as a studio monitor speaker, as are its bigger brothers. So why are active speakers nearly

universally professional? Active speakers haven't been successful in the consumer market because power is relatively cheap these days and every receiver comes with five or seven channels of power. Customers don't see the need for buying power twice.

Furthermore, cosmetics are much more important to consumers than performance. Also, for professionals, active speakers tend to conserve valuable rack space. That is, you don't need to provide shelf space for five or seven channels of power with active speakers, so they are naturally more suitable for that environment. And there is less need for piano gloss finish.

Of course, you know that I have ignored the modern "subwoofer" market where practically all products are active in that they have an electronic crossover, filters of one type or another, and an internal power amplifier.

How about price? When I first purchased the LSR6325p the list price was $400, but you could buy mail order for $309. Today the MSRP is $525, but I've seen them advertised for $419. Not exactly cheap, but given modern dollar value pretty reasonable.

Before I get to performance, consider the speaker's faults. Nothing is ever perfect. What's wrong with the LSR6325p? Well, it has "professional" cosmetics: A dark gray matte finish with the tweeter recessed into a waveguide with no grille; thus you must be careful with handling to avoid denting the tweeter. Because it's active, you need an AC outlet within 6′ or an extension cord.

Also, there's no "auto-on" function. That is, when the speaker is turned "on," it doesn't shut down when no input signal is present like modern subwoofers. That means it draws a small amount of power at idle, and if you want to shut down the entire system, you'll need to turn off each speaker individually. In my case, I just leave them on all the time.

Furthermore, while the power switch and volume control are conveniently mounted on the front panel, the inputs are recessed on the rear and moderately tough to reach. Other controls are also recessed on the back (80Hz high pass, boundary compensation, and +1.5dB and -1.5dB tweeter level adjustments at 2.3kHz) and require a small pry bar to flip small levers. The operating controls work as specified, but are small and difficult to read. Otherwise, I can't find anything to complain about.

PERFORMANCE

Is it true that speaker measurements often don't correspond to sound quality performance? Well, think about our hearing mechanism. What can we actually *hear*? Our eardrum is a small tympanic membrane, but only a small amount of sound actually contacts that and the rest of what we hear is through bone and body conduction. So the *only* thing we can actually *hear* is sound pressure. But, because we have two ears separated in space, we can also sense arrival time. Our perception of frequency, direction, and distance is actually a function of the brain interpreting arrival time and sound pressure at the two ears and body/bone conduction.

Therefore, a sound reproduction transducer will sound the most natural when it reproduces the sound contained on the program device as closely as possible to the original—not the original sound but the signal that left the microphone terminals and was modified by the production team. The way you test for that is to measure sound pressure and directivity. The more closely the sound pressure matches the test signal (usually a noise source), the more natural it will sound.

Why, then, do people "prefer" different speakers? Well, none of them is perfect, not even the JBL, and there are tons of nuisance bias sources available that influence preference. That's one thing about measurements—they don't care about preference—and they also can be imperfectly used, but at least they can't "prefer" a given source of imperfection over another.

However, there's more to it than middle level frequency response. Dynamics are also an issue. But you can measure those, too. In my case I test the low-frequency capability with the Linkwitz/Keele 6.5 cycle ramped sine signal. It's not a true sine wave, but it allows you to drive the speaker into overload without burning it out. I drive the speaker with a 1/3 octave preferred frequency until 10% distortion is reached. I use 10% because at that level of sound pressure the speaker will still sound "clean," but any further drive will cause distortion to increase exponentially as it moves into nonlinearity.

So, yes, measurements are a key factor in speaker evaluation. Because no speaker is perfect, sometimes you need to interpret the value of imperfections with listening, but the more even the frequency response on- and off-axis, the more likely the speaker will sound natural and mimic the actual source. Therefore, the most even response with regard to sound pressure will sound the most like the recording as long as the system has the requisite dynamic capability. And a noise signal with proper averaging and display will tell you which speakers will sound the most natural.

I got my first pair of JBL LSRs in June of 2005. I was really impressed and bought that pair as review samples and replaced my dipolar side surround speakers (Paradigm Active ADP-450) with them. Then as I began upgrading my bedroom system, I acquired three more units to use as left, center, and right front channel speakers in 2006 and 2007.

So my samples reported here are all 3-5 years old and allow you to see whether JBL could make the same speaker twice (sometimes manufacturers have difficulty with that) and whether they retain their new performance over time.

MEASUREMENTS

Frequency response:

- Monitor on-axis 85Hz to 20kHz ±2.3dB

Bass Limit:

- Monitor: 92dB SPL at 62Hz at 2m (<10% distortion)

Control Action:

Monitor:

- 80Hz high pass: actual frequency = 97Hz
- +1.5dB at 2.2kHz: actual response +1.3dB at 2.2kHz
- -1.5dB at 2.2kHz: actual response -1.6dB at 2.2kHz
- Boundary compensation: begins at 500Hz with -2dB at 150Hz and -4.5dB at 62Hz

I used the figure of merit 10% distortion because operating characteristics of drivers (using DLC Design DUMAX) show that when a speaker has reached the end of its linear operating range (BL product has fallen to 70% of the rest position value or the suspension compliance has increased by a factor of 4), the unit will still sound clean, but distortion increases exponentially with further drive.

The early (2005) LSR6325P Studio Monitor has incredibly smooth frequency response (**Fig. 1**, which actually shows two curves. The first was taken when the speaker was new (2005) and then five years later). The speaker has retained as-new performance for at least five years. The 2005 pair were the flattest loudspeakers I ever tested.

The early (2005) speaker had excellent

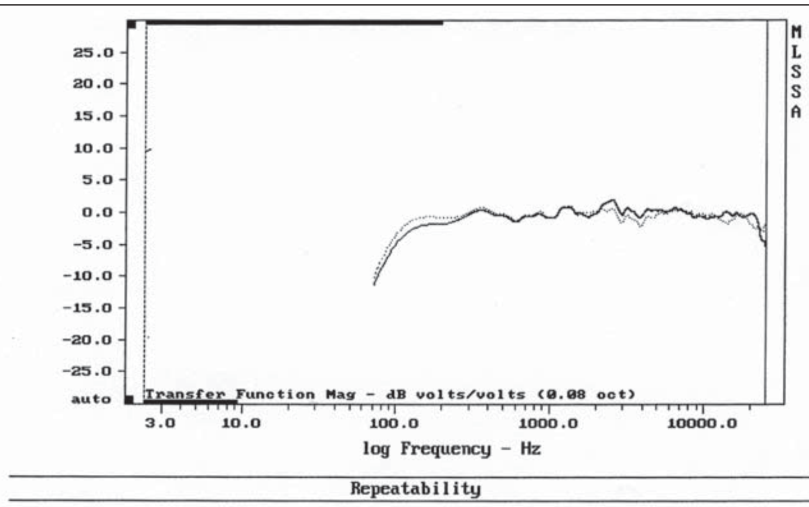

FIGURE 1: Original pair of LSR6325p taken five years apart. There is no significant wear or other anomalies with five years of use.

off-axis directivity (**Figs. 2** and **3**). This is the best off-axis directivity of any speaker I've ever tested above 60Hz. This LSR6325P monitor has better performance than nearly every consumer loudspeaker out of the hundreds I've examined over the past decade. Although the speaker, like most 5.25″ woofers, has limited low-frequency extension, the unit has relatively strong dynamic capability to 62Hz for a product in this size class. Most of the controls, except for the high-pass filter, had operating characteristics that matched the specifications closely. Boundary compensation is an EQ function that will help those who choose to use these speakers as computer monitors.

The frequency response of the newer (2007) model as measured new and three years later is shown in **Fig. 4**. You see the speakers again retained as-new performance for at least three years, but you see a problem when comparing the newer speaker to the 2005 version (**Fig. 1**). The second set of the three speakers has a 19kHz peak of about 6dB. You can also see the peak in horizontal

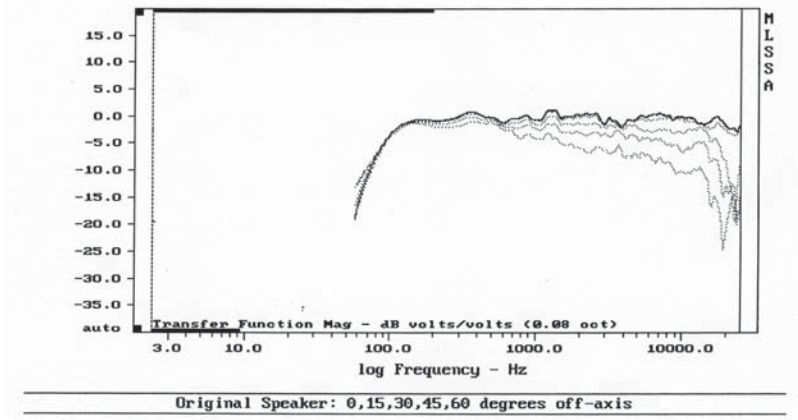

FIGURE 2: Original set of LSR6325p on-axis and with directivity plots. Notice the smooth on-axis response and even off-axis directivity.

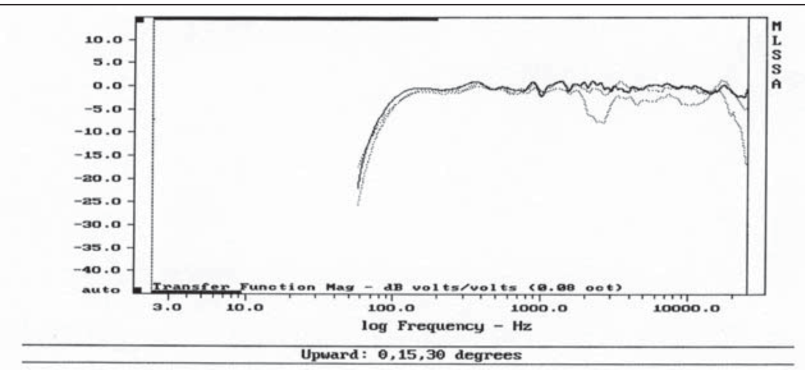

FIGURE 3: Original set of LSR6325p with vertical directivity.

directivity plots of the 2007 model (**Fig. 5**).

I believe this anomaly is virtually inaudible because there are no musical recordings with significant signal content at this high a frequency. In other words, the later ones sound the same to me as the first set. On the other hand, it was disappointing that JBL did not indicate the measured performance of the speaker that changed over time in any of the documents associated with the model.

To get a better idea what had changed in the LSR6325p over time, *audioXpress* sent the measurements in this review to JBL and they did respond to me. They stated that the only changes they have made to the speaker is a smoother, darker black finish which is now called the LSR6325p-1 and they have no explanation as to why three of my speakers have that high-frequency peak.

As this review is being submitted to print, JBL is trying to track down anything that could have gone wrong. This may be difficult to do because all of them are three to four years old. It is true that there possibly could have been individual parts that may have been out of specification for a set. And it is possible that although I bought one of the three from a different supplier, they could have been manufactured in a non-spec batch. Anyway, it doesn't seem as though JBL intentionally was selling out-of-spec products or changing parts without notification.

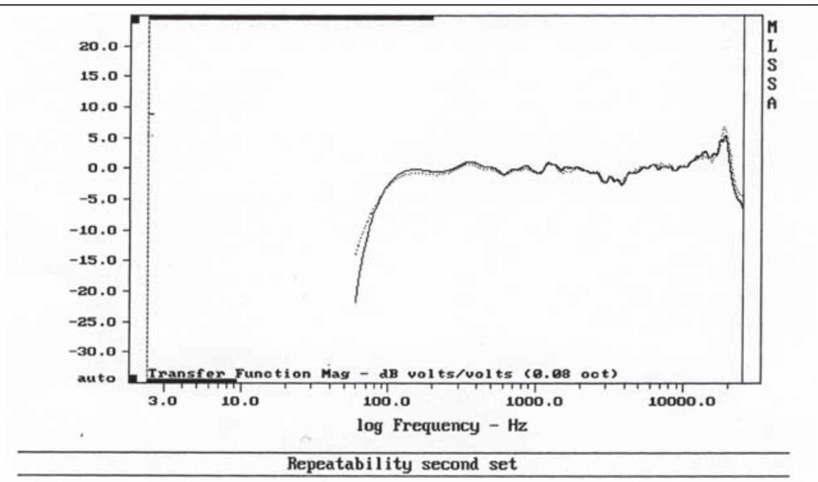

FIGURE 4: Second set LSR6325p taken three years apart. Again, no significant wear is seen. Notice the 19kHz tweeter peak.

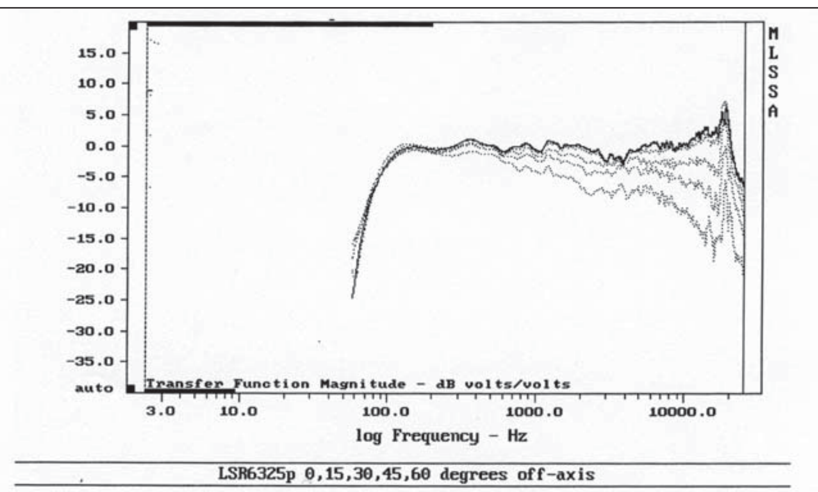

FIGURE 5: Second set LSR6325p on-axis and with horizontal directivity plots.

SUBJECTIVE SUMMARY

Listening to these speakers individually or as a stereo pair or as components in a surround system is simply natural. Highs are extended, mids are neutral, and the limited low-end sounds okay by itself but needs a subwoofer for best sound. And, best of all, the off-axis performance is smooth so reflections come back to listeners with natural tonal balance.

ADDITIONAL COMMENTS BY DAVID A. RICH

The specification sheet for the LSR6325P-1 on the JBL Pro website is very detailed. Tom could not transcribe all the information on the drivers and crossovers in this review. Many measurements for distortion and frequency response are also presented in the LSR6325P-1 datasheet including the so-called Spin-o-rama frequency response data set. They call it Spin-o-rama because 70 measurements are made in 10° intervals around the vertical and horizontal orbits of the speaker. Each of the six curves of the Spin-o-rama frequency response data set are produced from different averages of a select number of the 70 measurements. This data set is Harman Labs enhancement of the work done by Floyd Toole at the Canadian NRC.

A detailed understanding of the Spin-o-rama frequency response measurement and how they guided the design of the LSR6325P can be found in many AES papers by Dr. Toole and other members of Harman Labs as well as Dr. Toole's book, *Sound Reproduction* (www.amazon.com). You can get the information for free by searching for Dr. Toole's online paper "Making a good loudspeaker—Imaging, space and great sound in rooms" (www.infinitysystems.com/home/technology/whitepapers/inf-rooms_2.pdf). JBL Pro calls the design process Linear Spatial Reference technology.

Unfortunately, none of the Spin-o-rama or distortion graphs are presented in the datasheets for the Harman consumer products. Some other companies producing professional studio monitors also provide significant information on the design process they use in addition to providing a significant amount of measurements that are absent in the consumer world. A spec sheet for a Genelec product is a good example (www.genelec.com).

MANUFACTURER'S COMMENT:

We are looking into the tweeter anomaly in some of the samples of the LSR6325p tested. ***rr***

By Bill Reeve

Ground Loop Basics

Running an MP3 player through your car stereo can teach you a lot about ground loops.

Audio enthusiasts commonly encounter "ground loops," but often have difficulty recognizing and eliminating them. This article uses a real-world example (charging an MP3 player while it's playing through a car stereo) to provide an intuitive understanding of conducted ground loops and how to eliminate the noise they cause.

NOISY MUSIC

To experience this ground loop, play music from an MP3 player through your car stereo's auxiliary, or AUX, jack (usually a 3.5mm stereo jack). The sound should be clear and clean. Pause the music, and turn the car stereo volume all the way up. The speakers should be silent. Immediately turn the volume back down. Do not play music with the volume all the way up; you could blow out your car's speakers.

With the volume back down at normal levels, plug the MP3 player into a charger powered by the car's 12V DC accessory power jack (the "cigarette lighter" jack). An annoying, pulsating, and hissing sound should be superimposed on the music coming from the car's speakers. To hear this noise very clearly, pause the music, turn the car stereo volume all the way up, and then plug and unplug the charger from the accessory jack. The noise appears only when the MP3 player is charging. Again, immediately turn the volume back down before doing anything else.

Figure 1 shows plots of 1kHz sine waves generated by an MP3 player and measured at the power amplifier output. The top curve shows the nominal audio signal, and the lower plot shows the effect of ground loop noise when the MP3 player is charging.

This pulsating hiss is the result of charging currents flowing back to the car battery through the MP3 player's audio cable—a "ground loop." It doesn't matter whether the charger is a switching converter or a linear regulator, because the noise is caused by the battery-charging circuitry in the MP3 player itself. The noise texture will change as the charging circuit changes modes (the noise is worse when the MP3 player's battery is low).

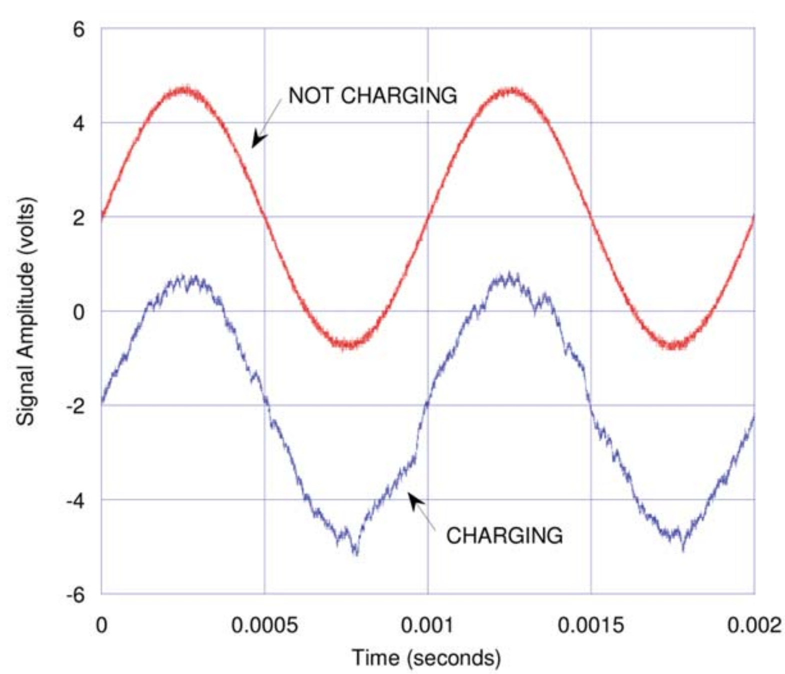

FIGURE 1: Typical ground loop noise caused by charging an MP3 player while playing it through a car stereo.

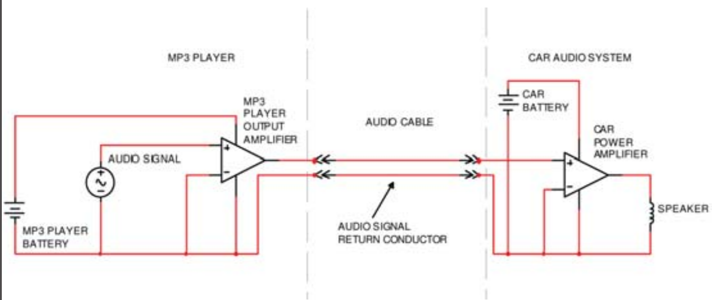

FIGURE 2: MP3 player connected to a car's sound system. There is only one return connection (the audio cable return conductor) between the two systems, and therefore, no ground loop.

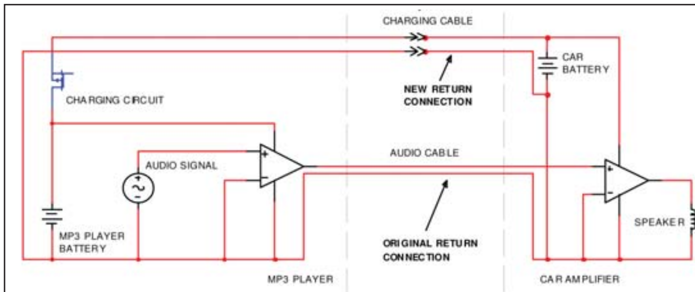

FIGURE 3: MP3 player connected to a car's sound system and charging from the car battery. Note that there are now two connections between the MP3 player and car battery returns—both the audio cable and charger cable return conductors.

CLEAN MUSIC

Figure 2 is a simplified schematic of an MP3 player connected to a car audio system. The MP3 player is powered by its own internal battery, and the car audio amplifier is powered by the car battery. Only the audio cable connects the two systems, and thus there is only one return connection (the audio cable return conductor) between the two systems, and no possibility of ground loops.

An important fact to remember when diagnosing ground loops is that electric current must return to its source. For example, every bit of current that flows out of a battery's positive terminal must return to that same battery's negative terminal—no exceptions. Applying this to **Fig. 2**, you can see that current from the MP3 player's battery is powering the MP3 player's output amplifier, which drives current along the audio cable's signal conductor, through the car stereo amplifier's input. All this current, because it originated from the MP3 player's battery, must return along the audio cable's return conductor, back to the MP3 player battery's negative terminal.

Current from the car battery powers the car stereo's power amplifier. This current flows through the speakers and back to the car battery's negative terminal. Neither circuit's current is flowing through the other circuit's return paths, so there is no ground loop and no ground loop noise.

NOISE FACTORS

Now, consider the case when the MP3 player is charging from the 12V socket in the car's dashboard (**Fig. 3**). Now you have charging current from the car battery flowing into the MP3 player's battery. The MP3 player's charging circuit is pulsing car battery current into the MP3 player's battery. Every bit of this current must flow back to the car battery (where it originated).

You want this charging current to flow back to the car battery through the return conductor in the charging cable (**Fig. 4**). But it is easy to see that some of the charging current (all of which originated from the car battery) will flow back to the car battery through the audio cable's return conductor—causing noise you can hear. The return path for charging current is supposed to be through the return conductor in the charger cable (**Fig. 4**). However, current will flow through all available parallel return paths (inversely proportional to their impedances), just as it does through parallel resistors.

Some of the charging return current flows back to the car battery negative terminal through the audio signal return (**Fig. 5**). This unintended return current path is called a "ground loop." This "ground loop" charging current flows through the audio cable return wire. You know that wire has impedance (which is both resistive and inductive), and Ohm's law requires that

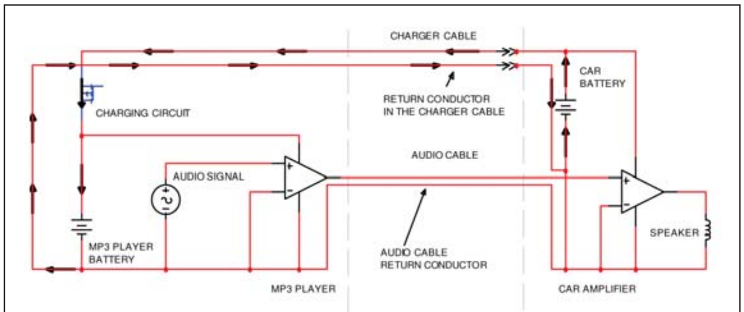

FIGURE 4: Desired charging current path from the car battery, through the MP3 player. You want this charging current to return to the car battery only through the return conductor in the charger cable.

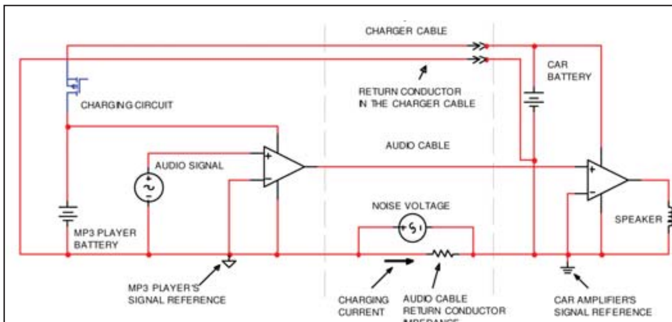

FIGURE 5: Some of the charging current returns to the car battery negative terminal through the audio cable return conductor. This current path is the "ground loop."

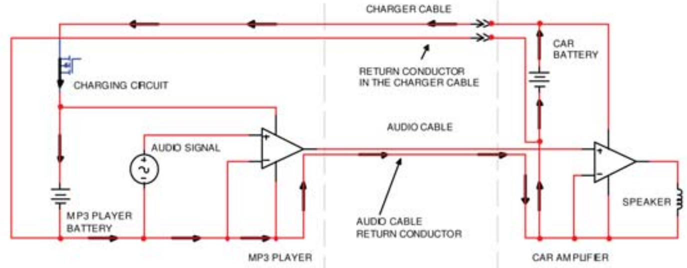

FIGURE 6: "Ground loop" charging current flowing through the audio cable return conductor impedance produces a noise voltage between the car amplifier signal reference and the MP3 player's signal reference. This noise voltage appears as hiss in the output sound.

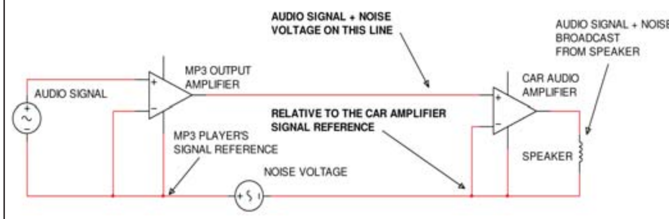

FIGURE 7: Simplified schematic showing noise voltage on the audio signal return conductor. The charging circuit is not shown for clarity.

current flowing through impedance produces voltage (V = IR).

This noise voltage (**Fig. 6**) is generated between the car's power amplifier signal reference (the car's frame) and the MP3 player's signal reference (within the MP3 player). From the car amplifier's point of view, this noise voltage adds to the audio signal voltage. You can show this clearly by simplifying **Fig. 6** in steps (**Figs. 7-9**).

Figures 7-9 are successive simplifications of **Fig. 6**. They show how, from the car stereo's point of view, the ground loop noise is added to the audio signal.

Figure 7 shows that the noise voltage moves the MP3 player's voltage reference up and down relative to the car stereo's signal reference. From the car stereo's point of view (looking back into the signal coming from the MP3 player), it sees the MP3 player's signal moving up and down relative to its own signal reference (the car stereo's signal reference). The car amplifier cannot tell what portion of the MP3 player's output is signal, and which part is noise. It sends both the audio signal and ground loop noise to the speakers.

This type of audio system is called "single-ended." High-end audio uses differential drivers that avoid this problem by sending both the signal (with its embedded noise) and the local reference (with the same embedded noise) to a differential receiver that subtracts the two voltages, recovering only the signal. If you assume the MP3 player's output amplifier is just a buffer—that all it does is lower the output impedance of the signal it's driving—you can remove it from your schematic (**Fig. 8**).

To make the picture even clearer, you simply slide the noise voltage to the left along the signal path so that it stacks up with the audio signal (**Fig. 9**). This clearly shows how, from the car stereo's point of view, the ground loop noise is added to the audio signal.

This, then, is the problem: Charging currents flowing back to the car battery through the audio cable return conductor superimpose a pulsating, hissing noise on the desired audio signal.

SOLUTIONS

An easy way to show that you understand the problem is to unplug the charger from the car's 12V power jack, and power the charger from a "standalone" 9V battery. The MP3 player will still charge, but the hissing noise will disappear. This is because charging currents can only flow back to the 9V battery through the charger cable—there are no other return paths (no "ground loops") back to the 9V battery (**Fig. 10**).

If you have a 110V outlet in your car, you can also eliminate the ground loop by using a transformer-isolated charger (**Fig. 11**). Although "standalone" charging power sources can solve the problem, another solution is to prevent any charging current from flowing back through the audio cable return conductor.

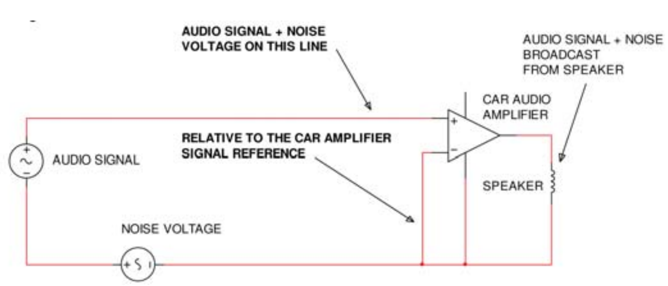

FIGURE 8: Removing the MP3 player amplifier from the schematic shows how the ground loop noise is added to the MP3's audio signal.

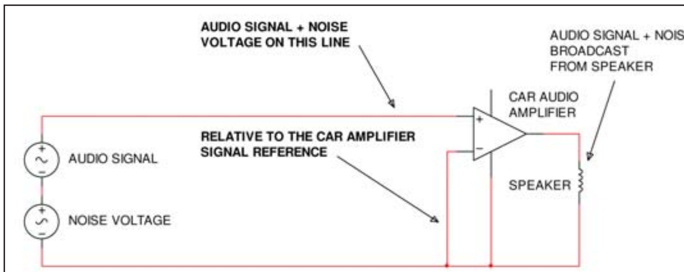

FIGURE 9: All I've done in going from Fig. 8 to Fig. 9 is slide the noise voltage to the left along the signal path to show how it stacks up with the audio signal. The car stereo amplifier sends both signal and noise to the speakers.

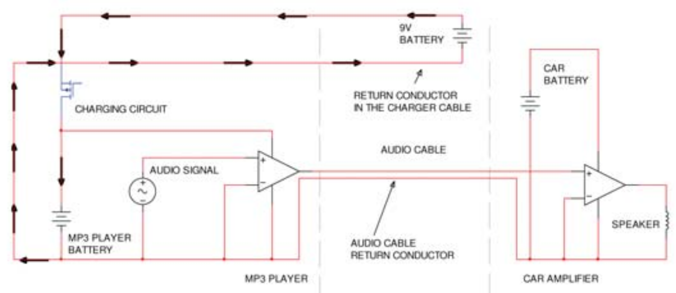

FIGURE 10: Charging the MP3 player from a standalone 9V battery eliminates the ground loop. No charging current flows through the audio cable. All the charging current that originates in the 9V battery must return to the same battery, and there is only one path—through the charger cable's return conductor.

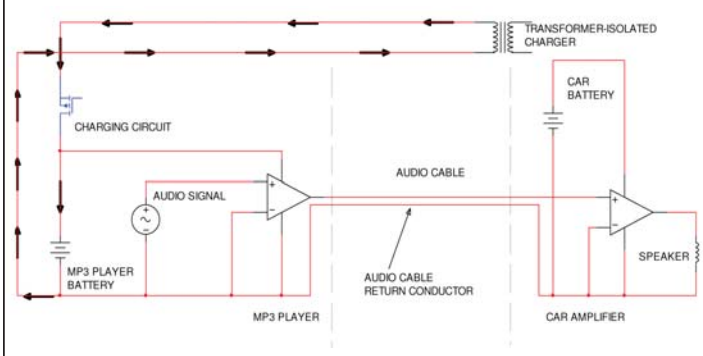

FIGURE 11: Charging the MP3 player from a transformer-isolated charger also eliminates the ground loop.

Placing an isolation transformer (available from many manufacturers) in the audio signal path "breaks the ground loop" (**Fig. 12**). This means the transformer prevents the charger return current from flowing through the audio signal return on its way back to the car battery. All the charging current must flow back to the car battery negative terminal through the charger cable's return conductor (where it was designed to flow).

However, reasonable questions to ask are: Why does the isolation transformer block the noise voltage while letting the audio voltage pass through? Because both are alternating (AC) voltages, how can the transformer tell them apart?

A transformer only "transforms" AC current that flows through its primary winding. **Figure 13** shows an AC signal being driven into a transformer's primary winding. The transformer induces an AC signal on the circuit attached to its secondary winding. But, notice that the driving AC signal must flow through the transformer's primary winding. This is how a transformer can tell signal from ground loop currents. The ground loop current cannot flow through the transformer's winding—it is only return current.

Figures 14 and **15** show the difference between the audio signal current path and the incomplete ground loop current path relative to the transformer. The signal current makes a complete circuit loop through the transformer's primary winding. However, the ground loop current cannot make a complete loop; it cannot flow through the transformer's

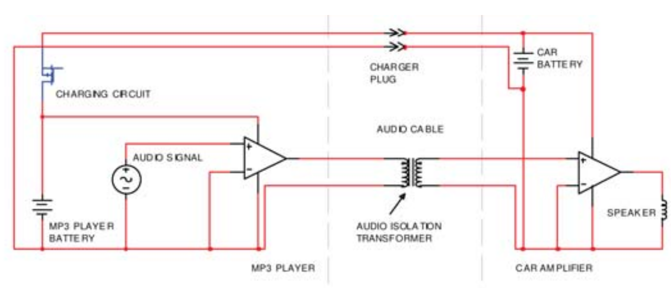

FIGURE 12: Adding an isolation transformer in the audio signal path prevents charging return currents from flowing in the audio cable return conductor. This eliminates the "ground loop" and cleans up the sound.

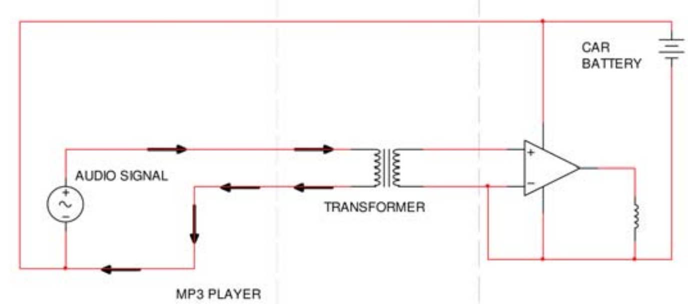

FIGURE 14: Arrows indicate the audio signal current path. Note how the audio current flows through the transformer's primary winding.

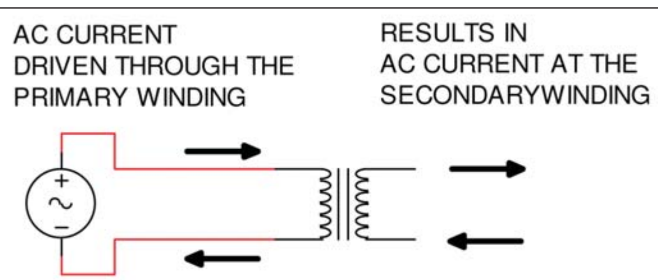

FIGURE 13: When current is driven through its primary winding, a transformer drives current through a circuit attached across its secondary winding.

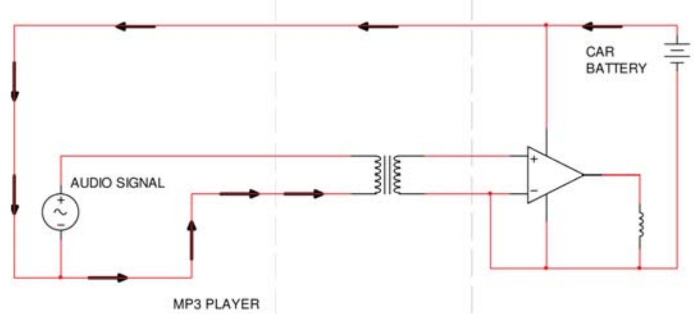

FIGURE 15: These arrows indicate the incomplete ground loop current path. Note how the ground loop currents cannot flow through the transformer's primary winding. Thus, they are blocked and cannot use this return path.

primary winding, and is thus blocked.

I've covered enough here for you to understand the basics of conducted ground loops. As you can imagine, ground loops can also involve higher frequencies and stray capacitances. For example, **Fig. 16** shows how inter-winding capacitance (capacitance between the primary and secondary windings of the isolation transformer) can form a high-frequency ground loop return path—one reason isolation transformers without internal shields are not very effective at high frequencies.

As car manufacturers appreciate this problem, they will probably build isolation into their AUX inputs. But for now, it is a classic example of a conducted ground loop.

The simple principles I have used in this example apply to all sound systems—indeed they apply to all electronic systems.

1. All current must return back to its specific, original source.
2. Current will take every available return path back to its source.
3. Stray current flowing in signal returns causes noise.
4. Be suspicious of multiple return connections between electronics.
5. Know where your currents are flowing!

So the next time you hear an unwanted hum or hiss, you can diagnose the problem by applying these basic principles and sketching simplified schematics. *aX*

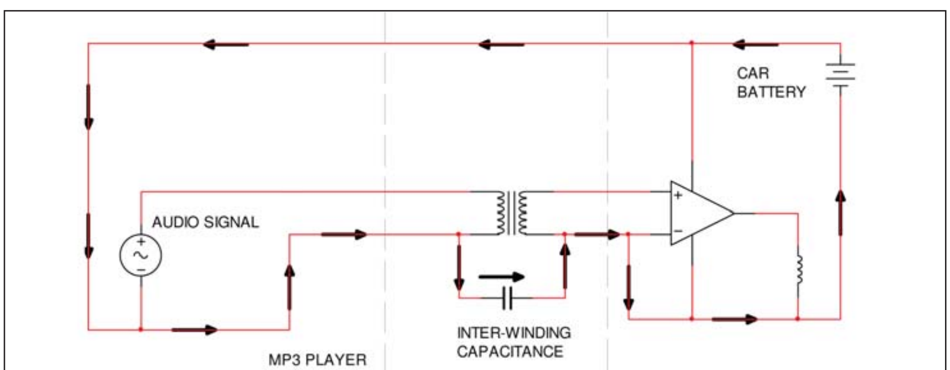

FIGURE 16: Inter-winding capacitance (capacitance between the transformer's primary and secondary windings) can allow high-frequency ground loop currents a path back to the car battery.

CONTRIBUTORS

Bill Reeve ("Ground Loop Basics") is a Director at Lockheed Martin Corporation in Palo Alto, Calif. He has graduate engineering degrees from the Colorado School of Mines and the University of California, Berkeley as well as a Masters in Electrical Engineering from Santa Clara University.

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

David A. Rich (Review: Infinity Classia C336 Floor Standing Speaker) received his MSEE from Columbia University and his Ph.D. from Polytechnic University of NYU. He specializes in the design of analog and mixed-signal integrated circuits and has taught graduate and undergraduate courses in integrated electronics and electroacoustics. Student work under his guidance, including a novel high-efficiency mixed-signal integrated power amplifier, has won numerous awards. His industrial positions include Technical Manager at Bell Laboratories. His portfolio has spanned the design of audio ICs for Air Force One to RF ICs for wireless cell phones, and his innovations have earned 14 patents. He is a Senior Member of the IEEE and has frequently served as chairperson for technical and panel sessions at IEEE conferences. He has been a member of the AES signal processing technical committee and has been Technical Editor for *Audio Critic*. He is the head of the music committee of the Bethlehem Chamber Music Society.

We visit
Aalt-Jouk van den Hul

What do grasshoppers, 300km wiring in a Hong Kong sound studio, Antonov 225 dorsal wing reinforcements, and a Chinese satellite have in common? The answer is Aalt-Jouk van den Hul, a kind of Renaissance man and audio manufacturer.

By Jan Didden

Jan Didden (JD): Dr. van den Hul, as most readers will remember, you established your reputation in the audio world with your phono tips and cartridges. How did you get involved in that?

Dr. van den Hul (VDH): Well, my first exposure to record replay was through my father, who had an early phono player, with a "stahlnadel" [steel needle—JD], and my mother let me play records when my father was at work. This was all confiscated by occupation troops during the war, so when the war was over I decided to build my own player, which I put together from a bicycle dynamo doubling as a motor, a multiplex "turntable," some "meccano" parts, and rubber bands. I even got it to rotate very close to 78rpm!

Later, when I was studying at Delft Technical University, I visited a company in Germany that cut and polished tips for pickup cartridges. They used the well-known "screwdriver blade" shape, which they were very proud of. Cautiously, I told their design engineer that, in my opinion, the blade shape was just cutting up the grooves at each playback. He didn't believe me, so I set up a demo for him.

We cut a sinusoidal groove in butter from a packet, then traced that groove with a screwdriver blade. Yes, it cut the groove to pieces, so he had to admit that his shape was not very good for the record.

They asked me whether I had a better idea. Now, at that point, I needed to be very careful. If I came up with a better idea, they would claim the idea as their own. So, I developed not one, but two, solutions. I handed one solution over to them and they immediately treated it as their own!

When they had finished their first tip, they came to me in Holland and asked me to put the tip in a cartridge. This was not a simple thing, especially when you realize that this tip was the only one in existence, and if I dropped it, the chances of finding it were pretty much zero! I had to remove the existing tip from a Philips GP400 cartridge without damaging the cantilever, fill the tip hole (which was too large) with compound, then drill a hole in that compound and mount the new tip in the right position and orientation. All through a monocular microscope.

I remember we tested it on a record from Jean-Toots Tielemans, the legendary Belgian jazz musician. It sounded fantastic! And I knew it would. I can visualize various concepts in my head, particularly how geometries affect sound reproduction, so I wasn't really surprised. The end result was that the Swiss left with that modified cartridge as well as with my record, neither of which I ever saw again!

Of course, the tip promptly became "their" tip, no longer a VDH tip! But I was so impressed with my own design and the way they manufactured it that I started to buy lots of those tips from them. That was pretty crazy because I had to pay them much, much more than the small royalty I would get from each sale! I almost went broke, but I still have those tips and I am still selling them, so it was a good investment. It also teaches you very quickly the ins and outs of doing business and how to safeguard your own interests.

JD: Did you go into technical design work when you left school?

VDH: I studied at Delft Technical University, and when I finished my studies I went into particle research at the university's particle accelerator, but at a certain point the research went into an offensive direction that I couldn't support, so I quit. After a stint in teaching, I decided to move on to the next station, and founded my own company to sell the phono stylus I had just developed. That stylus success, of course, opened up the possibility to do a lot of other things I am interested in, one of which is everything involving cabling.

After that, I never looked back and have been involved in music, in the broadest sense, ever since. Music and sound was my motivator, but I never looked at it in isolation; I always have been interested in issues surrounding audio.

JD: What was so special about that stylus tip?

VDH: It has a side rounding radius that is constant, no matter how low or high it rides in the groove. The result is that the high-frequency tracing does not depend on amplitude, which is not the case with a lot of other tip shapes (**Fig. 1**).

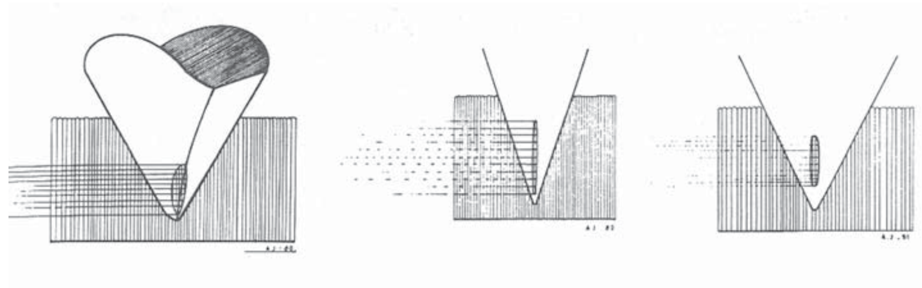

FIGURE 1: A revolutionary tip shape (from "Who Has the Perfect Tip?" by A.J. van den Hul, *TAA* 4/83 and *The LP Is Back*, www.audioXpress.com).

Now, patenting this tip was pretty hard. I had to travel to the States several times, and made presentations of slides I made from tip pictures taken through my father's microscope. But eventually I was able to convince the patent people that my tip was new and much better, and my US patent was granted.

At that point people started to notice me because the quality improvement of that tip was so evident. At that time Philips was already working on the CD, so they weren't excited about the prospect of this new tip.

I got a visit from a gentleman from the famous company who invited me to company headquarters in Eindhoven. They ushered me into a fancy restaurant and proceeded to feed me lots of very good port wine. Then they offered me 25,000 Dutch guilders (about 12,000 euro) if I would stop all my activities with these tips. I wasn't impressed by that figure and told them so, and that was the end of it.

JD: Were you also building cartridges at that time?

VDH: No, that came later. I started designing (and repairing—a great source of knowledge!) cartridges around 1987, just before the CD appeared. That was hard! Imagine winding coils with wire one-quarter the thickness of a human hair, or even less. After I taught myself to do it, I demo'd it to a group of audio journalists looking over my shoulder. Then it went fast: I started to design cartridges for Benz in Switzerland; I designed the Goldring Elite, and maybe six others. Goldring promised me one British pound per cartridge, but when the time came to pay they said they really didn't have the money. That's when I decided to build and sell my cartridges myself, and I've sold many thousands, worldwide.

JD: What's the main activity of the van den Hul Company these days?

VDH: Of course, we're still in the cartridge business. One interesting niche is providing cartridges for music companies that wish to reissue music for which they no longer have the master tapes! We then provide them with the high-quality means to regenerate the master, so to speak, from a well-preserved record.

The bulk of the company business these days is cables and cabling systems, but my own focus is still shifting. One of the things I've done recently is to develop some programs to help companies and shops that aren't doing so well. I've done a lot of work on that in the Ukraine, and apparently successfully, because I am at present an honorary citizen of the capital, Kiev. So I really enjoy going off on tangents not directly related to audio, which will always remain a strong interest and hobby!

JD: That golden key on your prize shelf is the key to the city of Kiev?

VDH: Yes. There are some other interesting items (**Photo 1**). You see this "White House" baseball cap? Worn by President George W. Bush on one of his jogging runs; I managed to hold on to it during my White House visit. Another dear memorabilia, but for other reasons of course, is a pair of bricks from the Dachau Nazi concentration camp, where my father was held. I had to put a lot of effort in getting my hands on that Concorde model (**Photo 2**), but it was worth it. I think this is one of the most beautiful planes ever made. A further development of the equally beautiful British Vulcan bomber.

JD: Getting back to audio, I notice you listen to classical music while at work.

VDH: Of course, never forget to listen to the music! That's why I spend at least a week every year in Vienna. I go to the Staatsoper or to the Musikverein. Now, the Musikverein is an interesting venue that is relatively narrow and long. Yet you hear the detailed placement and sound stage of the orchestra, whatever seat you are in.

I can easily identify the Wiener Philharmonic, the Wiener Symphonic, and the Berliner Symphonic. The Amsterdam Concertgebouw Orchestra is more difficult because they often have different guest conductors who change some of the characteristic sounds. The Berliner Symphonic, for instance, has a very constant and stable recording technique and equipment which makes their characteristic sound easily recognizable. It's also simply miked—just two or three, with possibly a few in the back of the venue to capture the acoustics.

Good sound engineering means that the orchestra and venue are clearly recognizable in the end product. But it puts a lot of responsibility on the recording engineer. He needs to have a first-class pair of ears and needs to be able to listen "through" the performance—with the entire attendant recording stress—to know how it will sound at the end of the process. He needs to have a strong image in his brain of how he wants it to be.

That's one of the strengths Decca had: Their SXL series is very high quality and very consistent, clearly recorded with the same equipment by the same people for the whole series. I don't think it has ever again been done so well, although EMI often came close. Good sound recording is not easy and we should recognize the incredible jobs some of these people do for us.

PHOTO 1: A.J. van den Hul with memorabilia and appreciations.

DGG also used quite aggressive A/D converters for their first digital recordings, which I thought were not very good. There was a lot of justified criticism but also a lot of praise for that crystal-clear digital sound, which I never understood. I mean, you've got ears, don't you? You just had to listen to know it was a step backwards from good analog. But it's extremely difficult to be objective, not to be dragged along with the mainstream, and influenced by expectations, and experiences in your youth. Your taste for music, for instance, is very strongly determined by what you are exposed to when young.

In my case it was the church organ. My father was the church organist and always played that music on the home organ. I remember that one day, early in the war, my father was playing Maarten Luther's "A strong castle is our lord" when a German military unit came to arrest him. The unit leader was also an organ player and he joined my father and they played together for a while. After they finished, the Germans left, to come back two minutes later to formally arrest my father, who, of course, was no longer there. A moment of humanity in a sea of madness that was.

JD: I remember that you used to write articles for the audio press, but haven't seen any lately. Is it something you stopped doing?

VDH: I'm not a journalist by trade; I wrote a few articles at the request of some early Dutch technical periodicals. Then at a certain point I wrote an article on a comparative test of a trio of video recorder technologies Betamax, Video 2000, and VHS. Unfortunately, my conclusions didn't align with the marketing and advertising stories. My article was changed beyond recognition, and I went into the editor's office and symbolically handed in my pencil. I stopped writing for periodicals but continued to put my thoughts in writing, as witnessed by my articles on the van den Hul website (www.vandenhul.com).

JD: Since you mentioned cables, can we talk a little about that? I remember when you came out with your carbon interlinks; I thought: Huh? How can that improve anything?

VDH: Ahhh! You see, when I worked with those tiny wires I started to become interested in what actually happens in a conductor and what can go wrong, so it was back to the study room again! The standard story is, of course, that electrons are responsible for transporting electrical current through a conductor, but that's only part of the story. It is accompanied by a modulated electromagnetic field. I visualize the process more in terms of electromagnetic fields than of electrons moving around, although both views are largely equivalent. It's just that the electromagnetic field view is easier for me.

PHOTO 2: A.J. proudly shows off his large scale Concorde model.

JD: The electromagnetic view led you to carbon interlinks?

VDH: It's hard to say how creativity works, but I remember wondering why everything always had to be from metal for low resistance, and the importance of low resistance. I started to think about designing a non-metallic cable and ended up with carbon. Not super-low resistance, but with a perfect lattice structure, much better than any metal cable could reach. No impurities, no structural defects.

So, don't look only at conduction. There's a well-known company here in Holland, Siltech (www.siltechcables.com), who advertised the fact that they dope their silver cables with 5% gold. On the face of it you might think it helps conduction. But a gold molecule has a much larger structure than a silver molecule. So by gold doping you introduce a lot of irregularities in that cable! Many of our cables are a combination of metal core with a carbon mantle. The carbon improves the conduction when there are small defects in the metal. It gives a better chemical protection and absorbs EMI better.

Here at our laboratories I have run a separate spur to the entry panel with our own hi-end mains cable. Now, you can say, how can it make a difference when I have 40 miles of ordinary wire ahead of it to the substation? But what that special cable does is filter out a lot of HF interference that is generated locally in our building from the myriad computers, test equipment, cell phones, and other gear. Filtering your mains signal and avoiding interference contribute to cleaner sound.

JD: You also use your own proprietary insulation on your cables.

VDH: Yes, I use halogen-free material in the insulation and jackets, without softening products, which we developed ourselves. That was a decision I made in 1990, to go away from PVC and make our own materials.

JD: Does that influence the performance of the cable as a conductor of electrical signals?

VDH: Yes, it does. Chemicals are always restless, always on the move. PVC migrates all over the place and eventually damages the metal structures of the conductor. Soft-makers, which you need to keep PVC supple and soft, are quite aggressive in this respect. And over time they evaporate, making the PVC shell hard. So, when you then bend or flex the cable, something has to give and very often it internally damages the conductor. Cables are needed to transport the music, but they must do that as cleanly as possible; the emotion is in the music and not in the cable!

JD: So coming back to the carbon cable, if I would measure it, would it have a rather high resistance?

VDH: About 28Ω for a meter. But that's of no consequence for an audio interconnect. Anyway, look at that [carbon] cable as a series of circular molecules of carbon and hydrogen where you remove a hydrogen atom from one position. **Figure 2** gives an impression of such a molecule. The position where the hydrogen atom is removed (light gray) becomes the position where the molecules latch together, and you get a perfectly regular structure. The difficulty in 1990 was how to manufacture such a cable.

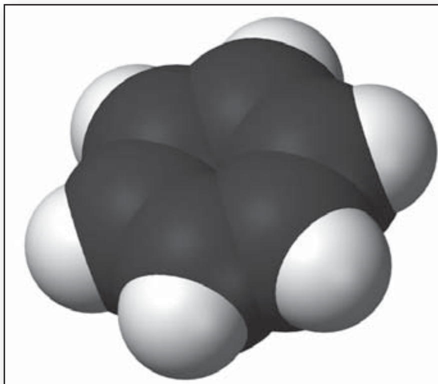

FIGURE 2: Example of a symmetrical molecule that can form highly regular lattices.

JD: Is there a clear relation between cable performance and price?

VDH: Sometimes people question the pricing of our cables, but there's always a good reason something costs more than a comparable product. For example, some cable manufacturers use the equivalent of a pair of drills, at opposite ends, to twist their cables. That is the best way to ruin the cable! It may still look good, but sound-wise it will be a disaster. I consider carbon fiber the best electrical connection you can have: very reliable, not chemically active, and extremely stable both electrically and mechanically and with a perfect internal structure for signal transfer.

JD: Dr. van den Hul, what, then, are your personal interests these days?

VDH: Actually, lots of my time goes to two areas. One area is a kind of fall-out of my carbon cables. I really like these fibers; some time ago I started a new company for developing all kinds of high-tech fibers. Not just carbon fibers, but also aramide-type fibers and even fibers made from basalt rock material!

[Points to a model of an Antonov 225 giant transport plane on his desk.] See that plane? Can you imagine the stress on whatever keeps those two wing halves together on top of the fuselage? When the plane taxis, and the wing tanks are full of kerosene, our carbon fibers keep them from falling off! No kidding! There's even a satellite in orbit that has a VDH carbon frame. Once people find out the advantages of that material, it really takes off. Selected HP and ASUS laptops have our carbon shells, as do some upscale Nokia mobile phones.

My other area of interest has more to do with ourselves as living beings. One thing I discovered was that rhythms that are prevalent in classical music often correspond with the signaling rhythms internal to the body, which make various organs work together. What most people don't know is that the heart can emit up to 136 different acoustical tones, and that organs react to those tones. So when the heart doesn't function optimally, it's not just the circulation that suffers. It has other detrimental effects on the body as well. My current interest is very much on how those internal body parts work together and communicate, and how that determines "the person" we see and experience from the outside.

I recently acquired an interesting combination of hardware and software that can analyze one manifestation of this "persona" through the electrical field a person generates [points to an oversized mouse-like object connected to a laptop]: the Human Body Field (HBF). Trying to draw conclusions from the body field is an old idea; the Russians already attempted to monitor Yuri Gagarin's body state from the electrical signals it emitted, when he—as the first human ever—circled the globe in a satellite.

The people at NES (www.nutrienergetics.com) developed the concept into powerful analysis software running on this laptop. Now, NES does not diagnose or cure illnesses and it's not a validated medical system. But, that said, I've seen some remarkable changes for the good in people whose HBF was analyzed and who were recommended certain remedies. So I find it fascinating enough to spend time on it and to delve deeper into the matter.

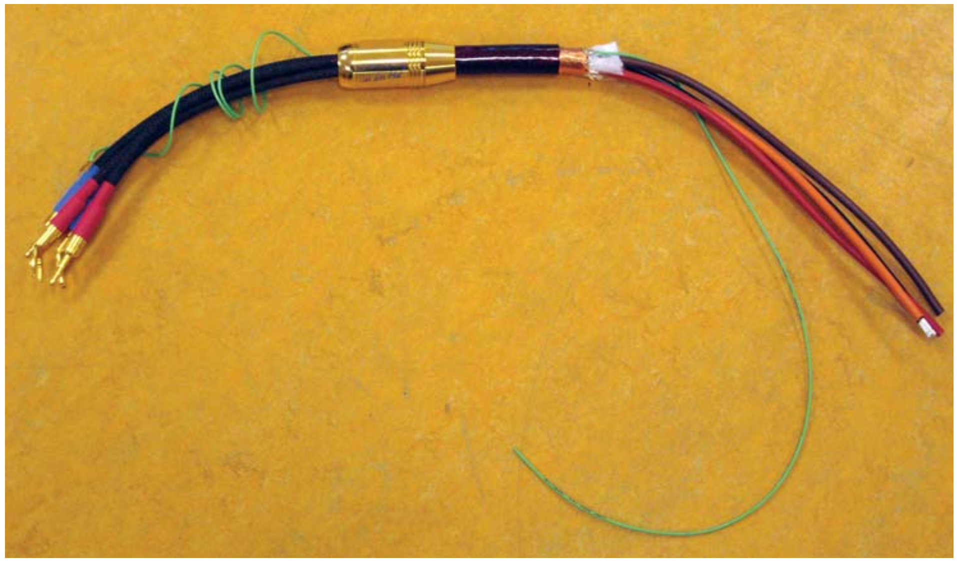

PHOTO 3: The construction of a bi-amping VDH cable.

JD: Back to audio; do you think that SACD will establish itself in time?

VDH: No, I don't think so. I think we will see the CD continue to be used for the near future and then gradually give way to downloads, either on-demand or as a purchase. Also, if the trend toward 3D video continues to grow, audio may even become just a supporting medium for video, with musical performances reproduced as a 3D holographic event supported by multichannel audio.

Interestingly, in many developing countries (and for this purpose that includes China), we see similar developments as we saw in the western world decades ago regarding hifi and hi-end audio. If you go to hi-end shows in our part of the world, most visitors are of age, while in the developing countries they are generally quite younger. If you go to hifi shows in Vietnam or Hong Kong, you get a déjà vu from what we had 30 years ago.

JD: Dr. van den Hul, thank you very much for your time and frankness. *aX*

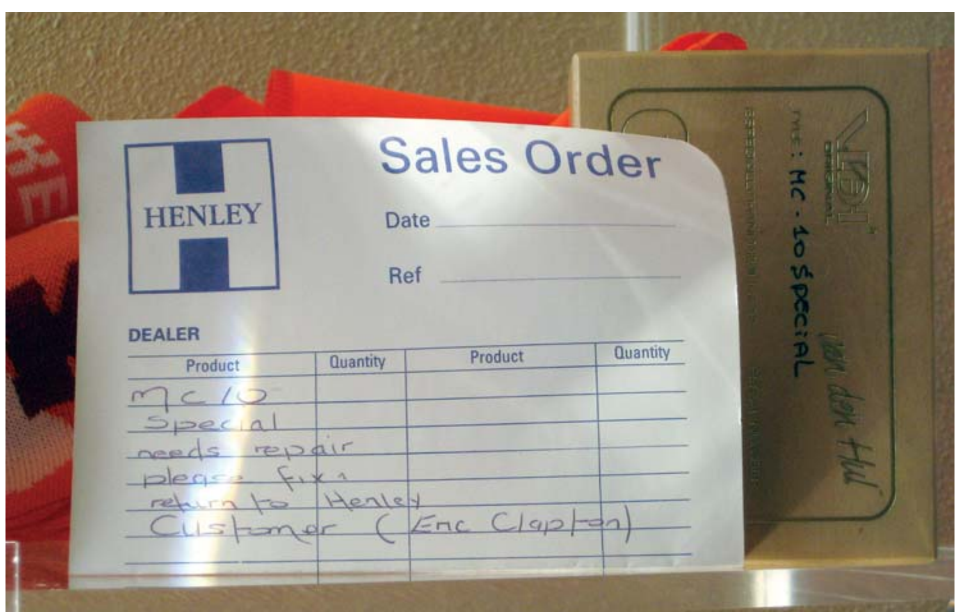

PHOTO 4: The sales order for repair of Eric Clapton's cartridge.

Infinity Classia C336 Floor Standing Speaker

By David Rich

	Infinity Division
	Harman International
	Corporate Headquarters 400 Atlantic Street Stamford, CT 06901
	www.infinitysystems.com
	800-553-3332

PRODUCT SPECIFICATIONS

- Low-frequency driver Triple 6½″ CMMD cone, stamped basket
- Midrange driver 4″ CMMD cone, stamped basket
- High-frequency driver 1″ CMMD dome
- Crossover frequencies 500Hz, 2800Hz; 24dB/octave acoustic
- Dimensions 48½″ H × 8½″ W × 10½″ D
- Finish of real wood section Cherry or high-gloss-black
- Weight 56.1 lb
- Price $899 each

Infinity has been offering popularly priced speakers designed to the Canadian National Research Council standard shortly after Dr. Floyd Toole decamped with some of his graduate students to Harman International. Soon many speakers emerged from their laboratory, including some in plain boxes in good-looking vinyl wood enclosures to be sold at very competitive prices in large-chain electronic stores under the Infinity brand.

Starting with the IL series in 2001 and evolving through the Alpha and Beta designs, Infinity had many best-buy products. I participated in a review of the $1000 IL 40 with Peter Aczel in my *Audio Critic* days, and they were a breakthrough in terms of performance for the price. Peter wrote of the IL 40: "Only when I switched to my reference was I reminded that there exists a further degree of refinement in loudspeakers."

I reviewed the Beta 40 for *Sensible Sound* in 2005. The Beta 40 was actually at a lower price of $800 per pair and was still a clear best bang for the buck. I did a deep dive in my review (Dec./Jan. 2006 Issue 108) on the design approach of Dr. Toole and his team. You can find the review online without the figures at: www.thefreelibrary.com/Infinity+Beta+40+loudspeakers.-a0139430354.

Infinity, of course, had more expensive speakers (the Prelude line) that had been designed to be the reference. Subsequently, Infinity introduced the Kappa line with its more stylish wood cabinet that split the difference in price. These more expensive lines slowly faded away, leaving the Beta as the top-dog floor-stander.

FEATURES

The Classia C336 replaces the Beta in the US, but has some relation to Kappa with the C336 curved, real wood, top and front bottom sections attached to a unique ultra thin (8.5″) wood cabinet (**Photo 1**). The wood pieces are made of layers of plywood laminated with a wood veneer. Multiple layers of polyurethane are then applied to the wood for a high-quality look. The cabinet tapers from front to rear. At 10.5″, front to rear the cabinet is not very deep. The sides of the speaker are clad in black vinyl and have vertical grooves spaced about an inch apart.

The grille frame is plastic with the older plastic plugs that fit into plastic holes to hold it in place, rather than the more modern magnets. Metal strips are at the top and bottom of the grille to enhance the high-tech look. The grille

PHOTO 1: C336.

PHOTO 2: C336 without grille.

frame shows no signs of being designed to prevent diffraction when it is in place.

With the grille cloth removed (**Photo 2**), which Infinity advises for improved sound quality, you see the exposed drivers surrounded by a black vinyl covered material. The light shape of the speaker cones against the black background is distracting and tends to draw the eyes to the speakers. I would have preferred real wood, as I have seen in other speakers in this price range, to match the top and bottom panels. The four plastic feet extend past the bottom of the speaker to stabilize what would be a wobbly design given the 55 lb tall and skinny cabinet.

The C336 has three 6.5″ woofers instead of the two found in the Beta 40. The height and weight of the speaker are deceptive. You would have expected a more voluminous cabinet relative to the Beta 40, but a Harman engineer reports that the shape and dimensions of the cabinet "when combined with extensive internal bracing and a volume partition below the bent wood decorative panel, the internal volume is approximately the same 32-liter internal volume as the Beta 40."

Infinity's proprietary CMMD cone material Infinity has been used on most of the company's designs since 2001. CMMD is a composite sandwich of ceramic alumina grown on the aluminum substrate. The aluminum core prevents the cone from shattering, while the alumina supplies strength. Infinity has a technology white paper on CMMD on its website (www.infinitysystems.com).

The surrounds of the C336 drivers are not butyl rubber surrounds or cast baskets as they are in higher-priced Harman products. Compared to the Infinity Beta, the C336 has a reduced midrange size from 5″ to 4″, although crossover points and slopes are similar. The cabinet construction and an added woofer increase the speaker's list price to $1,800 per pair. Discounts from list price are often available in authorized stores.

Infinity now has a speaker line at a lower price point than was customary for the group in the past. These Primus speakers are simple boxes in vinyl-colored wood and do not have the CMMD cones. The cabinets appear to be less well braced. I suspect the crossover is also simplified. The Primus P362 is the top-of-the-line 3-way floor-stander and has two 6.5″ woofers. It is list priced at $660 a pair.

This review has had a long gestation period because of a number of issues and it is more than a year late, which unfortunately brings us to a time that the speaker is being discontinued, although still a value. The most significant delay involved locating an engineer who could answer questions about my measurements. As you may be aware, Harman reorganized after Dr. Harman retired in 2008. I started the hunt for an engineer close to two years ago, but it was only in late February—thanks to the efforts of a new press representative whom I had identified a month earlier—that I finally received detailed responses to my questions as well as feedback on my measurements. Design engineer Emmanuel Millot saved the day. He is now based in France—no wonder he was hard to find. Not only did he provide text, but also 25 figures of simulations and measured results. In this review I will show just three because of space limitations.

As it turns out, many of my assumptions about speaker performance were wrong, so I am glad I did not publish without his guidance. I decided to proceed with the review even as the speaker is reaching its end of life because Emmanuel Millot provided some important insights into the design of these new thin, tower speakers with multiple woofers, and the measured performance of the C336 appears to reflect advances in performance over the Beta series.

The bottom end of a floor-standing tower speaker such as the C336 tends to be flatter in the upper bass than a small speaker and a subwoofer crossed over at 80Hz with the typical bass management system found in audio-video receivers. A subwoofer reproduces film sound effects well, and the small satellite speakers accurately reproduce the actors' voices and background music so the aberrations in the crossover area go unnoticed. Many music instruments produce notes in the 50Hz–150Hz range, and a floor-standing speaker is the better choice.

To get a satellite/subwoofer system to really work, you need a high-end active room EQ in both the satellite and the subwoofer channels. These high-end room EQs adjust the slope and frequency of the crossover to optimize the match of the speakers. I have achieved good results in satellite/subwoofer matching only with the expensive Anthem ARC and Lyngdorf room EQ systems. The room EQ systems in almost all AV receivers lack the computational power to do the job.

MEASUREMENTS

For my measurements, I used RPlusD, which is the latest quasi-anechoic speaker measurement software from AcoustiSoft (www.acoustisoft.com), which has much improved graphics capability over the older ETF product and can easily average data. I used a calibrated microphone and USB microphone preamp from iSEMcon in Germany that is sourced by AcoustiSoft in North America.

Figure 1 is a listening window plot taken at 1m back. I used the NRC definition of a listening window, which is an average of the on-axis, horizontal radiation measurements at ±15° and vertical radiation measurements at ±15°. Harman has a slightly different set of microphone placements to form a listening window. The NRC curve was flatter in my measurements of the C336.

I used a very fine 0.05 octave averaging to unearth details that would be smoothed over by the typical one-

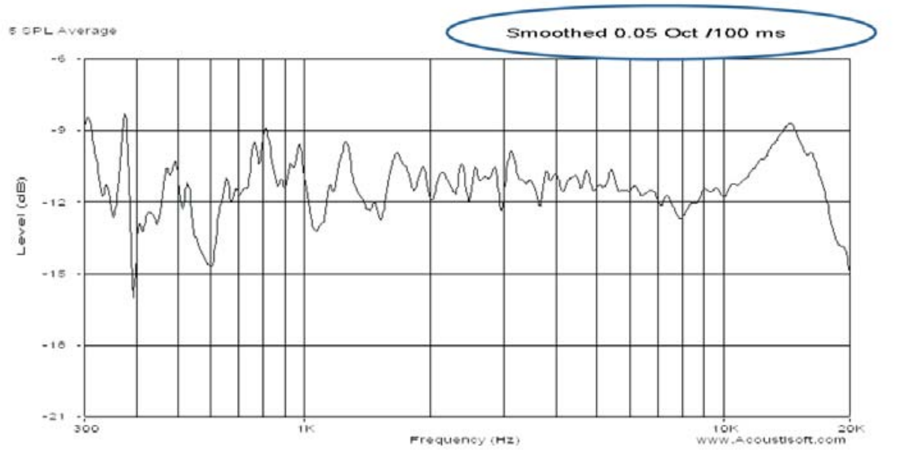

FIGURE 1: Infinity C336 NRC listening window (5pt) at 1m back. Quasi anechoic above approximately 300Hz.

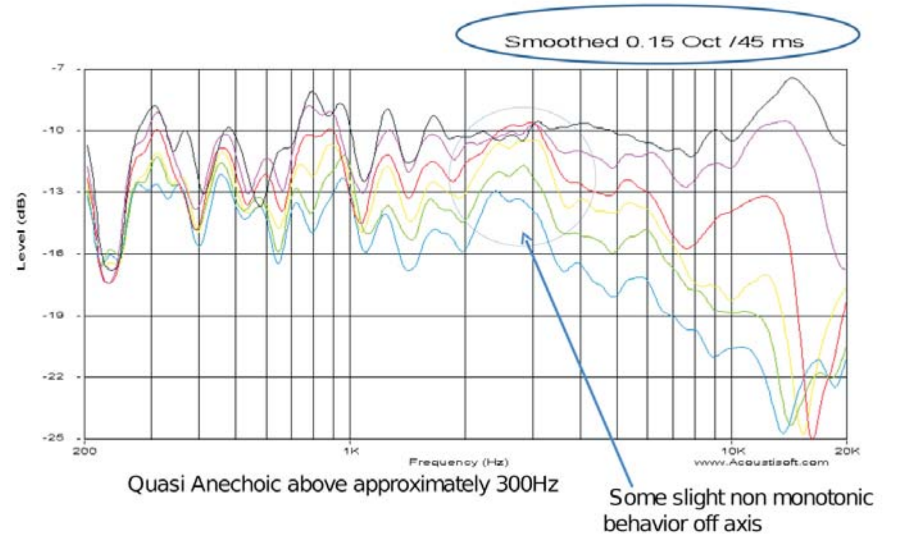

FIGURE 2: Infinity C336 horizontal radiation pattern from 0 to 75° off-axis in 15° steps at 1m back. Microphone at tweeter level.

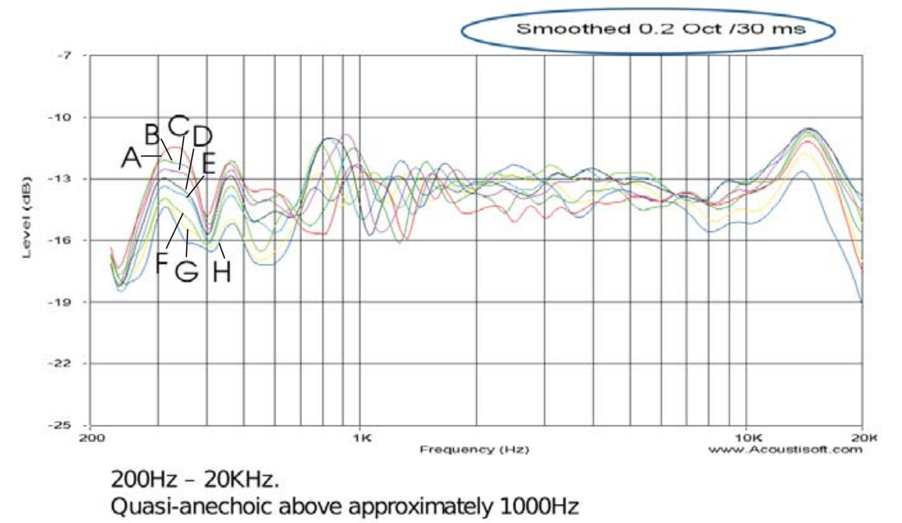

FIGURE 3: Infinity C336 vertical radiation pattern from −15 to +20° (A = 20°, B = 15°, C = 10°, D = 5°, E = 0°, F = -5°, G = -10° H = -15°) off the tweeter axis in 5° steps 1m back.

third octave averaging. The vertical axis spans only 15dB again making response variations easier to see. Most of the frequency response is within ±1.5dB even with the 0.05 octave averaging. Room effect may still be present until 400Hz, since I could not raise the heavy speaker off the floor to reduce reflections that corrupt the quasi-anechoic measurements. The dip at 600Hz is real and is discussed later.

The 15kHz tweeter resonance is not a defect since it shows up on Infinity-generated graphs I was supplied. I wanted more clarification on the resonance, but this was the one issue for which I could not get sufficient color from Harman.

Figure 2 shows the horizontal radiation of the C336. I measured it on the tweeter axis, which gives the flattest response, but this is not very critical for the C336 as highlighted in the vertical radiation patterns. **Figure 2** runs from 0 to 75° off-axis in 15° steps at 1m back. Smoothing was increased to 0.15 octaves to make the graph less noisy.

Figure 2 illustrates the hallmark of an Infinity design: The amplitude declines monotonically as the horizontal angle increases. This ensures that the room reflections in the horizontal direction (off the walls) are similar to the on-axis response. A typical speaker shows a rise in amplitude in the off-axis curves back to the flat response above the crossover frequency between the midrange and tweeter, as explained by the reduced off-axis response of the typical 4-6″ midrange at the crossover frequency being brought back to base line as the tweeter becomes active. The tweeter is normally not directional at the crossover frequency. A special waveguide on the Infinity tweeter matches the dispersion characteristic of the midrange to the tweeter at the crossover. Only between 2–2.5kHz is there non–monotonic behavior in the off-axis curves in the horizontal direction.

Dr. Toole has commented about the Harman design approach and the efficacy of monotonic off-axis radiation patterns. You can get more details by searching on the web for Dr. Toole's online paper "Making a good loudspeaker—Imaging, space and great sound in rooms."

Figure 3 is the vertical radiation pattern from -15 to +20° off the tweeter axis in 5° increments 1m back. Emmanuel Millot pointed out that the angles are double to what the listener hears 2+ meters back, so these measurements are stressing the limit of the speaker. Despite this, there is almost no activity above 1500Hz, which is an octave below the crossover to the tweeter. Harman's philosophy is to keep the off-axis response as close as possible in overall shape to a listening window response. In this case, the reflections are off the wall and ceiling. Of course, the response declines above 8kHz as the microphone becomes significantly off-axis to the tweeter. The off-axis tweeter response is also determined by the oval-shaped tweeter waveguide. With the oval shape, the horizontal and vertical radiation of the tweeter can be independently controlled.

Typical speaker designs have an energy dip around the crossover frequency due to interaction between the noncoincident speakers as different vertical angles alter the path length between the speakers. I wrote a tutorial on crossover design and driver interactions in the Dec./Jan. 2006 issue (#108) of *Sensible Sound*. This is the same issue containing the Infinity Beta 40 review.

Many listeners prefer the dip in the 2-3kHz range, which is sometimes called the BBC or Gundry dip. Some speaker manufacturers intentionally increase the size of the dip as the "voice" of the speaker by pairing certain types of drivers and crossovers. In general, some dip is unavoidable, but it is missing from the C336 as can be seen in **Fig. 3**. Depending on the Gundry dip's depth and width, the sound stage sometimes backs away from the listener and may make the speaker more

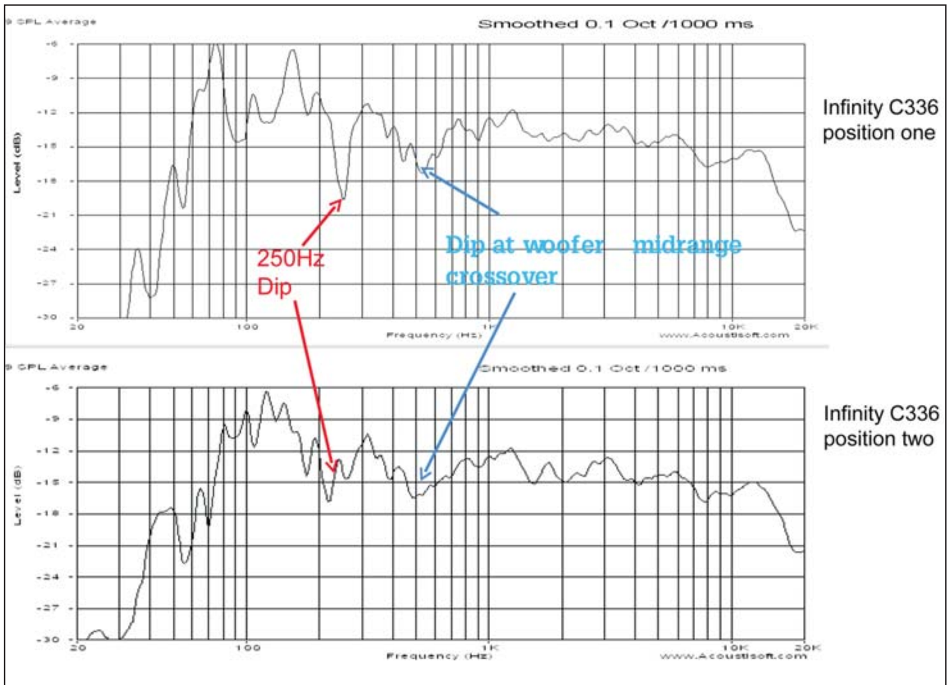

FIGURE 4: Infinity C336 in-room response at two different speaker placements.

forgiving of sub-optimal source material. Dr. Toole has named this type of voicing methodology the circle of confusion: "Listening through loudspeakers that are evaluated by recordings made with microphones that were evaluated by listening through loudspeakers."

Even the Infinity Beta 40 has a small dip due to its larger midrange, which has center further away from the center of the tweeter than the 4″ driver in the C336. But this alone cannot account for the absence of the dip in the C336. Improved engineering in the crossover also results in more desirable vertical radiation patterns. Moreover, the tweeter waveguide has evolved over time.

Examining the vertical radiation patterns around the 600Hz crossover from the woofer to midrange is more difficult because the C336 is too heavy to lift from the floor and the curves become room dependent as the microphone is lowered to get the radiation pattern in the negative direction. My attempt to converge the three woofers and the midrange at 1m will not be accurate. Moving to 2m renders the quasi-anechoic measurement useless because the reflected sound comes almost as fast as the first arrival providing no way to window it out. Accordingly, take the section of **Fig. 3** below 1kHz with a grain of salt.

ROOM RESPONSE

Figure 4 is the in-room performance of the speaker. I placed the speaker in my large family room that is 19 by 18 with a complex ceiling whose height runs from 12′ at the sides to 16′ at the center. The graphs in **Fig. 4** are an average of nine points taken around a 2 × 3′ square around the prime listening position, which is about 9′ back from the speaker. I placed the microphone at ear level for all nine measurements. As with all my measurements, the grille was removed. With the AcoustiSoft RplusD software, a measurement like this can be done in only a few minutes since it automatically rolls each new measurement into the average.

To bring out response changes of the speaker in **Fig. 4**, I used a tight 0.1 octave smoothing for this graph. To achieve resolution to 20Hz, I set the gate time of the measurement system to 1 second, which essentially makes the measurements equivalent to an RTA but with a higher resolution than the one-third octave response those

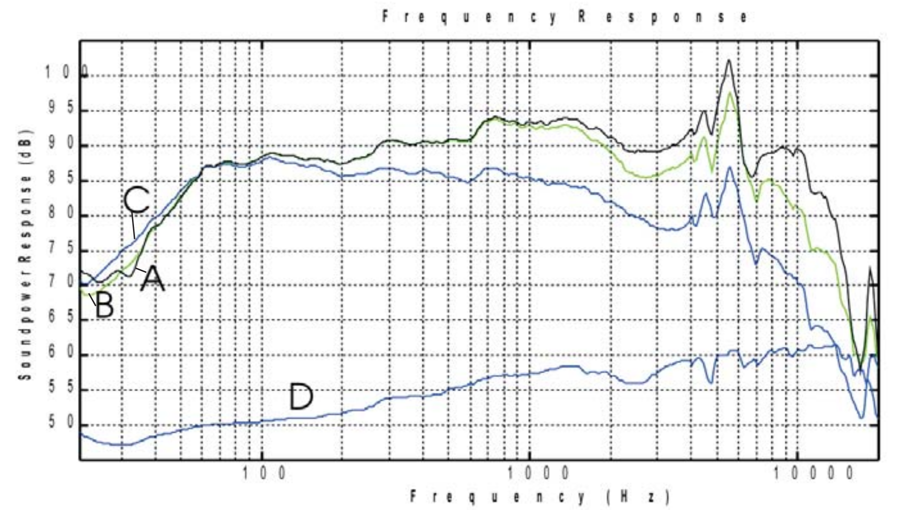

FIGURE 5: Measurement of Infinity C336 woofer array. Three woofers with no crossover. Graphic supplied by Harman International.

devices provide. The 24dB span in the vertical axis is tight for an in-room response curve emphasizing variation in the amplitude response. I show the speaker in two different positions in the room to separate room boundary effects from the speaker's intrinsic performance. Given the room size, many speaker placements are feasible.

As I have said in the past, a graph of the in room performance of a speaker cannot alone reflect the total performance of the speaker. As Dr. Toole details in his papers, the direct listening window response along with the first bounce reflections from the walls, the ceiling, and floor are equally important in the speaker-dominated region above 300Hz. **Figures 1–3** cull the data that the room response cannot provide.

That said, you can still get some significant information from the room response. The low-frequency limit at 6dB for both positions is 50Hz, about the same as the Beta 40. As I mentioned previously, the C336 might appear bigger than the Beta 40, but its internal volume is about the same. Does the extra woofer of the C336 offer an advantage? Perhaps, with the slightly higher sensitivity between 0.5 and 1dB that I measured around 90dB SPL.

I asked designer Emmanuel Millot to shed more light on the extra woofer in the C336, and he replied, "The design goals for the C336 were high output capability, low distortion, and low power compression." In the absence of an isolated room to deal with a 100dB SPL test level, power compression tests were not possible. High ambient room noise prevents me from measuring distortion. For this reason, I cannot objectively demonstrate the advantage of the woofer. You can see why I delayed publication until I had someone who could answer these key questions. The C336 played very cleanly at the highest amplitude levels; I could play big classical scores within its low-frequency limit.

The shape of the curves in the room dominating this portion of the spectrum (below 200Hz) in **Fig. 4** are similar to what I measured for other tower speakers with two woofers in the same position, suggesting the C336 has flat anechoic response at the bottom end and it is not boosted up to move fast off the floor at the dealer.

The virtual absence of the 2–3kHz dip in **Fig. 4** is expected against the backdrop of the 1m quasi-anechoic measurements in **Figs. 2** and **3**. At 15kHz, the top end is only 2dB down from the midband response. Some of this is due to the 15kHz resonance of the tweeter, but, even in its absence, the C336 has a stronger top end than most speakers in-room. The Beta 40s high-end started to roll off around 8kHz and was down 2 or 3dB below the C336 at 15kHz. I cannot directly compare them since I measured the Beta 40 in a different room with different software.

Figure 4 shows a sizeable dip in response at 600Hz as well as 250Hz. I tried all sorts of speaker and listening chair placements, but I could not get these to go away. The 600Hz dip is also evident in the listening window plot of **Fig. 1**. I had never seen these sort of dips in this listening room before from other speakers, although I have not tested a 3-woofer 3-way design before. With this configuration uncharted waters for me, I needed to refer to Harman engineers.

Emmanuel Millot writes:

"The overall woofer response encompasses transducer response, the low frequency alignment and other factors such as enclosure diffraction. In the C336, the result is a rising amplitude response with steps at 250 and 600Hz. When combined with the passive network, these steps manifest themselves as the 200 and 600Hz dips in the system response you have noted."

This is best understood with reference to anechoic frequency response curves in **Figs. 5** and **6** that were supplied by Emmanuel Millot. **Figure 5** is the response of the three woofers alone, and you can see the steps in the amplitude response. **Figure 6** is the response of the three woofers with the 600Hz crossover with a rolloff of a 4^{th}-order LR network with the resulting dips.

Figures 5 and **6** show the direct response(A), the listening window(B), the sound power (C), and (D) the directivity index (the difference between the listening window response and the sound power). For the discussion here, concentrate on trace A.

My vertical response curve was not reliable because the speaker was on the floor and the quasi-anechoic measurement failed, not only because I could not window out reflection, but 1m was too close for the four speakers (three woofers and the midrange) to converge correctly. **Figure 7** is a polar vertical radiation pattern supplied by Emmanuel Millot made in Harman's large anechoic chamber. Now you can see that the amplitude is reduced around 600Hz missing in my data. This data from a real anechoic chamber shows my measurements were inaccurate and shows the 600Hz dip is related to the crossover frequency, the nature of the driver placement of the baffle, and the crossover design.

You will notice from **Fig. 7** that the response variation in the vertical direction going up from 0° is very small around the 3kHz midrange/tweeter crossover region and correlates with what I measured in **Fig. 3** using the quasi-anechoic technique. The midrange and tweeter were closely spaced on the C336 baffle, so the sound converged at 1m. In addition, the higher frequency at the midrange tweeter crossover allowed reflections to be windowed out more easily. Again, the lack of amplitude variation in the midrange/tweeter crossover region is extremely impressive for any speaker, and especially one in this price range.

Subjective Evaluation

I mentioned Dr. Toole's circle of confusion earlier in this review. Harman Labs has a specially constructed room for double-blind matched level testing of speakers. Each speaker is rotated into place so that it occupies the same space as the other speaker under test. Trained listeners hear carefully selected music that allows performance differences to be quickly detected. The Harman Labs advocates test in mono for best discrimination of differences between the test speakers. Random switching is used to ensure that the subject is actually hearing differences. If the same speaker is played again, the test subject should give it similar ratings.

The speakers that did well in the controlled environment can be measured and compared with speakers that did not do well. The process is continued with many speakers until the measured design parameters for a good-sounding loudspeaker are clear. If you run through the AES papers published by the NRC and then Harman Labs, you see how they refined the process of controlled listening tests as well as the measurement procedures for speaker evaluation.

The comprehensiveness of Harman's subjective tests puts a reviewer at a distinct disadvantage unless he/she has a special room to do the double-blind tests. You can try to simulate Harman's approach with a set of listeners (only one at a time to ensure best seating), grading speakers from behind the screen. Anoth-

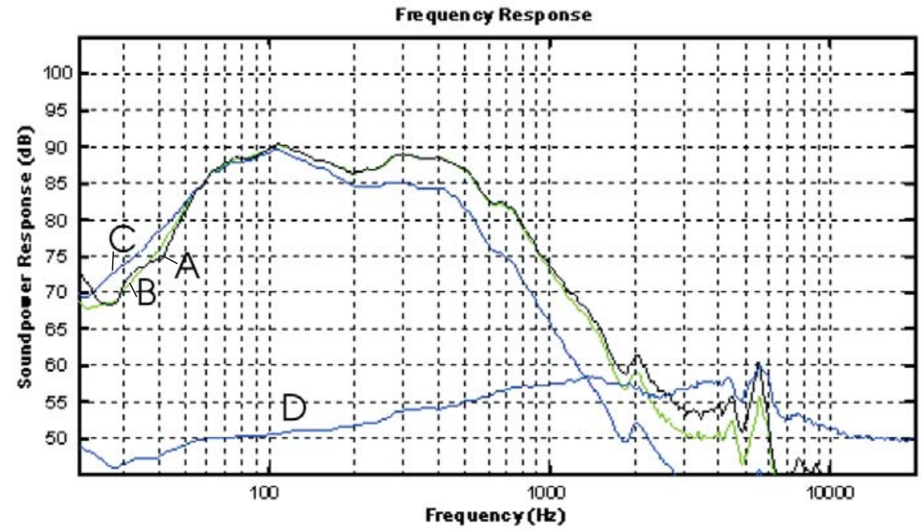

FIGURE 6: Measurements of Infinity C336 woofer array. Three woofers with 4th-order LR crossover at 600Hz. Graphic supplied by Harman International.

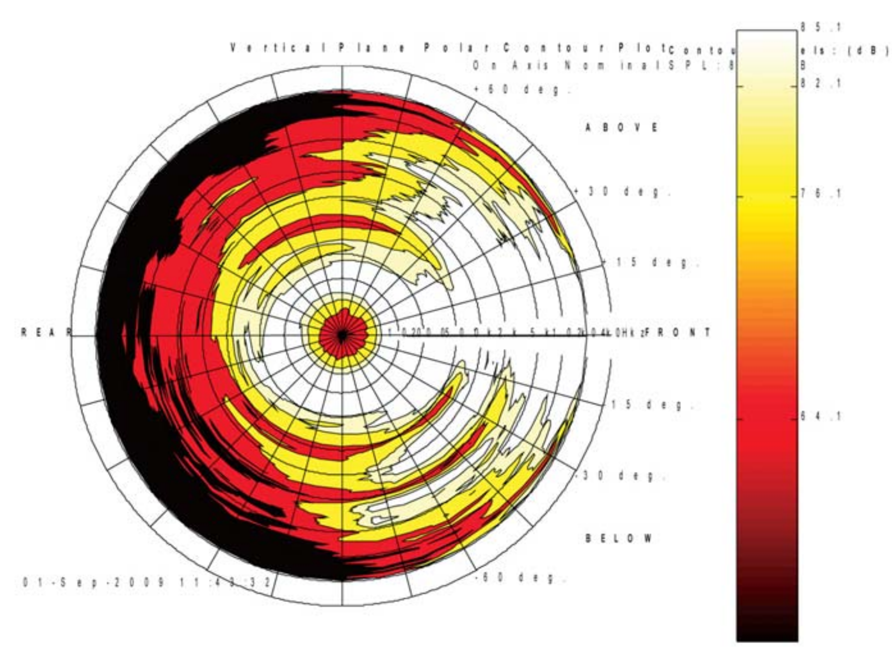

FIGURE 7: Infinity C336 vertical plane polar contour plot. Graphic supplied by Harman International.

er person, hidden from the view, could randomly switch between the speakers. The person needs to be hidden to avoid facial expressions from supplying clues to which speaker is playing.

As far as I can tell, the folks who go to this trouble today are working for the *Audioholics* website (www.audioholics.com). You can replace the person with an electronic box that does the random switching, but these are no longer commercially available. It is not a trivial task to design such a box, especially when switching speaker cables. I find it easier to drive each speaker with its own power amp and switch at line level.

I wimped out and listened to the C336 comparing them sighted to other speakers in the room at matched levels. One reason was the other speakers available were of sufficiently different sizes and voicings they could be easily identified. Unfortunately, I could not directly compare the Infinity Beta 40 because I sent it back long ago. I also used headphones to gauge the quality of my source material with the room taken out of the picture.

In the end, the C336 sounds like it measures. That was, after all, the objective when the speaker was designed, so the on-axis, early reflections, and sound power track each other and remain monotonic. Compared to most speakers, the C336 has a more apparent top-end which correlates with my measurements, both with the quasi-anechoic listening window (**Fig. 1**) and the in-room measurements (**Fig. 4**). The absence of any dip in the 2–3kHz area did appear to make the speaker more sensitive to the quality of the recording. Close miked recordings with lots of post-production EQ became fatiguing to listen to.

Of course, what is needed to make these recordings sound better is a parametric EQ to add in the dip and reduce the high end. Of interest is that many room EQ systems offer just this in the optional target curves. Dips at around 2kHz and options for increased high-frequency rolloff are available.

At the end of the day, the C336 is a more accurate loudspeaker compared to those voiced to taste. Such a speaker is much easier to EQ for badly produced recording than one that is already voiced and may have inherited poor early reflection frequency response curves in the process. That said, a control to reduce the top-end would have been useful as provided on some Harman-branded Revel and JBL Pro models. Most users do not have the appropriate equalizer in the signal path.

Down at the bottom, the speaker performed similarly to other floor-standing speakers with similar low-frequency cutoffs, but the C336 might have been a little cleaner at higher SPLs. I expect the real differences are apparent only when testing SPLs higher than I was willing to tolerate. In the frequency range of the midrange and tweeter, the C336 produces cleaner sound than the other speakers I had on hand at high SPL levels that I could tolerate higher.

I tried using the Anthem ARC room EQ to flatten the response of the C336 and the other speaker under test. The Anthem permits you to limit the lowest frequency to which the EQ is applied. I made sure the C336 low-end rolloff was unaffected by the EQ. The Anthem also limits the upper frequency to which it corrects. I set it at 300Hz, which is the room-dominated area.

The subjective result of the EQ was similar for the C336 as other floor-standing speakers I have tested. The level of adjacent notes smoothes out in the low brass, double bass, and tympani. The sound quality also changes as the first harmonics are brought into correct relation to the fundamental of the instrument. I recommend a good recording of the Britten "Young Persons Guide to the Orchestra" to hear these effects since the score isolates different sections of the orchestra. The double bass section playing in the low register is especially telling, as is the tympani section.

I was bothered by one subjective effect that does not show up clearly in the measurements. This is a tall speaker with the big array of three 6.5″ woofers at the bottom and the 4″ midrange and tweeter close to the top. This resulted in a subjective impression that the violins appear to be coming closer to the top of the speaker while the trombones and cellos sound as though they are further down the speaker. I tried the speaker at distances between 7 and 10′ with little change in the effect.

I thought the 600Hz response dip might have something to do with this, so I used the Anthem ARC room EQ system to remove the 600Hz dip, but did not allow it to EQ above one 1kHz. No improvement. Removing the 600Hz dip did not make a significant change in the tonal balance of the speaker.

I think the speakers' very flat response from 2kHz up to 15kHz may be partially responsible for the effect. With more energy coming from the top of the speaker, there is a greater tendency to localize the midrange/tweeter array. I have not seen many reviews about this sort of effect in tall towers and I may be more sensitive to it than most people. You will be able to quickly tell whether you are sensitive to it when you audition the C336. I expect a lower crossover point between the midrange and woofer would have solved the problem, but would have resulted in added cost in the midrange design to get it to go lower without increased distortion or compression.

In conclusion, the speakers measured better than others I recently auditioned, and subjective performance corroborated the results. Its excellent performance is obvious when compared to other similarly priced speaker designs with stylish real wood cabinets and more than three woofers in the so-called 2.5-way configurations that do not have an isolated midrange. The speaker has little identifiable colorations above the room-dominated area above 300Hz, and its dynamic limits are above levels at which I am willing to listen. ***rr***

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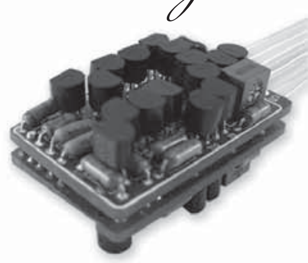
Ed Simon - AudioXpress, October 2009



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