HOW CLEAN IS YOUR JUICE?, PT. 2

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Design a Basshorn Speaker

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Six Channels For Home Theater

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New from **DH Labs**, the HDMI cables exceed 1.3 specifications and support up to 1440p video and up to 7.1 channel audio. Featuring surface treated silver-coated OFC copper conductors, the cables are encased in a nitrogen gas-injected dielectric for ultra-low loss. It also includes a double foil shield and high-density copper braid, and a custom die-cast connector assembly, housing 24-karat gold contacts. For more, go to www.silversonic.com.

Esoteric Sound has released three new audio products. The Rek-O-Kut De-Hisser removes hiss and noise from vintage or modern phonograph records as well as tape recordings of any format.

Featuring high input impedance and accurate record compensation, the Rek-O-Kut Re-Equalizer accurately compensates RIAA equalized preamplifiers. The Rek-O-Kut Archival Elliptical Stylus Kit, consisting of five Stanton 500 compatible, elliptical diamond styli and one conical sapphire stylus, will help overcome noise in records caused by wear and abuse. To learn more, visit www.esotericsound.com.

Ben Duncan Research has released StdBrd-01, a Fiberglass Veroboard based on over 30 years' experience with design and prototyping analog electronics and high quality audio equipment. Features include eight busses, 6 and 7mm edge surrounds, and a fiberglass substrate. The board, suited to digital circuits, valve/tube circuits, and more, can be used in upgrades and additions, field service and repairs, as well as industrial R&D. For more, please go to www.benduncanresearch.org or www.BDR-UK.dial.pipex.com.

New from **Phoenix Gold** (phoenixgold.com), the Ryval Series subwoofer drivers are available in 10 and 12″ models with single and dual voice coil configurations. Including laminate-treated paper cones and high excursion triple layer UV-treated black foam surround, the V10S and V10D subs handle up to 150W RMS and can be mounted in a sealed or ported box. The V12S and V12D have a maximum power handling of 600W, 20-20Hz frequency response, and a top-mount depth of 5.4″.

The **Parasound** Halo JC2 is a stereo preamplifier used to switch between audio source components and to control the output level of the connected power amplifier. Employing fully balanced inputs and outputs, the John Curl-designed JC 2 has standard unbalanced connections, as well as six line-level inputs. It includes an RS-232 interface, four 12V triggers, and remote control repeater input and loop out jacks. Visit www.parasound.com to learn more.

The O 410 Midfield Monitor, from **Klein + Hummel**, is a tri-amplified three-way loudspeaker, featuring magnetically shielded drivers. With a high capacity vented enclosure that cleanly extends bass response down to 34Hz, its drivers are powerd by 340, 160, and 180W hybrid class A-B amplifiers. Its mains voltage can be switched between 100, 120, and 230V; it contains a dimmable display for protection; and flexible acoustical controls adapt the monitor's response to suit its environment. You can find more information at www.klein-hummel.com.

New from **Sunfire Corp.**, the Sub-Rosa In-Wall and On-Wall subwoofers couple 2700W with StillBass' anti-shake technology. The SRS-210W combines dual 10 High back emf drivers, and features inverted surrounds and die-cast baskets, while the SRS-210R is 3.75″ deep. For more, go to www.sunfire.com.

The JBL Story—60 Years of Audio Innovation is a new book by the late John Eargle. Offering a historical perspective on the people and the products of the audio company, topics include stories behind the development of innovative applications for consumer products, systems installations, as well as engineers and colorful record producers, musicians, and

technicians who have used and contributed to the company's products over the years. It is available from Old Colony Sound Lab (custserv@audioXpress.com).

An Excellent DIY Basshorn

Analyzing measurements to determine when a basshorn speaker attains or departs from the target design; this design method is from the work of Dr. W. Marshall Leach, Jr., and articles by Dr. Bruce Edgar.

Builders usually choose a suitable woofer and design a basshorn cabinet around the woofer characteristics. I will begin a theoretical design using this method for the JBL model 2226H 15″ (38cm) basshorn driver.

A HYPOTHETICAL HORN

In order to gain the maximum output at the horn cutoff frequency, I use reactance annulling by way of a sealed back chamber. Leach states that reactance annulling occurs just above the driver fs, and this condition can be imposed if the hyperbolic horn equation annulling parameter (M) is a specific value. Stated another way, the lumped (driver and rear chamber) capacitive reactance cancels the inductive reactance of the throat and horn path. Thus the horn is purely resistive near the cutoff frequency.

Compared to other expansions, the hyperbolic area expansion gives the greatest output near the horn cutoff frequency. Because the hyperbolic curve is long and expansive near its mouth, its area expansion is *not* easily built using wood. To maximize cutoff frequency output, choose a flare rate in which the woofer's capacitive reactance begins to peak. This is usually a few (0-10) Hz above the driver resonance. The capacitive reactance in **Fig. 1** shows a peak beginning from 44.3Hz (measured fs) and reaching a peak near 60Hz.

For a perfect 50Hz full wavelength horn, this driver will be suitable for a horn cabinet that has a length of one wavelength of 50Hz; i.e., 271.8cm (107″) in length and a mouth diameter of 222.5cm (87.6″). A throat diameter of 23.7cm (9.33″) will maximize the acoustical output of the basshorn in its bandwidth. The Leach equation for the throat St is St = (2 * 3.14159 * fo * Vas * Qa)/((1+alpha)*345)) in units of m^2. The horn length, mouth, and throat dimensions are not always practical, but you can simulate this easily.

What would the ideal (50Hz) horn look like? In **Figs. 2** and **3** I simulate and plot the JBL 2226H woofer using the requirement of a full-size horn. The horn parameters are determined using reference 1.

One noticeable characteristic for the ideal hyperbolic horn is that Z-mag (bottom line) is nearly flat (i.e., highly damped). The Z phase (top line at

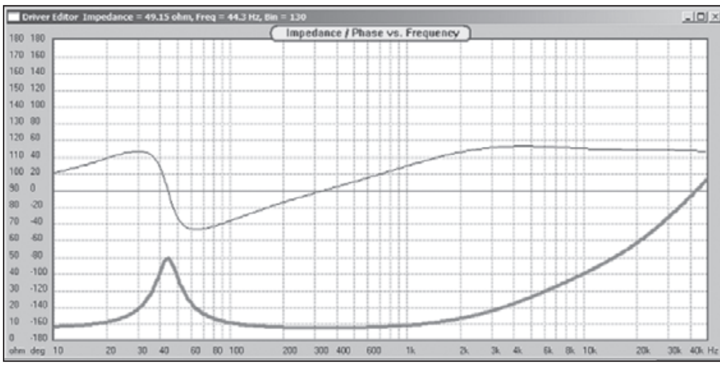


FIGURE 1: SoundEasy simulation shows impedance curve of JBL 2226H woofer. Below fs (44.3) the woofer phase is inductive and above fs its phase is capacitive peaking near 60Hz.

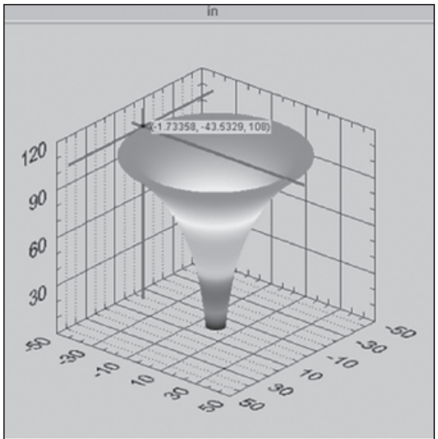


FIGURE 2: A perfect ideal circular 50Hz hyperbolic horn. Mouth dimension is 87.6″ diameter and the throat diameter is 4.7″.

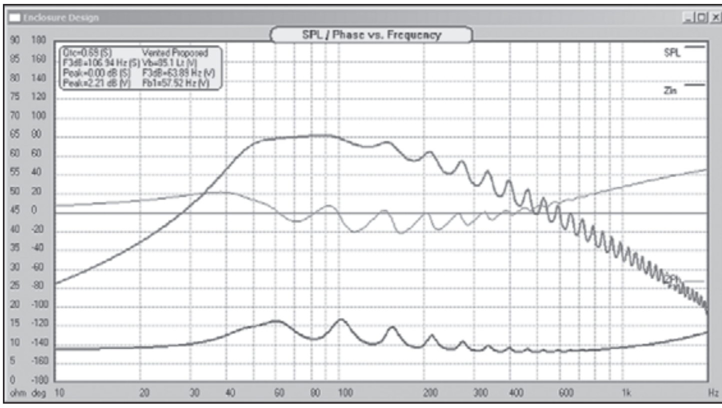


FIGURE 3: Theoretical frequency response of 2226H driver. At the 20Hz vertical frequency line, the bottom line is impedance magnitude, the top line is impedance phase. The middle is frequency amplitude.

10Hz) is essentially cancelled above fc (at the 50Hz cutoff frequency) compared to the driver by itself, as in **Fig. 1**. Its frequency bandwidth is roughly two to three octaves.

The squiggle in the horn output is due to the model's discontinuity at the horn mouth reflecting back into the throat. Because the horn length is nearly 10′ long, frequencies at three octaves above the cutoff frequency, 50Hz (fc), begin to roll off due to the wavelength being only one-tenth the fc dimension. At three octaves above fc these frequencies are not coupled well by the throat and at three octaves above fc these frequencies tend to obey the "intensity dissipates by 6dB when the distance is doubled" rule inside the horn expansion walls!

This 50Hz horn output level compared to its optimized bass reflex is shown in **Fig. 4**. Notice the horn's output is an acoustic 5dB higher in its passband and a minimum of 3 higher in the stopband range!

A CORNER HORN DESIGN

Now, the full-wave horn is impractical to build, so a little engineering compromise will make life easier. Design the horn for one-eighth the space (the corner position) and use the corner or multiple cabinets (for sound reinforcement) to couple the mouth size for loading down to fc. Using the Leach equation, a 1/8 size, 50Hz horn requires an estimated length of 1.54m (61″) and a mouth diameter of 76.2cm (30″ diameter).

When the horn is full size, the throat diameter should be 23.7cm (9.34″); i.e., throat vs. cone surface area ratio (St/Sd) is 0.5. For a 1/8 size horn, Leach's equation for the throat (St)—to maximize the horn output in the passband—is not applicable. When the horn is shortened, it violates the horn equation solution.

The rule of thumb for a shortened basshorn is to use a St/Sd of 0.5 to 0.8 for bass woofers. The lower value (0.5) will give slightly higher fh (upper cutoff), and the upper value (0.7) will give a slightly higher output near fc. Thus, use 0.7 St/Sd for this 1/8 size horn design.

Leach and Edgar explain that when the horn length is not taken out to full size the back chamber volume is usually near 2× the optimum size of a full-size horn. Through my own building (and simulation tools) of horns, I have confirmed this to be true. **Figure 5** is a plot showing the 1/8 size horn SoundEasy simulation.

If this horn were placed in an anechoic environment, its response would look like the **Fig. 5** simulation plot. This is the point I would like to stress. (Some readers may have used other simulation tools that account for a corner.)

Notice that the horn cutoff frequency (fc) is higher, the impedance magnitude (Z) begins to show peaks, and Z-phase shows it is no longer resistive above fc, but favoring capacitive reactance (or negative phase) in the passband.

But unless you're hanging basshorns in free air, you will always have them on a ground plane. If you add a ground plane or an equivalent second identical cabinet to couple the bass response, its response would look like the **Fig. 6** plot. Essentially this is a model of the horn with its area expansion taken out to a quarter space horn length size; i.e., 1.95m (77″).

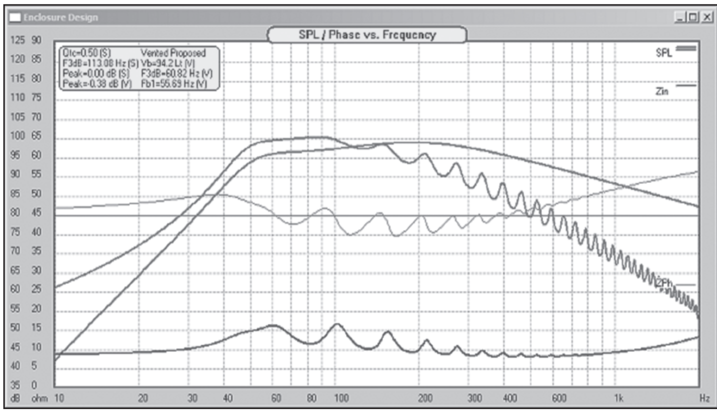


FIGURE 4: The 2226H vented box frequency response data is shown for comparison to horn output. At 50Hz the horn has 5dB higher output.

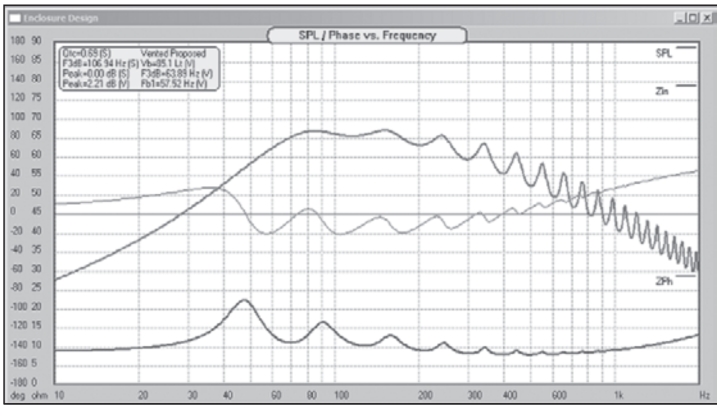


FIGURE 5: The 2226H frequency, phase, and impedance of a 1/8 size horn. The cutoff frequency is now higher and the impedance begins to peak at fs.

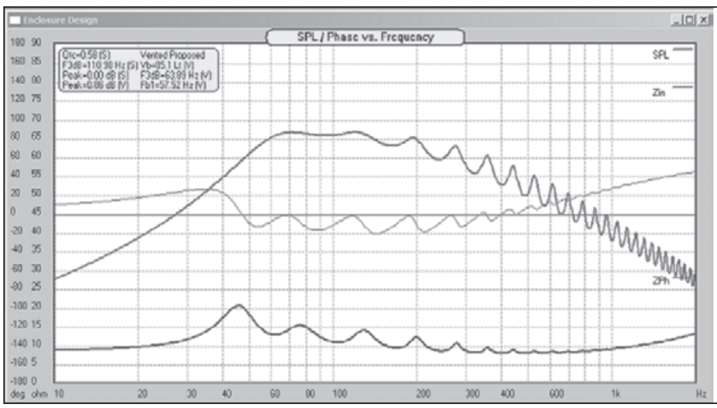


FIGURE 6: The 2226H quarter space horn frequency response, impedance, and phase.

As the area expansion approaches a full-size horn, the impedance phase is canceled due to the horn's (back volume and driver) capacitance canceling the horn's inductive peak, which is at just above the driver's fs value. Should you miss the target of the back volume, it will affect the fc output, and a high impedance peak of the driver and Vback shows up—either above (too small back volume) or below (too big back volume) the target fc. The simulations in **Figs. 7** and **8** each show a different volume—a back chamber volume of 3× and a volume of 1/3× for the quarter space horn simulated.

Figures 7 and **8** show impedance magnitude and impedance phase when a horn's rear chamber is mistuned. Assuming a 50Hz hyperbolic horn is quite near the theoretical horn expansion, the uncanceled rear chamber of the horn's SPL will look like **Fig. 7** or **8** when improperly tuned.

I have come across commercial bass cabinets with a higher or jagged cutoff frequency due to practical or cost-saving errors in the hyperbolic area expansion and again an incorrect back volume. I intend to show you what a correctly tuned horn looks like after building it. At the other extreme, if you happen to build your hyperbolic horn whose length is too long (i.e, beyond 50Hz quarter wavelength), the frequency response will be choked off due to the horn's own resistive path extending to a frequency below 50Hz, not taking advantage of the driver and Vback chamber canceling the horn inductance.

A REALIZED HORN

I have built several basshorn speakers using a combination of the Leach and Edgar approaches. I will share some aspects of the design here and present some measurements to prove the theory works!

(I once naively designed and built a horn path for 30Hz, with the driver fs at 35Hz. When I measured the SPL, the response was rolled off much higher than 30Hz. I later figured out I needed to use a woofer that had an fs at or below 30Hz.)

The driver I chose for the horn is the JBL 2226H 381mm (15″) woofer. Again, a 50Hz quarter space bass horn requires a mouth size diameter 77cm (30.3″), an estimated length of 155cm (61″), and a chosen St/Sd (because the horn is not a full wavelength) of 0.7. The circular estimate of this horn's hyperbolic expansion and 3D picture is shown in **Fig. 9**.

The 50Hz quarter space circular horn designed as a rectangular horn with a fixed width of 50.8cm (20″) will require a hyperbolic expansion as shown in **Table 1**. Notice that the goal of 30″ diameter (or 738 in^2 area) mouth is reached at 54″ length because I chose to start from a throat size of 613cm^2 (95 in^2); i.e., St/Sd = 0.7, because a shortened horn violates the horn equation.

A realized rectangle horn will only estimate the area expansion of the ideal circular horn shown in **Fig. 9**. As I mentioned, the chosen back volume will be twice the theoretical full space horn volume.

This realized 50Hz horn uses diagonal reflectors at the corners because they minimize parallel walls and thus minimize standing waves as the sound travels along the horn path. Dr. Edgar explains this in reference 2. All bass horns I have built used diagonal reflectors with great success.

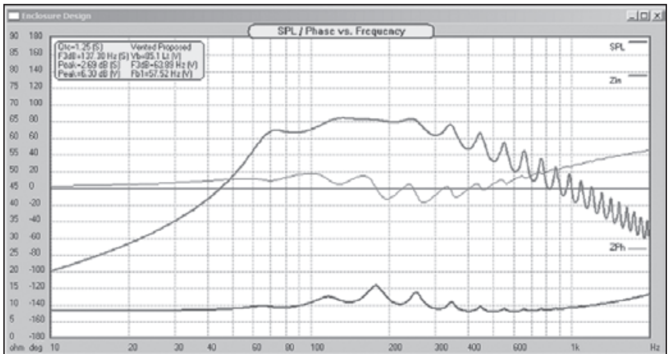


FIGURE 7: The rear volume is one-third the size of a full-size horn. This chokes off the lower frequencies. The impedance peak now shows up near 180Hz, and for the most part the impedance phase remains resistive.

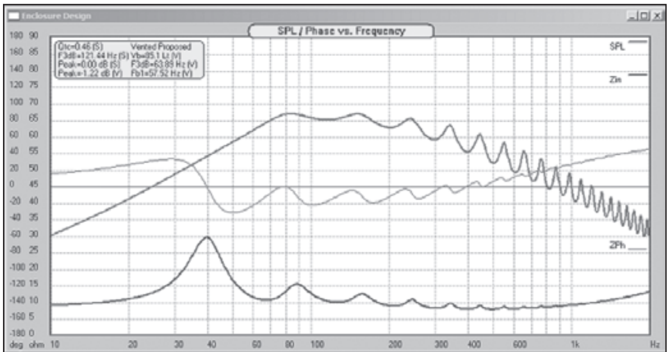


FIGURE 8: The rear volume is three times the size of a full-size horn. An impedance magnitude rise (at 40Hz) appears below fc (50Hz) frequency. The impedance phase shows inductive and capacitive peaks.

Figure 10 shows this bass horn based on all the information presented thus far. For brevity, the throat baffle is not shown. The front and side cutout is shown; the center vertical and horizontal pieces at the top are braces to keep vibration to a minimum.

I recommend ¾″ 11-ply Baltic birch (birch actual thickness is 23/32, so your drawing should be accurate to 1/32 of an inch). To cut costs as I did, use 3/4 MDF for the sides.

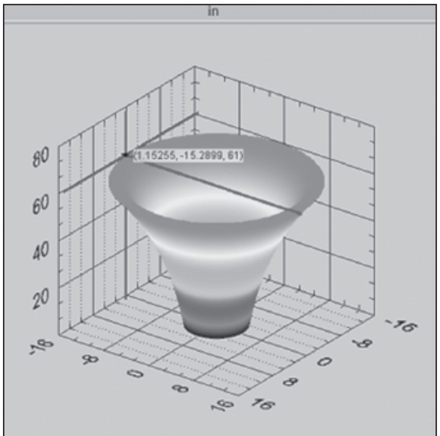


FIGURE 9: JBL 2226H woofer corner (1/8) space horn.

Baffle bracing is critical here. I used two pieces of birch to brace between the throat baffle and bottom facing piece. Cut a square or rectangular throat baffle and place the braces directly over the throat opening at two midpoints across the baffle. Like the front vertical brace, the baffle braces are cut, shaped, and glued on the edge thickness. If you omit the baffle, the horn driver SPL will flex the bottom piece of wood even at moderate levels, robbing you of thumping bass.

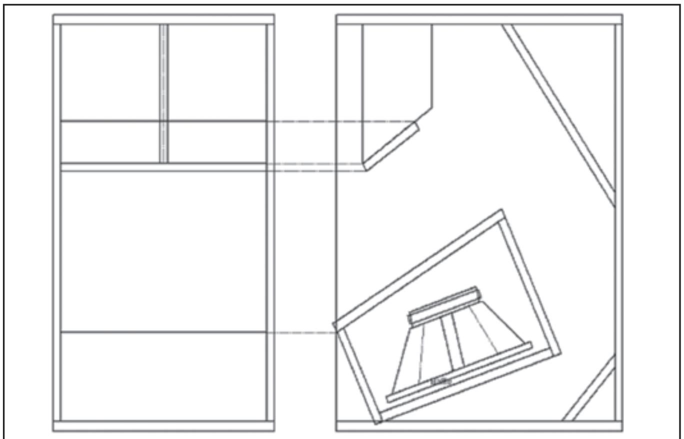


FIGURE 10: The front and side view of my 50Hz drawing design uses JBL 2226 15″ woofer and a sealed back chamber.

MEASUREMENTS

When measuring bass horn speakers, you often need to take the DUT outside away from any large obstruction or walls (other than the ground). When a large obstruction is nearby, the effect will show up as a shift in phase measurement, provided the path length from the driver to the measurement microphone is removed. Close mike measurement indoors will usually measure the horn speaker, including the floor and a wall, depending on where the mike is placed.

Figure 11 is the measured SPL and phase of the 50Hz horn. (Unfortunately, due to cable shortage at the time, I placed the DUT near a wall when I measured outside, so the phase shift between 12-40Hz shows this reflection clearly. Thus, a null at around 12Hz is evident.)

The **Fig. 11** plot is for a single cabinet placed on the ground. The response has a dip below 100Hz that is most likely due to the rectangle horn layout error estimating a circular expansion. The huge null between 500-600Hz is due to wave cancellation of the 2226H driver cutout opening that provides room for cone movement. JBL recommends a cutout opening of ~35.56cm (14″) when front-mounting the driver. As you can see, there are no nulls in the bottom end, thus proving Edgar's hypothesis of using diagonal corner reflectors and 90° bends.

Figure 12 shows the identical horn, but with the number of cabinets dou-

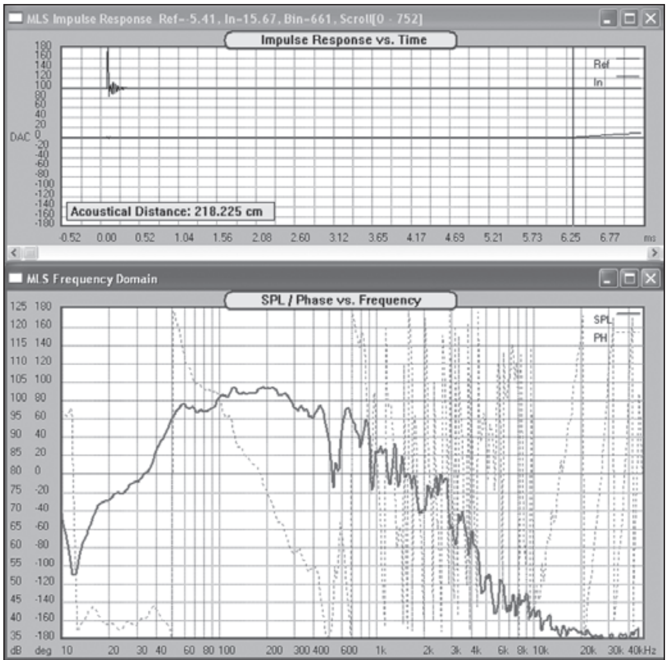


FIGURE 11: SPL and phase of 50Hz horn of Fig. 10.

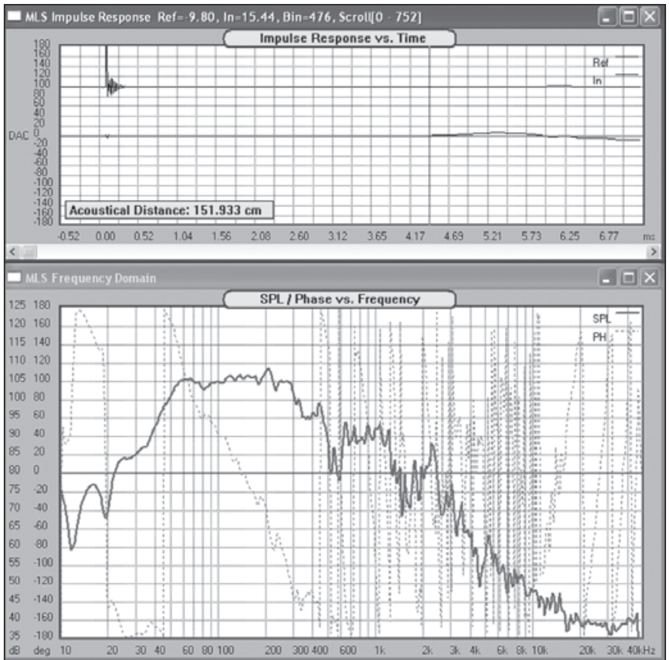


FIGURE 12: Double cabinet SPL and phase of 50Hz horn of Fig. 10.

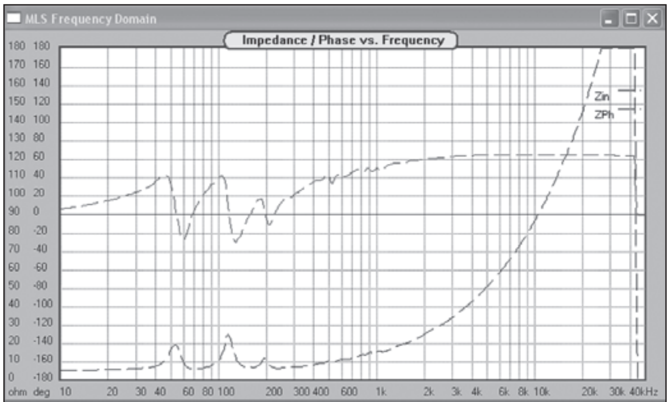


FIGURE 13: Impedance magnitude and phase of Fig. 10 horn.

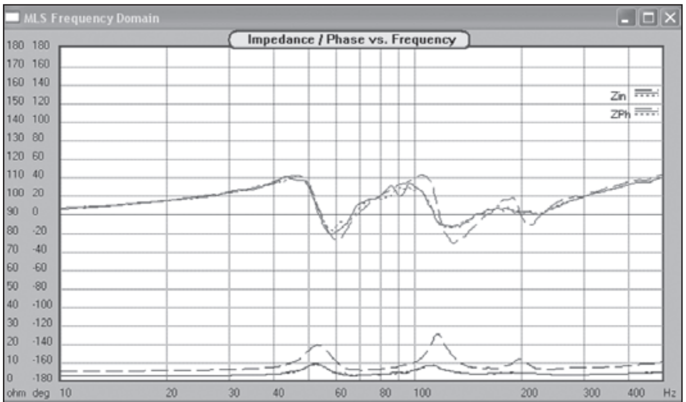


FIGURE 15: Same as Fig. 14 with double cabinet stacked on its side.

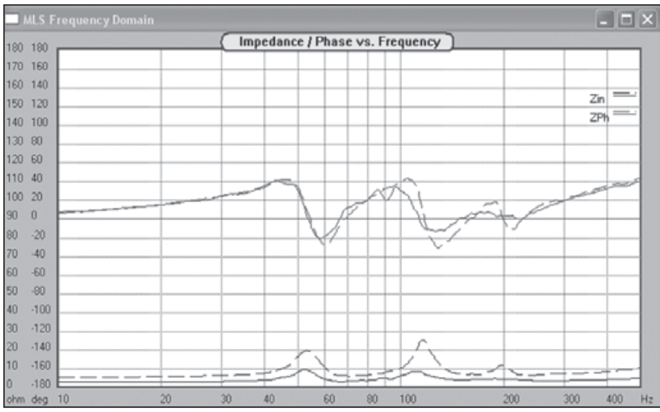


FIGURE 14: Two parallel and side-by-side basshorn cabinet impedance magnitude and phase drawn (solid lines) on top of the single basshorn impedance and phase (dashed line) as in Fig. 13. Peaks in magnitude are flattened and the impedance phase approaches resistive reactance.

bled (i.e., two cabinets side by side). The acoustical distance (driver to microphone) is 1.51933m, and the microphone is 1m from the ground.

The big difference here is that the bottom end no longer dips below 100Hz. It extends down to 50Hz and with this the 3dB bandwidth has shifted lower, from ~450 to ~300Hz. Between 50-300Hz (±2dB) this double cabinet bass horn will outperform any equivalent vented cabinet.

It appears that the design is on target, but how close is the design to the target back volume and horn expansion layout? To determine this, look at the impedance-phase response. **Figure 13** is the impedance-phase plot (0° at 10Hz) and impedance-magnitude plot (7Ω at 10Hz) of a single cabinet. The cabinet is standing vertically, i.e., as in **Fig. 13**. When taking impedance measurement, you should note the mounted position of the horn. (It makes a significant difference by affecting the mouth size.) As you can see, the measured plot has the characteristics discussed in the case of the theoretical horn that was simulated for a quarter space horn.

With the cabinet doubled and with the vertical cabinet position (as in **Fig. 10**), the impedance-phase and impedance-magnitude are shown in **Fig. 14**. As I mentioned, the Z reactance approaches zero, above fc, as the double cabinet's 50Hz is nearer to the full horn mouth size.

Just to prove a point, I placed the double cabinet on its side stacked on the floor and measured the impedance-phase and impedance-magnitude. Because the longest mouth dimension is closest to the ground, this will add more coupling to the mouth. Thus, the Z plot in **Fig. 15** is for two **Fig. 10** cabinets stacked on their sides. The measurement line is the dotted line. Clearly, the impedance-phase approaches zero (i.e., canceling).

Should you decide to build a horn, it is possible there will be other factors that produce resonance above fc. My design is one of the best-behaved bass cabinets I have built yet. In general, the more complex the horn layout, the more resonances appear.

One final note: Midbass or midrange horn design using a cone driver requires a different approach. Thus my use of basshorn in this discussion. *aX*

REFERENCES

1. *Introduction to Electroacoustics & Audio Amplifier Design*, Dr. W. Marshall Leach, Jr., 3rd Edition, 2003, ISBN 0-7575-0375-6.

2. "The Monolith Horn," Dr. Bruce Edgar, *Speaker Builder*, June 1993.

3. "The Show Horn," Dr. Bruce Edgar, *Speaker Builder*, February 1990.

4. SoundEasy Software version 10.0, Bodzio Software, Bohdan Raczynski.

TABLE 1

enter M annulling parameter 0.577273

enter m flare rate 0.0231294

enter throat area ft^2 0.663

enter desired length inches 61

enter fixed width of horn 20

string =

Z(frthroat)	Area/width	Area inch^2
0	4.7736	95.472
1	4.9045	98.09
2	5.0425	100.85
3	5.1879	103.76
4	5.341	106.82
5	5.5022	110.04
6	5.6717	113.43
7	5.8499	117
8	6.0372	120.74
9	6.2341	124.68
10	6.4409	128.82
11	6.6581	133.16
12	6.8861	137.72
13	7.1254	142.51
14	7.3766	147.53
15	7.6402	152.8
16	7.9167	158.33
17	8.2068	164.14
18	8.511	170.22
19	8.83	176.6
20	9.1645	183.29
21	9.5153	190.31
22	9.8829	197.66
23	10.268	205.37
24	10.672	213.45
25	11.096	221.92
26	11.54	230.79
27	12.005	240.09
28	12.492	249.84
29	13.003	260.05
30	13.538	270.76
31	14.099	281.97
32	14.686	293.72
33	15.301	306.03
34	15.946	318.93
35	16.622	332.44
36	17.33	346.59
37	18.071	361.42
38	18.848	376.95
39	19.661	393.22
40	20.513	410.27
41	21.406	428.12
42	22.341	446.83
43	23.321	466.42
44	24.347	486.94
45	25.422	508.44
46	26.548	530.95
47	27.727	554.54
48	28.962	579.24
49	30.256	605.12
50	31.611	632.22
51	33.03	660.6
52	34.517	690.34
53	36.074	721.48
54	37.705	754.09
55	39.413	788.26
56	41.202	824.04
57	43.076	861.52
58	45.039	900.77
59	47.094	941.89
60	49.247	984.95
61	51.502	1030

By Andreas Schilloff

A Home Theater Preamp

With this six-channel preamp, you'll be able to accommodate the latest audio formats.

Is it worthwhile for an audio hobbyist to try and build a multi-channel preamp in the age of multi-format digital audio? The answer is "yes," especially for those who relish separate amplifiers and active frequency dividing networks. Inexpensive home theater receivers do not likely contain multi-channel analog outputs to support a multi-amplifier system.

Obtaining an expensive home theater preamp limits you to the digital audio formats embedded therein at the time of purchase. The chipset used for decoding is often touted as an advantage for audio quality. However, if you wish to take advantage of newer formats such as multi-channel PCM, Dolby True-HD, or DTS-HD available on Blu-Ray and HD-DVD discs, the unit may not be able to decode these. Instead, you will need to resort to using a set of analog multi-channel inputs along with an audio decoder internal to the player or other source.

PREAMP OVERVIEW

The preamp contains inputs for four sources, three of which are six-channel inputs and one stereo input. In my system (**Photo 1**), these are connected to a Dolby Pro-logic decoder (for TV sound), a Blu-ray disc player, a DVD audio/video player, and a CD changer. The disc players decode multiple audio formats internally and deliver them via multi-channel RCA outputs.

The preamp contains five outputs (more on that later). The main channels are sent to a three-way active crossover network and then subsequently through separate MOSFET power amplifiers. The center channel and surround right/left channels are connected directly to respective amplifier channels.

The reason for including only five outputs is that the "subwoofer" input channel is summed with the main channels. This "subwoofer" signal is a combination of the Low Frequency Effects channel (Dolby 5.1 soundtracks) and low-frequency information that has been re-directed from the center and surround channels. This is a matter of personal preference because my system does not contain typical subwoofers, but rather a set of robust three-way main speakers. Should you choose to build your own preamp, you could modify the design to include a dedicated subwoofer output.

The front panel of the rack cabinet (**Photo 2**) contains a rocker power switch, three function push buttons, and a red bezel window. Mounted behind the bezel are an infrared detector and the seven-segment LED displays. The rear panel (**Photo 3**) contains 25 RCA connectors necessary for the audio component connections.

The LED display indicates which of

PHOTO 1: Author's audio/video system.

the four sources is currently selected. When the attenuation up or down command is given, the attenuation level is displayed (0 to −60dB in 2dB steps). For my application, a gain > 1 is not necessary given that digital audio players have sufficient output levels to drive the power amplifiers into clipping (overloaded). However, you may choose to modify the design for additional gain to suit your application.

The preamp is programmed to respond to NEC type IR commands that are generated by a "universal" TV remote. The attenuation level is controlled with arrow keys on the remote. Numbered buttons 1-4 set the input source selection. I also added a mute function ("0" key) that sets the attenuation level

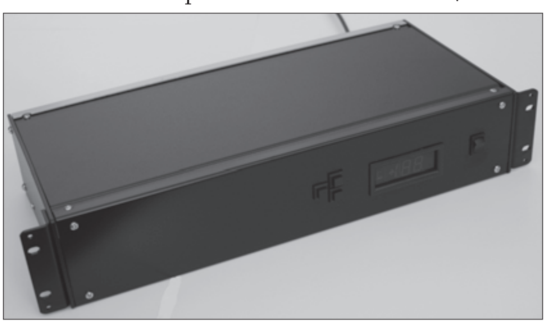


PHOTO 2: Front panel.

PHOTO 3: Rear panel.

to >-100dB, which is great for dealing with annoying commercials. This mute feature is only available from the remote control.

Another added capability of the preamp is that relative levels of the individual channels may be set from within the software. This is useful when the internal decoding of your equipment does not allow for proper balancing of the surround system.

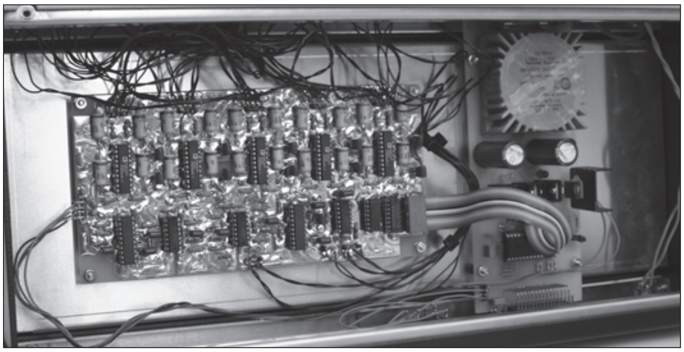


PHOTO 4: Preamp PC and power supply (right) boards.

THE POWER SUPPLY/ MICROCONTROLLER BOARD

The power supply board (**Photo 4** far right) is a single-sided PCB that contains the power supply, the microcontroller IC with associated components, and the IR detector and seven-segment LED displays. The power supply is made up of a toroidal transformer T1, rectifier bridge D1, filter capacitors C1/C2, and four fixed-voltage regulators. Regulator output stability is ensured with the use of tantalum output caps. The ±15V levels are used to supply the op amps on the audio board. All other ICs use the ±5V levels. Note: The four TO-220 package regulators use small heatsinks; however, I used a slightly larger 5W rated heatsink for the +5V regulator due to the heavier current demand.

The PIC microcontroller contains 2K bytes of flash program memory, which is easily programmed using a PICSTART Plus development programmer. This PC-based application allows changes to be made in minutes. The program listing, along with a flowchart explaining the IR remote portion of the code, is posted on the *aX* website (www.audioXpress.com). The program decodes NEC 6121 IR commands, which are 32 bits in length and are derived from Pulse-Position Modulation. I used the 6121 encoding because it was available from an existing TV remote and not used by any other equipment that I own.

The 22 I/O pins of the PIC16F57 are fully used in the preamp design. Port A is connected to the input switches SW1-SW3 and pullup resistors R15-R17. The Sharp IR receiver U7 is also connected to Port A. This three-terminal device contains the IR detector, amplifier, limiter, and 38kHz bandpass filter necessary to receive IR commands from across a room.

Port B I/O pins are connected to the multiplexed common-cathode LED displays (one overflow digit and a two-digit display). The LED displays are mounted on a 30-pin right-angle IC socket directly adjacent to the IR detector. Port C I/O pins control the function of the audio board via a 16-pin DIP header cable.

The schematic for the power supply

board is provided in **Figs. 1** and **2**. **Figure 3** is the component-placement diagram for the PCB. The single-sided etch pattern is also provided in **Fig. 4** (Note: The photo tool is negative in polarity due to the etch process used.) Construction of the PCB is fairly straightforward. I placed small circular pads in the pattern to locate the mounting holes.

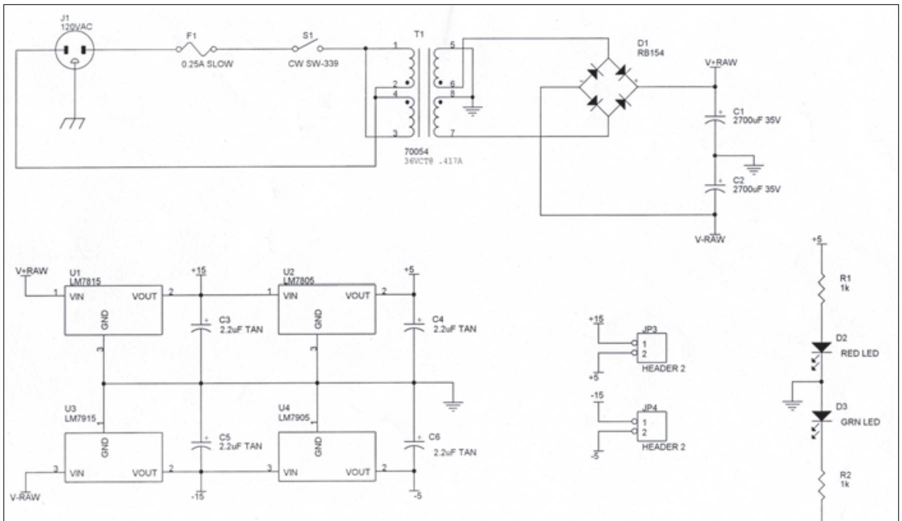


FIGURE 1: Preamp power supply.

THE AUDIO BOARD

The schematic diagram for the audio board (**Photo 4**) is shown in **Fig. 5**. The three sets of six-channel inputs (and single stereo input) are connected through DC blocking capacitors to quad analog switches. The SSM2404 is produced by Analog Devices and is designed especially for audio applications. Noise and distortion figures are negligible. Digital logic levels from the microcontroller select the input source.

The outputs of the analog switches are loaded with 10kΩ resistors. The selected audio signals are then fed to two LM1973 three-channel log step attenuator ICs. These devices are also intended for professional audio applications. I chose stepped attenuator devices over any type of potentiometer attenuation control because even expensive ones wear and induce noise over time. The attenuation level is controlled by serial data sent from the microcontroller. Different offsets may be applied to any of

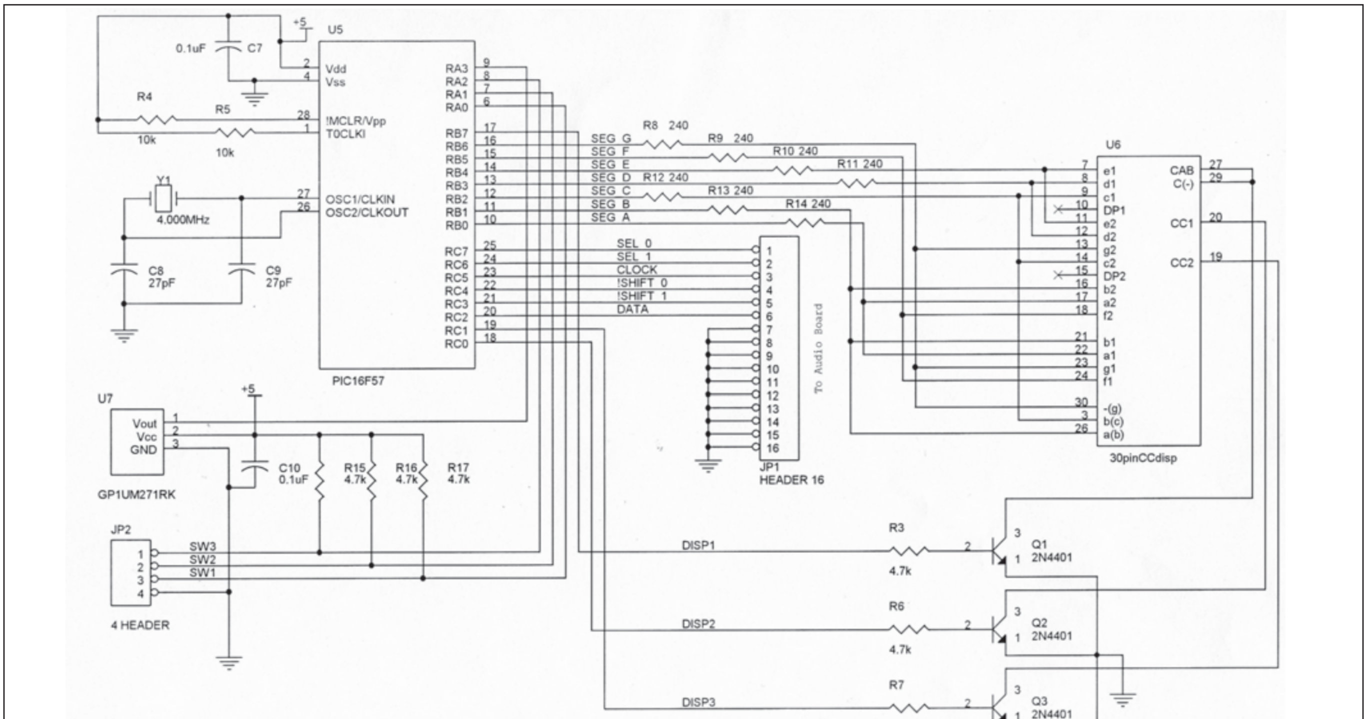


FIGURE 2: MCU/display circuit.

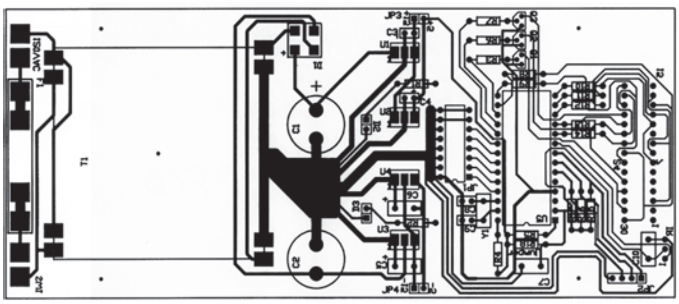


FIGURE 3: Component placement.

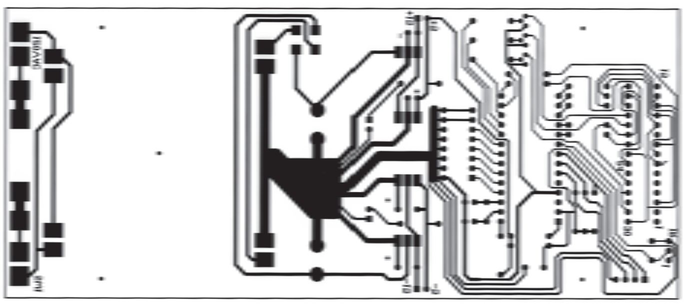


FIGURE 4: Printed circuit layout.

the six channels by the program, because each attenuator channel is individually addressable.

The outputs of the attenuators are buffered using LF347B bi-FET op amps. You may substitute more specialized devices; however, FET inputs are necessary to maintain clickless operation. The 100Ω series resistors prevent signal degradation when driving higher capacitance audio cables.

The purpose of op amp U5 is to sum the subwoofer channel with the main channels. Again, this has been done to suit a specific application (and admittedly a preference). You may modify the design to provide a dedicated subwoofer output if so desired.

The audio board is a double-sided PCB in which the top layer is designed primarily as ground plane. This feature, along

PARTS LIST

Power Supply/Microcontroller Board			
Ref.	**Value**	**Manufacturer**	**Digikey.com #**
C1, 2	2700µF 35V	Panasonic	P11247-ND
C3-6	2.2µF tan	Kemet	399-3550-ND
C7, 10	0.1µF	Panasonic	P4593-ND
C8, 9	27pF	BC Components	1430PH-ND
D1	RB154, 1.5A 400V	Bridge	RB154MS-ND
D2	Red LED	Generic	
D3	Green LED	Generic	
F1	0.25A slow	Generic	
JP1	Header 16	16-pin IC socket	ED3116-ND
JP2	4 header	Generic	
JP3, 4	Header 2	Generic	
J1	120V AC	Generic	
Q1-3	2N4401	Fairchild	2N4401-ND
R1, 2	1k	Yageo	1.00KXBK-ND
R3, 6, 7, 15-17	4.7k	Yageo	4.75KXBK-ND
R4, 5	10k	Yageo	10.0KXBK-ND
R8-14	240	Yageo	240XBK-ND
S1	SW-339	CW	SW339-ND
T1	70054	Talema	TE70054-ND
U1	LM7815	National Semi	LM340T-15-ND
U2	LM7805	National Semi	LM7805CTFS-ND
U3	LM7915	National Semi	LM7915CT-ND
U4	LM7905	National Semi	LM7905CTFS-ND
U5	PIC16F57	Microchip	PIC16F57-I/P-ND
U6	30 pin CC disp	Stanley	404-1131-ND, 404-1136-ND
U7	GP1UM271RK	Sharp	425-1139-ND
Y1	4.000MHz	Abracon	535-9057-ND
Preamp audio board			
Ref.	**Value**	**Manufacturer**	**Digikey.com #**
C1-3, 5, 11-14, 21, 22, 27-30, 35-38, 43, 44	0.1µF	Panasonic	P4593-ND
C4, 10, 15, 20	10pF	BC Components	1426PH-ND
C6-9,16-19, 23-26, 31-34, 39-42	10µF	BC Components	4211PHCT-ND
C45-48	2.2µF tan	Kemet	399-3550-ND
JP1	Header 16	16-pin DIP socket	ED3116-ND
JP2-15	4 header	Generic	
R1-3, 5, 6, 12-15, 17	100k	Yageo	100KXBK-ND
R4, 16, 28, 29, 35	100	Yageo	100XBK-ND
R7-11, 18-27, 30-34, 36-41	10k	Yageo	10.0KXBK-ND
U1	74LS00	TI	296-1626-5-ND
U2	74LS139	TI	DM74LS139-ND
U3-5	LF347B	TI	LF347B-ND
U6, 8-10, 12	SSM2404	Analog Devices	SSM2404P-ND
U7, 11	LM1973	National Semi	LM1973-ND
Miscellaneous			
Item	**Value**	**Manufacturer**	**Digikey.com #**
Rack chassis	RMC19038, RMP1908	Hammond	HM738-ND, RMP1908-ND
LED Bezel		PRD	PRD180R-ND, PRD180B-ND
Selector switches (3)		E-switch	EG2041-ND
RCA connectors (25)		Switchcraft	SC1134-ND, SC1146-ND, SC1147-ND
30-pin display socket		MILLMAX	ED58427-ND
16-pin DIP cable		CW	C6PPG-1606M-ND

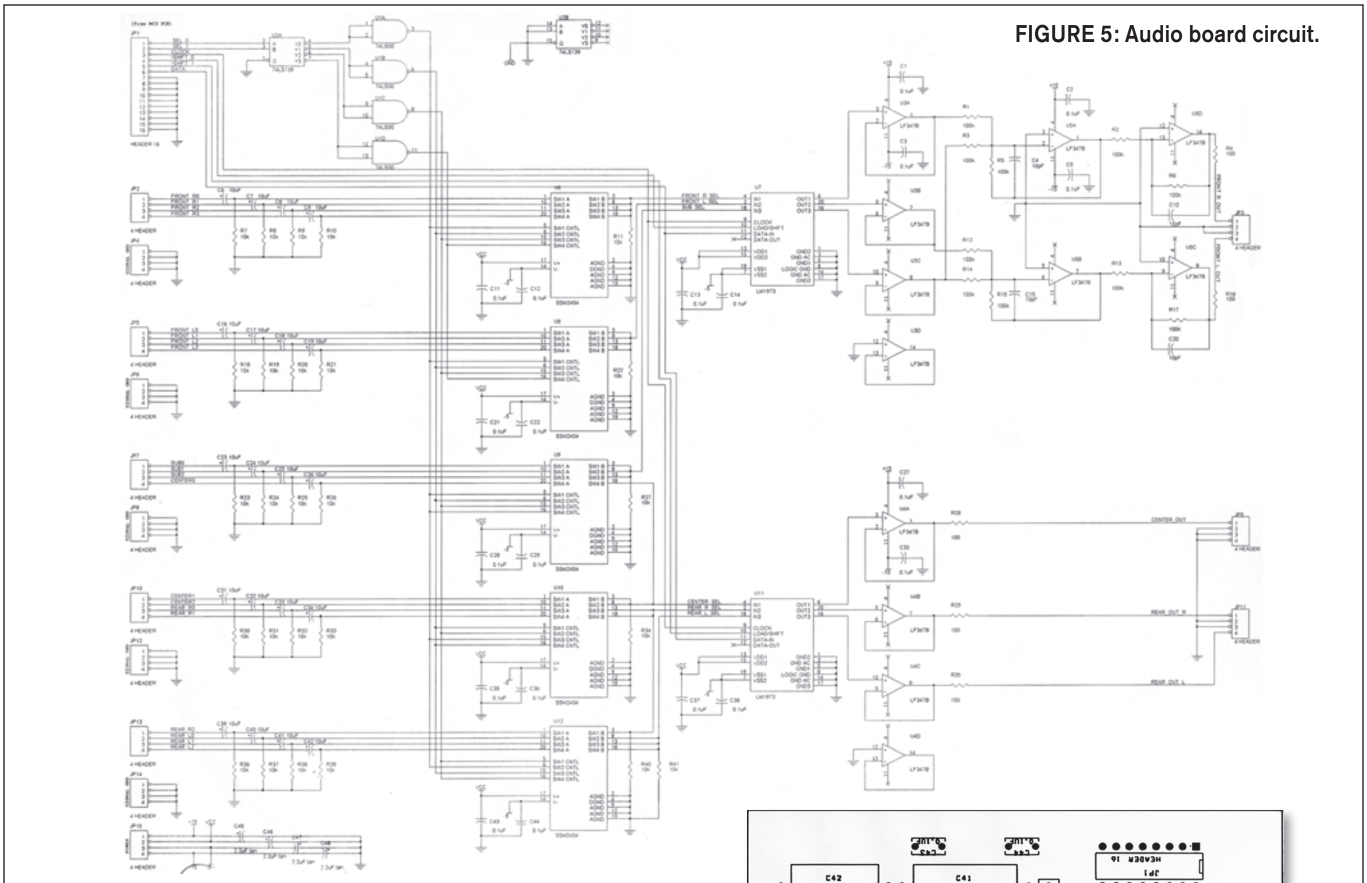


FIGURE 5: Audio board circuit.

with generous power supply bypass capacitors, helps ensure good electrical performance. The layout is available in **Figs. 6** and **7** (top and bottom side, respectively). The component layout is shown in **Fig. 8**.

The PCB has been designed without any vias, such that all front-back connections are made with only component leads. Hint: I achieve better results drilling from the bottom (main routing) side of the PCB. This reduces the chance of damaging a pad-trace connection due to drill bit deflec-

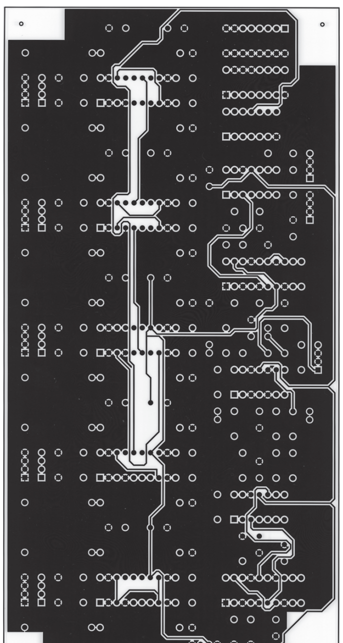


FIGURE 6: Audio board—top.

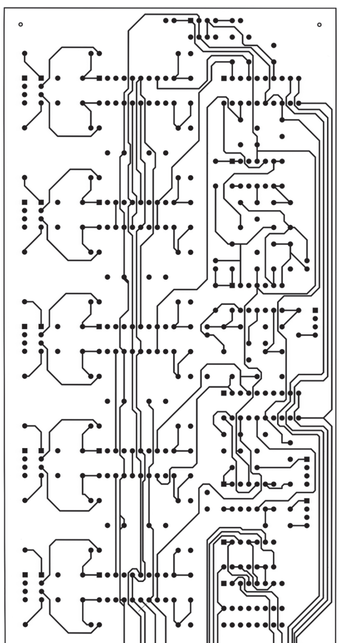


FIGURE 7: Audio board—bottom.

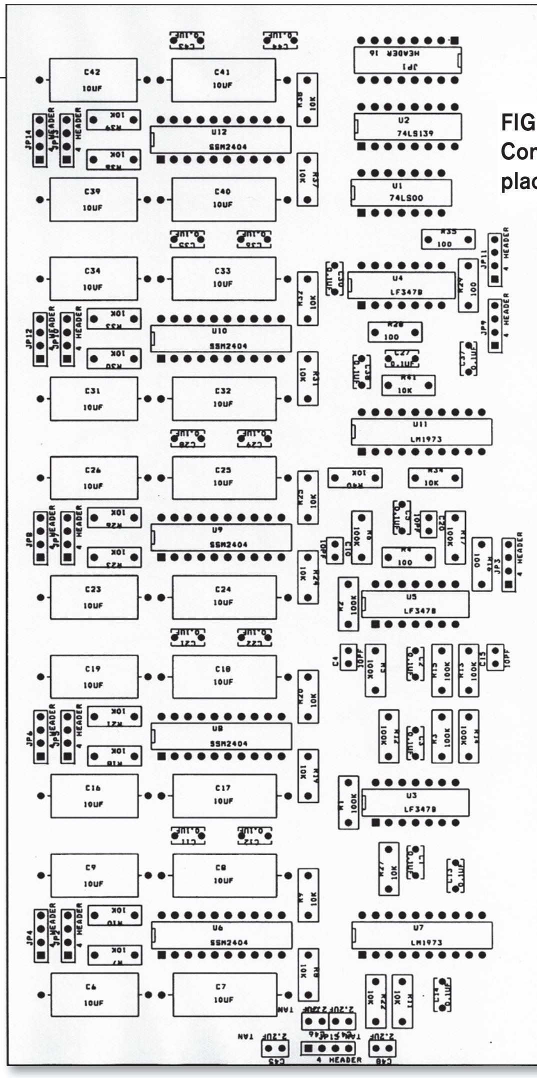


FIGURE 8: Component placement.

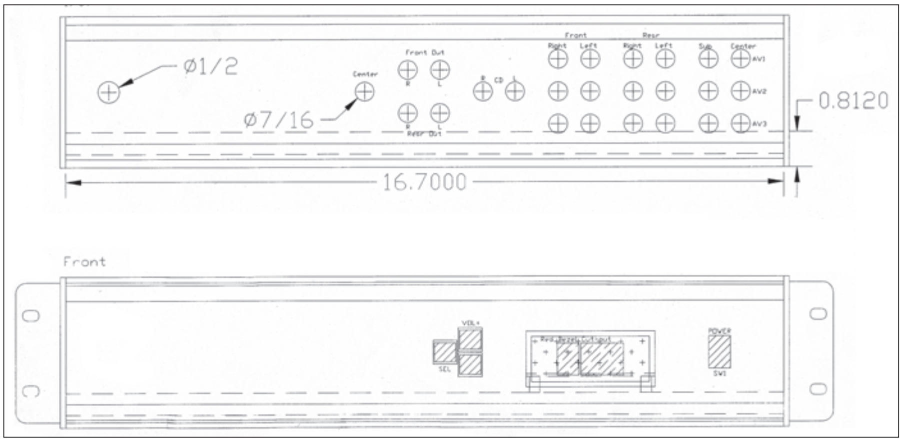


FIGURE 9: Template for front and back panels (45%).

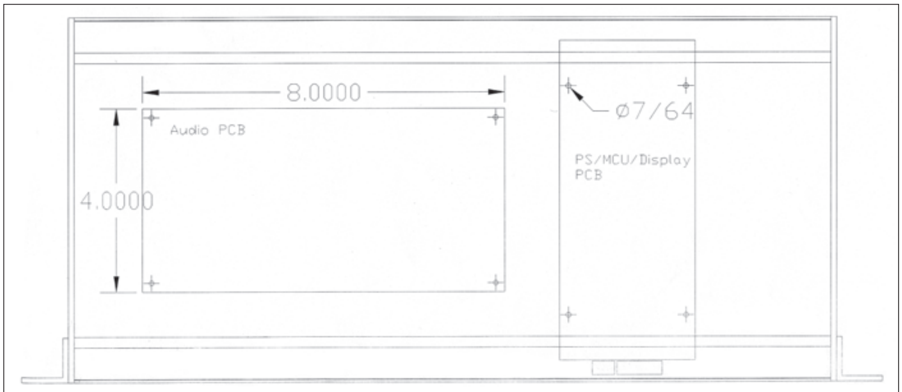


FIGURE 10: Top view template (43%).

tion or positioning error.

THE CHASSIS

The preamp was fabricated in a Hammond RMC19038 19″ × 8″ rack cabinet. I mounted the circuit boards on an optional RMC1908 chassis panel using ¼″ × ¼″ nylon spacers and 4-40 hardware.

I used an inkjet plotter to produce the [reduced as indicated] CAD drilling templates (**Figs. 9** and **10**). Once you cut these to size and affix them to the proper surface, you can use them to locate the drilled holes for the RCA connectors, PCB mounting holes, switch cutouts, and so on. Cut the display openings using a template along with a nibble tool and some patience. Use draw filing to clean the edges after using the nibble device.

FINAL WIRING AND TESTING

During assembly, I first verified the testing of the power supply, display, and remote functions by using a simpler version of the final program and then installed the audio board. After wiring the power supply, DIP ribbon cable, and RCA connectors, I applied power without any ICs inserted into the sockets. Voltage and logic levels were verified for each of the functions prior to installing all ICs.

The power supply and input/output connections were made using 26 GA twisted-pairs to reduce any crosstalk. You could also use shielded audio cable; however, it proved too difficult to manage so many I/Os in a limited space.

The ground plane of the PCB is connected to the power through redundant wires in the ribbon cable between the two boards. As best practice, I purposely made the ground connection much shorter than other power supply connections. Also note: Each RCA connector is connected separately to the ground plane of the PCB to minimize the production of ground loops. Finally, to complete the testing, inject a 1kHz, 2V peak sine wave with a waveform generator and verify the proper outputs and levels with an oscilloscope.

The preamp described here has worked flawlessly for many hours of use since I installed it into my system. It is the link that integrates a multi-amplifier, homegrown audio system with the latest sources without necessitating the purchase of an expensive home theater preamp. aX

By Dennis Colin

The "Amplibass" Control

This natural-sounding bass-boost circuit restores the deep bass attenuated by record producers and the Fletcher-Munson effect, without "muddying up" the midrange.

You may need bass boost for one of three reasons:

1. Most popular recordings, and even some classical, have the naturally powerful and deep bass attenuated, because these waveforms are often of higher amplitude than the rest of the audio signal. Producers reduce the bass because of the infamous "loudness wars," in which sales are proportional to dB SPL.
2. The Fletcher-Munson effect: At the usual lower-than-live playback volume, hearing sensitivity of low frequencies falls off faster than at midrange and treble frequencies.
3. Loudspeaker bass rolloff.

Another reason is simply the pleasure of hearing and feeling clean, powerful deep bass. (My wife says that's a "guy thing." She's right about me!)

A MUSICAL EXAMPLE

Consider "Bayou Country" by Creedence Clearwater Revival, now available on SACD (Analogue Productions CAPJ8387SA). One of my favorite songs is "Born On The Bayou," which I must have played over 300 times since its debut around 1968 (on the flip side of the smash hit "Proud Mary"). In the key of "E," the electric bass frequently plays the open-E string (41.2034446Hz for you mathematicians).

With the lowest notes of an electric bass (and acoustic bass viol), the fundamental is several dB lower in amplitude than the 2nd and 3rd harmonics. This, plus the aforementioned record-production bass attenuation, makes it difficult to hear those deep fundamentals. With "Born On The Bayou" (and most rock recordings, old and new), I need as much as 20dB boost at 40Hz to obtain satisfying reproduction of the low fundamentals. And that's with speakers inherently flat to below 40Hz.

"AMPLIBASS" FREQUENCY RESPONSES

Figure 1 shows the bass boosts with four values of the 100k control potentiometer resistance. With $R_p = 0$, the response is flat (-0.16dB at 20Hz, -0.01dB at 20kHz). The potentiometer values (R_p) shown give 40Hz boosts of 20, 15, 10, and 5dB. Note that as the boost is decreased, so is the "turnover" (start of boost) frequency. For example, with 10dB boost at 40Hz, the 3dB point is 104Hz, and the boost is only 0.28dB at 200Hz.

The maximum slope (with 20dB boost at 40Hz) is –6dB/octave, because the boost circuit is first-order. However, the "turnover" point is sharper than a pure first-order response because of an added small dip effect (described later). In **Fig. 2**, the upper curve is without this dip effect; the 3dB point is 430Hz, instead of the 261Hz with the lower curve. Both curves are for 20dB boost at 40Hz; the lower one is the same curve shown at the top of **Fig. 1**.

OUTPUT NOISE

The measured output noise with full bass boost is 31µV RMS unweighted (-90dBV) and 9µV "A" weighted (-101dBV). With no boost, the noise is 9.5µV unweighted (-100dBV) and 7.7µV "A" weighted (-102dBV).

CIRCUIT DESCRIPTION

The input signal is split into two paths: a flat frequency response path, and a LF-boosted path with variable gain. Unlike conventional tone controls, this has the advantage that with the boost control set to zero, only the unaltered flat signal is transmitted to the output.

The input at J1 is AC-coupled by C1 and R2, RF lowpass-filtered by R3 and C2, then buffered by voltage-follower U1A. U1B serves as a differential amplifier. Without C11, R11, and C12, it would have unity gain and a flat frequency response. The function of C12 will be described later, but C11 and R11 equalize the effect of C12 so the flat-response signal from U1A is fed unaltered to the output at J2.

The function of U2 is to provide an amplified signal having a nearly constant frequency downslope of –6dB/octave, starting at 10Hz. A simpler circuit could do this, but the one used minimizes noise buildup. R7 and C8 provide the 10Hz rolloff pole, but the non-inverting configuration makes the downslope flatten at 57Hz. If R5 were shorted and R4 increased by R5's value, then this—with C5—would produce a lowpass pole at 57Hz and cancel the transmission zero (slope flattening) mentioned, resulting in a constant –6dB/octave downslope. I will describe the function of R5 later (R5 and C5 produce a slope-flattening zero at 343Hz).

The output of U2A is AC-coupled (through C9 and R8, forming a 7.7Hz LF rolloff) to the inverting input of U2B, which provides a variable level of the downsloping signal, adjustable with P1, the bass control potentiometer. Of inverted polarity, it's summed into the output with non-inverted polarity by being fed into the inverting input of U1B, through R13 and R12.

The function of C12 is to provide a rolloff pole at 203Hz. If the value of

C12 were 195nF, this pole would be at 343Hz, the frequency mentioned where R5 and C5 flatten the downslope above 10Hz. These would then cancel, resulting in a constant downslope without flattening. *If* C12 were 195nF.

PHASE SHIFT

The would-be constant downslope would have a phase shift very close to –90° at frequencies much above 10Hz. So, at the frequency where this signal sums into the output with the same level as the direct (flat response) signal, the overall response would be +3dB. This could be called the turnover frequency (f_t), where the bass boost is 3dB.

Now back to the actual 330nF value of C12. Because its rolloff (203Hz) is lower than the 343Hz slope-flattening point of C5/R5, the overall response of the boost signal approaches a -12dB/octave slope between 203 and 343Hz. This also increases the –90° phase lag by a maximum of –15° (at the geometric center frequency of 264Hz). Thus, near 264Hz the boost signal's phase is –105°.

If you were to sum a flat signal with a 2nd-order downslope signal (having a –180° phase shift), there would (ideally) be a complete null where the two signals had the same amplitude. (This is why one driver's polarity must be inverted in a 2nd-order Butterworth speaker crossover.) Here, the –105° phase shift, being closer to –180° (same angle as

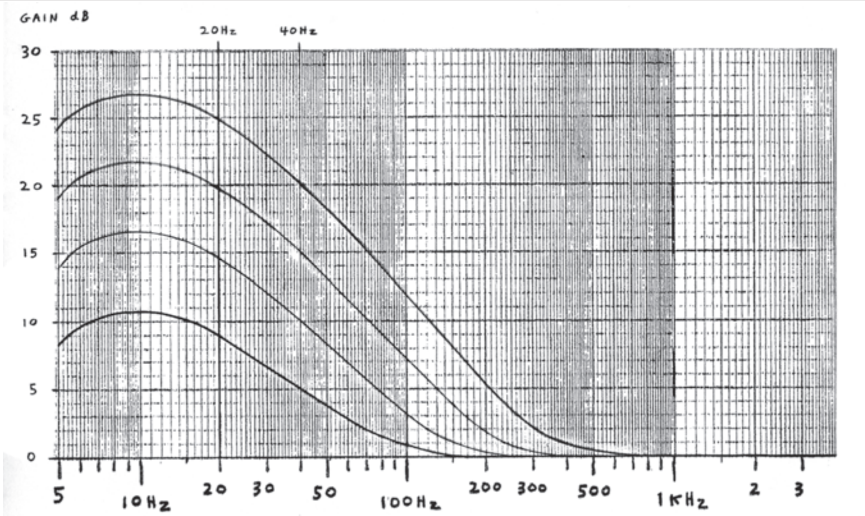


FIGURE 1: Frequency response, "Amplibass." Top to bottom: R_p = 97.3k, 52.5k, 26.9k, 11.7k. f_{3dB} = 261Hz, 167Hz, 104Hz, 57Hz.

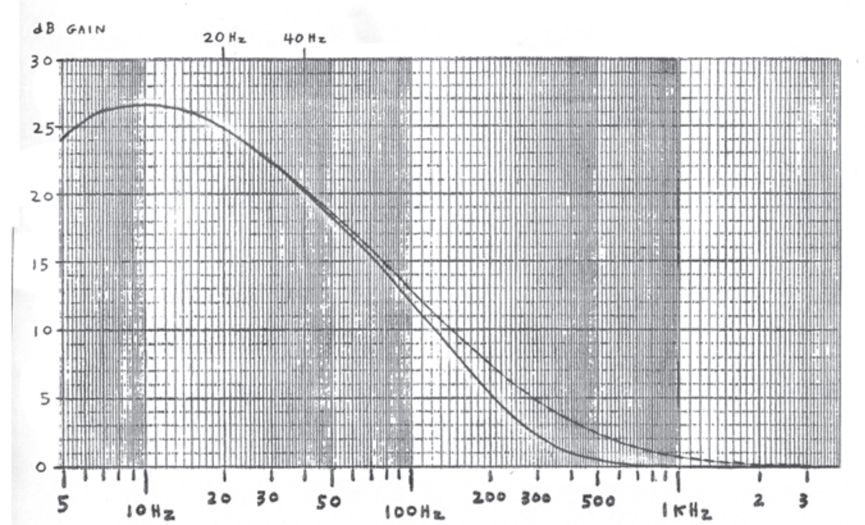


FIGURE 2: Frequency response, R_p = 97.3k. Top: C12 = 195nF (pure 1st-order). Bottom: C12 = 330nF (value used).

+180°) than to 0°, produces a small dip in the overall summed output response, if the turnover frequency is near the 203-343Hz range.

I selected those frequencies so the dip effect wouldn't produce an overall dip, but rather only lower the boost around the turnover frequency so as to sharpen the start-of-boost "corner." The effect is shown in **Fig. 2**. Second- and higher-order filters can use resonant poles and/or zeroes to sharpen the rolloff or boost corners (e.g., Butterworth or Chebyshev). A 1st-order response can't be resonant, but here I use a small amount of added resonant dip to produce what could be called a "maximally flat" 1st-order response. So rather than "cut corners" here, I decided to sharpen them!

Actually, the response is slightly overcompensated. Not shown in **Figs. 1** and **2**, the response does dip to a maximum of –0.18dB at 400Hz, when set for 10dB boost at 40Hz. For an exact maximally flat response (no dip), the Q of the resonant dip effect would be $1/\sqrt{3} = 0.57735$. This produces a boost at the turnover frequency of 1.761dB, where it's about 0.8dB here.

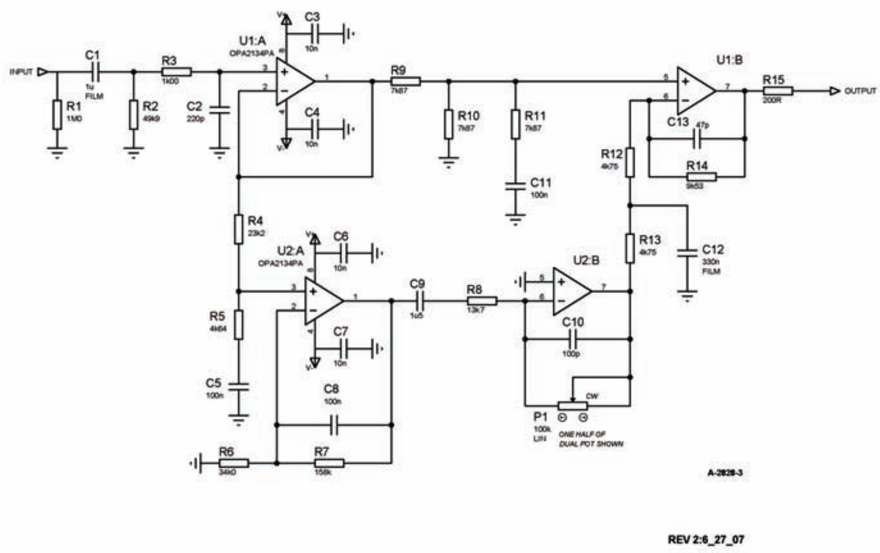
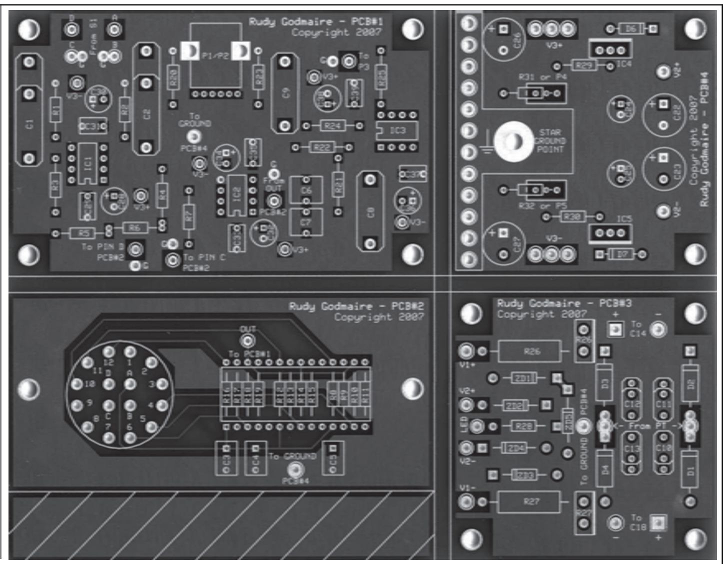
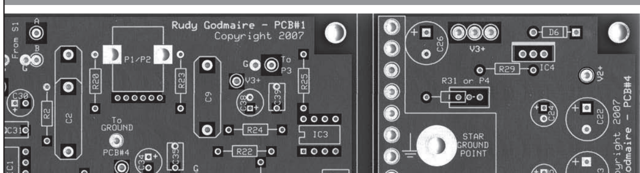


FIGURE 3: Schematic, "Amplibass" control. One channel shown of two.

BENEFITS OF THE OVERALL TOPOLOGY

1. The boost responses minimize effects on the midrange without resorting to 2nd-order responses (which would be too steep).
2. Using C12 to extend the downslope, just before the output op amp, filters noise (above 200Hz) from the boost circuit.
3. Because the boost signal has its own separate path, when R_p is set to zero the boost circuitry is removed from the audio path.





4. The control pot (P1) implements a simple linear gain control. This allows, if desired, the U2B circuit to be replaced by an inverting voltage-controlled amplifier IC. Then, the bass boost could not only be remote controlled, but also electronically "ganged" with a remote volume control. By having the controls inverted, the bass boost would increase as the volume decreases; furthermore, the boost could be made to start only when the volume decreases to a predetermined level.

This could produce a very effective and wide-range loudness compensator. It would allow you to hear, for example, 20Hz organ fundamentals over a wide volume range.

Table 1 shows 40Hz boost versus R_p values. **Table 2** is the parts list.

A NOTE ON AUDIO QUALITY

I conducted an "intrinsic fidelity" test on this circuit. With zero bass boost, I did blind A/B and extended listening comparisons between the unit and a bypass connection. I used high-quality SACD music. The gains were matched to 0.02dB. I heard absolutely no difference, and my hearing is good (recently checked).

IMPULSE AND STEP RESPONSES AT MAXIMUM BOOST

Figures 4 and **5** show the response to a "virtual impulse," a 200µS square pulse of +8V amplitude (this input not shown; it's too narrow to be visible on the much longer time scales). Also not visible, but present at the output, is the flat-response-transmitted short input pulse. The impulse rate in **Fig. 4** is 100Hz. Time scale is 2mS/div.

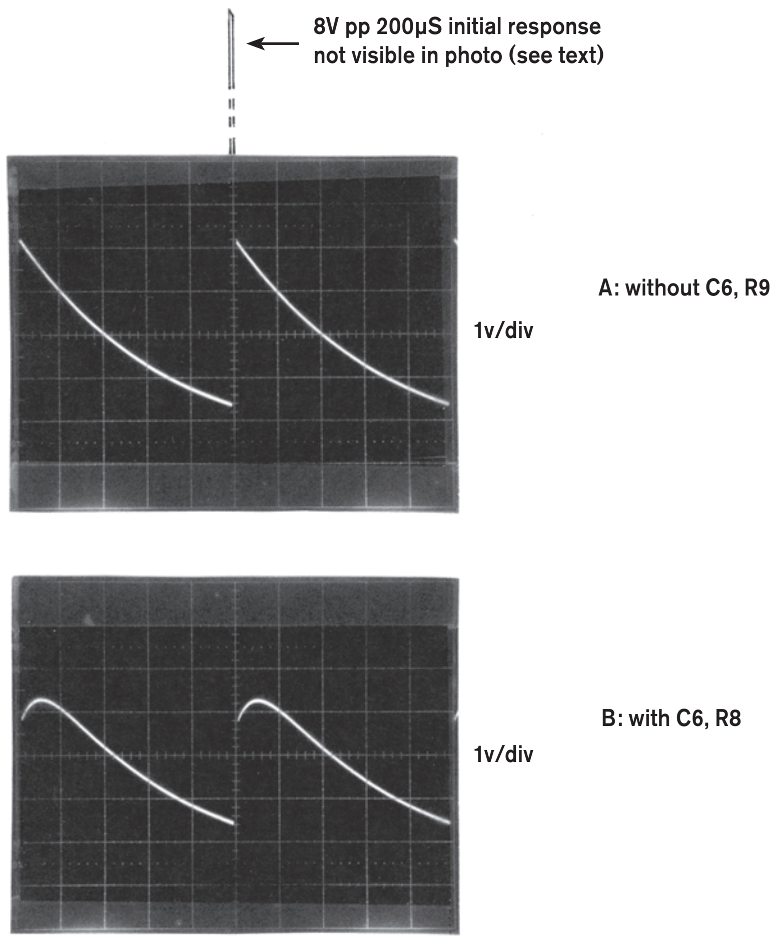
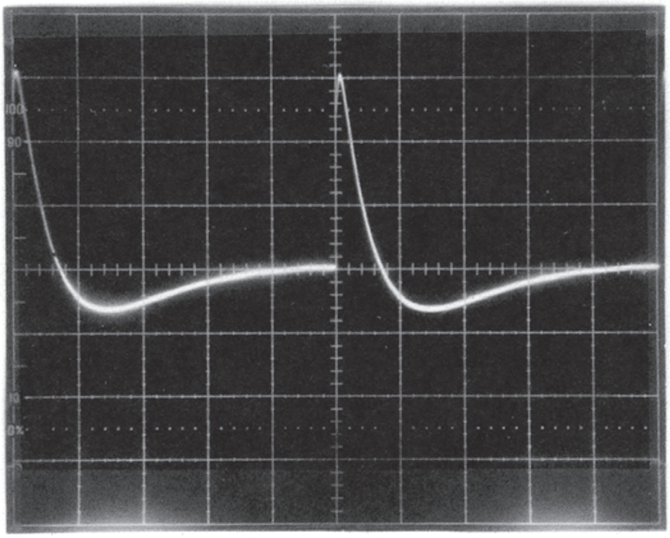


FIGURE 4: LF boost portion of impulse response, impulse rate = 100Hz, maximum boost.

Figure 4A is the output with C12 = 195nF (no sharpening). The exponentially decaying output step is from the time integration (with 10Hz rolloff) of the 1st-order bass boost. **Figure 4B** shows the effect of the turnover sharpening phase shift (C12 = 330nF). The slightly resonant effect on the impulse response is seen in the curvature following the step.

In **Fig. 5**, the impulse rate was lowered to 10Hz, showing the response on a longer time scale, 20mS/div. As with **Fig. 4**, the initial flat response transmitted 8Vpp 200µS pulse is not visible.

Figure 6 shows the response to a 5Hz, 0.2V pp square wave. The output amplitude of 5.4Vpp is from the large LF gain (26.5dB maximum at 10Hz). Note the freedom from ringing characteristic of 1st-order networks. With the boost control turned to zero, the square-wave response (not shown) is virtually perfect, as is the audio band frequency (and phase) response.



1v/div

20mS/div

FIGURE 5: LF boost portion of impulse response, impulse rate = 10Hz, maximum boost.

TABLE 1 40Hz boost vs. Rp

dB at 40Hz	Rp kΩ
-.07	0
0	0.24
1	2.80
2	5.01
3	7.16
4	9.37
6	14.2
8	20.0
10	26.9
12	35.6
14	46.3
16	59.6
18	76.3
20	97.3

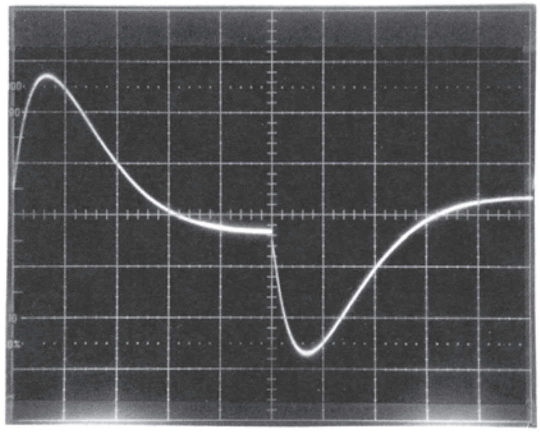


FIGURE 6: Response to 5Hz, 0.2Vpp square wave, maximum boost.

HOW DOES IT SOUND?

To my ears it provides the most natural-sounding and effective bass boost I've yet heard. I realize that it's difficult for an audiophile to be unbiased about his or her new toy. But I am unbiased; for example, I've tried many times to design the perfect speaker. But my efforts never sound quite as good as speakers designed by Joe D'Appolito, and I have no problem recognizing and admitting that.

I think a "close second" is the bass control in the Scott 222D tube amp (see my article, "A FET Standby Switch for Tube Amps," *aX* Dec. '06). On page 24 is shown the Scott amp's tone control responses. At maximum bass boost, the response is +13dB at 40Hz and +3dB at 250Hz.

EXAMPLES OF BOOST SETTINGS

With speakers or headphones flat to below 40Hz, I generally use 15-20dB at 40Hz with most rock recordings, 5-15dB with blues, 0-5dB with jazz, and no boost (except at low volume) with well-recorded classical and other acoustic bands. One exception to the latter is with pipe organs—sometimes I use the maximum boost (25dB at 20Hz) to hear and feel those low pedal fundamentals shaking me and the house! *aX*

TABLE 2 Parts list, per channel

Ref	Description	Size	Mfr.	Mfr./P/N	Dist.	Dist. P/N
C13	cap, 47pF					
C10	cap, 100pF					
C2	cap, 220pF					
C3, 4, 6, 7	cap, 10nF, film				DK	PS1103J
C5, 8, 11	cap, 100nF, film				DK	EF1104
C12	cap, 330nF, film				DK	EF1334
C1	cap, 1µF, film				DK	EF1105
C9	1µ5, film				DK	EF1155
J1, 2	RCA jack				PE	091-1120
P1	dual pot, 100k, Lin			one required for stereo unit		
R15	Res, 1%MF, 200Ω	¼W				
R3	Res, 1k00	¼W				
R5	Res, 4k64	¼W				
R12, 13	Res, 4k75	¼W				
R9-11	Res, 7k87	¼W				
R14	Res, 9k53	¼W				
R8	Res, 13k7	¼W				
R4	Res, 23k2	¼W				
R6	Res, 34k0	¼W				
R2	Res, 49k9	¼W				
R7	Res, 1% MF, 158k	¼W				
R1	Res, 1M00	¼W				
U1, 2	dual op amp			$2.63 from DK OPA2134pA		

DK: Digi-Key, PE: Parts Express

Testing Panasonic's WM-61A Mike Cartridge

This microphone study measures how distortion affects performance.

I recently had access to a high precision Bruel & Kjaer Sound Level Meter type 2230 equipped with a ½″ pre-polarized condenser microphone type 4155. This gave me the opportunity to test the Panasonic WM-61A omnidirectional back electret microphone cartridge. I had ordered the units used for testing from Digi-Key (part number P9925-ND) at the end of 2006.

This article presents the results of these tests. I investigate how a cheap microphone cartridge compares with a high precision, high price (at least for the audio amateur) microphone. The article is divided into two parts: first, I tested the microphones as provided by the manufacturer, and, second, I modified the Panasonic microphones according to Mr. Linkwitz's recommendation. The purpose of this modification is the reduction of the microphone distortion at high SPL levels.

I placed all the cartridges on the front side of an aluminum tube, which had an inner diameter of 7mm, an outer diameter of 10mm, and a length of about 10cm. A complete Panasonic microphone together with the Bruel & Kjaer Sound Level Meter is shown in **Photo 1**.

I performed the evaluation of the microphones with the following measurements:

A) For the standard Panasonic microphones
Test A1: the low-frequency response of the standard microphone.
Test A2: the middle- and high-frequency response of the standard microphone.
Test A3: the phase response of the standard microphone.
Test A4: the distortion of the standard microphone at the level of 115 dBSPL.

B) For the Linkwitz modified Panasonic microphones
Test B1: the low-frequency response of the modified microphone.
Test B2: the middle- and high-frequency response of the modified microphone.
Test B3: the phase response of the modified microphone.
Test B4: The distortion of the modified microphone at 125 and 130 dBSPL levels.

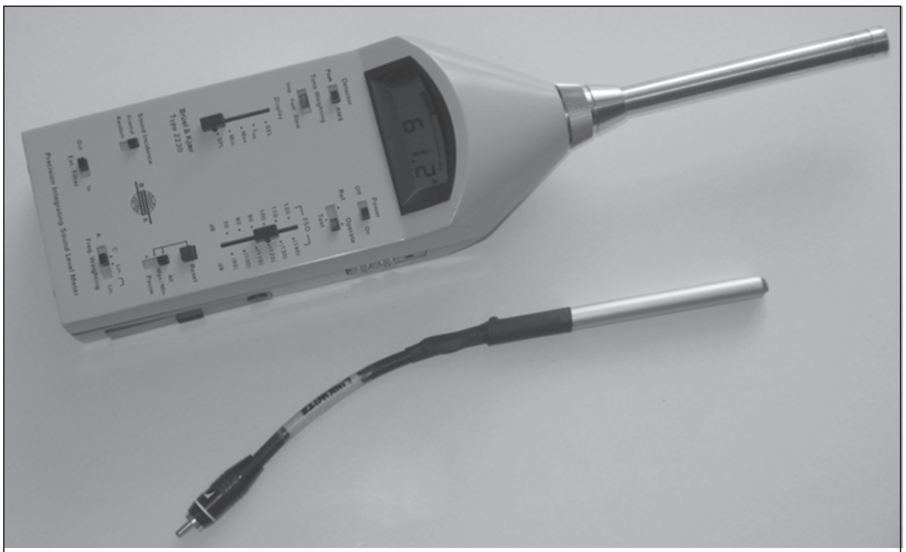


PHOTO 1: The Bruel & Kjaer sound level meter and the microphone with the Panasonic cartridge.

THE TESTING METHOD

For this simple testing method, I used an acoustic source and measured the frequency response with the reference B&K microphone. Then I used the same source to measure the response of the uncalibrated Panasonic microphone. I compared the two measurements and determined the response of the uncalibrated microphone.

I placed the reference B&K microphone and the Panasonic microphones as close together as possible. Actually, the Panasonic microphone was supported on the body of the B&K microphone as shown in **Photos 2** and **3**.

I also performed some measurements for the evaluation of the test method. I took two frequency response measurements of the Bruel and Kjaer microphone: the first with the B&K alone, and the second with the Panasonic microphone supported on the B&K in order to determine the influence of the Panasonic microphone to the response of the B&K. The result is shown in **Fig. 1.** The two responses are very close. Differences exist in the range from 5 to

10kHz with a peak about +0.6dB and in the range from 15 to 20kHz, which is down about -0.5dB.

According to the calibration chart provided by Bruel & Kjaer for this specific microphone, the response is absolutely flat up to 11kHz. Then the response rises with a peak of +1.2dB at 16kHz. The response is flat again at 20kHz. Based on this, I conclude that the following measurements will have an accuracy better than ±1dB.

For the testing of the standard microphones I used four samples. I connected the microphones to a preamplifier that provided a DC supply voltage to the microphone of 4.1V and had a gain of 16dB. It is a modified version of the original Mitey-Mike preamplifier (*Speaker Builder* 6/90, p. 12).

TEST A1

The test setup is shown in **Photo 2**. Each Panasonic microphone was tied together with the B&K microphone. For this test I used an 8″ woofer without a box or baffle as the acoustic source. The microphones were close to the center of the woofer, and I took a frequency response measurement for every microphone. **Figure 2** shows the results.

The responses of samples 1 and 2 were very close to the B&K with a maximum deviation of 0.5dB at 10Hz. Samples 3 and 4 had a maximum deviation of 0.7dB at about 40Hz, but otherwise they were close to the B&K except at the very low frequencies down to 10Hz where the deviation is about 1 to 1.3dB.

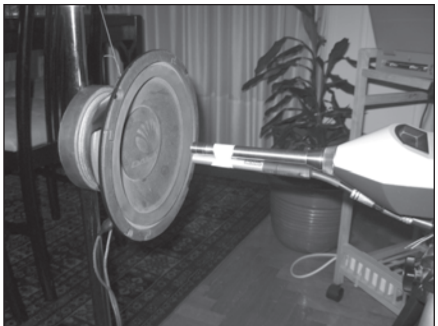


PHOTO 2: The test setup for the measurement of the low-frequency response using an 8″ driver.

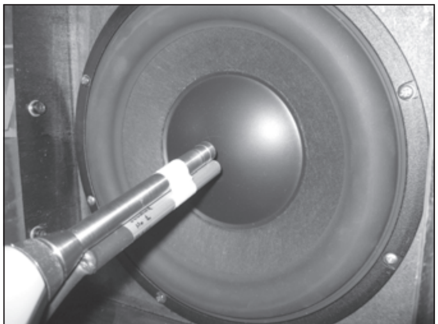


PHOTO 3: The test setup for the measurement of the low-frequency response and the distortion using a 10″ driver.

TEST A2

I performed this test by using a loudspeaker with an Audax 130mm woofer/midrange and an Audax TW025 tweeter as the acoustic source. The microphones were 1m from the loudspeaker.

The height of the microphones was about 1.2m from the floor. This setup gave me an impulse response that was reflection free for the first 3 milliseconds. The results of the tests are shown in **Fig. 3**. All the microphones have similar responses, which are flat up to about 2kHz. Then they measure +2dB at about 6kHz, +3 to +4dB at about 10kHz, and finally a broad peak of about 3.5 to 5dB at about 12 to 14kHz.

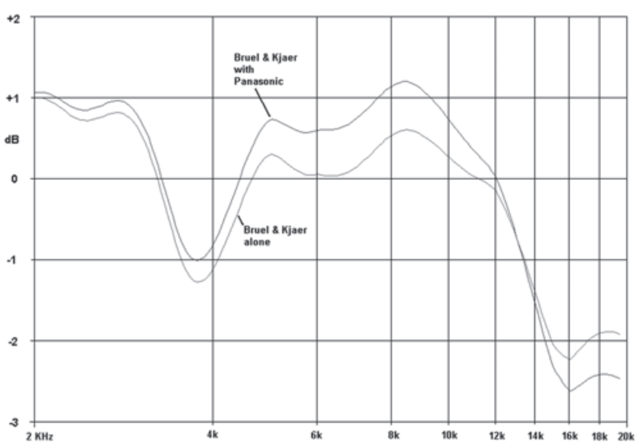


FIGURE 1: Differences in the responses of the Bruel & Kjaer microphone due to the proximity of the Panasonic microphone.

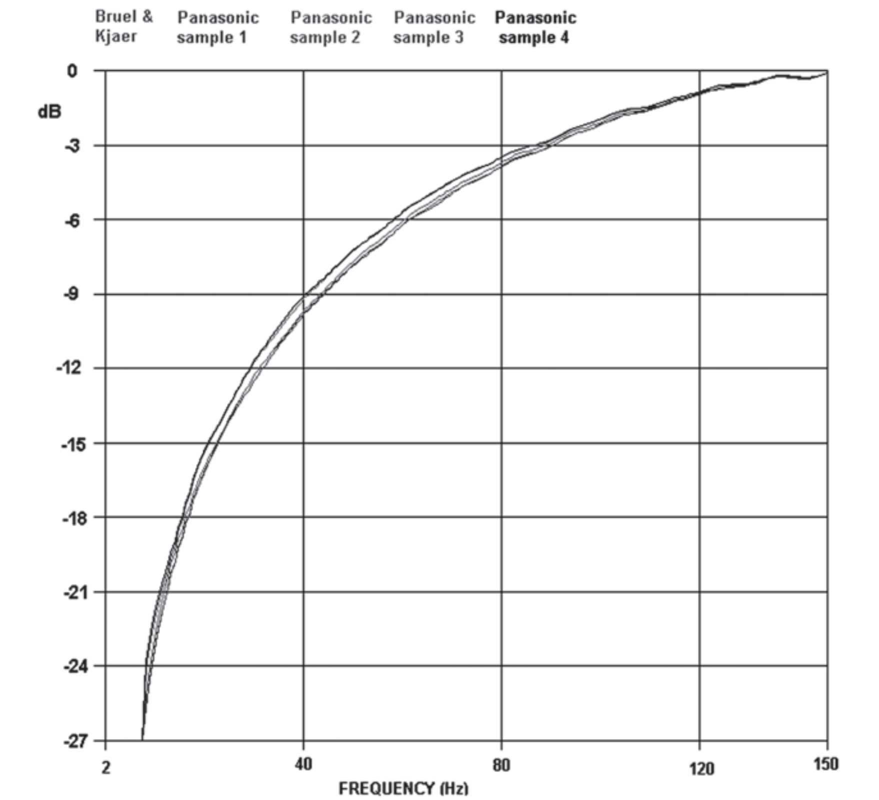


FIGURE 2: The low-frequency response of the standard microphone.

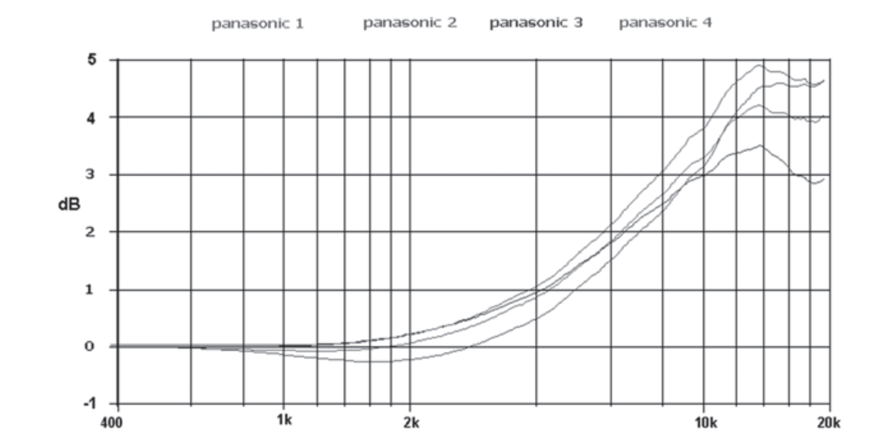


FIGURE 3: The middle- and high-frequency response of the standard microphone.

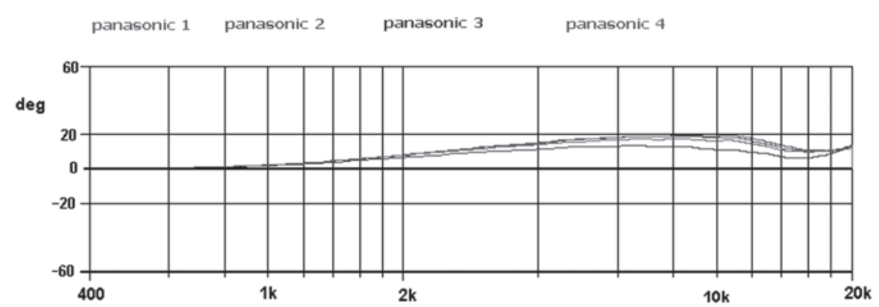


FIGURE 4: The phase response of the standard microphone.

TEST A3

The phase response was measured with the same setup as for the test A2. The results are shown in **Fig. 4**. The phase is flat up to about 1kHz, then +10° at about 2.5kHz, and +20° in the range from 6 to 12kHz.

TEST A4

According to the technical specifications of the B&K microphone, its second harmonic distortion is 0.13% at the level of 120 dBSPL, 0.23% at 125 dBSPL, and then increases linearly up to 4% at 150 dBSPL. The third harmonic distortion is 0.06% at the level of 140 dBSPL and increases linearly up to 0.6% at the level of 150 dBSPL. No data is given for the distortion at higher harmonics.

The measurement of the distortion of the Panasonic microphones at high levels is not an easy task because I don't have an anechoic chamber available. The measurement should be unaffected (as much as possible) by the room influence, so the only solution to this problem is the near-field measurement. Each Panasonic microphone was tied up with the B&K microphone (as shown in **Photo 3**) and placed very close to the center of the cone of the loudspeaker, which was used as the acoustic source.

I used different speakers according to the frequency. For the low frequencies of 15, 58, and 165Hz, I used a 10″ Peerless XLS woofer; for the frequencies of 580Hz and 2kHz, I used an Audax 130mm woofer/midrange; and for the frequency of 4k an Audax tweeter. I measured the distortion for all frequencies at the level of 115-dBSPL-peak. The level was monitored for each measurement with the B&K Sound Level Meter. Above that level the distortion of the Panasonic microphone increases rapidly and above about 120 dBSPL you cannot use the mike.

The results are shown in **Table 1**, from which you see that the distortion of samples 1 and 4 is much better than the distortion of samples 3 and 4. At 15Hz the distortion of the microphones is close to the distortion of the B&K. As the frequency increases, the distortion of the Panasonic also increases and the average distortion is from 2 to 5

TABLE 1 Distortion of Standard Mikes at 115 dBSPL Peak						
Hz	Bruel & Kjaer (%)	Average of the four Panasonics (%)	Panasonic sample 1 (%)	Sample 2 (%)	Sample 3 (%)	Sample 4 (%)
15	5.3	6	5.9	6.8	5.5	5.8
58	0.24	0.6	0.24	1.2	0.7	0.24
165	0.16	0.71	0.5	0.74	1	0.6
580	0.16	0.65	0.4	1	0.8	0.4
2k	0.32	0.81	0.65	0.84	1	0.7
4k	0.17	0.82	0.5	1.1	1.1	0.6

times worse than the distortion measured by the B&K.

An interesting note is that the distortion of samples 1 and 4 at the frequency of 58Hz is comparable to the distortion of the B&K. Conclusively, the distortion levels of the Panasonic microphones are good at a level of 115 dBSPL.

MODIFIED MIKE TESTS

The modification of the microphones is described on the S. Linkwitz site (www.linkwitzlab.com/sys_test.htm#Mic). The purpose of this modification is to reduce the distortion of the original Panasonic microphones by changing the connection of the capsule to the built-in FET amplifier stage.

This mod is not very easy to perform and requires some delicate work. Expect to spend some time and some cartridges before you succeed with this modification.

Here are my recommendations:

a) Avoid heating the cartridge. Use a low temperature when soldering the cables on the capsule.
b) Don't solder the ground cable to the body of the capsule. The heat may destroy the capsule. I use conductive epoxy to connect the wire on the body of the capsule.
c) Cover the whole backside with glue to avoid low-frequency rolloff due to a possible broken air seal, which might have been caused by the modification (Linkwitz recommendation).

Two problems might appear with the modified microphones: either some loss in the low-frequency response or some loss in the sensitivity of the microphone. You must check both for a properly working microphone.

I modified four Panasonic microphones for the tests. The microphone preamplifier was similar to the one described in the Linkwitz site. The gain was 15dB and the DC voltage supply to the microphone was -9V. (A negative supply voltage is required after the modification of the microphones.)

TEST B1

I used the setup shown in **Photo 3**. The speaker was a 10″ Peerless XLS woofer in an open baffle. The microphones were, again, at the near field of the woofer, close to the center of its apex.

I took a frequency response measurement for each microphone in comparison with the B&K microphone. **Figure 5** shows the results. The responses of

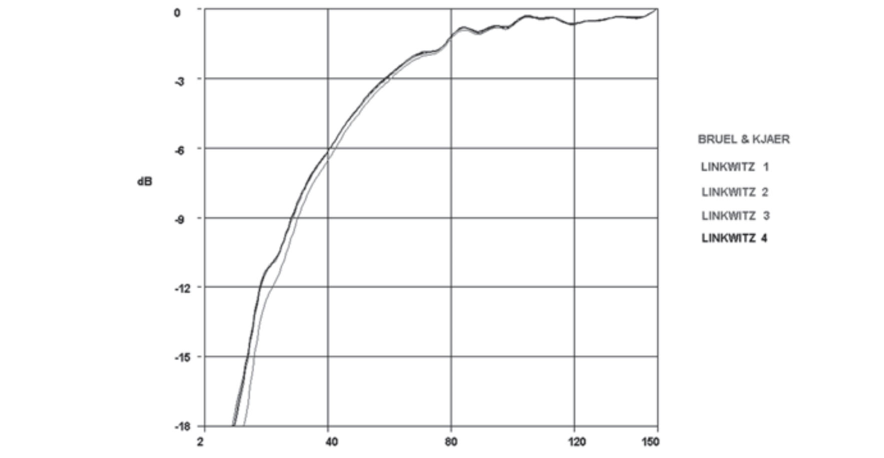


FIGURE 5: The low-frequency response of the modified microphone.

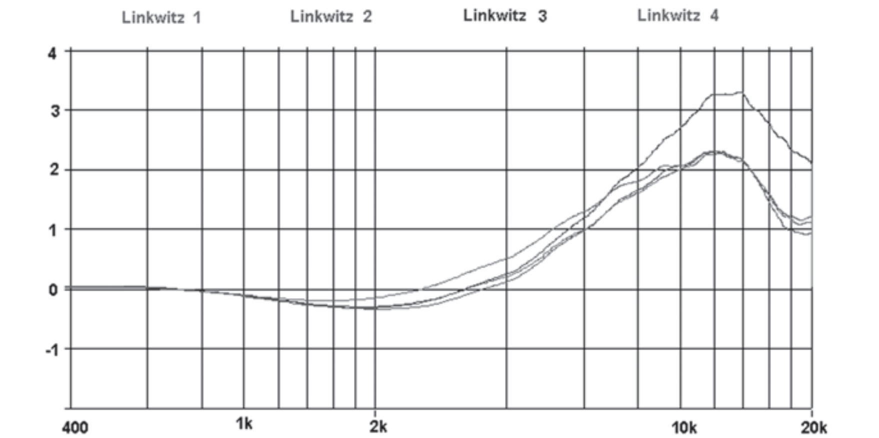


FIGURE 6: The middle- and high-frequency response of the modified microphone.

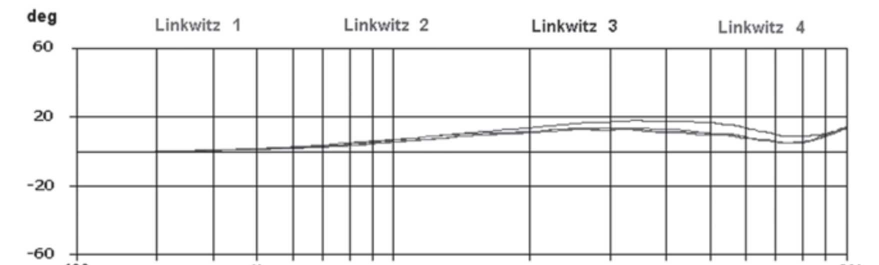


FIGURE 7: The phase response of the modified microphone.

TABLE 2 Distortion of the Linkwitz-Modified Mikes at 125 dBSPL Peak

Hz	dBSPL	Bruel&Kjaer (%)	Average Panasonic (%)	Linkwitz 1 (%)	2 (%)	3 (%)	4 (%)
15	125	44	48.3	55.2	46	46	46
58	125	0.86	0.77	0.8	0.7	1	0.6
165	125	0.26	0.61	0.7	0.6	0.34	0.8
580	125	0.29	0.34	0.36	0.33	0.26	0.4
2k	125	0.45	0.87	0.9	0.83	0.74	1

TABLE 3 Distortion of the Linkwitz-Modified Mikes at 130 dBSP

Hz	dBSPL	Bruel&Kjaer (%)	Average Panasonic (%)	Linkwitz 1 (%)	2 (%)	3 (%)	4 (%)
165	130	0.26	0.61	0.7	0.6	0.34	0.8
580	130	0.56	0.54	0.56	0.53	0.47	0.6

samples 2, 3, and 4 are within ±0.1dB from the B&K response down to 20Hz and go to -1dB at 10Hz. Sample 1 is not as good and has a deviation of -0.4dB at 40Hz, -1.5dB at 20Hz, and -3.7dB at 10Hz.

Maybe the backside of this microphone was not covered very well with the glue that I used, resulting in an air leak. According to Linkwitz, this could result in some loss at the low frequencies.

TEST B2

This setup is similar to the one described for test A2. The results are shown in **Fig. 6**.

As you can see, the peak in the response is much lower than the peak of the standard microphones. Because I didn't measure the microphones before the modification, I cannot tell whether this is due to the modification.

The responses of the four samples are flat up to about 1kHz, then down –0.3dB at 2kHz and +0.3dB at 4kHz. Above this frequency the response rises with a peak of +2.3dB at about 12kHz for samples 1, 2, and 4 and +3.2dB for sample 3. At 20kHz the deviation is from +1 to +2dB.

TEST B3

The test setup is similar to the one described for test A2. The phase response is shown in **Fig. 7**. It is flat up to 1kHz, then about +10° at 4kHz, and +11 to +17° from 8kHz and above.

TEST B4

The measurement of the distortion of the modified Panasonic microphones was a difficult job. In addition to the problems already described for the standard microphones, I also needed to contend with the higher sound level. Because I didn't have access to an anechoic room, I performed all the measurements in my typical listening room. And although I used the near-field method, the level of the sound was very uncomfortable (even using earplugs).

I measured the distortion of the microphones at 15Hz, 58Hz, 165Hz, 580Hz, and 2kHz at a sound level of 125-dBSPL-peak. Additionally I measured the distortion at 165Hz and 580Hz at a level of 130-dBSPL-peak.

The sound level was always monitored with the B&K meter.

For the frequencies of 15, 58, and 165Hz, I used a 10″ Peerless XLS woofer, and a 130mm Audax woofer for the frequencies of 580Hz and 2kHz. The results for the measurement at 125-dBSPL-peak are given in **Table 2**. Indeed, the modification is very effective because the distortion of the modified microphones at this high level is very good. Actually, it is about the same with the B&K microphone at 15Hz, 58Hz, and 580Hz, and about double at the other frequencies.

The results of the measurement at the level of 130-dBSPL-peak are given in **Table 3**. Again, the results are excellent, especially for sample 3. All the modified microphones at 580Hz have distortion comparable or even better than the B&K microphone.

CONCLUSION

Keep in mind that all the measurements presented in this article were obtained with samples from a single order, so I assume that the microphones are from the same manufactured lot. It is possible that microphones from different lots may present different results. Having said this, here are my main conclusions from the tests:

a) You can use the standard microphones without any calibration for frequencies up to about 2 to 3kHz. Above this limit, if you require accuracy, the microphones should be calibrated.

b) The distortion of the standard microphones is acceptable, providing that the peak sound pressure level is always below 115-120 dBSPL.

c) You can use the microphones modified according to Linkwitz without any calibration for frequencies up to about 4 to 5kHz. Above this limit, if accuracy is required, the microphones should be calibrated.

d) The Linkwitz modification offers a dramatic improvement on the distortion, making the microphone suitable for near-field measurements. aX

By Iain McNeill

A Mains Analyzer—How Clean Is Your Juice? Part 2

With the design complete, it's time to build and test the PC boards.

Once I had completed the design with simulations and procured all the parts, I started playing around with a box layout. I put the outlet sockets on the front and the four wall wart transformers at the back. Then I laid out the shunts in the middle with the prototype circuit board and squared everything off by putting the audio transformer on the left.

Checking the size of the panel meters, I saw that they would fit above the shunts, in between the outlets and the transformers. I then drew a line around all the parts, added ½″ to all dimensions for good luck, and started looking for a suitable box. Once the box was on order, I set about assembling the PCB.

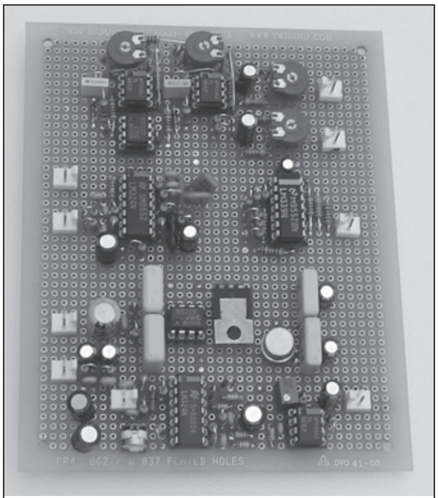


PHOTO 1: Prototype PCB, top side.

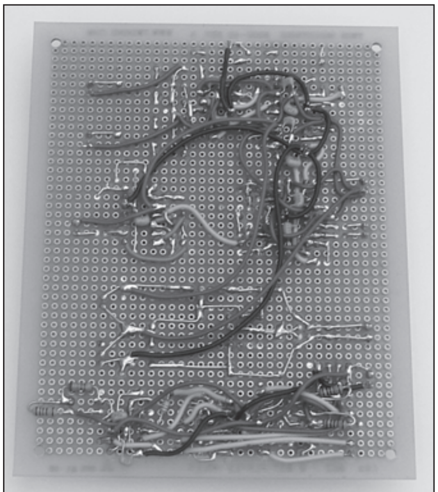


PHOTO 2: Prototype PCB, bottom.

ASSEMBLY

My preferred technique for building prototype circuit boards is using matrix or vector board, as with many of the projects in *audioXpress*. I use the small round plated-through-hole variety because I find this most resistant to accidental shorts yet flexible enough to simplify wiring. Starting with an IC or other active device and placing the Rs and Cs right next to their appropriate IC pins, I can achieve quite high component density (**Photo 1**). This is good for sensitive circuits because it minimizes loop areas and therefore EMI susceptibility and emissions.

I always take care to make sensitive nodes such as op amp inputs as short as possible by placing those components first. A quick mock-up on grid paper in pencil can speed up assembly and allows you to try a couple of options without having to get out the (de)soldering tools. I like to use leaded components because you can use the leads to actually wire up the circuit, bending and forming them just like real PCB traces.

Once all the components are inserted, soldered, and wired, I usually find that the only wiring left is power and ground. OK, maybe there're a few signal wires to run, but these are usually inter-stage connections, and using a neutral color such as yellow or white allows easy identification of these useful nodes for debugging. The prototype board I built is shown in **Photos 1** and **2**.

The only drawback to this prototyping technique occurs when you must change a component or, heaven forbid, add or redesign a stage. Not that this ever happens to me! It's always the component with a really complex lead formation that's soldered in many places that needs to be replaced, and usually I just clip the part off at the component body and solder the new one on the old component leads. Easy! However, judicious simulation beforehand pays big dividends at this stage.

Here's an important point regarding creepage and clearance: As with all high voltage designs, beware of lead-to-lead spacing to ensure you don't suffer from catastrophic flashover. The breakdown voltage for air is approximately 25kV per inch, or about 1000V per millimeter, which makes it seem quite easy, but this specification is for DC; when AC voltages are present you must use the peak voltage. You also need to consider that the mains voltage of 120V has a tolerance to it; I like to assume ±20%, so 120V AC RMS nominal becomes 212Vpk worst-case.

It's probably very unlikely, but not impossible, that the ground wire on the outlet under test could become connected to a different phase under a fault condition. I don't like surprises, so I decided to design for this event, and thus the worst-case peak voltage becomes 424Vpk. Double it, just in case it's a damp morning, and you want to withstand almost 1000V—now you can be confident that you won't have any flash-over. The pad to pad separation on my vector board was a bit less than a millimeter, so I made sure that there were two rows of unused pads between the line voltage and current measurement circuit and the ground leakage measurement circuit.

Once my box arrived, I was able to do a dry run and see how all the parts went together. The front panel was predetermined for the mains outlets, so I measured everything and then produced a 1:1 scale drawing, centering the outlets on the front panel. I printed it out on my printer, taped it to the front panel, and drilled the screw holes and the four corners of each cutout. With a sharp blade I scored the outline of the cutouts using the corner holes and then used a Dremel to cut along the lines. Not as good as a punch or a nibbler, but sometimes you must use what you have.

Similarly, I made a drill template for the BNC connector, and after working out how much space all the wall warts would take, I drilled the five holes in the remaining space. I hot-glued all the wall warts to the back panel and left them to set.

Now I needed to align everything on the top panel. I made a template for the panel meters and added holes for four 1″ standoffs on which to mount the PCB. I then centered this complete module in the space between the wall warts, outlets, and monitor transformer to get a location on the top panel. I drilled, scored, and Dremeled as before. As you can see in **Photo 3**, one of the large holes became a bit too large—once again my trusty hot glue gun saved the day. **Photos 3–5** are pretty self explanatory, showing how it all came together.

TESTING

I started testing the PCB on its own with a signal generator and a scope, just to make sure everything was wired correctly. Using a signal of –40dBV input, I was able to see the amplifiers, filters, and the rectifiers were doing what I expected. After increasing the signal to 0dBV and sweeping the frequency, I noted that the glitch detector turned on right about 400Hz, and if I did bursts of 1V tone, the LED flashed for a good second, even catching just 1 cycle of 1kHz. It was time to go live.

Call me paranoid, but I unplugged the mains tester from the mains outlet every time I wanted to stick my hand in the box to connect or disconnect something. There's no real energy storage in the tester, so you don't need to wait for components to discharge, but I prefer to get into a calm, progressive rhythm when working with dangerous voltages. If I become tired or frustrated, I do something else for a while. I'm still alive to vouch for the efficacy of this method! However, to make the narrative less te-

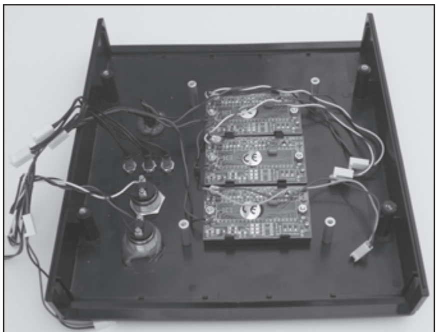


PHOTO 3: Case top assembly without PCB.

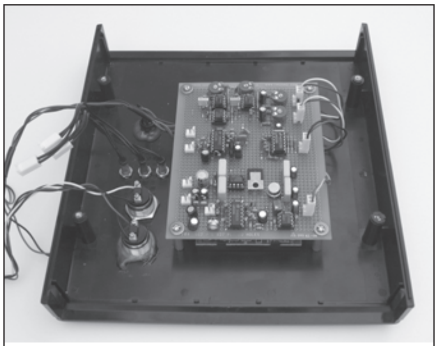


PHOTO 4: Case top assembly with PCB.

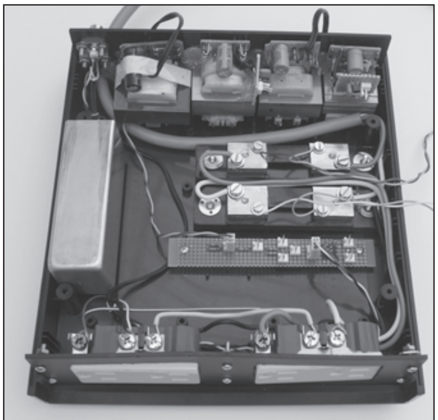


PHOTO 5: Case bottom assembly.

dious, I'll omit all my plugging and unplugging from the mains supply (trust me, there was a lot).

Setting up the meters is relatively easy. For the line voltage meter, I first set zero by removing the mains supply connector from the resistor divider board. With the 1.4kΩ resistor still in place shorting the input, I could adjust the last op amp offset voltage to zero the panel meter. I then reconnected the resistor divider board, tagged my multimeter on the live-neutral output terminals, and adjusted the gain so the panel meter read the same as my multimeter.

I zeroed the line current meter in the same way, with nothing plugged into the tester outlet sockets. I adjusted the offset pot to get a zero reading on the panel meter. To calibrate, you need to pull some current out of the tester, and I chose my soldering iron because it was handy.

Removing the neutral wire to the tester outlet sockets, I inserted my multimeter into the mains circuit, taking care not to short anything with the alligator clips and double-checking the meter jacks and settings (those 2A fuses are hard to find!). Again, I adjusted the measurement gain so the panel meter read the same as the multimeter. Setting the calibration at such a low percentage of full-scale probably isn't the most accurate way of calibrating this meter, considering the linearity problems of the rectifier stage, but my multimeter went only up to 2A so I had little choice. Besides, if a reading on a rack of equipment is approaching 20A, then it's getting too close to the circuit breaker rating anyway and corrective action is in order.

There is a bit of interaction between the zero and the cal pots on the ground leakage meter because the gain pot is ahead of the offset adjust stage. As you turn the gain up, any noise in the input signal produces an output at the rectifier stage, which you then must readjust with the offset null. A couple of iterations got me close enough, but if I were doing the project again I'd use the same scheme as the line voltage and current meters in which the cal pot is last in the chain.

All through this, the glitch detector LED has been on. Using my multimeter, I could see an AC voltage out of the filter and a DC voltage on the rectifier/RMS stage. I assumed it was 60Hz and I would need to improve the filter. That could wait for now so I chose to ignore it, not realizing that it was actually working perfectly!

OBSERVATIONS

I connected the monitor output of my mains tester to a line input on my EMU Systems soundcard and recorded a few seconds of mains waveform. After editing the .wav file to extract a couple of cycles, I plotted the mains waveform against a pure sine wave of the same amplitude and calculated the error between them as shown in **Fig. 7**. The mains waveform was obviously distorted but is still recognizable as a sine.

I wanted to see just how bad this mains waveform was, so I calculated the spectrum and plotted it against that of the pure sine wave of the same amplitude (**Fig. 8**). Both are windowed with a Blackman Harris window, and I used an FFT size of 8192 points as I did for all the spectra presented here. I then integrated all the energy from the linear FFT bins into 1/12 octave bands to get a smooth curve on a log frequency axis.

There are many fine software packages that will do this for you. CoolEdit/Adobe Audition is just one, but there's nothing quite like doing it the hard way, by hand. I use Matlab as my analysis tool of choice because it gives me infinite flexibility in how I handle the data, which is both a blessing and a curse.

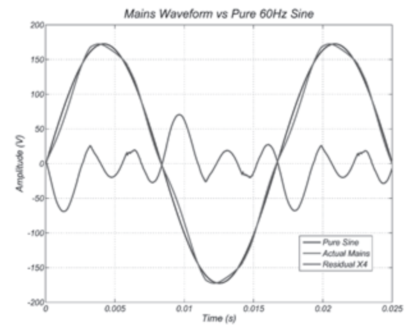


FIGURE 7: Mains waveform compared to pure sine.

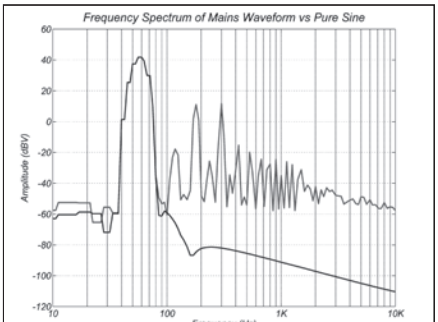


FIGURE 8: Spectrum of mains waveform vs. pure sine.

This first spectral analysis of the mains waveform on the monitor output was a real eye-opener. I stared at it dumbfounded, thinking there must be a calibration error. After checking and rechecking, I realized the math was correct. I still couldn't believe it and went straight back to my design notes to see where I'd messed up. Everything looked OK, so I ran a few more tests to try and debug this problem.

First I calibrated the monitor output circuit itself. Disconnecting the mains supply to the resistor divider R66-68, I connected the soundcard output to the top of the divider. I played a Maximum Length Sequence (MLS) pseudorandom noise signal into the divider and recorded the output on the monitor output jack. After processing the recorded signal, I plotted the monitor response (**Fig. 9**).

Everything appeared to be in order here so I put everything back together. I thought the problem may have something to do with the wiring in my house, so I made several recordings with different loads. First, I turned on my office lights (which are on a dimmer switch) to about 50% and took a reading, thinking that there would be significant noise generated by the triac switching in the dimmer. My laptop had a switching charger so I plugged that into the tester as well and taped a 3W, 100Ω resistor to the output and took another recording.

I then took readings for a 7A space heater and a 17A microwave (as measured on the line current display). Finally, I took a recording with the mains supply to the voltage sense resistor divider disconnected just to make sure I was actually measuring what I thought I was measuring. I was taking measurements using my laptop with a battery supply for the outboard soundcard so leakage currents shouldn't have been a problem, but I wanted to make sure.

The results are shown in **Fig. 10**. Reassuringly, the measurement for the background noise floor with the resistor divider disconnected was at least 30dB down at all frequencies, so it would seem that I really am measuring the mains supply. Amazingly, this background level represented 100µA of mains leakage flowing through R74, the 600Ω on the transformer output. Maybe a bigger box to try to reduce the roughly 2nF of stray capacitance causing this current would have been prudent, but I was happy with the 30dB signal-to-noise that I had.

The mains spectra for different loads through the mains tester didn't produce significantly different results. It wasn't until I got up to 17A through the box to power my microwave oven that the spectra changed noticeably. There's a big increase in the 2nd and 3rd harmonics with about 300mV RMS of 120Hz and over 2V RMS of 180Hz noise. The 3V RMS of 300Hz is relatively unchanged, although still bothersome. On this particular outlet, the line voltage dropped only 2V for the 17A load, which would suggest a mains source impedance of around 118mΩ. I was quite close to the panel, so I guess you could get maybe 5 or 10 times that on a really long run.

Everything was behaving properly and it began to sink in that this was, in fact, what was coming out my wall socket. The mains supply is noisy, really noisy. About 10V RMS noisy.

WHERE THIS HIGH-FREQUENCY NOISE COMES FROM

When you think about all the appliances connected to the mains supply, which ones present a purely resistive, linear load? I can't think of any. Even a simple incandescent lamp goes through a temperature-dependent resistance function every mains cycle. DC supplies only really conduct in short bursts when the mains supply is greater than the voltage on the smoothing capacitors and the rectifier diodes are forward biased.

Switched mode power supplies are prolific, promoted by their environmentally friendly power efficiency. They take their power in pulses at many times the mains frequency and probably contribute much noise at frequencies beyond the 20kHz limit of my analysis tools. Supply all these devices through a highly reactive grid network, and it's no wonder the mains supply is distorted.

HOW BAD CAN THINGS GET?

Is my experience normal or do I have grounds for complaint? I searched the Internet for specifications and information. Most power companies think in terms of current because this is what they have to manage; the volts are fixed, after all. Most of the specifications I have found refer to current distortion rather than voltage distortion and use a couple of different Total Harmonic Distortion (THD) definitions.

One method computes the harmonics against the fundamental, the other against the total combined signal. This latter definition, favored by the utilities, is known as Total Demand Distortion (TDD). The astute will notice that TDD can never exceed 100%—what specmanship! The reality is that high THD in the current waveform is almost the norm considering the lack of linear loads, but when the harmonic content approaches the magnitude of the fundamental 60Hz supply, it doesn't really matter whether it's 90% THD or 190% THD. The grid is being stressed and customers are probably having serious problems. The power companies must keep control of harmonic content.

I found one voltage specification from Europe, EN50160 (or maybe EN61000), that suggests the following spectral limit to mains harmonic content. The limit scales with whatever the local line voltage is, which makes sense, and is specified in terms of the harmonics. I converted the spec for 60Hz and 123V and derived the limits for my house as shown in **Table 2**. Adding this limit to the spectral plot of **Fig. 8**, I was horrified to discover that things could actually become much worse before the power

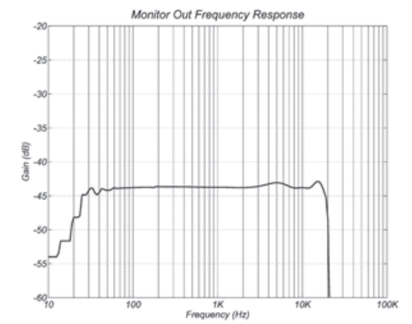


FIGURE 9: Monitor output frequency response.

company would get involved, as you can see in **Fig. 11**.

Figure 11 also shows a measurement I took at a different site—an industrial park in a factory building that was 50% powered by solar panels on the roof. Not only was there the heavy machinery of neighboring businesses, HVAC units, and office appliances, but there was also the step-up circuits, inverters, and synchronization systems of the solar generator. Add in all the noise from the electronic ballasts of the interior lighting, and you begin to understand where the extra 20dB of high frequency (>1kHz) noise comes from. The voltage waveform of this industrial supply is equally ugly (**Fig. 12**).

It turns out my residential supply was actually better than average quality! This was becoming downright depressing. The thought of actually connecting this to any of my gear made me feel queasy.

WHY SHOULD I CARE?

The most severe problems caused by this harmonic content affect the power utility company[2,3]. For instance, large capacitor banks used to correct power factor have lower reactance at higher frequency so the harmonic content causes overheating and premature failure. Multiple-of-three harmonics (3^{rd}, 9^{th}, 12^{th}, and so on), known as triplens, cause large currents to flow in the neutral line of three phase supplies causing overheating and possible failure. Transformers need to be derated to prevent overheating due to highly distorted current waveforms.

Some problems that affect consumers

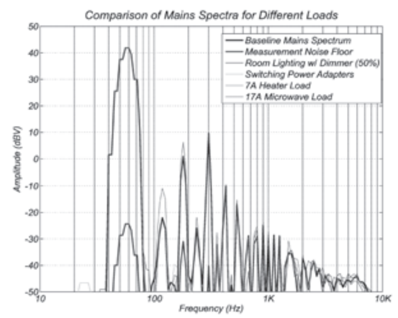


FIGURE 10: Spectra of mains waveform for various loads.

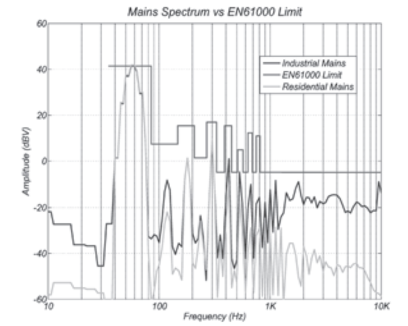


FIGURE 11: Mains spectra for industrial and residential service vs. EN61000 limit.

Table 2

Harmonic order	relative voltage(%)	frequency (Hz)	limit (dBV)
1	100	60	41.8
2	2	120	7.8
3	5	180	15.8
4	1	240	1.8
5	6	300	17.4
6	0.5	360	-4.2
7	5	420	15.8
8	0.5	480	-4.2
9	1.5	540	5.3
10	0.5	600	-4.2
11	3.5	660	12.7
12	0.5	720	-4.2
13	3	780	11.3
14	0.5	840	-4.2
15	0.5	900	-4.2
16	0.5	960	-4.2

in the home: nuisance tripping of circuit breakers and fuses has been attributed to high harmonic content[2,3] of the mains supply. The harmonics more easily couple capacitively into adjacent wired networks such as the telephone system, causing irritating buzz on phone calls or, more problematically, degrading DSL connections due to high error rates. The high neutral current problem in three phase systems can manifest itself as a significant voltage between neutral and ground, which exacerbates common mode hum problems. If I were starting this project knowing what I know now, I would definitely add a fourth meter to measure this neutral-ground voltage.

The biggest concern from my perspective is the additional burden on the power supply design to make sure all this high-frequency noise isn't propagated into any precision, sensitive audio electronics. Having designed many power supplies, crossovers, and filters, I know how hard it is to get deep attenuation with real-world passive components. With some trepidation this time, I got out my trusty napkins to see whether I was in trouble.

Assume for now that your speaker puts out 90dBSPL (at 1m) for an input signal of +9dBV (2.83V) and that you want the mains hum to be inaudible, say below 30dBSPL. Then the total RMS noise level input to the speaker must be below –51dBV. From my measurements, with multiple harmonics at 10dBV, I'd say you need a good 70dB attenuation from mains input to speaker output to ensure this. Just to illustrate this by means of a typical 100W class B transistor amp, that attenuation is provided by the following means.

The mains transformer steps down all voltages by a factor of 4, or -12dB. The power supply filter capacitors provide only 20dB of attenuation due to the fact that the Effective Series Resistance (ESR) of most large caps is of the order of hundreds of milliohms versus transformer secondary source impedance of an ohm or so. The remaining 38dB of attenuation must be provided by our friend negative feedback; by comparing the output voltage to the clean (you hope) input voltage and driving the output transistors with an error signal, the output transistors can be made to block any noise on the power supplies. This is PSRR, and a careful design can achieve upwards of 60dB.

In my house, with my class B amplifiers, it looks as though I have a small margin (my speakers are quiet). But what about a tube amplifier with a step-up power supply? Now the harmonics are gained up by the mains transformer. Fortunately, in this case the currents flowing are much reduced so the supply source impedance can be increased.

For instance, adding 10Ω in series with the HT line before the filter capacitor will achieve an extra 20dB attenuation, with the ESR of the filter cap being unchanged. But in a design with no feedback, you need to ensure that somehow you provide the remainder of the required attenuation. I haven't played with feedback-less tube designs, but I now have a whole new respect for them and their designers!

Another major red flag goes up for class D amps. In class D you typically switch the output transistors hard on and hard off at a very high frequency and then filter out as much of the high-frequency noise as possible. You can't filter out any audio frequency noise because that would defeat the object of the amp.

The class D amps I have seen connect the power supply, noise and all, to the output with a very low impedance, typically just a few ohms. That's only a couple of dB of PSRR for a typical 8Ω speaker. This implies that any noise on the power supply of a class D amp effectively connects directly to the output. Buyer beware!

Finally, preamplifiers for low-level transducer input are even more sensitive. For a low impedance moving magnet cartridge putting out 500µV or –66dBV,

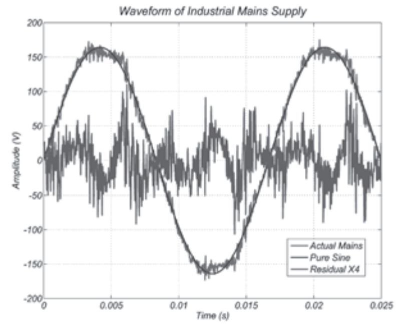


FIGURE 12: Industrial mains waveform vs. pure sine.

you need an impressive 160dB PSRR from your mains supply. This makes a compelling argument for batteries, especially if you factor in mains leakage current effects as well.

WHAT NOW?

Having been alerted to the fact the wall outlet is really a conduit for broadband noise, I feel obliged to take a look at the spectrum up to the RF bands. My soundcard samples up to 192KS/s, which will give me almost 100kHz bandwidth, although I'm not sure how wideband my audio transformer is. Beyond that, I'll need to visit some of my radio engineer friends and borrow a real spectrum analyzer. I'm sure there's some interesting surprises higher up the spectrum just waiting to be discovered.

Never again will I be satisfied with a simple transformer, bridge, capacitor power supply—it's just not good enough for audiophile-quality electronics. The

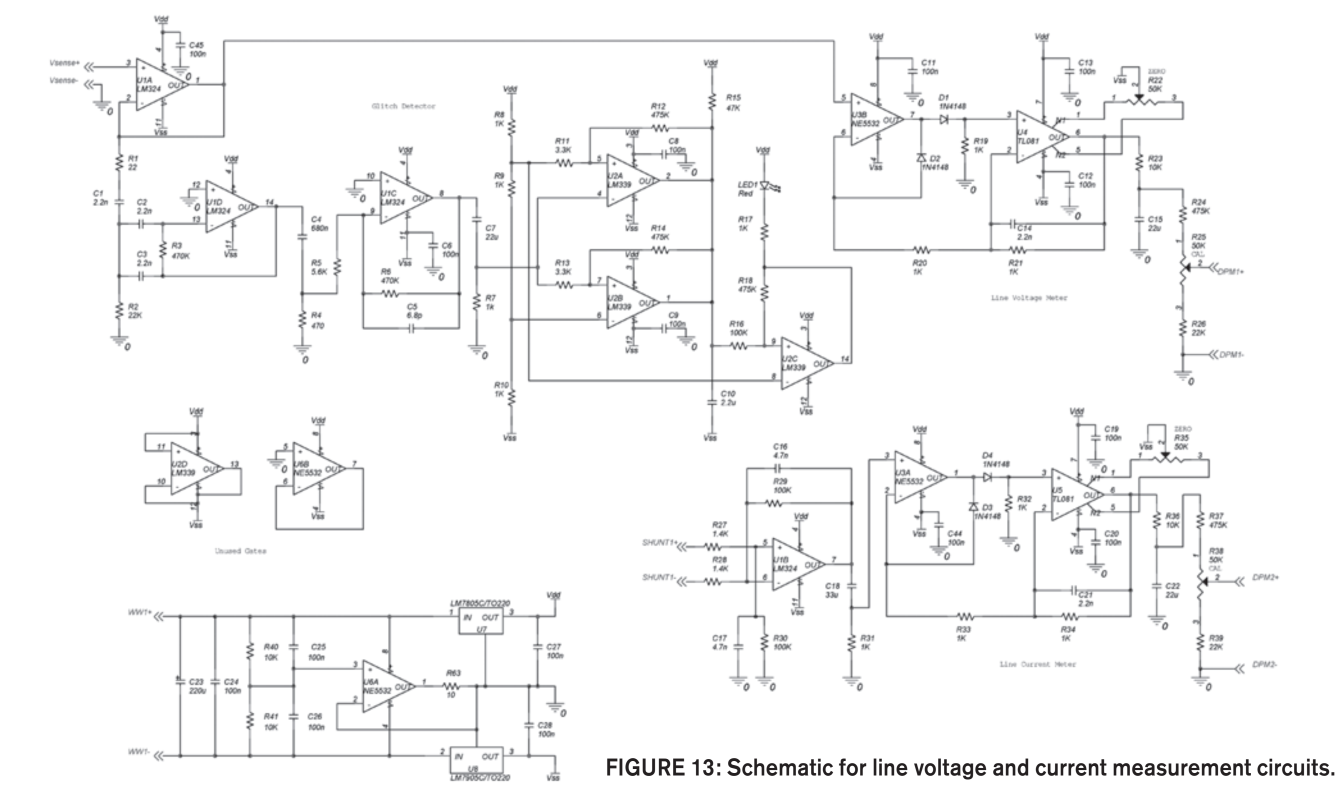


FIGURE 13: Schematic for line voltage and current measurement circuits.

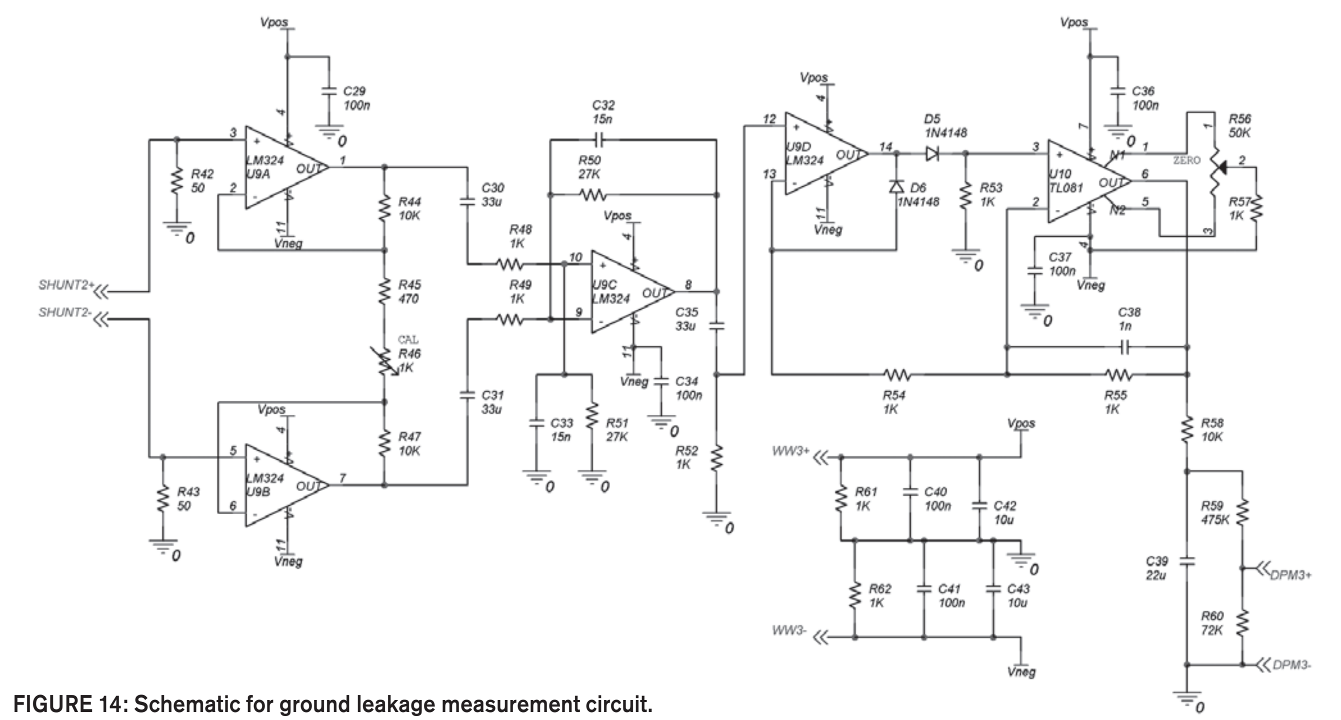


FIGURE 14: Schematic for ground leakage measurement circuit.

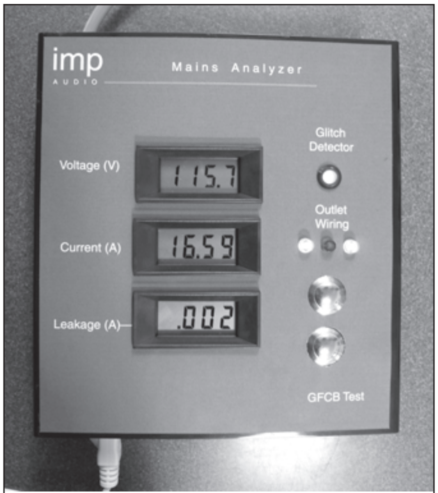


PHOTO 6: The completed Mains Analyzer in action.

power supply for the 21st century needs to provide appropriate impedance to the mains supply from DC to light and much more broadband attenuation than a transformer-capacitor can provide. I should no longer contribute to the problem, so I want to research power-supply designs that present a purely resistive load, keeping the current waveform sinusoidal and in-phase with the voltage waveform. Noble goals, I'm sure.

Finally, for the mains tester, when I get a spare afternoon, I'm going to drop the impedances around the line current input amplifier to reduce the rectifier offset due to mains leakage. I think R27, 28 = 50Ω, R29, 30 = 3300Ω, C16, 17 = 120nF will help center the zero pot (it's way over to one end on mine) while keeping the gain structure intact. I'm also considering changing the ground leakage current meter to be a ground-neutral voltage meter, although it may be better to just add a switch to the existing voltage meter—the jury's still out on that idea. In the meantime, I'm going to collect some data on ground leakage currents in typical installations.

There's also the open question of what's the right threshold for the glitch detector. At the moment it seems good for my house. On a quiet day (electrically speaking) it goes out and I can make it go on and off by turning on my office lights. But then I know I have a particularly quiet supply. aX

REFERENCES

2. Power System Harmonics: Causes, consequences and potential financial impact. www.npl.co.uk/electromagnetic/clubs/dclfclub/meetings/7_12_05_meeting/j_milanovic.pdf.

3. Power System Harmonics, PG&E Power Quality Note. www.pge.com/docs/pdfs/biz/power_quality/power_quality_notes/harmonics.pdf.

PARTS LIST

Reference	Description
R63	10Ω, 1%, ¼W
R1	22Ω, 1%, ¼W
R42, 43, 68	50Ω, 1%, ¼W
R4, 45	470Ω, 1%, ¼W
R74	600Ω, 1%, ¼W
R67	845Ω, 1%, ¼W
R7-10, 17, 19-21, 31-34, 48, 49, 52-55, 57, 61, 62	1kΩ, 1%, ¼W
R27, 28	1k4Ω, 1%, ¼W
R11, 13	3k3Ω, 1%, ¼W
R5	5k6Ω, 1%, ¼W
R23, 36, 40, 41, 44, 47, 58	10kΩ, 1%, ¼W
R2, 26, 39	22kΩ, 1%, ¼W
R50, 51	27kΩ, 1%, ¼W
R15	47kΩ, 1%, ¼W
R69-71	68kΩ, 1%, ¼W
R60	72kΩ, 1%, ¼W
R16, 29, 30	100kΩ, 1%, ¼W
R3, 6, 12, 14, 18, 24, 37, 59	470kΩ, 1%, ¼W
R46	1kΩ trimpot (any)
R22, 25, 35, 38, 56	50kΩ trimpot (any)
R64, 65	25A shunt, 50mV (2mΩ)
R66	40kΩ, 5%, ½W (see text)
R72, 73	10kΩ, 5%, 3W (see text)
C5	6p8F, 5%, 16V, ceramic
C38	1nF, 5%, 16V, ceramic
C1-3, 14, 21	2n2F, 5%, 16V, ceramic
C16, 17	4n7F, 5%, 16V, ceramic
C32, 33	15nF, 5%, 16V, ceramic
C6, 8, 9, 11-13, 19, 20, 24-29, 34, 36, 37, 40, 41, 44, 45	100nF, 20%, 16V, poly
C4	680nF, 10%, 16V, poly
C10	2µ2F, 20%, 16V, electrolytic
C42, 43	10µF, 20%, 16V, electrolytic
C7, 15, 22, 39	22µF, 20%, 16V, electrolytic
C18, 30, 31, 35	33µF, 20%, 16V, electrolytic
C23	220µF, 20%, 16V, electrolytic
LED1	LED, red, ultrabright
D1-6	1N4148, 100V, 200mA signal diode
U1, U9	LM324, quad op amp, 14 pin DIP
U2	LM339, quad comparator, 14 pin DIP
U3, 6	NE5532, dual op amp, 8 pin DIP
U4, 5, 10	TL081, single op amp, 8 pin DIP
U7	78M05, +5V regulator, TO220
U8	79M05, -5V regulator, TO220
U11, 12	78L09m +9V regulator, TO92
J1, J2	dual outlet, 3 prong
J3	BNC connector, chassis mount
WW1-4	AC adaptors (wall warts), various (see text)
T1	audio transformer, 600Ω, 20Hz – 20kHz
V1, V3	neon, amber, chassis mount with resistor
V2	neon, amber, chassis mount with resistor
SW1, SW2	pushbutton, chassis mount
DPM1-3	digital panel meter, 3½ digit, 200mV f.s.

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TUBES

SOUND SOLUTIONS

AUTHOR INDEX

STATEMENT OF OWNERSHIP, MANAGEMENT, AND CIRCULATION

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Total Distribution	7,470	9,217
Copies not distributed	1,947	1,783
Total	9,417	11,000

I certify that the statements made by me above are correct and complete. Publication No. 787-840. Edward T. Dell, Jr., Publisher.

VENDORS

Yard Sale

Wanted

JVC AXZ-911BK 100W per channel amp
JVC FX-1100BK FM tuner
JVC RX-222 Digital equalizer
Pioneer PL-3F turntable
Kenwood KD-35R turntable
Onkyo TA-2800 cassette tape deck
Pioneer CT-A9X cassette tape deck
Pioneer F-99x digital tuner
Optimus Mach II home stereo speakers by Radio Shack
Pioneer CS-707 home stereo speakers
Pioneer A-88x, BK 120W per channel integrated amp
Onkyo P-304 preamp
Onkyo M-504, 100W per channel power amp
Luther Wright
16901 Parkdale Road
Petersburg, Va. 23805

I need to contact someone who knows vacuum tube fundamentals, values, and so forth. Also someone who knows bandpass filter designs (experimental project).
Kendrick Sellen
164 South Main St.
Carbondale, Pa 18407-2655
kendricksellen@hotmail.com

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Ad Index

THOR TWEAKS

I have a pair of Joe D'Appolito Thor transmission line speakers with the Madisound upgraded crossovers mounted in the bottom cavity. After reading a letter by N. S. Francis of Norwich, England, in *Hi-Fi News* (May 2005), I decided to look into the cavity in the bottom of the Thors with the idea of trying the paving stone block trick.

I started out by making two concrete blocks about 30mm bigger than the bottom profile of the Thors with holes corresponding to where the spikes were. These blocks were about 50mm thick. I then left the concrete blocks to dry for some months, and then painted them satin black.

Unscrewing the crossovers from the Thor enclosures, I placed them on some foam plastic about 20mm thick; I also filled the bottom cavity with some dry sand in plastic supermarket bags. Running some BluTack around the edge of the bottom of the Thors and removing the spikes, I bolted the concrete blocks to the bottom of the Thor enclosures using the spike T-nuts to bolt into.

The result was:
Much tighter, better defined bass
Much better soundstage and stereo image
A much more detailed midrange.

This is the same result that N. S. Francis discovered in May 2005. I am unsure what, if any, effect vibration has on crossover components, but I suspect it may have some. Removal of the open cavity certainly does.

This has also added quite a lot of mass to the enclosures, which were pretty non resonant anyway, being made of solid 100+ year-old recycled Rimu—New Zealand native hardwood. A very worthwhile cheap improvement to a fine-sounding speaker.

Alan Smith
Dunedin New Zealand
jbandaj@clear.net.nz

CDs used:
Benjamin Britten, Variations on a theme of Frank Bridge Naxos 8557200
Jennifer Warnes, The Well SD8960
Jacques Loussier, Trio Telarc CD83544

CHIP UPDATE

As a followup to my article ("High End 120W MOSFET IC Driven Amp," June '07, p. 30), I have been corresponding off and on with Troy Huebner and Mark Brasfield from National Semiconductor, which offers a very comprehensive application note on the use of the LM4702 and MOSFETs (AN-1645). There is also a new driver chip introduced by National which will source much more current—the LME49810.

Jack Walton
Short Hills, N.J.

PHONO PREAMPS

I've been a fan of Dennis Colin since I built my first Aries synth module back in the late 70s. I recently became interested in electronics again after needing a high-gain low-noise mike preamp for field recording. So I fired up my old Weller and built a preamp based on the THAT 1512 op amp. The breadboard version sounded very good, but the perfboard version had a lot of noise at high gains—I suppose due to the lack of a ground plane.

Just as I was building my mike preamp, I downloaded and read your article "The LP 797 Ultra-Low Distortion Phono Preamp" (Sept. '07, p. 6). I was intrigued with the description of your construction techniques—especially after stumbling across a lot of info online regarding "Ugly Construction"—a technique used by many amateur/ham radio builders. The article included some photos of your preamp, but I'd love to see more closeup shots of your construction techniques on both projects.

I plan to tear my preamp circuit down and rebuild it on a solid copper plane PC board and would benefit greatly from having a look at the techniques you employed in making your low-noise preamps. I hope you can post more pictures online or publish them in an upcoming issue of *audioXpress*. Keep up the great work...I always look forward to reading your articles!

Kim Cascone
kim@anechoicmedia.com

Dennis Colin responds:

Thank you very much for the compliments.

PHOTO 1: Groove Master phono preamp.

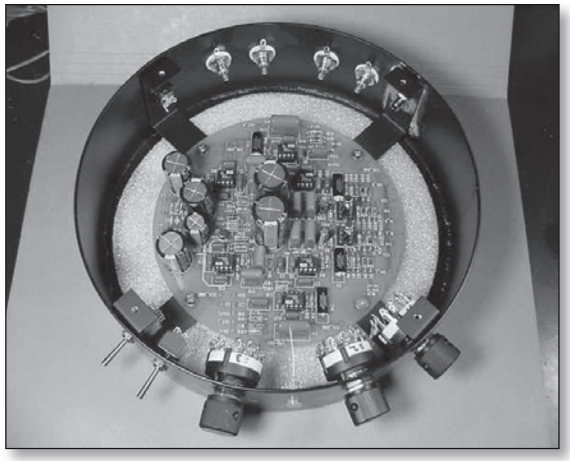

PHOTO 2: Inside the Groove Master.

Because the LP797 is so crammed in the too-small box, I'd rather not open it for photos. But the "air bridge over ground plane" construction is easy to describe:

1. I start with a piece of unetched PC board material (single- or double-sided; doesn't matter). Then I take DIP sockets for the op amps (and the JFETs here), and use 16-pin sockets so there are eight unused socket terminals.
2. I bend the inside four pins per side outward, then solder the downward-facing two pins at each socket corner to the solid copper board surface.
3. I solder the op amp supply-bypass caps from pins 4 and 7 to ground, with short cap leads.
4. I also solder all the other parts between appropriate op amp (and JFET) socket pins, if the paths are short enough. When needed to prevent shorts, use Teflon tubing on the component leads.
5. For long paths, I give balsa wood pieces to the board to provide insulated tie points, and also to mount large caps, and so on, by gluing them to these balsa strips. Crazy Glue works well and fast; you can use plastic or hardwood instead of balsa. The best Crazy Glue type cyanoacrylate (CA) I know of is "Zap-A-Gap CAT," available from HobbyLobby.com. Unlike Crazy Glue, it fills gaps and hardens like a rock. Don't get any on your skin! Hobby Lobby also sells a solvent, "Z-7 Debonder," which is made by Pacer Tech.

This technique is easy to describe, but very laborious. However, it works extremely well. I've built stable oscillators up to 13GHz in this way.

Good News
I recently had (conventional) PC boards made for the phono preamp, and they work fine. The circuit uses the same "core" as in my article, but has provision for switchable loading (100Ω, 500Ω, 1k, 47k, 75k, 100k; 50pF, 100, 150, 200, 250, 300pF) and gain (36, 42, 48, 58, 64, 70dB), and a stereo/mono switch. The two accompanying photos show the PC board and complete prototype, called the "Groove Master." I'm having difficulties finding a company that can screen the round enclosure graphics cleanly (or produce high-quality labels). When (or if) this, and reasonably priced assembly, are solved, I plan to market the unit.

My website, ColinElectronics.com, will post any progress information. But for now, I'm considering offering a kit or at least the blank PC board, enclosure, and schematics.

Regarding your preamp: I think that if you do a conventional PC board layout (instead of perfboard and without the labor of a ground-plane breadboard), it should likely be successful if you have lots of bottom-side ground plane, especially under the active devices and input area. Poor grounding doesn't directly cause noise; it's likely that your perfboard unit had RF oscillations at high gains—this can intermodulate to produce audio noise.

Please feel free to contact me at dcolin@worldpath.net; I would enjoy talking with you about general audio and the good old Aries days!

CABLE TESTS

In response to John Loudenback's assertions in Oct. '07 *aX* (Xpress Mail, p. 49), I have followed the cable discussion since about 2000. I responded to an article that again claimed audible influences of cables. The author assumed that I had a bias against cables being able to affect sound. This is simply not the case. In a home session I finally had to convince myself by conducting a cable test between a Goertz flat cable, an Audiquest, and a home-rolled one that no difference was perceivable, although I had been a proponent of "better cables" for years.

It is also fallacious to claim that the ear must be "trained" to perceive such differences. This argument is always put forth by the proponents of "cable sound."

Politely put, this is simply hogwash.

The simple truth is: as long as someone knows which cable is in the chain, the differences are, let me quote: "...a curtain withdrawn...even my wife heard it in the kitchen doing the dishes...blew me away...the $500 equipment sounded like $5000 worth...blacker blacks, higher highs..." You get the drift.

Funny enough, though, once the information—and only the information which cable is just hooked up—is removed, everybody so far has failed any and all cable tests. Pure and simple. I have in my database a collection of tests made:

www.hydrogenaudio.org/forums/index.php?sh owtopic=33951,
www.vxm.com/21R.64.html,
www.hometheaterhifi.com/volume_7_4/audio-cable-shootout-part-1-12-2000.html—how not to conduct a test,
www.verber.com/mark/cables.html,
www.bluejeanscable.com/articles/humrejection.htm,
www.aes.org/sections/pnw/pnwrecaps/2000/lampen/,
www.provide.net/~djcarlst/abx_wire.htm,
www.hometheaterhifi.com/volume_11_4/feature-article-blind-test-power-cords-12-2004.html, and
www.provide.net/~djcarlst/abx_pwr.htm

Add to that tests made in Germany on two occasions by hififorum de. and tests made by David Messinger of HIFI AKTIV, a high-end dealer in Austria, and the conclusion I draw is that anybody who claims that cable sound is a certainty is, to put it mildly, economical with the truth.

For myself, the discussion is over, I just get a bit irate when obvious untruths get peddled in perpetuity.

Peter M. Moritz
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TUBE DATA

There is an intriguing point in Mr. Touzelet's article, "Simple Approximations of Tube Anode Characteristics" (Sept. '07, p. 18). I've tried to build the Excel sheets based on his article, but the handmade (on my old and affordable HP-41C) and/or MS-Excel calculations do not match the values shown in table 2, page 18. So, I took a look in Mr. Norman Koren's website www.normankoren.com/Audio/Tubemodspice_article.html and, to my surprise, the same error was there.

So I'm confused. If I use the "base 10 logarithm (Log)" as shown in both articles, the values do not match, but if I use "Neperian Logarithm (LN)" everything becomes right (or seems to be right).

I'm not a mathematician, but we normally use *Log* for base 10 numbers and *LN* for base "e" numbers.

Luciano Peccerini Júnior
Brazil

Pierre Touzelet responds:
First of all, I would like to thank Mr. Luciano Peccerini for having tried to develop by himself the proposed software for tube characteristics simulation. I am sure he has enjoyed the time spent to do the work and see how rewarding this effort can be from an intellectual point of view.

It is exactly the objective I wanted to achieve when publishing this article in

CONTRIBUTORS

Richard A. Johnson ("An Excellent DIY Basshorn," p. 8) became excited about high-performance horn speakers in his teens when he started DJing. He once begged a guy at the wood shop to cut dimensioned material for his first speaker cabinet. After finishing school, he bought himself all the wood shop tools he could use to build speakers. He earned his BS EE at the New Jersey Institute of Technology. He has experience with wireless handset development and cellular technologies. He currently develops graphical user interfaces (GUI) using Labview for the test and measurement industry.

Andreas Schilloff ("A Home Theater Preamp," p. 16) lives with his wife Kristi and dog Sally in the small city of Binghamton, NY. He has a BS degree in mechanical-electrical engineering from the Rochester Institute of Technology. The majority of his career has been spent working as a manufacturing engineer for a high-end PCB division of a global EMS company.

Dennis Colin ("The Amplibass Control," p. 22) has demonstrated the audibility of phase distortion at Boston Audio Society, and has designed the "Omni-Focus" speaker (bipolar coincidental with phase-linear first-order crossover), ARP 2600 analog music synthesizer, 1kW biamp and PWM supply at A/D/S, and Class D amps.

George Danavaras ("Testing Panasonic's WM-61A Mike Cartridge," p. 28) graduated from National Technical University of Athens, Greece in 1986 with a degree in Electronic Engineering. He currently works in the R & D division for a Greek telecommunication company. His hobbies include design and manufacturing of audio crossovers, amplifiers, and loudspeakers.

Iain McNeill ("A Mains Analyzer–How Clean Is Your Juice? Part 2," p. 34), an avid reader since the days of *Speaker Builder*, resides in Aptos, Calif.

The audio hobby for **Alex Arion** ("A Greek Baroque System," p. 52) began in the magic '60s in Communist Romania in his grandmother's attic, where, together with two friends, he installed a small "illegal" laboratory. Their first attempt was a beginner's guitar amplifier and a professional guitar pickup. The tubes used included ECC40s and EF40s. The rectifier was a 5µ4, Russian equivalent. They mounted all in a hat box that they found in the attic. After finishing his studies and many years of research, Alex moved to Greece, where he continues to build different tube-based and solid-state amps.

audioXpress, and I am really satisfied with this result. Concerning the difficulty he has to face with the logarithm, the usual rule agreed by mathematicians is the following:

Logx, when written with a capital letter (L) means the logarithm base "e" of x.

logx, when written with a small letter (l) means the logarithm base "10" of x.

So the logarithm used in my article is the logarithm base "e".

LOW FREQUENCY EFFECT

In "Analog Bass Control for Magnepans" (Aug. '07, p. 10), what does "LFE" refer to? It is an input to the ABMS, but its function is never explained.

Charles King
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George Danavaras responds:
Thank you for your interest in my article. The LFE input refers to the Low Frequency Effect channel of a multi-channel system. The LFE channel has a limited bandwidth up to 120Hz so it delivers bass-only information, which usually is directed to a subwoofer.

In the ABMS, there is an input for this channel, which then is mixed with the low frequencies coming from the other channels. The total signal containing the low frequencies of all the channels is driven to the subwoofer output.

More information for the LFE channel can be found at www.dolby.com/assets/pdf/tech_library/38_LFE.pdf.

HELP WANTED

Can anyone supply me with a simple circuit using a tube as a VCA voltage control amplifier?

Craig Kendrick Sellen
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CORRECTION

The schematics of Figures 1 and 4 in "A Single-Ended Class A Hybrid Amp, Part 2" (Nov. '07, p. 26) may baffle anyone trying to build this design without the Aikido boards. There are no connecting dots where required on all those crossing schematic lines. I needed to refer to my collection of *Tube Cad Journal* articles to find the proper circuit connections (the *TCJ* article has the connection dots). The associated *Tube Cad Journal* article is the 8 November 2004 issue, see the "Safer version of the Aikido Amplifier" that John Broskie shows on page 4 of 4.

There should be a dot at the junction of the plate of V1a and R4, at the junction of the plate of V2b and R8 (but *not* where the line to C1 line crosses the lines to R7 or R15). The schematics have a common mistake: placing a connection at crossing lines in a schematic. This is forbidden by IEEE and ANSI standards. All connections must be made only at the "T" intersections of lines. The reason for this is that if a connection dot that is required at a crossing line is accidentally left out of the schematic, the circuit will not work properly. This can waste time and money, and may in some instances present a danger to the finished unit or the builder.

Chuck Hansen
Ocean, N.J.
aX

A Greek Baroque System

In 2005, after the winter holidays, I decided to build a special amplifier. I needed a Class A (no more than 50W), but the design needed to be impressive—eventually on wheels—to permit clean, sweet sound. I called Aris the architect, and thus began the process of building The Tower (**Photo 1**).

We finished the impressive-looking, three-tiered unit in one week, including the tests. We made three versions—each with some electronic differences, but the general view was the same. It's a decorative piece, which we needed to match the furniture.

As the photos show, the unit features a lot of massive wood and many Baroque ornaments. We only made custom prototypes, not production models. We sent the first one to the *Sound/Image* magazine for tests and evaluation, and received favorable results.

CIRCUITRY AND POWER STAGE

Because most listeners do not need more than 45-50W, I chose a Class A PP with 2xKT88 or 6550 with automatic bias (180Ω, 50W resistor) decoupled by two capacitors. I had the opportunity to find the tubes in Athens and in my numerous trips to my former country Romania. The common-cathode phase splitter uses the 6FQ7 double triode former TV horizontal and vertical oscillator, which has very big anodes, good dissipation, and did the job perfectly. Of course, it needs to balance the AC tensions, so the two RA resistors should not be the same.

I did not connect the lower tubes as triodes or UL because of the power losses and the need for more input tension.

PHOTO 1: Three-tiered Baroque Tower.

In place of the 6FQ7, I used E80CC and the Russian 6H1P, and the results were not much different. The two excellent quality OTRs (output transformers), with double C core type, were manufactured in a little shop by Tim, who helped me to experiment with all my crazy ideas. The transformers were mounted in special non-thermal plastic black boxes.

A vintage magic eye (6E5) and two VU meters complete the power stage plus some golden ornaments to make a big impression! The eye is not dancing with the music, but tells you that the

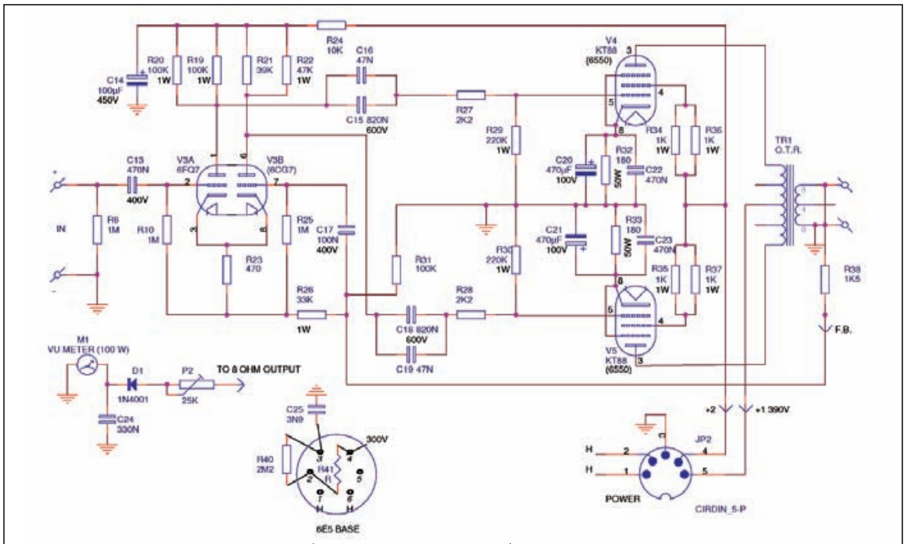

FIGURE 1: Amp power stage (one channel shown).

second HI voltage supply is OK after the standby is in the ON position. Generally speaking, as you can tell from the circuit diagram (**Fig. 1**), I used economical parts to keep the price down.

THE PREAMPLIFIER

The preamp (**Fig. 2**) is a classic design with two NOS double triodes: one 12AT7 (ECC81-Siemens) in the line stage and two 12AX7s (ECC*3, Mullard) for the phono stage. The line stage has one preamp tube direct-coupled to the second, which is a cathode-follower, and provides output for the phase splitter to the third level.

The RIAA equalization with carefully sorted, excellent quality passive parts is good enough for most MM pickups (Shure, Stanton, and so on –10 to 50mV RMS signal). The line stage has 0dB (about 1V RMS) signal sensitivity for all inputs. The volume potentiometer (100k-log) is the Alps model, and the Mode switch is also a brand name. The knobs on the second level are custom made, gold-coated from massive bronze.

THE POWER SUPPLY

My first dilemma involved the power transformers: How many to use? Eventually, I decided on one to separate the filaments—a huge, about 35cm, iron core, American made, with 18000 Gauss and a big window. I calculated it and left it up to Tim to wire and isolate the monster. Amazingly, no problems occurred after many hours of operation, with both full and partial power.

The first HV is tube-rectified by two EYY13s, which are respectable NOS East German tubes. I used an old technique, soldering the strange base of EYY13 in an octal one, to fit also to a 5U3 rectifier. This old tube works at 400V/250mA.

The second HV for the G2 and the rest of the tubes is stabilized with the help of a ECL82 (6F3P) Russian equivalent, or the American 6BM8. The stabilized voltage may be adjusted between 320-380V/50mA. The heaters for the phono stage are rectified simply, without stabilization. The power tensions are transported to the second and first levels with special cables and fit to screwed connectors on both sides.

Last summer in Athens during a heat wave of 40° C (over 100° F), I decided to test the endurance of one of the amps. Amazingly, it did not explode.

Tubes and different parts in stock used in construction. **NOS tubes**, brand names, in their original boxes: Haltron (England), TFK (Germany, Russia) Tungsram, Tesla, KEN, JAN, Sylvania (USA), Siemens, Mullard, Philips, Ultron.
Electrolytic capacitors: Nippon Chemicon, Jensen, Sankyo, Frako (Germany), Mallory, Plessey.
Coupling capacitors: MKP-Siemens, Mial, IPRS (Romania—mylar), Konweg, Hunts, Holko, MPW
Resistors: MLT (Russia), Rosenthal, Caddock, IPRS *aX*

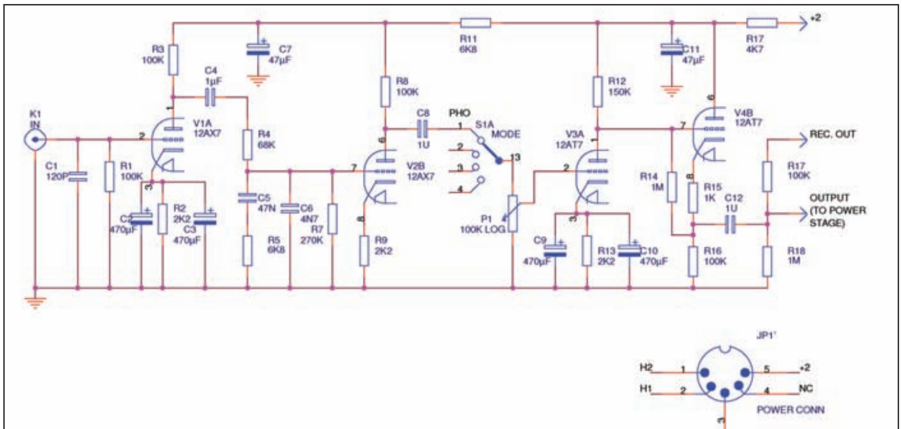

FIGURE 2: Line/phono stage preamp (one channel shown).

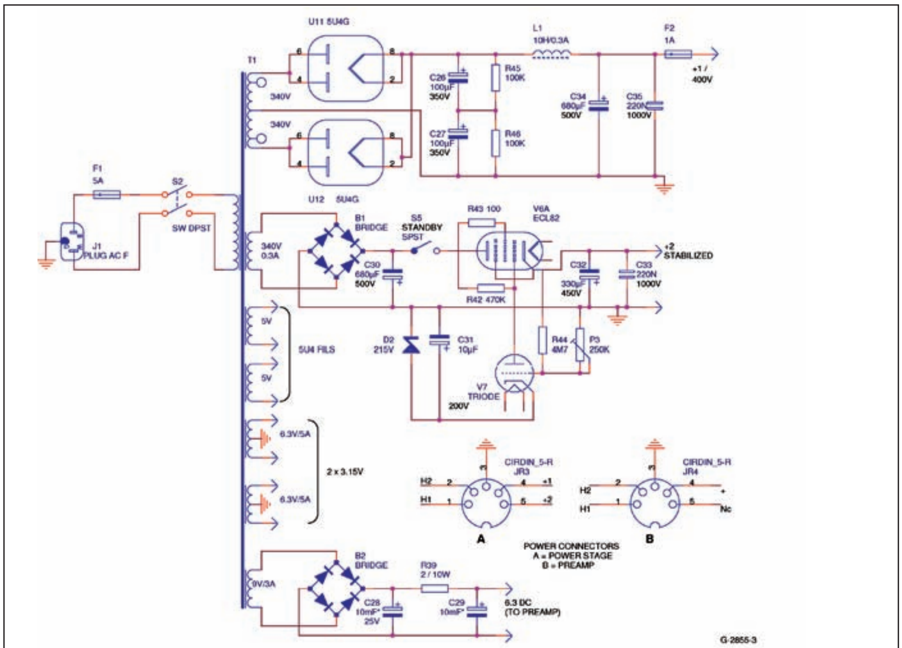

FIGURE 3: Power supply for both channels.

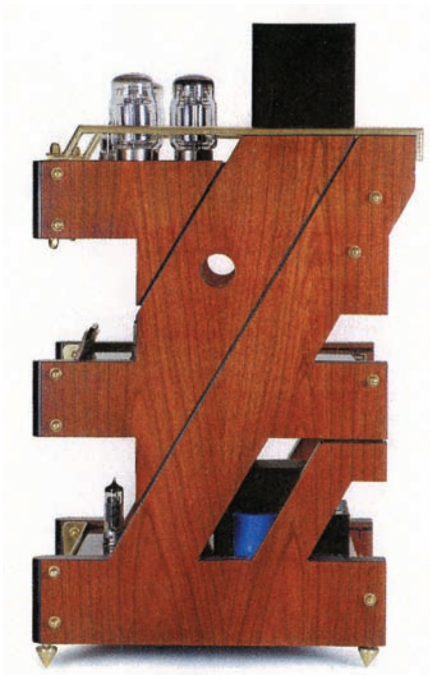

PHOTO 2: Tower in profile.