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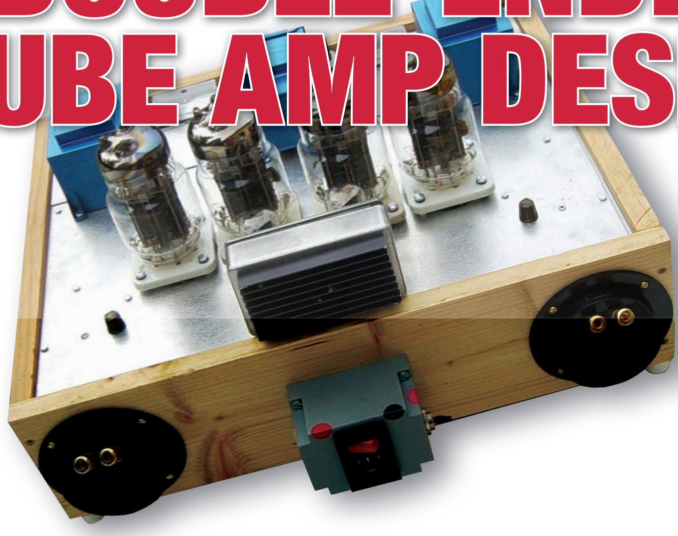
"DOUBLE-ENDED" TUBE AMP DESIGN



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A Double-Ended Two-Channel Amp

Here's a new kind of output transformer for SE amps.

DIY audiophiles considering building a single-ended (SE) amplifier are prepared to accept a limited power output—compared to a push-pull amplifier—for many reasons. First of all, they know that two output valves in a push-pull configuration can deliver quite a lot of power; whereas, the same tubes in a single-ended circuit deliver about one-quarter of that power. In a SE amplifier, the magnetic circuit of the output transformer is constantly (and from the start) magnetized by the idle plate direct current of the power valves, up to—and sometimes over—half of its capacity, leaving less than half of the total headroom to the alternating currents of the amplified signal.

On the other hand, in a push-pull amplifier's output transformer, the two half primaries produce opposed fields that cancel each other. As a consequence, these OPTs (output transformers) can be smaller for the same power handling.

Mostly for these reasons, SE amplifiers (commercial or custom built) seldom have power outputs in excess of 10W (more frequently 5 or 6W), unless high voltage power valves, such as the 845 or 211, are used. Naturally, the latter amplifiers are more expensive, because they require more sophisticated and critical circuits and components, not to mention the danger of very high voltages. Times have changed, and new solutions have made some configurations obsolete, or, to be more precise, have allowed SE amplifiers to become more like push-pull ones.

My article in July 2004 ("High Power SE 6C33C Amp," p. 20) described a SE amp delivering 50W per channel. This article will provide more information on how to further simplify the whole construction and reduce the costs. I will not deal with theoretical ideas or expectations. The results, having been obtained with units already assembled, combine the advantage(s) of the push-pull (mainly power) to the wonderful character of the SE: sensitivity, smoothness, simplicity, and warmth (due to the 2nd harmonic content, easily tolerated by our ears, as against the third, fifth, seventh, and so on harmonics produced by the push-pull).

I will use triodes—except in the power supplies, of course—and it will have direct-coupled stages, working in Class A. The layout belongs to the family of the SE amplifiers, but, to be more specific, it is *double ended*.

SE HANDICAPS

Finding them does not take much effort nor time. The responsible component is the output transformer. The more power you need in the SE layout, the sooner this critical unit becomes bulky and heavy, unless you can find a way to avoid magnetizing that half of the core's headroom that I mentioned previously.

At first glance this looks impossible, but it's not. Don't forget that this is already obtained with the push-pull transformer, in which the voltage applied to the center tap separating the two primary half windings flows to the valves through two coils having an opposed winding sense. Because the number of turns is the same, the two magnetic fields produced in the core by the DC (direct current) cancel each other, provided the currents are of the same intensity. This simply means that the full core magnetic headroom (or most of it) is available for the alternating current voltage swings produced by the output valves.

In a push-pull amplifier, each channel is completed with one output transformer (**Fig. 1**), and the two half primaries receive opposed modulations from the "push" valve and the "pull" valve. This is compulsory; otherwise, the alternating current swings would also oppose and cancel each other. For this reason,

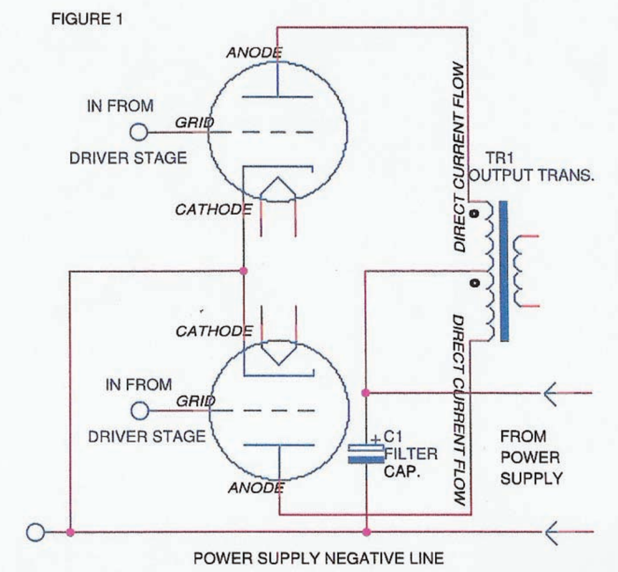


FIGURE 1: Push-pull amp circuit.

either a phase inverter or a phase splitter is necessary to feed the push and pull valves' control grids with opposed signals.

Also, in the PP OPT (push-pull output transformer), there is only one secondary that collects the induced signals and transfers their energy to the loudspeaker[1].

A typical PP amplifier—from the input terminals to the output ones of *each* channel—includes the following:

1. A voltage amplifying stage (generally one or two valves)
2. Phase splitter or phase inverter system
3. Driver stage
4. Power stage
5. Output transformer

Plus, the relative power supplies for the heaters and anodes.

Figure 2 displays the schematics of one channel of a well-known classic unit, the famous Williamson. A more modern layout has been introduced by Menno van der Veen with his ST40 amplifier (info@mennovanderveen.nl).

ACTION

Consider only three of the stages—1, 3, and 4, one set for the right channel and one for the left channel, skipping, for the time being, the inverter or phase splitter system. *Do not* use an output transformer for *each* channel, but *just one* novel output device for both channels[2].

This device (**Fig. 3**) might at first look like a PP transformer, because the theoretical layout is the same, but, in practice, there are some significant differences. First, the two primaries are not closely coupled, as in most PP transformers, but they are located each on one branch of a "C" or "UI" core, or a three-phase lamination stack, with the middle leg removed (**Photo 1**). Second, there are two secondaries (one for each channel output), coupled as closely as possible to their respective primaries[3].

REACTION

The two SE channels have been enclosed in just one output transformer, the SC-SCC-SET (split-core, stereo common circuit, SE transformer, as explained in footnote 2). What are the main advantages? Well, you saved the cost of *one* output transformer (and you certainly know that the good ones are quite expensive), and you have also reduced the overall weight and bulkiness.

But does it work? Yes, and more than satisfactorily, provided its construction follows certain rules. Besides, it has opened the door to many additional features, as you will see later.

I built (after many bench prototypes) three complete amplifiers (**Photos 2-5**) with the above principles. The results, especially the quality of sound, were quite good. Even after listening to this

PHOTO 1: Cores and bobbins.

amp for a long time, it still gives you joy and relaxation.

OPT BEHAVIOR

I can assure readers that the result is quite surprisingly good, even concerning the crosstalk between channels[4].

I mentioned that the output transformer, with its novel layout, works well with a phase inverted signal. Along with co-inventor Giovanni Mariani, head of the R&D department of Graaf, Italy, we considered the alternative of "without phase inversion." He tended to discard the latter solution, insisting on the fact that, as with a push-pull transformer, the phase inversion condition delivers good power in the bass range and that feeding it with same-phase-swings lets each primary interfere with the opposite one, with the result of reducing the inductance and consequently restraining the range in the lower frequencies. No doubt he is certainly right, but, after some tests, I found that with a good initial primary inductance and a loose coupling factor between the two channels' windings[5], on a suitable magnetic core (UI – C or EI with the middle branch removed), the OPT is still working satisfactorily, for those who appreciate the midrange more than the deep bass[6].

Here are some test results with the OPT modulated by same-phase signals, as well as opposite ones, on the first two amplifiers[7]:

1. With phase inverted swings, much more bass is heard and measured and the output power is almost the same level as the one obtained with a push-pull circuitry. The high-frequency limit depends essentially on the structure of the primary/secondary windings (tightness of the coupling, number of interleaves, and so on) and of the core material quality.
2. Without phase inversion, the high-frequency range does not change noticeably, but the bass range is less extended (in the OPT the –3B point at the low end shrinks to about 45Hz as opposed to the original 10-11Hz). Therefore, there is a loss of power in the bass frequencies, starting at about 100-120Hz[8].

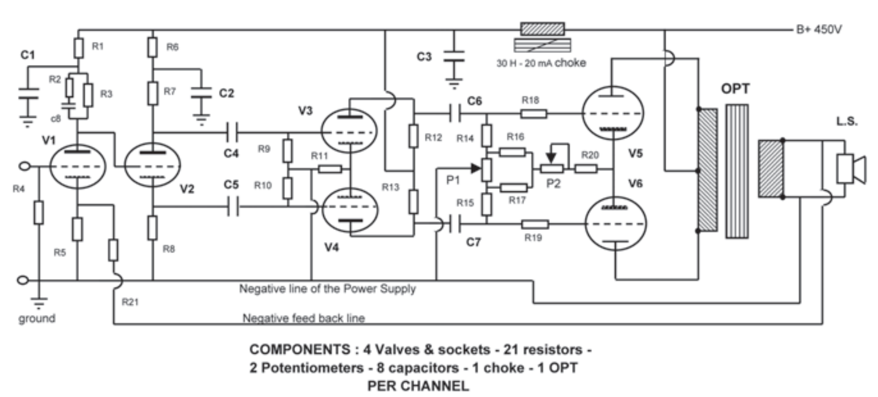


FIGURE 2: General layout of the Williamson amp.

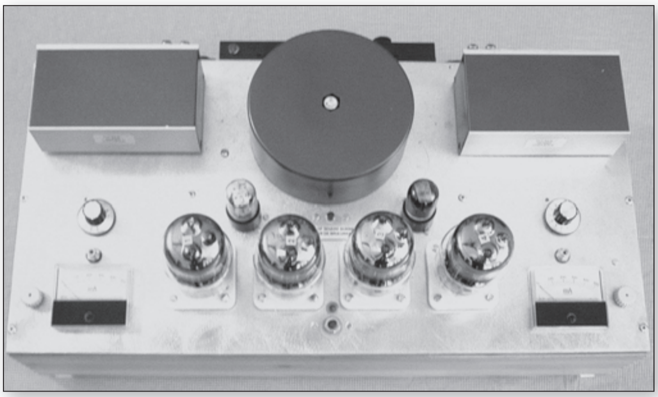


PHOTO 2: Duplex.

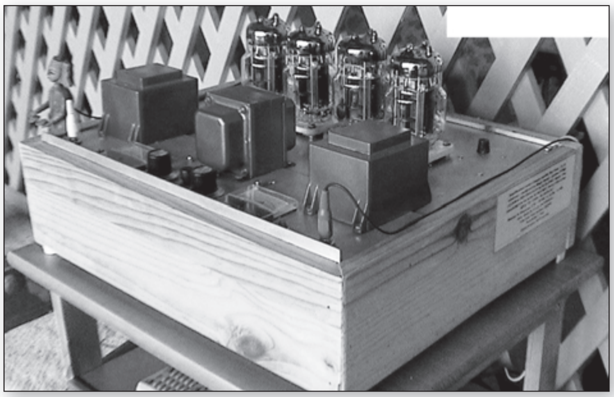


PHOTO 4: Ray's duplex amplifier.

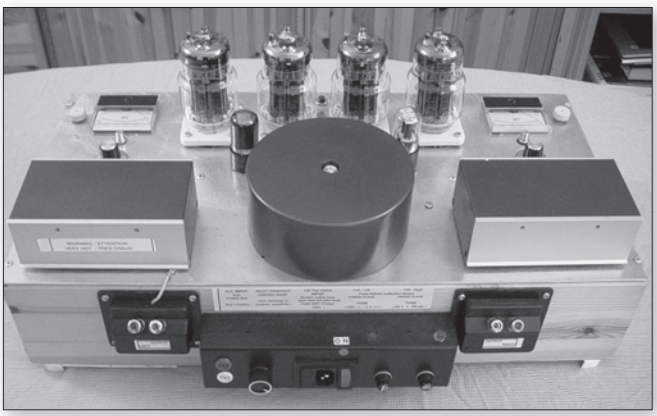


PHOTO 3: Rear view.

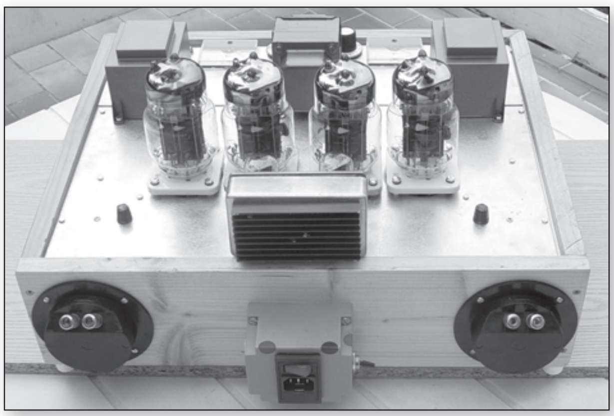


PHOTO 5: Ray's duplex amplifier, rear.

3. In both cases, the leakage between the channels was negligible, in the mid-range and above, and did not affect the stereophonic image. On the other hand, the crosstalk was more intense in the lower frequencies area (**Table 1**).

CONSTRUCTION GUIDELINES

Follow these tips when building the basic SC-SCC transformer for the double-ended amplifier circuit of **Figs. 4a** and **4b**

a) Magnetic core.
You need a C core having a cross-section area between 7 and 9cm^2 (1.1 and 1.4 in^2)
The model reference in Europe is AD32 and has the following dimensions:
Outside length: 169.9mm (6.7″)
Outside width: 97.2mm (3.83″)
Inside length and width (window): 114.3 × 44.4mm (4½ × 1¾″)
Cross sectional area: 25.4 × 31.7 (1 × 1¼″).

b) The horizontal internal dimension of the two bobbins (left and right channel) on which the primaries and the secondaries are wound must be larger by 4 or 5mm, because it will receive six layers of a magnetic band 2cm wide (visible on the sides in *Photo 9*).

c) The suggested layers of the primary and secondary windings are listed below. The total number of turns of the primaries must be as close as possible to the stated ones.

1. Close to the iron: 145 turns of 0.5mm enameled wire followed by a second layer with the same number of turns. These two layers (totaling 290 turns) must be separated by a thin sheet of insulation paper. The turns must be wound close to each other and must not overlap.
   Note: the inward wire will be connected to the anode; the outgoing will be soldered to the inward wire of the next primary turns layer.
2. Insulation: one presspahn sheet wrapped by an adhesive polyester film (such as the 3M 1350F).
3. One layer of secondary wire 1.6mm diameter (45-46 turns).
4. Same insulation as #2.
5. Another pair of primary layers as #1.
6. Same insulation as #2.
7. A second layer of secondary wire as #3.
8. Same insulation as #2.
9. Another pair of primary layers as #1.

Total primary turns = 145 × 2 × 3 = 870, secondary = 46 × 2 = 92

d) The turns ratio will then be approximately 9.4, meaning a primary impedance of 700Ω on a load impedance of 8Ω ($9.4^2 \times 8$).

e) The suggested wire diameter is suitable for a maximum idle direct current of 500-550mA, as opposed to the suggested total current of 460mA (230mA per valve).

f) Take care to properly insulate the layers, as well as the outgoing wires (with sleeves), because the potential difference between primary and secondary (as well as primary and ground) exceeds 850V (DC plus AC). Conservatively, I recommend a 1.5kV insulation.

g) You must solder the primary layers in series. As mentioned, the first inward wire will be connected to the anode and the last outgoing wire (the one that comes out from the external primary layer) to the B+. Due to the primary layers being a pair (left to right and then right to left, or vice versa), all their wire terminals will be on the same side. The secondary wires will also be soldered in series, but, in this case, the intermediate terminals to be soldered together will be on opposite sides, unless the second layer is wound from right to left (if the first one was wound from left to right). As you can imagine, custom-building the output transformer requires much attention and patience.

Other observations on the results of the tests:

- Case #1 (input signals inverted): The output voltage across the secondaries is flat from below 20Hz to about 20kHz (Note: the slight drop in the high frequencies is due to the inverter network).
- Case #2 (input signals with same phase): There is a heavy drop at the low frequency end.
- Case #3 (crosstalk): Considering that the noticeable interference is located in the bass range, the channel separation—i.e., the crosstalk level—is acceptable. The mid and high frequencies are spared, as stated here:
  20Hz = 38% (-8.4dB)
  200Hz = 8.6% (-21.3dB)
  2000Hz = 1.1% (-39dB)
  20000Hz = 1.6% (-35dB)
  Placed more than 4m apart, the speaker enclosures reduce the effect of the crosstalk when a bass note is reproduced.
  Generally speaking, the sound with the SC-SCC output transformer can be defined as "full" or "consistent," occupying the entire stage without interruption.

RECOMMENDATIONS

I recommend you first build the driver stage, which should be totally independent from the power stage. From my experience, the schematics of this stage—shown in **Fig. 4a**, together with its power supply (#1) shown in **Fig. 4b**—can be considered to represent a

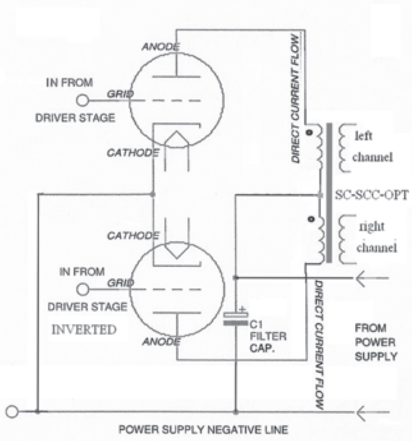


FIGURE 3: SC-SCC-SET circuit version.

universal driver. This driver stage provides for all the necessary conditions to let the power stage do its work—that is, bias supply and transfer of the amplified signal. You will notice that no blocking capacitor is present between the driver and the output stage[9]. Consequently, the power stage becomes drastically simplified.

Figure 5 shows a suggested layout of the components (see also **Photos 7** and **8**) of a driver unit under construction for an efficient and trouble-free driver stage, which should also include its dedicated power supply (**Fig. 4b**).

Before proceeding, first make sure that the driver section works properly. Building the whole amplifier at once can turn into a nightmare, if you make some mistakes here and there. Better to proceed step by step.

ADVANTAGES

A completely independent driver stage is easier to build and check for proper operation. Here's why I referred to this driver as a universal one.

A power stage basically comprises two major, important elements: the power valves and the output transformer. You must size the latter according to the power valves' characteristics, as far as impedance (a rule of thumb suggests that it be two to four times the internal resistance of the valve—or more, but with a loss of power), as well as the DC idle current and the AC power handling capacity. On the other hand, to operate satisfactorily, the valves' grids need to be "biased," generally at a negative level, and receive an AC signal with a large enough swing to produce full power output (normally—and this is called Class A—with a peak amplitude not exceeding the negative bias level).

While the task of providing this swing is definitely the driver stage's responsibility, the negative bias can be provided for in different ways, typically with the self biasing or fixed bias supply systems (**Figs. 6-8**). In the direct coupling modulated bias layout (DCMB), the bias is supplied by the anode load resistor of the pilot valve located in the driver's stage. As a consequence, the set of components of the power stages is reduced to the essential ones (valves and output transformers).

The average negative bias for a 6C33C-B set to 80-90V, with a voltage of 220-230, between cathode and anode (k and a), corresponds to an idle current between 200 and 250mA. This value, however, depends a lot on the valve's conditions (brand new or used valves with different characteristics, and so on). The 300B would require the same bias, but the voltage between k and a should be about 400V, in which case the resulting idle current would be approximately 75-85mA. With the EL34, you have -35V, 350V, 75mA. For the 2A3, the figures are -45V, 240V, 55mA (for other valves, please consult the previous articles in *audioXpress* or the technical characteristics of the valves published by the manufacturers or in some Internet sites).

POWER HANDLING

During the tests on these amplifiers, I also had the opportunity to vary the idle

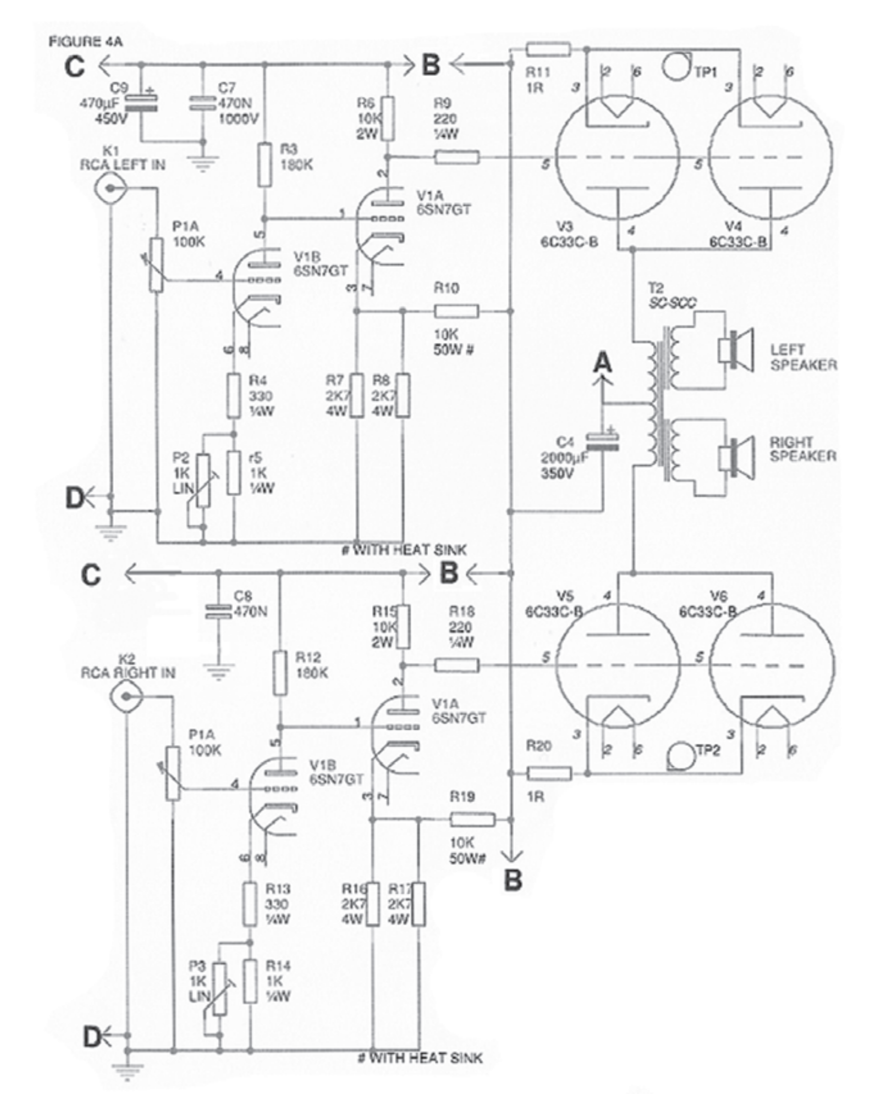


FIGURE 4A: Driver and output stages.

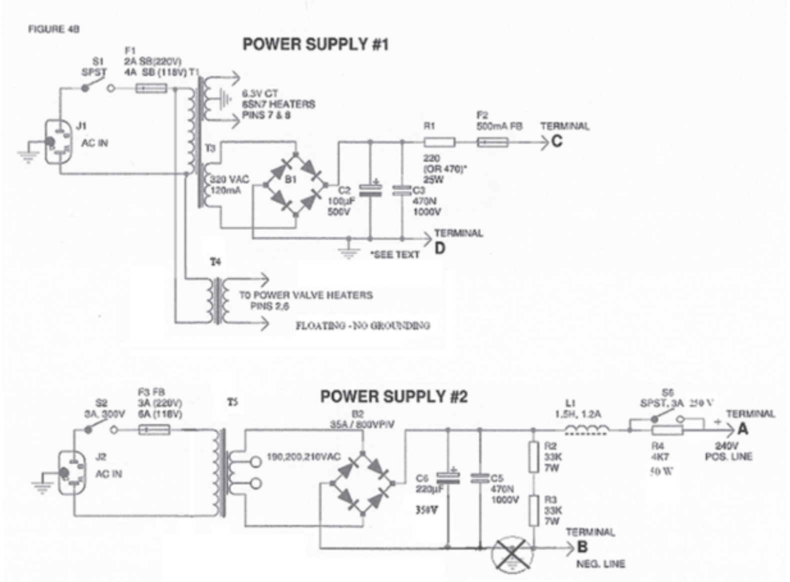


FIGURE 4B: Power supplies.

current while listening to the amplifier, and I noticed that (obviously, I would say now) the sound changed somewhat. I tried the amplifier with different idle plate currents, ranging from 100 to 250mA DC, using the variable resistors (P2 and P3 in **Fig. 4a**) that adjust the 6C33 valves' plate consumptions (dissipation) and went from a Pa of 23W (100mA) to over 50W.

I am aware of two effects when the idle current level is increased:

1. a different damping factor (due to the fact that a valve operated at a higher anode current has a lower plate resistance)
2. more bass[10], for the same reason.

The output power at midrange frequency is not particularly affected, according to the tests I made. You could use these features to reduce the valve's stress (when the idle current is set at a lower level) if the music is soft without bursting sections, or a high bass content.

CIRCUIT DETAILS

Some measurements (average):

- Frequency range: (at 20W of power output)
  a) 40-50Hz to 30kHz at –3dB (without the inverter)
  b) 11Hz to 29kHz at -3dB (with the inverter).
- Total harmonic distortion: 1% at 10W, 4% at 20W measured at 1kHz.

Needless to say, the output transformer's quality is very important if you want to obtain these results. The sources are indicated at the end of the article, but, if you want to build it yourself, the basic details are available. They are intended for private and noncommercial use.

I must state that it is not very easy to build. Anyway, you can contact me at these e-mail addresses for further information: ari.polisois@wanadoo.fr or Lpolisois@yahoo.it (for Ari).

This new kind of compensated audio output transformer for SE amplifiers is quite tolerant and does not require critical calculations. Even the primary impedance (depending on the turns' ratio as well as on the impedance of the load connected to the secondary) could vary to a certain extent without unacceptable consequences (for the amplifiers described, from 300 to 600Ω). Needless to say, the more accurate the work is, the better the results in terms of frequency range and distortion.

CONCLUSION

The amplifier's circuitries described allow for some interesting listening alternatives, such as more or less bass, more or less idle power fed to the output valves (irrespective of the integrated volume control adjustment), and must be considered as an attempt to explore new ways for better listening that would please demanding audiophiles. These features might suggest that they make the amplifier more complicated, but, luckily, once you have found the alternatives that better suit your taste, you can stop playing with the soldering iron, knobs, and connections and listen to the music in your preferred way.

The 6C33C DCMB amplifiers shown in the photos delivered an extremely pleasant sound—precise, "cool" but also explosive (and this includes the higher setting of the idle current level) when required, and "consistent" (sorry, but I could not find a better word to qualify the difference I noticed, compared to a SE amplifier fitted with a pair of output transformers, instead of a common magnetic core one).

You will need to resolve for yourself the question of whether the layout described is single-ended or double-ended?

FOOTNOTES

1. Disregarding the losses, the power transmitted by the primary (VA = volt amperes) is the same in the secondary. $Va = Vp \times Ip = Vs \times Is$.
   Therefore, if you have 100V and 0.1A in the primary winding, the corresponding power is 10VA. If the turns' ratio primary/secondary is, for instance, 10, you would have 10V at the secondary terminals and the same power available (10V × 1A).
2. This novel OPT has been designed as SC-SCC-SET, meaning: Split core-Stereo Common Circuit–SE Transformer. It's a novel transformer, patented by Ari Polisois and Giovanni Mariani (Chief Engineer R&D/Graaf/Modena—Italy), INPI Patent # 03 13898. It was presented at the Audio Engineering Society convention held in Paris in May 2006, following the presentation of the first generation of the basic self compensated transformer—INPI patent # 01 02457 at the AES convention of May

SUGGESTED LAYOUT OF THE DRIVER SECTION - View from underneath.
(each square : 1 x 1 cm.)

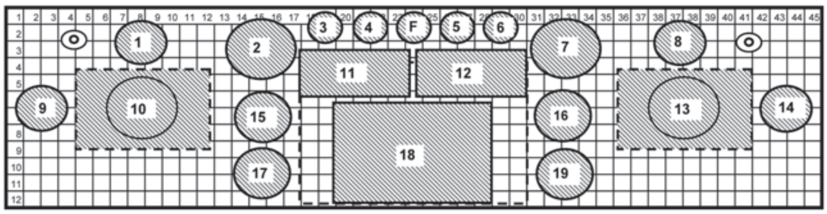


FIGURE 5: Suggested layout of the driver section, view from underneath.

2005 (Barcelona, Spain).

3. For an extended high-frequency range, a good number of interleaves is recommended: four primary and three secondary would give excellent results.
4. The reason might be the following: In the iron core, two opposed magnetic fields generated by the DC flowing in the primaries fight steadily, tending to cancel each other. Depending on the amplitudes of the left and right channels' alternating currents, the above balance is upset, but the DC fields dominate the scene, if they are much stronger than the AC ones. However, more likely, the real cause of the situation is the very close coupling between each primary with its secondary, combined with the loose coupling between these two sets (see also note 5). The tighter the coupling primary/secondary and the weaker the coupling between the windings of the two channels, the less crosstalk you have.
5. The loose coupling is improved by the "side flux escapes" that are described in the construction sidebar.
6. I needed to modify the inverter set (the bass boost box, shown in **Photo 6**) by introducing a bass control. It is simply a pair of variable resistors (5k each) inserted between the RCA inputs (left and right channels) and the primaries of the inverting transformers. At their maximum resistance, the –3dB point at the low frequency end moves from 10Hz to 40Hz.
7. They both use two 6C33C-Bs per channel, with about 25W output.
8. However, with the two paralleled 6C33C-Bs per channel, giving an average internal resistance close to 60Ω, the basses can be heard quite loud (see Menno van der Veen's book, *Modern Audio Amplifiers*, chapter 5.5). Because an amplifier without phase-inverter

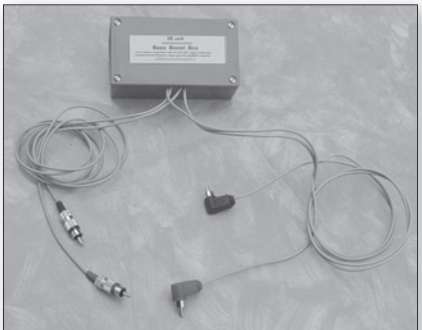


PHOTO 6: Bass boost box—input phase inverter transformer.

FIGURES 6-8: Bias configurations.

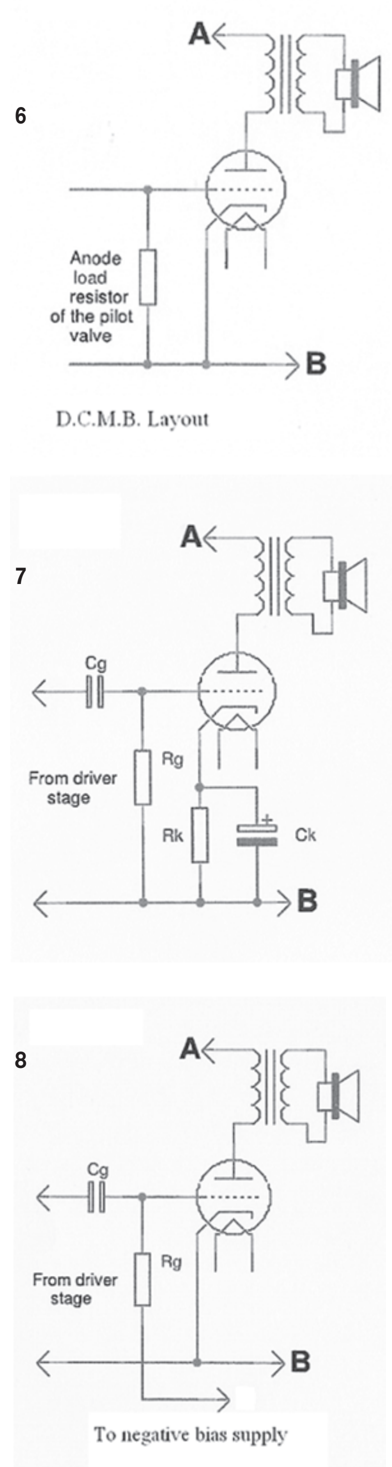
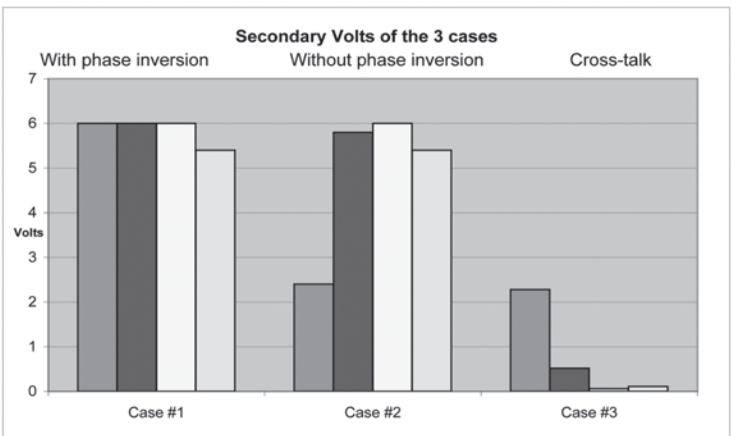


TABLE # 1
AC Volts measured at different frequencies on the secondary of the SC-SCC-SET Ref. 160902-Rev-0106 novel output transformer.
(Reference frequency 1 kHz - with 6 VAC output)

Frequency	Case #1	Case #2	Case #3
Four steps	Normal use of the OPT **with input signals inverted** *(best results)*	**No phase inversion** applied to input *(low level of bass)*	**Cross-talk** leakage from the active channel (measured on the other one)
20 Hz	**6**	2.40	2.28
200 Hz	**6**	5.80	0.52
2 kHz	**6**	**6**	0.06
20 kHz	5.40	5.40	0.11
	Volts	Volts	Volts



Bars : from left to right 20 Hz - 200 Hz - 2kHz - 20 kHz.

Case # 1 — Left channel and right channel sine wave input signals of same amplitude, but 180° apart. ***This layout gives the best results.***

Case # 2 — Left channel and right channel sine wave input signals of same amplitude, with the same phase. ***This layout lacks of bass frequencies.***

Case # 3 — With the left channel input voltage same as above and no voltage applied to the right channel input : ***the left channel signals leak to the right channel mainly on the low frequencies.***

is much simpler and less expensive (fewer valves, passive components, space and time), some of you might prefer alternative # 2. This choice is also justified if you usually listen to CDs recorded with enhanced bass (such as many popular ones), or because your loudspeakers are booming. Or you could try the following:

a) when a lot of bass is welcome, use a phase splitter transformer (Sowter—UK—references 8920-3603-8584 or model 3B—BassBoostBox from A2Belectronic@wanadoo.fr, as per **Photo 6**) connected between the CD player's output and the amplifier's input RCA sockets, or, alternatively,

b) when disturbed by the excessive bass level, use a direct connection between the CDP and the RCA input sockets of the amplifier, without any phase splitter or inverter.

c) the inverter box with a bass control (see note #5).

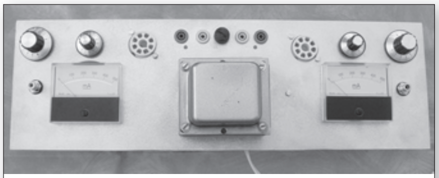


PHOTO 7: Driver unit under construction.

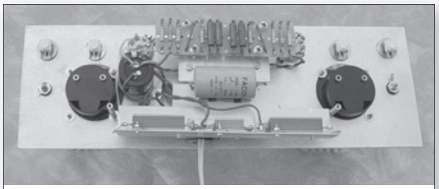


PHOTO 8: Driver unit under construction, inside view.

Solution b definitively reduces the bass level. It is necessary, in this case, to reverse one of the speakers' connections, if these are normally set for the high bass level option, in order to have the speakers working with the same phase.

9. This layout, designed as DCMB (direct coupling modulated bias) is described in the article mentioned (*aX* July 2004) as well as in other sources on the net. Schematics and comments can be found at the following sites: www.audiodesignguide.com, or *Valve Magazine* www.bottlehead.com, or www.Plitron.com, or you simply ask for it at ari.polisois@wanadoo.fr).

10. Menno van der Veen's book, Chapter 5.12.

The formula is

$$F\_3L = (Z/2\pi Lp) \times (\beta/\beta+1)$$

where F_3L is the lowest frequency, Z the impedance, Lp the primary inductance, and $\beta = Ri/Z$ from which it is obvious that Ri (the internal resistance of the valve, inversely proportional to the anode current) plays an important role. *aX*

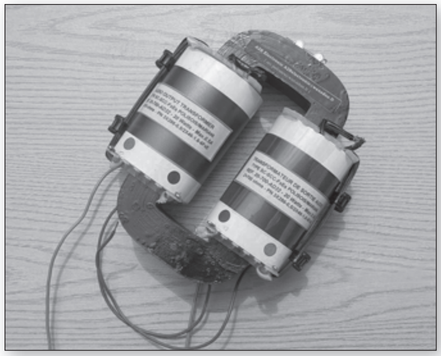


PHOTO 9: SC-SCC OPT.

Driver-Induced Vibrations

In order to generate a sound, a driver must vibrate. But what happens when one or more drivers are placed on the same baffle?

Let's take a look at the classic loudspeaker: a wooden parallelepiped with two or more drivers mounted on its front panel through screws or bolts, with the driver's frame rigidly coupled to the panel, or with a foam gasket between the two. In the exact moment that the cone begins to move, it starts to transmit—through the frame and the screws—vibrations to the cabinet and to the other drivers that are fixed on it. These are the contact conduction vibrations.

Moreover, you must consider another sort of transmission of the vibration, the one through the air that is generated by the back of the cone. This kind of conduction is less evident in open baffle systems, because there is no back enclosure, as well as in closed box systems because the large amount of absorbing mats used in this kind of loudspeaker helps to reduce the energy transmitted through the air.

The result of these two types of conduction is that besides the driver, the cabinet also sounds. . . and it sounds loudly. In fact, in 1975 Barlow[1] measured the sound output of a birch plywood square panel, excited by a driver placed on the inside of the panel, and determined that at the fundamental bending resonance the panel was almost acoustically transparent. In the same year, Stevens[2] presented an even more interesting work: he measured the undamped back panel sound output of a 50 ltr box, made of 18mm chipboard, showing peaks 10dB lower than the driver output. These dB values are worrisome, because they can be easily heard[3]. Further and more recent studies by Backman[4] and Moriyasu[5] compared various cabinets of similar volume and panel thickness, but built with different kinds of wood (chipboard, MDF, and plywood), and found minimal resonance differences among the three; nevertheless, it was noticed that MDF and plywood were better at damping the higher modes.

DEFINING THE PROBLEM

To summarize, the driver-induced vibration makes the box sound; therefore, it produces coloration, masking, and detail loss.

Thanks to Iverson[6], you already know that over 1200Hz the panels do not bend very much; therefore, your goal is to raise the panel resonance frequency. To accomplish this, you need to work on material, dimension, thickness, density, rigidity, and damping of the panels to be used. More precisely, with different values for each topic, increasing the thickness and rigidity of a material raises its frequency of resonance. The same happens when you decrease the panel dimension, while increasing the mass and damping will produce a lower resonance frequency.

To build a rigid structure, you can start by using thick panels: for a volume up to 30 ltr, I recommend woods with at least 18mm of thickness; from 30 to 50 ltr 25mm and above 50 ltr it's wise to go with at least 30mm of thickness. I also suggest some kind of corner joint in the side panels, with the front and back panels inserted in the box, as I did in the Auri loudspeaker (*audioXpress* 10/06).

To further increase the box rigidity, instead of employing the classic rectangular form, you can try a curved one, such as B&W 800 or Sonus Faber Guarneri Homage, which, however, doesn't come cheap. Another recent technique is the *translamination* (also called *layertoning*) adopted by Tad and Seventh Veil, in which the box is made by overlapping many wood sheets. As you can imagine, cost and wood waste increases quite a lot. An example of translamination is visible in **Photo 1**.

Another way to stiffen the structure is to double or triple the number of panels forming the box. Moriyasu[5] reported a reduction of the resonance modes when using a double panel of MDF, in com-

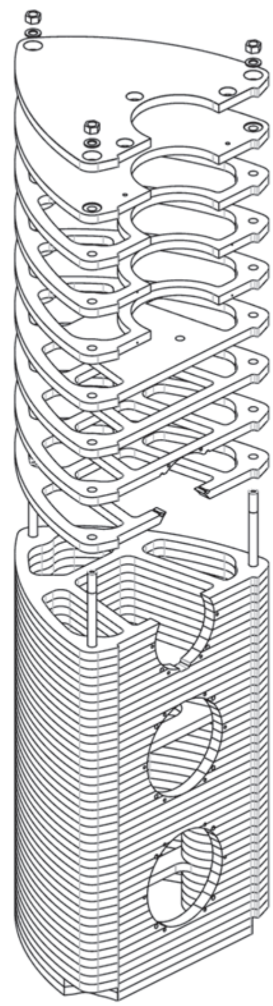


PHOTO 1: The translamination, used with TAD Model-1 loudspeaker.

parison to the single 19mm thick panel, but the increase in weight might be an obstacle, particularly if the speaker dimensions are huge.

The last two cards to play are the bracing (which helps to raise the panel resonance frequency) and the panel damping (which instead lowers the frequency). Bracing is more effective in the vibration suppression, in comparison to damping, but you can use both. But let's take a closer look at these two techniques.

BRACING

The first step in the war against vibration is the use of reinforcements inside the box, made of high-density hardwood and fixed to the panels with screws and glue. To maximize the contact area between the parts, make a hole in the brace larger than the screw, so the screw threads won't bite into the brace, while the use of epoxy glue or formaldehyde-based adhesives (not to be used in a too thick layer, otherwise mechanical strength is reduced) adds rigidity.

Vertical bracing is more effective than horizontal, and it is important to place it not in the center of the panel; if the cabinet space allows, you can use more braces set at different distances from each other.

Moreover, you can brace the driver magnet as well, to better distribute the energy involved. Just don't cover the decompression hole, if present, and don't interfere with the back cone wave radiation, to reduce the *regurgitation effect*.

The second step is shelf bracing and involves the use of windowed panels, so that four sides of the box are tied together. The best results are achieved with four holes—rather than a single hole—of oval or circular form, rather than rectangular. It is important that the holes are large enough and the thickness of the shelf not too excessive, so the holes don't act as reflex ports. You can use more shelves to increase the cabinet rigidity and therefore its resonance frequency. Just remember to place them so they don't interfere with the driver back emission.

Another step—for sure more complicated—is to build a Matrix™-like structure (**Photo 2**), in which a two-axis grid forms an advanced brace, coupling all the cabinet panels with each other. Those interested in experimenting with such a brace should carefully choose the woofer to be used. Because the Matrix™ offers to the driver an uneven load (at least more uneven compared to a cabinet without Matrix), a rigid thick damped cone is the better choice. B&W itself adopts drivers with pulp and Kevlar cone, or Rohacell and carbon fiber one, avoiding aluminium cone.

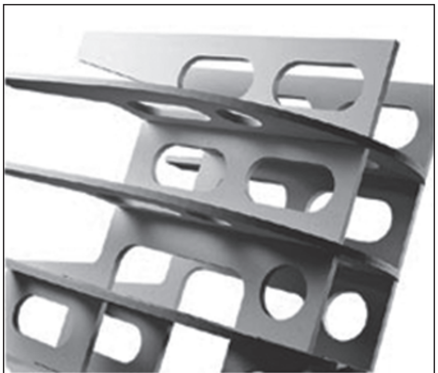


PHOTO 2: B&W's Matrix™.

DAMPING

All materials have the capability of dissipating the energy they receive, properly quantified by the damping loss factor. The higher the value, the better the damping, and a value of 0.6 corresponds to a very good dissipation. Metals usually have a low loss factor, while viscoelastic materials are capable of better damping by transforming the mechanical energy into heat. To raise the damping loss factor of a material, in this case wood, all you need to do is to stick a sheet of damping substance on it, according to one of the following techniques: the extensional damping and the constrained layer damping.

Fig. 1 illustrates the extensional damping (ED): the energy is dissipated by the deformation, due to compression and extension strain, of both layers; increasing the damping layer thickness increases the damping.

The ED is economical and easy to apply, besides offering good results. Some damping material includes lead (which, however, increases speaker weight), the Isodamp C3202-50 by EAR SC, the Fonomat 425 by AZ Audiocomp, the Antiphon LD13, and roofing felt. Be sure to make a strong bond between the damping layer and the base layer to maximize the contact area, as well as to make an age-proof union.

The constrained layer damping (CLD) consists of a three-layer sandwich (**Fig. 2**), in which the *bread slices* are the wood layers and the *ham* is the damping layer. The energy applied to the constraining layer is partially transmitted (shear strain) and partially dissipated (hysteresis) by the damping layer, so the resulting energy that reaches the base layer has been reduced, and its vibrations too. A thicker damping layer doesn't correspond to a better damping loss factor, as you see in the ED. The best result comes with the layers glued together using a high shear stiffness adhesive.

While more efficient than the extensional damping, the CLD also offers many variants: it's possible to use different materials and thicknesses for the constraining and base layers, as well as employ different damping layers. The downside is the cost, weight, and time to determine the best variant to use, which implies measurements with an accelerometer. For the damping layer I suggest the Fonomat 225 and the Fonogel by AZ Audiocomp, and the EAR SC CN-12. Plywood is the standard material with CLD.

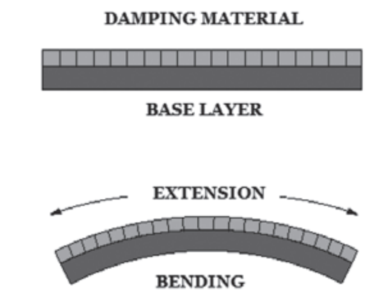


FIGURE 1: Extensional damping.

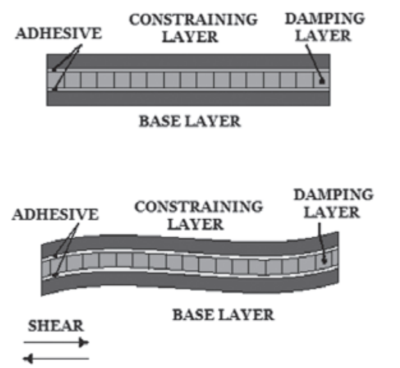


FIGURE 2: Constrained layer damping.

CONTACT VIBRATION REDUCTION

After this look at how to combat vibration, it's now time to see what you can do to reduce the energy conduction itself. From Newton's third law of motion, you know that most of the mechanical energy generated by the driver is transferred to the panel through the frame; therefore, if you decouple it, you reduce the energy that reaches the panel and, consequently, its resonance amplitude. Another positive effect of chassis decoupling is a reduction of driver interaction, because the same energy that reaches the panel goes to the other drivers, too.

If you use a foam gasket between the driver frame and the panel, you are already applying a decoupling; however, the screws used to fix the driver are another means of vibration transmission. Using some rubber inserts, such as EAR SC Isoloss or Well-nut or Flex-loc, will better isolate the driver. Solving the problem seems an easy task, but it's not, because the decoupling material has its own *transmissibility* curve, which says that at certain frequencies the insert and the gasket can act as an amplifier more than a damper.

Look at the past for some examples of driver decoupling: In 1980 Linkwitz used some soft rubber grommets to attach the Kef B110 (*Speaker Builder* 2, 3, 4/80). B&W, for many years, has decoupled the mid and high cabinets by using Raychem's Isopath. . . even if this is useful to diminish the interactions among the drivers more than to reduce the energy that reaches the cabinet panels. KEF, in the golden years of research, used a resilient mounting for the woofer and midrange, to decouple them from the enclosure.

In his article about driver isolation, Moriyasu[5] used a Peerless 831858 woofer with the Well-nuts (**Photo 3**), and the accelerometer measurement showed a reduction of the secondary modes. In another article about driver isolation, Jones[7] used the EAR SC E-610 to decouple a Pioneer Q13ER71 woofer (only 13cm diameter), and the conclusions affirm that driver diaphragm motion is minimally affected by decoupling the drive unit from the cabinet. In addition, magnet motion and cabinet vibration improve with the decoupled drive, all confirmed by preliminary listening tests.

I decided to conduct my own study on driver interaction, using a plywood baffle (83 × 47cm large and 1.7cm thick) on which were fixed: a 10″ woofer using screws and T-nuts (this will be the exciting driver); a 7″ low-mid driver (Focal 7K4412), the DUT, coupled to the panel with T-nuts or decoupled with Well-nut (1 × 1.3cm M3). Both drivers used a 3mm thick foam gasket between the frame and the wood. To eliminate any


PHOTO 3: The Well-nut rubber insert.

crosstalk, I used different generators and amplifiers for each driver.

The first measurement was the electric response of the 7″ Focal (DUT), while the exciting 10″ woofer (EW) played a sine wave signal at 250, 430, and 1000Hz. **Figure 3** shows a reduction of the resonance peak when using the DUT and Well-nut (white curve), with respect to the DUT and T-nuts (black curve); the reduction diminishes in amplitude as frequency

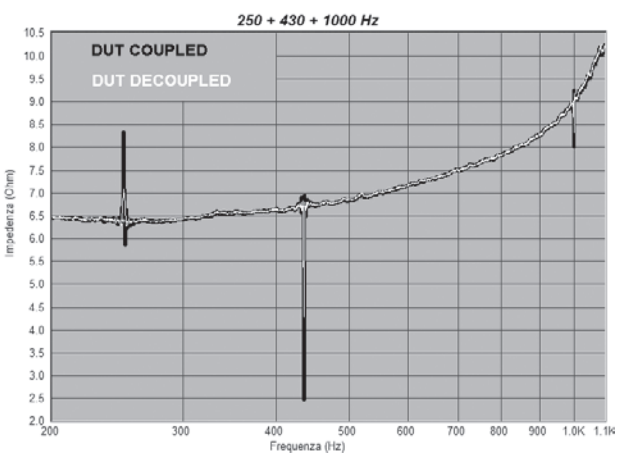


FIGURE 3: DUT impedance response comparison, with Well-nut (white line) and without (black line); the exciting woofer was playing a 250-430-1000Hz tone.

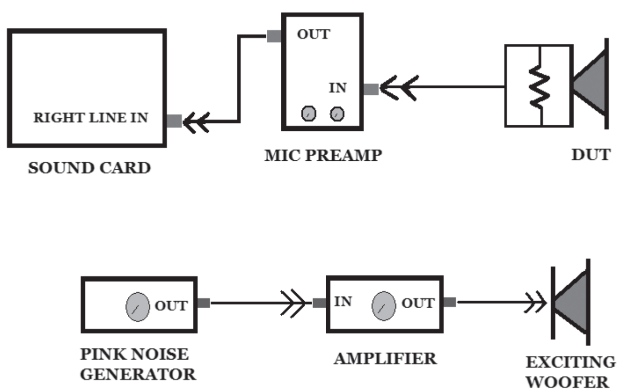


FIGURE 4: Connection using the DUT as a microphone.

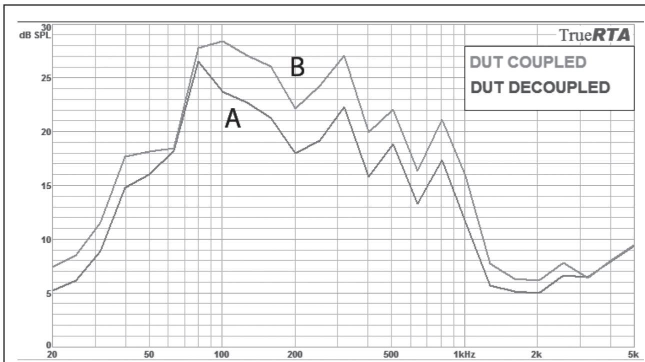


FIGURE 5: DUT used as a mike, with Well-nut (A) and without (B).

increases, as you would expect. The measurement was done using the freeware program *Speaker Workshop*.

Another kind of measurement arose from a conversation with my friend Valerio Russo, who suggested using the DUT as a microphone. The connections are visible in **Fig. 4**. The DUT is connected to the mike preamplifier input, while the preamp output goes into the sound card line input; the EW is attached to the amplifier output, whose input is linked to a pink noise generator. I used a real-time analyzer (TrueRTA), and the amplitude values are not absolute.

For this kind of measurement, the Focal mid-woofer was appropriate because it has a rigid cone, a soft suspension, and a wide frequency response. I placed a 0.2Ω resistor in parallel to the driver to simulate the amplifier, as in real life, and therefore used its electric damping. **Figure 5** shows how using the Well-nut reduces the energy generated by the exciting woofer in a wide range, with the exception from 65 to 80Hz where the curves are overlapped.

So far the Well-nut has helped lower the contact vibrations that reach the driver, but does its use have any collateral effect? To try to answer this question I investigated some more, starting with the impedance response of the DUT and Well-nut and then of the DUT and T-nuts (in both cases I also used a gasket foam). The resonance peak, at the driver Fs, was lower in amplitude when using the Well-nut, bringing a 6% reduction in the driver Qms. I expected this, because the driver-baffle coupling is less rigid employing a damping material.

The next verification was the near-field acoustic measurement of the two configurations, and **Fig. 6** shows the comparison. From 38Hz the curves are identical, with the 1100Hz dip slightly more evident in the rigidly mounted driver; below 38Hz the differences are negligible.

Moving to the far-field measurement, **Fig. 7** compares the two impulse responses, in which the DUT and Well-nut have some more evident peaks, but this is nothing to worry about. In fact, both the step response and the frequency response, calculated from the impulse response, have very similar curves.

Figures 8-9 show the waterfalls of the *cumulative spectral decay*, measured with the mike 10cm from the driver dustcap, using the ARTA software. Here, too, it's hard to notice any difference between the two configurations.

In sum, is driver decoupling the way to go? Unfortunately, there are two answers. Using the Well-nut helps to damp the driver-induced vibrations, but, at the same time, the driver-baffle coupling is less rigid. On one side you gain by reducing the cabinet sound and driver interaction; but on the other side you lose in details and transient response. Each project needs an evaluation of which side is more effective overall, supported by a music listening test.

CONCLUSION

Driver-induced vibration is a problem with many faces, but there are many ways to reduce its effects, because elimination is quite impossible. A ready-to-use recipe is not readily available, but this article does offer some useful guidelines. My personal recipe for a hypothetical three-way loudspeaker is

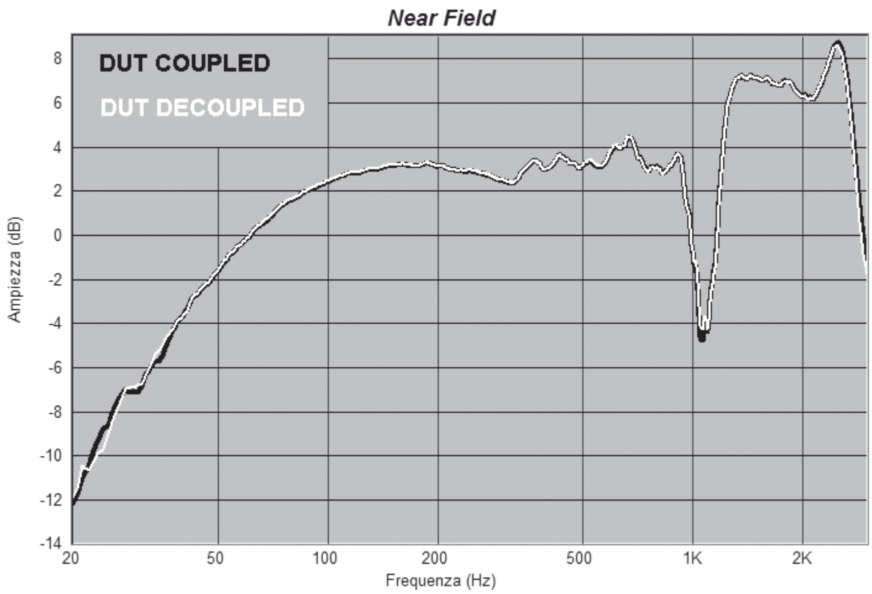


FIGURE 6: Near-field response, with Well-nut (white line) and without (black line).

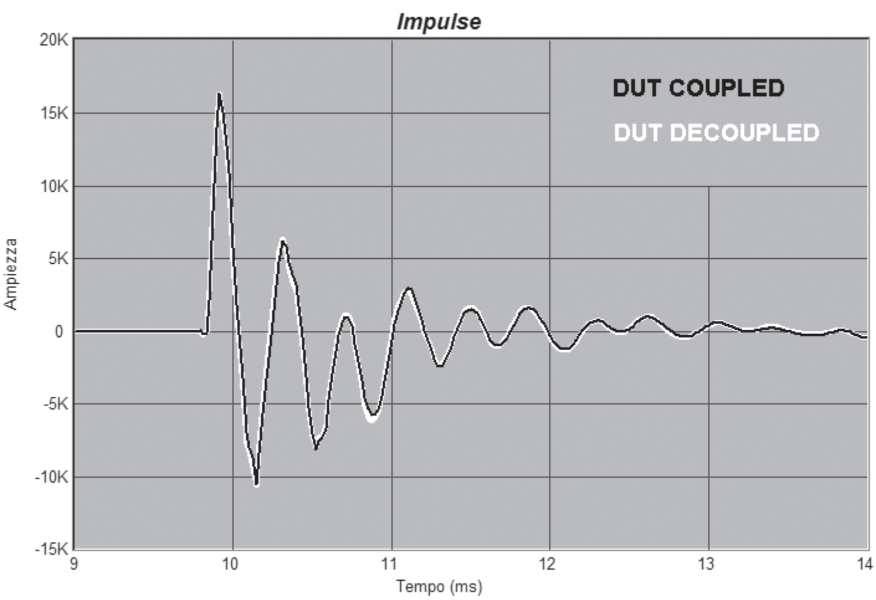


FIGURE 7: Impulse response, with Well-nut (white line) and without (black line).

to use an enclosure for just the woofer, made of plywood with many braces so the panel resonance is well above 1000Hz, assuming the driver will be used until 500Hz.

The midrange is in another enclosure, decoupled from the woofer. This enclosure uses dense wood, such as MDF, and damping mats (lead is a very good choice, because it adds mass); this lowers the cabinet vibrations below 500Hz. And, finally, you can mount the tweeter in the same enclosure as the midrange, but use the Well-nuts to fix it on the panel. aX

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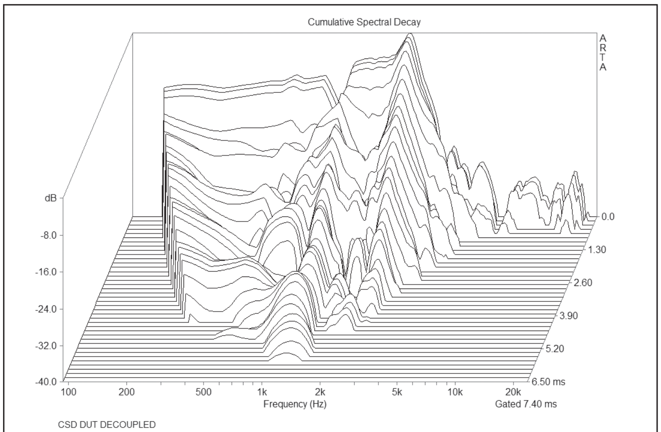

FIGURE 8: Cumulative spectral decay with Well-nut.

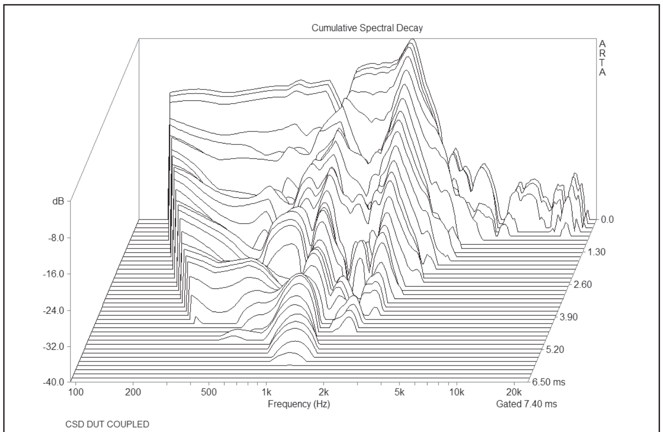

FIGURE 9: Cumulative spectral decay without Well-nut.

By Paul Loewenstein

A Dynamic Microphone Soundcard Amplifier

A good tutorial on interfacing microphones to sound cards.

This microphone amplifier is designed to interface the Shure SM10A dynamic microphone to a SoundBlaster card or other computer microphone input with a 5V through 2.2kΩ bias to the microphone connector ring. It enables speech recognition veterans of DOS DragonDictate to continue using the DragonDictate microphone with Dragon Naturally-Speaking.

PRINCIPLE OF OPERATION

In **Fig. 1**, the open circuit voltage gain is provided by the grounded emitter stage formed by Q1 and R2. This is followed by a unity-voltage-gain emitter-follower stage formed by Q2 and R4. R3 provides negative feedback to better define the gain, improve linearity, provide DC bias, and reduce the output impedance.

C2 rolls off the open loop gain above audio frequencies to prevent oscillation. It also attenuates any radio frequencies picked up (see "Effects of Nonlinearities" section).

C1 and C3 provide DC isolation at the input and output, respectively. C4 reduces power rail excursions.

DC BIAS

Q1 base is one diode drop (about 0.6V) above ground, drawing current from Q2 emitter through R3, across which there is a negligible DC voltage. Q2 base is therefore about 1.2V above ground. Taking into account the 2.2kΩ in the soundcard between the ring and the 5V supply, the ring settles at about 3.5V.

Open Loop Gain. The DC voltage across R2 is 3.5-1.2 = 1.3, so at room temperature the voltage gain of the grounded emitter stage is about 40 × 1.3 = 52.

Closed Loop Gain. The model for calculating the closed-loop gain (**Fig. 2**) contains:

- An ideal amplifier of gain $-A$ representing the gain of the grounded-emitter stage with $A = 52$.
- Resistance x representing R1 in series with the microphone impedance; the impedance of an SM10A is 223Ω, so $x = 323Ω$.
- Resistance z represents the input impedance of the grounded-emitter stage. This is β × the emitter impedance, which is R2/A. The lower bound on β is 270, giving a lower bound on z of 29kΩ.
- Resistance y represents R3 (10kΩ).

Analysis of this circuit gives the closed-loop gain as: $-Ayz/xy + yz + (1 + A)zx$

Substituting for A, x, y, and z gives a (negative) closed-loop gain of 19.

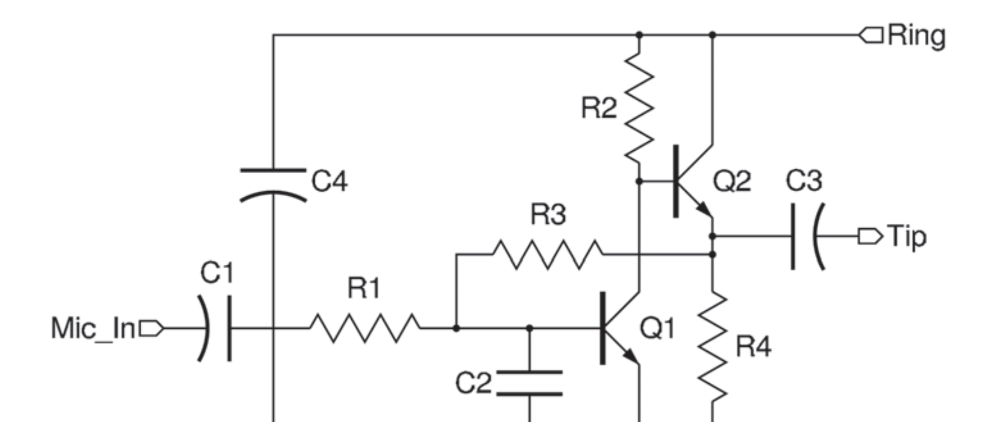

FIGURE 1: Amplifier schematic.

NONLINEARITIES

The emitter impedance of a bipolar transistor is inversely proportional to the emitter current. This results in two sources of nonlinear gain:

1. The gain of the grounded-emitter stage.
2. The output impedance of the emitter follower, which results in nonlinear gain depending on how the output is loaded.

THE GROUNDED-EMITTER STAGE

The gain of a common emitter stage is proportional to the emitter current and therefore proportional to the voltage drop across R2. The gain when the output signal is high is less than when the output signal is low. Assuming a 1mV signal from the microphone, the signal is about 20mV at Q1 collector, which means that the open loop gain can vary by almost 2%. However, the negative feedback limits the variation of the closed loop gain to about 0.7%.

THE EMITTER-FOLLOWER STAGE

The output impedance of the emitter-follower stage is inversely proportional to Q2 emitter current. With a 20mV signal, this can vary by about 6% around about 200Ω. However, the soundcard microphone input impedance is about 2.2kΩ, so the effect on open-loop gain is only about 0.3%. The effect on closed-loop gain is only about 0.1%. In addition, the nonlinearity is opposite to that of the grounded emitter stage, so it subtracts from the effect due to the grounded emitter stage.

Capacitively loading the output can

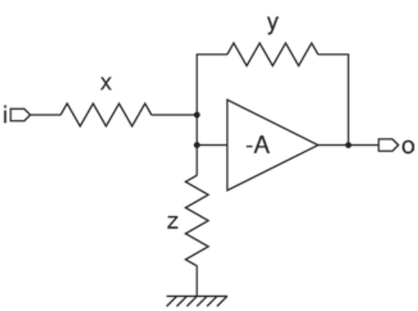

FIGURE 2: Close-loop gain model.

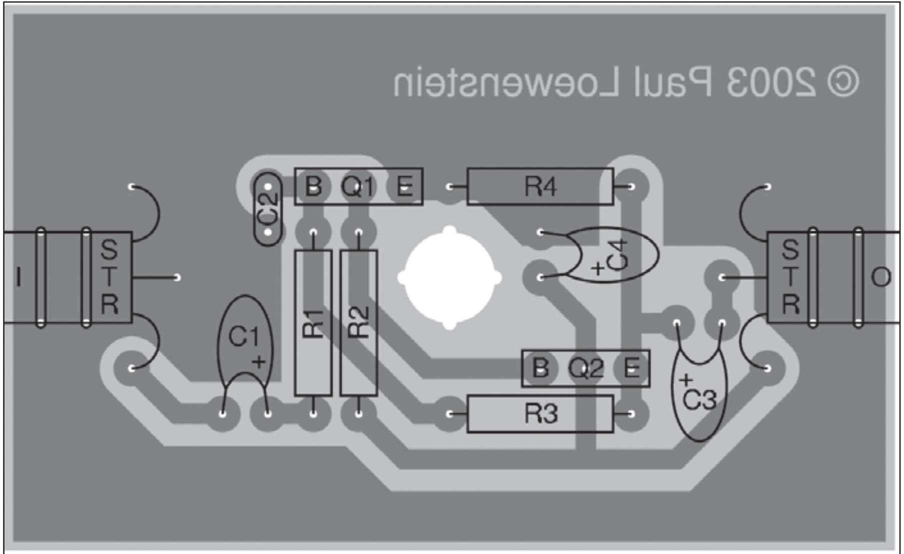

FIGURE 3: Printed circuit layout.

introduce more nonlinear distortion. However, a significant effect would require several nanofarads of capacitance, much more than any likely parasitic.

EFFECTS OF NONLINEARITIES

For variation of gain, as a function of signal level introduces two components to the signal:

1. Frequencies double that of the original (0.6% variation of gain introduces 0.15% second harmonic distortion). Given the high harmonic content of human speech, this is unlikely to adversely affect speech recognition.
2. A DC offset proportional to the signal level. At first sight this may seem inconsequential, but it turns the amplifier into a radio receiver unless you take steps to attenuate radio frequencies[1].

MECHANICAL DESIGN

The printed circuit design is shown in **Fig. 3**. You can use Radio Shack's smallest project box to house the amplifier. It has a convenient stud in the middle which provides a press-fit mount for a 0.1″ pitch perforated board with one hole drilled out to 5mm.

Drill holes for the cables in each end with the box clamped tightly closed.

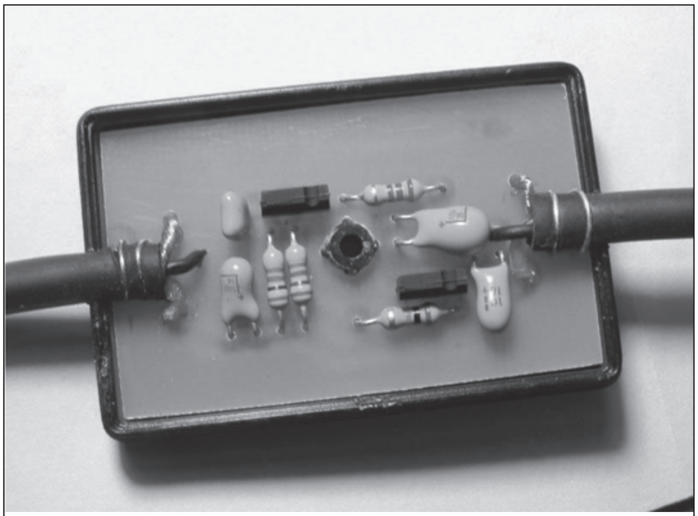

PHOTO 1: Amplifier in Radio Shack box.

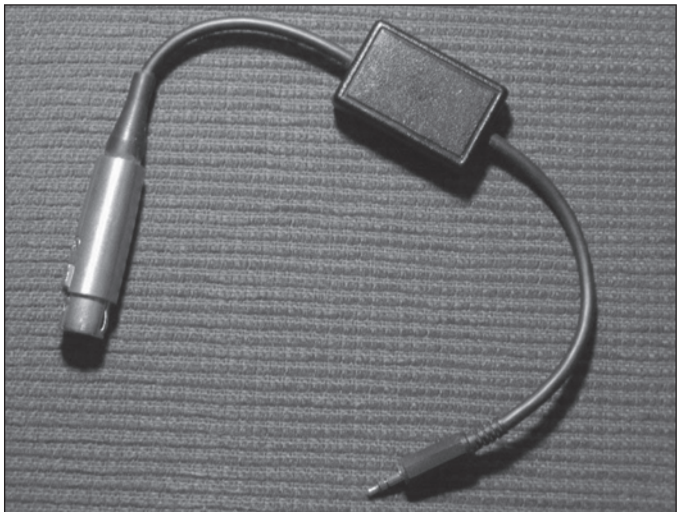

PHOTO 2: Complete amplifier with cables.

Parts List			
C1	tantalum capacitor	10µF	16V
C2	ceramic capacitor	47nF	50V
C3	tantalum capacitor	10µF	16V
C4	tantalum capacitor	10µF	16V
R1	carbon film resistor	100Ω	0.25W
R2	carbon film resistor	5.6kΩ	0.25W
R3	carbon film resistor	10kΩ	0.25W
R4	carbon film resistor	2.2kΩ	0.25W
Q1	silicon NPN transistor	NTE16	
Q2	silicon NPN transistor	NTE16	

Then insert the board, cables attached, into the open box (**Photo 1**), close the box, and secure it with a single self-tapping screw. The complete assembly with cables and connectors is shown in **Photo 2**.

FOOTNOTE

1. The first version of this amplifier rolled off the loop gain using a capacitor to the collector of Q1. When conditions were right it happily picked up KGO at 810kHz AM, causing great confusion to the speech recognition. *aX*

7W Power Amp with Cathode NFB

This push-pull 6V6-based amp design delivers powerful sound, while successfully dealing with the issue of negative feedback.

As far as a tube power amplifier is concerned—at least here in Japan—NFB (negative feedback) has long been a topic for argument among amateur amp builders. Some insist that NFB improves THD, while others maintain that NFB kills the liveliness of reproduced sound. I can hardly judge. However, with low THD and a good damping factor non-NFB amps often sound cool and vivid.

To determine the effect of NFB, I first built an experimental amp using triode tube 6CK4. I placed the selector on a beta circuit to produce three different NFB operations—non-NFB, 6dB NFB, and 18dB NFB. I must confess that my poor circuit design produces a harsh sound at non-NFB operation, while 18dB NFB sounds somewhat static. Unfavorable sound of the non-NFB seems to me due to the poor linearity of 6CK4. So, I am convinced that the comments made by most amateur amp builders are almost correct.

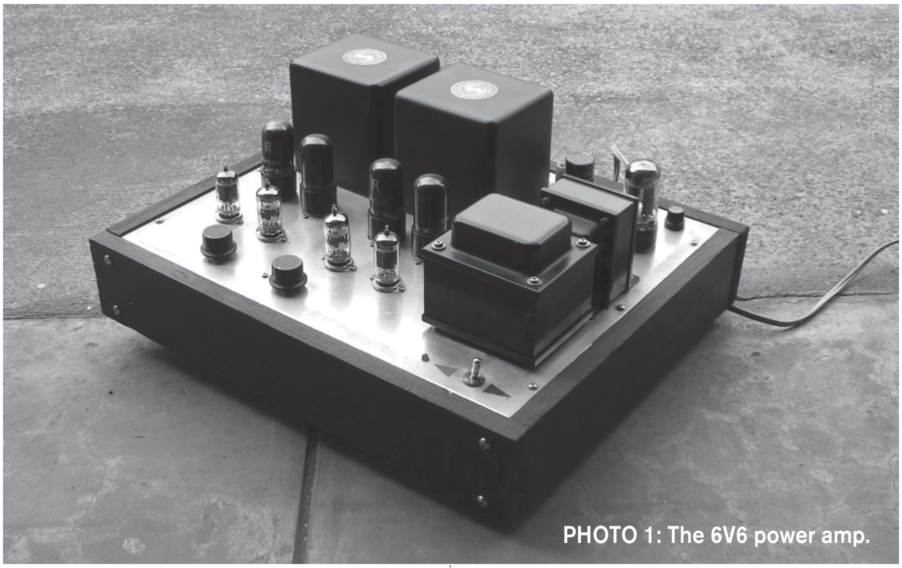
PHOTO 1: The 6V6 power amp.

Also, I realize that a designer must make the amp with low THD level without NFB, which should not be used as a trick to improve the measured performance. Should the non-NFB amp realize low THD and a good DF, the designer would be praised. If you try to design a non-NFB amp, you must carefully select a good power tube with favorable linearity. In my opinion, the benefit of NFB is to stabilize the amp

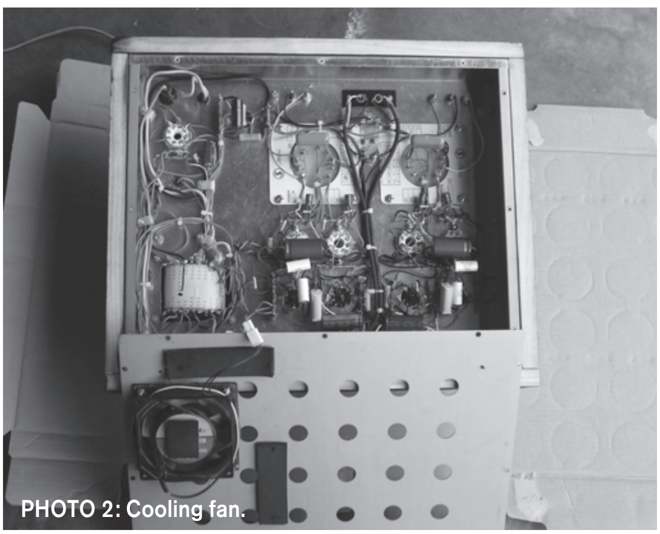
PHOTO 2: Cooling fan.

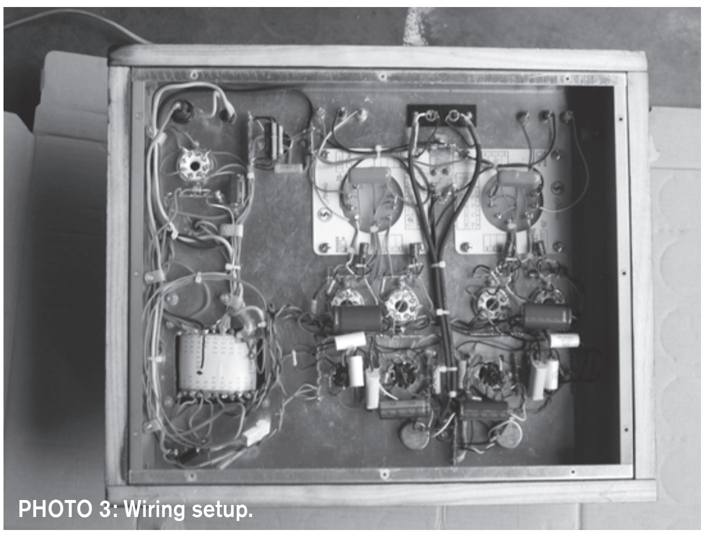
PHOTO 3: Wiring setup.

and keep the performance longer even though the tubes become aged. In this regard, I think this is why commercial amp manufacturers adopt the NFB technique.

6V6 is a good choice if you do not need big power because many amateur amp builders claim it is a nice tube with a tractable character. Also, 6V6 is popular in many brands. The only problem is its high internal impedance.

As a matter of fact, when I first built this amp without cathode NFB, measurement revealed the poor result. The output impedance measured 9.8Ω, which is higher than the 8Ω usual speaker impedance. I am therefore determined to include cathode feedback (CNFB), which dramatically reduces the output impedance to 6.4Ω. How CNFB is valid for the output impedance depends on the tube character. 6V6 seems to react well to it.

CIRCUIT TOPOLOGY

The circuit diagram (**Fig. 1**) is nothing remarkable. Basically, it is a totem pole first stage with a long-tailed pair that drives a pair of 6V6s. A textbook explains that the totem pole guarantees low THD, wide frequency range, low output impedance, and so on. But to me, the totem pole has another charm. Because the cathode voltage level of the upper tube or the plate voltage level of the lower tube is almost half of the B+ given to the tube, this defines the voltage level of the next stage grid when the first stage and the driver stage are directly coupled and make it easy to calculate the long-tailed pair performance. Heater bias is provided so the cathode-heater voltage stays within the limit.

I believe readers are familiar with the long-tailed pair, so I need not describe it further. A 33k resistor between the first stage grid and input prevents oscillation in the event a nervous preamp such as antique AGI511 is used to drive this power amp. You must solder the resistor quite close to the grid pin; otherwise, the high frequency will be spoiled.

The Taiwan-made output transformer has a wonderful wide frequency range. This OPT can take the power up to 30W, which may be too much for this 7W amp; however, the larger OPT guarantees deep bass. The power transformer is apparently weak in terms of electric current.

My experience indicates that you can retrieve more DC current with a choke coil and a tube rectifier than the transformer catalog states, but then, it dissipates more heat. Therefore, I installed a cooling fan just under the transformer. The result is beautiful.

WIRING

There are various ways to wire a tube amp. If you compare a classic Western Electric amp with a famous QUAD

PARTS LIST

Tube	
12AT7 (2)	RCA
12AU7 (2)	Philips ECG
6V6 (4)	Russian made, brand unknown
5AR4	Sovtek

Transformer	
OPT (2)	James 6238h
Power	Noguchi PMS170M
Choke coil	Noguchi PMC1520H

Cooling fan
AC115V 130mA

Volume
50kΩ B (2)

Capacitor	
10µ 450V	
22µ 350V	
100µ 100V (4)	
100µ 250V (2)	
100µ 350V (2)	
0.2µ 400V (4)	film
0.47µ 400V (4)	film

Resistor
220Ω ½W (4)
2k ½W (2)
2.2k ½W (4)
33k ½W (6)
100k ½W (2)
330k ½W (4)
560k ½W (2)
560Ω 2W (4)
5.6k 2W (2)
27k 2W (2)
27k 5W (3)
100k 5W

Chassis	
Aluminum	40cm (W) 33cm (D) 8cm (H) 1mm thick ready-made

Fuse
2A slow blow

Miscellaneous
Tube sockets, speaker binding posts, RCA jacks, and so on are all normal ones I purchased at shops located in the famous city of Akihabara.

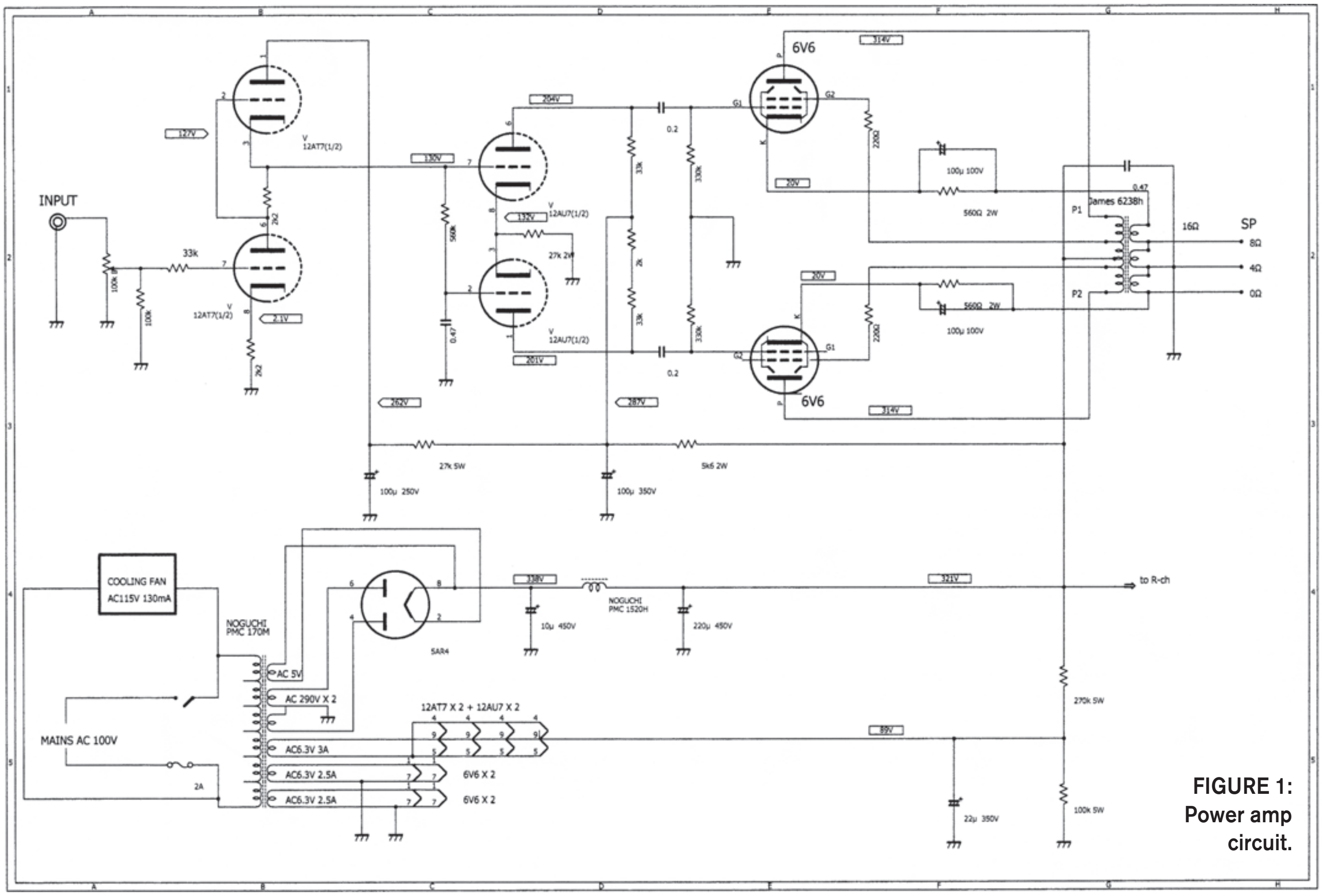

FIGURE 1: Power amp circuit.

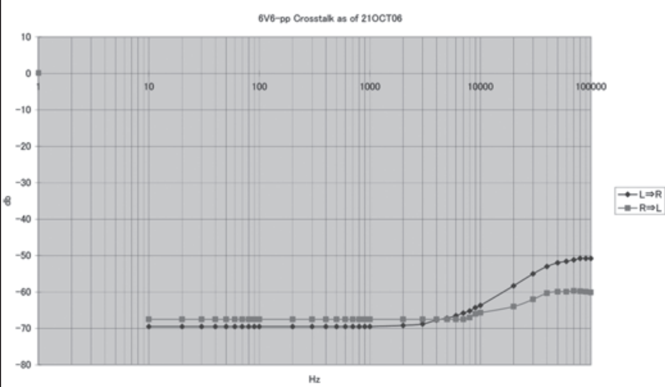

FIGURE 2: Crosstalk measurement.

amp, you should recognize the wiring style is a bit different. Yet, both amps claim to be a first-class product.

When I built this amp, I did not use the so-called ground line. In principle, I wired ground of each stage, then hooked up all ground wiring together at the handmade earth terminal placed close to the OPTs. I hoped that the method would improve crosstalk and residual noise. In addition, I have tried some small tricks which seemed to do something for the sound. With a non-NFB amp, actual manufacturing is more important.

MEASUREMENT

Basic measurement—THD, frequency response, and so on—showed good performance, except for the crosstalk. However,

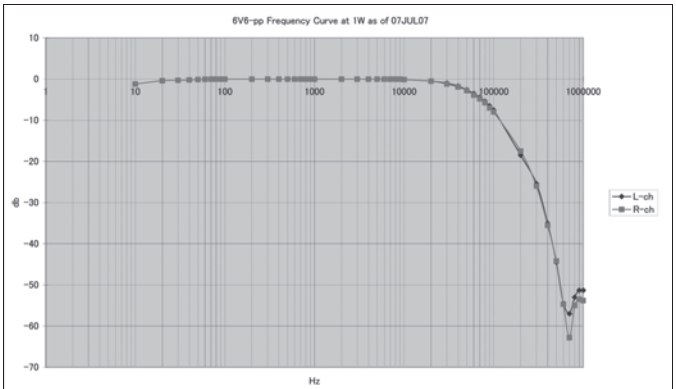

FIGURE 3: Frequency curve at 1W.

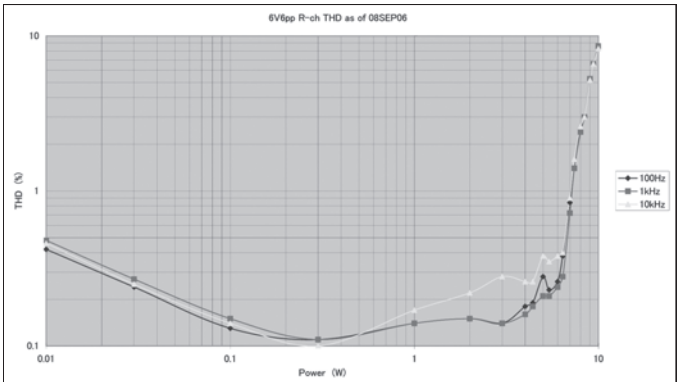

FIGURE 4: THD response.

MEASUREMENT EQUIPMENT

- Audio Analyzer: NF Circuit Design Block DM155B
- Signal Generator (1): NF Circuit Design Block CR-131
- Signal Generator (2): Leader LAG 120A
- Oscilloscope: Iwatsu SS-6050
- Digital Multimeter: Fluke 79

you should understand that the crosstalk includes noise. I am rather confident of a good performance, if you consider that this amp does not apply an overall NFB. The details are indicated in the graphs (**Figs. 2-4**).

Data not included in the graphs:
Total gain: 35dB
Residual noise (Input open)
L 1.8mV RMS   R 1.9mV RMS
Residual noise (Input terminated)
L 1.6mV RMS R 1.4mV RMS

THE SOUND

How this amp reproduces the sound is another matter. I believe that any amp built by an amateur should be evaluated by a third party rather than the builder himself. Therefore, I brought this amp to the RGAA (Rajio Gijutsu Audio Amateur Club) annual meeting last October.

Rajio Gijutsu (Radio Technology) is a leading amateur audio magazine with a long history. The publisher organizes RGAA for its subscribers and offers an opportunity to demonstrate the members' products. They provided me with JBL 4312 speakers, while I brought the Revox B226 CD player and Trio L-07C preamp for my sound demonstration. I purchased the Revox and Trio units as junk pieces, and I completely restored them. (Restoration of old equipment is also my hobby.)

I replayed various CDs from Beethoven's symphony to Frank Sinatra's vocals. The comments made by several members were almost the same: They could hardly believe the amp has only 7W output power because the sound was so powerful. Bass is deep, while the mids and highs are sweet and mellow. I am convinced that this amp has a typical sound of that without overall NFB. *aX*

 By Are Waagbø

Shunt or Not

After testing various shunt configurations, this Norwegian author arrives at a sophisticated shunt regulator version.

I had been looking for the ultimate audio voltage regulator for a long time. But thanks to the well-known Norwegian audio designer Erno Borbely (living in Germany at the moment), it became a reality! Testing more shunt models with selected components and numerous listening tests led to one current-source-fed super shunt, developed by Erno. I developed the PSU, which, together with the sophisticated shunt, sets a new standard for audio voltage regulators.

This regulator greatly improves musical handling. The music gets a new dimension: time, timing, and space! And I don't know of another regulator that has such explosive and live musical performance! Sound image is deep and broad, and it also has details from low down to high frequencies! This low-noise PSU/shunt is meant to power all high end units: preamp, DAC, phono, and so on. Both my DAC and phono were battery powered before, but no more.

SHUNT OR SERIES REGULATOR?

At the outset, I must admit that I am not an electronic engineer, just a passionate 50-year-old hi-fi amateur, with many years of practice and experience! Many of my projects have succeeded, and some of them have not!

Some time ago I became preoccupied with the idea of finding a good audio voltage regulator for my preamp. I had been searching the net, and so I encountered many super low-noise, high-speed series regulators. Unfortunately, none of them described how they sounded. I hoped for a glimpse of the famous STAX shunts, but I gave up. I found some other shunts, but they were not symmetrical designs or did not have the voltages or current I needed.

So I bought one of the series regulators—the JSR 03/05, which is super low noise, super fast. The funny thing about this discrete design is that it is supposed to have preregs with LM 337/317, but I left them out, because these preregs sort of compressed the music in one of my older Sulzer designs.

This super design could do no better than Erno's all-FET series reg, and I had to admit that the JSR 03/05 was a waste of money. I made a simple zener shunt from some old zeners I had, and it was not bad at all. While it was noisy, there was something different about the musical presentation, which I thought was positive. Erno sent me the diagram of an updated shunt model. And from here, one thing led to another, because of the great improvements in the sound.

Several listening tests led us to a final shunt version, designed by Erno himself. This final shunt outclassed all the other tested shunts, and is now incorporated in Erno's unique all-FET series, known by many audiophiles. Besides its excellent audio performance, the shunt is capable of absorbing transients in the mains power, coming from refrigerators, oven thermoswitches, and lighting armatures. And also because it delivers a constant-current with constant voltage, there will be no pumping or oscillation of the transformer and capacitors voltage.

Shunts also have an ultralinear low impedance all over the frequency band and into MHz. It is also called a parallel regulator because it sits parallel to the power supply. In this manner the audio is connected directly to the power supply and does not affect the audio idle or amplifier feedback, but with a series regulator it will. I think this is the main reason for the shunts giving the better musical performance (**Fig. 1**).

The shunt idea is also based upon shunting current greater than consumed.

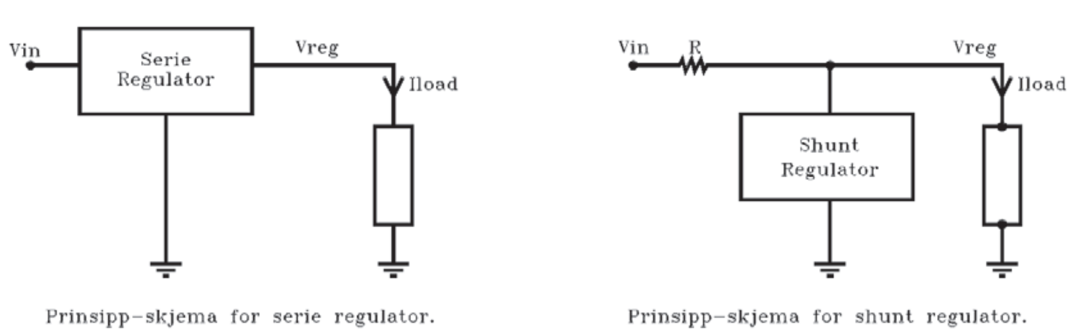

FIGURE 1: Series/parallel regulator.

In my case, I shunted about 400mA for my preamp using 200mA. This amount is overkill, but I had to start somewhere, and mostly all the tests were shunted at about 400, just to eliminate at least one parameter. For later tests I shunted about 330mA, and I don't think I noticed a degradation of the sound. Greater shunt current gives lower impedance.

The drawbacks are that the shunts consume more power and, much like a Class-A amplifier, became hot, because all the power that is not consumed by the amplifier will end up in the shunt output transistors. Another factor is that the supply and shunt will be bigger than the traditional PS/regulator combination.

SHUNT TYPES

The **zener shunt**, the simplest of all shunts, consists of feed resistor R, the zener diode, and a capacitor. Resistor R determines the total current that should be sinked by the zener and the load! It is a bit noisy, but I have replaced 3-pin LM regulators with this in buffers with a gain of 2 with good results! This is also the first one I tried on my preamplifier (**Fig. 2**).

The **simple shunt regulator** shown in **Fig. 5** (left) consists of two transistors, and because of the gain factor, it actively decreases ripple and also noise. You will notice at once the much lower noise of this regulator. Zener diode Vz = 22 gives about 23-24V, and ideal PSU voltage should be 6-7V above shunt voltage (regulated voltage).

If you look at **Fig. 5** (right), you will notice that the feed resistor 27R is replaced by Erno's constant-current regulator using a MOSFET. The input resistor of 2R2 gives a current of approximately 300mA. This constant-current

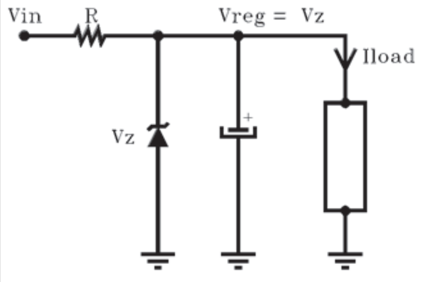

FIGURE 2: Zener shunt regulator.

fed shunt gives an incredible improvement in sound and sound stage. More about this later.

With the **all-FET super shunt** (**Fig. 6**) you will notice the same constant-current regulator as in the simple super shunt. The regulator, which is designed by Erno, uses the sophisticated all-FET topology! The even greater gains in this design offer better regulation and, of course, lower noise. I am now using this with my phono preamp (gain 70dB).

Even greater gain is offered by the **op amp shunt**, and this should theoretically be the best of all the regulators, with lowest noise of them all! You will once again see the same constant-current regulator as in the simple shunt and the all-FET shunt (AFS). Despite the lowest noise of all, it cannot cope with the all-FET super shunt's musical performance (**Fig. 7**).

LISTENING TESTS

I have been using battery power for both DAC and MC phono for quite some time. For my preamp I have been using Erno's all-FET series regulator. And I have already tried out a zener shunt for my preamp, despite the higher noise level. I told Erno I liked the distinct spacious and dynamic sound. A more important impression was that the loudspeakers became sort of invisible. After I received a diagram from Erno on the simple shunt, it took me just a few hours to hardwire this.

With the simple shunt, I could hear the lower noise as well as greater image and sound stage. Dynamics are even better, and bass is deep and powerful, with good punch low down. Even dynamics and bass are greater than my series regs.

I had been playing my new shunts for a few days before Erno faxed me the constant-current regulator, which replaced the feed resistor 27R. I also hardwired this CCS and a solid heatsink onto an output board, as I did for the shunts. I adjusted the current to be about the same as before. Shortly after I was ready to listen and was very excited.

My new constant-current fed simple shunt had less or very little gain compared to just the simple shunt, with feed resistors. Also, noise seemed to be lower, with even greater and deeper sound stage. Dynamics and low down attacks were super, and so was detailing. This shunt gives you a realistic sound stage, like sitting in the third or fourth row of a concert. The dynamics are like having my amplifiers bridged, which they are not. I was so impressed by this regulator I built another very soon afterwards and was so impressed that I threw away my battery supply (eight Panasonic HV12-07 and 40,000µF capacitors, and the charging equipment).

With all respect to the Panasonics, they tended to be flat and dull compared to my new shunts. I even used this with my phono: Erno's all-FET MC, gain 64dB. Even with lots of noise, I still preferred this to both series regs and batteries. After I conferred with Erno, we agreed to do something about the noise. He wanted me to test

TIPS WHEN BUILDING

Always test the shunts one after another for oscillation, with load. I like to be sure of a stable condition and tested with no load condition, but only for a short moment. And be sure to test with the shunts built into the cabinet. I recall one occasion in which there was no oscillation outside the cabinet, but there was inside when hooked up to the amplifier! I solved this by insulating input/output grounds, and grounded only outputs to the chassis via 0.1µF!

Another consideration is that the shunt PSU and regulator use more space because of bigger transformers and capacitors and larger PCBs and heatsinks. You also must transfer the heat away. You might consider building a separate cabinet for the shunt supply, including transformers. The ideal enclosure is vented top and bottom. If you want to use a closed box, you must make some slots or holes in the top, bottom, front, or back plates. My enclosures are shown in **Figs. 3** and **4.**

You could also install a little fan (6 × 6 cm) for better cooling. I used an old cell phone charger 5-6V DC to feed a little 12V fan, which is practically noiseless. It is also a good idea to mount the capacitors in the coolest place in the cabinet.

For phono use you should have the transformers in a separate enclosure, at least 1m away. And use thick cables 0.75-1.0mm^2 for the supply, preamp, or DAC. For long supply cables, I have mounted a pair of Panasonic 2200µF or 3300µF capacitors in the preamp or DAC cabinet close to the amplifier V+/V-.

For power plugs I use the cheap DIN U 8 or DIN C 8, and all my power connects have the same pin-out and are compatible. I have used these for more than ten years, and despite heavy use, they have never failed. —**AW**

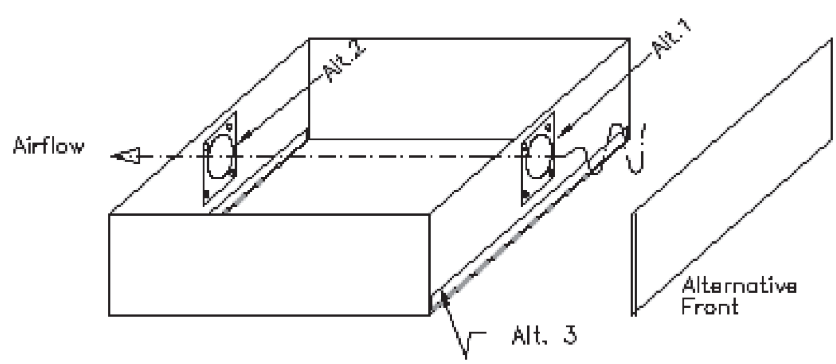

FIGURE 3: Suggested cabinet and cooling.

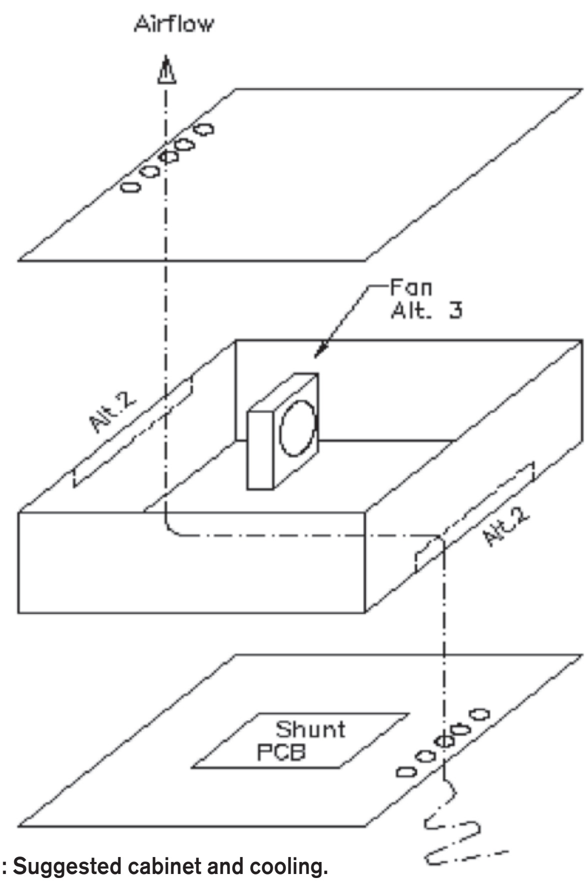

FIGURE 4: Suggested cabinet and cooling.

two completely new shunt models, one discrete design and one op-amp based. And, of course, they should be constant-current fed. They would be readily assembled, tested, and sent to me.

I had to admit I didn't believe further improvements to the sound by the new shunts were possible. Still I hoped when the new shunts arrived. I had already made a pair of plug connections to plug in or out of the different shunts!

ALL-FET SUPER AND OP-AMP SHUNTS

A brief listening test of both shunts indicated that they both had extremely low noise and produced deep sound stage and super detailing mid and top! But the greatest difference was the much smaller sound stage. The worst was that bass could not cope with mid and high frequencies; and they lacked punch low down, too. I was surprised, and a bit disappointed.

I went through the diagram for the shunts, and I noticed that Erno used other output transistors and a source resistor. With all respect for the 2SK2030/J313, I wanted to replace them and short out the source resistor! I used my own transistors, K 1530/J 201 for CCS and the same or IRFP 150/9140 as shunt elements. It happened to be the only choice of power transistors in all the shunts. But every attempt to alter the components of these low-noise shunts was rewarded with oscillation!

Very frustrated, I gave up on the op-amp shunt for the moment and concentrated on the all-FET super shunt. This one at least became stable with a compensation capacitor to the output driver, C11 and C12. I did more tests with other components and had some real bad experiences. It was partly my own fault, because I used a different PSU, transformer, and caps. I had to go back and do the tests again, but this time with the same PSU.

Listening to the upgrade, I noted that the bass, mid and high frequencies all played together now, in a manner similar to the CCS/SS, with a deeper and broader sound stage. It was never grainy, and seemed to have extended low and top. The ideal use was, of course, for my phono, because of the very low noise, but I was also using it for my DAC. I loved the CCS/SS for the very close sound stage, although it was noisier and a little bit grainy on some occasions! It was difficult to decide which of them I preferred, so I used both, switching between them.

FINAL UPGRADE

Trying to make op amp shunt sound better led to the final upgrade of AFS. I tried more op amps, and did the same transistor mod, as with the AFS. With the LF411, it played very convincingly, with all the details but the smallest sound stage. It also had incredible dynamic and punch, even better than the AFS. I wondered about this for a few seconds, then I remembered that I had changed the 47µF ELNA with MV-AX Panasonics, in an earlier attempt to stop oscillation. I wanted to try this cap mod with AFS.

I don't regret changing these caps, because now this shunt plays like a refined CCS/SS, with extended bottom and top ends. And, of course, the dynamic was increased, and the sound stage is very like CCS/SS. Building a second one later just confirms the improvements with even more open and clean sound stage. I now have four AFS regulators; the latest was to my hybrid tube headphone amp (and AKG 701). Before this it was fed with Panasonic batteries and capacitors.

IMPRESSIONS BY SHUNTING POWER

I have no fancy loudspeaker cable, interconnects, nor perfect speakers, but I

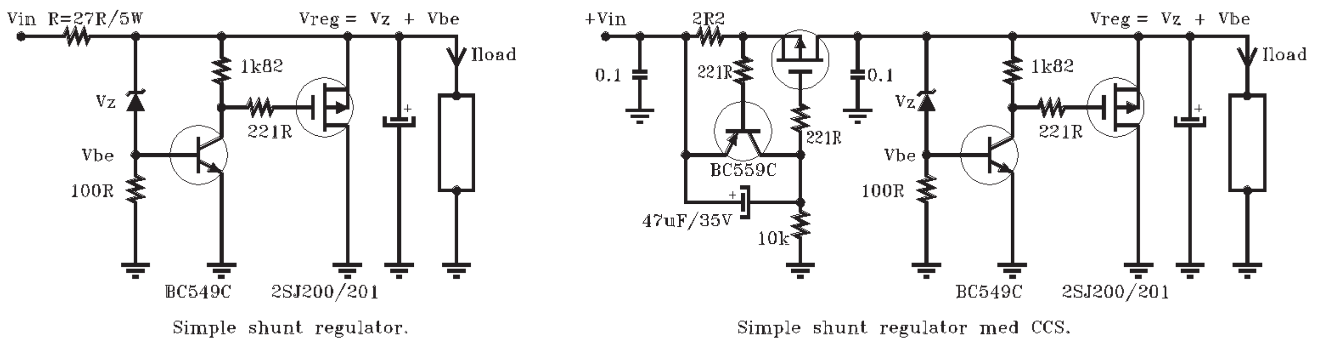

FIGURE 5: Simple super shunt.

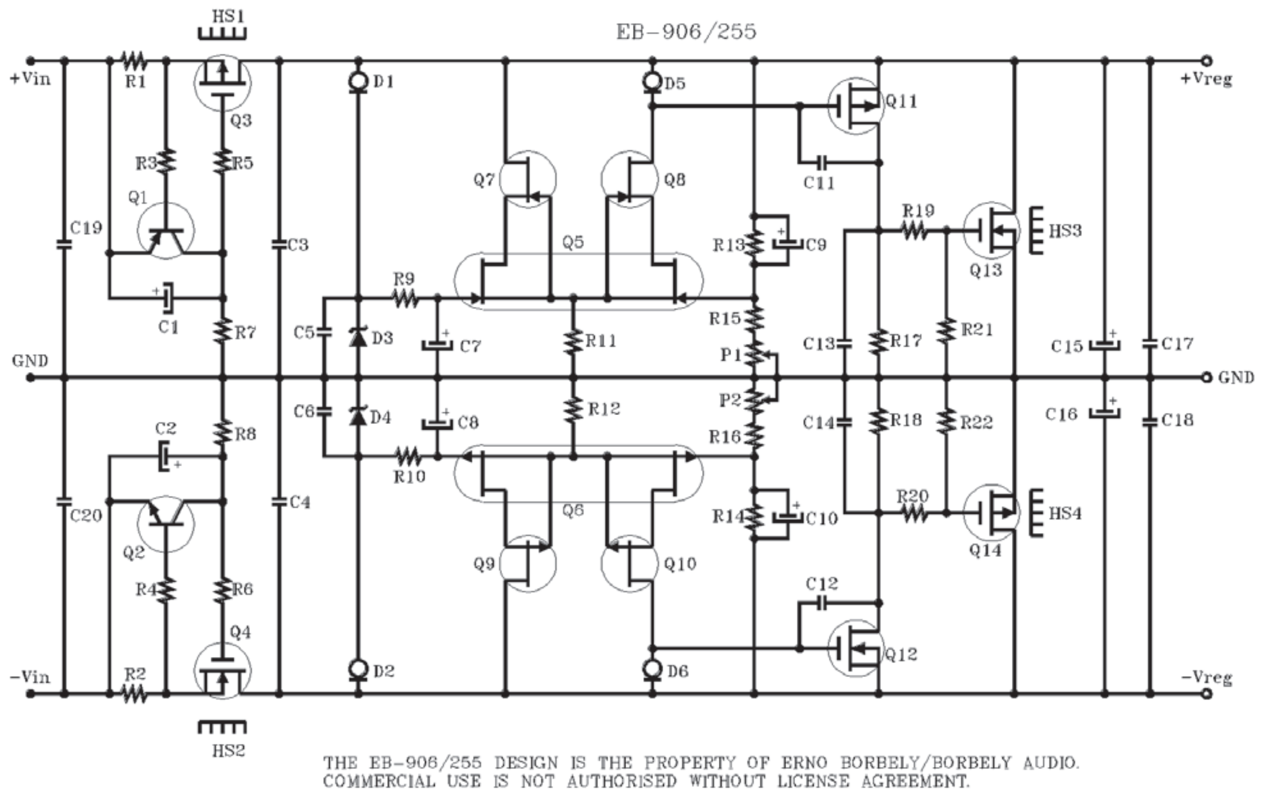

FIGURE 6: All-FET low-noise super shunt regulator.

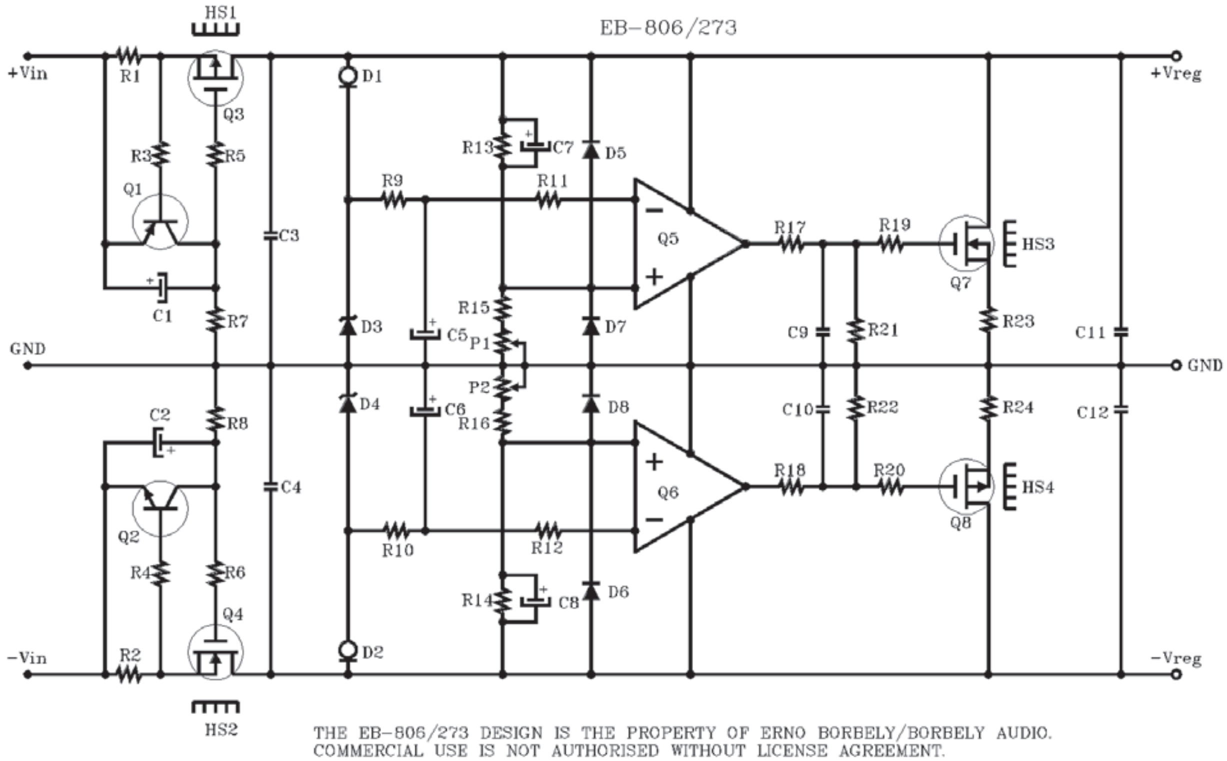

FIGURE 7: Op amp MOSFET dual low-noise super shunt regulator.

have never played such realistic music. I don't think you will find any cable that will give you the same improvement as this shunt does! I have tried many cables myself, and some are good. But I refuse to accept any cable that costs more than a high-quality industrial cable. Think of what cables record studios are using.

UPS AND DOWNS

Testing the AFS and op amp shunt was frustrating from start to finish, working with both at the same time. I had to put away the op amp shunt for a moment, because all my modifying attempts were rewarded with oscillation or bad sound. The AFS remained stable after just one compensation capacitor, and I modified it more times without trouble. I also tested two transformer sets and two capacitor banks. This expensive transformer and caps reversed the whole progress, and I had only myself to blame. The moral of the story is to use exactly the same transformer set and capacitors I used for the successful CCS/simple shunt. I had to do all the listening tests again.

SELECTING COMPONENTS

A very good help when selecting components has been Erno's ladder attenuator, with Caddock and Dale resistors. This attenuator has an incredible open and transparent sound. The genius of it is that the signal goes through only two resistors, regardless of attenuator position! At the same time, in balanced applications, it will be 100% balanced in all positions, just as the feed resistors are equal. See **Fig. 8** ladder attenuator.

POWER SUPPLY DEMANDS

To get the best out of the shunt, you must have a rigid, stable power supply! That means bigger transformers and greater capacitors than used with the series regulators. I have also tested some transformers.

My first choice is the R-core transformer, which has incredibly clean sound and transparent sound stage. The mechanical noise here was the lowest of all. They smelled good, too. I tested the 120 VA, and used two of them! The importer is Borbely Audio.

My second choice was the Plitron toroids from Canada. I tried two of the 160 VA, which produced a very clean sound, with a little warmer presentation in the mids and lows. Also these had very low mechanical noise.

My third choice, and the best above 100 VA, was the fully encapsulated transductor, formerly Noratel. These sounded best of the big toroids, and had low mechanical noise. Some of the other big—not encapsulated, or center encapsulated—units had punch and bass, but tended to add a gray mist to the sound, not so fully transparent as the best. The small toroids—30 to 50 VA—seemed to compress the music and lacked attack and bass. These were also very noisy, despite a maximum rating of 1A and consuming not more than 0.2-0.3A! I also selected the capacitor and rectifiers from listening tests when I earlier tested series regulators. For signal audio amps and power amp driver boards, I always use dual secondaries and two bridge rectifiers (**Fig. 9**).

For the rectifiers I have used both BYW28, 4A, and HFA08, 8A. The only capacitor is Panasonic FC, which has low ESR and low impedance. But you must have many of them paralleled. I suggest minimum 20,000µF per rail for shunting 200mA, and 30,000µF

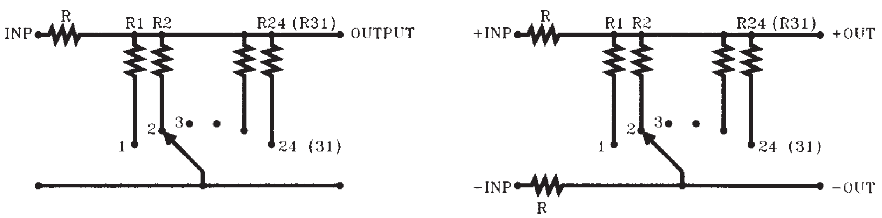

FIGURE 8: Ladder attenuator

for 300mA. Use as many as you can afford, which gives a more distinct and rock steady listening image.

The Panasonics have simply no comparison, with their ultra smooth frequency band and no coloring. You could, in fact, mix in some Sanyo MV–AX components, but not too many, because they tend to sound a bit clinical. I have a drawer full of new Nichicon, ELNA Cerafine, ALS 30, PEH 169 /200 capacitors, and they all sounded bad. And just paralleling the Panasonics with the other caps degraded the sound.

AT LAST

I wish to thank Erno Borbely for his contribution to this project, and his assistance in working out problems. For shunt or PSU components or specs, visit www.borbelyaudio.com, which has all you need for a shunt or other hi-end information. If you are planning to build any amplifier, I cannot think of any better choice than his new-generation all-FET audio series. *aX*

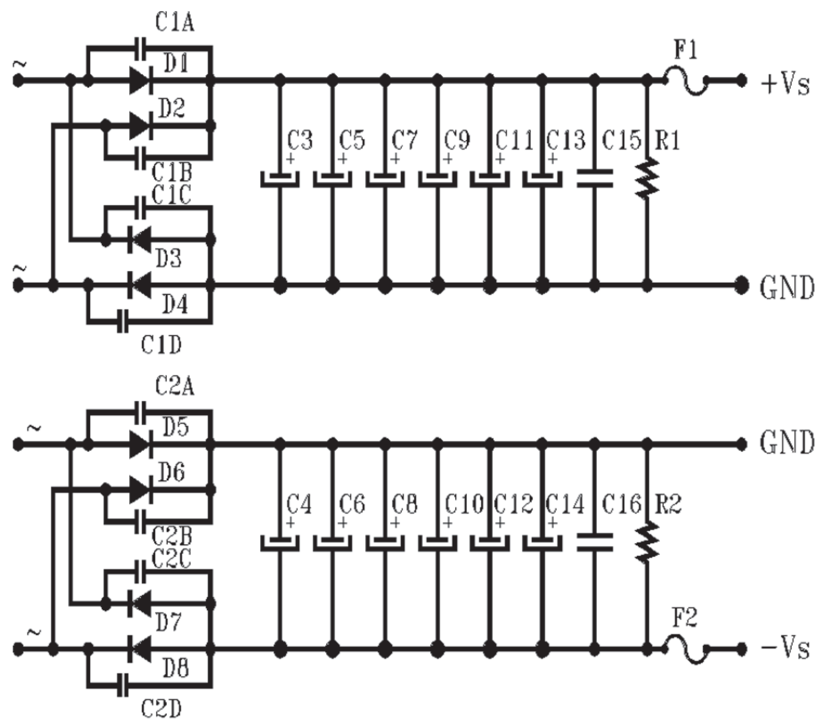

FIGURE 9: ± power supply for shunt regulator.

AES Multi-Tracks in New York 2007, Pt. 1

Here are some highlights from last fall's AES Convention in New York City.

Almost 21,000 professionals invaded New York's Jacob Javits Convention Center for the 123rd Audio Engineering Society convention. Attendees had to choose from a diversity of topics in more than 150 sessions including paper presentations, panels, tutorials, workshops, master classes, and special events on broadcasting, education, pro-audio history, live sound, architectural acoustics, and hearing. Coincident was the Broadcast Conference—co-presented by the AES and Dolby—which covered the entire broadcast chain from facility design to the receiver. Almost 500 companies had exhibits and demos. There was far too much for me to experience it all.

LIVE SOUND SYMPOSIUM: SURROUND LIVE 5

Fred Ampel (Technology Visions; www.TechnologyVisions.com) produced and chaired the fifth Surround Live event, with major support from the Sports Video Group and sponsorship from Sennheiser, Neural Audio, Harris, Digico, Beyer, and Klein & Hummel (which provided the loudspeaker systems). Strictly speaking, Surround Live is not part of the formal AES convention, but is an official symposium event that is always held in conjunction with the convention. Surround Live 5 (SL5) drew well in excess of 100 attendees. A surround sound system was set up around the seating area for presenters' demonstrations. Ampel used Audio Control's Iasys for initial calibration, which shortened the process.

Due to scheduling conflicts, the event was held in the Broad Street Ballroom, within a block of the New York Stock Exchange, instead of its usual location within a short walking distance from the convention center. The ballroom is bare marble and concrete with acoustics like the world's largest bathroom! Ampel had heavy drapes hung around the seating area, which helped a great deal and enabled most of the demonstrated effects to be heard.

FEATURED SPEAKERS

Kurt Graffy (Arup Acoustics; www.Arup.com), SL5's keynote speaker, talked about how you determine a sound's location—the inter-aural level difference (>1.5kHz) and inter-aural time difference (<0.5kHz; a 1.5ms time difference moves a sound from centered to one side), the head-related transfer function (including the pinnae effect), the ISO-226 equal loudness curves (which look like the Stephens curves, slightly different from the Fletcher-Munson curves), masking, cochlear-cortical filters, and other physical and psychoacoustic characteristics. Human hearing is nonlinear with frequency, time, level, and angle (vertical and horizontal). Factors that you perceive and affect what you hear include depth, width, distance, direction, balance, timbre, dynamics, and direct-versus-reverberant sound. There are substantial environmental distortions to dimensionality including room reflections, noise, reverberation time, room modes, speaker location(s), and listener(s) location(s). His overall message was that hearing is complex, and you make many pro-audio decisions based on simplified models.

Jim Hilson (Dolby Labs; www.Dolby.com) addressed the pressing issue of levels and Dolby Digital metadata. DialNorm is a parameter to normalize the dialog level to -31dBFS. Dolby Digital playback (decoding) will drop the level from the number set in metadata to -31dBFS; thus if DialNorm is set to –27, program level will be lowered 4dB. He mentioned three measurement periods (a 10-second average, a 10-minute average, and a full program average), emphasizing that the program should be of normal dialog, not shouting or whispering, and should not include other loud effects or music, which lead to an incorrect value. DialNorm decoding is mandatory and always active by default in all Dolby Digital decoders (some decoders offer a menu choice to turn it off). For the full dynamic range experience it is critical that only one DialNorm value be set for the entire program, as is the case for almost all movies.

Tom Sahara (senior director of remote operations, TNT Sports, Turner Networks; www.Turner.com) spoke about the difficulties in getting good live-event sound for broadcast. Problems include mike selection and placement (for example, miking the track at NASCAR races involves consideration of levels, dirt, and possible rain), a contract sound-mixer monitoring in a remote truck he's not familiar with, while keeping the mix balanced for surround, stereo, mono, and playback systems from movie/home theaters to hand-held devices.

His fundamental premise is to keep it simple. He picks the viewer's sound field perspective, plus how fast and often it changes—should the viewer's sound field match the image for the in-car camera as well as various close-up and far-field shots; should the sound transitions be hard-cut or cross-fades, and so on. The surround channels' sounds must be carefully chosen to not distract the viewer from the screen and to ensure stereo and mono mixdown compatibility, particularly because most viewers use stereo or mono playback. Mike Pappas (KUVO-FM) emphasized that point, saying that at least 30% of his audience listens on small radios in mono and that he is careful to ensure they get good sound.

Sahara pointed out that at each step in production-through-playback there are many potential causes of sound problems. He strongly recommends—and practices—staying away from the "this is how we've always done it" approach and enter each situation by gaining a thorough understanding of how the considered technology works, what it can and can't do, and how it can most effectively be used to produce the sonic reality sought. Program creators are challenged to deliver sound that is compatible with mono, stereo, and poorly set-up surround systems, yet produce a sound that blows the minds of those with great, properly calibrated playback systems.

Intelligibility is the most important characteristic in design and delivery of broadcast sound. Sahara has determined that the announcer should have a maximum average level around +12dBA (slow-averaged) over the effects, and the surround levels should be –6dBA compared with the effects. If an LFE is used, its level should be even with the effects' level. He ended his presentation with an impressive video excerpt from a NASCAR race, with the image of the cars, and their sound, coming toward the camera and receding off to the right-rear.

Michael Nunan (CTV Television; www.CTV.ca) has long said that in a surround system the front channels are not merely stereo-plus-centered-dialog, but should be used to produce a sound field continuum across the screen. His production manager complained that while watching hockey, football, or even ice skating, the announcers are center-front even when they're not on the screen—their disembodied voices interfering with the front sound field. Nunan produced a demo of sporting-event sound fields with two announcers, one in the left-surround and the other in the right-surround. With the right levels, and properly set-up surround playback, there was no problem understanding the announcers while more fully ex-

periencing the front sound field. The announcers being behind was not distracting from the event; in fact, it allowed better integration of the image and sound field. Obviously there must be consideration for badly set-up surround systems, plus stereo and mono mixdowns, but I found the demonstration thought-provoking.

Other presenters were Fred Aldous (Fox Sports), Randy Conrod (Harris), and Mike Pappas (KUVO radio). The panel discussion "Surround Operational Issues: What Happens When It Leaves the Truck?" was moderated by Ken Kerschbaumer (Sports Video Group), with Ron Scalise (ESPN Remote Operations Audio Project Manager), Jim Starzynski (principal audio architect for NBC/Universal Advanced Engineering Growth Center), Robert P. Seidel (VP Engineering & Advanced Technology, CBS Television Network), Bruce Goldfeder (Director of Engineering, CBS Sports), and Randy Conrod.

Beyerdynamic (www.northern-america.beyerdynamic.com) demonstrated its Headzone® earphones that take a digital audio input and reproduce a stereo or surround sound source with "out-of-head localization of the center channel" and offer limited room size and acoustics adjustment, plus head tracking to keep the image at the "stage." These are fine if you don't need a speaker-reference, but the Smyth Research system should be a better fit for location mixers who must work in an unfamiliar environment, such as a remote location truck at an NFL game.

Surround for live events is challenging, but when done well, involves the viewer much more than traditional sound approaches and adds lots of fun and a sense of reality to the program. Surround Live 6 will coincide with the 125th AES convention in San Francisco next fall.

SAMPLING THE CONVENTION

Perception

"Room Reflections Misunderstood?" (AES preprint 7162) Siegfried Linkwitz (Linkwitz Lab; www.LinkwitzLab.com) evaluated a dipolar versus a monopolar speaker for "perceived differences in their reproduction of acoustic events" in a domestic living room. His criteria were timbre, phantom image placement, and sound stage width. He claims they sound similar even though their frequency responses and sound fields are significantly different, plus the effect is "further enhanced by extending the dipole behavior to frequencies above 1.4kHz." The monopole speaker is omnidirectional below about 3kHz.

Linkwitz noted six "impediments to creating a realistic impression of an acoustic event:" non-uniform polar response, inadequate dynamic range (not able to reproduce realistic levels), speakers too close together and asymmetrically placed, room treatment that changes the spectral balance of reflected sound, room equalization above low frequencies, and recordings with too many microphones in separate sonic spaces. He seems recently to have realized that off-axis frequency response needs to closely match the on-axis response—a fundamental truth that is decades old to many.

Linkwitz initially tunes his systems with test equipment, but performs fine tuning by ear, with source material he knows well.

SIGNAL PROCESSING FOR ROOM CORRECTION

"A Low-Complexity Perceptually Tuned Room Correction System." (AES preprint 7263) The instigation behind this effort by James D. Johnston and Serge Smirnov (Microsoft; www.Microsoft.com) was that in many rooms speakers are not identical, locations are not ideal (asymmetrical and at various distances), and room acoustics are antagonistic. This results in non-optimal reproduction. Johnston explained their "room-correction algorithm that restores imaging characteristics, equalizes the first-attack frequency response of the loudspeakers [in the critical bandwidth, the first 6-8ms is the most important for room EQ, even at higher frequencies], and substantially improves the listeners' experience by using relatively simple render-side DSP in combination with a sophisticated room analysis engine that is expressly designed to capture room characteristics that are important for stereo imaging and timbre correction."

BROADCAST SESSIONS

"Audio for HDTV: DialNorm." This panel discussion involved chair Andy Butler, Mike Babbitt (Dolby Labs), Tim Carroll (Linear Acoustic; www.LinearAcoustic.com), and Robert Seidel (CBS). Seidel said that an acceptable Leq(A) needs to stay within about a 7.5dB range to not be bothersome to listeners. CBS receives many program files with widely varying levels and incorrectly set DialNorm, and has had many instances of a quiet program segment just before a loud commercial, causing an unacceptable jump in measured and perceived level (sometimes this is unavoidable because the program's level, while OK overall, happens to have a quiet segment just before a commercial break). Because of these problems, CBS sets the DialNorm metadata to –31, effectively turning off the volume leveling function in all receivers.

Turning off the DialNorm function results in the widest variation in playback levels possible. Steve Lyman (Dolby Labs) told me that the DialNorm and DynRng metadata parameters must be processed by all Dolby Digital decoders, and that the default in consumer equipment is almost universally ON, attempting to level the inter-program volume and compress the dynamic range of the soundtrack. Because of the nonlinear transfer function in the playback processing, turning the DialNorm function off by setting it to –31 exacerbates the level-change problem.

The author continues his look at the extensive offerings at the 123rd AES Convention in next month's issue. aX

review | By Julian J. Bunn

Speaker City's
PS10 Subwoofer Kit

PHOTO 1: The PS10.

This kit comprises a black MDF cabinet, 200W amplifier module, a 10˝ woofer, acoustic damping, speaker gasket, and associated hardware (**Photo 1**). It arrived in two large boxes from Speaker City in Burbank.

The kit is advertised as requiring no soldering, and the construction as simple. There were no instructions included in either box. However, after an examination of the components, it was obvious how everything fit together. The amplifier module, the face plate of which is shown in **Photo 2**, clearly was intended to fit in the aperture at the rear of the pre-made MDF cabinet. The rear side of the amplifier contained some chunky capacitors and impressive-looking power transistors mounted on substantial heatsinks (**Photo 3**).

The front of the cabinet (**Photo 4**) shows the heavy 10˝ woofer from Goldwood Sound, model GW-10PC/4, with a specification of 400W power between 39Hz and 5kHz, and 94dB SPL per meter squared at one meter distance. This was clearly intended to be mounted in the front of the cabinet, using the provided gasket and long screws.

I began by mounting the amplifier module in the cabinet and attaching the cabinet feet (**Photo 5**). I then used the acoustic stuffing to fill the rear section of the cabinet around the amplifier (a view of the interior of the enclosure is shown in **Photo 6**). At this point I noticed a small area of damage to the enclosure that had possibly occurred in shipping—not serious, but spoiling the otherwise perfect condition of the kit (**Photo 7**).

I was unsure how tightly to pack the stuffing, so I operated on the principle that half of the provided material should go in the rear section of the cabinet (between the amplifier module and the cross brace), and the remainder in the front half, behind the speaker. **Photo 8** shows the fully stuffed cabinet before attaching the woofer.

I used the spade ends on the wires from the amplifier, rather than baring the wires and attaching them to the attractive gold banana plug sockets on the woofer (**Photo 9**). Finally, I carefully placed the gasket around the lip of the woofer and screwed it into the cabinet (**Photo 10**).

The total assembly took less than 45 minutes. For the listening tests, I added it to my existing system, placing it on the stone hearth centrally located between my Jean Marie Reynaud Milles-

PHOTO 2: Faceplate of the KG-5150 amplifier module.

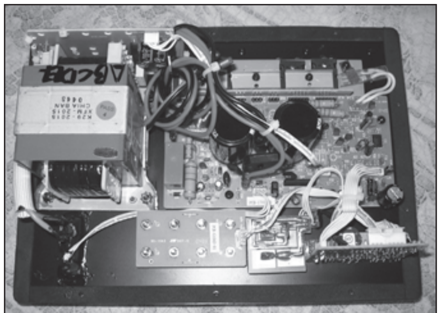

PHOTO 3: Circuit side of the amplifier module.

PHOTO 4: The front of the MDF cabinet, showing the woofer opening and two ports.

ime V loudspeakers, which are accompanied by my Acoustic Triangulators at the left and right rear positions. I cabled it to the subwoofer output of my Pioneer VSX-815 home theater amplifier's subwoofer line output, checked the connections, and applied power.

SUBJECTIVE IMPRESSIONS

The controls on the faceplate at the rear of the subwoofer include an on/off switch, a toggle switch for selecting 0 or 180° phase (required when matching the phase to that of the main speakers), a toggle switch that enables a low-pass filter, and two potentiometers: one for the characteristic frequency of the low-pass filter between limits of 50Hz and 100Hz, and the other for setting the output level of the amplifier.

With the low-pass filter selected, it is sometimes difficult to tell whether the woofer is operating at all: The amount of audible sound is, of course, much reduced. I attached a pink noise generator to the line inputs and was able to distinguish between the 50Hz and 100Hz settings of the low-pass frequency pot. With the low-pass filter out of circuit, there is much more signal, and I decided to leave it that way for some music listening tests.

I found the adjustment of the output level difficult: I wanted to turn it up so I could tell it was making a difference, but then all bass was too intrusive. After tinkering around for a while, I used the very handy automatic system provided by the Pioneer VSX-815, which uses a microphone input and a set of test tones to automatically set the levels for all speakers attached to it, and which involves placing the microphone at the usual listening position.

After going through that process, I settled down to listen carefully. One passage that I usually play when evaluating sound systems is the Allegro non Troppo movement of Shostakovich's 8th Symphony (Haitink, Concertgebouw Orchestra, Decca 411 616-2), which was especially impressive with a noticeably fuller and more exciting big drum sound at the end of the movement. Following that, I tried a change of mood, selecting the track "Sorrow" from Pink Floyd's "A Momentary Lack of Reason" (EMI CDP 7 48068 2), which contains some of the most fruity, bassy, and deeply distorted electric guitar that Gilmour produces. Again, I was most impressed by the extra sensations the subwoofer produced. It was a definite enhancement to the musicality of the whole system.

Over the next weeks and months, I listened to many different types of

PHOTO 9: The Goldwood Sound 10″ woofer.

PHOTO 5: Amplifier mounted, and feet attached to the cabinet.

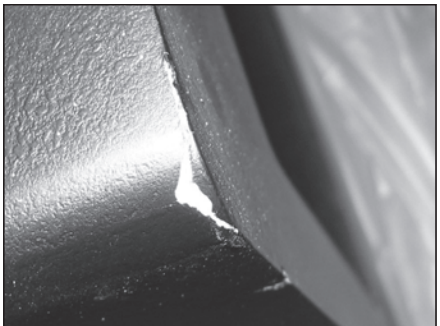

PHOTO 7: Small area of damage to the enclosure.

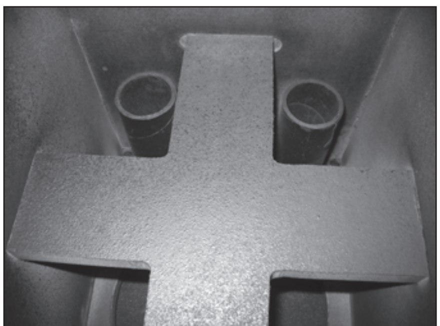

PHOTO 6: Interior of the enclosure showing the two ports and cross-shaped brace.

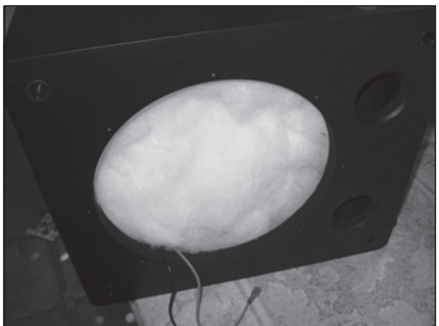

PHOTO 8: The fully stuffed cabinet, ready for mounting the woofer.

material. The most "in your face" difference is definitely experienced when watching movies; there are many very deep rumbles, atmospheric noises, and such that simply go unnoticed without the subwoofer. Explosions, of course, become moving experiences (literally), and several times while watching action/adventure films I feared for the foundations of my house, or wondered whether there was an earthquake.

In conclusion, I remain very impressed and pleased with this subwoofer. It does its job well, and simply worked well from the outset, after an easy construction procedure. My only complaint is its uninteresting shape, blackness, and size. I suppose if you want deep bass, you need to sacrifice space in your living room for something big like this.

OBJECTIVE MEASUREMENTS

For these measurements I used the Ivie Technologies IE45 Real Time Analyzer system (www.ivie.com), equipped with a Type II calibrated microphone. The setup is shown in **Photo 11**. The microphone, inserted in the IE45 jacket, was positioned about 2″ from the face of the subwoofer. The IE45 software, running on a Samsung Q1 Ultra Mobile PC, was set up in 1/12 octave RTA mode, with power averaging of the dB levels in each band of the spectrum. The IE45 software was also simultaneously instructed to generate a pink noise signal on the output jack of the PC, which was fed to the line input on the rear of the subwoofer.

Using this arrangement, I made several measurements, and the results can be seen in **Fig. 1**. The vertical scale shows dB SPL at the microphone. The curve (A) shows the response with the subwoofer's low-pass filter disabled (turned off). The curve peaks at around 100Hz with a level of 90dB or so, and falls off either side with frequency. There is still appreciable output over 1kHz. (Note that the curve is not weighted, and that the broadband dB SPL at the microphone is 101.7dB, as shown in the lower right-hand window on the display.)

The curve (B) shows the response with the low-pass filter enabled, and the frequency pot adjusted to its maximum, which is marked 100Hz. The curve (C) shows the response with the pot adjusted midway (75Hz), and the curve (D) with it adjusted to minimum (50Hz). As you can see, this pot appears to simply alter the frequency at which the rolloff begins. The rolloff appears to be about 20dB/octave. aX

PHOTO 10: The woofer mounted in the cabinet.

PHOTO 11: Measurement setup, showing the subwoofer, the Ivie Type II microphone, inserted in the IE45 jacket, and connected to an ultra mobile PC running the IE45 RTA software.

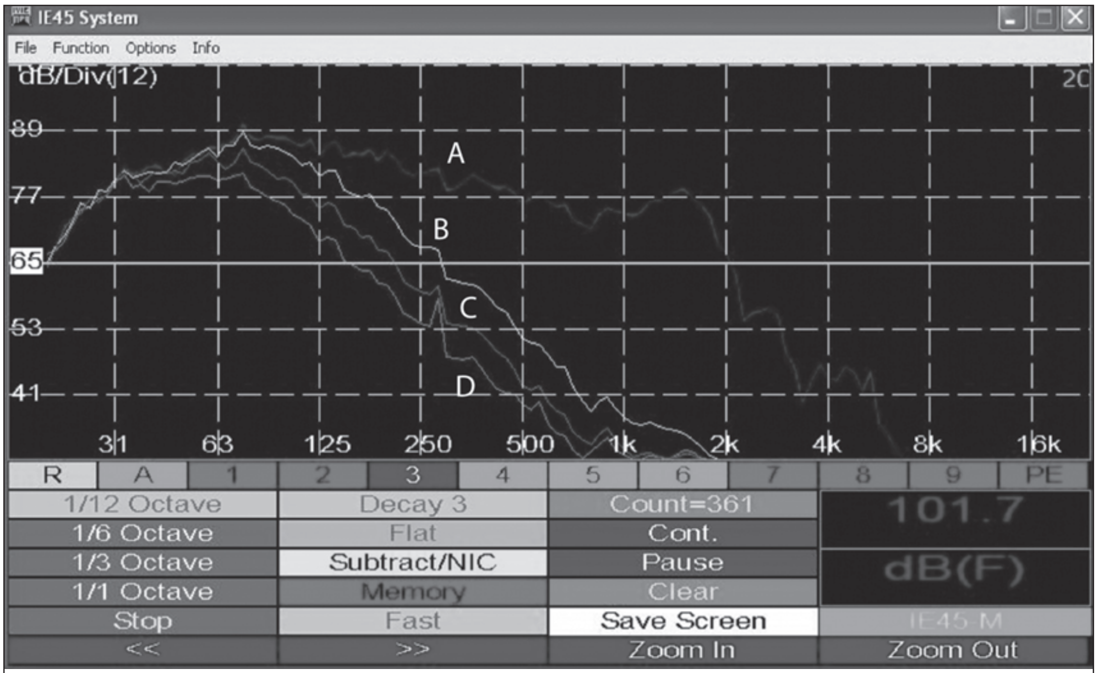

FIGURE 1: The IE45 RTA software showing measurements of the subwoofer's measured response to a pink noise signal.

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"Yard Sale" is published in each issue of *aX*. For guidelines on how subscribers can publish their free ad, see our website.

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What's New on the *aX* website?

Supplemental Information:

PC Board and Wiring layouts for Bill Christie's "A Hybrid High-Fidelity Power Amplifier," (*aX* 1/08).

Appendices for Andreas Schilloff's "A Home Theater Preamp," (*aX* 12/07).

Articles from Past Issues:

"In Praise of Wall Warts," by Paul Stamler (*aX* 1/08).

Charles Hansen's review of the Lundahl 1592 Transformer (*aX* 1/08).

Gary Galo's "Lundahl and Jensen Meet the Monarchy M24 (*aX* 1/08).

"A Mains Analyzer—How Clean Is Your Juice? Part 2," by Iain McNeill (*aX* 12/07).

"Testing Panasonic's WM-61A Mike Cartridge," by George Danavaras (*aX* 12/07).

"The "Amplibass" Control," by Dennis Colin (*aX* 12/07).

"A Mains Analyzer—How Clean Is Your Juice? Part 1," by Iain McNeill (*aX* 11/07).

"A Single-Ended Class A Hybrid Amp, Part 2," by Stephen W. Moore (*aX* 11/07).

"Intrinsic Fidelity Testing," by Dennis Colin (*aX* 11/07).

Charles Hansen's "Tube Imp Update," (*aX* 11/07).

For information on these and others, visit www.audioxpress.com. And don't forget to check out the links to our other magazines, *Voice Coil* and *Multi Media Manufacturer*!

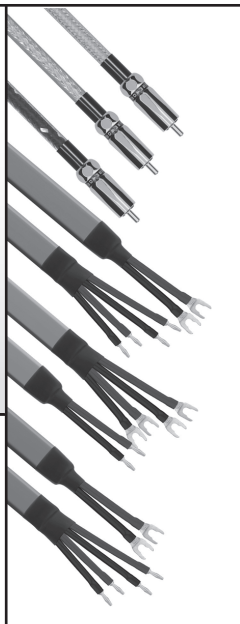
New from **Listen Inc.**, SoundMap is a Time Frequency Analysis software package, ideal for impulse response analysis and detection of loose particles in loudspeakers. Its software enables detailed analysis of signals simultaneously in both the time and frequency domain, and offers four different analysis options. For more information, check out www.listeninc.com.

Pianist/Composer Leslie Pintchik has released her second CD, "Quartets," her followup to "So Glad To Be Here." Featuring two different quartets, the CD includes members of Pintchik's working trio of herself on piano, Scott Hardy on bass, and Mark Dodge on drums. The first quartet has Satoshi Takeishi playing percussion, while the second has saxophonist Steve Wilson. The disc has five original songs and three unusual arrangements of well-known standards, and a composition by Hardy. Put out by **Ambient Records** as a multi-channel Super-Audio CD, "Quartets" features a 5.0 surround sound mix along with the standard stereo mix. For more information, please go to www.lesliepintchik.com.

New products from **MartinLogan** include the Purity and Source hybrid electrostatic loudspeakers. Purity features an internal 200W amplifier, precision performance crossover engineering, and Bass Level Control, with a horizontal dispersion of 30° and 4kΩ impedance on binding posts. The Source speaker has an 8″ aluminum cone woofer, an adjustable base design, and includes a custom wound audio transformer, and features 90dB/2.83V/meter sensitivity. To learn more, visit www.martinlogan.com.

The Alcyone, from **Cabasse** (www.cabasse.com), is a compact loudspeaker offered in a complete 5.1 system consisting of a matching Santorin 17 active subwoofer with 6.5 driver, 200W built-in amplifier and five satellites plus a full set of cables. It features peak power handling of 350W, sensitivity rating of 91dB and 8Ω impedance. The flexible Alcyone has a spherical enclosure with a perfect 60° arc of dispersion with zero phase error.

Wireworld's Equinox5[2] interconnects and speaker cables incorporate the same revolutionary Delineated Neutralizing Array technology in the company's reference audio cables. The unique geometry and structural design provide a significant improvement in audio fidelity and superior isolation from noise and interference. The Equinox single and bi-wire speaker cables feature the stacked conductor arrangement for improved flexibility, resulting in a series of 12 flat conductors stacked diagonally within a flat overall outer profile. The interconnects and speaker cables feature Ohno Continuous Cast copper conductors, and the proprietary connectors have silver-clad Oxygen-Free Copper contacts. For more, visit www.wireworldcable.com.



The Diablo Sub Panel, from **Artcoustic** (www.artcoustic.com), is a sealed cabinet featuring 10 woofers that deliver low frequency response down to 20Hz and extend up to 120Hz. It comes with the Artcoustic PA-300 professional amplifier, and is 5.75″ high and 20″ wide and deep.

New from **Enhanced Audio Inc.**, the DC Seven is a new version of the company's Diamond Cut Audio Restoration Tools. It employs an all-new technology designed to allow significant noise reduction without distortions on difficult brass or vocal passages and now includes a Narrow Crackle filter to the Filter Menu. The DC Seven also introduces artifact suppression to allow significantly more aggressive noise reduction, a new High Noise mode for the virtual valve tube simulator, an EZ Enhancer, a Virtual Phono Preamp, and an Active Music Database that allows users to access, arrange, list, and store their entire collection with ease. To learn more, go to www.enhancedaudio.com.

MAINS ANALYZER

I have researched the shunt current sensing method used in "A Mains Analyzer part 1" by Iain McNeill in 11/07 *aX*. I contacted engineers I worked with at the electric company, and at a major engineering-construction company where my brother works. While not specifically addressed in the National Electric Code (NEC), they all agree it is a bad practice to use shunts for AC current sensing on any AC line application. Only current transformers are mentioned as AC current sensors in the NEC.

My personal practice is to use current transformers to meter current flow for AC voltages above nominal 48V, or Hall-effect devices for high voltage DC. Shunts do not provide galvanic isolation from the power conductors. Small 5VA Class 1 current transformers are readily available on eBay at reasonable prices, in current ratios from 30:5 up to 200:5.

Shunts do not provide galvanic isolation from the power conductors; they are in essence bare conductors and are directly connected to external measurement equipment. This is why the power utilities use current transformers exclusively to monitor AC line currents. Aerospace electric power follows the same practice. I am particularly concerned where the proposed Mains Analyzer might be built and used by a novice unfamiliar with safety regarding the AC line. Admittedly, I have not seen how Mr. McNeill has packaged his analyzer. He may have totally precluded contact with the AC line voltage by the user.

Another issue for me is the use of the LM324 for sensing the small mV shunt current signal. Even at a line current of 20A (the maximum for a 120V AC circuit), there will only be 40mV of signal available from the 25A shunt. The offset voltage for the LM324 is ±2mV, or 5% of the maximum shunt signal.

At lower current levels the Vos will become a proportionally greater part of the signal error. The two leads from the shunts to the op amp circuits will see very high magnetic fields proportional to the load current. These leads would need to at least be twisted together in an attempt to prevent a large 60Hz error voltage from being impressed on the sense circuit, and perhaps twisted-shielded pairs would be required. Also the DC output of wall adapters WW1 and WW3 are not supplied with a voltage regulator, and the PSRR for the LM324 is only 65dB. Much better quad op amps are available.

Consider the specific failure mode effects for various failures in the design of Fig. 1 on page 10 of the *aX* article.

1. If the mains-side neutral is open, the line current shunt will be elevated to full AC line voltage through whatever load is connected to the outlet sockets. There will be no power from any of the wall warts. The line current op amp will have AC line voltage on its inputs with no ±Vcc to the circuit and the LM324 will fail (note that the use of Vdd and Vss in Figs. 2, 4, 5, and 6 is incorrect for an op amp). The situation is not as bad for an open mains ground, depending on the chassis ground configuration of the load and the amount of load imbalance on the two 120V AC sides of the 240V AC service entrance power.

   The line current shunt will be elevated to full AC line voltage through whatever load is connected to the outlet sockets. [I should have added here this would only happen with ungrounded, 2-wire power cords.] There will be no power from any of the wall warts. The line current op amp will have AC line voltage on its inputs with no ±Vcc to the circuit and the LM324 will fail.

   Applying the AC line to the inputs of the LM324 with no V+ rail DC, yet still having an inductive/capacitive path through the wall wart back from its grounded substrate back to the AC line, may cause current flow from substrate out to inputs. The 1k4 resistors may not limit the substrate current flow to 50mA and can damage the LM324 (see Notes 3 and 7 of the NSC Data Book). I have, in fact, seen this happen in the 400Hz AC systems I have worked with, where the older voltage sensing circuits used lower value wirewound resistors than we use today.
2. If R68 opens, very high voltage will be impressed on the primary of the T1 audio transformer, causing it to fail, perhaps catastrophically. There may then be a short between the two windings and anything connected to the J3 monitor output could see full line voltage. My suggestion is to use a small step-down power transformer with its primary directly connected to the 120V AC line for the monitor output.
3. With R68 or R67 open, the Glitch Detector circuit and Line Voltage DPM inputs will see full AC line voltage and will fail.

Only fireproof resistors should be used where directly exposed to AC line voltage. This includes the input resistors to the sensing circuits, due to the failure modes just described.

Power engineers are required to monitor at least to the 13th harmonic for line voltage distortion products. That would require a flat response to 780Hz for the 60Hz line, so Mr. McNeill's filter cutoffs of 340-400Hz won't give him the full power quality picture; of course, he is not running qualification on a piece of electrical equipment either. The third harmonic is generally considered to be the worst, due to electromagnetic loads such as transformers and motors (see my three-part series "AC Power Line and Audio Equipment," *audioXpress* Sept.-Nov. 2001).

As an aid to Mr. McNeill's Glitch Circuit detection settings, ANSI C62.41 1980 (IEEE Std. 587-1980) "Guide On Surge Voltages in Low-Voltage AC Circuits" calls out the time-voltage limits for AC line transients. "Low-Voltage" here refers to less than 1kV, including 120/240V AC residential AC power. I wrote a Power Line Monitor project article in the April 1998 issue of *Popular Electronics* with four distinct detectors for Outage, Sag, Surge, and Spike levels based on what was then ANSI/IEEE C6.41. It latched LED indicators after detection of any of these events. With modern microchip controllers, you could place the number of detected events into memory for later analysis.

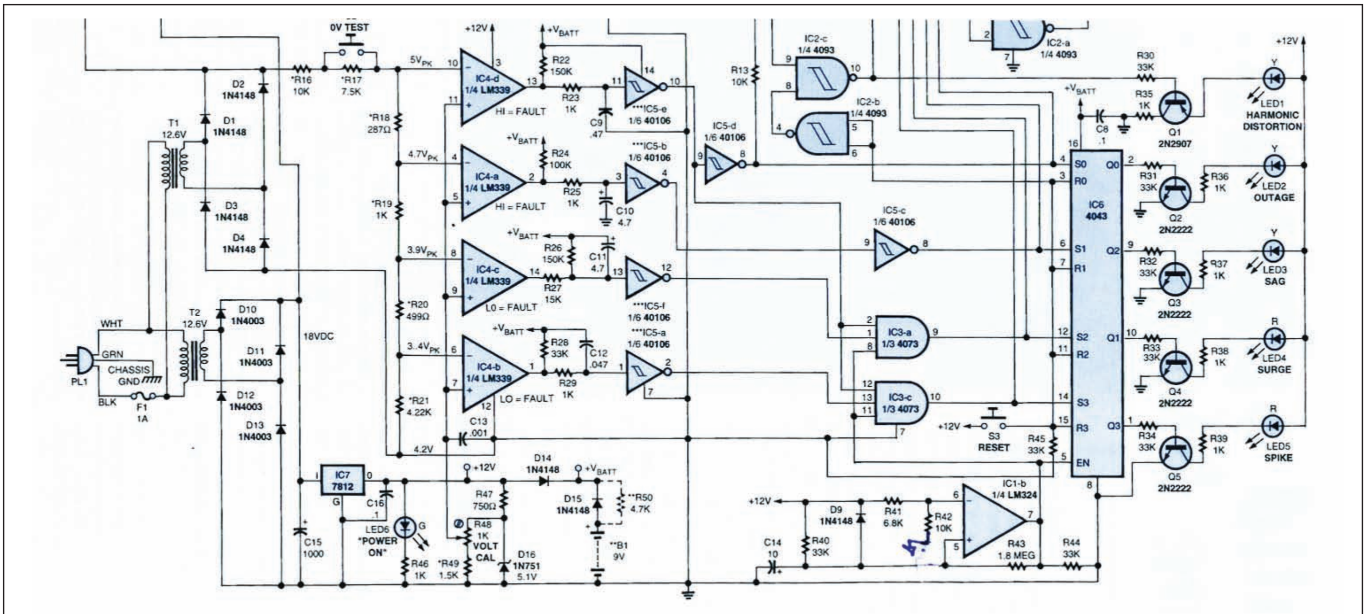


Figure 1: Voltage fault sensing circuit from *Popular Electronics*.

Figure 1 shows the voltage fault sensing circuit from the *PE* article (Schematic note: I used only "T" intersections for line connections in my supplied artwork. When *PE* re-drew the schematic they used crossing intersection dots for some of the connections. This is not permitted by ANSI/IEEE standards.) If I were to do this project today, I would include current fault sensing as well.

I hope I am not sounding too critical of Mr. McNeill's work. His basic concept is quite good. I just want to point out these few safety issues from my experience as a power engineer in both the aerospace and electric utility industries. I look forward to Part 2.

Chuck Hansen
Ocean, NJ

Iain McNeill responds:

I would like to especially thank Charles Hansen for taking the time to review and comment on my article. I have read and learned from Mr. Hansen's articles for many years, so it was quite an honor to have him comment on my work. And don't worry, you can never be too critical when it comes to safety.

Maybe I should reiterate how safety is achieved in my design. All live circuits are completely enclosed in a plastic enclosure and all exposed metal parts have galvanic isolation from the live mains circuits to withstand voltage of 1500V AC achieved through the use of air clearance and/or insulating layers. Exposed metal parts include switches, connectors, and mounting screws.

I suspect the exclusive use of current transformers in the power industry is to do with lightning surges and other fault transients on installed equipment; this device is not intended for permanent installation. I think it is important to separate safety issues from reliability issues here. I believe this device is safe, as I have ensured that there is basic and supplementary insulation from the mains circuits as is common in consumer electronics products. I didn't go to the lengths of addressing regulatory compliance or ensuring fault condition tolerance because the device was intended as a DIY project for those comfortable working with high voltage.

Checking my design notes on the LM324 offset question, I don't see this as a significant problem. The output is DC blocked in all cases and so all the offset voltage does is reduce effective headroom. For the line current meter, my calculations show worst case output DC offset at 71.43 * 0.002 = 0.143V. Supply voltage is +4.75V worst case minus 0.143V offset minus another 0.7V for output transistor Vcesat yields 4.05Vpk (2.86V RMS) worst case maximum output, which equates to a 40mV input signal equivalent to the target 20A.

I completely agree with Mr. Hansen that there are much better op amps available than the LM324, which has been around for over 25 years, as far as I remember. I didn't want to use my good (expensive) op amps on this utility application and the 324s were what I had lying around. I was actually feeling quite pleased with myself for finally clearing them out. I went back to check accuracy on the line current meter and found that the accuracy has a dome shape having good accuracy at the extremes (as you would expect for the zero/cal setup) and peaking at +5.5% at 13A as shown in **Fig. A**.

The accuracy below 2A is pretty appalling due not only to the offset problem outlined by Mr. Hansen but also due to low level signal nonlinearities in the precision rectifier. For me, this is quite adequate because I'm only looking for an indication that a circuit's capacity is being approached. For those with higher accuracy needs, substituting a better op amp will help.

I twist signal pairs together as a matter of routine and did so here also, as you recommend. There probably is some induced error in this wiring, but I was able to compensate adequately using the zero and cal pots, so I wouldn't personally call it a "large" error. The PSRR issue is moot because the LM324 is fed from a regulated supply, a fact which Mr. Hansen could not know as the complete schematics are in part 2—my apologies. I should have re-written the article somewhat after we decided to split it into two parts.

Regarding the failure modes you cite:

Open Neutral: I don't think this is a problem. As you say, the line current shunt will be pulled up to line voltage, pulling the measurement circuit and wall-wart secondary supply up with it. However, the wall wart primary is also only connected to the live conductor. There could be some small

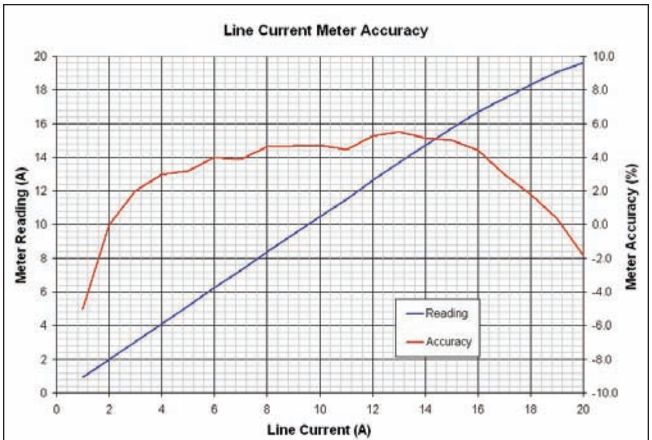


Figure A

leakage current flowing to ground through parasitic capacitance which, even if it were 100pF, would only amount to 4.5µA.

Just to be sure, I tried this with a 100W bulb plugged into the outlet and the incoming neutral conductor disconnected. As you would expect, all the displays were blank, but the red neutral-ground neon and amber live-ground neon were illuminated indicating the fault condition. When I replaced the neutral conductor everything worked as normal.

Open R68: As designed, R68 is in parallel with the transformer primary for a combined impedance of 21Ω. With R68 out of the picture we're left with just the transformer primary impedance of 37Ω. There is under 3mA of current flowing in the chain in both cases although with R68 open, all 3mA will flow through the transformer primary. The voltage across the primary will increase from 60mV to 110mV. Neither of these is particularly challenging for the audio transformer, and, with its 1500V AC withstand rating far exceeding the 205V AC worst case maximum mains voltage, I don't believe there is a safety issue here.

Open R67: You are quite right. If R67 were to fail open, 120V AC would be connected to the line voltage and glitch detector circuits, albeit current-limited by the 40kΩ resistor, R66, and would probably destroy the entire line current/voltage/glitch detector section. Having said that, it would take some pretty severe over-voltage to cause this to happen, and my calculations show that R66 would blow first. R66 reaches its 500mW power rating at 3.5mA, where the power in the 125mW R67 is only 10mW.

The use of flameproof resistors is a good idea to make this project even safer. I believe the common metal film leaded resistors today are flameproof.

Regarding monitoring to the 13th harmonic: I used an audio transformer for the monitor output for exactly this reason, and this gives me bandwidth far in excess of the 13th harmonic at 780Hz. In fact, as you can see from the spectra in Part 2, I'm seeing mains-borne noise all the way up to 10kHz. I still haven't gotten around to measuring the RF spectrum on the mains, but I suspect that there is significant noise well past the audio band.

Thanks for enclosing a copy of your mains monitor project from *Popular Electronics*; it was interesting to review and compare. I see your design as more of a long-term "sentinel" device and as such requires higher precision and consideration of reliability factors. However, my motivation in designing this project was for a quick and easy sanity check that would highlight possible problems for further investigation and as such it has been a very handy tool that I have used many times in the year since I built it.

This letter is in reference to Iain McNeill's "A Mains Analyzer—How Clean Is Your Juice? Part 1," (*aX* 11/07, p. 8). I'm highly interested in this topic of power quality and gladly welcome such a project (even though excellent, off-the-shelf solutions already exist from Dranetz and made affordable by eBay).

However, I found the article lacking for several reasons: No photos. (Which makes me wonder if this prototype was ever finished?) Safety: The author talks the talk, but I think he missed the boat. For some reason, he ignored the universal requirement for overcurrent protection. It's generally required by insurance underwriters for all line-connected equipment to have a fuse, breaker, thermistor, or thermal cutout in the primary circuit unless the circuit is inherently current-limited as in a small step-down transformer. (His wall-wart supplies would fall under this exception.) For some reason, he rules out fuses early on, but doesn't say why. (Perhaps, he was trying to use a fuse as a cheap shunt?)

I would also recommend he use current transformers instead of those 25A shunts. This would provide isolation from the power line and should allow running the digital meters directly off a single DC supply. (I know, lab-standard current transformers are quite expensive, but, you can get service-grade ones that provide ×1/×10 ranges designed to work with clamp-on ammeters quite cheaply.)

Failing that, the shunts should be placed before the neon bulb indicators, so that an open shunt would be indicated immediately. He should also move the shunt from the neutral leg to the hot leg, because an open neutral stops the flow of current, even though the tester and downstream equipment is still hot to the chassis! (Likewise, fuses and switches should only be placed in the hot leg.)

To measure line voltage safely, he could simply use a 12V step-down transformer and set his meter multiplier for 10.

Finally, to derive an isolated scope output, he could float the scope power off an isolation transformer, because I've found that the waveform becomes severely clipped if you float the signal. I'm guessing that's caused by an impedance mismatch. (It also clips if you use a voltage divider shunting the line.) But, because the power line has a characteristic impedance of 25Ω, a 12.5Ω dummy load (since we're only seeing half of the 240V pole-mounted transformer) would dissipate 1100W continuously and piss off Al Gore tremendously.

Jerry Nutter
audio.carp@verizon.net

Iain McNeill responds:

I'd like to thank Mr. Nutter for his interest in my article. I suspect many of his questions have been answered by part 2 of my article. I designed and built the analyzer about a year ago and have been using it regularly since then in many applications. My need for such a device predates that, and, as I mentioned in the article, I performed a fairly extensive search on the Internet before deciding to "roll my own."

On your recommendation, I looked at the Dranetz products and couldn't find one that was as simple to use as mine or had the same features. All the Dranetz products appear to be designed for site monitoring in a permanent installation. I wanted something that would give me a reading in minutes without the use of tools. However, repeating my search today, just to see whether the landscape had changed, I did find a product that includes most of the features of my analyzer from Extech (their part number 380803) retailing for around $700.

With regard to safety, I spent much time thinking through the various failure scenarios before starting the design and originally intended to fuse the device. However, as I explain in the article, my concern was that I couldn't guarantee that the mains circuit I was connecting into was correctly wired. I have encountered swapped live and neutral conductors in several residences where the homeowner has incorrectly replaced outlets, and if a fuse in the (assumed) live conductor failed, due to power-on surge, for example, then any equipment plugged into my analyzer could conceivably float up to line voltage. The logical solution is to use a breaker that would interrupt both live and neutral in the event of overcurrent, but these are large and expensive and there is always one of these in the customer service panel.

As long as the mains circuit in the analyzer is designed to greatly exceed the 20A maximum residential circuit current, everything is safe; the analyzer works much as an extension cord in this sense. I would also point out that the load part of the analyzer, as you mention, is inherently current limited. This includes the neon indicators, GFCB test circuit, and voltage sense resistor divider. Anything plugged into the analyzer will have its own overcurrent protection.

The use of current transformers is an equally valid choice in my mind and would allow the use of a pair of wall warts (the DPMs still need a floating supply). It so happened that I had a huge pile of wall warts available and the cost and design simplicity of shunts was more attractive. As with many consumer electronics products today, isolation is provided by the use of a plastic enclosure.

My decision to place the neon indicators upstream of the current sense was so that their currents, admittedly small, would not be included in the load current measurement. I would argue that an open shunt would be indicated by 0V on the voltmeter display, but now we're splitting hairs; I firmly believe that the service panel breaker will trip before these 25A shunts fail open.

Your comments regarding the correct placement of current sense shunts, fuses, and switches are valid as long as you can guarantee a correctly wired outlet; this is not the case in my experience. One of the first tests I performed on this unit was to verify the analyzer operated correctly under all miswiring conditions including live-neutral swap and neutral-ground swap. I am glad to say I have never encountered a live-ground swap, but the analyzer will safely alert you to this lethal condition as long as you plug the analyzer into the wall outlet first before plugging equipment into the analyzer.

The problem with using a stepdown transformer for the voltage sense functions is that they are optimized for 60Hz operation. (This is also why I used an audio transformer for the monitor output.) My concern was that a simple 12V step-down transformer wouldn't preserve the voltage ratios of the harmonics to the 60Hz fundamental.

For the glitch detector to provide a valid indication, I needed a wide bandwidth sense voltage, and resistors are a cost-effective way to achieve this. Once again, safety isolation is provided by the enclosure.

Finally, I don't think floating the scope power off an isolation transformer is an acceptable solution to the isolation problem. If I remember correctly, this was a practice used back in the 50s and 60s for working on televisions with hot chassis. Fortunately, this design technique is no longer used, so floating test equipment off the mains supply is not necessary and the integrity of safety ground can be maintained throughout.

My implementation, using a wideband audio transformer to provide an isolated, single-ended monitor output, works just fine as you can see from the figures in part 2. I suspect the clipping you have seen may be due to ground loops and/or leakage currents conducting in input clamping devices, but without examining the actual conditions I can only guess. Impedance matching is only important where you need to control reflections, and while this may be important in the solution of the high frequency mains-borne harmonics problem, it is, as you point out, highly inappropriate for 60Hz mains power delivery.

Thanks again for allowing me to discuss my design decisions, and I hope you enjoyed the final part of the article.

GOOD-LOOKIN' HORN

Concerning Ed Simon's Combination Horn from January '07 (p. 10), I am wondering whether we could have more dialogue about new developments with the speaker system. It appeared to have many positive attributes, although Ed found some shortcomings. I would like to further explore this system, which has a higher WAF than my current hi-fi setup. His other articles have been equally impressive and useful for our hobby.

Kevin Tuttle
Durant, Okla.

MIDRANGE ID

I would like a clarification concerning project # 10, Mr. Bones (*Speaker Building 201*, by Ray Alden, p. 133). In the general description of this project and in table 11.9 as well, it is stated that the Morel MW113 midwoofer is being used; however, in the description of the midrange section of the crossover and also in the accompanying schematic, it seems that the Morel MW114S is the one that's being used. Which is the driver that has actually been used?

Dimitris Katramados
Athens, Greece

Ray Alden responds:
I would like to thank Mr. Katramados for pointing out this discrepancy. To immediately answer his question, it is the Morel MW113 that is being used, not the MW114S. The MW114S is mentioned in error in the crossover schematic on the bottom right of p. 134. I suspect that, because graphic designers had to draw this diagram independent of my manuscript, this error crept in and went undetected during my final reading.

It can be a great disappointment to authors who provide designs of complete speaker systems in their books that parts, in particular drivers, go out of production so soon after publication. Personally, I tried to choose drivers that had a long shelf life, and happily many in *Speaker Building 201* are still available. Morel, a company founded in 1975, focused on using advanced materials such as rare earth magnets and hexagonal-shaped wires in their voice coils. In fact, the MW114-S midrange is the immediate replacement for the MW113; it used a neodymium magnet system instead of the double-magnet ferrite used in the MW113. If you wish to read a detailed PDF spec sheet for the MW133 go to: www.madisound.com/catalog/PDF/morel/mw113.pdf?osCsid=0412d63acb55ada0160a06190bef627f

If you wish to purchase the MW113 midrange for use in this system, I have discovered that Madisound still has 13 available; they can be found on the Internet at: www.madisound.com/catalog/product_info.php?manufacturers_id=141&products_id=597

"Mr. Bones," so named because of its more skeleton-like construction, is a very fine system that I hope more builders will come to enjoy.

CIRCUIT CONNECTIONS

I read Alex Arion's "Greek Baroque System" (*aX* 12/07) with interest. I have been a professional electronic circuit designer for more than 55 years, going way back to the tube days and continuing into the transistor and IC days. I am still active at it.

I looked at his circuit diagram (p. 52) and something caught my eye. I know that in a long-tailed pair of triodes connected as phase inverters, the undriven triode (the common cathode part of the pair) has somewhat lower gain than the driven triode. To compensate for that, you usually use a somewhat larger plate-load resistor in that stage than in the driven stage.

In his diagram, this is not the case at all. The plate load of the driven side is much larger, more than double the value of the undriven side. This must be wrong; otherwise, the output-stage will be totally unbalanced AC wise.

Then I noticed that the long-tail resistor R26 is connected to the output transformer secondary, in some form of feedback configuration. This cannot be a correct connection, because the bottom of the long tail is a nonlinear input to the amplifier. You want negative feedback to a linear input to the amplifier, not a nonlinear input.

There is an attempt at negative feedback by connecting V3B grid back to the input, but because of the long-tail connection this feedback is to a large extent cancelled. Not only that, but because the author needed to decrease the plate loads on the wrong side of the long-tailed pair to balance the drive to the output amplifier, the feedback must effectively be positive rather than negative feedback.

The result of overall positive feedback in an amplifier is that the gain is increased. This is often OK, but harmonic distortion is increased as well. The frequency response of the amplifier becomes worse as well. In addition, the output of the amplifier as seen by the speaker load increases as well. With a pentode-connected output stage, the output impedance is pretty high to start with, and by adding positive feedback it becomes even higher.

You can only imagine what this does to the overall performance of the amplifier-speaker connection, but you know one thing—that this amplifier would not care about what kind of speaker cables you would use. Any speaker-cable impedance would be totally swamped out by the huge output impedance of the amplifier. It has become a current source rather than a voltage source, as required by most loudspeaker loads.

Next oversight is present in the preamp circuit on p. 54. R14 and R15 on Fig. 2 serve no purpose whatsoever, and R14 should be removed, while R15 should be replaced with a short. The circuit will work better that way, and will be circuit-wise more correct.

My last objection is to the schematic for the power supply, which shows D2 as being a 215V zener diode. That must be a misprint for a 21.5V zener diode. The regulator cannot possibly work with a 215V zener diode. According to my calculation, the output voltage wants to be more than 5000V using a 215V zener diode. That is not possible with the components shown on the schematic. Also, the filament connections of the rectifier tubes U11 and U12 are shown incorrectly; they should obviously not be shorted together at the tube-base as shown on the drawing.

Hans J Weedon
Salem, Mass.

Alex Arion responds:

Thank you for your letter. Unfortunately for me, you are right in all your observations regarding the "Baroque Amp." So, in the power stage (p. 52) R31 is 100Ω, not 100kΩ! In this case, all the problems you indicated about the nonsymmetry and the negative feedback will disappear! Of course, the OTR's secondaries should not be shorted to ground, but only the downer side. The upper side (8Ω) should go to the speakers.

About the preamp (p. 54), here I wanted to divide the DC (direct coupling) to not use the cathode follower before its lifetime. Of course, it will work OK with your simplifications. Now, in your last objection (the power supply), of course you are right that the zener diode is 21.5V, and the filament connections for the two rectifiers are shorted! Mea culpa, all the mistakes are mine, because I'm not good at computer graphics, but I will improve myself everyday!

Once again, I thank you for your competent observations that only a man belonging to our generation, having huge experience in tube construction, could make.

Alex Arion's "A Greek Baroque System" is one beautiful piece of art, and I can only imagine that it is a beautiful-sounding system. Many thanks for sharing it with the world!

Jon Geissinger
Elliottsburg, Penn.

CORRECTION

Rickard Berglund's article, "The Many Uses of a Hybrid Van Scoyoc Circuit" (Oct. '07), has a major problem—the circuit as shown won't work. Instead of cross-coupling, the schematic has each feed from the source followers going to the grid and cathode of the same side. The signals will largely cancel out. I suspect a drawing error.

A version which I modestly think is a bit more elegant uses p-channel JFETs to accomplish the same task. For a bit of extra gain, the biasing resistors can be bypassed. I've built versions of this circuit using a wide variety of tubes; because the AC balance of a cross-coupled inverter is equal to (mu + 1)/mu, a high mu tube (e.g., 6SL7) is preferred. FETs can be selected for best DC balance—they're cheap.

Stuart Yaniger
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Rickard Berglund responds:

Yes, there is an error in the cross-coupling section. The idea of using p-channel FETs is very elegant. **aX**

CONTRIBUTORS

Ari Polisois ("A Double-Ended Two-Channel Amp," p. 6) resides in France. He previously wrote "High Power SE 6C33C Amp," in the July 2004 issue of *audioXpress*.

Claudio Negro ("Driver Induced Vibrations," p. 16) is a retired commercial airplane pilot, and has been interested in speaker building since the age of 13 when he built a zero offset three-way speaker with a self made tweeter acoustic lens. He also became interested in electronics, building a mixer based on a Radford preamp and a Marantz MM pre-schematics, with separate PSU enclosure. In 1985 he was classified sixth in an Italian magazine DIY speaker contest, after which he took a long break from DIY audio. His other interests are listening and playing hard rock music, photography, computers, and online car racing.

Paul Loewenstein ("Dynamic Microphone Soundcard Amplifier," p. 22) has a BA in 1971 from the University of Cambridge, MA in 1974, and Ph. D in 1994. His early career at Schlumberger in the UK and France involved analog systems from high-frequency IC amplifiers, vibrations of thin shells, and transients on high-voltage power lines. His professional activity has been purely digital since his move to the US in 1981, working for Schlumberger, National Semiconductor, Mitsubishi, and Sun Microsystems.

Hiroshi Obama ("7W Power Amp with Cathode NFB," p. 26) was born in 1943 in Tokyo, Japan. After having mastered economics, he joined the airline industry, where he met several colleagues who were audio enthusiasts and taught him to design and build amps. He enjoys restoring old audio equipment as well as building his own amps, including solid-state units.

Are Waagbø ("Shunt or Not," p. 30) resides in Norway. This is his first article for *audioXpress*.

David J. Weinberg ("AES Multi-Tracks in New York 2007, pt. 1," p. 38) is an engineering consultant and technical journalist on audio, video, and film technology. He provides audio and home theater engineering consultation and professional on-location digital audio recording services to companies, radio stations, and individuals. He brings to his work an MSEE, a First Class Radiotelephone license, and over 40 years of continued study and active involvement in the audio, video, and computer industries. He is Vice Chairman of the Audio Engineering Society's DC section, a manager in the Society of Motion Picture and Television Engineers' DC section, and membership officer of the Boston Audio Society (www.BostonAudioSociety.org).

Julian Bunn (Review: Speaker City's PS10 Subwoofer Kit, p. 42) is a member of the Professional Staff at the California Institute of Technology (Caltech), where he is engaged in researching computing solutions for the next generation of CERN's particle physics experiments, investigating contaminant dust particle motion from a future Mars lander, and simulating the motion of the Basilar membrane in the human cochlea. In the early 90s he wrote the AIRR and WinAIRR applications for loudspeaker measurement, which are sold by audioXpress. Recently he formed his own company, Bofinit Corporation (www.bofinit.com), which specializes in development of audio applications for the Pocket PC. In this context, he formed a strong relationship with Ivie Technologies (www.ivie.com), where he holds the position of Senior Scientist. He is the main software architect of Ivie's IE33 audio measurement device. In his spare time he has recently built a 70s vintage Minisonic 2 synthesizer, a SE tube amplifier, and a four digit Nixie tube clock. Julian has a lovely wife, Sarah, and three beautiful daughters, Jessica, Georgina, and Isabella.