SPEAKER MODS

Nice article by Ed Simon ("A Combination Horn You Can Build," Jan. '07, p. 10). His description of the bass properties of the horn sounds remarkable.

Is it possible to eliminate the midrange-supporting section of the speaker and optimize the horn for reproduction of bass frequencies only? What would that look like?

Since the woofer would not be used for midrange frequencies in this application, how about configuring it so that the woofer fires inward, and the output of the horn is facing the listener instead of facing into the corner?

Angel Rivera
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Ed Simon responds:

Yes, you can use this as a bass only speaker. Do not cut out for the woofer in the front of the box. Mount the woofer in the filter board that forms the top of the Z section. Extend this board 2″ in length to connect to the front of the box. The rear chamber should now be sealed.

You can now decide whether you want to make a bass reflex loading on the rear chamber of the woofer by putting a port tube in the front. I would do this near the bottom of the chamber on the face of the loudspeaker. If you are placing a small speaker on top of the woofer, you may wish to place the port high near the full-range unit.

If you want to try this inexpensively, make the test enclosure out of several pieces of corrugated cardboard glued up to about 1″ thickness. Any old 8″ speaker should work. The lighter and stiffer the cone, the better for this application. I would not use a speaker designed to be a small box woofer because they tend to have a lower output and the heavier cone does not match as well to the horn load.

If you try this, let us know how it goes.

LPS TO DVDS

First, I would like to compliment Victor Staggs on his excellent article, which was both thorough and informative ("Transferring LPs to DVDs in High Resolution," Feb. '07, p. 14).

The reason I'm writing is because I'm also planning a similar project, and I have some additional information that may be of use. (Note that I am planning to do this on a PC.)

I would strongly suggest that unless you plan to follow Victor's excellent instructions exactly—using similar hardware and software—you become familiar with the process of producing and recording DVDs in general before making decisions about DVD formats.

Converting analog to digital. The entire part of the process involved in converting LPs into a hi-res digital format is the same regardless of your final choice of medium. The digital files will be immediately playable on a computer—or may be output to an outboard DAC. You will need to be sure that your computer card and DAC have compatible connections—TOSLINK or coax.

You should also be aware that DACs vary in the formats they support; many older DACs may not support 96/24 digital or higher, and some newer ones may still not support 192/24. You *can* do this part before even deciding what DVD format to use.

DVD audio formats. There are two entirely different choices available for putting audio on a DVD: DVD-Audio and DVD. The method that Victor uses is considered DVD (he is producing a standard video DVD with hi-res PCM audio and blank video). You should be able to play Victor's method on most DVD players—although some may limit their digital output format in terms of resolution (some players simply will not output 192/24 audio from their digital ports—and some may not be willing to do 96/24).

There may even be players out there that do not support playing 96/24 PCM audio from a DVD—not all players support all formats in the standard. The main problem here is that this is often simply not specified in the player's spec sheet. DVD players are cheap—this just means that you might need to search for one that does what you need.

You may also have a problem with software—a lot of DVD recording software (especially low-cost programs and shareware for the PC) simply fails to specify what formats it supports. They say "put audio here"—you drop it in and see whether the software accepts it or not. Worse, some software will automatically down-sample the audio without telling you. The settings may be somewhere in the options or you may need to call product support for an answer.

I suggest making sure that the software you choose accepts using hi-res audio before experimenting (and that it will not generate an error if the video stream is blank). The software that Victor used actually produces the DVD content and outputs it as a video file—as I understand it.

Depending on the software you choose, some PC software will allow you to separately specify audio and video content—in which case you may be able to simply not specify a video file (the step of producing the combined audio/video file is carried out internally).

The other alternative choice is to use DVD-Audio, which is a different standard and somewhat more difficult. To play it you specifically need a DVD-Audio compatible player. There is a DVD-Audio copy-protection scheme that limits the digital output resolution of tagged commercial disks (or blocks it entirely); this is

not an issue on disks you record. In addition, however, many players do not play the bits from the DVD straight out the digital output (this is called "bit transparent" when it is mentioned). Another issue is that there is very little software that supports recording in true DVD-Audio format—and most of it is expensive.

One benefit to DVD-Audio is that it supports a type of lossless compression that can double the capacity of a disk—the downside is that the codec is licensed by one company and costs about $2500. Odds are that Victor's choice is easier to implement and more universally playable. The quality should be similar, but there are "audiophile quality DVD-Audio players" out there that may be better than combined DVD units.

DVD players. There are just a few facts about DVD players in general that you should know:

- some DVD players will not play DVD-recordable media (the ones that do cost no more than those which do not, just be sure you have one that does)
- DVD-recordable media is divided into DVD-R ("dash R" or "minus R") and DVD+R ("plus R"); they are similar and most computer drives support both, but your recorder and player must support the one you choose or it won't work (some players support only one or the other)
- some DVD players do not support hi-res digital output or even have a digital output (be sure yours does if you plan to use an outboard DAC)
- most DVD players have a coax (S/PDIF) digital output and/or an optical (TOSLINK) output; some DACs have only one or the other input; make sure both of yours support a common format if you plan to use an outboard DAC
- many DVD player TOSLINK outputs do not support 192/24 digital format (apparently much of the standard TOSLINK hardware tops out at 96/24); on some of these machines the S/PDIF output is also limited to 96/24; on some it supports 192/24

I hate to make it seem so complicated, but this is one of those subjects that becomes very convoluted once you "get off the path."

One final note: Some DVD software has the option of producing a disk that is immediately playable. This usually involves a menu option and should result in a disk that plays as soon as you put it into the drive—so there is no need to push any buttons (or remember what to push without a screen connected).

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TRANSISTOR NOISE

Some questions are raised in the February issue of *aX* about transistor noise ("Xpress Mail," p. 60). If the letter writers will send me—via *aX*—a few of each type of their transistors in question, I will measure them and report back on the results via *aX*. I use a Quan-Tech Labs Transistor Noise Analyzer.

I have found the noise specs to be wildly different in some cases from the spec sheets. Some with high noise specs are very quiet and vice versa. It appears that as factory processes "improve" over the years, the effects/changes on the spec sheets don't change from the original. This seems more likely with transistor numbers that have been around for a long time.

The bottom line is that they need to be checked/tested. Then you find out the noise with varying DC currents and voltage to find optimum noise value conditions.

Richard Marsh
Cool, Calif.

CF FOLLOW-UP

I wish to thank Christopher Paul for his in-depth analysis of the relative performance of the cathode through mu follower to SRPP ("The Cathode Follower and Its Weaker Siblings," Jan. '07, p. 22). The final comment made in the article disturbs me somewhat, because, although the cathode follower (CF) may demonstrate the "gold standard" for the measurements undertaken, we have no basis to assume that it may in any way sound superior.

An FFT harmonic distribution profile—either as measured or from a Spice simulation—would point the way more clearly as to the advisability, or not, of discarding the SRPP.

Wayne L. Wright
New Zealand

Christopher Paul responds:

I thank Mr. Wright for his interest in my article and acknowledge his concerns about how the circuits sound. I do, however, question his certitude that it offered no guidance at all on this topic. Surely it's clear which stage would suffer the most sonically when powered from a less than clean supply or driving a potentially bandwidth-limiting capacitive load.

I have generally steered clear of addressing questions of what sounds superior in my articles. I believe that this topic is complex and requires more work to address authoritatively than I care to attempt. Mr. Wright's assertion that there is a correlation between an FFT spectrum and sonic quality is debatable, unless we're discussing gross distortion. There is much literature supporting a number of different views on this topic. Which should we adopt?

Suppose that we were to accept a specific correlation, no mean assumption. To generalize about the relative distortions, it would be necessary to test a wide variety of valves as SRPPs, CFs, and MFs over a range of bias conditions while driving a number of different loads, capacitive and not. And you would need to consider the balance of distortion characteristics from tube to tube in the pair of SRPP triodes, because that configuration tends to subtract its triodes' even (2nd order) distortion as it averages its (3rd order) odds. As I said, a lot of work.

What may be instructive, however, is to simulate the AC grid-to-cathode voltages of my article's SRPP and CF at common output levels. I think it's fair to say that these voltage levels will indicate the relative amounts of distortion that each triode is generating. I'll assign mu = 30 and gm = 0.012 mhos (siemens). With equal output voltages across a load of 100K, I get SRPPbottom/CF = 1.07 and SRPPtop/CF = 0.74. For a 10K load, I get SRPPbottom/CF = 0.93 and SRPPtop/CF = 0.75.

The first thing you see from these results is that there are no significant differences in the drive levels of the three triodes. The second thing is that the average of the SRPP levels is slightly less than the drive of the CF. Because the SRPP averages the odds but subtracts the evens, I'd expect from the SRPP less 2nd order (the total amount depends on the balance in the distortion characteristics of the two triodes) and only slightly less 3rd order harmonics. You can be the judge as to whether these small differences mean anything.

In summary, considering the small differences expected in distortion and the big differences in Gain, Zout, and PSRR, I'll stick by my article's limited sonic implications listed here, and especially to the conclusion that if you choose to use the SRPP, you'll now know what you're losing.

TRANSFORMER MAGIC

I think Bob McIntyre did a great job on his 25L6 "Mighty Mouse" amplifier, which appeared in April '06 (p. 34). I'm amazed that Bob got great results using power transformers as audio output transformers. In spite of the great sound, I'm sure that there are those who don't feel comfortable substituting audio output transformers with power transformers, and have shied away from building the amp. If you wanted to build the "Mighty Mouse," and use audio output transformers, what impedance would you use?

Because the power transformers Bob uses are 240V AC CT:12V as connected, the primary-secondary ratio is 20:1. If squared, and multiplied by the load impedance connected to the secondary, 8Ω, as is the formula for determining primary impedance, the result is $20^2 \times 8$, or 3200Ω C.T. In looking through the Antique Electronic Supply catalog, there is one such transformer available as a replacement in Marshall guitar amplifiers, but it's rather large, maybe too much so, for the "Mighty Mouse" chassis.

A second option is to use the Hammond P-T1609 (10,000Ω CT-:4-8-16Ω) and connect the UL leads to the plates instead of the blue and brown wires which should have their ends taped, or soldered to an empty terminal lug. The red wire would be connected to B1 (150V DC) as usual. Also, the 8Ω speaker would be connected to the P-T1609's 4Ω tap, instead of the 8Ω tap. This configuration provides the 20:1 ratio for 3200Ω C.T.:8Ω.

The P-T1609, like all Hammond 1600 series output transformers, has its UL taps at 40% of the primary. To determine the UL-to-UL impedance, multiply the primary value by 0.16. If, for example, the primary impedance is 10,000Ω CT, the UL-to-UL impedance is 1600Ω CT, not 4000Ω (40% of 10,000), as you might believe. To understand why this is, you need to diagram a 10,000Ω CT:4-8-16 transformer with 40% UL taps (located toward the center tap) as I have done in **Fig. 1**, based on the necessary turns ratio.

The ratio of the transfomer is found using the formula

$$\sqrt{\frac{Z_{pri}}{Z_{sec}}} = \sqrt{\frac{10000}{16}} = 25(:1)$$

The 25:1 ratio tells you that the primary will contain 25 times as many turns as the secondary. The example in **Fig. 1** has 180, therefore the primary has 4500. The center

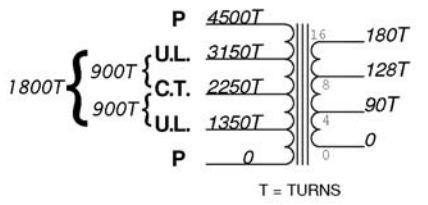

FIGURE 1: Winding diagram for 10,000kΩ CT:4-8-16Ω with 40% UL taps.

tap is located at 2250 turns, and the UL taps at 1350 and 3150. The UL winding, 40%, consists of 1800 turns, 900 turns each side of the center tap.

The secondary, which again has 180 turns, is tapped at two locations. Because the 4Ω tap is located in the middle of the 16Ω secondary, it will be found at the 90 turns point.

To find the 8Ω location, you must find the ratio of 10,000Ω:8Ω by using the earlier mentioned formula

$$\sqrt{\frac{10000}{8}} = 35.36$$

Next, divide the number of primary turns by this result

4500 ÷ 35.36 = 127.26 (128)

The 8Ω point is at 128 turns—not 135 turns, the halfway point between the 4Ω and 16Ω taps, as some might assume.

Now knowing that the diagram in **Fig. 1** is correct, you can determine the UL-to-UL value. Simply divide the number of turns in the UL winding, 1800, by the 180 turns in the secondary, to get 10, or 10:1. To find the UL-to-UL impedance, square this result and multiply it by 16

$10^2 \times 16 = 100 \times 16 = 1600\Omega$ CT.

The UL-to-UL value is 1600Ω CT, or 16% of the primary value. Because all Hammond 1600 series transformers have 40% UL taps, you can determine their UL-to-UL value by multiplying the primary impedance by 0.16 as well.

By knowing their UL-to-UL impedance, you can use Hammond 1600 series transformers in many more applications.

Neal A. Haight
Castro Valley, CA

LISTENING TESTS

Why doesn't someone write an article for readers who don't possess a lot of bench test equipment (or no test equipment at all)? Then, other readers could suggest recorded source material—not just special test CDs—to use for a really comprehensive listening test to "flex the muscle" of their system?

I have no test equipment; however, there are certain recordings I always go back to so that I can tell whether I picked up a quality component or, whether the ones I've rebuilt are at their optimal performance.

As much as I feel like a kindred spirit to Dennis Colin and Charles Hansen, the "Rube Goldberg"-type technical minutiae doesn't communicate the "worthwhile immediacy" any audiophile felt (or not) if they'd been sitting right in front of those Scott tube amps. The principle I go by is that an audio component—regardless of price—will never match the sound of live music (even, probably with a $37,000 Audio Note Ongaku monoblock amp). That's not its job. But what it should do is extract every ounce of signal from the source it can in the most distortionless and transparent way.

I prefer to do my listening tests with old, analog reel-to-reel tapes. (I was handed down a library of them by my father—all still in good condition, too.) I believe (be it my hangup or be it true) that these represent the closest thing to a "time travel back to" vintage studio performance (because they don't amass the kind of surface degradation vinyl records are susceptible to).

The deck I use is a (rebuilt) AKAI 1730-SS, two speed (7½ and 3.75 ips) quadraphonic recorder from 1973. (It originally sold for $349.) I've replaced the following components:

- all the capacitors on the playback and record preamplifiers as well as on the power supply
- the TO-220-type voltage regulator with a NTE-brand substitute
- the power cord with 12AWG wire and a heavy duty Hubbell plug (this had the most dramatic effect on the consistency of the motor's reaction)
- the heads with later AKAI glass and crystal composition heads (as opposed to earlier alloy and ceramic cast heads) from another machine I'd taken apart. I now know this is

not a weak link in my system.

I've also replaced the following parts on my vintage Marantz 1060 (from 1971):

- all the coupling and filtering caps on the preamplifier
- the driver-stage transistors on the power amp
- the bridge rectifier with a new, 6A/200PRV
- the DPDT/TV-5 power switch with a newer Alps replacement. (I pried open the old power switch after removing it, and couldn't believe all the oxidation which had crept in.)

When the budget allows, I also plan to upgrade the output stage filtering caps (from Nichicon, VZ-Series general purpose) to really "elite" Cornell Dubilier, "MLP" Series Flatpack 50-year-life caps. (Digi-Key now offers these, and the ones I need are the 80V DC, 3300µF rating. . .at $87.43 each!) Are they worth it? With all I've already done, I need only turn the volume up to the 8:45 position to hear it at full blast. This amp is supposed to deliver 60W per channel, and even with famously insufficient vintage Acoustic Research model 2ax's (1968), I must be getting well over 50W per channel with no problem.

As for the recordings I'd like to mention:

1. *Arthur Fiedler: Superstar*. Boston Pops. Polydor 5008, 1970. Listen to the grand spectacle of their arrangement of the overture to Andrew Lloyd Webber's *Jesus Christ Superstar*. Also, an instrumental version of the 1969 country hit *Gentle on My Mind*, in which you should be able to clearly delineate the rhythm of the second horn section trombones from the glaring French horns in the first horn section. Even with just a two-channel recording, you should hear the echoic reverb of the percussion (due to the album having been made inside of Symphony Hall) on this track.

2. *Glory Of Love*. Herbie Mann, A&M 3001, 1967. A lot of jazz purists will dismiss this album as being "kitschy, lounge lizard pre-Fusion"; however, in one track, *Oh, How I Want to Love You* (composed by Mann himself), you should reproduce the kick and thump of Grady Tate's drums without any muddiness. Also, the extended guitar solo (by Eric Gale, whom I don't know whether or not he was famous at the time) should sound as though it's a Gibson flailing right in front of you. (The only way I can describe this song—which is an instrumental—is that it's a mix of a tango, the Pied Piper, and B.B. King style guitar thrown all together! People really took chances in the Sixties!)

3. *Revolver*, The Beatles, Capitol 2481, 1966. I think this is the best album the Beatles made (they were too imitative of other people before this point and became too gimmicky afterwards). On *Here, There, and Everywhere*, the soundstage should clearly indicate the place of John and George's harmonizing (as well as finger-snapping) behind Paul's lead vocal. On *She Said She Said*, listen to how John and George each try to synchronize the lead and rhythm guitar parts on their respective Rickenbackers. On *Good Day Sunshine*, you should clearly pick up Ringo Starr mumbling something (I think it's "she does") after the first line of (I believe) the third verse, when Paul sings "She looks good."

Thank you for catering to this soulful hobby.

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CONTRIBUTORS

Ed Simon ("Sound Strobe Review," p. 6 and "Speech Intelligibility, Part 2," p. 34) received his B.S.E.E. at Carnegie-Mellon University, and has installed over 500 sound systems at venues including Jacob's Field, Cleveland, Ohio.

Walt Jung ("Sources 101, Audio Current Regulator Tests for High Performance, Part 1: Basics of Operation," p. 10) is a noted analog designer and author of the classic *IC Op-Amp Cookbook*.

William Eckle ("Building a Center Channel for an Altec A7-800," p. 24) is featured on the website www.wmeckle.com.

Dennis Colin ("A Low-Noise Measurement Preamp" p. 26) has demonstrated the audibility of phase distortion at Boston Audio Society, and has designed the "Omni-Focus" speaker (bipolar coincidental with phase-linear first-order crossover), 1kW biamp and PWM supply at A/D/S.

Rudy Godmaire ("A Flexible Subwoofer Amplifier, Pt. 4," p. 40), a sales consultant for Bell Canada, has been interested in DIY audio since 1998.

Darcy Staggs ("DeNoising A Vanilla Preamp," p. 46) is a retired engineer. His work experience began with the Apollo project, then defense analysis, writing management software at IBM in Stockholm Sweden, and later more defense analysis back in the US. His later years as a consulting engineer involved product design, machine design, business development, and any other fascinating challenge that came in the front door.

Rickard Berglund ("Audio Aid: Build This Six-Transistor Low-Noise MC Preamp," p. 48) has worked in research in analytical chemistry, and in industrial electronics design. He is now engaged in construction and manufacturing of tube amplifiers.