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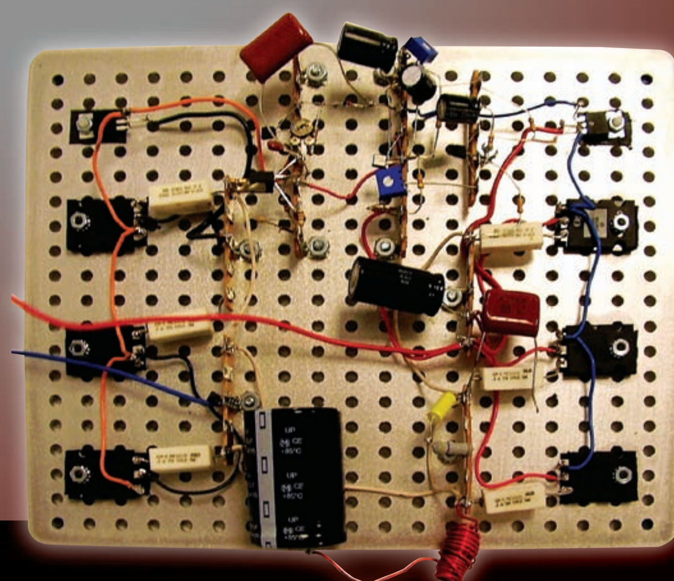

Simple ways to Test Your Circuits

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Magazines: A Primer

We are often asked why *audioXpress* is so expensive. Compared to what? Well, *Stereophile* for one. The simple answer is that the old maxims of supply and demand apply. Buy few and the cost is higher, buy a lot and you get a quantity break. The last time I looked, our friends at *Stereophile* were about four times the size of our list of subscribers and newsstand readers. And you can subscribe to *Stereophile* for about one-third the amount you pay for *aX*.

Delivery of the larger magazines is now cheaper per copy. The Postal Service, founded to expedite information in aid of an informed citizenry of a participatory republic, is now a fully capitalist enterprise working to avoid losing money. It is cheaper per copy to deliver a big mag for a lot of technical reasons I won't bore you with.

Today all magazines are paying more for delivery across the board. Oil is one large reason, raising the cost of everything which goes into producing, printing, and delivering the product. Why not publish electronically? Because, so far, subscribers prefer "a hold in the two hands object."

Advertisers prefer big magazines. Quantity is something advertising agencies understand, especially with technical magazines whose content they often seem not to understand. If it is bigger, the hit rate is likely to be bigger.

Promotional budgets of large mags are large. When *Stereophile* was first purchased from J. Gordon Holt, the price was low, but the buyer owned a Mercedes agency and was obviously not poor. Unfortunately, I have never owned a Mercedes agency. I could only afford a mailing of 100,000 to launch *Audio Amateur* in 1970, which netted about 5,000 responses. *Stereophile* was relaunched after purchase with a huge effort and rose steadily to over 80,000.

We spend all we can regularly to promote *aX* these days, sometimes with a lot of help from other vendors, sometimes not. But we have never been very big. Big enough to provide a living for those of us who produce our magazines, but never a fat bottom line.

Of course, these two periodicals are selling different content. *Stereophile* readers are mostly treated to a vicarious pleasure of a dream of owning stratospherically expensive audio gear which only the fabulously rich can actually purchase. In some ways the prices tell the buyer something about how important he or she is, making normally priced equipment emotionally unacceptable. But what fun to dream about playing with this month's expensive toys.

audioXpress and its several predecessors offers readers a way to take audio in hand. We like to know how the equipment works, what makes it better, and believe that if it is technically better it is likely to be more satisfying to listen to. We also love to be able to build or modify speakers, amplifiers, preamps, and almost anything having to do with reproducing sound. We like to do it ourselves—or look over the shoulders of others who do it, which is our form of vicarious experience.

You have doubtless noticed that *audioXpress* looks as though it has been on a rigorous diet. Not so. The old law of supply and demand still works in publishing like everywhere else. Profitability requires a balance of 60% editorial to 40% advertising. When ad sales slip, we must cut editorial content. Sadly, we have a rich cupboard well-stocked with great audio projects. Authors are producing excellent manuscripts in amazing quantities. Alas, they will be published much more slowly unless things change.

Advertisers like traffic to their websites or their phones. This gives them the sense that their messages are getting through to readers and the feeling their money is well spent. They feel even better if readers find their offerings so appealing that they order merchandise. If you spend no time with the ads in each issue, you are jeapordizing the health of your publication. Our advertisers are an unusual band often providing unique products and services. If you want *audioXpress* to thicken, spend some time getting to know our advertisers.—E.T.D.

Upgrading a Classic **Pilot Amp**

This amp construction project is vintage all the way—with old iron and coke-bottle tubes.

In the December 2005 issue of *audioXpress* ("Junk Box Tube Amp Jubilee," p. 24), I wrote an article on constructing amps based on the Pilot 410 monoblock. As I stated in that article, this circuit lends itself to the use of many different types of tubes, output transformers, and power transformers. My friend Larry and I have built many of these types of amps, which have not only worked upon first being fired up, but have also sounded very nice and been extremely stable. I encouraged readers to dig around in their parts boxes and build one or two. Apparently, many of you did, because I have received many e-mails and notes from people who truly enjoyed the construction. I have built a number of them for some of my friends, who have enjoyed them as well.

PHOTO 1: The completed amplifier.

VINTAGE COMPONENTS

Many of the ones we built used transformers from the late 50s and early 60s, as well as some new production ones from Hammond. I decided I wanted to build a very vintage unit (**Photo 1**), such as was constructed in the 40s and 50s before there was readily available amateur audio equipment at a reasonable price. I would use only octal tubes, very vintage transformers, carbon composition resistors, and period wire and chassis.

I decided to use 6L6Gs for the right look (nothing like coke bottle tubes for a vintage look). Digging around, I found a very high-quality Thordarson CHT output transformer, which was big, weighed about 8 lbs, and had both 8000 center tapped (ct) and 5000 ct primaries rated at 15W with 70mA current per plate. The connections for both primary and secondaries are on the bottom of the case, but the back featured an impedance selection with a pin plug for 2, 3, 4, 6, 8, 16, 250, and 500Ω. I was able to date it in a reproduction catalog from the late 40s. The only problem was I only had one of them and I wanted to build a stereo amplifier. I put a search online and waited.

About 1½ years later I got a hit that led me to a very similar unit. The difference was this one had the ID tag on the top of the can instead of on the front. It was also missing the bottom mount plate, but it was the same size and the rest of it appeared identical. I purchased the unit, which was identical to the first, except for the noted differences. I can only assume that they were produced in different years, but with the same model designation.

I was able to make a suitable mount plate and now had a pair of transformers—I was on my way! I wanted to run about 295V on the plates, so I dug around and found a power transformer with 500 ct primary at 225mA, with two 6.3V 4A and 5V 3A filament windings. While I have no idea of the date of manufacture of this unit, I painted the top with wrinkle paint to match the factory finish on the outputs.

After a little more digging, I found a steel chassis from an old rack-mount telephone amplifier that used 4 and 5 pin tubes. It was very heavy steel and had 6 coupling transformers; "cat gut" wrapped wiring harness with cloth-covered wire, and some very nice oil caps under the chassis. After stripping the excess nonusable parts and sockets, I started laying out the chassis.

When you use a salvaged chassis with holes already punched and drilled in it, you must be creative. I arranged and rearranged the sockets and transformers until I covered the unwanted holes, while using the pre-punched socket holes in a pleasing arrangement. After drilling and punching the additional needed holes, I sent the chassis to the powder coat shop for a finish of gray and black antique. (This used to be called "hammered" finish, but the powder coaters refer to it as antique vein). I touched up the outputs with satin black (they were already wrinkle-finished).

I used a 5V3 coke bottle as a rectifier tube, and 6SN7 dual triodes for the preamp and phase inverters. I looked for a coke bottle version of the 6SN7, but the only ones I found were European versions (CV181), and the ones I have run across were generally over $100 each. I recently found a pair of coke-bottle-shaped CV1285s, which I will substitute for the 6SN7 (for significantly less than $100 each).

I wanted to continue my vintage theme by using as many older parts as possible, but I did use new gold-plated input jacks, newer banana output jacks, and a new volume control.

I decided to use the original oil caps in the power supply (**Photo 2**), as well as their original mounting hardware and spacers. I used the original cloth-covered wire salvaged from the telephone amp for all the wiring. Internally, I used all NOS carbon composition 1 and 2W resistors for their vintage look and to stay with my theme (I know, I have scoffed at using these old beasts and ridiculed people who swear by them).

The only newer-style components I used were the coupling caps and electrolytic caps. I planned on using some salvaged coupling caps, but after I sorted through about 50 I had saved, and found not one "on value" and only a few that were not leaky, I abandoned the idea. I chose to use the newer version of these parts because of the unreliability of older wax coupling caps and old electrolytics. I would have used some oil-impregnated coupling caps, but I couldn't find any that weren't worth a fortune to their sellers. I plan to replace the newer coupling caps with some older Sprague oil caps if I find some that are reasonably priced.

As you can see from **Photo 3**, there is plenty of room to work under the chassis. I used a bus wire to "float" all the grounds from the chassis, except where the input jacks are connected to chassis ground and the bus wire. This is to prevent ground loops. I was very surprised when I fired up this amp and it had a hum!

COMPLETED AMPLIFIER

Because I used the bus wire system (**Photo 4**), I assumed the noise problem was something else and proceeded to swap preamp tubes, change and add filter caps, and even rerout filament wiring, all to no avail. When I asked my friend Larry (I have mentioned him before—we share a tube amp addiction) to take a look at it, he found that I had not attached the ground terminal of the volume control to the bus wire. As soon as he added it, the amp became dead quiet.

I think many of us face the same issues; we sneak in a little time to work on projects whenever we can, and sometimes just miss something. When you spend a lot of time on a project, it is almost impossible to find a simple error. It is very nice to have a friend such as Larry, who can look over your work and spot your errors quickly. Or, sometimes if you set the project aside for a while and then go back to it, you may notice your mistake.

I have not done any distortion or power measurements on this amp yet, but I was struck by how great it sounds! Larry and I both agree that it sounds the best of any of the other 12-15 of these amps we have built using the same circuit.

SOUND FACTORS

The question arises as to why it sounds so great. Is it the "old iron" outputs?

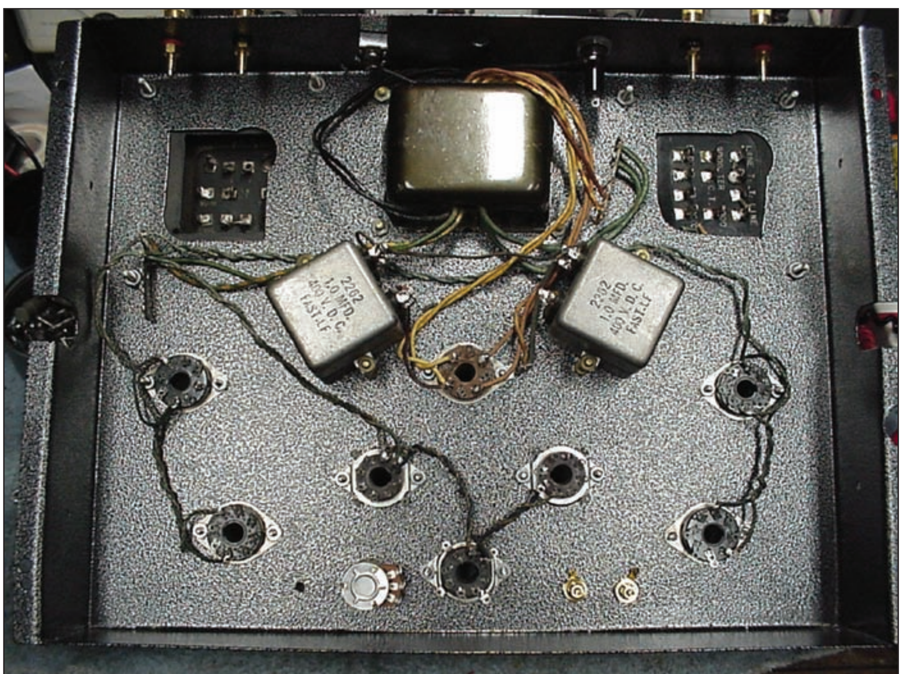

PHOTO 2: Tube wiring.

Might be, although we have used vintage iron in most of the amps we have built. We have used Sherwood, Fisher 400 and 500, Heathkit A4 and A5, and Knightkit transformers, as well as some unknowns. The Thordarson transformers are huge and heavy, especially for their 15W rating. They are every bit as heavy as Dynaco Mark III/IV outputs rated at 60W.

One of the things I think we can all agree on is that these Thordarsons were made when quality of construction was a prime consideration in US products. Many of the others we have used were built at the same time, and we agree that they all sound different, but none sound as good as this one.

Is it the coke bottle outputs? This is pretty easy to check, and I have tried various 6L6 tubes in the amp. Honestly, I haven't been able to detect a difference with the 6L6s. Larry disagrees with me on this; he says they sound different, but not enough to account for the kind of difference we hear in this amp. To be fair, I have found some significant difference in the sound of NOS output tubes versus Chinese and Russian imports, but I think the reliability of the NOS is better than anything made today.

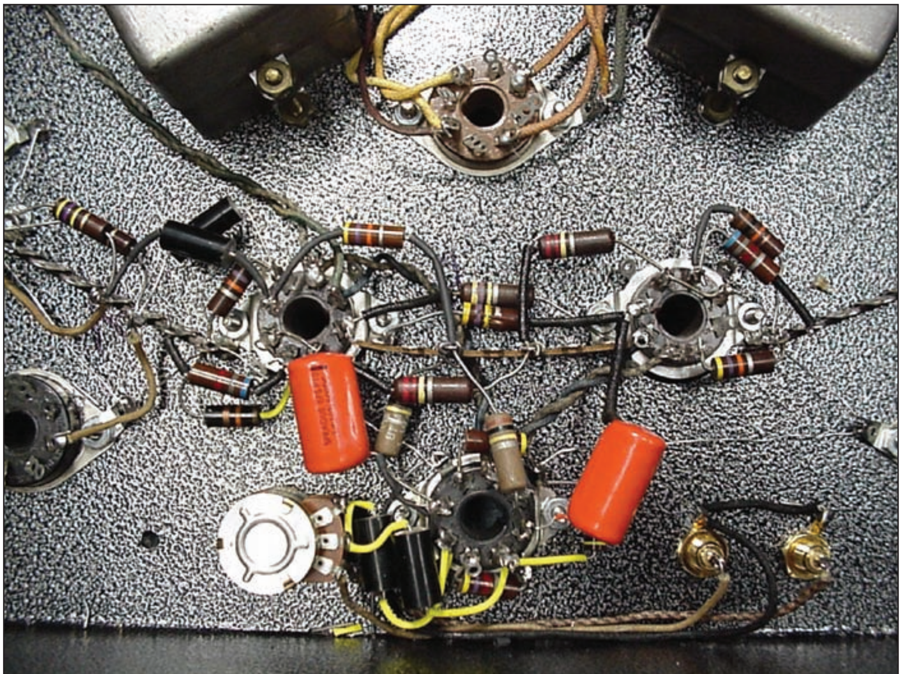

PHOTO 3: Grounding the amp.

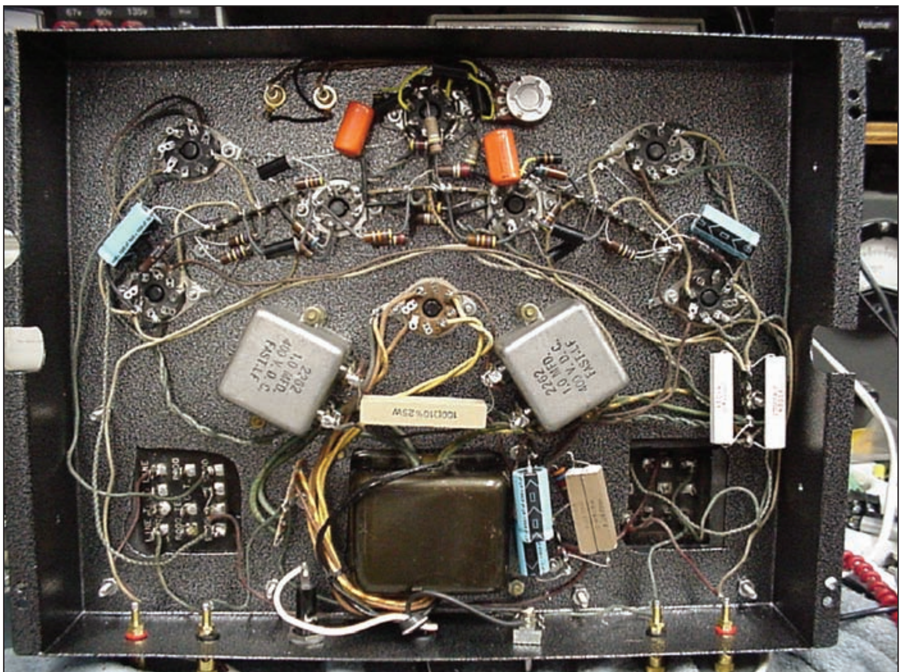

PHOTO 4: All hooked up.

Is it the 6SN7 input and phase inverter tubes? Over the years, quite a number of very well-known and intelligent audio people have stated that octal preamp tubes are better sounding than 7- or 9-pin tubes. I am not sure about this one either. The other amps we have built used a variety of input and phase inverter tubes including 6C4s, 12AU7s, as well as 6SN7s. I suppose I could build some socket adapters and try 7- and 9-pin tubes, but that would require much more effort than I am willing to expend. I do believe that you can hear a huge difference in preamp tubes!

If you want to experiment with this, try a half-dozen various 12AX7s (or whatever common ones you use) in your favorite preamp. After a pretty short exposure to various tubes, you can detect some differences. Noise levels and microphonics are the easiest to hear, but the treble smoothness is also generally apparent after some critical listening. I have found that some of the newer production tubes are every bit as good as NOS ones, so I would not spend a bunch of money on smooth plate Telefunkens. (I couldn't hear the difference in these and other NOS quality tubes.)

What about the tube rectifier and oil caps in the power supply? This is the first of my amps that used a coke bottle rectifier tube and oil caps. The stereo version in the original article used silicon diode voltage doublers and it sounds very good, not much different from this one, but maybe not quite as smooth. We have used a variety of tube and diode rectifiers in the 410 clones, and even though they all sound a little different, I am not sure why.

Many people believe the oil caps are superior to electrolytic caps in power supplies, and I think it may make a difference in single-ended amps, but I am not sure that this is true here. I do have the oil caps bypassed with a 100mF Sprague electrolytic because of the hum issue I mentioned. After Larry found my wiring error, I neglected to remove the electrolytic cap. It might just be the oil caps. The only way this is going to be solved is to take them out of the circuit. Not really sure I want to do that.

Could it be the carbon composition resistors? I am sure that I will hear a lot on this one! I have criticized carbon comp resistors for their troublesome noise characteristics. I have tried to avoid them in preamp circuits, and have used them only in restoration work, where the appearance of the components is as important to collectors as the function of the unit. Noise from them is an issue, but in the case of this amp there is no noise, even with tapping on the various resistors (a technique used in troubleshooting noise).

In this amp I used a variety of NOS and used carbon comp resistors, but discarded any used ones that were off value more than 5%. This question must remain unanswered, because I have no intention of substituting all the resistors in this amp to find out.

Last but not least, is it the old cloth-covered wire from the 40s? I think this one is very easy. I have never been able to hear a difference in the "sound" of wire. I have tried silver wire, Teflon coated wire, and heavy wire. All with the same results—no difference in sound. I generally use quality—but sometimes salvaged—wire and prefer to use solid wire over stranded, but have yet to be convinced that there is a difference in sound.

The same goes for speaker wire. I use Monster™ wire or heavy zip cord (for short runs with small speakers) and haven't seen the need for spending extraordinary amounts of money for speaker wire.

Many years ago Frank Van Alstine challenged high-end cable manufacturers to send him their cables for a scientific evaluation. Only one manufacturer responded. After measuring them, he said they were nice, satisfactory, but not superior. Frank believes that if the amplifier is designed correctly there should be no difference in sound. I tend to agree with him. He thinks that the capacitance of some cables could unbalance the output circuits in solid-state amplifiers, especially poorly designed ones. I think this is one of those personal choices.

At any rate, I don't think the hook-up wire makes the difference. Now, if someone wants to send me some $1000/ft cables to evaluate, I would be happy to try them. But don't send $10K cables, because I can't afford the insurance rate increase.

So, after considering all these factors, I conclude that it might be the combination of all these things acting together. I think the combination of extremely good iron, vintage good tubes, overall high-quality NOS USA components, and, of course, good construction techniques have resulted in a very superior amplifier. The bass is solid without boominess, the midrange is liquid, and the treble is extremely smooth.

This amplifier has become my "reference" unit in my shop, and I find myself listening to it frequently, rather than working on a project.

I welcome your thoughts and ideas. aX

By Ethan Winer

Everything You Wanted to Know About **SPL Meters**

How to use this handy device to calibrate loudspeaker levels and measure frequency response in a room.

SPL meters are intended mainly to measure absolute sound volume levels, but they have other uses as well. Sound waves are rapid changes in air pressure, to which our ears respond. Therefore, SPL stands for *Sound Pressure Level*, though you'll sometimes see them called Sound Level Meters (SLM).

In this article, I look at a number of practical applications for this seemingly simple device. I use the Radio Shack meter shown in **Photo 1** for my examples because it's inexpensive and commonly available. Note that SPL meters are "reference" devices, so they must be calibrated at the factory using known absolute volume levels. The Radio Shack meter claims accuracy within 2dB. You can also have an SPL meter calibrated by an independent lab, but that's usually reserved for the more expensive meters used by professional acousticians.

PHOTO 1: The popular SPL meter from Radio Shack costs about $50.

In truth, SPL meters are quite simple—a built-in microphone and preamplifier are connected to a readout that displays the absolute sound level in decibels. Most SPL meters also have a range control to accommodate a wide range of sound levels. For example, the Radio Shack meter can report levels between 50 and 126dB, or a span of 76dB. This might not seem large on the surface, but decibels are a logarithmic ratio, so the actual span is nearly 10,000 to 1!

—Some common applications of SPL meters used as originally intended to measure absolute volume levels are:

- To be sure your rock band's outdoor gig doesn't exceed the maximum volume set by local ordinance.
- By law enforcement for the same reason.
- By highway engineers to assess pavement surface noise, or when designing barriers to shield nearby houses.
- In a recording studio to ensure consistent levels from one mixing session to another.
- As a reality check for how loud you're listening to avoid hearing damage.
- To brag to your friends about how loud your system can play!

An SPL meter is also useful for assessing relative levels, for example, to calibrate your 5.1 surround system to ensure that all speakers play at consistent volume levels. Calibration products, such as the *DVD Essentials* test DVD that I use, include signals that play through each speaker one at a time. This way you can adjust your receiver for identical levels from each speaker, and also adjust your subwoofer's volume relative to the other five speakers.

You simply place the SPL meter where your head would be while listening, then play that section of the DVD. As each speaker sounds the test signal, you make a note of the level reported by the SPL meter. Then you can use your receiver's controls to adjust the volume for each speaker, so they're all the same. Unlike the directional microphones commonly used by singers and radio announcers, SPL meters contain an omnidirectional microphone, so it doesn't matter which way the meter is pointed.

THE ANALOG VERSUS DIGITAL DEBATE

No, not *that* debate! At various times, Radio Shack has offered two meter types—analog and digital—which refers only to the display. I prefer the digital version for one important reason: When used at the slow setting to average sound levels that vary rapidly over time, it's *much* easier to read the digital display because it is steadier.

When using an SPL meter to calibrate loudspeaker levels, you can use either static sine waves or pink noise as the test signal. Sine-wave test signals for level adjustment are usually 1kHz, which is the center of the midrange. In this case the SPL meter will display a single number that corresponds to the volume received by its microphone. Because the tone's level is constant, there's no need to average readings over time.

However, using sine waves is not a good idea for matching loudspeaker levels because they cause standing waves in the room. Standing waves produce peaks and deep nulls in the frequency response that are highly positional. If you play a 1kHz sine wave and note the level on the meter, then move the meter only a few inches, the level will likely be very different. Room acoustic treatment—especially absorption at the first reflection points—reduces the change in volume at different nearby locations but doesn't avoid it completely.

Therefore, a much better signal source for matching loudspeaker levels is pink noise, which contains multiple frequencies. If the SPL meter's microphone happens to be in a null location for 1kHz, a nearby frequency, such as 1.1kHz or 912Hz, will not be in the same physical null. Therefore, the big advantage of pink noise is its inherent averaging of volume level versus frequency. Taken as a whole, the measured volume will be fairly accurate.

Pink noise contains all frequencies and sounds like tape hiss. Unlike white noise, which contains equal energy at every frequency, pink noise is filtered to have less treble. Technically, the level falls at a rate of 3dB per octave, so each octave contains the same energy level as all other octaves, rather than the same energy for each Hz of bandwidth. Pink noise is less irritating to listen to than white noise, and you can also play a pink noise test signal much louder without risking damage to your tweeters. The *DVD Essentials* DVD takes this one step further and filters the pink noise to contain only midrange frequencies. The bass response in most domestic-size rooms varies wildly with position—much more than in the midrange—so filtering the noise ensures more consistent and reliable readings when used for matching loudspeaker levels.

While pink noise is better overall for loudspeaker level matching than individual sine waves, there is one drawback: Because noise is by definition random, the volume constantly fluctuates both overall and at each frequency. So when viewed on a conventional signal meter, such as the VU meter in a cassette deck, the needle dances around making it difficult to read. The variance is typically several dB, though it can be 8dB or even more at very low frequencies. So it might display 80dB for a moment, then 77dB, then 84dB, and so forth.

To get the true picture, you need to watch the meter carefully for ten seconds or even longer and mentally average all the numbers. And this is why I prefer Radio Shack's digital SPL meter over the older analog model. The digital meter does a better job of averaging the variations over time.

ROOM ACOUSTIC MEASUREMENTS

Most people have no idea how large an influence their listening rooms have on the overall frequency response of their systems. Numerous peak/null spans of 20dB or even larger are not only common but typical, especially below 300Hz. It astounds me that people will obsess over frequency response errors of less than 1dB in electronic gear, yet totally ignore a room response such as the one shown in **Fig. 1**, which, again, is absolutely typical. Note the peak/dip pair at 110 and 122Hz, where the response varies a staggering 32dB across a range smaller than one musical whole step.

There are two basic ways to measure the acoustic properties in a room. The simple method plays test signals at different frequencies, which you monitor on the SPL meter and draw as dots on semi-logarithmic graph paper. The most accurate way to measure the raw frequency response of your speakers and room is to use individual sine waves at 1Hz increments for 300Hz and below, and pink noise in third octave bands above 300Hz. You can buy test tone CDs having sine waves at each of the 31 standard third octave frequencies, but they are not useful in my opinion.

For example, the room response in **Fig. 1** has a peak at 110Hz and a very deep null at 122Hz. But the standard third octave test frequencies are 100Hz and 125Hz. So measuring only at 100Hz and 125Hz completely hides the real response.

Furthermore, as I explained earlier, sine wave levels are highly positional at all frequencies, so such test CDs are not useful at mid and high frequencies. Other commercial test CDs contain pink noise filtered into each standard third octave band. While that works fine at mid and high frequencies, third octave averaging lacks sufficient resolution at bass frequencies to see the true response. This is why I recommend sine waves at 1Hz intervals for low frequencies, and third octave filtered pink noise above 300Hz. You can download a collection of low-frequency MP3 files to produce your own bass-range test tone CD for free from my company's website: www.realtraps.com/test-cd.htm.

MODAL RINGING DECAY TIMES

So far I've considered only the frequency response of loudspeakers and the room they're in, as measured with an SPL meter. However, assessing room acoustics properly also requires measuring in the time domain. Many people use their SPL meters mainly as microphones to feed a computer sound card and software for room measurements. Because the Radio Shack meter has a line-level output, you can plug it directly into the line input of a sound card without needing a separate microphone preamp.

There are many popular and affordable programs available for measuring room acoustics. The advantage of software—versus using simple test signals and an SPL meter—is the ability to measure not only frequency response but also modal ringing, individual reflections, and reverb time. Small rooms don't really have true reverberation such as you'll find in an auditorium or gymnasium, but the same metrics are often used. In small rooms the reflections produce a series of individual echoes that decay fairly rapidly, versus true reverb that first swells over time and then decays.

I use the ETF and R+D software for Windows that costs $150 for both from www.etfacoustic.com. For Mac users, I recommend FuzzMeasure, avail-

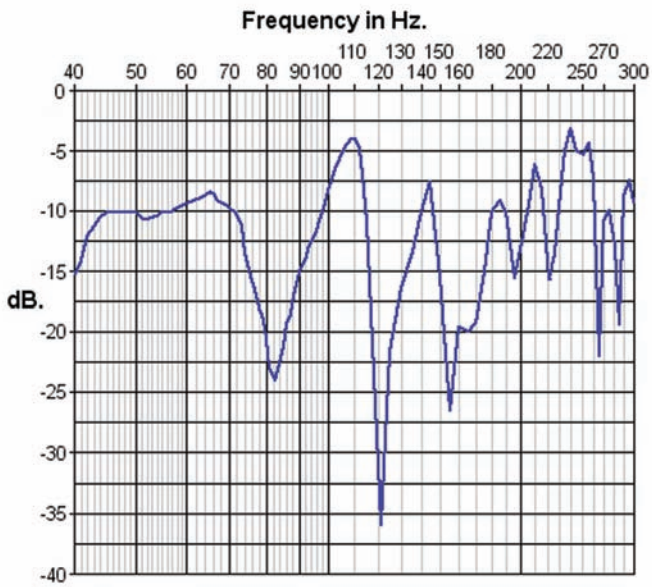

FIGURE 1: This frequency response was measured in a typical untreated listening room 16′ long by 10′ wide.

able for $150 from www.supermegaultragroovy.com. There's also an excellent freeware program called Room EQ Wizard (REW) that runs on both Windows and Mac OS, available at www.hometheatershack.com/roomeq. Although REW is intended mainly for adjusting an equalizer to compensate for poor bass response (not recommended, as explained below), it's a fairly full-featured room analysis program in its own right.

Unlike the simple response graph in **Fig. 1**, room analysis programs can also display modal ringing using what's called a *waterfall* plot. Ringing is a big problem in all small rooms, and it causes some, but not all, bass notes to linger even after the bass player stops the note from sounding. This is similar to the "boing" sound you get when you clap your hands in an empty room or stairwell, but it happens at low frequencies that can't be tested with handclaps. Ringing is caused by room modes, which are resonances that occur at frequencies related to the room's dimensions. **Figures 2a** and **2b** show a pair of ETF graphs that I measured in the RealTraps test lab when completely empty except for one loudspeaker, the measuring microphone, and computer. This room is about 16 by 11 by 8′ high, and you can see that the response is just as terrible—though absolutely typical—as the response graph from the other room shown in **Fig. 1**.

As you can see, **Fig. 2b** shows the same frequency response as **Fig. 2a**, but it also shows the decay time of each mode frequency. In this type of graph, the "mountains" come forward over time. Modal ringing is the main cause of the problem commonly known as "one note bass," in which all bass notes sound more or less the same, regardless of their actual pitch. With a waterfall graph, you can clearly see that each peak in the response is accompanied by ringing that decays slowly over time.

Ringing is a huge problem because it causes some bass notes to sustain and overlap subsequent notes. This results in indistinct and muddy-sounding bass, because no matter which note is played, the room's ringing sounds at the same dominant frequency. And because the ringing tones also linger, there's more total energy in the room at those frequencies. Using simple test signals and an SPL meter measures only the frequency response and ignores the ringing. Likewise, using handclaps to test a room produces mostly midrange frequencies, so that won't reveal modal ringing problems either.

MICROPHONE ACCURACY

It's worth mentioning that inexpensive SPL meters use inexpensive microphones that are not as accurate as professional microphones. Fortunately, Radio Shack's meter is fairly accurate at bass frequencies, though above 1kHz its accuracy is much worse. I use a precision DPA 4090 microphone for room testing; but at about $700, it costs much more than the Radio Shack meter. However, even if the microphone in an inexpensive meter is not absolutely accurate, it is still useful to assess relative changes, for example, to see how the response and ringing in your room improve as bass traps and other

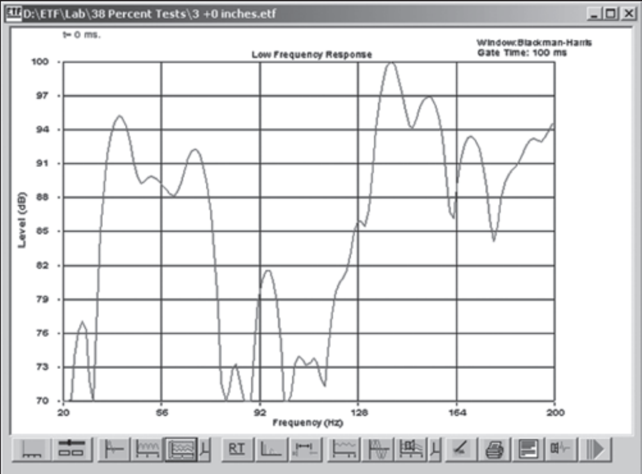

FIGURE 2A: The raw low-frequency response measured in the RealTraps test lab.

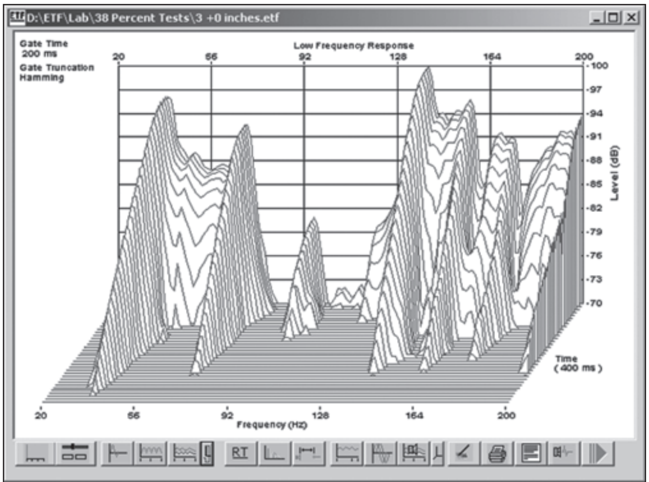

FIGURE 2B: A waterfall plot derived from the same measurement shown in *Fig. 2A*.

treatment are added.

Figures 3a and **3b** show a comparison of my Radio Shack SPL meter and a precision-calibrated AKG microphone, both at low and mid/high frequencies. As you can see, the Radio Shack meter's response is within 1dB of the much more expensive microphone up to about 800Hz, then deviates substantially at higher frequencies. You'll find "calibration" curves on the Internet that claim to compensate for the Radio Shack meter's inaccuracy, but these are mostly worthless if well-intentioned.

As you can see in **Fig. 3a**, the Radio Shack meter's response is fine at bass frequencies, so no correction is needed. But above 1kHz it's unlikely that these meters are consistent from unit to unit, anyway, for various reasons. So there's no assurance that someone else's meter has the same frequency response as yours. Furthermore, measuring a microphone's frequency response is not a trivial task—I can't imagine that most people who post correction curves are skilled enough or have the tools to do that correctly.

NUTS AND BOLTS

The only knob you need to adjust on a Radio Shack SPL meter—when used as a microphone front end for software room analysis—is the range. Set your computer's software mixer for maximum input level, then use the range switch to control the actual signal level going to the computer sound card. For other applications it helps to understand what the various knobs and switches do. There's no need to duplicate the owner's manual here, so I'll mention these features briefly just for the benefit of those who do not (yet) own SPL meters.

The weighting switch selects one of two frequency response curves the meter uses to bias the display. Our ears hear midrange frequencies more readily than extreme bass or treble, so when the goal is to assess *perceived* volume—how loud something actually sounds—the A weighting curve is preferred. With A weighting, sound containing mainly midrange frequencies displays a higher value than very low or very high frequencies of the same physical SPL level. But when measuring the frequency response of a loudspeaker or room, you use C weighting, because in that case you want the true response. Note that the Radio Shack meter always powers up with C weighting engaged.

The response button selects fast or slow meter response times. For most uses, it doesn't really matter which speed you select because the speed affects only the bar-graph portion of the display. The actual numbers never change so quickly that you can't read them. The default for response speed is fast.

The Radio Shack SPL meter also has buttons that let you tell it to remember the loudest, softest, and average levels it measures over time. This way you can leave the meter unattended for a while, then come back later to note the results.

CORRECTING ROOM ACOUSTICS

Because so many people use SPL meters to measure room acoustics, I'll conclude by explaining why you should *not* try to equalize a system for a flat frequency response. It may seem that you could use an equalizer to reduce peaks and raise nulls to obtain a flat response from your loudspeakers and room. But in practice it simply does not work, and in most cases using EQ results in a worse sound, not better. One big problem is that peaks and nulls are highly positional. So any correction that you apply will be valid

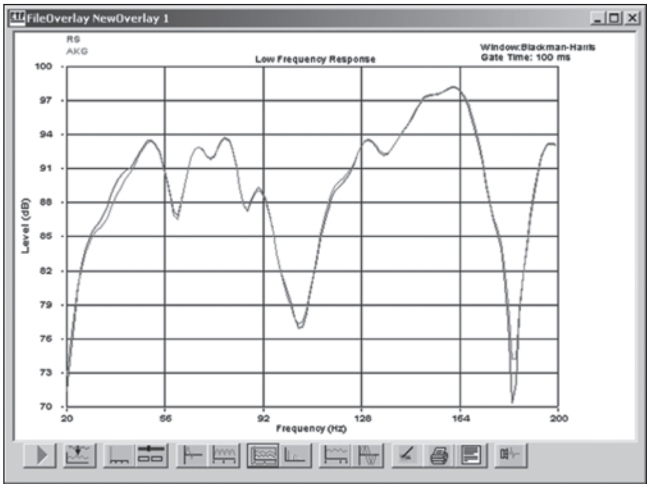

FIGURE 3A: This graph compares the response at low frequencies of my Radio Shack SPL meter to an expensive calibrated AKG microphone.

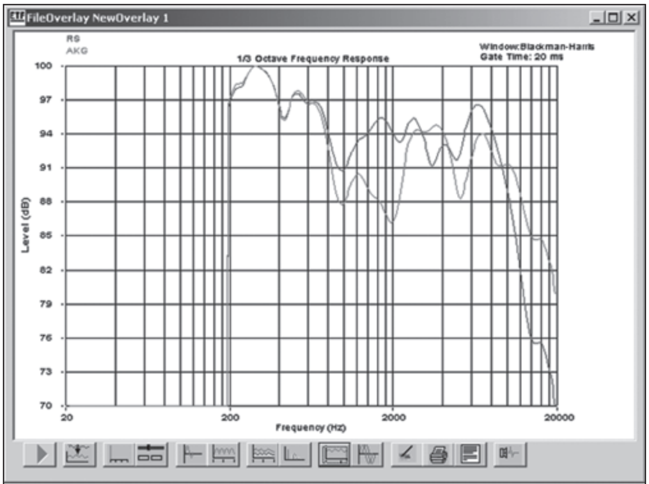

FIGURE 3B: Graph similar to *Fig. 3A*, but it compares the microphone responses at mid and high frequencies.

only for a very small physical area.

Indeed, in most rooms it's not even possible to get the response flat for both ears at the same time. Moreover, modal ringing, as mentioned earlier, is at least as damaging as the peaks and nulls, and EQ cannot improve that. Nor can EQ counter the typical nulls that are 10 to 30dB deep without blowing up your amp and speakers.

Newer systems claim that their use of sophisticated DSP (Digital Signal Processing) overcomes the problems with equalizers, but they suffer from the same problems for the same reasons. The more correction that is applied, the smaller the improved physical area becomes. So using either EQ or a DSP system is guaranteed to make the response worse elsewhere in the room, even the next seat over on a couch. The only place an equalizer makes sense is at the very lowest octave below, say, 40 or 50Hz, where conventional bass traps tend to be less effective. Even then, you should use an equalizer to reduce peaks only—never to try to bring up a null. *aX*

By Bjørn Kolbrek

Horn Theory: An Introduction, Part 2

The author continues his look at the various horn types and how they work.

SPHERICAL WAVE HORN

The spherical wave (or Kugelwellen) horn was invented by Klangfilm, the motion picture division of Siemens, in the late 1940s[26, 27]. It is often mistaken for being the same as the tractrix horn. It's not. But it is built on a similar assumption: that the wave-fronts are spherical with a constant radius. The wave-front area expansion is exponential.

To calculate the spherical wave horn contour, first decide a cutoff frequency f_c and a throat radius y_o (**Fig. 20**). The constant radius r_0 is given as

$$r_0 = \frac{c}{\pi f_c} \tag{18}$$

The height of the wave-front at the throat is

$$h_0 = r_0 - \sqrt{r_0^2 - y_0^2} \tag{19}$$

The area of the curved wave-front at the throat is

$$S_0 = 2\pi r_0 h_0$$

and the area of the wave-front with height h is $2\pi r_0 h$. Thus for the area to increase exponentially, h must increase exponentially:

$$h = h_0 e_{mx} \tag{20}$$

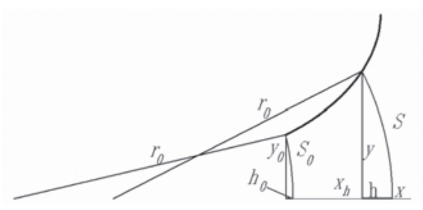

FIGURE 20: Dimensions of a spherical wave horn.

where x is the distance of the top of the wave-front from the top of the throat wave-front and

$$m = \frac{4\pi f_c}{c}.$$

Now that you know the area of the wave-front, you can find the radius and the distance of this radius from the origin.

$$S = 2\pi r_0 h$$

$$y = \sqrt{\frac{S}{\pi} - h^2} \tag{21}$$

$$x_h = x - h + h_0 \tag{22}$$

The assumed wave-fronts in a spherical wavehorn are shown in **Fig. 21**. Notice that the wave-fronts are not assumed to be 90° on the horn walls. Another property of the spherical wave horn is that it can fold back on itself (**Fig. 22**), unlike the tractrix horn, which is limited to a 90° tangent angle.

The throat impedance of a 100Hz spherical wave horn—assuming wave-fronts in the form of flattened spherical caps and using the radiation impedance of a sphere with radius equal to the mouth radius as mouth termination—is shown in **Fig. 23**. You can see that it is not very different from the throat impedance of a tractrix horn.

LE CLÉAC'H HORN

Jean-Michel Le Cléac'h presented a horn that does not rely on an assumed wave-front shape. Rather, it follows a "natural expansion." The principle is shown in **Fig. 24**. Lines 0-1 show the wave-front surface at the throat (F1). At the point it reaches F2, the wave-front area has expanded, and to account for this, a small triangular element (or, really, a sector of a circle) b1 is added.

The wave-front expansion from b1 (line 3-4) continues in element a3, and an element b2 is added to account for further wave expansion at F3. The process is repeated, and the wave-front becomes a curved surface, perpendicular to

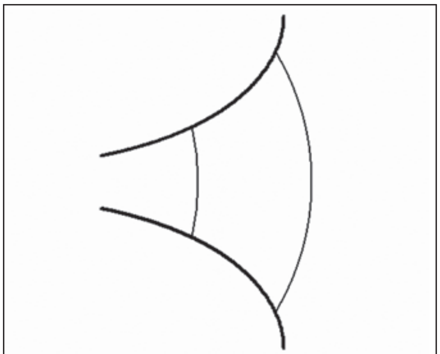

FIGURE 21: Assumed wave-fronts in spherical wave horns.

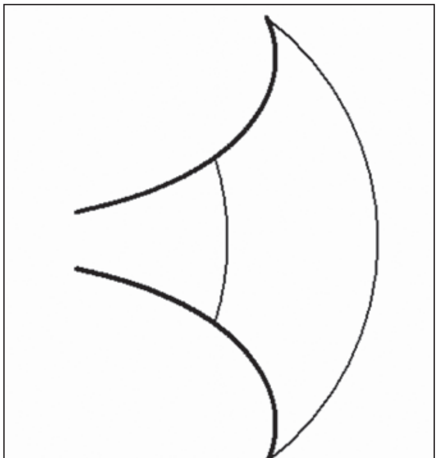

FIGURE 22: Spherical wave horn folding back.

the axis and the walls, but without making any assumptions regarding the shape prior to the calculations. The wave-fronts are equidistant from each other, and appear to take the shape of flattened spherical caps. The resulting contour of the horn is shown in **Fig. 25**.

The wave-front expands according to the Salmon family of hyperbolic horns. There is no simple expansion equation for the contour of the Le Cléac'h horn, but you can calculate it with the help of spreadsheets available at http://ndaviden.club.fr/pavillon/lecleah.html

OBLATE SPHEROIDAL WAVEGUIDE

This horn was first investigated by Freehafer[28], and later independently by Geddes[6], who wanted to develop a horn suitable for directivity control in which the sound field both inside and outside the horn could be accurately predicted. To do this, the horn needed to be a true 1P-horn. Geddes investigated several coordinate systems, and found the oblate spheroidal (OS) coordinate system to admit 1P waves. Putland[7] later showed that this was not strictly the case. More work by Geddes[29] showed that the oblate spheroidal waveguide acts like a 1P horn for a restricted frequency range. Above a certain frequency dictated by throat radius and horn angle, there will be higher order modes that invalidate the 1P assumptions.

The contour of the oblate spheroidal waveguide is shown in **Fig. 26**. It follows the coordinate surfaces in the coordinate system used, but in ordinary Cartesian coordinates, the radius of the horn as a function of x is given as

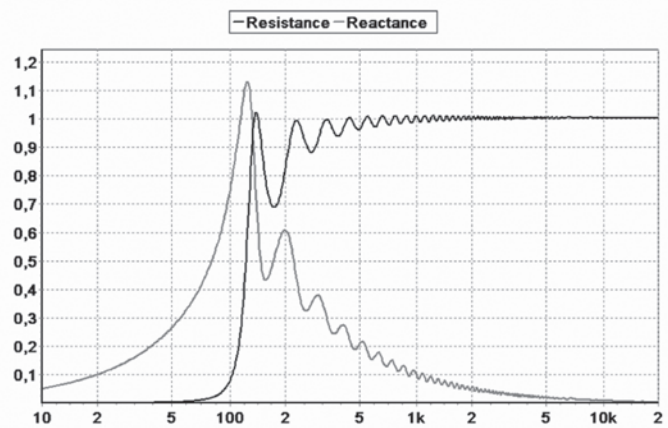

FIGURE 23: Throat impedance of a spherical wave horn.

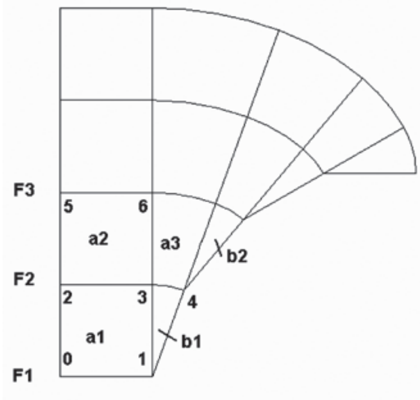

FIGURE 24: The principle of the Le Cléac'h expansion.

$$r = \sqrt{r_t^2 + \tan^2(\theta_0)x^2} \qquad (23)$$

where
r_t is the throat radius, and
θ_0 is half the coverage angle.
The throat acoustical impedance is not given as an analytical function; you must find it by numerical integration. The throat impedance for a waveguide with a throat diameter of 35.7mm and θ_0 = 30 is shown in **Fig. 27**.

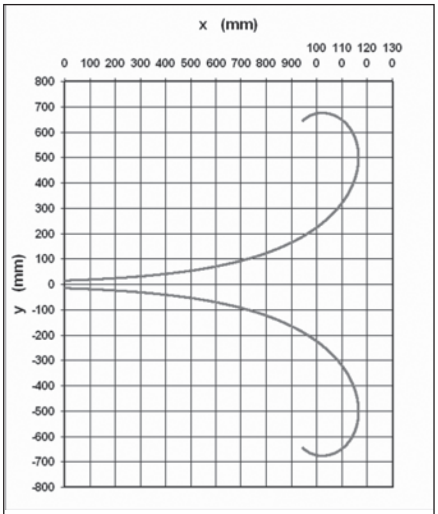

FIGURE 25: Contour of a Le Cléac'h horn.

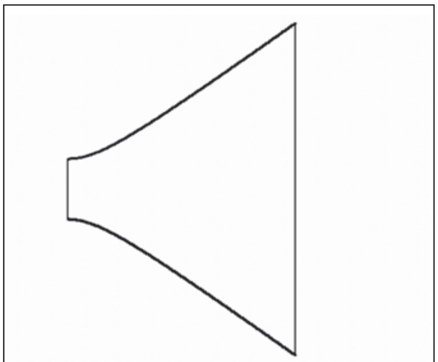
FIGURE 26: Contour of the oblate spheroidal waveguide.

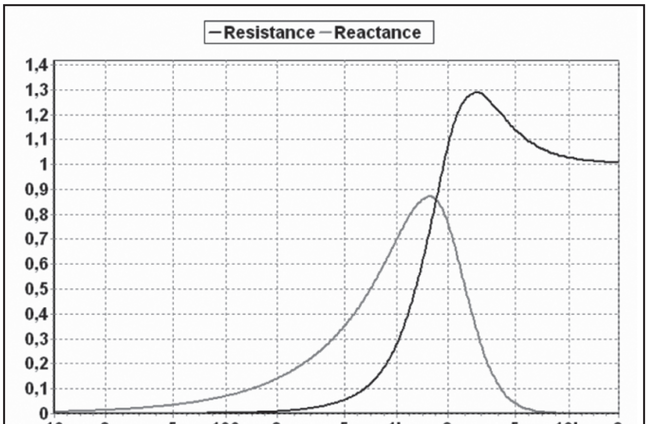

FIGURE 27: Normalized throat impedance of a 60° included angle infinite oblate spheroidal waveguide.

The OS waveguide does not have a sharp cutoff like the exponential or hyperbolic horns, but it is useful to be able to predict at what frequency the throat impedance of the waveguide becomes too low to be useful. If you set this frequency at the point where the throat resistance is 0.2 times its asymptotic value[30], so that the meaning of the cutoff frequency becomes similar to the meaning of the term as used with exponential horns, you get

$$f_c = \frac{0.2c}{\pi}\frac{\sin\theta_0}{r_t} \qquad (24)$$

You see that the cutoff of the waveguide depends on both the angle and the throat radius. For a low cutoff, a larger throat and/or a smaller angle is required. For example, for a 1″ driver and 60° included angle (θ_0 = 30), the cutoff is about 862Hz.

The advantages of the OS waveguide are that it offers improved loading over a conical horn of the same coverage angle, and has about the same directional properties. It also offers a very smooth transition from plane to spherical wave-fronts, which is a good thing, because most drivers produce plane wave-fronts.

The greatest disadvantage of the OS waveguide is that it is not suitable for low-frequency use. Bass and lower midrange horns based on this horn type will run into the same problems as conical horns: the horns become very long and narrow for good loading.

To sum up, the OS waveguide provides excellent directivity control and fairly good loading at frequencies above about 1kHz.

OTHER HORNS

Three other horn types assuming curved wave-fronts that are worth mentioning are: the Western Electric horns, the Wilson modified exponential, and the Iwata horn. What these horns have in common is that they do not assume curved wave-fronts of constant radius.

The Western Electric type horn[17] uses wave-fronts of constantly increasing radius, all being centered around a vertex a certain distance from the throat (**Fig. 29**).

In the Wilson modified exponential horn[31], the waves start out at the throat and become more and more spherical. The horn radius is corrected in an iterative process based on the wall tangent angle, and the contour lies inside that of the plane-wave exponential horn, being a little longer and with a slightly smaller mouth flare tangent angle (**Fig. 28**). Unfortunately, the Wilson method only corrects the wave-front areas, not the distance between the successive wave-fronts.

There is not much information available about the Iwata horn[32, 33], just a drawing and dimensions, but no description of the concept. It looks like a radial horn, and seems to have cylindrical wave-fronts expanding in area like a hypex-horn with T = √2. The ratio of height to width increases linearly from throat to mouth.

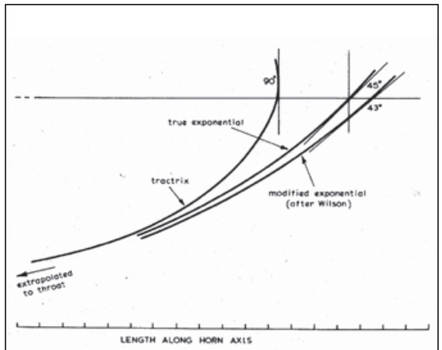

FIGURE 28: Comparison of the exponential horn with the tractrix and the Wilson modified exponential horn[22].

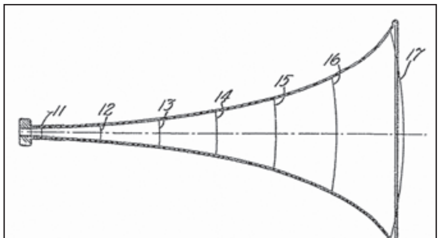

FIGURE 29: Wave-fronts in the Western Electric type exponential horn[17].

DIRECTIVITY CONTROL

Control of directivity is an important aspect of horn design. An exponential horn can provide the driver with uniform loading, but at high frequencies, it starts to beam. It will therefore have a coverage angle that decreases with frequency, which is undesirable in many circumstances. Often you want the horn to radiate into a defined area, spilling as little sound energy as possible in other areas. Many horn types have been designed to achieve this.

For the real picture of the directivity performance of a horn, you need the polar plot for a series of frequencies. But sometimes you also want an idea of how the coverage angle of the horn varies with frequency, or how much amplification a horn gives. This is the purpose of the directivity factor (Q) and the directivity index (DI)[34]:

Directivity Factor: The directivity factor is the ratio of the intensity on a given axis (usually the axis of maximum radiation) of the horn (or other radiator) to the intensity that would be produced at the same position by a point source radiating the same power as the horn.

Directivity Index: The directivity index is defined as: $DI(f) = 10 \log_{10} Q(f)$. It indicates the number of dB increase in SPL at the observation point when the horn is used compared to a point source.

Because intensity is watts per square meter, it is inversely proportional to area, and you can use a simple ratio of areas[35]. Consider a sound source radiating in all directions and observed at a distance r. At this distance, the sound will fill a sphere of radius r. Its area is $4\pi r^2$. The ratio of the area to the area covered by a perfect point source is 1, and thus Q = 1. If the sound source is radiating into a hemisphere, the coverage area is cut in half, but the same sound power is radiated, so the sound power per square meter is doubled. Thus Q = 2. If the hemisphere is cut in half, the area is 1/4 the area covered by a point source, and Q = 4.

For a horn with coverage angles α and β as shown in **Fig. 31**, you can compute Q as

$$Q = \frac{180}{\sin^{-1}\left(\sin\frac{\alpha}{2}\sin\frac{\beta}{2}\right)} \qquad (25)$$

Most constant directivity horns try to act as a segment of a sphere. A sphere will emit sound uniformly in all directions, and a segment of a sphere will emit sound uniformly in the angle it defines, provided its dimensions are

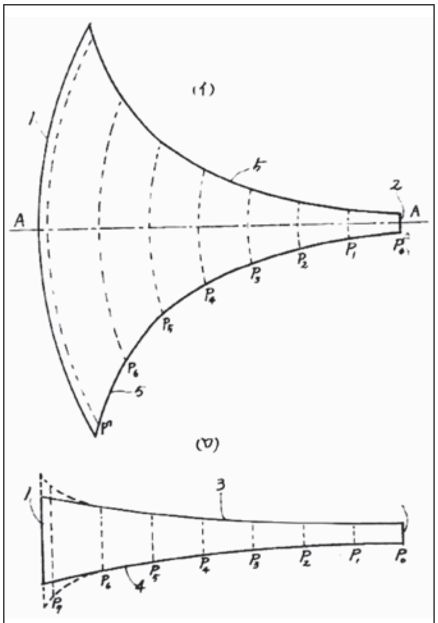

FIGURE 30: Contour of the Iwata horn[32].

large compared to the wavelength[11]. But when the wavelength is comparable to the dimensions of the spherical segment, the beam width narrows to 40-50% of its initial value.

A spherical segment can control directivity down to a frequency given as

$$f_I = \frac{25 \cdot 10^6}{x \cdot \theta} \tag{26}$$

where

f_I is the intercept frequency in Hz where the horn loses directivity control,

x is the size of the horn mouth in mm in the plane of coverage, and

θ is the desired coverage angle in degrees in that plane.

You thus need a large horn to control directivity down to low frequencies.

Most methods of directivity control rely on simulating a segment of a sphere.

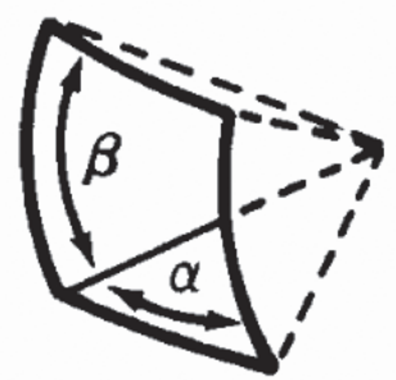

FIGURE 31: Radiation into a solid cone of space defined by angles α and β.

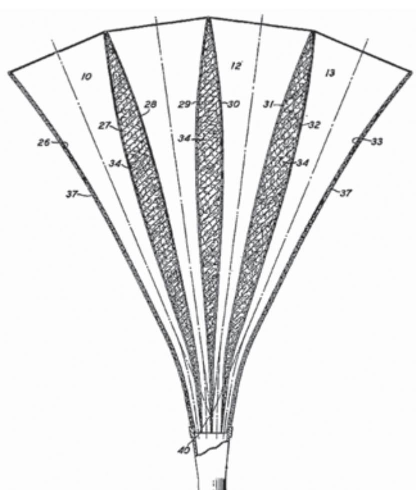

FIGURE 32: Multicellular horn[38].

The following different methods are listed in historical order.

MULTICELLULAR HORNS

Dividing the horn into many conduits is an old idea. Both Hanna[36] and Slepian[37] have patented multicellular designs, with the conduits extending all the way back to the source. The source consists of either multiple drivers or one driver with multiple outlets, where each horn is driven from a separate point on the diaphragm.

The patent for the traditional multicellular horn belongs to Edward C. Wente[38]. It was born from the need to accurately control directivity, and at the same time provide the driver with proper loading, and was produced for use in the Bell Labs experiment of transmitting the sound of a symphonic orchestra from one concert hall to another[39].

A cut view of the multicellular horn, as patented by Wente, is shown in **Fig. 32**. In this first kind of multicellular horn, the individual horns started almost parallel at the throat, but later designs often used straight horn cells to simplify manufacture of these complex horns. As you can see, the multicellular horn is a cluster of smaller exponential horns, each with a mouth small enough to avoid beaming in a large frequency range, but together they form a sector of a sphere large enough to control directivity down to fairly low frequencies. The cluster acts as one big horn at low frequencies. At higher frequencies, the individual horns start to beam, but because they are distributed on an arc, coverage will still be quite uniform.

The multicellular horn has two problems, however. First, it has the same lower midrange narrowing as the ideal sphere segment, and, second, the polar pattern shows considerably “fingering” at high frequencies. This may not be as serious as has been thought, however. The -6dB beam widths of a typical multicellular horn are shown in **Fig. 33**. The fingering at high frequencies is shown in **Fig. 34**.

The beam width of a multicellular horn with different number of cells is shown in **Fig. 35**[34]. The narrowing in beam width where the dimensions of

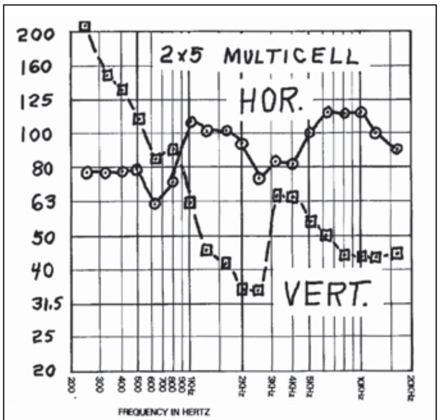

FIGURE 33: -6dB beam widths of Electro-Voice model M253 2 by 5 cell horn[42].

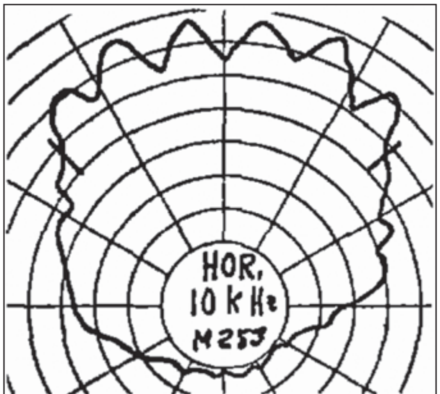

FIGURE 34: High frequency fingering of EV M253 horn at 10kHz[42].

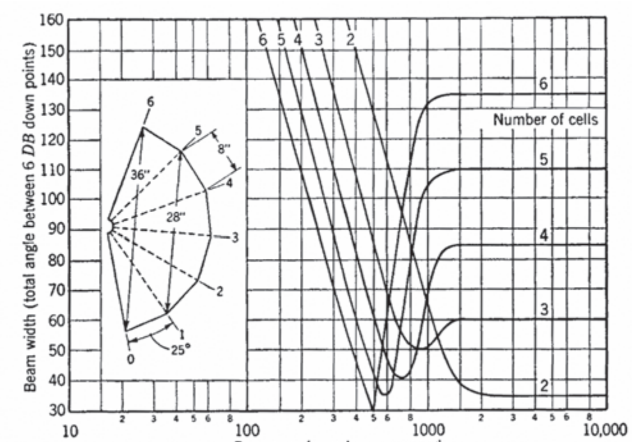

FIGURE 35: Beam width of a multicellular horn constructed as shown in the insert[34].

the horn are comparable to the wavelength is evident.

RADIAL HORNS

The radial or sectoral horn is a much simpler concept than the multicellular horn. The horizontal and vertical views of a radial horn are shown in **Fig. 36**. The horizontal expansion is conical, and defines the horizontal coverage angle of the horn. The vertical expansion is designed to keep an exponential expansion of the wave-front, which is assumed to be curved in the horizontal plane. Directivity control in the horizontal plane is fairly good, but has the same midrange narrowing as the multicellular horn. In addition, there is almost no directivity control in the vertical plane, and the beam width is constantly narrowing with increasing frequency.

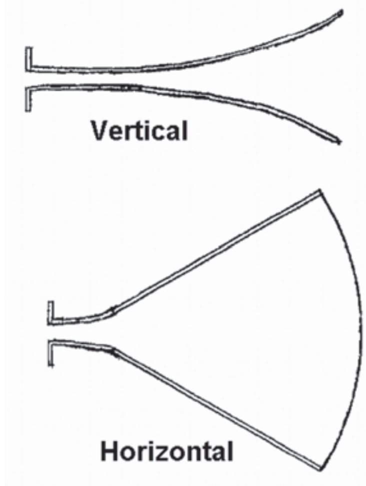

FIGURE 36: Profile of a radial horn[42].

REVERSED FLARE HORNS

The reversed flare horn can be considered to be a "soft diffraction horn," contrary to Manta-Ray horns and other modern constant directivity designs that rely on hard diffraction for directivity control. This class of horns was patented for directivity control by Sidney E. Levy and Abraham B. Cohen at University Loudspeakers in the early 1950s[40, 41]. The same geometry appeared in many Western Electric horns back in the early 1920s, but the purpose does not seem to be that of directivity control[17].

The principle for a horn with good horizontal dispersion is illustrated in **Fig. 37**. The wave is allowed to expand in the vertical direction only, then the direction of expansion is changed. The wave-front expansion is restricted vertically, and is released horizontally. The result is that the horizontal pressure that builds up in the first part of the horn causes the wave-front to expand more as it reaches the second part. That it is restricted in the vertical plane helps further.

Because the wave-front expansion is to be exponential all the way, the discontinuity at the flare reversal point (where the expansion changes direction) is small. In addition, the change of curvature at the flare reversal point is made smoother in practical horns than what is shown in the figure.

CE HORNS

In the early 1970s, Keele, then working for Electro-Voice, supplied an answer to the problems associated with multicellular and radial horns by introducing a completely new class of horns that provided both good loading for the driver and excellent directivity control[42].

The principle is based on joining an exponential or hyperbolic throat segment for driver loading with two conical mouth segments for directivity control. The exponential and conical segments are joined at a point where the conical horn of the chosen solid angle is an optimum termination for the exponential horn. Keele defines this as the point where the radius of the exponential horn is

$$r = \frac{0.95 \sin\theta}{k_c} \tag{27}$$

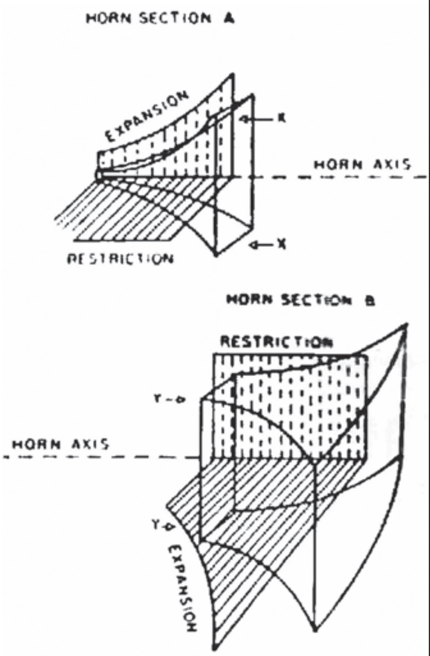

FIGURE 37: Wave-front expansion in reversed flare horns[41].

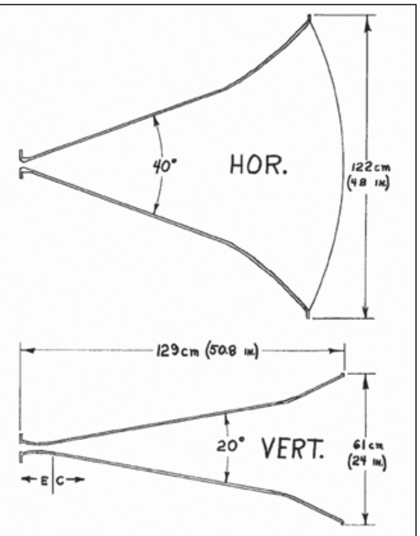

FIGURE 38: Example of the Electro-Voice CE constant directivity horns. This horn covers 40° by 20°[42].

where
r is the radius at the junction point,
θ is the half angle of the cone with solid angle Ω,

$$\theta = \cos^{-1}\left(1 - \frac{\Omega}{2\pi}\right), \text{ and}$$

k_c is the wave number at the cutoff frequency, $k_c = \frac{2\pi f_c}{c}$.

The problem of midrange narrowing was solved by having a more rapid flare close to the mouth of the conical part of the horn. Good results were obtained by doubling the included angle in the last third of the conical part. This decreases the acoustical source size in the frequency range of midrange narrowing, causing the beam width to widen, and removing the narrowing. The result is a horn with good directivity control down to the frequency dictated by the mouth size.

For a horn with different horizontal and vertical coverage angles, the width and height of the mouth will not be equal. The aspect ratio of the mouth will be given as

$$R = \frac{X_H}{X_V} = \frac{\sin\frac{\theta_H}{2}}{\sin\frac{\theta_H}{2}}, \tag{28}$$

or, if θ_H and θ_V are limited to 120°,

$$R \approx \frac{\theta_H}{\theta_V}. \tag{29}$$

The lower frequency of directivity control will also be dictated by the mouth aspect ratio. Substituting equation 26 into equation 29 and solving for the ratio of intercept frequencies, you get

$$\frac{f_{IH}}{f_{IV}} \approx \left(\frac{\theta_H}{\theta_V}\right)^2. \tag{30}$$

For a 40° by 20° (H-V) horn, the vertical intercept frequency will be four times higher than the horizontal intercept frequency.

MANTA-RAY HORNS

The Altec Manta-Ray horn sought to solve the problems of the CE horns, mainly the inability to independently specify the horizontal and vertical intercept frequencies[43]. To achieve directivity down to a lower frequency in the vertical plane, the vertical dimension of the mouth must be increased. Because the dispersion angle is smaller, the expansion must start further back, behind where the horizontal expansion starts. The result is the unique geometry shown in **Fig. 39** (although it's not so unique anymore).

At the point where the horizontal expansion starts, the wave is diffracted to fill the width of the horn, and dispersion is controlled by the horn walls.

The Manta-Ray horn incorporates the same rapid mouth flaring as the CE horns to avoid midrange narrowing, but does not use radial expansion of the walls. The reason for this is that radial walls produced a "waist-banding" effect, in which the horn lost much energy out to the sides in the upper midrange. This effect cannot be seen in the polar plots for the CE horns, which suggests that "waist-banding" can be a result of the Manta-Ray geometry, and not solely of radial wall contours.

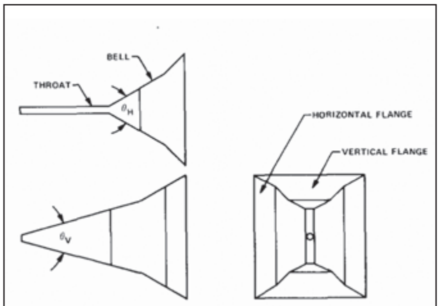

FIGURE 39: The Manta-Ray geometry[43].

NEW METHODS

Most newer constant directivity designs have been based on either the conical horn, some sort of radial horn (including the JBL Biradial design), or diffraction methods such as the Manta-Ray design. The only notable exception is the oblate spheroidal waveguide (covered previously) introduced by Geddes.

The general trend in horns designed for directivity control has been to focus on the control issue, because it is always possible to correct the frequency response. A flat frequency response does not, however, guarantee a perfect impulse response, especially not in the presence of reflections. Reflected waves in the horn at the high levels in question will also cause the resulting horn/driver combination to produce higher distortion than necessary, because the driver is presented with a nonlinear and resonant load. (See next section.)

DISTORTION

As mentioned, the horn equation is derived assuming that the pressure variations are infinitesimal. For the intensities appearing at the throat of horns, this assumption does not hold. Poisson showed in 1808 that, generally, sound waves cannot be propagated in air without change in form, resulting in the generation of distortion, such as harmonics and intermodulation products. The distortion is caused by the inherent nonlinearity of air.

If equal positive and negative increments of pressure are impressed on a mass of air, the changes in volume of that mass will not be equal. The volume change for positive pressure will be less than that for the equal negative pressure[44]. You can get an idea of the nature of the distortion from the adiabatic curve for air (**Fig. 40**). The undisturbed pressure and specific volume of air ($\frac{1}{\rho}$) is indicated in the point P_0V_0. Deviation from the tangent of the curve at this point will result in the generation of unwanted frequencies, the peak of the wave being stretched and the trough compressed.

The speed of sound is given as

$$c = \sqrt{\gamma \frac{p}{\rho}} \qquad (31)$$

where
γ is the adiabatic constant of air, γ = 1.403.
You can see that the speed of sound increases with increasing pressure. So for the high pressure at the peaks of the wave-front, the speed of sound is higher than at the troughs. The result is that as the wave propagates, the peaks will gain on the troughs, altering the shape of the waveform and introducing harmonics (**Fig. 41**).

There are thus two kinds of distortion of a sound wave: one because of the unequal alteration of volume, and another because of the propagation itself. This last kind of distortion is most noticeable in a plane wave and in waves that expand slowly, as in horns, where distortion increases with the length propagated. Both kinds of distortion generate mainly a second harmonic component.

Fortunately, as the horn expands, the pressure is reduced, and the propagation distortion reaches an asymptotic value, which can be found for the horn in question, considering how it expands. It will be higher for a horn that expands slowly near the throat than for one that expands rapidly. For example, a hyperbolic-exponential horn with a low value for T will have higher distortion than a conical horn. For an exponential horn, the pressure ratio of second harmonic to fundamental is given as[44]

$$\frac{p_2}{p_1} = \frac{\gamma+1}{2\sqrt{2}} \frac{p_{1t}}{\gamma p_0} \frac{\omega}{c} \frac{-e^{-mx/2}}{m/2} \qquad (32)$$

where
p_{1t} is the RMS pressure of the fundamental at the throat,
p_1 is the RMS pressure of the fundamental at x,
p_2 is the RMS pressure of the second harmonic at x,
p_0 is the static pressure of air, and
m is the flare rate of the exponential horn.

You can see that distortion increases with frequency relative to the cutoff frequency. This is easier to see in the

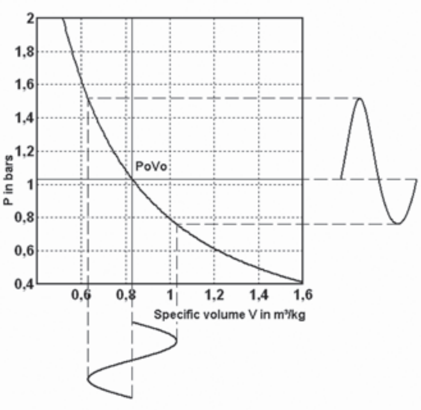

FIGURE 40: Adiabatic curve for air.

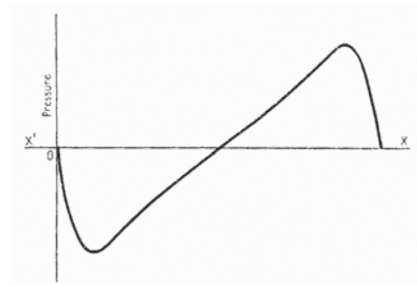

FIGURE 41: Distorted waveform due to non-constant velocity of sound[5].

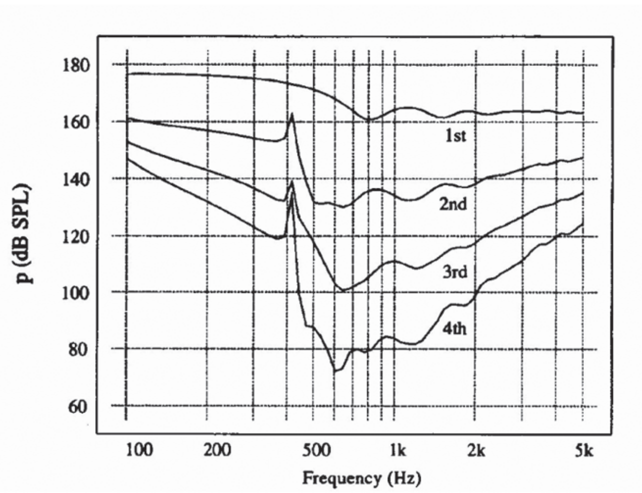

FIGURE 42: Level of harmonics at the throat for a sinusoidal mouth sound pressure level of 150dB, frequency sweep[45].

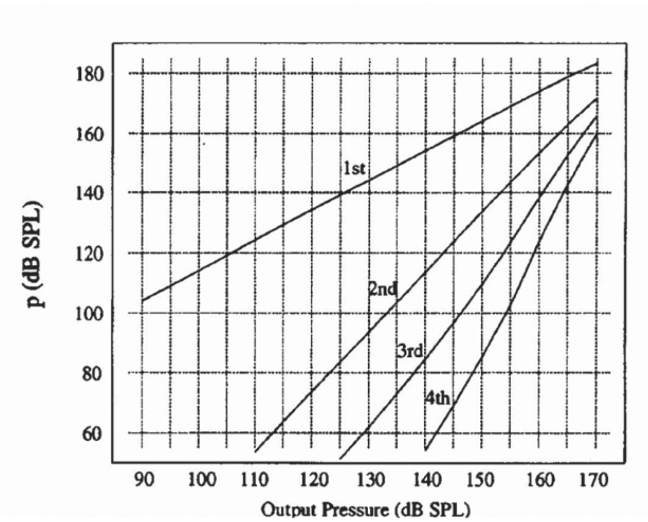

FIGURE 43: Level of harmonics at the throat for a 1kHz sinusoidal wave at the mouth, level sweep[45].

simplification for an infinite exponential horn given by Beranek[34]:

$$D_2[\%] = 1.73 \cdot 10^{-2} \frac{f}{f_c} \sqrt{I_t} \qquad (33)$$

where

I_t is the intensity at the throat, in watts per square meter.

Holland et al.[45] have investigated the distortion generated by horns both with the use of a computer model and by measurements. The model considered the harmonics required at the throat to generate a pure sine wave at the mouth (backward modeling), and also took reflections from the mouth into account. For a horn with a 400Hz cutoff and 4″throat, and a mouth SPL of 150dB, the distribution of harmonics is shown in **Fig. 42**. The peak at the cutoff frequency is due to the very high level required at the throat to generate the required SPL at the mouth.

Figure 43 shows the level of the harmonics at the throat at 1kHz for a given SPL at the mouth. Measurements showed that the prediction of the second harmonic level was quite accurate, but measured levels of the higher harmonics were higher than predicted. This was recognized as being due to nonlinearities in the driver.

As you can see from the results, the level of harmonics is quite low at the levels usually encountered in the home listening environment, but can be quite considerable in the case of high-level public address and sound reinforcement systems.

One point I need to mention is the importance of reducing the amount of reflection to reduce distortion. At the high levels involved, the reflected wave from diffraction slots or from the mouth will not combine with the forward propagating wave in a linear manner. The result will be higher distortion, and a nonlinear load for the driver. A driver working into a nonlinear load will not perform at its best, but will produce higher distortion levels than it would under optimum loading conditions[45].

Directivity of the horn also plays a role in the total distortion performance[46]. If the horn does not have constant directivity, the harmonics, because they are higher in frequency, will be concentrated toward the axis, while the fundamental spreads out more. This means that distortion will be higher on-axis than off-axis.

HIGHER ORDER MODES

At low frequencies, you can consider wave transmission in most horns as one-dimensional (1P waves). When the wavelength of sound becomes comparable to the dimensions of the horn, however, cross reflections can occur. The mode of propagation changes from the simple fundamental mode to what is called higher order modes. The behavior of these modes can be predicted for the uniform pipe and the conical horn[47, 48, 49], and it is found that they have cutoff frequencies below which they do not occur.

In 1925, Hoersch conducted a theoretical study of higher order modes in a conical horn, and calculated the equipressure contours for two kinds of modes. The results (**Fig. 44**) show the equipressure contours including both the radial and non-radial vibrations. The left part of the figure shows a pattern that resembles what Hall measured in a conical horn (Fig. 16). For a flaring horn such as the exponential, however, the higher order modes will occur at different frequencies at different places in the horn[8].

Higher order modes will also be generated by rapid changes in flare, such as discontinuities, so the slower and smoother the horn curvature changes, the less the chance for generating higher order modes.

The effect of the higher order modes is to disturb the shape of the pressure wave-front, so that directivity will be unpredictable in the range where the modes occur. According to Geddes, they may also have a substantial impact on the perceived sound quality of horns[50].

CLOSING REMARKS

In this article, I have tried to present both classical and modern horn theory in a comprehensive way. A short article like this can never cover all aspects of horns. But I hope it has provided useful information about how horns work, maybe also shedding light on some lesser known aspects and research.

Finally, I would like to thank Thomas Dunker and David McBean for proofreading, discussion, and suggestions.

aX

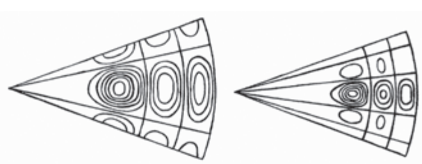

FIGURE 44: Higher order modes in a conical horn[47].

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 By Jan Didden

aX Visits **Burning Amp '07**

The Burning Amplifier Festival did not live up to its name. No amplifiers or any other equipment burst into flames during the event. What did burst into the open was a vibrant, very alive community of audio DIY enthusiasts of true global proportions, presenting completed designs and research projects that could hold their own against many professional products.

PHOTO 1: The focus at the BAF was, of course, listening to music.

You know how it is. Bring together a few people sharing the same hobby and before long they will plan a get together, an "event." The online audio DIY community of diyaudio.com tested the water with its first last summer in the UK.

Not to be outdone, a group of moderators of diyaudio.com put an organization together for a US-based event, the Burning Amplifier Festival (with a wink to the Burning Man Festival at Black Rock Desert). Held in the old firehouse of San Francisco's Fort Mason, against the spectacular backdrop of SF bay and Alcatraz, BAF07 opened its doors on Oct. 21 at 10:00, and closed down 12 hours later. The event was organized around group demos.

Participants were assigned to five groups, each with a 2 hour slot to demo and explain their projects. Each group had several speaker systems, amplifiers, sources, cabling, and so on, which were swapped in and out during the listening session, which gave a very good impression of differences (or not) between components. The show was a good mix between an organized event and group interactions.

One highlight of the show was the visit of Nelson Pass, pulling up with a pick-up truck literally full of "goodies" to give away. But the main focus was on listening and discussions about the usual subjects such as sound quality, tube versus solid-state engineering, and speaker building (**Photo 1**).

HIGHLIGHTS

Amplifiers, of course, abounded, although I didn't see many single-ended designs. Several participants demonstrated Pass-based DIY amps, including a towering version of the F4. The amp was still being finished at the show and Nelson Pass wasn't above lending a hand (**Photo 2**). An interesting hybrid FET/tube amp was shown which sported a switched-mode high-voltage supply. Because of that, the amp was unusually small in size (**Photo 3**).

The trend in speakers was toward full range, simple but high quality. One speaker, the BOFU BIB (where BIB stands for Bigger Is Better) drew a lot of attention with an excellent sound quality and a build price of just $100 a pair. It used a Pioneer full-range unit with a tiny tweeter (**Photo 4**).

Iain McNeill showed a very professional DSP crossover, based on an AD Blackfin DSP evaluation board and Matlab software. The system drove three-way speakers through six dedicated power amps (based on Doug Self's Blameless amps). The sound was well

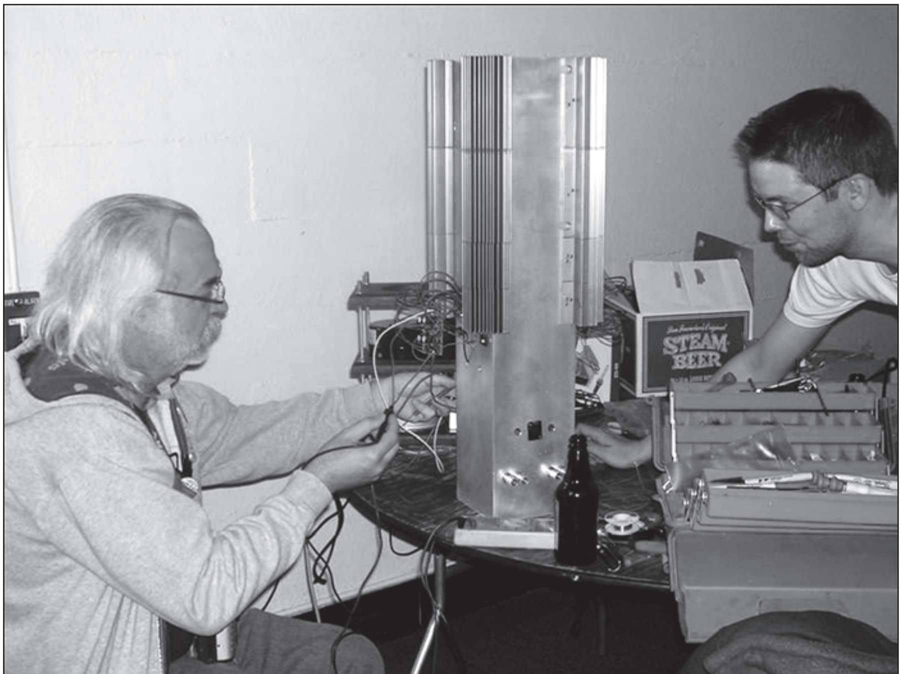

PHOTO 2: Nelson Pass (left) helping out to finish Mark's DIY version of the Pass F4.

integrated and the driver takeover appeared seamlessly. This was clearly an advanced project.

In the side rooms two sets of test equipment were set up. Jack Hidley of NHT demonstrated a complete Klippel speaker driver test bench, complete with Audio Precision hardware and a laser interferometer. Folks who brought drivers with them had no trouble finding him! The laser interferometer was used to track cone movement as a function of drive level and frequency **(Photo 5)**. The demo showed all kinds of interesting things, such as a cone that "jumps out" at a certain drive level and then vibrates around a new neutral position which is actually away from the mechanical rest position without drive.

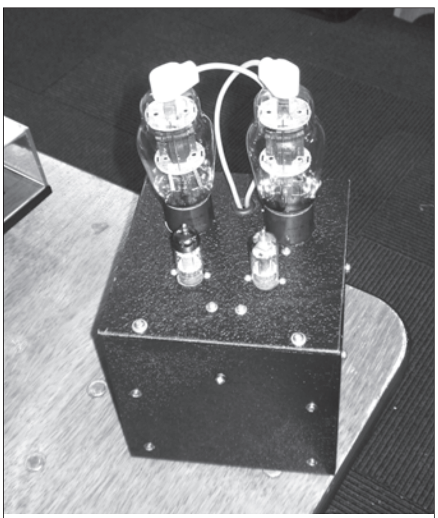

PHOTO 3: Richard's compact hybrid amp with a switched-mode high voltage power-supply.

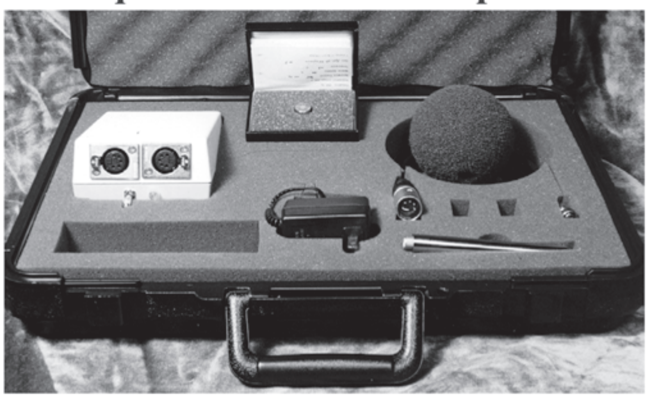

PHOTO 4: The BOFU BIB $100/pr speakers were a full family project.

In another corner Kent Leech had an Audiomatica Sofia tube curve tracer hooked up to his laptop and ran curves for people's tubes.

By all accounts, Burning Amp 2007 was a resounding success. Participation ranged from a sizeable Californian group to enthusiasts coming all the way from Denmark, The Netherlands, Lithuania, Canada, and Serbia. For anyone reading this magazine or following the online communities, it's clear that amateur audio is alive and well. I'm looking forward to BAF08! *aX*

PHOTO 5: Klippel driver measurement system.

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Fig. 12 (Circuit Boards) from Jack Walton's "The Noise (Analysis) Machine," (*aX* 3/08).

Updated versions of Rudy Godmaire's "A Flexible Subwoofer Amplifier, Parts 1-4," to accompany his PC boards.

Articles from Past Issues:

"AES Multi-Tracks in New York 2007, Pt. 1," by David Weinberg (*aX* 2/08)

Product Review: Speaker City's PS10 Subwoofer Kit, by Julian J. Bunn (*aX* 2/08).

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By Ed Simon

The Secret Art of Breadboarding

You can test a circuit idea with one of these quick 'n' dirty breadboarding approaches.

I have an audio power amplifier circuit that is simplicity itself. It uses ten bipolar transistors, two MOSFETs, and an op amp. My simulation program shows it has less than .001% distortion and can deliver 1000W into an 8Ω load.

BACK TO BASICS

Unfortunately, common sense told me it would not work, but just to be sure I built a trial version. It oscillated, was noisy, overheated, and was completely useless. That is why before I design and order a printed circuit (PC) board, I find it useful to test out the circuit. Sometimes, when I build a one-off for my own use, it never sees a PC board.

There are many ways to build a trial circuit. You could start with a piece of wood, drive small brass tacks into it, and use these for solder terminals. To be completely accurate, you could use a real breadboard!

If you want to try this method, you might build a crystal radio. You will need a cardboard tube, some magnet wire, a wire coat hanger, a 1N34 germanium diode, a crystal earphone, perhaps four Fahnestock clips, a few nails, brads, or tacks, and a piece of wood. The larger the diameter of the cardboard tube, the better. You can use the center of a roll of bathroom tissue or an oatmeal container (**Photo 1**).

I used a piece of 3″ mailing tube, which was very rigid, making winding the coil easier. I put a small hole in each end of the tube to hold the wire and I wound as much of the surface of the tube as I could with a single smooth layer of 22-gauge wire (you can also use thinner or thicker wire). It is important to be sure the wire does not overlap a previous winding. You can use a spot of glue to hold the wire. I used almost 6 ounces of wire to cover my tube.

You can coat the entire coil with polyurethane to hold it in place while sanding it. Some folks might just use varnish or shellac, but shellac—although traditional—doesn't really hold well enough. If your coil is wound really tightly you might skip this step.

Using 220-grit sandpaper, I then sanded the insulation off a band across the top of the coil to allow for the tuner rod to have good contact. If you accidentally wind one wire over another, then you might just sand through it. Be careful.

Nail your tuning coil to your board. Arrange a piece of coat hanger that has also been sanded clean to act as a tuning lever. Use your Fahnestock clips for the antenna and earphone.

This is a classic school science fair project, or, if you make it look old, something of a collectable reproduction. But it really is a simple way to breadboard a circuit.

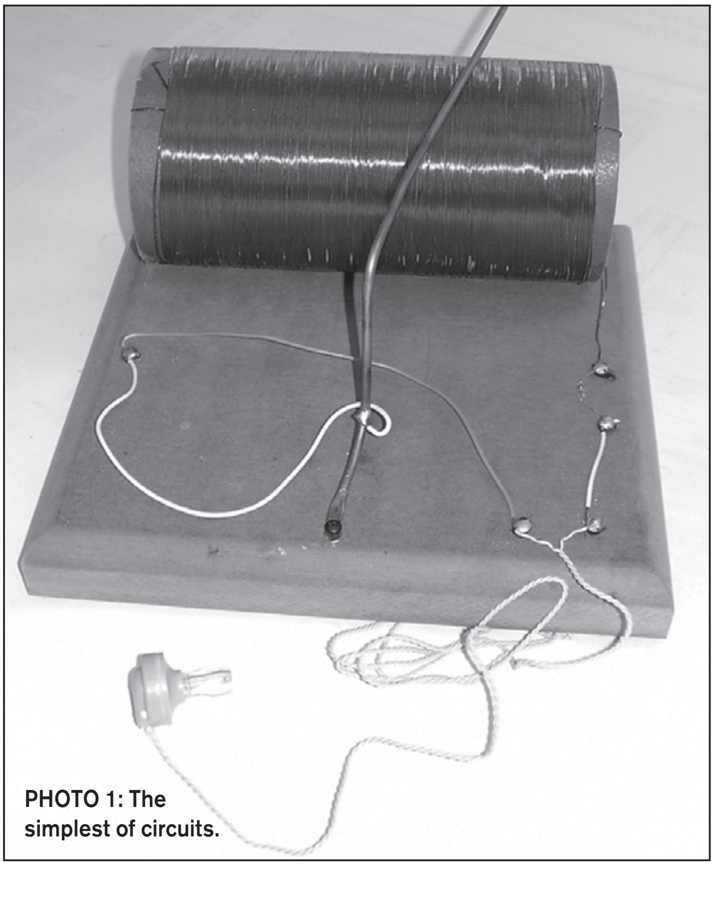

PHOTO 1: The simplest of circuits.

SWEEP GENERATOR

For those who want something fancier, the next step up is a small metal plate, which can be a piece of printed circuit stock or thin brass or copper "shim stock." You can buy brass shim stock at many hobby outlets. I used to use the stock 0.032″ thick, but as prices have risen, I tend to use more of the 0.020″ material.

The reason for using copper or brass is so that you can solder directly to it. It provides a very good ground plane. The parts that are grounded are also used as standoffs for the other parts. Air makes a great insulator for the other parts. If you need an extra support, you can add an extra power supply bypass capacitor or even a very high value resistor. Sometimes you can make a complete box out of brass and use it for both chassis and case.

Photo 2 shows an RF sweep generator I built using two voltage regulators, a simple saw tooth generator, and a voltage-controlled oscillator module. It puts out an RF signal from 80MHz to about 120MHz and is used for sweeping the FM band. Outputs are provided to connect to an oscilloscope to display the frequency response of a tuner's front end. On one connector is the diode detector to get the DC level for the display.

You can see how lots of bypass capacitors are used as standoffs. This is not particularly neat, but is quite practical for these frequencies and ease of modification. Because I needed only one for a few uses, this is the final product. The front panel has RF out and RF return. The back panel has horizontal, vertical, and common to the display.

A variant of this technique is to use the circuit board material and cut insulated areas with a razor knife or engraver. This gets you closer to a PC board type of product.

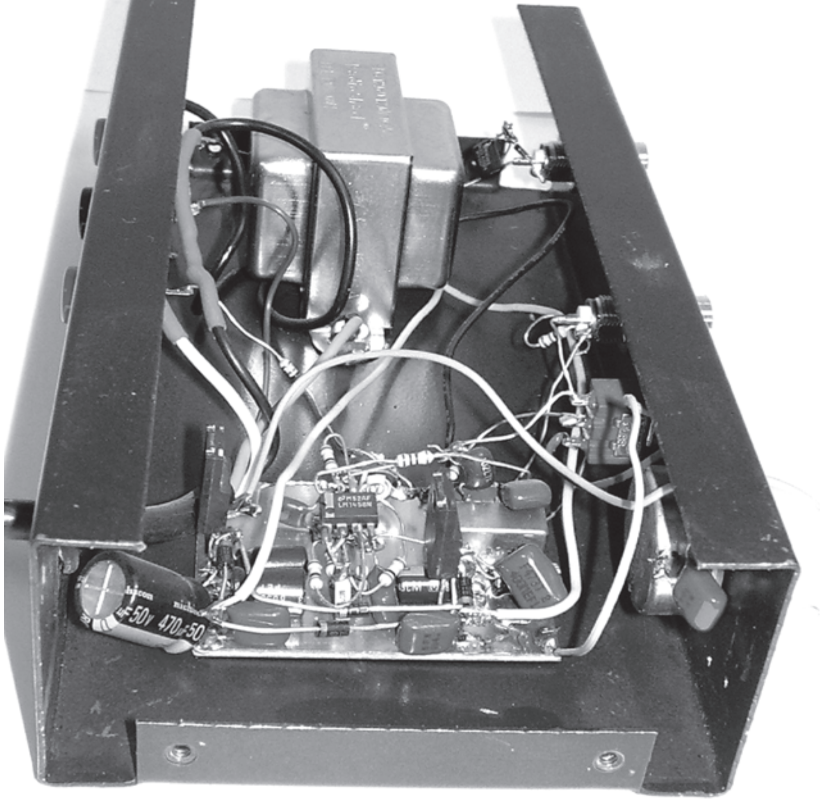

PHOTO 2: RF sweep generator.

SIMPLE CIRCUIT

A slightly fancier method is to use perfboard, which is available from most electronics suppliers. The most common type used today has small 0.042″ holes on a 0.100″ matrix. You can use push-in "flea" clips to hold components or just wire them through the holes.

Photo 3 is a simple circuit for using white LEDs as a light source. Note that the current limiter is a capacitor and this circuit is fed by AC through a diode bridge. This method gives a physically more secure product than the metal-based breadboard.

Many people prefer to use a universal circuit board to hold the trial circuit. There are lots of different types of these boards available—from the catalogs at Digi-Key, Mouser, or Old Colony. When I am having circuit boards made, there is often extra space on the panels that I do not need for the project at hand, so I add a few extra IC patterns and traces to give me a few prototype boards.

OSCILLATOR

Photo 4 shows a 100kHz calibration

oscillator I needed to calibrate an old oscilloscope. The 9V battery snap was removed from a dead battery and reused. The basic circuit is a multi-vibrator with the crystal in the feedback loop. Although you could trim such a circuit to WWV with a receiver, in this case such accuracy was not needed. The universal card was from Radio Shack.

Short lead lengths are very important in a circuit such as this, so this is not particularly neat. When I was in a college lab class, Tron always insisted that his wiring be perfectly neat and square. With his complete routing list and by doublechecking each wire as he installed it, it took him longer to build his circuits than anyone else. He insisted that they always work the first time. The teaching assistant who had to stay for two extra hours after the two-and-a-half hour lab session was not very pleased when it did not work!

A test circuit is just that. Do what it takes to test your ideas. It needs to be neat enough to allow for modifications, but if it is not a keeper, you may wish to invest your time in something more productive.

So far these breadboards have been for simple circuits. A method that I use is to take a punched metal plate and mount terminal strips on it. The plate can hold heavy components and the terminal strips the lighter ones. If you do a neat job, the finished plate can stand alone or be mounted in a wood case. The plate shown just has four rubber feet stuck onto the bottom.

Photo 5 shows a prototype power amplifier. This layout is very easy to modify for both topography of the circuit and change components to see what effect different values have on the per-

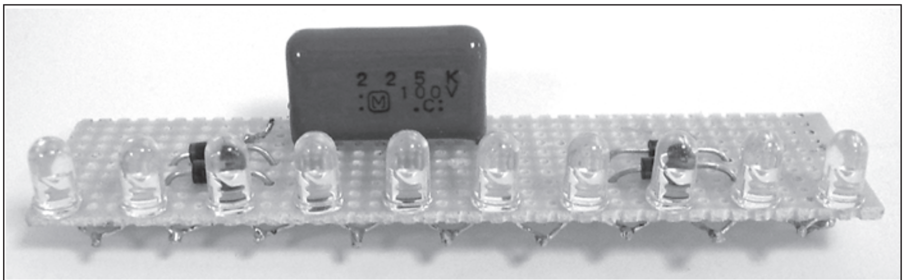

PHOTO 3: Simple circuit.

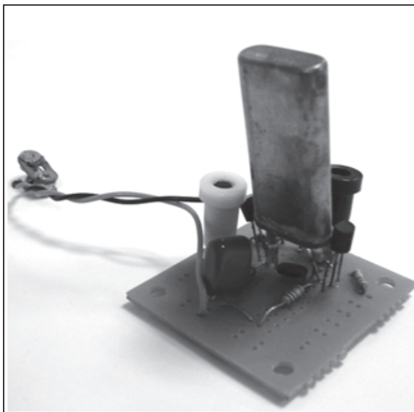

PHOTO 4: 100kHz calibration oscillator.

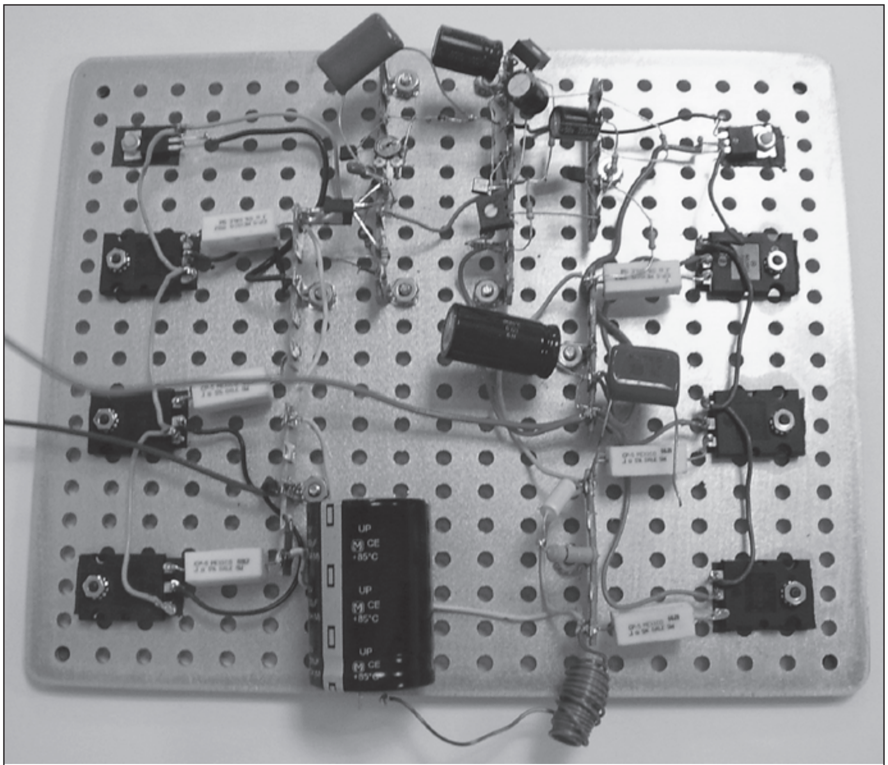

PHOTO 5: Prototype power amplifier.

formance. The transistors are mounted directly on the plate, which also doubles as a bit of a heatsink. The terminal strips are available from Mouser.

The hole spacing is a bit tricky. One axis is ½″ spacing, the other is 1cm. I can install both English and metric components on the plate. If you are building for show, you should be sure all the components are anchored at both ends on a terminal. Wire can then run between the terminals to make the connections. You can order metal parts such as these custom made or from folks such as Design, Build Listen.

Of course, another breadboard method is to start with a circuit board that is similar to what you want to end up with. You can then build the extra circuits in the air above and use the existing pads as tie-ins and supports. You can split traces with a sharp knife to add or change the circuitry. Sometimes you can take an existing card and add a small breadboard to it for just the experimental circuitry.

Many people will design a circuit, test it with simulation, and then lay it out as a PC board. I have found that simulation is just that. It is not the actual operation, and there are usually small differences—and sometimes gigantic ones—that exist in the real world. That is why I like a quick test bed. Even with whiz-bang computer programs, the best simulator is still a soldering iron. aX

New from **Dynakit**, the Mark III is a 60W mono amplifier kit featuring a non-magnetic "mirror polished" chassis, and a 600V (curve) rated quad capacitor of 80/40/30/20µF. The Mark III is also offered with an optional cover as well as either new production or NOS tubes. For more information, please visit www.dynakitparts.com.

TEAC America Inc.'s new hi-definition earphones, the Audio ZE-1000, feature patented 5-layer silicone ear pads and dual hi-definition balanced armature drivers. Four integrated micro speakers are included with the inCores, and there is an airtight seal within the ear canals. The listener is no longer forced to maximize volume, where ambient noise is high, keeping ears safe in the long run. The inCore Audio ear pads come in three sizes, and have 109dB sensitivity, 15Ω impedance, and 20Hz-20kHz frequency response. For additional information, contact www.teac.com/DSPD or mnguyen@teac.com.

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Mouser Electronics signed 64 suppliers for the year 2007, representing all the major electronic product categories including semiconductors, optoelectronics, interconnects, and more. Some of the newer companies added to Mouser's lineup were SkyeTek, Avago Technologies, Cypress Semiconductor, Delphi, and Pericom. To learn more, go to www.mouser.com.

As the newest additions to **BG Radia**'s Z-Series, the Z-92 and LCR Z-62 feature an advanced three-way design with unique coaxially mounted mid and high frequency planar ribbon drivers. The Z-92 is 44.25″ high, with its Neo 3PDR ribbon tweeter mounted in front of the Neo10 driver. The Z-62 is 7.7″ high, and both drivers employ two 6.5″ aluminum cone woofers with rubber surrounds. Contact Peter@hoagland.us for further information.

CAIG Laboratories' new DeoxIT Wipes are solvent and VOC free, RoHS compliant, and come pre-saturated and individually packaged. They have a unique fast-acting deoxidizing solution that cleans, enhances, lubricates, and protects electrical conductivity on all metal connectors and contacts. Deoxit, which acts as a lubricant and assists with difficult or tight connects, dissolves the oxidation and sulfides along with dirt, dust, and grime as well as protecting against these foreign elements from reattaching to the metal surfaces. For more, please go to www.deoxit.com.

The CRM-2BIP on-wall bi-pole speaker, from **Sunfire Corp.**, contains a dual side-firing ribbon configuration which produces wide dispersion for a huge rear sound stage. Also featuring a compact 4.5 high back-emf woofer, the CRM-2BIP handles 400W power and uses Intelligent Tweeter Shaping, which tailors the speaker's high frequencies to the listener's tastes and/or the acoustics of the room. For additional information, contact Peter@hoagland.us or www.sunfire.com.

Monster Cable's new Cable-It is a cable hiding solution that lets home theater owners finish their system off in style. From small to large sizes, Cable-It can hold between 3 and 13 cables, ranging up to 50′. If you'd like to learn more, contact Mike Schroeder at (212) 388-1400.

The **Cerwin Vega** Passive Series of professional non-powered subwoofers consist of the CVP-121 and CVP-118, capable of 4kW peak and 1400W peak, respectively. The CVP-121 features 32Hz frequency response, and a cast aluminum grille that functions as a heatsink for the sub's driver, while the CVP-118 has a maximum SPL of 131dB, featuring a singular pole cap that can accommodate up to two top boxes for maximum configurability. For additional information, please visit www.cerwin-vega.com.

From **AudioQuest**, the ITC System's new gold-plated connectors feature a captive center pin that securely grips onto the center conductor using a unique spring-tension system for maximum contact and conductivity. Available in F, BNC, and RCA connector configurations for 18-AWG, 20-AWG, and 24-AWG coax cable, Instant Tool-less Connectors are easily reusable. More information is available at www.itcplug.com.

Snell Acoustics released the Model IC C7 In-Cabinet LCR and IW C7 in-wall LCR THX Ultra 2 Certified Loudspeakers, featuring the D'Appolito Array. Both feature 1″ SEAS silk dome tweeters, two 4.5″ SEAS treated paper midrange drivers, and two 8″ copolymer cone woofers for dynamic bass response. The IC C7 is an acoustic suspension design, and the IW C7 has a solid aluminum, acoustic suspension enclosure with full bracing and asphalt damping, and a perforated steel grille. For more, go to www.snellacoustics.com.

SOUND ADVICE

I picked up the Jan. '08 issue of *aX* and wanted to let you know that the article on tweaks ("Of Tweaks, Mods, and Evidence," p. 16) was the most interesting in a long time. Could you ask Bill Waslo what sound cards he would recommend? My computer doesn't have one and I'd like to get one, especially for this purpose.

Joe Wdowiak
flyfishonly1@hotmail.com

Bill Waslo responds:

There have been a number of soundcards that have worked well for this kind of testing. My favorite is called the ESI "Juli@," which is a pro level (but reasonably priced) 24bit/192kHz sampling PCI card. A PCI card like this would only do with a desktop computer, though. There is an advantage to using a portable battery-operated laptop computer instead to do the tests, in that you can avoid any hum-inducing ground loops that might otherwise result from the AC power line grounds.

If you want to use DiffMaker with a laptop computer, I've had good luck with the EMU 0202 USB, a USB2.0 type 24bit/192kHz card, also reasonably priced. Another one that is downright cheap to buy, if you can find it, is the "Soundblaster Audio ZS Notebook," a PC Card (PCMCIA) device that can sample at 24bit up to 96kHz. I would use the ZS Notebook only at the 96kHz or 24kHz sample rates, however, because the other rates are provided only by using a somewhat signal-degrading form of "sample rate conversion." If you have only USB1.0 on your laptop, consider an M-Audio "Transit USB," which does well at 24bit/48kHz.

The card must, at bare minimum, provide stereo line-level recording inputs (most laptop computers offer only a mono microphone input for recording) and line-level playback outputs. Some built-in sound functions in desktop computers have even worked well (one particularly good one I found is one that used the Analog Devices "SoundMAX" chipset).

Diffmaker itself can be used to do a particularly sensitive test of soundcard quality. Just play a high quality short WAV file out of the soundcard's line outputs and cable it back into the soundcard's line inputs and re-record it at the same sample rate. That gives you two WAV files of the same thing, one being the original and the other being one that has passed through both the A/D and the D/A conversion of the soundcard. Then use DiffMaker to extract the difference between these two files. Doing this test of a perfect card would leave silence. I haven't found one that leaves a silent difference result yet, but the Juli@ card comes close, leaving a weakly audible bassy sound, probably because of low frequency phase shifts in its AC-coupled analog circuitry.

ARRAY TESTING

In reference to Mr. Fitzmaurice's article, "The CurveArray" in the January 2008 issue of *audioXpress* (p. 8), he did not report any listening observations about the system, except that it has higher sensitivity and has a point source characteristic. However, the point source claim is not completely accurate. The apparent sound image changes with listening position. To confirm this, I constructed a 33″ line of closely spaced drivers to see what effects could be heard. The listening tests were made at a 10′ distance, somewhat like the distance used for home theater.

As expected, the straight line configuration has a strong output when the listener is opposite the center of the system. The image also appears to be at the center, but as the listening position moves to the left or right, the image appears to follow the listener to the left or right. However, this effect is almost entirely negated by the drastic loss of output at the mid and high frequenices as the listener approaches the ends of the line.

In the convex arrangement where the listener is directly opposite the center, the image again appears to be at the center, but the listening level is reduced. As the listening position moves left or right, the image again tends to follow the listener left or right, but only moderately so. The response sounds very uniform from left to right, even when the curvature approaches a semicircle.

In the concave orientation, the image again seems to be at the center and is slightly higher in level than the convex orientation. However, the image, surprisingly, is reversed as the listener moves from side to side. When the listener moves to the left, the image appears to move to the extreme right, and so on.

Although the response measurements supplied in the article indicate benefits in the concave arrangement, they do not account for how we hear and the shifting of the sound image with respect to the listening position. Monophonic voice in the center channel will not appear to be at the center unless you are seated exactly in the center. There is also the question about amplitude and phase response of a phantom image reported in my last article on hearing[1].

It would seem that a single driver in the center would be a stable point source that will eliminate the image from shifting and also retain the coherence of the original sound. Accuracy could be preserved and the image would remain in the direction of the driver regardless of the listener position.

Roger Russell
rogerr4@earthlink.net

1. "Sounds and Hearing, Part 1," Roger Russell, *audioXpress*, October 2007, p. 28.

Bill Fitzmaurice responds:

First, a horizontal line array should only be used when you need the sensitivity that an array provides, and placement options won't allow a vertical array. If you can use a point source center channel, that's all well and good. But when the rest of your HT speakers are high-sensitivity, your center channel must be as well. While an Altec A7 or a Fostex or Lowther loaded backhorn center is seldom a viable option, a small line array often will be.

That being said, Roger's observations are correct. With a concave array the audio image will seem to emanate from whichever driver set is closest to on-axis with respect to the listener, so it will shift to the right when the listener is to the left, and vice versa. In theory this may seem problematic, in practice it isn't.

To understand why, listen to a TV with eyes closed and try to find the audio source. Most mono TVs have the internal speaker on one side of the screen or the other, and it's easily discernible which with eyes closed, as the only information your brain is processing is the audio triangulation provided by your ears. Now open your eyes, watch the TV, and try to pinpoint the audio source again. It will seem to come from the center of the screen, irrespective of the actual location of the speaker. The reason? Your brain gives priority to visual stimuli over audio, and it shifts the perceived sound source from where your ears say it's coming from to where your eyes are telling you it's coming from.

My own HT is probably a worst-case scenario, with my CurveArray located 3′ below and 20″ to the right of the center of the 42″ screen. With my eyes closed, the center channel is clearly sourced nearly 4′ from the center of the screen. With my eyes open, the source is dead center. Roger's testing would have had a different result had it been done in conjunction with a visual source rather than audio only.

Regarding a convex versus concave array, a concave array suffers far less from comb-filtering and lobed response than convex. Because the imaging superiority of the convex array is rendered moot by our visually dominant brain, the real-world advantage belongs to the concave alignment.

AUDIO LEMMINGS

After reading the exchange between Steve Nugent and Dennis Colin in the January 2008 "Xpress mail" (p. 47), I could only shake my head in amazement. Here it is the end of 2007, and there are so many people out there who still don't get it. I am afraid that the audio community is doomed to live in purgatory for another few millennia before the question of blind testing is resolved. I'll start by saying that I am in the camp of Mr. Colin as far as blind testing goes. I also agree with Mr. Nugent that even with a well-run blind test of one component, you cannot say with certainty how another component will perform unless you also test it as well.

That said, Mr. Nugent is guilty of

using psychological pressure on the readers instead of valid testing to convince them that unless you have the best equipment or the best hearing you are relegated to the mass of unwashed who will never be able to participate in audio Nirvana. That is nothing but heaping, steaming, gurgling piles of brown stuff. It is truly a sad state of affairs in this industry that we still have people using scare tactics to try to make a point instead of using reliable testing. His inference that if you don't automatically hear the superiority of this cable or that resistor, you must be deficient is pure bunk.

Mr. Nugent uses the analogy of a Ferrari and a Geo to describe proof of performance of add-on products. The ability to go faster, a measurable result, is the proof. And how is this done? By measuring the velocity of the cars or who gets across the finish line first. The cars are put to a test and the results verified.

This is not to say that all things that affect your audio experience at any given time are measureable. Listening can be a very emotional experience. In fact, producing a very satisfying emotional experience has always been the goal of this hobby for me. However, that very fact demands that when determining the effect of any given component, the preconceived notions of what should happen during that experience must be removed. Note that I did not say that the emotions themselves should be removed, only the preconceived notions. The only way to reliably do that is with a blind test, preferably double-blind.

I will not state that there are no cables that will not make a sonic difference. I will state that I have tested many in double-blind tests and heard no differences. More important, I have run double-blind tests with a manufacturer who makes cables and speakers as well. This person heard no differences with his cables in my system and then blamed it on my speakers. We then repeated the same tests using his own speakers and he again was not able to hear any differences. Could there have been other factors that interfered with the results? Possibly, but the point is that with enough controlled listening tests, a degree of confidence about the results will emerge.

Doing uncontrolled listening only leads to susceptibility to listener bias about what should be heard. And all the classical condemnations of double-blind testing being too fast and producing too much pressure only apply to tests that are poorly run. I have seen tests in which an ABX box was left with a user for over one month in his house to use to test sonic differences of two amplifiers when used in his system. There was no rush and no pressure from other observers to achieve one result or another. Was that a lot of work? Yes, but the results were rock solid and very believable.

Finally, saying that certain parts are sonically superior because so many people are buying them that they are in short supply is truly ludicrous. An analogy would be that jumping over a cliff to your death is a good thing because all the lemmings do it. There are huge numbers of experiences in human history where people go in a particular direction because of peer pressure that may not result in positive results. So let's not be audio lemmings and be forced by peer pressure to believe that you must be able to hear differences in certain "golden" components lest you be deemed an inferior clod. Do yourself a favor and prove those conclusions in an unbiased way.

Tom Perazella
Sleepy Hollow, Ill.

SUB DRIVER GOALS

I would like to thank Tom Perazella for his inspiring article, "The Kiss Bass Project" (Sept. '07 *aX*, p. 21). As Mr. Perazella said, the amplifier he used in his project has been discontinued. If you use the substitute he recommends, you will need to change the speaker enclosure size. The physical dimensions of these amplifiers are different. The dimensions of the discontinued amp were 10 3/8″ W × 10 7/16″ H × 3¼″ D, with an enclosure cutout of 9″ W × 9¼″ H. The dimensions of the new amplifier are 9 13/16″ W × 9 13/16″ H × 5″ D, with an enclosure cutout of 8½″ × 8½″.

To keep the idea of using the plate amplifier as the back of the enclosure, you need to reduce the outside width and height from 10¾″ to 10″ and adjust the depth to meet the desired volume. Remember that the box volume displaced by the new amplifier is different not only because of its overall dimensions but because it has an enclosed back. The Vb of the Kiss Bass enclosure 0.537ft^3 is the total box volume of 0.792ft^3 less 0.0459ft^3 for the driver and 0.209ft^3 for the amplifier.

Mr. Perazella made no reference to damping in his article. By damping an enclosure, its size can typically be reduced but usually at a cost of slightly raising the F3. Was this ignored to keep it simple, or to keep the F3 as low as possible? Should adding some stuffing be a consideration?

Also, how was the testing done for the response graph? I assume the results shown include the room response. If Bass Box 6 Pro is used to predict the response for the Tang Band W8-740C driver in a 0.8ft^3 enclosure, the results are quite different. This design software predicts an F3 of only 65.3 with a low total system Q (Qtc) of 0.462.

Figure 1 predicts the cone displacement and amplitude response superimposed on the same chart. It confirms that there are indeed adequate output levels of 106dB. I would personally say that the design goals have been surpassed in this regard because I believe anything above 105dB is adequate for even a medium-size room. The right cursor shows the predicted F3 of 65.3Hz. The left cursor highlights that the Tang Band driver is capable of achieving these output levels without exceeding its Xmax of 12.0mm.

I would also like to review Mr. Perazella's goals for this subwoofer project. They were:

- Clean bass
- Predictable performance (not ultimate)
- Use a simple method to achieve results
- Use key driver parameters to tune enclosure
- Simple design (no vents, passive radiators, transmission lines, servos, and so on)
- Small design (I assume Mr. Perazella means 1ft^3 or less)
- Designed for small rooms
- Sealed box
- Modest bass requirements (reasonably deep but not very high SPL lev-

els)

- Cone displacement capable of frequency response and output level
- Fb of 40Hz
- F3 of 40Hz

I respond in this genre because I believe that many hi-fi enthusiasts share similar goals.

Mr. Perazella stated that it was three years ago when he was looking for the drivers for this project. The science of driver design has grown somewhat since then. Are there new drivers now available that are better qualified to meet or exceed these goals?

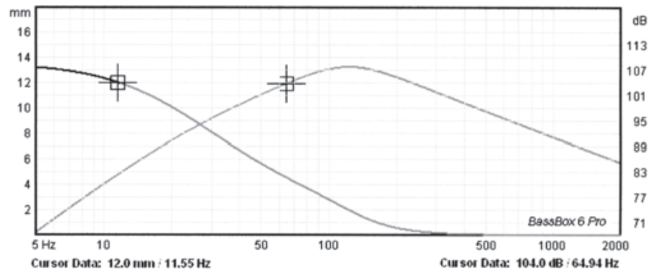


FIGURE 1: Predicted cone displacement and amplitude response.

Figure 2 is a XY Scatter chart, in which each data point represents a unique driver in an "optimum" volume, sealed and undamped enclosure. I have found a chart of this type very helpful in driver selection. The 128 drivers depicted are all available from one distributor and represent models less than 10″ in diameter from ten different manufacturers. The chart shows the volume and the predicted F3 for each speaker. The trendline substantiates the generalization that larger enclosures have lower F3s. The data point (0.2, 69) for the Tang Band W8-740C is highlighted. Upon close examination of this chart, it appears that another driver, also highlighted, could be considered. Drivers from other manufacturers may also be good candidates.

The driver that represents the data point (0.7, 50) is the Dayton SD215-88 shielded 8″ DVC subwoofer, available from Parts Express as part number 295-480. Consider a design with this driver using Mr. Perazella's design goals, and his "keep it simple" idea to use a plate amplifier for the back of the speaker enclosure. This design will combine the voice coils with parallel wiring. Although this driver has a lower power handling rating than the Tang Band W8-470C, it has a higher SPL rating. The result is a higher SPL at rated power of 108dB for the Dayton versus 105.8 for the Tang Band.

At a regular price of only $23, this driver becomes even more attractive. There is a catch, however. If you use the recommended box volume of 0.7ft^3 to achieve a Qtc of 0.707 and F3 of 50Hz, you find that the Dayton driver is not capable of producing the 108dB without exceeding its Xmax of 6mm. Nonlinear distortion increases when a driver exceeds its Xmax. This is because the magnet system will exert less control over the voice coil when fewer coils are within its gap. This problem is not serious for three reasons.

One, the design goals of this project do not require you to drive the speaker at full power. Two, the mechanical limit of the suspension or the distance traveled before the voice coil former contacts the back plate should be larger than Xmax. Three, the introduction of this type of nonlinear distortion is subtle and not audible unless severe.

Figure 3 shows how far the diaphragm and voice coil of the driver will travel with an input power level of 80W and with 36W. Notice that the plot line rises above the Xmax level below 58Hz with 80W of power and below 26Hz with 36W. **Figure 4** predicts the amplitude response and system impedance for the Dayton DVC superimposed on the same chart. It shows four things.

One, at 36W the design meets the output goal for a small room. Two, the design meets the F3 goal of 50Hz. Three, the left cursor shows the output is 12dB down at 26Hz providing enough protection for the driver ensuring Xmax is not exceeded. Four, the right cursor shows the box tuning at about 51Hz. **Figure 5** shows the predicted response of the Dayton DVC (black line) and the Kiss Bass (shaded line).

To help keep things in perspective, for those DIYers who are not restricted to size or simplicity, these drivers can produce even lower F3s if put in larger vented enclosures.

I am currently pursuing drivers from other distributors that will fit the bill. I will follow up with my findings. I would like to thank Tom Perazella once again for his inspiring article.

Tom Finn
Minneapolis, Minn.

Tom Perazella responds:
Many thanks to Mr. Finn for the kind words about my article. I have always tried to pick subjects that would provoke an interest in readers and it appears that I have succeeded in this case. I would like to reiterate that the purpose of this work was to show people who may have thought that speaker building was a complicated and time-consuming endeavor that in some cases it could be quite simple.

There is no doubt that building full-range speakers can be quite involved. However, subwoofers are good entry-level projects for many reasons including the fact that the long wavelengths involved minimize some of the stick problems that you face in mid-range and high-frequency designs. Simple sub projects can often outperform commercial designs for less money.

Mr. Finn is correct that if you want to retain the design using the subwoofer amplifier as the back wall, you would need to change the dimensions for amps having different dimensions. It is a simple matter to

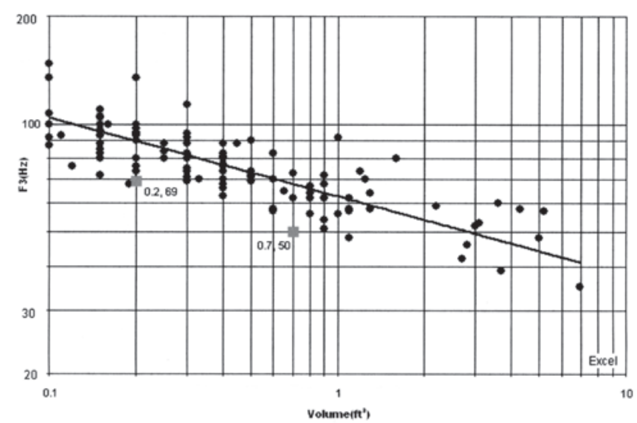


FIGURE 2: Driver responses.

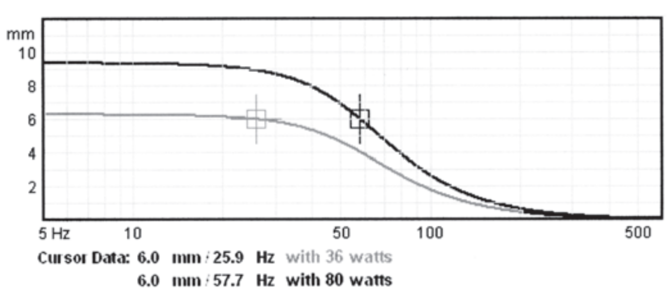


FIGURE 3: Predicted cone displacement.

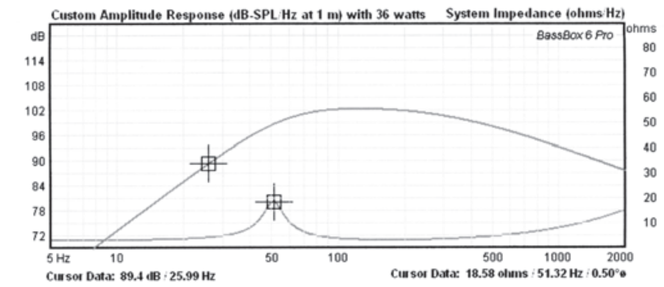


FIGURE 4: Predicted amplitude response and system impedance.

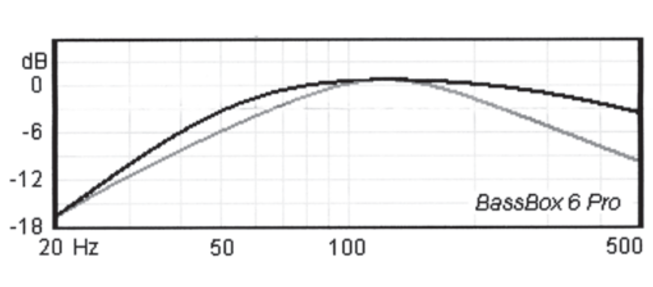


FIGURE 5: Normalized amplitude response (dB-SPL/Hz).

re-calculate the dimensions to reach the required volume. If the resulting shape is not to your liking, you can forgo the "amplifier is the back wall" method and use a more conventional design that results in a more acceptable configuration. That's the fun of doing it yourself.

The testing was done close miked, with a very small distance from the microphone to the center of the dust cap. In this configuration, room gain is minimized.

When it comes to choice of driver, you are truly blessed by the number of competent drivers available today. My choice of the drivers was determined by what was readily available having high Xmax. One of the most important characteristics of a good designer is to be a good librarian. Doing the research on what devices are available to meet the project's needs can result in a very satisfying project at a reasonable cost. Mr. Finn is correct that there are other drivers that would be satisfactory in this type of project. Again, making this choice is a large part of the satisfaction of designing a custom component.

I would like to clarify the discussion of Xmax by Mr. Finn. First, Xmax is not simply determined by the magnetic structure of the driver, but also the changes in compliance versus travel of the suspension. The two characteristics used to determine Xmax are Xmag, the behavior of the magnetic structure, and Xsus, the behavior of the suspension. **Figure 6** is a datasheet for a 15″dual voice coil Dayton driver that I use in a larger system. You can see the values for Xmag, Xsus, and Xmax. If the linear ranges of either the magnetic structure or the suspension are exceeded, Xmax will be limited. I have seen drivers that were limited by either Xmag or Xsus.

It is not just the voice coil former striking the back plate that can cause a limit to acceptable performance caused by the suspension, although if that does happen you will certainly have a very sudden and audible problem with the output. Even when exceeding Xsus limits, you are not necessarily going to strike the back plate, although the distortion will rise dramatically from suspension nonlinearities.

With regard to the audibility of the distortion mentioned by Mr. Finn in point three, I partially agree. Certainly a small amount of distortion caused by mild excursions beyond Xmax are less noticeable at lower frequencies. However, looking at the graphs, it is clear that once you get beyond the Xmag or Xsus points for this particular driver, the departure from linearity progress at much faster rates, translating to a very fast rise in high levels of distortion that will be audible. The linearity change in this driver, although fast, is slower than other drivers I have seen, especially those with under hung voice coils. In a nutshell, the performance of the driver goes south very quickly when Xmax is exceeded.

Over the years, I have become an Xmax junkie. It used to be that Xmax was at a premium, especially when amplifier power was low and the requirements for reasonable output levels demanded that a short voice coil in a short magnetic gap be used to provide high sensitivity. Now, driver designers routinely substitute sensitivity for higher linear excursion because of the availability of plentiful low-cost clean amplifier power.

In this particular design, although the 8″ Dayton driver mentioned will provide an acceptable level of output before exceeding Xmax, the Tang Band driver will provide double the Xmax for only $15 more based on its sale price. The Dayton driver will also run out of linear excursion before the amplifier runs out of power, not a good match. Considering the cost of the amplifier, wood, and sundries, the driver cost increase is insignificant for double the displacement.

When I think about speaker designs and linear displacement, I always remember the statement by Luigi Chinetti, who drove for and managed Ferrari USA, that went something like this: "The only substitute for cubic inches is rectangular money." To try to get extra performance from a driver with limited volume displacement requires some of the techniques such as servos that were not explored here because of cost considerations and complexity. Even when those extra steps are taken, there is only so much additional linear output that you can achieve when Xmax is limited. Given one driver with "X" amount of Xmax versus another in the same cost neighborhood with "2X" Xmax, all else being equal, I'll take the

Thiele-Small Parameters @ x=0

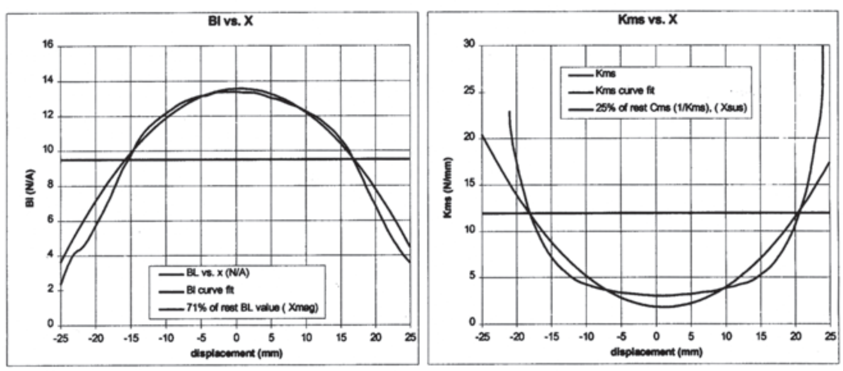
Fs	19.76 Hz
Qe	0.3715
Qm	7.2576
Qts	0.3534
Vas	281.4106 ltr.
Sd	0.0774 m²
Re	2.7619 ohm

Electro-Mechanical Parameters @ x=0

Mmd	181.71 g	
Cms	0.0003 m/N	0.334419 mm/N
Rms	3.3177 ohmm	
Sd	0.0774 m²	effective radius
Bl	13.3796 N/A	156.97 mm
Re	2.76 ohm	

X Parameters from Curve Fit

Xmag	16.34	Xsus	19.29	Xmax	16.34
Mag Center	0.55203	Sus Center	1.1116		
Bl	13.5637	K	1.8054		
Bl x coef A	0.00124	K x coef A	-0.0337		
BL x^2 coef B	-0.00112	K x^2 coef B	0.0152		



manual Bl fit
-17 mm to 18 mm

manual Kms fit
-20 mm to 23 mm

FIGURE 6: Dayton driver specs.

CONTRIBUTORS

Edward T. Dell, Jr. (Editorial—"Magazines: A Primer," p. 6) is editor/publisher of *audioXpress*, *Voice Coil*, and *Multi Media Manufacturer* magazines.

Bruce Brown ("Updating A Classic Pilot Amp," p. 8) is a registered pharmacist who works in the medical research area for a major pharmaceutical company. He has been experimenting with electronics for over 35 years, remaining actively interested in electronics, building kits, and "home brew" audio. Online auctions have stimulated new interests. He has just recently become interested in restoring vintage audio equipment and writing articles to assist other hobbyists. His wife, Kristin, has been invaluable help with editing his ramblings. He currently uses many restored and homebuilt audio amplifiers and welcomes communication about vintage equipment and restorations. He can be reached at Tuninfork@aol.com.

Ethan Winer ("Everything You Wanted to Know About SPL Meters," p. 14) has been a professional musician, composer, circuit designer, recording engineer, audio instructor, computer programmer, technical writer, and consultant since the 1960s. He was a contributing editor for *PC Magazine* for many years and has written more than 70 feature articles. Ethan now designs acoustic treatment products and runs RealTraps, in New Milford, Conn. Contact him at www.realtraps.com.

Bjørn Kolbrek ("Horn Theory: An Introduction, Part 2," p. 20) is an electronics engineer in Norway, and works as an R&D engineer for Scana Mar-El designing remote control systems for ships. His hobbies include building horn loudspeakers, amplifiers, and amateur radio gear. For the last four and a half years, he has studied horn loudspeaker technology—both theoretical—and practical—with Thomas Dunker.

Jan Didden ("*aX* Visits Burning Amp '07," p. 30) lives in the Netherlands and is a longtime contributor to Audio Amateur publications.

Ed Simon ("The Secret Art of Breadboarding," p. 34) received his B.S.E.E. at Carnegie-Mellon University. He has installed over 500 sound systems at venues including Jacob's Field, Cleveland, Ohio; MCI Center, Washington D.C.; Museum of Modern Art Restaurants, New York; The Coliseum, Nashville, Tenn.; The Forum, Los Angeles; Fisher Cats Stadium, Manchester, N.H.

higher Xmax driver every day.

It was very rewarding to read the excellent response from Mr. Finn and I am looking forward to the results of his searches for other high-quality drivers. This level of competence and dedication to the hobby bodes well for its future. Again, thanks to Mr. Finn and happy building.

SERVO PROJECT

I read with interest the DVC Servo Subwoofer article by Daniel Ferguson (Nov. '03 and Sept. '04). The article tweaked my interest, as I had been working along similar lines using 10″ DVC subwoofers for audio systems for my two "student sons," who have limited space and, of necessity, limited power (dorm rooms). However, I had shelved the project due to lack of inexpensive bookshelf speakers of adequate quality (they're very picky with regard to music quality!).

Now the time has come to revisit the project and I want to thank Daniel Ferguson for doing the bulk of the work for me. However, I have a couple of questions regarding the servo subwoofer filter schematic (Fig. 3, Sept. '04) and the differentiator.

1. For the HP Sallen & Keys filter sections (Op amp B of Fig. 3), from the data given in the article (R1 = 17.2kΩ and R2 = 70kΩ) the calculated turnover frequency is 20.8Hz. The Fs for the 12″ Dayton DVC woofer used is 20.7Hz. Is this a design point to tame the woofer resonance? Happenstance? Or were these values arrived at empirically?

2. If you model the response of the same HP filter, the characteristic response has a slight hump at the turnover point. I am of the opinion that a Bessel (linear phase) response might be a better design. Has Daniel Ferguson any comments on this?

Ian Freeman
Fremont, Calif.

Daniel Ferguson responds:

Thanks for taking the time to read my article on the DVC servo subwoofer. I'm planning to retire in the next few months and plan to revisit this project in an attempt to make the design more theoretically correct. Anyway, on to your questions.

Regarding the relationship between the filter corner frequency and the woofer's free-air resonance frequency, there isn't any. Furthermore, there is no connection between the filter bandpass shape and the driver/box combination. The filter merely provides the reference response shape that you want the servo to reproduce.

The box in the article was intentionally undersized to produce a strong resonance peak for the purpose of demonstrating the error correction capability of the servo—how well it could change its uncorrected response shape to conform to the filter's shape. In fact, it is theoretically much better to start any servo design with the best open-loop response possible so that the servo must "work" the least amount possible. In this case, the box should have been much larger to get closer to a closed-box Q of 0.7. Then, the system should be more stable and capable of greater dynamic output with less amplifier power.

Concerning the actual trimpot settings, they were as close as I could get these single-turn pots to make the filter have a corner frequency of 20.0Hz with a Q of 0.707. The shape was verified by my relatively expensive RTA to be accurate to within ±½dB. But again, the choice for the filter shape is somewhat arbitrary. You could easily choose to not be that ambitious and set the corner frequency at 25 or 30Hz and make the system more stable.

As to whether to choose a Bessel response over a Butterworth response for a closed-box woofer system, I think that's a matter of individual preference. One of the attributes of the servo is that you can make the reference shape anything you wish. So if you prefer a Bessel response to a Butterworth, you're free to set the pots for a Q that will approximate that.

The only thing that occurs to me is that the debate over which of these two is better is based on open-loop characteristics of loudspeaker dynamics, and I'm not sure it applies in a servo environment. All I think you would get from a non-flat tuning is more response droop at the low-end. Having said that, it would cause the servo to work less and therefore increase system stability.

I hope all this sheds some light on the system tuning. aX