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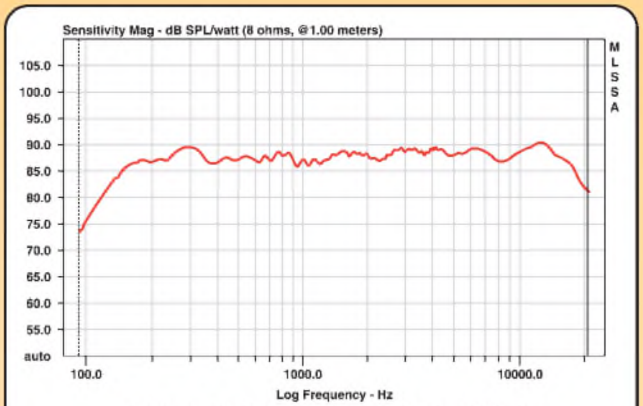
RD-40	264-706	40"
RD-50	264-702	50"
RD-75	264-700	75"



Representative Frequency Response Graph of RD Series

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This unique planar element was specifically designed as a replacement for the Carver planar element. The 48" ribbon was used in the Carver Amazing loudspeakers as well as in several Genesis loudspeaker models. Instructions and necessary hardware are included to make direct drop-in replacement possible.
Mfg. Part # RD48
PE Part # 264-704

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The Neo series planar transducers feature Bohlender Graebener's latest technology in a compact and inexpensive package. The Neo series drivers differ from conventional dome tweeters in that there is virtually nothing between the electrical signal and the sound- just an almost weightless diaphragm. No wonder the Neo series drivers deliver clean, airy, transparent sound that is inherently natural and musically pleasing. The Neo technology is available in a 3" model in raw dipole form, a 3" model with faceplate, and an 8" dipole unit. All Neo models have a 4 ohm impedance and a sensitivity of up to 96dB at 2.83V/1m.


Neo3 Neo3FP Neo8

PE Part #	Mfg #	Power Handling	Frequency Response	Dimensions (L x W x D)
264-718	Neo3	20W	2,000-20,000	3-1/2" x 2-5/8" x 1/2"
264-720	Neo3FP	20W	2,000-20,000	5-1/8" diameter x 3/4" deep
264-712	Neo8	50W	200-20,000	7-7/8" x 3-1/2" x 1/2"

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"What One Wants"—An Appreciation of Henry Kloss

By John Milder

Henry Kloss, whose death in February was both expected and hard to accept, was the first person I ever heard use the term "appropriate technology," back in the 1960s. He wasn't talking, though, about bringing solar power to the Third World, or designing a methane-powered car, or any of the other buzzy uses of the term. What Henry cared about—was passionate about with an intensity that could be withering—were products doing what they *should*. Doing what would really *satisfy* someone. Doing what you really *wanted*. Or doing what really *could* be done with a given product or technological advance or overlooked piece of technology.

But Henry never actually talked about what he, I, we, or you wanted. He was definitely not at home with personal pronouns. It was always what "*one*" wanted, needed, and so on—as in: "What one really wants here is...". Henry used that genderless, impersonal "one" constantly, usually as austerely as possible. It always gave some added techno-weight (and occasionally a touch of the "royal we") to whatever he was advocating.

Henry loved to advocate, especially for something that he believed everyone else was overlooking. Or for some simpler, more direct way to do what other people were overdoing. And he advocated for products by actually making them.

PROTOTYPICAL INVENTOR

The basics about Henry have been pretty well covered in various obituaries, from the succession of audio/video companies with which he was involved—Acoustic Research, KLH, Advent, Kloss Video, Cambridge SoundWorks, Tivoli—through his many "firsts," to a sartorial style that said "inventor" as soon as you met him. And yes, he was in a way that prototypical inventor, waking up rumpled every morning, proceeding to a cluttered office/lab with an unbelievably piled-up desktop (in and on which he could indeed find anything right away), and focusing in a very, very directed way on "the next thing," whatever it was.

But there was much more to it than that. His chinos-and-oxford-shirt style, complete with rolled-up sleeves, was also pragmatic to the core. He could do anything—including go out to a fancy lunch with investors, lawyers, or media types—on a moment's notice. Just let him whip out the old utilitarian tweed jacket and the plain black tie that was always somewhere around his desk—and off he could go to the Ritz.

And his mindset was anything but the abstracted tinkerer's. He was engaged in the life of the community, interested in socio-political developments of all kinds, widely read and definitely "cultured" in the best sense, a terse but very funny conversationalist. He also loved music, almost always classical. He was a weekly regular at Boston Symphony concerts. He wasn't much for pop, rock, or jazz, and he hated electronic instruments on recordings and audio equipment. ("There's no standard of reality to work from," he insisted. "Who knows what these things are *supposed* to sound like? One will never be able to tell what a speaker really is like from that stuff.")

That was indeed Henry. But it doesn't begin to cover the man who could regard you with clenched jaw and palpable mistrust one moment as you tried to explicate something, and who could then say: "Well, *yes*" with great grace and appreciation when you made yourself clear.

Nor does it indicate the incredible standard to which his combination of shyness, assertiveness, and rigor held those who worked with him. Or the fun that could be involved in that, and the tremendous stimulation. This appreciation of Henry can't do him justice either, but I hope it can fill in some of the cracks for those who didn't know him and those who did.

Henry Kloss (Photo by Edward T. Dell, Jr., 1972).

IRREDUCIBLE MINIMUM

I went to work for Henry at KLH in 1964, about five years after I'd bought one of his early successes, the Model Six loudspeaker. I'd bought it because it sounded unusually musical to me. But what I didn't realize until I was KLH's advertising director, spending long hours with Henry and trying to accurately represent what he was doing, was that the Model Six was so likable because it was the first fully successful example of what he liked to do most—finding successively simpler, more direct ways to make something really satisfying to people.

Take "two-wayedness," of which the Model Six was the supreme example. (Nothing was more Henry than terms such as "two-wayedness" and "three-piecedness.") What had helped him succeed so well at using two-way speaker design in the Model Six was the first of a series of Henry-designed tweeters that could go way down into the midrange and still behave beautifully at high frequencies. To Henry, the added complexity of a third driver in a speaker system, and of a more complex crossover network, stood in the way of doing something really good that lots of

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ABOUT THE AUTHOR

John Milder was Director of Advertising and Product Definition at KLH from 1965 to 1970, and then Advent's ad agency through 1979.

The Process of Design.

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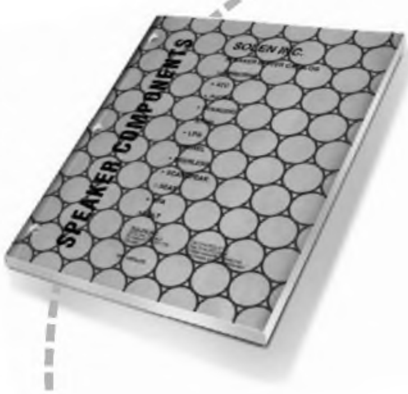

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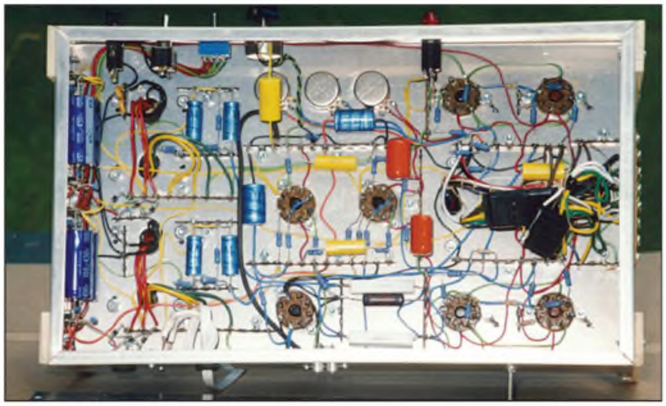

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.

JOHN STUART MILL

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Henry Kloss

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people could afford. The Model Six definitely was that product, very appropriate to the time. And it definitely made KLH's reputation.

Until I met Henry, I'd always associated the word "elegant" with "snazzy" or "luxurious" or what-have-you. But Henry used it in the strict, accurate, engineering sense. Elegance meant doing it exactly right, with nothing missing that should be there and not an ounce of excess fat that shouldn't be there.

The tweeter in the Model Six, with its amazing "elastomeric" suspension and its typically pragmatic, in-house-produced diaphragm, was the first in a series of amazing little drivers. The Model Eight and Twenty-One FM radios came out of small-driver design. So did the Advent Radio, and so, eventually, did the little Soundworks sub/sat systems and (just yesterday) the Tivoli radio. The little drivers helped Henry pursue his ideal: the irreducible minimum.

When I went to see him at Cambridge Soundworks and marveled at what the first Soundworks sub/sat system did, he immediately pointed to a still newer driver about half the size on his littered workbench. "One could do some interesting things with *that*," he said.

So one could. And today's Tivoli radio probably has taken the idea furthest, yielding the ultimate example of Henry's irreducible minimum. But basically, elegance and possibility were what always drove him, and when the two came together, he could act with amazing speed and directness.

PURSUIT OF ELEGANCE

One weekend in 1967, I took home a pair of records that *Stereo Review* had sent me to review for technical quality. They were the first two records made from master tapes that had been processed with the use of the Dolby S/N Stretcher, a mysterious and expensive black box that worked in four different segments of the frequency spectrum to reduce background noise on master tapes. One of these first two "Dolbyized" records was a Vox release of something or other, recorded with such a liberal helping of Vox's over-the-top reverberation that it didn't much matter what else might be going on in the recording.

But the other, Vanguard's recording of Stravinsky's *Histoire du Soldat*, was incredible! The absence of background noise—of all *kinds* of background noise—produced such crystalline clarity that the record sounded to me almost like an entirely new medium.

Monday morning I was in Henry's lab/office first thing to lay my newfound treasure on him. He listened for a few minutes in KLH's littered demo room, mumbled something inaudible, went back to his lab, and started making phone calls. Within an hour or so, he had tracked down Ray Dolby in London. And by the next afternoon, he was flying off to England to convince Dolby to make a new product.

While Henry was as impressed as I was by the Vanguard recording, what his mind had leapt to in those very first few minutes of listening was the idea of a single-band version of Dolby's complex studio system—one that would focus specifically on tape hiss by itself. The black box for recording studios was all well and good, but what Henry envisioned immediately was a consumer product, something that could go into a home tape deck to drastically improve the quality of consumer tape recording. His next-day trip to London was to talk Dolby into doing it—immediately!

The contrast between Ray's methodical, no-loose-ends approach to engineering and Henry's let's-get-on-with-it urgency must have made for a fun couple of days. But Henry got what he wanted and Ray got the beginning of the rest of his life.

The ensuing KLH Model 40 tape deck, which we introduced at the next year's New York Audio Show in a hair-raising comparison between one of Victor Campos's carefully selected 15-ips studio tapes and the 3¾-ips copy of it he had made on the Model 40, was the most amazing demo prop since the first acoustic-suspension loudspeaker. But by the time we were having fun wowing roomfuls of people with it, Henry's mind was long since on to the

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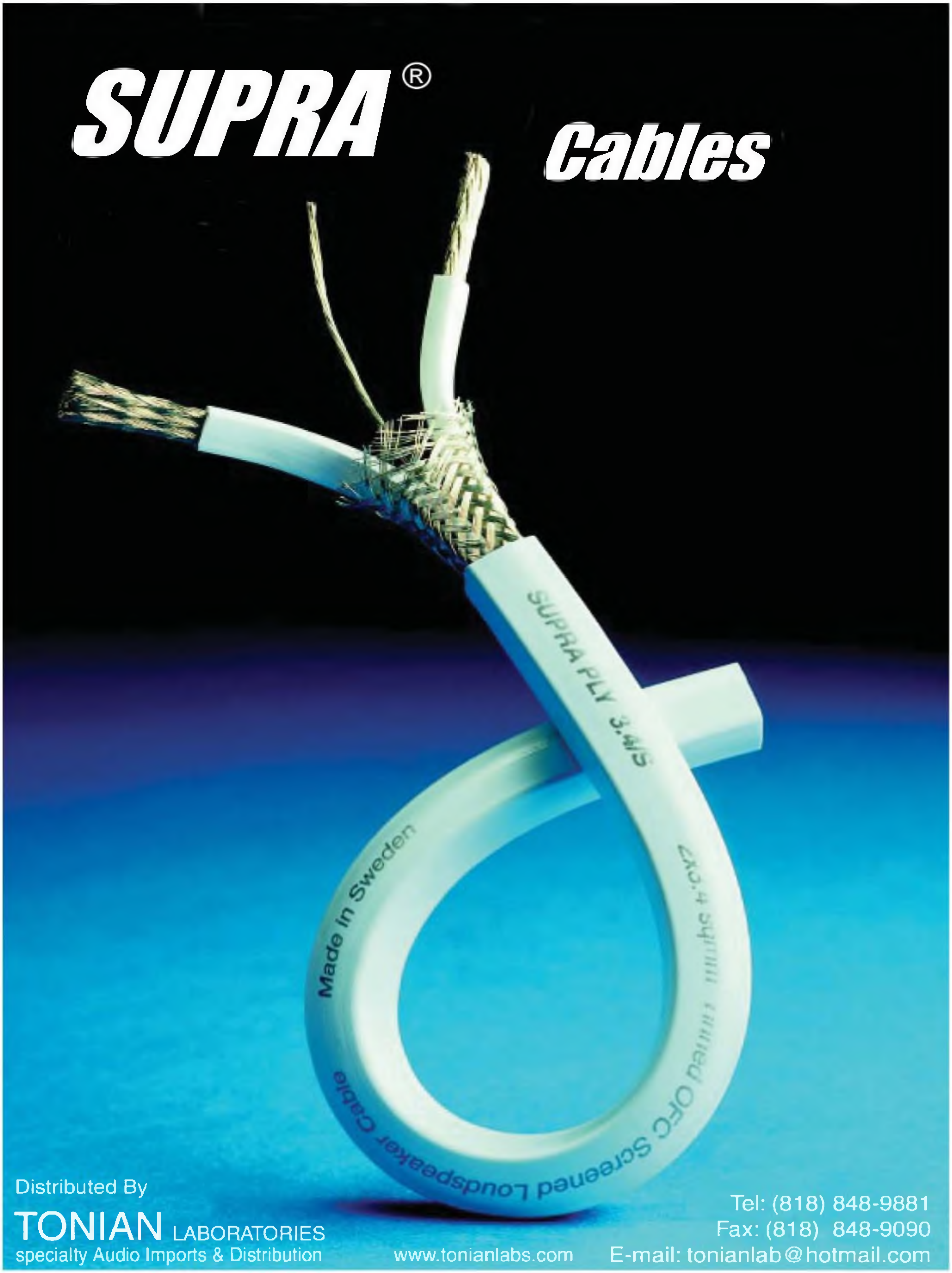

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THOR: A D'Appolito Transmission Line

With this exceptional design for some exceptional drivers by an exceptional designer, transmission-line ownership is well within your grasp.

By Joe D'Appolito

During CES 2000 I met with Norwegian and US representatives of SEAS. They described their new Excel line of drivers to me and asked whether I could design a flagship loudspeaker using these drivers that would highlight their extraordinary capabilities. They wanted something other than a vanilla sealed or vented box.

I suggested a transmission line, pointing out that the non-resonant behavior of this enclosure assured that we would hear the full capabilities of these drivers free of "boxy" coloration. The SEAS folks agreed enthusiastically, and the THOR transmission-line project was born.

THE THOR-EXCEL TRANSMISSION-LINE LOUDSPEAKER

Transmission-line (TL) loudspeakers have long enjoyed a small but dedicated following, especially in the DIY community. The advantages of TLs are well known. They are essentially non-resonant enclosures, producing a deep, well-controlled bass response. For a given driver, bass response can often extend well below that produced with either a vented or sealed enclosure using the same driver. Above a few hundred Hz, the line-filling material completely absorbs the driver back wave, giving the TL an open, non-boxy sound.

Unfortunately, the TL has not enjoyed wide commercial popularity due to the lack of a good design theory and the additional complexity of enclosure fabrication relative to the more conventional vented and sealed enclosures. Recently, however, work by G.L. Augspurger has appeared in the technical literature[1] and in *audioXpress*[2,3,4] that, while not providing a complete theory of design, has given us an excellent starting point. This, coupled with modern PC-based acoustic measurement systems, allowed me to converge quickly to an optimum design for the new Excel W18EX001 woofers.

The present design uses an MTM driver configuration in a tapered, folded line uniformly filled with Dacron pillow stuffing. Tapering the line greatly increases the frequency range of bass augmentation produced by the line. Using two mid-bass drivers exciting the line at slightly different points reduces mid-bass ripple.

The resulting line produces a uniform 3–4dB bass response lift from 110Hz all the way down to 20Hz with less than 1dB ripple. The –3dB point is 44Hz. Contrast this against 65Hz for a similarly damped sealed enclosure. Below 45Hz TL bass response falls off at 12dB per octave, compared to the 24dB/octave fall-off rate of a vented system. In most rooms useful bass response extends well down into the 30Hz region.

Above 2500Hz the system crosses over to an Excel T25CF002 tweeter. Several hundred hours of laboratory testing and listening have gone into producing a seamless transition between the mid-bass and tweeter drivers. You literally cannot tell where the woofers leave off and the tweeter begins.

The Excel product line from SEAS was introduced in 1994 as a showcase for the company's best ideas and technologies. Originally comprised of only five models, the Excel line has expanded to ten products, with additional designs in continuous development.

THE W18E001 WOOFER

Building a "better" mid/woofer required a complete rethinking of nearly every component in the driver: the cone, the magnet system, the surround, and the basket.

PHOTO 1: In the lab, ready to test and trim the line.

The Magnesium Cone

The advantages of metal cones are well known. They remain virtually pistonic throughout their passband, and do not suffer from midband cone edge resonance problems so common in paper and other soft cones. Prior to the development of the magnesium cone, virtually all metal cones used some form of aluminum alloy. While aluminum is an easy material to form either by stamping or spinning, it also suffers from its share of acoustic drawbacks.

To keep the moving mass reasonably low, the cone must be quite thin. For an 18cm woofer, the nominal thickness is approximately .18mm. This, unfortunately, results in a cone with numerous high Q breakup modes starting at about 5kHz and extending beyond 10kHz.

SEAS therefore decided to search for a material with a potential for greater stiffness than aluminum. Magnesium was attractive because its specific gravity was only 1.7 versus 2.7 for aluminum. This meant that, for the same cone mass, a magnesium diaphragm could contain almost 60% more material by volume than aluminum. This gave the potential for much greater stiffness and internal damping of the cone with no increase in mass. Acoustic testing of prototype magnesium cones immediately revealed the benefits over the aluminum cone: the breakup modes had been largely reduced to a single, well-defined peak that could easily be suppressed via simple notch filtering.

The question was how to produce the cone? Magnesium does not lend itself to bending or shaping in the thickness required for a loudspeaker cone. That left the only option of die-casting. Fortunately, a small magnesium foundry close to the factory was able to cast the rough cone. But getting to the finished cone would require that the remaining processes be developed in-house.

SEAS developed a special machining process to remove the precise amount of material necessary to shape the cone and achieve the proper mass. Through much experimentation, a cone of varying thickness between .26mm and .33mm was found to be the ideal solution.

All that remained was the finishing process to give the cone an attractive look and prevent it from corroding over time. For this, a chemical etching process was developed, followed by a coat of protective lacquer on the front and rear surface, giving the cone its unique appearance.

The Excel Motor

To gain the greatest advantage from the magnesium cone, an exceptional magnet system was required. The key design goals of the Excel motor were to: 1) Reduce the levels of eddy current distortion and flux modulation, thereby reducing harmonic and intermodulation distortion, 2) Stabilize the inductance of the voice coil under all excursion conditions to reduce modulation distortion, and 3) Improve the heat transfer from the coil and pole piece to the outside air to reduce voice-coil temperature and subsequent voltage sensitivity modulation.

These goals were accomplished by incorporating two heavy copper rings fitted above and below the magnet gap defined by a T-shaped pole piece, which was press-fit into a bumped back plate. To further enhance the heat transfer capability, a solid copper phase plug was fitted to the top of the upper ring. The stationary phase plug replaces a conventional dust cap and thereby eliminates the acoustic resonator behind the dust cap. At the same time, the excellent thermal conductivity of the phase plug aids tremendously in heat dissipation, while the air movement from the cone over the phase plug also serves to cool the motor.

The Excel Basket

A high-performance motor and cone should not be mechanically or acoustically limited by a less than optimal basket. For the W18E001, an entirely new state-of-the-art, die-cast zinc basket was developed. The casting is extremely stiff, maintaining precise alignment of all mechanical parts, and providing a stable and secure mounting surface for the cabinet.

At the same time the rear of the basket is designed to be as open as possible, using thin but strong "arms" that minimize early reflections at the rear of the cone. The area behind the spider is completely open as well, eliminating air compression and "chuffing noise."

Complete specifications for the Excel woofer are listed in *Table 1*.

TABLE 1
EXCEL WOOFER SPECIFICATIONS

Nominal impedance	8Ω	Voice coil resistance	6.1Ω
Frequency range	40–2500Hz	Voice coil inductance	0.4mH
Short-term maximum power	250W*	Bl factor	7.2N/A
Long-term maximum power	100W*	Free-air resonance	31Hz
Characteristic sensitivity	86.5dB SPL	Moving mass	15.5g
Voice-coil diameter	39mm	Suspension compliance	1.6mm/N
Voice-coil height	16mm	Suspension resistance	1.4Ns/m
Air gap height	6.0mm	Effective cone area	126cm^2
Linear coil travel (p-p)	10.0mm	V_{AS}	37.0 ltr
Magnetic gap flux density	0.88T	Q_{MS}	2.5
Magnet weight	0.42kg	Q_{ES}	0.39
Total weight	1.75kg	Q_{TS}	0.34

*IEC 268-5

WHICH TL GEOMETRY?

After describing the performance of straight TLs in the first two parts of his series for *Speaker Builder*, Augspurger details five alternate geometries in Part 3[4] that provide certain benefits over a straight pipe. Of these, two will be used in the THOR system, and the benefits of a third will be obtained by alternate means. The particular geometries are the tapered line, the offset driver line, and the coupling chamber line. The benefits of each are as follows.

1. Tapering the line broadens its fundamental resonance and thereby increases the frequency range of constructive pipe output. The f_3 value is typically 0.8 times f_p. Attenuation of upper harmonics is comparable to a straight line. Augspurger recommends tapers in the range of 3:1 to 4:1.

2. Offsetting the driver from the closed end of the line by one-fifth its length reduces the first passband dip, thus smoothing low-frequency response. However, f_3 must be set about 20% higher than f_p for the flattest response.

3. A coupling chamber between the driver and the pipe inlet lowers the fundamental frequency of the combination. The coupling chamber compliance combines with the resistive acoustic impedance of the damped line to produce a first-order low-pass filter that increases high-frequency attenuation.

It was clear to me at the outset that I wanted to use a tapered line to get a low f_3 with fairly broad low-frequency reinforcement. Additionally, with the MTM driver configuration one driver is automatically offset from the closed end of the line. This driver offset will mitigate somewhat against the low-frequency extension provided by the taper, but will help to reduce midbass response ripple.

Finally, folding the line provides additional high-frequency attenuation, somewhat like that obtained with the chambered line. With these considerations in mind, I describe the initial layout and sizing of the THOR TL.

TABLE 2
EXCEL TWEETER SPECIFICATIONS

Nominal impedance	6Ω	Voice coil resistance	4.7Ω
Frequency range	2–25kHz	Voice-coil inductance	0.05mH
Short-term maximum power	200W*	Bl factor	3.1N/A
Long-term maximum power	90W*	Free air resonance	500Hz
Characteristic sensitivity	88dB SPL	Moving mass	0.37g
Voice-coil diameter	26mm	Linear voice coil travel	1.0mm
Voice-coil height	1.5mm	Effective piston area	7cm^2
Air gap height	2.5mm	Magnet weight	29g
Magnetic gap flux density	0.88T	Total weight	0.36kg

*IEC 268-5 using 12dB/octave Butterworth high-pass filter at 2500Hz

SIZING THOR

A big advantage of sealed and vented box design is that Thiele and Small, among others, have established strict relationships between driver parameters and box volume and, in the case of vented designs, box tuning, for a specified frequency response. This greatly simplifies the design process for these systems.

TABLE 3

Design	Q_{TS}	f_3/f_S	f_S/f_p	V_{AS}/V_p
Tapered	0.33	1.6	0.50	1.00
(Nom. 4:1)	0.35	1.525	0.533	0.90
	0.41	1.3	0.63	0.60
Offset	0.33	1.6	0.74	1.00
Speaker	0.35	1.525	0.80	0.90
	0.41	1.3	0.94	0.60

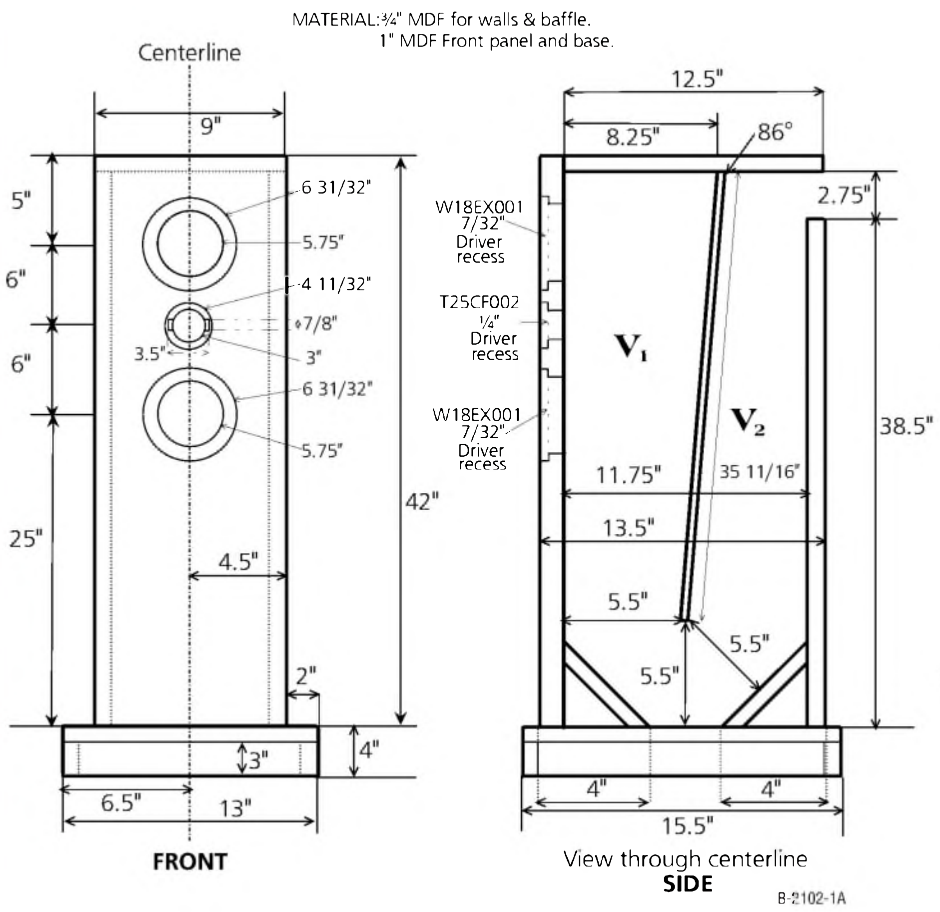

FIGURE 1A: Front baffle layout for the prototype transmission line and view through centerline.

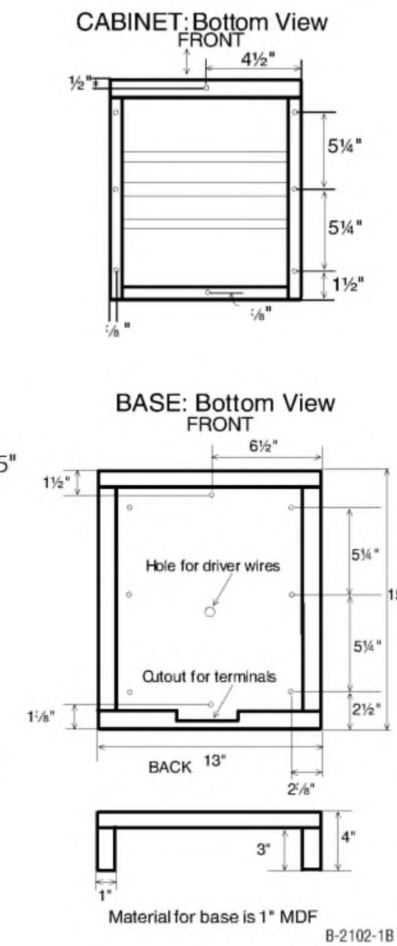

FIGURE 1B: Diagrams of bottom views of cabinet and base.

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The significance of Augspurger's work is that for the first time we now have relationships between the driver parameters f_S, Q_{TS}, V_{AS}, and the TL frequency, f_P, and volume, V_P. Strictly speaking, his relationships are not unique, but they do represent an excellent starting point and give us confidence that a good design can be attained.

The starting point for TL sizing is Augspurger's table of extended system alignments given in Part 3 of his series. Portions of that table are reproduced here in *Table 3*. These alignments are optimum in the sense that they approximate the response of an equal volume closed box, but with reduced cone ex-

EXCEL MILLENNIUM TWEETER

The history of dome tweeter development at SEAS has been a long and successful one, going back more than 30 years. SEAS' first dome tweeter was also one of its best known and most produced. This was the original type H087, 1½″ dome tweeter, used in the legendary Dynaco A25 loudspeaker. This landmark loudspeaker, manufactured by SEAS, was sold in the hundreds of thousands, and served as many a budding audiophile's introduction to true high-fidelity sound.

Producing the early dome tweeters was labor intensive, and considerable skill on the assembly line was required to produce a product with consistent quality. A sticky "doping compound," used to both seal and damp the dome diaphragm, was applied by hand after the tweeter was assembled. Obviously, the amount of the material and the evenness of application were critical to obtaining the desired frequency response.

Since those first designs, much research has been done to simplify and stabilize the process for producing soft domes. Other, noncloth materials, such as supronyl (polyamid) plastic, were successfully used as substitutes. But these, too, were far from ideal, because their performance was highly dependent on ambient humidity and temperature. By the late 1980s, promising new cloth materials were becoming available which allowed the cloth to be treated prior to forming the dome. In this way, the advantages of cloth could be realized without the need to coat the dome after assembly.

Today, SEAS manufactures all of its dome diaphragms in house, using special vacuum-forming equipment also designed and built by SEAS. Cloth diaphragms are produced from a proprietary material called "SONOTEX®." The SONOTEX process pre-coats the fabric four times with a damping/sealing material, giving a nearly ideal combination of acoustic performance and high consistency.

For the Millennium tweeter, SEAS designed a special two-piece diaphragm consisting of a SONOTEX dome with a SONOMAX® plastic surround. This combination results in a diaphragm with very linear behavior and large excursion capability.

THE HEXADYM® MAGNET SYSTEM

The ceramic-magnet-based magnet system found in most tweeters has remained basically unchanged for many years. The tweeter's magnet system performs two separate functions:

1. Supply the proper amount of magnetic flux to the voice coil
2. Allow the acoustic energy generated from the rear of the dome, the surround, and the voice coil to be fully absorbed within the tweeter body without reflections, resonance, or pressure build-up.

Ceramic magnet systems are easily able to supply the necessary magnetic energy. But they also get in the way of the rearward radiated energy. The construction of the system with a ring magnet covered by top and back plates produces a large cavity in the area between the pole piece and the inside of the magnet.

This cavity, when excited by the movement of the voice coil, produces resonance and pumping effects that will directly impact the performance of the tweeter. Another secondary cavity sits between the dome surround and the magnet system's top plate. With such an enclosed ceramic system, the energy build-up directly behind the vibrating surround cannot be vented away from the magnet system's top plate.

The new, patented Hexadym magnet system in the Excel Millennium tweeter completely eliminates any enclosed cavities within the tweeter structure (*Photos A* and *B*). Instead of a single ring magnet, the Hexadym system uses six radially magnetized neodymium bar magnets mounted on a hexagonal pole piece. This compact configuration produces large openings around the pole piece, allowing virtually all air movement generated by the diaphragm and voice coil to be vented directly into the rear chamber. The Hexadym magnet system also allows airflow produced directly behind the dome's surround to be vented into the rear chamber through four generous openings in the top plate.

The mechanical construction of the Millennium tweeter also reflects the no-compromise approach used in the dome and magnet assemblies. The front plate and rear chamber are constructed of extremely rigid, die-cast zinc. This provides a virtually non-resonant enclosure for the tweeter, while simultaneously conducting heat away from the magnet system. —*John Stone*

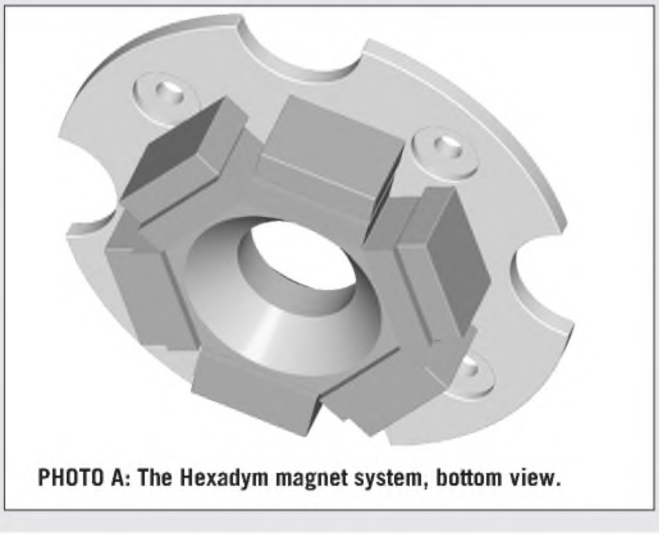

PHOTO A: The Hexadym magnet system, bottom view.

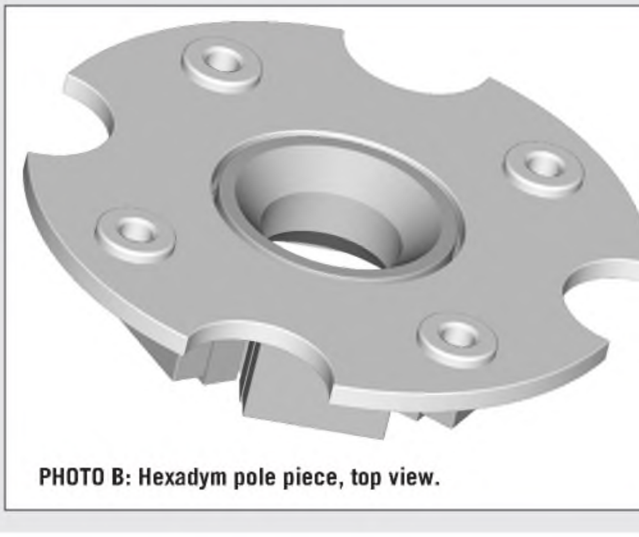

PHOTO B: Hexadym pole piece, top view.

cursion when the damping is adjusted for a ±1dB passband ripple. Fortunately, the TL is a non-resonant system so that the optima are broad. As you will shortly see, significant departure from the alignments Augspurger recommends can be made with little loss of performance.

The driver parameters needed to enter *Table 3* are f_S, Q_{TS}, and V_{AS}. Values of those parameters for the Excel W18E001 woofer are given in *Table 1*. The line $Q_{TS} = 0.35$ (which includes the effect of the crossover coil resistance on driver Q) in *Table 3* has been interpolated from the Q_{TS} values of 0.33 and 0.41 given in Augspurger's Table 2 of reference [4]. Notice that the columns headed f_3/f_S and V_{AS}/V_p are the same for both the tapered and offset speaker configurations for the same Q_{TS}. Thus f_3 and V_p will be the same for both, but the line lengths will differ.

From *Table 3*, the predicted f_3 is:

$$f_3 = 1.525 f_S = 1.525 \times 31\text{Hz} = 47.3\text{Hz}$$

For the tapered line, f_p is calculated to be:

$$f_{pT} = \frac{f_S}{0.533} = \frac{31}{0.533} = 58.2\text{Hz}$$

The line length, L_p, is then one quarter

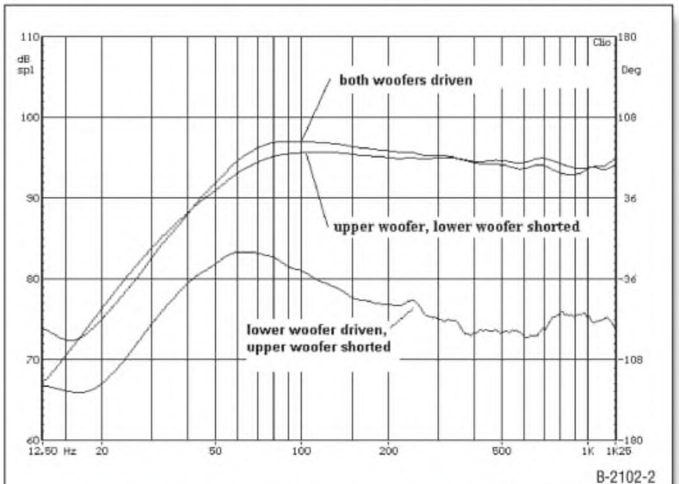

FIGURE 2: Near-field measurements on an MTM speaker.

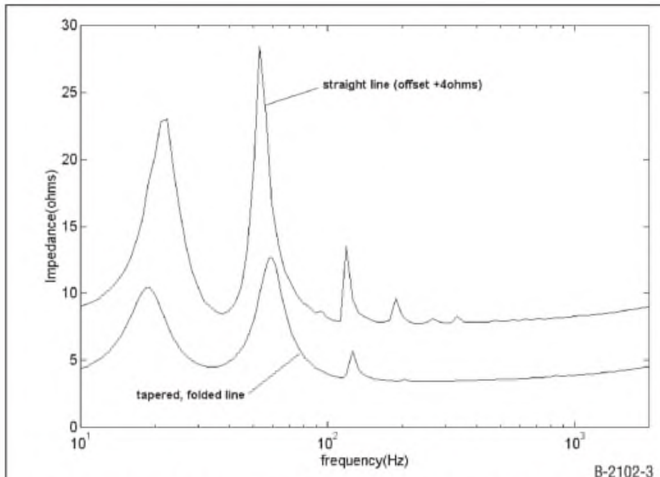

FIGURE 3: Unfilled transmission-line impedance comparison.

LISTENING CRITIQUE

BY DENNIS COLIN

Here is a loudspeaker that appears to be without audible coloration; that's how it sounded to me. Now that's quite a statement, deserving of intense scrutiny. After all, Joe D'Appolito designed these speakers, so of course I thought they *should* be excellent.

Jiddu Krishnamurti, the (less than should be) famous observer of the human condition, taught that thought itself is a corruption of free observation. You must instantly forget all the past—beliefs, memory, attitude, and so on, as you do when surprised by something new—in order to freely and completely *see* the new, the present.

I have no trouble doing this when listening to (not analyzing) music I like. When reviewing an audio component, I listen; if I hear a sonic anomaly I focus on it, then call on the analysis tool called thought to attempt a description, e.g., "3dB dip at 2.5kHz," "blown woofer," and so forth. But then I shut this off and just listen again. After each piece of music, I write down my impressions.

You may still question whether I have a subconscious desire to believe a Joe D'Appolito-designed speaker must be good, or a simple desire to please a master. To this I mention that I've also designed speakers, including one I want to be the world's best. So my biases include a competitive factor that would incline me to go the extra mile to find fault with anyone else's speaker, including Joe's.

But I was able to disregard all biases—positive and negative—as I sat down and just listened.

THE SOUND

Compared to any forward-only-facing speaker, I like the extra sense of ambience that a bipolar can provide, even if synthetically derived. To me, this can satisfactorily compensate the loss of original hall ambience in two- (and even 5.1) channel format limitations. This, however, I find true only in a highly damped room, such as my living room. In Joe's room, where I auditioned his speakers, there's more liveness (although very smooth), and I think a forward-only-facing speaker is the best. My impressions are:

1. Ambience. While desiring the presence of surround speakers (which Joe provided momentarily, resulting in astonishingly good 3-D ambience), this review is meant to be in pure two-channel stereo only. And as such, I heard an absolutely seamless, smooth, and deep stereo soundstage, even well off-axis. Not a hint of gaps, phasiness, or loss of tonal naturalness.

2. Tonal naturalness ("Presence" in the sonic ratings chart). I could simply hear no flaws, not even subtle colorations! The speakers are so acoustically transparent, though, that I had no trouble hearing many recording deficiencies, including those of the 16/44 CD medium. But mind you that free of the usual speaker anomalies, I thoroughly enjoyed the music.

One recording, even though a CD, is remarkably clean in detail and resolution: the Turtle Creek Chorale (I've mentioned its "goosebump factor" previously). On these speakers, I've *never heard more natural-sounding voice reproduction*, period. The performers sounded right there in front of me, yet the vocal fades into reverberation sounded infinitely far off. For a while I wanted to be a Tibetan monk so I could hear this all day!

3. Bass, midrange, treble, and balance. Sorry, nothing to comment on except personal perceptual flawlessness.

4. Bass. Now here's an area for comment: These are transmission-line (TL) speakers after all; TLs are supposed to have "different" bass. Before hearing these, I used to think "What's the big deal? TLs are just basically highly-damped open-back cabinets, aren't they?" Well, no!

First of all, the bass I heard *was* superbly damped; not a trace of "hangover" or emphasis. But the bass was also very deep and powerful. At one point, I was startled to hear a large bass drum impact shake me and the room down to at least 25Hz; as of this writing I don't know the speakers' f_3 or the bass room gain, but I do know I heard powerful and clean output to at least 25Hz, very surprising from a pair of 6½″ woofers per channel.

Second, the bass *quality* was even more impressive. For example, with Jacintha singing "Georgia on My Mind," the bass viol was the most natural and present sounding I've heard (not to mention Jacintha's voice and all else on the recording).

5. Transient Response and Image Clarity. A very good test is "Percussion Fantastic" (Fimco 017).

On these speakers, every detail of every percussion instrument was there with pristine immediacy and focus. With large tubular bells, for example, I clearly heard the subtle but lush midrange "knock" sound just before the blossoming resonance of the bell overtones, all spatially and temporally correct-sounding. From the deepest drums to what sounded like tiny (1″?) triangles (whose *fundamental* was probably 5kHz), the speakers never added any confusion.

6. Overall Impression. The speakers appear to reproduce whatever is fed them with flawless transparency. So well, they ruthlessly reveal any recording or medium deficiency. At one point, after criticizing some of Joe's recordings (which are much better than the average CD, mind you), I attempted to remove any sense of personal offense by saying "Joe, these speakers must be first-rate to reveal such fine details of recording imperfection!"

Now, how can you argue with that if you've designed the speakers? I felt like a politician making that statement, but nevertheless it's my true feeling.

7. One More Comment—To Sub or Not to Sub. Not! First, these speakers don't need any, unless you need response down to 5Hz. Second, I don't like separate sub (or any) woofers—I'm aware of a lack of coherence on well-recorded bass transients.

With these speakers, there was no audible separation or lack of coherence anywhere in the audio spectrum; the response in Joe's room sounded flat down to 25Hz. Of course, Joe *could* install eight 18″ woofers (per channel) into the walls and design it to extend the response flat to 3Hz at 130dB SPL.

OTHER SPECIFIC RECORDING IMPRESSIONS

A Chorus Line—Excellent including bass; opening percussion with great depth estimated to 25Hz (with possible room-gain contribution).

Carmen—Reproduced very clearly at peaks above 100dB SPL; speakers had no problem with the 200W/per channel amplifier probably driven near clipping.

Beethoven "Fastoral"—Tonality good, but recording seemed to have constricted ambience.

Fanfare for the Common Man—Very natural, deep, and spacious.

Chopin/Rubenstein—Hauntingly good music; piano recording technique sounded somewhat distant and dull. Not from speakers; other piano recordings could be first-rate.

Tannoy Hi-Fi Sound Sampler—One of the best recordings of string bass. Speakers delivered perfect-sounding tonality and space/time coherence. The overtone structure, instrument resonance, and sense of live string pulsiness sounded not just separately—good, but coherent live instruments.

Dvorak Symph. #9, Solti—Absolutely excellent in all regards.

BSO, von Stade—Very good instruments, voice somewhat overloaded in recording (present at any playback level, plus if it were the speakers, the instrument sound would have been intermodulated with the voice; it wasn't).

Dvorak Piano Quartet—No problem with *this* piano recording and reproduction thereof! One of the most analog-sounding (or more correctly, non-digital-sounding) CDs I've heard.

Up to this point, I hadn't seen any measurements. But now it's time to open the secret envelope. *(to page 16)*

of a wavelength at f_{pT}.

$$L_{pT} = \frac{c}{f_p} = \frac{13584}{4 \times 58.2} = 58.4\text{in}$$

In this equation, c, the speed of sound in air, is taken to be 13584 in/sec. A similar calculation for the offset line gives a line frequency of 39.2Hz and a line length of 86.6″. The total line length actually used is 81″. With this length an f_3 of 44Hz is achieved.

Column 4 of *Table 3* tells how to compute the TL internal volume as a function of V_{AS}. From *Table 3* you find that:

$$\frac{V_{AS}}{V_p} = 0.9$$

Now V_{AS} for the MTM configuration is twice the V_{AS} of a single driver or 74L. Solving for V_p you get:

$$V_p = \frac{V_{AS}}{0.9} = \frac{74}{0.9} = 82.2\text{L} = 2.905\text{ft}^3 = 5021\text{in}^3$$

You must establish at least two dimensions of the enclosure before you can use the volume calculation to get the final dimension. Here some practi-

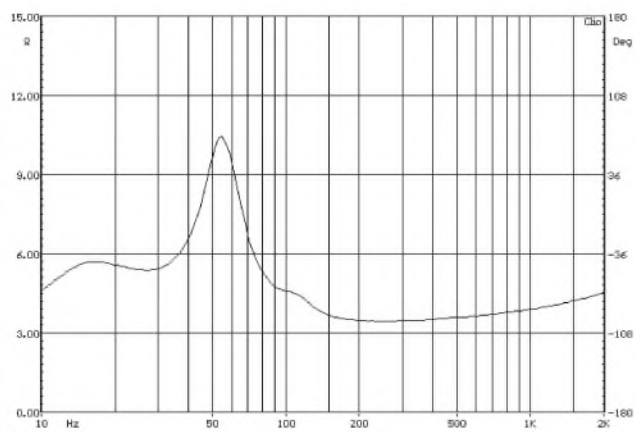

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FIGURE 4: TLine woofer impedance with 13 oz filling.

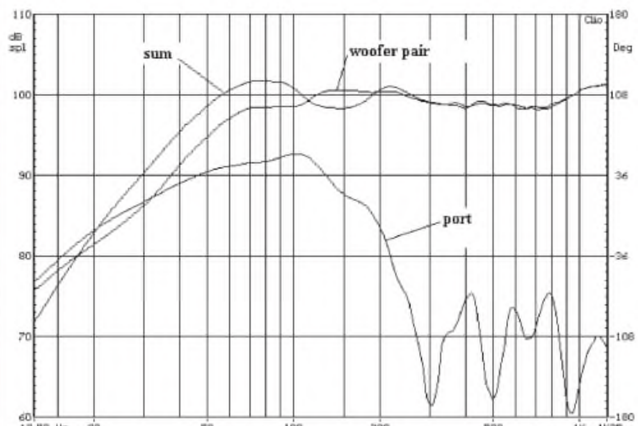

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FIGURE 5: TLine responses with 13 oz filling.

cal considerations entered.

I knew I wanted a tweeter height of 34–35″ to match ear height at my favorite listening position. I also knew from earlier experience with woofers of this size that a front baffle width should be no greater than 9″ for uniform horizontal polar response. Finally, I wanted to isolate the crossover from the main acoustic volume by placing it in the base of the enclosure.

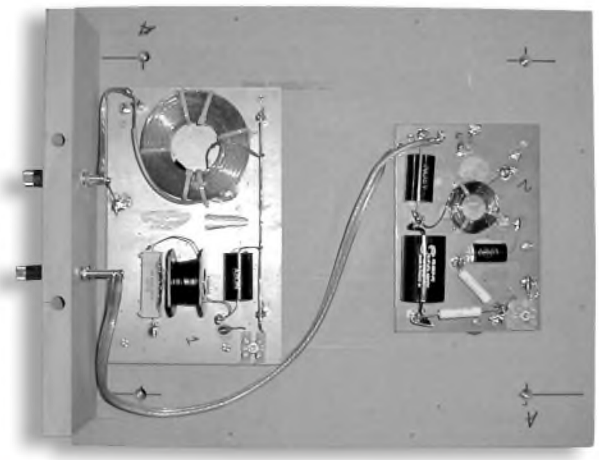
These considerations led to a trial layout for the front baffle of the THOR TL shown in *Fig. 1A*. Now all that was needed was to determine the interior and exterior depths of the line. A first cut at line depth went like this. Assuming 0.75″ MDF for the sides and top leads to an internal width and height of 7.5″ and 41.25″, respectively.

The internal depth, d, is then computed as follows:

$$d = \frac{5021}{7.5 \times 41.25} = 16.25\,\text{in}$$

To get the external depth you must add the thickness of a 1″ front baffle, a 0.75″ internal baffle, and a 0.75″ rear panel for an overall depth of 18.75″. This number was considerably deeper than I wanted and would lead to a rather large and heavy enclosure.

PHOTO 2: Prototype crossover.



(from page 14)

COMMENTS ON MEASUREMENTS

1. Joe has often said "Horizontal frequency response over a 60° arc is a good measure of perceived frequency response." Suffice it to say that *Fig. 16* agrees with my perception.

2. Regarding bass extension—*Figure 13* shows an LF –3 of 44Hz, with an ultimate LF slope of about 12dB/octave (similar to a closed, not vented box). As previously mentioned, I heard what I estimated to be strong 25Hz output. *Figure 13* is down about 12dB at 25Hz, so I would say room gain is helping here.

Since room gain (called "cabin gain" in a car) boosts LF output (re free air) at 12dB/octave, this can very well compensate a speaker's 12dB/octave rolloff. This is also true of closed-box systems, but the THOR TL had absolutely no "box-like" sound; bass was superbly natural.

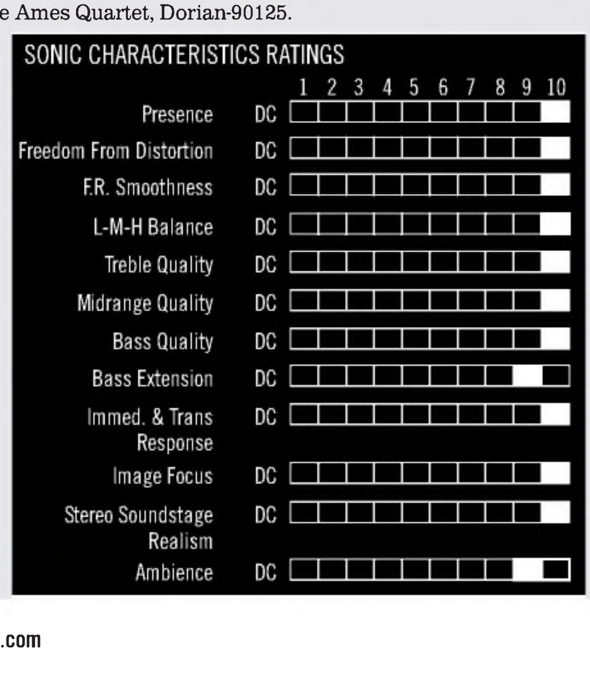
ABOUT ROOM GAIN

In free space, a small (re bass wavelengths) source must deliver constant air *acceleration* to radiate a constant sound *pressure* level versus frequency, because at lower and lower frequencies, less and less of a wavelength is "grabbed" by the source. Below f_3 of a closed-box speaker, however, the cone excursion (and air-volume *displacement*) is constant; acceleration (thus SPL) falls off at 12dB/octave. *Figure 13* shows THOR to do this also.

But in a confined space (room), below the frequency where the longest room dimension is about half wavelength, the air becomes pressurized as a whole. Thus, a constant speaker volume displacement produces air pressure cycles (SPL) of constant amplitude versus frequency; with no leakage (room and speaker), this would extend all the way to DC. So eight 18″ woofers mounted in a wall *could* produce 130dB SPL at 3Hz—not recommended if you value your hearing (and walls)!

RECORD REFERENCES

1. Chesnokov: "Spaseniye sodelal," Turtle Creek Chorale, Timothy Seelig, cond., Track 3, HDCD Sampler, Reference Recording RR-905CD.
2. Hamlisch/Morita: "A Chorus Line," Turtle Creek Chorale/Dallas Women's Chorus, Timothy Seelig, cond., Track 4, HDCD Sampler, Reference Recording RR-905CD.
3. Jacintha, "Georgia on My Mind" from "Here's to Ben," First Impressions Music, FIM XRCD 020.
4. "Carmen Fantasy" from "Percussion Fantasia," Harold Faberman and the All Star Percussion Ensemble, First Impressions Music, FIMCD 017 (HDCD 24-bit recording).
5. Bizet-Shchedrin, "The Carmen Ballet," Orchestre Philharmonique de Monte Carlo, James DePriest, cond., Delos 3208.
6. Beethoven, Symphony No. 6, "The Pastoral," Hanover Band, Roy Goodman, cond., Nimus Records, N15099.
7. Chopin, "The Nocturnes," No. 2 in D-flat major, Artur Rubenstein, Piano, Musical Heritage Society 523870T.
8. Tannoy HiFi Series Sound Sampler, band 11, "Im Uomini, In Soldati," Mozart, Cecilia Bartoli, Wiener Kammerorchester, GyOrgy Fischer, cond.
9. Dvorak, Symphony No. 9, "From the New World," Chicago Symphony Orchestra, Sir George Solti, cond., London 410 116-2.
10. Berlioz, "Nuits d'ete," Frederica Von Stade, BSO, Seiji Ozawa, cond., CBS Master Works, MK 39098.
11. Dvorak, Piano Quartet in E-flat Major, OP. 87, 2nd movement, The Ames Quartet, Dorian-90125.



At this point I made a number of arbitrary decisions. I chose a line taper of 3:1 and limited the overall depth to 13.5″. This led to the internal layout of the line shown in *Fig. 1A*. Placing the interior baffle at an angle produces the desired taper. A side, but important, benefit of the interior baffle is that it adds greatly to enclosure rigidity, effectively clamping the side panels together and largely eliminating side-wall vibration.

The resulting layout has a throat area of 61.875 in^2, which is roughly 1.6 times the combined diaphragm area of the two drivers, and an exit area of 20.625 in^2. As you will see, this departure from Augspurger's recommendation has little effect on f_3.

FILLING THE LINE

In Table 1, Part 3, of the Augspurger article[4], the author recommends packing densities for four filling materials. The optimum packing density is a function of line length. For my first trials I used polyester fiber ("Poly-Cat" polyfill available at Wal-Mart).

For a line length of 81″ the recommended packing density is 0.78 lbs/ft^3. To get the total amount of polyester needed you must calculate the TL internal volume. From *Fig. 1A* the internal volume is calculated to be

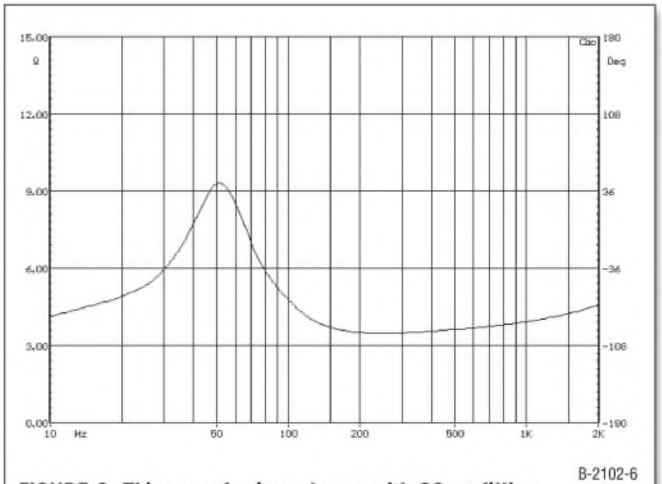


FIGURE 6: TLine woofer impedance with 26 oz filling.

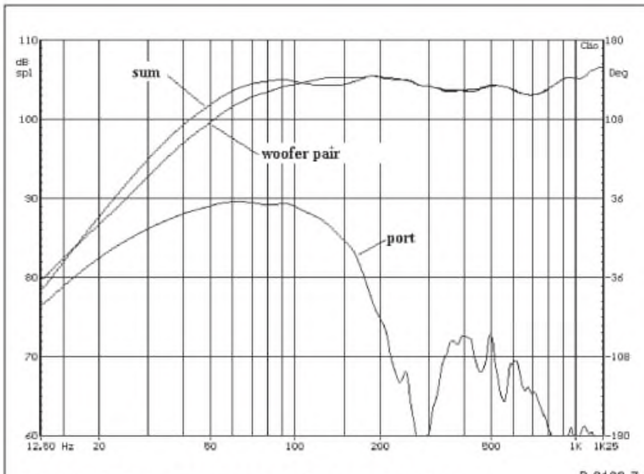


FIGURE 7: TLine responses with 27 oz filling.

1.934ft^3, and the total amount of polyfill needed is then $1.934 \times 0.78 = 1.61$ lbs = 24.1 oz.

In the past, some authors have suggested varying the packing density along the line length, but Augspurger found no particular advantage to this in his studies and recommended a uniform density. Getting a uniform packing density is a bit tricky, however, because the line volume per unit length is changing due to the taper.

Referring again to *Fig. 1A*, the volumes of the two line sections, V_1 and V_2, are found to be 1.208ft^3 and 0.725ft^3, respectively. These volumes represent 62.5% and 37.5% of the total line volume, respectively. Now to get a uniform packing density, you should place approximately 15 oz of the polyfill in V_1 and 9 oz in V_2.

THE APPROACH TO LINE TRIMMING

Once TL dimensions are set, final trimming of the TL packing density is done using a sequence of electrical impedance and acoustic measurements. I could jump to the final result, but I think the various steps in the process are instructive because they can be used in general to trim any transmission line. I will also take this opportunity to compare results from the tapered line with an equivalent straight line driven with a single woofer.

The acoustic measurements are similar to those used by Augspurger in his article. Near-field woofer and port SPL measurements are taken using the CLIO measurement system in the MLS mode. The near-field technique is used to overcome the effects of low-frequency standing waves.

In this technique, the microphone is placed very close to the driver diaphragm (<0.1″) to swamp out diffraction and room effects. At low frequencies where the diaphragm acts like a rigid piston, the measured near-field response is directly proportional to the far-field response and independent of the environment into which the driver radiates. Based on the diameter of the W18 woofers, the near-field woofer measurements are valid up to about 860Hz.

TABLE 4
THOR SYSTEM SPECIFICATIONS

Frequency range(Hz)	40–25000	Sensitivity	89dB SPL
Short-term maximum power*	400W	Recommended amplifier	50–400W
Long-term maximum power*	200W	Dimensions(mm)	$229 \times 1060 \times 343$
Crossover frequency	2500Hz	Bass loading	Transmission line
Impedance	4Ω		

*IEC 268-5

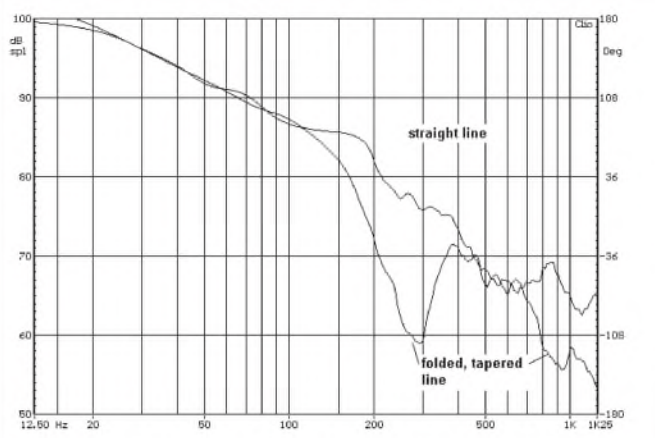


FIGURE 8: Woofer-port transfer function comparison. B-2102-8

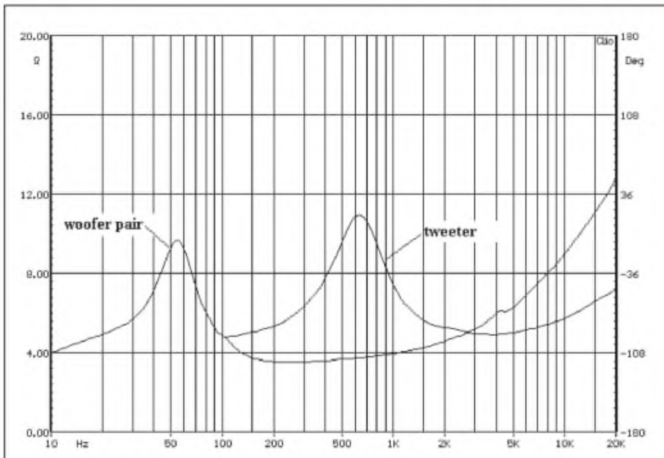


FIGURE 10: Woofer pair and tweeter impedance. B-2102-10

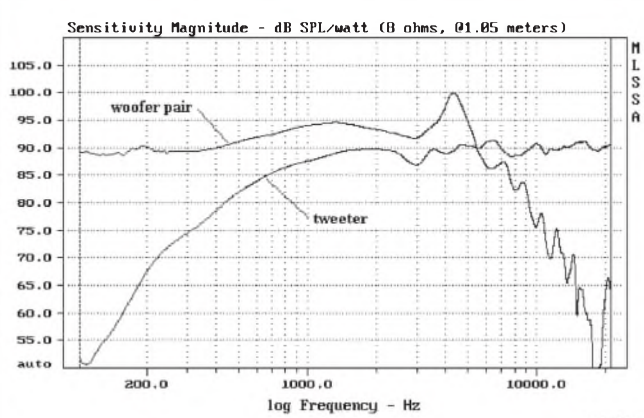


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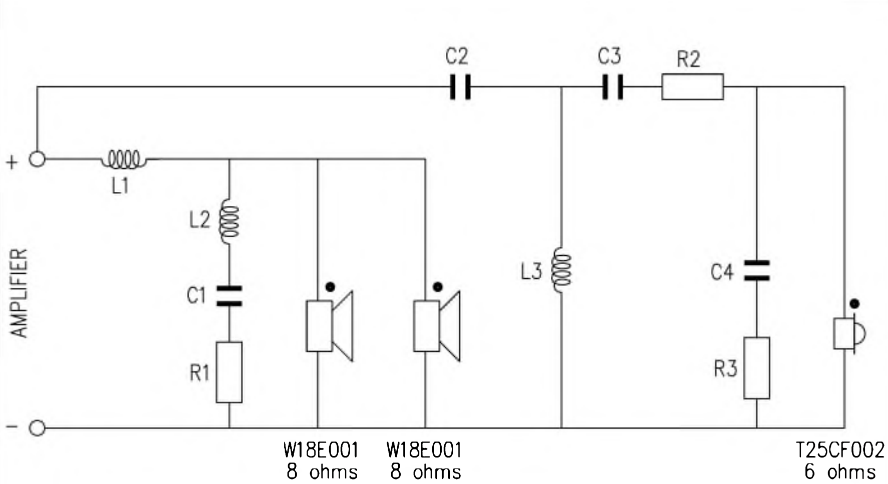
FIGURE 9: Woofer pair and tweeter frequency responses above 100Hz.

PHOTO 3: Prototype TL.



COILS	CAPACITORS	RESISTORS
L1=1.00mH – ø1.63mm (#14AWG)	C1 = 8.2μF	R1 = 0.5 ohms
L2=0.15mH – ø0.81mm (#20AWG)	C2 = 8μF	R2 = 2.0 ohms
L3=0.24mH – ø0.81mm (#20AWG)	C3 = 18μF	R3 = 12.5 ohms
Air–core	C4 = 2.2μF	Metalic oxide or non–inductive wire wound
	Polypropylene or polyester foil	

B-2102-11A

FIGURE 11A: Crossover network for THOR and ODIN Mk3.

CLIO works in the time domain and produces both amplitude and phase response data. The woofer and port responses are measured separately and then added, taking proper account of phase and woofer/port area differences to get the complete low-frequency response of the line. This process is described in detail in Chapters 4 and 7 of my book, *Testing Loudspeakers*[5]. *Photo 1* shows the lab setup for testing and trimming the line.

When measuring the port near-field response, you must place the microphone in the plane of the port exit. The port measurement is then corrected by multiplying it by the square root of the ratio of port area to the combined area of the two woofers. This correction is:

$$\text{port response correction} = \sqrt{20.625/3906} = 0.727$$

After correction, the port response is added to the two woofer responses to get the complete near-field TL response.

There is a potential problem with near-field measurements of woofer response with the MTM configuration. If both woofers are driven simultaneously, the near-field response of one woofer can be contaminated by the output from the second woofer because they are so close together. This is illustrated in *Fig. 2*.

The results shown in *Fig. 2* are for an MTM speaker using two 5.25″ woofers in a sealed enclosure, but the results will apply equally well to THOR. In this series of tests, the microphone is placed about 0.1″ in front of the upper woofer dust cap.

Figure 2 has three plots. One plot is the near-field data taken at the upper woofer with the lower woofer terminals shorted. A second plot shows the acoustic response measured at the upper woofer with the lower woofer driven and the upper woofer shorted. Below 60Hz, the lower woofer signal is only 10dB below the upper woofer output.

The third curve shows the response at the upper woofer with both woofers driven. You see that this plot is contaminated with some of the output from the lower woofer. In practice, the near-field response of each woofer is measured separately with the other woofer shorted, and then both responses are added to get the total low-frequency response of the woofer pair.

So far I've spoken only about acoustic

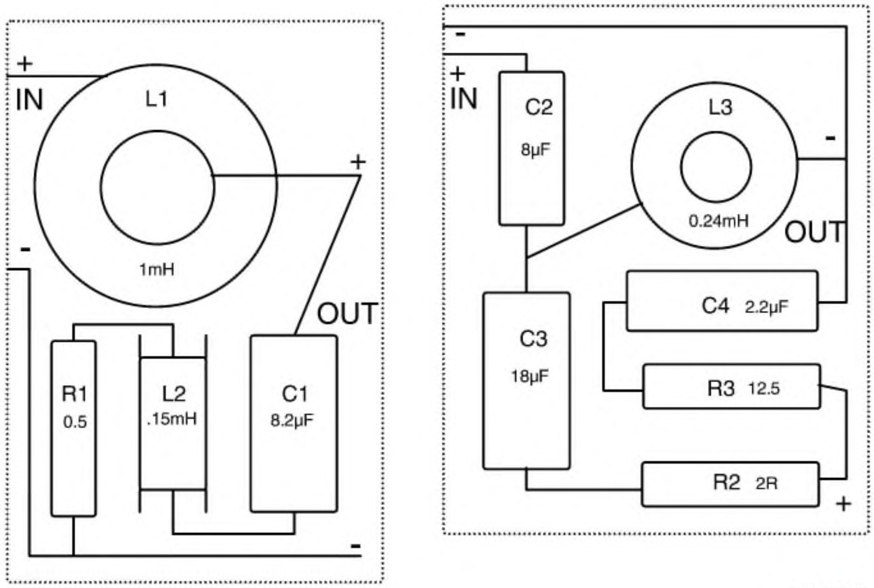


FIGURE 11B: Crossover layouts.

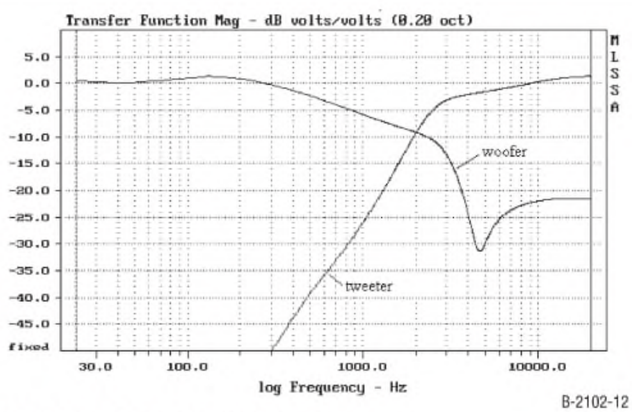


FIGURE 12: Crossover voltage responses.

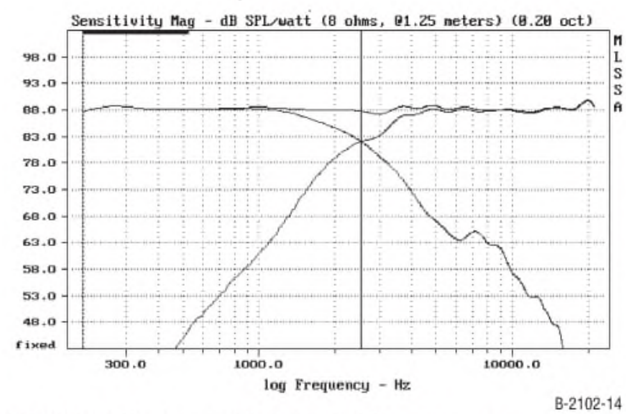


FIGURE 14: System and individual driver responses.

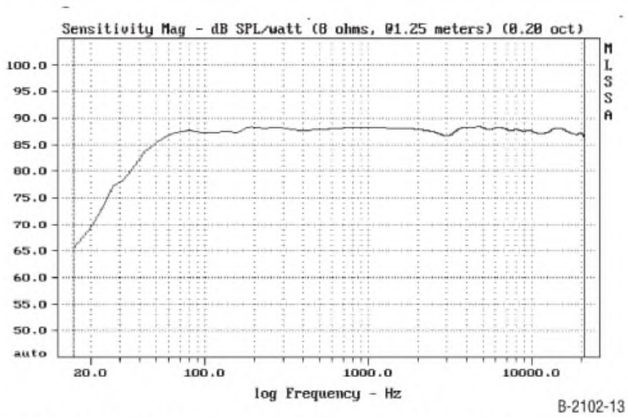


FIGURE 13: Full-range on-axis frequency response.

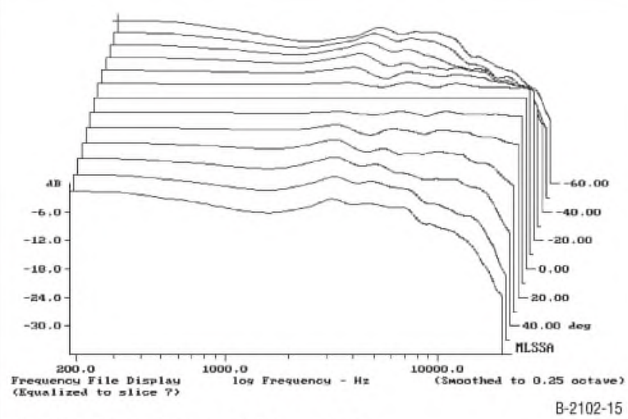


FIGURE 15: Horizontal polar response.

measurements, but you can also tell a great deal about TL performance from impedance data. *Figure 3* compares the measured woofer impedance of the tapered, folded line with an equivalent straight line. Both lines are unfilled. The straight-line plot is offset by +4Ω to ease the comparison.

First notice that both lines exhibit the double-peaked curve of a vented loudspeaker. Thus, up to first order, the unfilled line acts much like a bass reflex speaker. However, you also see additional impedance peaks due to higher-order modes of the line. Beyond the first two peaks, the straight-line TL shows four additional peaks with increasing frequency.

Contrast this with the tapered, folded line where: 1) the minimum between the two lower peaks occurs at a lower frequency, 2) the curve about the minimum is broader and shallower, 3) all peaks are more highly damped relative to the straight line, and 4) only a total of three peaks are seen in the folded, tapered line versus six for the straight line.

These results support the contention that a tapered line has a lower fundamental resonant frequency and a broader range of support. The absence of higher-frequency peaks is due to folding the line. You will see this more clearly later on when you compare transfer functions for optimally damped straight and tapered lines in *Fig. 8*.

TRIMMING THE LINE

Rather than going directly to the "optimum" calculated packing density, I first packed the line uniformly to half the recommended density; i.e., 13 oz of polyfill. The impedance plot for this condition is shown in *Fig. 4*. Notice that the first impedance peak just below 20Hz is almost gone. The same is true for the third peak just above 100Hz. The impedance curve looks almost like that of a closed box speaker.

Responses of the woofer pair, the port, and their sum are shown in *Fig. 5*. The summed response shows a peak-to-peak ripple below 500Hz of ±1.7dB. The low-frequency f_3 point relative to 500Hz is 41Hz. Below 41Hz response falls off at 12dB/octave.

Finally, observe that the port output augments woofer output by 4–5dB at all frequencies between 20Hz and 110Hz. From these results you can conclude that the lightly damped TL acts like an underdamped closed box system with 4–5dB increased low-frequency output capability.

The impedance curve of the optimally filled TL (24 oz) is shown in *Fig. 6*. Now all traces of line modes are gone and the curve is almost purely second-order like that of a closed box. The line is now essentially non-resonant. Responses of the woofer pair, the port, and their sum are shown in *Fig. 7*. The summed response ripple is ±0.6dB and f_3 is 44Hz. Bass augmentation averages 3–4dB from 20Hz to 100Hz.

From these results you can conclude that for a fixed line length there is a trade-off between ripple response and f_3 controlled by line damping. You also see that the line can be damped effectively by observing only the impedance curve. Damping should be adjusted until all traces of line modes just disappear from the impedance curve.

From the impedance plot you can compute an equivalent Q_{TC} for a second-order system with the same impedance curve using any of the procedures outlined in Chapter 2 of reference [5]. The value obtained is Q_{TC} = 0.55, indicating that the woofer pair is almost critically damped.

The woofer/port transfer function plot shows the acoustic output at the TL port produced by the acoustic input to the line coming off the rear of the woofer cones. If you compare the woofer/port transfer function for an optimally damped straight line against an optimally damped tapered and folded line, you get the plot shown in *Fig. 8*. Between 100 and 400Hz and again above 700Hz there is much less high-frequency acoustic output from the port of the folded, tapered line. This greatly reduces ripple in the 100 to 400Hz range relative to that of a straight line.

DESIGNING THE CROSSOVER

With the line optimally damped, my efforts now turned to the design of the crossover. Crossover design for me is a three-step process. First, I placed all drivers in the prototype enclosure of *Fig. 1A* and made acoustic and electrical measurements on them. The measurements include acoustic frequency and phase response, acoustic phase center and electrical impedance. This process is described in detail in Chapter 7 of reference [5].

I then enter this data into one of the many crossover optimization programs I have to develop a preliminary crossover design. Lest you think that this process is automatic and that the software does all the work, be warned that these optimization programs are quite dumb. They cannot decide on an optimum crossover topology and they do not know which components should be optimized and which should be left alone. This is where the "art" of crossover design with optimization software comes in. The software saves many hours of experimentation, producing a preliminary design that gets you quickly into the ballpark, but the designer must pick the right crossover topology and guide the optimization process to a reasonable result.

In the last step I built the preliminary crossover and auditioned it extensively, and used these listening tests for the final tailoring of loudspeaker performance.

CROSSOVER DESIGN CRITERIA

In designing a crossover I have two primary requirements: 1) flat on-axis first arrival response and 2) uniform horizontal polar response. Directional cues so important to imaging are determined primarily by a loudspeaker's first arrival response, which should be relatively flat to avoid amplitude distortion of the directional information.

However, the overall frequency balance of a loudspeaker as perceived by a human listener is a combination of direct and reflected sound. Off-axis energy arrives at the listening position after reflection off the walls. In typical listening rooms this energy arrives well within the Haas fusion zone, a time interval starting just after the first arrival and extending out to 40–50ms. Even if the on-axis response is flat, poor off-axis response can produce a perceived colored frequency balance.

For good stereo imaging and proper spectral balance from side-wall reflections, the horizontal polar response off-axis curves should be smooth replicas of the on-axis response with an allowable exception for the natural rolloff of the tweeter at higher frequencies and larger off-axis angles. (Our ear-brain combination tends to reject higher-frequency side-wall reflections.)

There are several other important quantitative measures of speaker performance, but these are not controlled directly by the crossover network. See my many loudspeaker test reviews in *audioXpress* for a complete discussion of these other measures.

INDIVIDUAL DRIVER TESTING

Figure 9 shows the quasi-anechoic frequency response (first arrival response) of the woofer pair and tweeter with the microphone placed on the tweeter axis

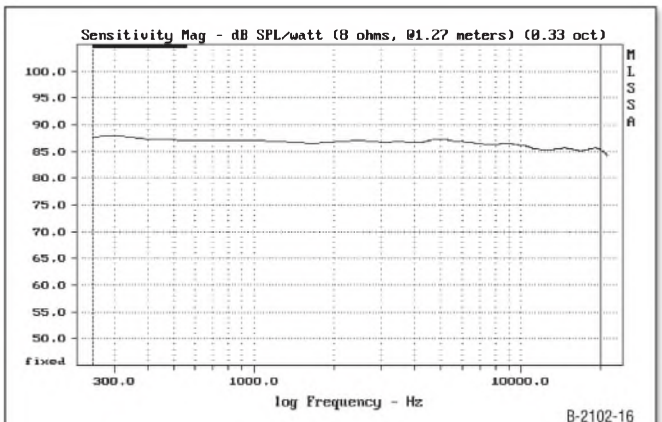


FIGURE 16: THOR average horizontal response over ±30°.

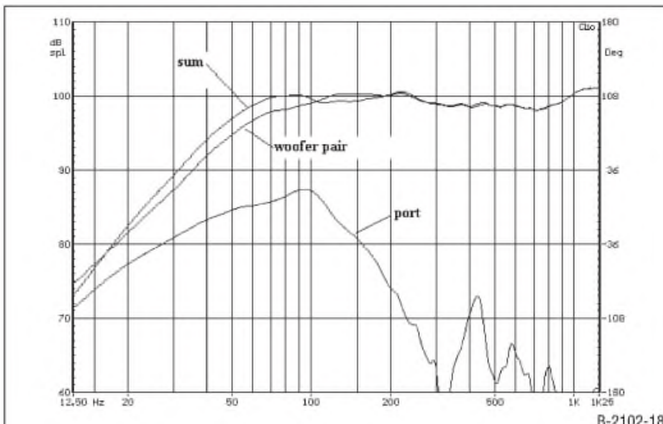


FIGURE 18: TLine responses: 13 oz front and 7 oz batting.

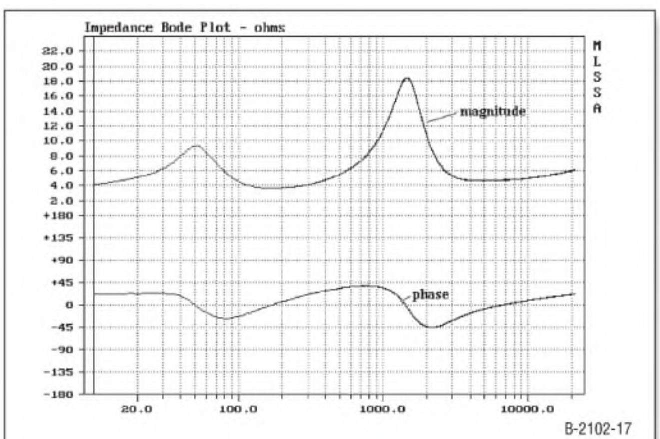


FIGURE 17: THOR impedance magnitude and phase.

PHOTO 4: Parts kit from Madisound.

at a distance of 1.25m. The plot scale covers 100Hz to 20kHz. The data is then normalized to 1m to get driver sensitivity. (Woofer pair response below 100Hz is determined via near-field techniques previously discussed.)

Tweeter response averages 90dB SPL/1m/2.83V above 2kHz. Below 2kHz tweeter response falls off smoothly with a slightly over-damped response.

Starting at about 1500Hz the woofer pair response falls 5dB with decreasing frequency, reaching a uniform level of about 90dB at and below 400Hz. The fall-off is due to the spreading loss characteristic of all woofers on narrow baffles ([5], Chapter 4). The woofer peaks to 100dB at 4.4kHz and then falls off at an average rate of 24dB/octave one octave above that frequency.

Frequency responses of the woofer pair and tweeter overlap between 1.2kHz and 5kHz, suggesting that a preliminary value of 2.5kHz for the crossover frequency would be a good place to start design. This frequency may be subject to change depending upon the resulting horizontal polar response.

Woofer pair and tweeter impedances are plotted in *Fig. 10*. The woofer pair impedance of *Fig. 7* has been extended out to 20kHz.

CROSSOVER TOPOLOGY SELECTION AND OPTIMIZATION

I favor in-phase, i.e., even-order, crossovers for most applications because they are the least sensitive to inter-driver phase differences and timing errors. In the case of the MTM configuration they also limit off-axis response in the vertical which greatly reduces floor and ceiling reflections. For the THOR TL the goal was to design a fourth-order acoustic crossover response at a crossover frequency of 2500Hz.

The woofer crossover must accomplish three functions: 1) control the response rise between 400Hz and 1.5kHz, 2) suppress the 100dB woofer peak at 4.4kHz, and 3) provide the final high-frequency rolloff, which, when com-

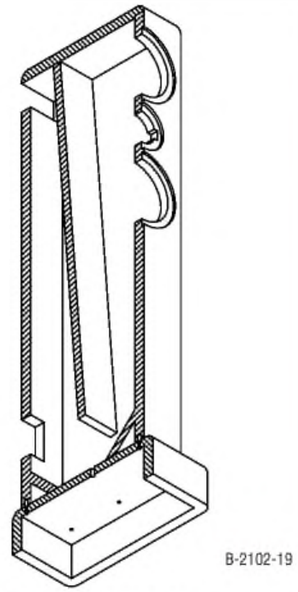


FIGURE 19: Enclosure plans.

PHOTO 5: An enclosure version from Madisound.

bined with the woofer's natural response, produces the desired 24dB/octave acoustic decay.

To adequately protect the tweeter the electrical rolloff of the tweeter crossover must be at least 12dB/octave at all frequencies below crossover. The tweeter's acoustic rolloff below 1kHz begins at a rate of 6dB/octave and transitions to 12dB/octave below 300Hz. The tweeter crossover must therefore produce an electrical attenuation starting at 18dB/octave and then transition 12dB/octave to attain the desired overall 24dB/octave acoustic rolloff.

With these points in mind the crossover circuit topologies I finally selected are shown in *Fig. 11*. Look at the woofer pair crossover first. There is a tendency in crossover design to separate the basic crossover action from the specialized functions of spreading loss correction and response peak suppression.

There is also an often-unthinking use of Zobel impedance compensation when better performance is often obtained without one. This leads to overly complex crossovers. The woofer crossover topology I finally settled on combines the three required functions with an economy of parts and results in absolutely astounding and seamless driver integration.

Woofer and tweeter crossover voltage transfer functions after optimization are shown in *Fig. 12*. For those of you with some circuit theory background, the woofer crossover is a third-order electrical filter with a second-order zero. The woofer crossover voltage response is explained as follows.

L1 provides an initial rolloff of 6dB/octave starting at 400Hz to compensate for rising response of the woofer pair. The R1, C1, L2 triple forms a series-resonant shunt that comes into play around 2500Hz. It produces a 31dB notch at the woofer peak and provides additional high-frequency rolloff. Resistor R1 controls the depth of the notch. Finally, beyond 10kHz the woofer crossover response flattens out, but that is OK because the woofers are falling off at 24dB/octave above the notch.

With a Zobel, the woofer crossover response would continue to fall off above 10kHz at a 6dB/octave rate. Without a Zobel, however, the rising impedance of L1 is matched by the rising impedance of the woofer pair voice coils, resulting in no net electrical rolloff.

Figure 12 also shows the 18dB/octave rolloff required by the tweeter below 1kHz. The tweeter crossover output is down 36dB at 600Hz, the tweeter's measured resonant frequency. The transition to 12dB/octave occurs below the scale of the plot.

Crossover parts values are also listed in *Fig. 11A*. It is very important to use the specified coil wire size for L3. Below 300Hz L3 coil resistance dominates over coil inductive reactance so that the crossover looks like a double RC filter giving the required 12dB/octave attenuation. A larger wire size would reduce coil resistance and push the transition frequency down to a lower value. Resist the urge to use a larger wire size. *Photo 2* shows the prototype crossover.

FREQUENCY AND POLAR RESPONSE TEST RESULTS

Photo 3 shows the prototype TL ready for testing in my lab. *Figure 13* shows the full-range quasi-anechoic frequency response obtained with the microphone placed on the tweeter axis at a distance of 1.25m. Response is flat within ±1dB from 200Hz to 20kHz. Low-frequency f_3 is 44Hz. Sensitivity averages 88dB SPL/1m/2.83V.

Figure 14 shows system frequency response and response of the individual drivers on an expanded frequency scale. On this plot the crossover frequency is highlighted at 2526Hz, satisfyingly close to the target crossover of 2500Hz.

Horizontal polar response is examined in *Figs. 15* and *16*. *Figure 15* is a waterfall plot of horizontal polar response in 10° increments from 60° right (–60°) to 60° left (+60°) when facing the speaker. All off-axis plots are referenced to the on-axis response, which appears as a straight line at 0.00°. Thus, the plotted curves show the change in response as you move off-axis.

For good stereo imaging the off-axis curves should be smooth replicas of the on-axis response with the possible exception of some tweeter rolloff at higher frequencies and larger off-axis angles. For home theater applications a more restricted high-frequency response may be desirable.

From *Fig. 15* you find that the –3dB beam width at crossover is ±50°. There is a bit more off-axis droop around 1500Hz, but the –3dB beam width is still ±45°. Above 15kHz and at angles greater than 40° there is a fairly steep fall-off in response that is characteristic of 28mm tweeters with a recessed dome. But, as I indicated earlier, this performance is perfectly acceptable. The –3dB beam width at 15kHz is still ±25°.

The average horizontal frequency response over a 60° arc is a good measure of perceived frequency response. This average response is plotted in *Fig. 16*. Relative to 1kHz, response at 10kHz is down only 0.9dB. At 20kHz the figure is 1.4dB. This plot, in particular, shows THOR's excellent in-room frequency balance.

THOR's impedance magnitude and phase are plotted in *Fig. 17*. The minimum impedance of 3.6Ω occurs at 180Hz. The impedance peak of 18.3Ω at 1.5kHz is caused by the interaction of the woofer and tweeter crossover networks forming a parallel resonance at that frequency. The maximum phase angle of 45° occurs at 2140Hz, but the impedance magnitude at that point is 10Ω. The system impedance is rated at 4Ω.

PRACTICAL CONSIDERATIONS

After many months of operation, the Dacron pillow filler settled in the second half (the rising part) of one of the lines. This occurs only in the second half of the line because it expands toward the bottom of the enclosure giving little support to the filling material. The settling did not appear to affect performance, but the problem can be avoided altogether by using either Acousta Stuf (available from Mahogany Sound) or Dacron Quilt padding in the second half of the line. Performance will be the same with either solution.

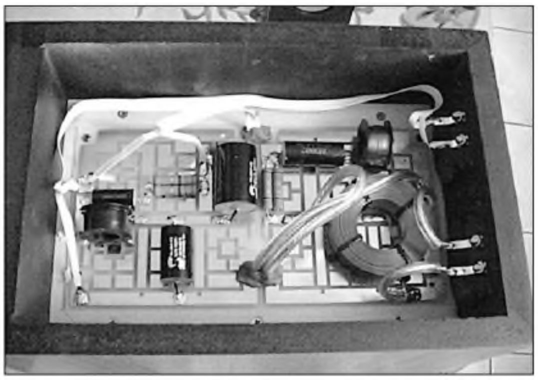


PHOTO 6: Madisound's finished crossover.

In the case of Acousta Stuf you will need 21 oz of material divided into 13 oz for the first half and 8 oz for the second half of the line. This material must be thoroughly teased out to fill each volume.

Alternatively, you can fill the second half of the line with Dacron Quilt padding, which will retain its shape when placed in the line. You will need about 9 oz of the material. Cut it into three 7.5″ wide strips.

The first strip should equal the length of the last half of the line. The second and third strips should be two-thirds and one-third the length of the first, respectively. The longest strip fills the second half of the line, while the second and third strips fill two-thirds and one-third of the lower portions of the line, respectively. Low-frequency response using the quilt padding is shown in *Fig. 18*.

CONSTRUCTION

I will not give detailed instructions for building the THOR enclosure. Enclosure plans are given at the end of this article (*Fig. 19*) and also are available on the SEAS website at www.seas.no. We have provided a cutting guide (*Fig. 20*) that also specifies the total amount of material needed for each enclosure. Any experienced woodworker should be able to follow the plans without difficulty.

For those of you who do not care to build the cabinets from scratch, enclosures are available from the sources listed at the end of the article. A complete kit of parts including drivers and crossovers is also available from these sources. *Photo 4* shows the parts kit provided by Madisound. One version of the enclosure also available from Madisound is shown in *Photo 5*, filled and ready for driver installation. *Photo 6* shows a finished crossover mounted in the base of the enclosure.

SUMMARY

In this article you have seen that Augspurger's work represents an excellent starting point for the design of transmission-line loudspeakers. His recommendations on packing density versus line length are right on target. Once a prototype line is built, the optimum packing density is easily determined experimentally with a sequence of acoustic and/or electrical impedance measurements. Similar acoustic and impedance measurements on the drivers mounted in the prototype enclosure then provide the data for rapid CAD design of a trial crossover network. ❖

REFERENCES

1. G. L. Augspurger, "Loudspeakers in Damped Pipes—Part One: Modeling and Testing; and Part Two: Behavior," 107th JAES Convention, 24–27 September, 1999, Preprint No. 5011.

2. G. L. Augspurger, "Transmission Lines Updated, Part 1," *SB* 2/00.

3. G. L. Augspurger, "Transmission Lines Updated, Part 2," *SB* 3/00.

4. G. L. Augspurger, "Transmission Lines Updated, Part 3," *SB* 4/00.

5. J. A. D'Appolito, *Testing Loudspeakers*, Audio Amateur Corporation, Peterborough, NH, 1998.

SOURCES FOR THOR KIT PARTS

Zalytron Industries: www.zalytron.com

Madisound: www.madisound.com

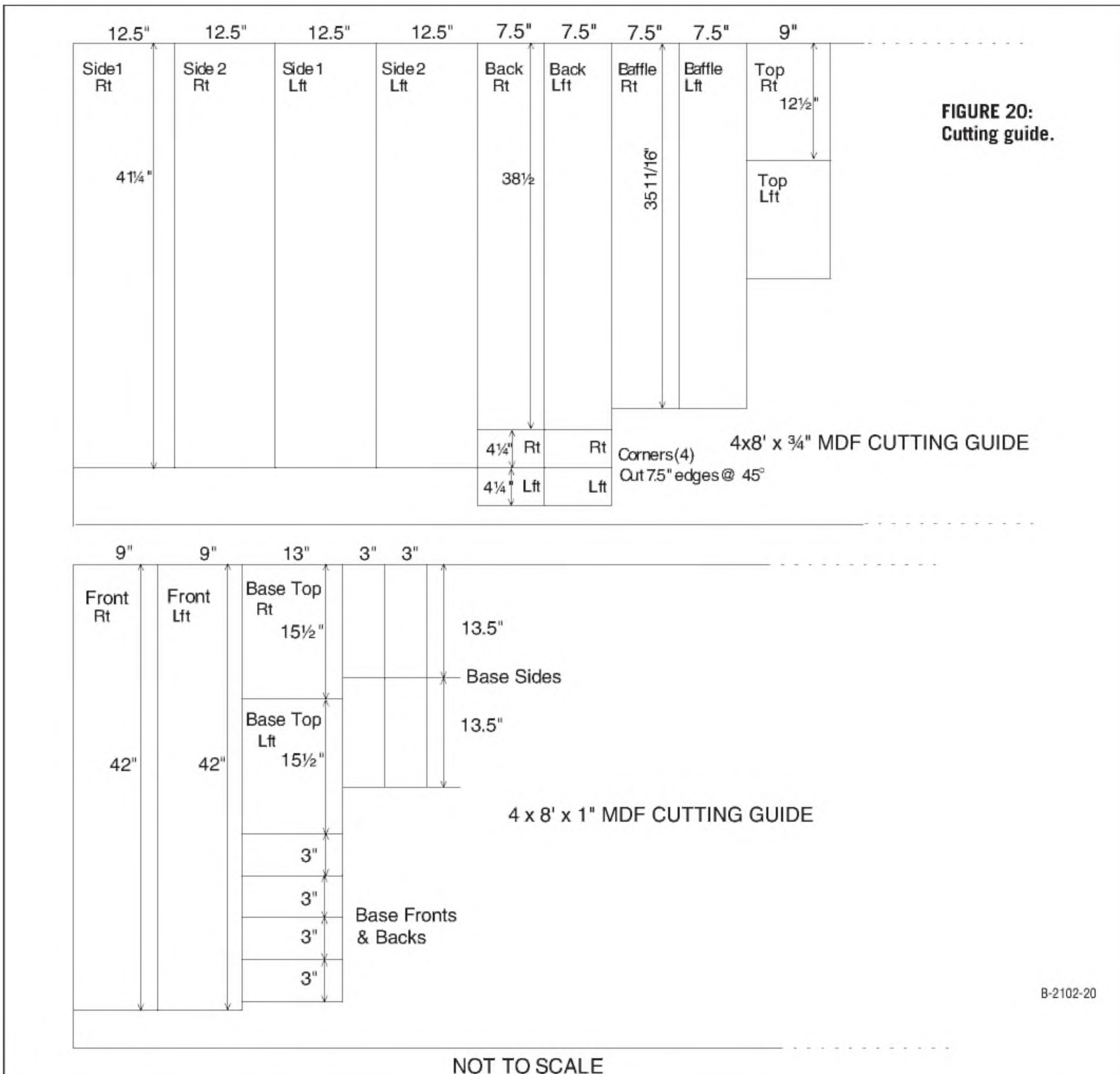


FIGURE 20: Cutting guide.

THOR CUTTING GUIDE

If you can find remnants of 1″ stock for the front panels without having to purchase a full sheet, then the parts for the base may easily be adapted to be cut from the stock remaining in the ¾″ sheet of MDF. Maintain the same outer dimensions for the base.

If you must buy a full sheet of 1″ MDF, the lumberyard will probably do straight cuts for dividing the heavy panel to get it delivered more easily. I had mine cut at 39″ to transport it. If your lumberyard attendant is really helpful, two cuts at 9″ and one at 13″ will do most of your work for you.

If you must cut your own pieces, set the saw width to 9″ for the front panels, then to 13″ for the top of the base and the front and back panels of the base. Reset for 3″ to cut off the base fronts and backs. The base sides can be cut with the 3″ setting, then the four base fronts and backs cut from the end of the 13″ cut. Since the crossover boards fit easily into the base, input connectors are best located on the rear of the base. Rout out an appropriate amount of the rear wall to thickness which will accommodate your choice of connector hardware.

The cutting guide for the ¾″ sheet is also not very economical. If your lumberyard cuts a 51″ piece from one end, you'll probably be able to get the two pieces into your car. Set your saw for 12½″ for the four sides, then to 7½″ for the backs and the T-line baffles. A 9″ setting will provide the width for the tops.

The four corner pieces, which make up the "turns" for the corners of the line, should have their long edges cut at 45° angles, 4¼″ wide. Carefully trim one end of the baffle at an angle of 86°.

PARTS IDENTIFICATION

ONE-INCH-THICK MDF

NO. PCS	SIZE (IN.)	PART
T Line Cabinet		
2	42 × 9	Front
Base		
2	5½ × 13	Top
4	13 × 3	End
4	13½ × 3	Side
THREE-QUARTER-INCH-THICK MDF		
T Line Cabinet		
4	41¼ × 12½	Side
2	38½ × 7½	Back
2	35¹¹⁄₁₆ × 7	Baffle
2	12½ × 9	Top
4	7½ × 4¼	Corners (Bevel 7½ side @ 45°)

PLEASE NOTE:

SEAS Fabrikker AS of Moss, Norway, has published a very early version of the THOR cabinet drawing based on a preliminary sketch provided by author D'Appolito which has been on several websites for many months. The drawing differs in a variety of dimensions based on 1″ MDF for the walls, top, and back panels. A new drawing is available on their web site http://www.seas.no/thor.html conforming to walls of ¾″ material. The diagrams, cutting guides, and all dimensions published in *audioXpress* conform exactly to the D'Appolito prototype, with the exception of the base where the prototype is made of ¾″ MDF.—**E.T.D.**

The All-FET Line Amp

This audio expert from Germany shares with us his latest design—this one an all-FET line amp. **By Erno Borbely**

In Parts 1 and 2 of my JFET articles in *Audio Electronics*[1,2], I described the three basic circuits: common-source, common drain, and the differential amplifiers. Using these circuits you can build up all low-level circuitry in phono preamps, line amps, crossovers, mike preamps, mixers, and so on. In fact, quite a few readers ordered small quantities of the JFETs for experiments. I had the impression that most of them wanted to make some simple crossover circuits, but a couple also experimented with mike preamps.

Figure 1 shows the common-source amplifier from Fig. 7A in Part 1, with minor modifications. Because I intend to use 24V supplies in this series, I have redrawn the amp with this in mind. In order to have maximum voltage swing at the output, I changed the drain resistor to 5k62. The source resistor has been increased to 100R, to produce 2mA drain current with the particular JFET used (I_{DSS}: 7–8mA). The drain is sitting at approximately 12V DC.

This simple amp has a gain of 33 times; i.e., 30dB. The frequency response is 300kHz! The distortion is dominated by second harmonics, which is to be expected from a square-law device. At 1V RMS I measured 0.4% and at 3V RMS it increased to just over 1%. The linearity can be improved by increasing the supply voltage, as we have seen in Part 1[1].

This very simple amp would probably make many SE fans happy. Unfortunately, the circuit is not complete, in that the output is at a DC voltage of 12V, and requires a capacitor to isolate it from the next amp in the chain. However, capacitor coupling—whether in tube or in FET amps—always invites lots of discussion on the quality of the caps used (especially from those arm-chair amateurs who never build anything), so I always try to avoid them if possible!

Thanks to the availability of complementary devices in semiconductors (npn and pnp bipolars, and n-channel and p-channel FETs), it is easy to design DC-coupled circuits that do not use caps in the signal path. This article, therefore, focuses on DC-coupled amps.

THE TWO-STAGE SE AMP

Figure 2 shows a two-stage JFET amp, in which the second stage, also a common-source amplifier, level-shifts the DC of the first stage to 0V. The drain voltage of JFET 1 is no longer required to sit at half of the supply voltage, because it is not responsible for the output voltage swing. The voltage drop across the drain resistor should, in fact, be as small as possible, to allow the second stage to work at maximum drain/source voltage.

I usually work with about 3V, which is a compromise between enough voltage to turn on the second-stage JFET, and enough to produce a reasonable gain. I selected 10mA drain current in the second stage. A small trimpot is inserted in the source circuit of the second stage, which allows trimming the output to 0V. The second-stage JFET must be selected for I_{DSS}>10mA, to avoid operating it with positive bias.

The open-loop gain (the feedback re-

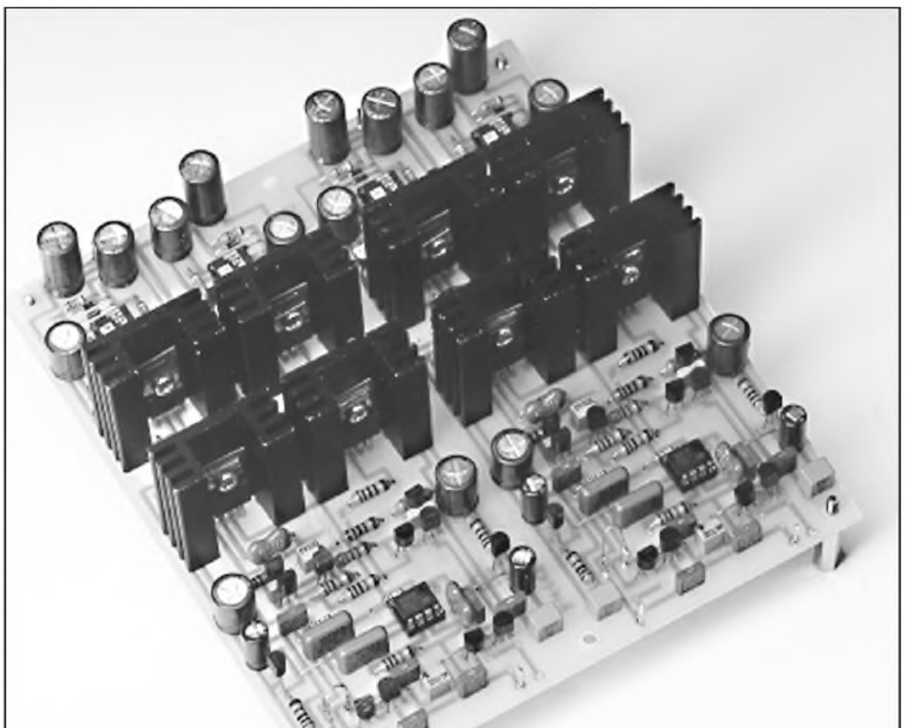


PHOTO 1: The EB-1198/116 all-FET line amplifier with FR-4 PCB and Caddock/Tantalum resistors.

sistor RF is open) is about 36dB and the frequency response is 150kHz. The reason for the relatively small increase of open-loop gain compared to the single-stage one is that we have reduced the gain in the first stage by using a smaller drain resistor, and the second stage also operates with a small(ish) gain because of the small drain resistor and the local feedback in the source. THD has been reduced a bit for the same output level, because of the higher available voltage swing in the second stage: 1V = 0.3%, 3V = 0.93%. Maximum output voltage before clipping is >10V RMS.

TABLE 1
THD MEASUREMENTS FOR AMPLIFIER SHOWN IN *FIG. 4*

OUTPUT V RMS	RF = 1k (GAIN: 20.9dB)	
	1kHz THD	10kHzTHD
1V	0.0015	0.0035
3V	0.0024	0.0075
5V	0.005	0.0145
6.5V	0.0075	0.02
8V	0.012	0.027
10V	0.029	0.044

I have indicated a feedback resistor RF from output to the source of the first stage. However, applying a useful feedback is not easy, since the source resistor, setting the operating point in the first stage, is fixed at 100R. If you want to have, say, a closed-loop gain of 10×, you would need an RF = 1k, which would load the second stage too much. Also, the open-loop gain of the amp is too low to have a useful amount of gain applied for feedback.

OPEN-LOOP GAIN

You can increase the open-loop gain, and thereby the available gain for feedback, by increasing the gain in the second stage. The easiest way to do this is to replace the 2k43 resistor (which supplies 10mA) with a 10mA constant-current source, such as the E-103 (*Fig. 3*). The E-103 has a dynamic impedance of >100k and increases the open-loop gain by 20dB when the amp is loaded with a 10k resistor. Measurements show that this in itself does not improve the linearity of the stage, but now you have more gain that you can apply as feedback.

To address the load capability of the second stage, I have added a source follower to the amp (*Fig. 4*), described in Part 2[2] (Fig. 16B). The source follower adds a number of advantages to the amp: it isolates the feedback resistor from the high output impedance of the second stage, thereby further increasing the open-loop gain, and it provides a low output impedance to drive the feedback resistor and outside load. Closed-loop gain with RF = 1k is 20.9dB. The closed-loop frequency response is more than 1MHz! *Table 1* shows the THD measured on this amp. Overall a simple, but very good, SE line amp!

THE COMPLEMENTARY GAME

Although this SE amp functions well, I have two issues with it. The first is its DC stability. As described in Part 1[1], the transconductance curve of JFETs is temperature-dependent, and unless you operate it at zero tempco, you will have a temperature drift. Of course, a few mV of drift is acceptable in most audio applications; however, here we will try to minimize this as much as possible. Naturally, the open-loop linearity can also be improved. Enter the fully complementary circuitry.

Consider the circuit in *Fig. 3* as half of such a complementary circuit. Imagine folding down this circuit along the horizontal ground line, naturally replacing the n-channel with p and vice versa. *Figure 5* shows this basic 4-JFET configuration. The trimpot is used to adjust the output offset to 0V.

I have kept the same currents in both

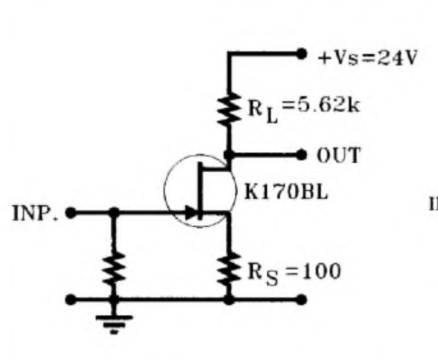


FIGURE 1: The common-source amplifier.

A-2094-1

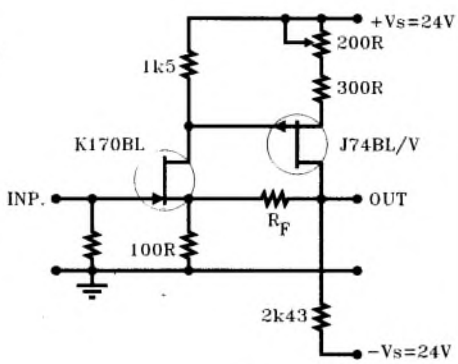


FIGURE 2: A level shifter converts it to a DC-coupled amplifier.

A-2094-2

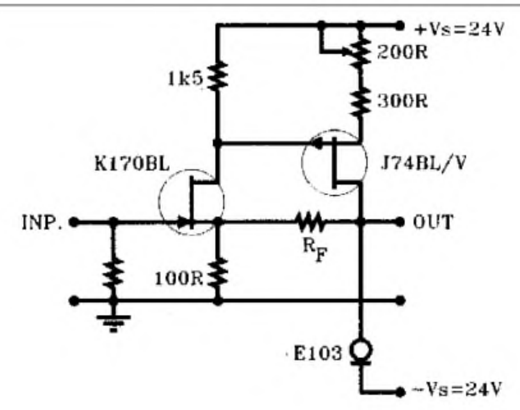


FIGURE 3: The constant-current source increases the open-loop gain.

A-2094-3

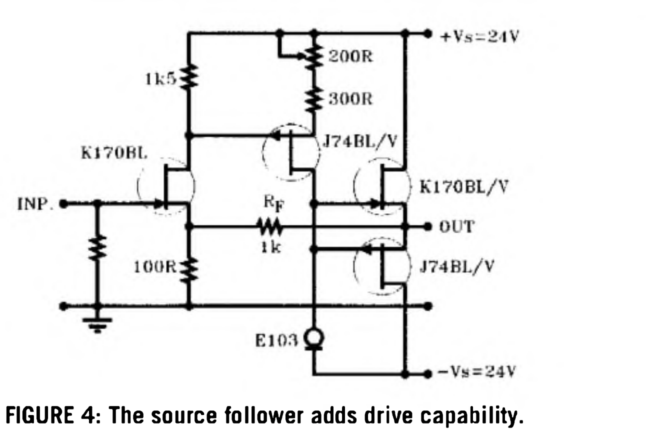


FIGURE 4: The source follower adds drive capability.

A-2094-4

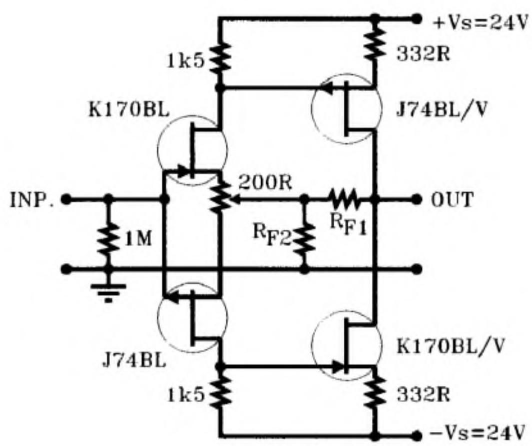


FIGURE 5: The basic 4-JFET complementary amp.

A-2094-5

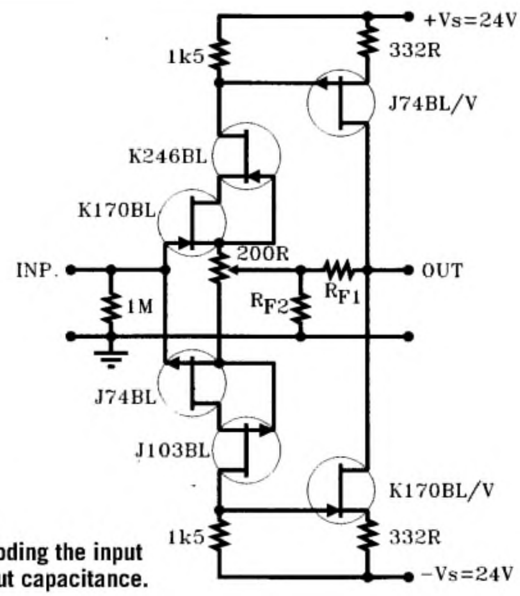


FIGURE 6: Cascoding the input reduces the input capacitance.

A-2094-6

input stage and second stage at 2mA and 10mA as before. Since the second stage consists of two JFETs connected together at their drains, they act as constant-current sources to each other. The amp therefore becomes a transconductance amplifier, converting input voltage to output current. The conversion back to voltage happens when you connect a resistor from the output to ground as a load resistor, or back to the negative input as a feedback resistor.

Here a feedback resistor is shown, representing the load at the output. The open-loop voltage gain is therefore very much dependent on this load resistor. I am usually using RF1 = 2k21 feedback resistor; open-loop gain is equal to 42dB with this load. The open-loop frequency response is about 150kHz, same as the SE circuit.

Open-loop distortion is almost an order of magnitude less than for the SE circuit, indicating that the complementary circuit, depending on the matching of the JFETs used, is cancelling some of the second harmonic distortion. DC stability has also improved significantly due to the use of complementary devices cancelling the drift.

There are many more possibilities in terms of applying feedback than with the SE circuit, and you can produce a number of useful circuits. If you select RF1 = 2k21 and RF2 = 221R, you will have a line amp with 20.9dB gain. You can also make RF2 = 2k21 for a 6dB amp, which is very useful as a buffer between your CD player and your power amp, when you need a volume control in-between. If you remove RF2 completely, you will have a unity gain buffer. *Table 2* shows the THD for this buffer with different gain settings.

TABLE 2
THD MEASUREMENTS FOR THE 4-JFET AMP SHOWN IN *FIG. 5*

OUTPUT	OPEN-LOOP THD		10× GAIN (20dB)		2× GAIN (6dB)		UNITY GAIN*	
	1kHz	10kHz	1kHz	10kHz	1kHz	10kHz	1kHz	10kHz
1V	0.032	0.033	0.0025	0.0047	0.0012	0.003	0.0011	0.0022
3V	0.095	0.097	0.0055	0.015	0.0016	0.0058	0.0016	0.003
5V	0.165	0.17	0.0095	0.021	0.0032	0.012	0.0032	0.0068
6.5V	0.22	0.23	0.013	0.03	0.0048	0.016	0.0047	0.012
8V	0.28	0.3	0.018	0.04	0.007	0.022		
10V	0.37	0.4	0.028	0.06	0.0115	0.033		

*Measurements limited by HP 339A output voltage.

TABLE 3
THD MEASUREMENTS FOR THE "SUPER BUFFER" SHOWN IN *FIG. 8*

OUTPUT	OPEN-LOOP		10× GAIN (20dB)		2× GAIN (6dB)		UNITY GAIN*	
	1kHz	10kHz	1kHz	10kHz	1kHz	10kHz	1kHz	10kHz
1V	Noise	Noise	0.0015	0.0025	0.0012	0.0021	0.0012	0.0021
3V	0.025	0.05	0.0011	0.0015	0.0011	0.0015	0.0011	0.0015
5V	0.035	0.084	0.0011	0.0024	0.0011	0.0022	0.0011	0.0022
6.5V	0.05	0.11	0.0011	0.0026	0.0011	0.0022	0.0011	0.0022
8V	0.07	0.15	0.0011	0.0034	0.0011	0.0028		
10V**	0.12	0.4	0.0012	0.007	0.0012	0.0047		

*Measurements limited by HP 339A output voltage.
**The amp starts clipping at 10V RMS.

REDUCING THE INPUT CAPACITANCE BY CASCODING

The input capacitance of the basic 4-JFET circuit in *Fig. 5* is about 28pF. This is normally no problem when the source impedance is lower than 10k. However, the input capacitance of the JFETs is voltage dependent, and when driven from high impedance it might cause distortion.

You might recall that we solved this problem in Part 1 by cascoding the input JFET with another one (Fig. 9A/B in Part 1). This circuit is shown in *Fig. 6*. The cascoding reduces the input capacitance to less than 2pF, and there is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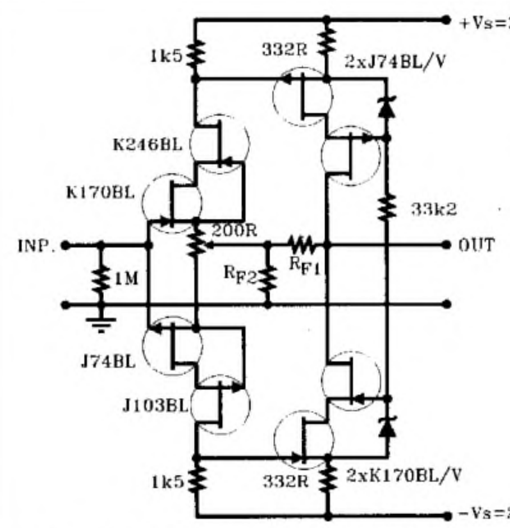


FIGURE 7: Cascoding the second stage improves the high-frequency linearity.

A-2094-7

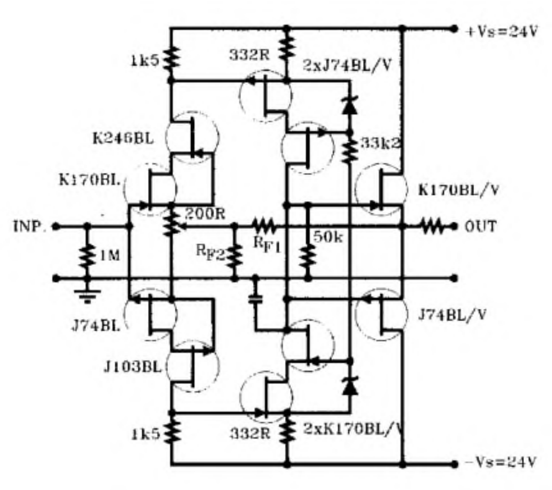


FIGURE 8: The "Super Buffer."

A-2094-8

no sign of any distortion with high source impedances.

Cascoding the second stage as well (*Fig. 7*) can make further improvements. This helps reduce the internal capacitances and capacitance modulation. However, all these circuits have a high output impedance and because the open-loop gain is limited, even the feedback cannot reduce it to very low levels.

Of course, this is not an important factor in normal line amp applications; however, when used as a unity gain buffer in filter applications, it is essential. I have therefore added a complementary source follower to the output. *Figure 8* shows the final all-JFET "Super Buffer."

Since there is no load on the second stage, the open-loop gain is very high. In fact, it is too high for my liking, and I usually reduce it with a resistive load on the second stage. With 50k the open-loop gain is 67dB. Open-loop frequency response is close to 20kHz.

The THD for four different gain settings is shown in *Table 3*. Note the very low open-loop distortion indicating very good linearity. Due to the voltage drop across the bias circuit in the second-stage cascode, the amp starts to clip just around 10V RMS. This is responsible for the slightly increasing THD levels at 10V. A higher supply voltage would allow more output without clipping.

The amp needs a series resistor at the output (47Ω) for stability. The actual output impedance of the amp is about

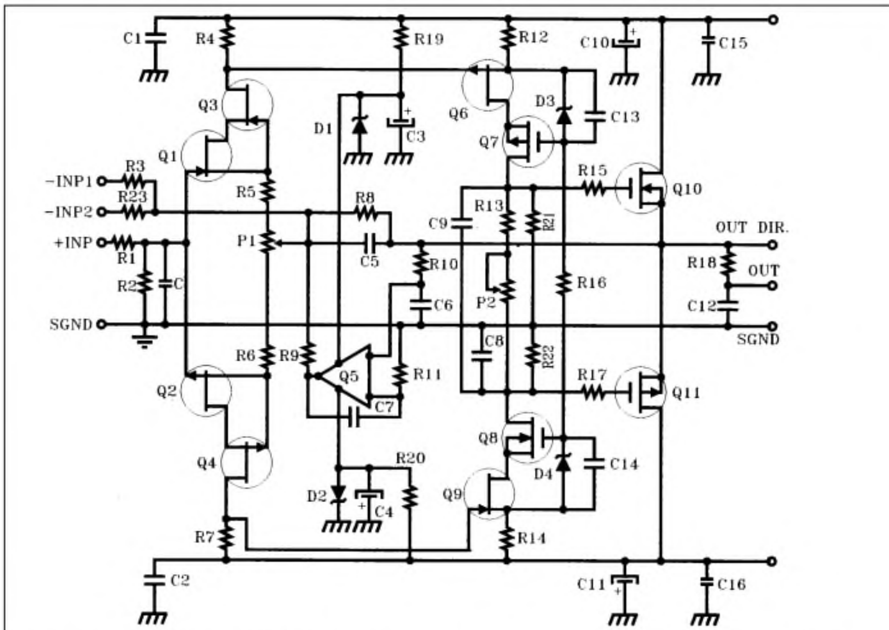


FIGURE 9: The EB-1198/116 all-FET line amp.

A-2094-9

2Ω, and with the series resistor it's 50Ω. The amp can drive loads down to 1kΩ, with no increase in THD. The frequency response with 10× gain is more than 1MHz! In unity gain it has a rise time of <100ns, indicating a frequency response well into the MHz region.

The circuit also shows very good noise performance. Measured in the 10× gain position with 2k21/221R feedback resistors, the input noise is <0.4μV over the audio bandwidth. You can further reduce the input noise by paralleling several input stages. See the input stage of the all-FET phono preamp on my homepage (www.borbelyaudio.com).

THE ALL-FET LINE AMP (EB-1198/116)

I have used the amp shown in *Fig. 8* in many line amp and filter applications with very good results. It's simple, inexpensive, and has an extremely high resolution. However, for a commercial product I wanted even higher drive capability.

My goal was to be able to drive loads down to about 100Ω in pure Class-A, also allowing it to be used as a headphone amp. To be able to drive 100Ω in Class-A with up to 10V RMS, you need to have a bias current of 70mA. There are no JFETs that can withstand this power dissipation, and I used MOSFETs as output devices. The schematic of the all-FET line amp is shown in *Fig. 9*.

The input JFET-cascode, consisting of Q1-Q2-Q3 and Q4, is operated at 2mA, and Q1 and Q2 must be matched to ±10% of I_{DSS}. Q1 and Q2 are selected from the "BL" group, and, depending on I_{DSS}, they are adjusted for I_D = 2mA by R5 and R6. (For selection of JFETs see the test jig described in Part 1.) Q3 and Q4 must have a gate-source voltage of minimum 2V at this current, to provide proper drain-source voltage to Q1-Q2.

The second stage, consisting of JFETs Q6-Q9 and MOSFETs Q7-Q8, operates at just below 10mA. Select Q6 and Q9, which must be matched to each other, for I_{DSS}>10mA. It is easiest to select these from the "V" group, because the minimum I_{DSS} for this group is 10mA (I_{DSS}: 10-20mA). I am using MOSFETs as cascode devices because I wanted to try the circuit with higher supply voltages than the JFETs would allow. The output devices, Q10-Q11 (Hitachi TO-220 MOSFETs), are operating at approximately 70mA; proper heatsinking is therefore mandatory.

Although the DC stability of the amp is very good, I added a servo to ensure very low offset voltage, independent of the ambient temperature. Q5 is a JFET-input servo amplifier, providing tracking of the output offset to less than a couple of mV. The RC networks of R10-C6 and R11-C7 filter out all AC signals over 1Hz. The servo is therefore only operational under 1Hz, and should not affect the sound of the amplifier. Nevertheless, if you believe that it does, don't hesitate to try different op amps for this. D1 and D2 are IC shunt regulators, supplying Q5 with ±10V.

For an all-out line amp, the best resistors are the Vishay S102s and the Caddock MK132s. Personally, I prefer the Caddocks, which seem to be more neutral than the Vishays. However, you don't need to use all-Vishays or all-Caddocks to get super sound; only the input and feedback resistors (R1, R2, R3, R4, R7, and R8) should be from these. The rest could be Tantalum (I have found these give the best combination with the Caddocks), but DALE RN60s are also good (but only the ones using non-magnetic construction). The frequency compensating capacitors are polystyrene or Mica caps, the electrolytics C10 and C11 are ELNA CERAFINE, and the rest of the caps are WIMA and Roederstein.

The open-loop gain of the amplifier is kept at a relatively low 66dB by the two resistors R21-R22. Open-loop frequency response is 20kHz (–3dB)! Open-loop THD is very low: 0.055% at 1kHz, 3V RMS and 0.095% at 10kHz, 3V RMS. This is reduced to below the measurement limit of the HP 339A distortion analyzer when feedback is applied with resistors R8-R3.

Normally, R8 is 2k and R3 is 221Ω. This gives a closed-loop gain of 20dB. Changing the ratio of R8-R3 can reduce the closed-loop gain. The rise time of the amplifier is less than 200ns for an output of +10V. Closed-loop frequency response is over 1MHz! Output impedance is less than 10Ω at the "direct" output and is approximately equal to R18 at the "normal" output.

The all-FET line amp also has low-noise, wideband regulators on the PCB (schematic not shown here). These are using low-noise JFET-input op amp and MOSFET outputs. The power-supply requirement is ±29V unregulated voltage at 100mA per channel. A high-quality power supply with fast/soft recovery diodes and high-quality 10,000-22,000µF electrolytics is recommended. Two of the line amps can also be connected for balanced operation.

TESTIMONIALS

Finally, here are a couple examples of authentic customer feedback from the US and Canada:

"After living with the line amp for quite a few weeks now, I would say the following is what I hear:

- incredible resolution in loud, congested orchestral passages; e.g., locating triangles on top of full orchestra
- amazing sense of "air," space, width, depth; i.e., imaging is holographic
- startling resolution in what I call "leading edge transients" on, for example, brass, drums, and so forth.

I could go on and on, but, you'll think I'm hyperbolizing! I can't imagine improving on your design."

"It (the all-FET line amp) has extraordinary detail, very accurate harmonic structure, powerful macro and micro dynamics, and an extremely low noise floor allowing effortless transient attack, and vocal intonation and ambient decay to be vividly portrayed...." ❖

PARTS LIST FOR EB-1198/116

R1	100R Caddock MK132
R2	100k Caddock MK132
R3	221R Caddock MK132
R4, R7	1k5
R5, R6	Adj. for 2mA drain current
R8, R23	2k21 Caddock MK132
R9, R21,R22	100k
R10, R11	1M
R12, R14	330R
R13, R15, R17	100R
R16	33k
R18	47R
R19, R20	1k ROE WK2 1.1W
P1	200R COPAL RJ5W
P2	500R COPAL RJ5W
C, C5	10pF/500V PS, MICA
C1, C2	0.1µF/100V ROE MKT-1826
C3, C4	100µF/25V ROE EKO
C6, C7	0.22µF/160V WIMA MKP4
C8	150pF/500V PS, MICA
C9, C13, C14, C15, C16	0.1µF/100V ROE MKT 1826
C10, C11	47µF/35V ELNA CERAFINE
C12	560pF/500V PS, MICA
Q1	K170BL Matched to Q2
Q2	J74BL Matched to Q1
Q3	K246BL
Q4	J103BL
Q5	AD820AN Analog Devices
Q6	J74V Matched to Q9
Q7	J148
Q8	K982
Q9	K170V Matched to Q6
Q10	K216
Q11	J79
D1, D2	LM4040DIZ
D3, D4	LM336Z-5.0

MISCELLANEOUS

PCB: EB-1198/116/BUFFER45

Heatsinks Fischer SK75 37.5mm

Note: Resistors are Tantalum or Vishay-Dale, unless otherwise noted.

The EB-1198/116 kit, including power supply, volume control, connectors, and case is available from:
Borbely Audio
Angerstr. 9
86836 Obermeitingen, Germany
+49/8232/903616
Fax: +49/8232/903618
Web: www.borbelyaudio.com

ACKNOWLEDGMENT

Many thanks to Walt Jung of Analog Devices for his valuable comments and suggestions.

REFERENCES

1. Erno Borbely, "JFETS: The New Frontier, Part 1," *AE* 5/99.

2. Erno Borbely, "JFETS: The New Frontier, Part 2," *AE* 6/99.

A Practical Circlotron You Can Build

Meet the Circlotron, a high-powered, but seldom-used, amplifier that may be making a comeback, thanks to this author. **By John Stewart**

PHOTO 1: The author's Circlotron amplifier.

Have you ever wondered what a Circlotron is? If so, have you wondered why anyone would use this relatively exotic topology to build an amplifier?

In this article I show you how this circuit works and provide enough information so that you can build your own. You will also be able to select from several versions of it. This circuit is probably not for beginners. If you are new to the hobby, it might be best to try something simpler in order to gain some experience.

The Circlotron has been around for a long time, but not well understood. Nor has it been built often by amateurs. There are two outstanding reasons why the Circlotron has not been popular. To begin with, you need two completely independent high-voltage power supplies. The other problem is the requirement for an output transformer of one-quarter the impedance of those commonly available.

This second problem is the very reason someone took the time to develop the circuit in the first place. A low impedance audio output transformer is much easier to build than a similar device of high impedance. My amplifier is built on a 10 × 17 × 3 chassis from Hammond (*Photo 1*). When using 6L6GCs it is capable of 30W of clean audio. The version using 6550Cs easily makes 50W.

Don't forget that the potentials you will be working with are lethal.

FEATURES

1) Choice of output transformer configuration
   a) One custom-built to low impedance or
   b) Two off-the-shelf standard transformers or
   c) A transformer meant for a Dyna, and so forth, but wired differently.
2) Uses two off-the-shelf transformers for the power supply.
3) Allows triode or beam tetrode operation.
4) Runs with or without feedback.
5) Choice of output tubes provided by wide range bias system.
6) Standby and run modes.
7) Outstanding performance.
8) Low cost.

AMPLIFIER CIRCUIT DESCRIPTION

The most important feature of the Circlotron is its output circuit in relation to the audio load. In a common push-pull circuit, the power tubes appear to be in series as far as the audio signal is concerned.

In the Circlotron the output tubes and load have been rearranged so that the tubes are able to drive the audio load in parallel (*Fig. 1*). That will let you use an output transformer of one-quarter the normal impedance. It has been shown that lowering the impedance of an audio transformer is a great way of improving its performance. Leakage inductance and winding capacity are much less of a problem in low impedance transformers. See *Table 1* for the parts list.

So that you can follow the circuit easier, I have separated the power supply (*Fig. 2*) from the amplifier. The amplifier is completely balanced throughout and uses a two-stage differential input circuit. The differential amplifier neatly sidesteps some of the problems associ-

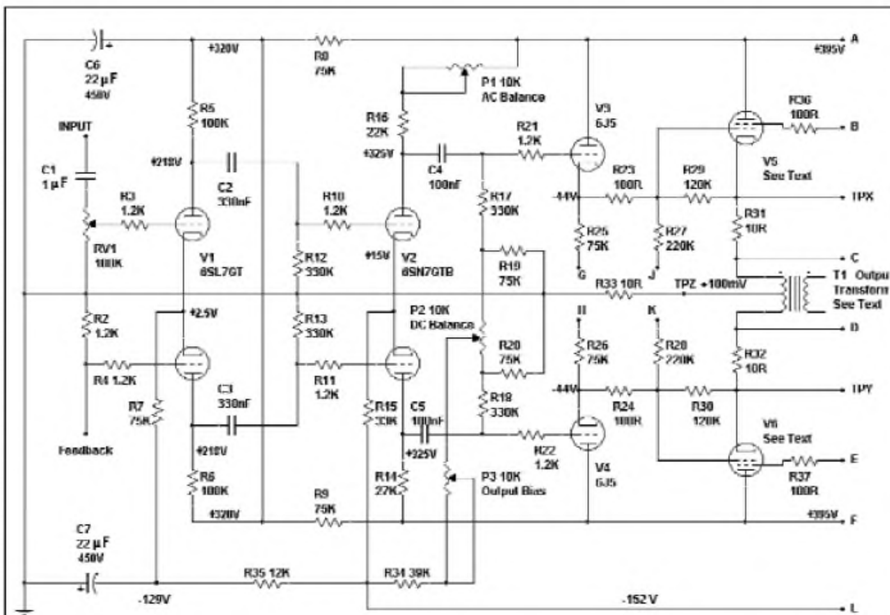


FIGURE 1: Circlotron circuit.

ated with phase inversion.

For this version of the diff amp the only drawback of note is the need for a negative power supply. In the days before low-cost silicon rectifiers, this produced lots of circuit complication. As a result, you will not see these circuits used often in amplifiers of the '50s. You will see in this power supply that a negative output is no longer a hindrance.

Voltages shown in *Fig. 1* are under no signal conditions. You will need to be careful while making voltage measurements while signals are present. At full output there will be about 250V peak signal on the wires marked A through F with reference to circuit ground. That could make a mess of a voltmeter set to measure low voltage DC on any of these locations.

The input signal is fed to one of the grids of a 6SL7GT (V1) on the left-hand side of the diagram. You could use a 12AX7 just as well. In many of my circuits I simply use the tubes I have on hand.

Gain of the amplifier is set by the 100k pot, RV1. If you intend to use full loop feedback, you can connect it to the other grid. Otherwise, just connect that point to your circuit ground.

Power for the first stage comes through the 75k resistors R8 and R9. Later you will see that the double high voltage supply is driven by the full audio voltage on the primary of the output transformer. The voltages are equal and opposite so that the two resistors provide the average with very little audio modulation. Any residual is removed by the 22μF capacitor C6.

Output of the 6SL7GT goes to a bootstrapped 6SN7GTB (V2). Again, you could just as easily use a 12BH7 at this location. The reason for the bootstrapping is to provide enough voltage swing for the output tubes (V5 and V6), which are connected to the output transformer as cathode followers, so they have no voltage gain. That means the 6SN7GTB driver plates must swing several hundred volts just to keep ahead of the output.

Without bootstrapping, you would need a much higher voltage power sup-

FIGURE 2: Circlotron power supply.

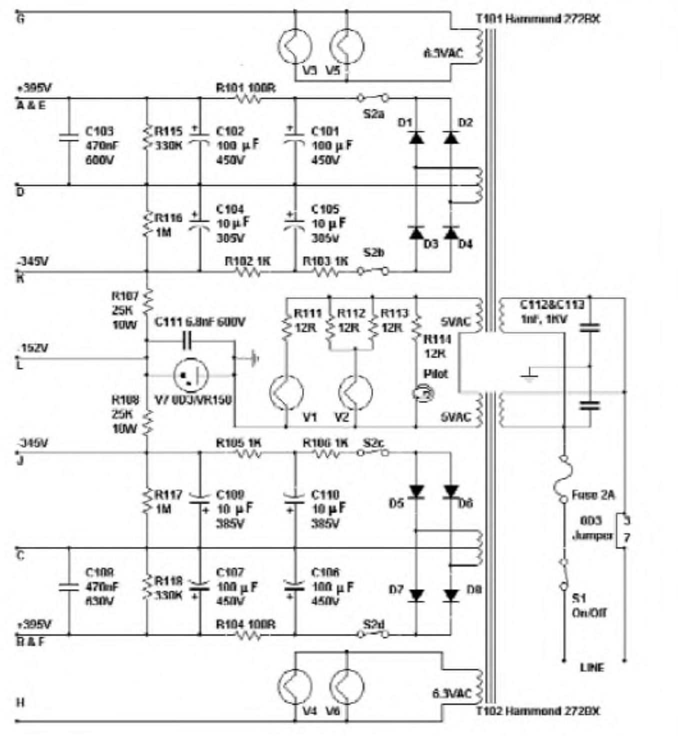


TABLE 1
PARTS LIST

C1	1.0μF, 400V DC
C2, C3	330nF, 600V DC
C4, C5	100nF, 600V DC
C6, C7	22μF, 450V DC
C101, C102, C106, C107	100μF,450V DC
C103, C108	470nF, 600V DC
C104, C105, C109, C110	10μF,450V DC
C111	6.8nF, 600V DC
C112, C113	1nF, 1000V DC
R1	not assigned
R2	1.2k
R3, R4	1.2k
R5, R6	100k
R7, R8, R9	75k
R10, R11	1.2k
R12, R13	330k
R14	27k
R15	33k
R16	22k
R17, R18	330k
R19, R20	75k
R21, R22	1.2k
R23, R24	100R
R25. R26	75k
R27. R28	220k
R29. R30	120k

R31, R32	10R
R33	not assigned
R34	39k
R35	12k
R36, R37	100R
R101, R104	100R
R102, R103	1k
R105, R106	1k
R107, R108	25k, 10W
R109, R110	not used
R111, R112	12R
R113, R114	12R
R115, R118	330k
R116, R117	1M
RV1	100k pot.(gain) and knob
P1, P2, P3	10k, 2W pot. and pot. locks
T1	See Text
T101, T102	Hammond 272BX
V1	6SL7GT
V2	6SN7GTB
V3, V4	6J5
V5, V6	See Text
V7	0D3/VR150
D1–D8	FR107 fast recovery, AE P-QFR107
S1	SPST Switch (On-Off)

S2	4PST Switch (Standby)
Aluminum chassis (10″ × 17″ × 3″)	Hammond 1444-32
Aluminum bottom plate (8″ × 16″)	Hammond 1434-30
#1815 pilot and holder	AE P-L114
Fuse 2A and holder	AE S-H258
Octal Sockets	AE P-ST8-801, need 7
Double Phono Jack	AE S-H310
6′ line cord	AE S-W105
Rubber grommet	AE P-H253, need 4
Assorted machine screws, nuts, and washers	

Terminal Strips

AE p/n	**lugs**	**quan**
P-0201H	2	2
P-0301H	3	7
P-0501H	5	6
P-0601H	6	5
P-0701H	7	2
P-0901H	9	7
S-H317	6 screws	1

* AE indicates Antique Electronics. Some parts are available at Radio Shack. All resistors are 2W unless otherwise noted.

ply for the driver to enable the circuit to provide full output. Coincidentally, the bootstrapping increases the stage gain by about 1dB. Negative return for both the 6SL7GT and 6SN7GTB is to a regulated negative 150V supply. The 6SL7GT gets a little extra smoothing through the action of the 22μF capacitor C7 and the 12k resistor R35.

You can see I have interposed 6J5 cathode followers (V3 and V4) between the driver 6SN7GTB and the output tubes in order to avoid capacity and resistance coupling. The 6J5s' cathodes are direct-coupled to their respective output grids through the 100Ω resistors R23 and R24. That avoids a problem called blocking. Norman Crowhurst provides a good explanation of that condition in a reprint of one of his articles in Volume 5 of *Audio Anthology* (pp. 119-22).

At this location you could use any of several *single* triodes, such as 6SF5 or 6C4. You should not use dual triodes at this location since there is a good chance their heater to cathode insulation will fail the first time you turn up the gain control. In the power supply I have provided electrically separated heater supplies to avoid that problem. The 75k resistors R25 and R26 are connected to the respective heater windings on the power transformers so that they follow the cathodes.

The 220k resistors R27 and R28 are returned to the negative power supplies. That allows for fast negative-going waveforms in this part of the signal path. Positive-going signals are not a problem here since they are pulled up by V3 and V4.

If the high voltage was on during warmup it would be possible for R27 and R28 to pull the power tube grids and the 6J5 cathodes all the way down and perhaps further since there would be no load on the power supply. That might cause some failures as well. That condition is prevented by the 120k resistors R29 and R30. During normal operation there is only a small signal difference between the grid and cathode of the output tubes so that there is negligible signal loss.

An important part of this amplifier are eight grid-stopper resistors. The output tubes in particular have very high transconductance, and the amplifier will likely oscillate at radio frequencies on its own if given a chance. Grid-stopper resistors lower the Q (quality factor) of the circuit according to the formula

$$Q = (1/R) \times SQR\ (L/C)$$

As little as 100Ω in a conductor interferes with the RF to the extent that it is eliminated. The audio is left intact. For the control grids any small non-inductive resistor from 1k to 10k will do. You should use 100Ω for R36 and R37 in the output screen grids in order to prevent power loss in the output stage.

The schematic shows the output tubes connected for beam tetrode operation. You can easily go to triode operation by changing the connections of the

TABLE 2
CIRCLOTRON PERFORMANCE SUMMARY (NO FEEDBACK CONNECTED)

CONDITION	WATTS	2 HARM	3	4	5	THD	DF	INPUT	REFERENCE
2X6L6GC									
as Tetrodes	1	0.13	0.12	~	0.09	0.14	2.725		J0801D
2X 125E,									
taps 1&5	10	0.29	0.49	~	0.12	0.66		0.29	J0801E
1000Hz	30	0.37	1.42	0.09	0.24	1.60			J0801F
	35	0.28	5.89	0.26	1.96	6.60	clipping		J0801G
2X6L6GC									
as Tetrodes	1	0.14	0.11	~	~	0.25	2.59		J08010
2X 125E,									
taps 1&5	10	0.28	0.41	~	0.07	0.60		0.30	J0801P
100Hz	30	0.14	1.57	0.20	0.24	2.30			J0801Q
2X6550C									
as Tetrodes	1	0.18	0.11	~	0.08	0.20	2.937		J0801R
2X 125E,									
taps 1&5	10	0.20	0.28	~	0.11	0.40			J0801S
1000Hz	30	0.35	0.78	0.12	0.10	1.00			J0801T
2X6550C									
as Tetrodes	1	0.11	0.10	~	0.11	0.22	1.926		J0801U
2X 125E,									
taps 1&6	10	0.20	0.20	~	0.11	0.38		0.23	J0801V
1000Hz	40	0.42	1.57	0.32	0.57	5.30			J0801W
2X6550C									
as Tetrodes	1	0.14	~	~	~	0.30	2.735		J08011
H300767,									
800R	10	0.37	0.14	~	~	0.50			J08012
100Hz	40	1.84	0.27	0.13	~	2.00			J08013
	50	1.76	0.92	0.26	0.29	2.30			J08014
2X6550C									
as Tetrodes	1	0.13	0.12	~	~	0.20	2.735		J0601A
H300767,									
800R	10	0.07	0.21	~	0.09	0.27		0.20	J0601B
1000Hz	40	0.31	0.14	0.14	0.03	0.45			J0601C
	50	0.36	0.60	0.10	0.34	0.90		0.47	J0601D
	55	0.60	1.65	0.04	0.97	2.00			J0601E
2X6550C									
as Triodes	1	0.14	0.11	~	0.08	0.15	3.63		J1201A
H300767,									
800R	10	~	0.18	~	0.08			0.27	J0501B
1000Hz	20	0.41	0.93	0.10	0.14				J0501A
2X6L6GC									
as Triodes	1	0.13	0.09	~	0.08	0.15	4.053		J0601T
H300767,									
1400R	10	0.10	0.74	~	~	0.90		0.354	J0601U
1000Hz	14	0.14	1.88	0.10	0.41	2.00			J0601V
2X6L6GC									
as Tetrodes	1	0.08	0.11	~	0.09		3.498		J0601K
H300767,									
1400R	10	0.07	0.49	0.03	0.05			0.262	J0601L
1000Hz	30	0.05	1.10	0.04	0.04				J0601M
	35	0.07	1.71	0.07	0.26				J0601N

100Ω resistors R36 and R37. You should connect the screen grid return through R36 directly to the plate of the same tube. Do the same with R37.

AMPLIFIER ADJUSTMENT

The amplifier has three controls that will enable you to adjust the operating conditions. The 10k pot P1 is used to set to minimum the even order (second, fourth, and so on) harmonic distortion at the amplifier output. If the output tubes are not a matched pair, there is a good possibility they will need unequal drive signals.

You will need equipment to measure or observe the distortion. You can use a distortion analyzer such as one of the HP 330 Series as a rejection filter and observe the remaining waveform on an oscilloscope. Adjust the pot P1 for minimum signal on the scope. The HP 330 Series instruments measures total harmonic distortion (THD).

Another method involves the use of one of the virtual measurement attachments connected to your PC. I used the system manufactured by Pico Technology to record all the harmonic distortion measurement results in this article. This system uses FFT to provide a measurement of each of the individual harmonics, on the fly. It allows you to observe the second harmonic while adjusting P1 for minimum.

If you don't have access to one of these measurement systems, then it is best to invest in a matched pair of output tubes. If you do, then you should replace the pot and the 22k resistor R16 with a single 27k resistor. That way the output of the driver circuit will be balanced into your output tubes.

The 10k pot P2, along with 75k resistors R19 and R20, is used to balance the DC conditions in the output stage. First be sure to disconnect any input signal and then set the gain control to minimum. Use a DC voltmeter set to a low range to measure the voltage difference between the test points TP1 and TP2. Then adjust P2 for 0V.

With the unit properly adjusted, there will be no current in the output transformer caused by the output tubes. A small current remains in the output transformer. It is primarily the diff amp return flow from the negative supply.

The 10k pot P3 and the 39k resistor R34 set the bias for the output tubes by way of V3 and V4. It is possible to set a wide range of operating points. You should monitor the output tube current by measuring the voltages across the 10Ω sampling resistors R31 and R32 while setting the bias.

Take care not to exceed the plate dissipation ratings of the output tubes you have chosen. With no signal applied, the voltage at these test points is quite low. However, if a signal were applied to the input, lethal audio voltages may be present.

Don't forget to buy some pot locks for each of these controls. After setup, you will want to be sure they don't inadvertently get changed.

THE POWER SUPPLY

You will be able to avoid an expensive and perhaps hard-to-find special power transformer for this unusual amplifier. It is possible to use a pair of identical off-the-shelf units. I used Hammond 272BXs in the final version of my amp. There is even a discount available when you buy two of them.

The resulting two high voltage supplies are identical. Each consists of a full bridge of silicon power diodes, D1 to D4 and D5 to D8 (*Fig. 2*). For the positive voltages each feeds into a 100µF capacitor, C101 and C106. That is smoothed by 100Ω resistors R101 and R104 feeding into 100µF capacitors C102 and C107.

The resulting 395V will drop to about 380V under full load conditions. Some may be tempted to use choke filtering at this point. You will find it unnecessary since the remaining ripple on each of the power supplies is cancelled in the output transformer. There is one not so obvious advantage the RC filter has over the LC filter. It is not resonant, so will not give your program material some extra unwanted coloration.

I have included the 330nF caps C103 and C108 as high-frequency bypasses. They are connected electrically and physically as close as possible to the output tubes and transformer. The 330k resistors R115 and R118 safely dis-

FIGURE 3: Circlotron output transformer—alternative 1.

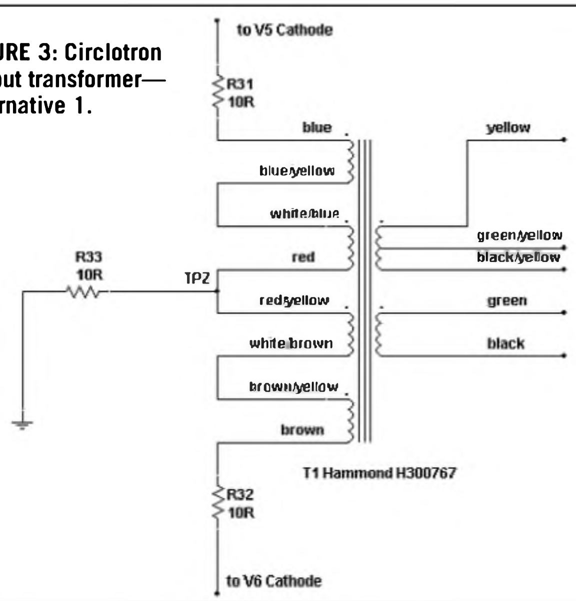
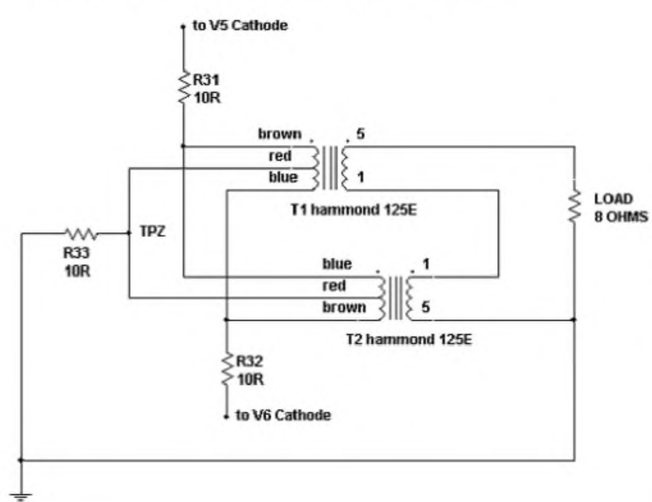


FIGURE 4: Circlotron output transformer—alternative 2.



charge the capacitors should the load become unhooked.

The negative voltages are derived in a similar fashion. You don't need as much current to be supplied by this part of the circuit. To avoid some heating in the transformers the 1k resistors R103 and R106 are used to limit the charging current into the 10µF caps C105 and C110. They also prevent the negative supplies from exceeding the voltage rating (385V) of the electrolytics I used. You could use 450V caps.

I again used RC filtering through the networks formed by R102/C104 and R105/C109. A safety discharge path is provided by the 1M resistors R116 and R117. The negative supply is further smoothed and regulated by the 0D3/VR150 gas regulator tube and the 25k resistors R107 and R108. C111, the 6.8nF cap, forms a high frequency bypass across the VR tube to remove some of its noise. R109 and R110 are not assigned.

I wanted the amplifier to be as free as possible of potential problems. Because output of the Circlotron power stage comes from the cathodes, there is a very good chance of heater to cathode insulation failure. Each transformer has a 6V winding so that they are used here to separately power their respective heaters. There is also a 5V winding on each transformer. I connected these in series and then used 12Ω resistors R111 through R114 to supply the heaters of V1, V2, and the pilot light.

The four-pole, single-throw switch S2 provides the amplifier with a standby and warmup mode. Pins 3 and 7 of the 0D3/VR150 regulator are wired into the primary of the power supply. The amplifier will not power up unless the regulator is plugged in.

The 1nF, 1kV caps C112 and C113 will reflect some of the crud coming off the line back into the wall. Be sure to use shielded cable at your input to prevent internal feedback and coupling to the power supply.

DETERMINATION OF OUTPUT STAGE LOAD IMPEDANCE

If you check your tube manuals you won't find much information regarding proper push-pull loading for the power tubes in this circuit when the plate and screen are run at the same voltage. Fortunately, the circuit lends itself to experimental determination of these conditions.

I had a relatively large output transformer of low DC resistance in my junk box. I connected it to the amplifier as shown in *Fig. 4* with no loading on its secondary. Then I used several large power resistors in the 1k and 2k range in various combinations to load the amplifier at points C and D. From these measurements I was able to determine the optimum loading for maximum power output.

The curves relating output power and distortion for pentodes and beam tetrodes indicate that as load imped-

TABLE 3
CIRCLOTRON PERFORMANCE SUMMARY (10dB FEEDBACK CONNECTED)

CONDITION	WATTS	2 harm	3	4	5	THD	DF	Input	Reference
2X 6L6GC									
as Triodes	1	0.11	0.13	~	0.10	0.06	12.05		J0801A
H300767,									
1400R	10	0.14	0.25	~	0.06	0.30		1.16	J0801B
1000Hz	14	0.17	0.79	~	0.20	0.80			J0801C
		R_{fb} = 8.4K							
2X6L6GC as									
Tetrodes	1	0.09	0.13	~	0.10	0.07	8.973		J0801H
2x 125E,									
taps 1&5	10	0.21	0.19	~	0.10	0.23		0.95	J0801I
1000Hz	30	0.30	0.67	~	0.19	0.70			J0801J
	35	0.18	5.35	0.19	2.19	6.50	clipping		J0801K
		R_{fb} = 15K							
2X6L6GC as									
Tetrodes	1	0.15	~	~	~	0.15	6.503		J0801L
2X 125E,									
taps 1&5	10	0.21	0.14	~	~	0.75		0.95	J0801M
100Hz	30	0.19	0.63	0.09	0.12	0.80			J0801N
		R_{fb} = 15K							
2X6550C as									
Tetrodes	1	0.10	~	~	~	0.20	8.10		J1201D
H300767,									
800R	10	0.09	~	~	~	0.15		0.67	J08016
100Hz	40	0.42	0.12	0.06	~	1.60			J08017
	50	0.49	0.44	0.09	0.15	2.30			J08018
		R_{fb} = 16.3K							
2X6550C as									
Tetrodes	1	0.10	0.10	~	0.10	0.06	9.137		J0601F
H300767,									
800R	10	0.10	0.07	~	0.11	0.11		0.69	J0601G
1000Hz	40	0.08	0.13	~	0.06	0.14			J0601H
	50	0.04	0.23	~	0.15	0.30			J0601I
	55	0.29	0.98	0.23	0.64	1.10			J0601J
		R_{fb} = 15.7K							
2X6550C									
as Triodes	1	0.10	0.11	~	0.09	~	10.37		J0501F
H300767,									
800R	10	0.08	0.08	~	0.08	~		0.89	J0501E
1000Hz	20	0.13	0.54	0.09	0.16	~			J0501D
		R_{fb} = 11.2K							
2X6L6GC									
as Triodes	1	0.14	0.10	~	0.08	0.15			J1201B
H300767,									
1400R	10	0.13	0.27	~	~	~	10.255		J0601W
1000Hz		R_{fb} = 8.4K							
2X6L6GC as									
Tetrodes	1	~	~	~	~	0.08	8.9		J0601O
H300767,									
1400R	10	0.12	0.21	~	~	0.23		0.80	J0601P
1000Hz	30	0.14	0.38	~	~	0.40			J0601Q
	35	0.22	0.73	0.05	0.23	0.80			J0601R
	38.5	0.24	1.83	0.04	1.05	2.20			J0601S

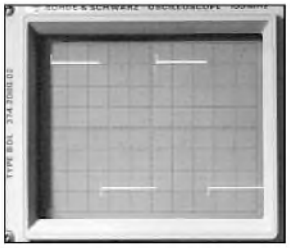


PHOTO 2: 1kHz with 10dB feedback.

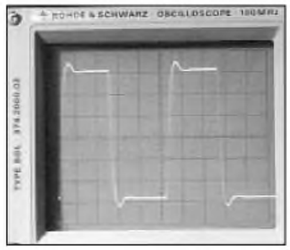


PHOTO 3: 10kHz with 10dB feedback.

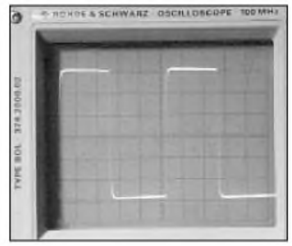


PHOTO 4: 1kHz with zero feedback.

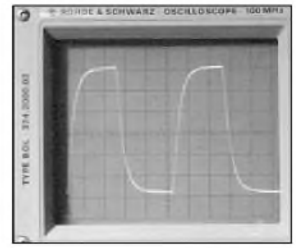


PHOTO 5: 10kHz with zero feedback.

ance is decreased the second harmonic will increase but the third harmonic decreases. The second harmonic will be cancelled in a properly balanced push-pull amp. For that reason I chose a load impedance about 10% less than optimum in order to reduce the third harmonic component.

THE OUTPUT TRANSFORMER

I tried three alternative hookups. The first and best of these is a custom-wound unit from Hammond (*Fig. 5*). It has four separate primary windings which allows matching 6L6GCs to 1400Ω or 6550Cs to 800Ω. The Hammond part number is H300767.

You could also use the 800Ω connection with KT88s or EL34s. The 1400Ω connection would be OK for KT66s. Secondary of the transformer is the normal Hammond configuration of two windings allowing 4, 8, and 16Ω loading.

A very good second alternative is that using a pair of Hammond 125E universal output transformers (*Fig. 4*). It occurred to me that if you could parallel-connect resistors, capacitors, or inductors to reduce the impedance, then why not transformers? The transformer primaries are connected in parallel so that they will reflect half the data sheet impedance if similar secondaries are used.

You will notice, in *Fig. 4*, that the transformer primaries are connected to each others' opposite connection, that is blue to brown and vice versa. The transformer secondaries are also reversed, so that the output signals are still additive. In many lower-cost audio transformers the secondary is wound around one end of the primary. This results in more stray capacity on one end of the primary than on the other. For the Hammond 125 series the secondary is wound around the brown end of the primary; this may cause problems unless you give it some consideration. In a push-pull amplifier, the high frequency balance of the output stage is disturbed, but by paying attention to this simple relation, you can avoid such a condition.

Keep in mind they are rated for 15W of audio each. That would be OK for the 6L6GC version of the amp. If you connect the secondary taps 1 and 5 of each transformer in series to an 8Ω load, you will reflect a load of 1500Ω to the output tubes, which will make them very happy. The damping factor is not as good as when the H300767 transformer is used.

The third transformer hookup allows you to use some of the commonly available push-pull output transformers (*Fig. 5*). The key here is that the two halves of the primary must be electrically isolated. You will find these in old amplifiers as well as replacement units now on the market.

If the two halves of the primary are paralleled, the impedance will be one-quarter of the normal hookup. For example, if you were lucky enough to have a transformer made for push-pull 6L6s with plate-to-plate impedance of 6600Ω, then the new connection would give you 1650Ω. That is close enough to satisfy the 6L6GCs in this version of the Circlotron.

You will still need to provide a DC return for the negative power supply. That is possible by using a pair of resistors bridged across the transformer primary. See R38 and R39 in *Fig. 1*. The resistances used will need to be rather large in order to avoid audio signal power loss at this point, since they form a load across the transformer primary.

That in itself is not a problem. However, DC return current in these same resistors adds too much to the output tube bias, tending to bias them off. For this reason I do not recommend this hookup.

Fortunately, most of the output transformers I just referred to come with ultralinear (UL) taps. That will neatly step around the biasing problem I referred to. The taps are normally at 20% or 43% of each half of the primary. That means you can use a much smaller bridging resistance which will have much less effect on the output tube bias. Even so, you will need to make a small adjustment in the bias network to bias the tubes properly.

For example, I had set the idling current of each 6L6GC in the circuit to 42mA in the circuit of the first output transformer alternative. When I connected this transformer alternative, which

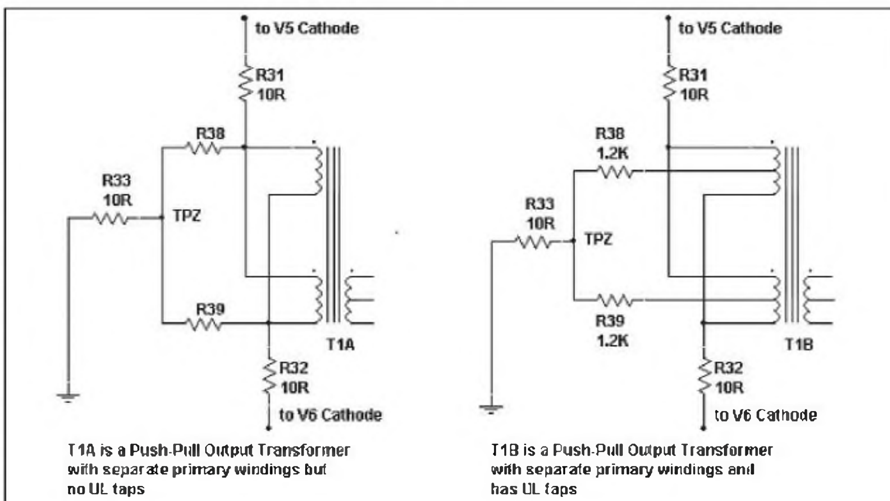


FIGURE 5: Circlotron output transformer—alternative 3.

has 43% UL taps, the new biasing conditions had reset the idling current to 30mA. Measuring the voltage drop across the bridging resistors, I found that 5.72V had been added to the bias.

The good news is that the audio signal voltage developed across the bridging resistors when the output was set to 12.5W was only 18.7V rather than 131V. That had allowed the bridging resistors to be reduced by 86%, a very worthwhile improvement in this part of the circuit.

PERFORMANCE

Summaries are given in *Tables 2* and *3* so that you can compare the amplifier performance both with and without feedback. Having two kinds of output transformers, two kinds of tubes connected as triodes or tetrodes, and measurements with and without feedback resulted in a dazzling array of test data. I managed to get 17 (not 16 as you would think) sets of test data with which to bewilder you. However, out of all this you will be able to arrive at some interesting conclusions and select a combination that best suits your needs.

Feedback has been normalized at 10dB so that valid comparisons can be made. In each case the feedback resistor needed to connect to the 8Ω tap is noted as R_{fb}. The amplifier AC balance adjustment has been set for best results in each case at 10W output. This may not be the best setting for lower power outputs.

The load resistor used in testing consisted of six 50Ω and one 150Ω resistor (all 10W) connected in parallel. The result measured 7.86Ω and was connected to the 8Ω output in each case.

I used no output matched pairs or special tube selections to get this performance. Cathode current for the 6550Cs was set at 70mA each for these tests. I set the cathode current for the 6L6GC tests at 53mA each. You could set these higher and reduce the odd order (third, fifth, and so on) distortion. The line voltage measured 116.

Individual harmonic distortion and total harmonic distortion have been measured at several power levels. Damping factor (DF) was calculated from the output voltage drop caused by connecting an extra 30Ω load in parallel to the 8Ω output connection at a 1W level. The columns marked "Input" contain the voltage needed to produce 10W at the output. Columns marked "Reference" are to keep me on track while sorting out this mass of data. The Lodestar AG-2601A test signal generator distortion measured 0.03%.

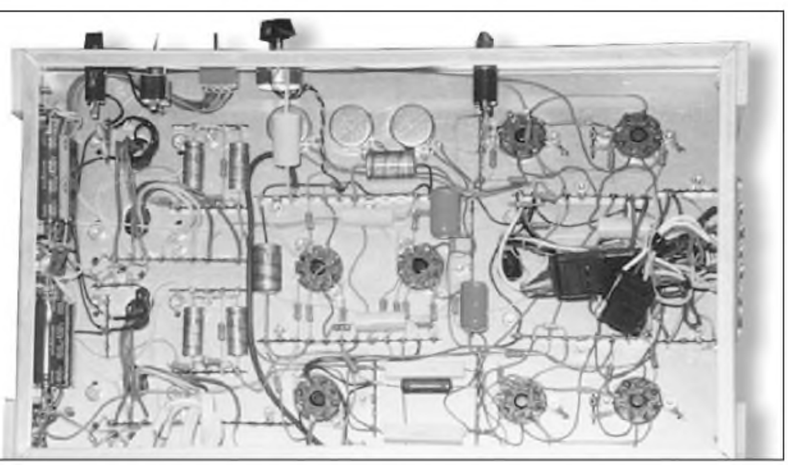
While recording the amplifier response to the 100Hz square wave, I made an interesting observation. Low-frequency phenomena on the scope don't photograph very well. That's because most scopes use a P31 phosphor, which doesn't have a lot of afterglow. I recall selling many scopes with a longer persistence phosphor, the P7, to users who needed the longer yellow afterglow.

This continued long after the advent of variable persistence scopes because the price and performance was good. For these measurements I used the storage capabilities of the Pico Technology Virtual Instrument. The responses at 1 and 10kHz were photographed straight from the scope (*Photos 2-5*).

With no feedback there is no overshoot. With 10dB feedback there is a minor overshoot but no ringing. This could be eliminated with phase correction networks. Form of the correction network will depend on the output transformer you use. All of the square-wave measurements were made at a level of 12V peak-to-peak.

In the photo of the amplifier underside (*Photo 6*) you may notice on the right-hand side what looks like automotive trailer electrical connectors. These formed a convenient way for me to quickly change the various output transformer schemes I wanted to try. You won't need those in your finished amplifier.

PHOTO 6: Underside of unit.



DISCUSSION OF RESULTS TABLES

Even without full loop feedback, the Damping factor for the tetrode-connected examples is a very triode-like 2–3. That will provide very good control of your loudspeaker. The triode-connected cases are even better in the range of 3 and 4.

Had the tetrodes been connected in the regular fashion (such as used by common amplifiers—Williamsons, Dynas & Mullard 520s), damping factors would have been in the 0.12 to 0.20 range before feedback had been applied. That is a huge difference.

Using two of the Hammond 125E output transformers as an alternative to the large output transformer results in a small decrease in the DF. That is caused by more copper losses due to the higher winding resistance in the signal path. Distortion at high levels and low frequencies increases as well. This is a result of the limited amount of iron available in these transformers. Let's hope Hammond comes up with a 125F—still a very usable alternative.

Output tubes connected as triodes result in even better DF, with or without feedback than when tetrode-connected. Distortion is marginally less at low levels, but power output capability is limited. ❖

EQUIPMENT USED

HP 200CD Oscillator
Lodestar AG-2601A Sine/Square Wave Generator
HP 302A Wave Analyzer
HP 334A Distortion Analyzer
Rohde & Schwarz BOL 4 Trace 100 MHz Scope
Radio Shack 22-168A DMM
Sanwa AX 303-TR Analog Multimeter
PicoScope ADC-100 Virtual Instrument
Electronic Workbench Software

The Infinite Box: Constructing a Subwoofer, Part 2

Now that you've put together the subwoofer enclosure, it's time to test the finished box. **By G. R. Koonce and R. O. Wright, Jr.**

The first playing of the completed subwoofer involved driving each voice coil independently to verify they both worked. Normal listening won't tell whether one or both coils are contributing. Next, GRK's normal test amplifier drove both voice coils in parallel, resulting in blowing the low amperage fuse protecting this channel. Since the unit was working, we decided to delay further listening and do performance testing until we could obtain a bigger amplifier.

We did initial testing on the subwoofer with the two voice coils connected in parallel and with no fiberglass lining on the walls of the DA (dead air) volume. *Figure 14* shows the input impedance to the subwoofer. The IB (infinite box) restricts the maximum low-frequency driver impedance to about 15.3Ω; it had exceeded 80Ω while baffle-mounted for the T/S parameter measurements.

The next testing consisted of measuring the cone (front) and rear-leakage (through the damping material) acoustic outputs via near-field technique. Note that near-field testing requires a low signal level to prevent microphone overload, and the reported results may not accurately represent the system's high-level response. *Figure 15* shows the measured magnitude of the cone and rear leakage with the proper relative levels for area difference. *Figure 16* shows the phase-shift results for the same signals.

For those not familiar with our earlier published IB work, we summarize the curves shown as follows:

1. The cone magnitude response for an IB has a slope near 12dB/octave from low frequency up to the peak value; there is no dip as with a VB (vented box). Here the peaking in the driver response modifies this slope somewhat.
2. The rear-leakage magnitude response for an IB has a slope near 6dB/octave up to some limiting frequency. The cone and rear-leakage levels will converge to the same level at very low frequency. The rear-leakage magnitude becomes rather rough at high frequency and can be suppressed by lining the dead-air volume walls with fiberglass. This was initially not done because of the intended subwoofer application, but should be done for a normal woofer application.
3. The difference in level between the cone and rear-leakage magnitude curves is the attenuation through the damping material at each frequency.
4. At low frequency the difference between the cone and rear-leakage phase shift will be 180°, because the cone rear radiation is out-of-phase with the cone front radiation. As frequency rises toward 100Hz, the two will tend to be more in-phase due to phase shift through the damping layers. Thus, over a portion of the bass region the rear leakage tends to augment the cone response—to a small degree—rather than suppress it.

At higher frequencies the phase shift of the rear leakage becomes complex and will add ripple to the system response as rear leakage alternately adds to or subtracts from the cone response. The magnitude of this ripple is insignificant compared to what your room will do to the bass response! You can find the phase shift through the damping material by subtracting the phase difference between the cone and rear-leakage curves from 180°.

SYSTEM FREQUENCY RESPONSE

You can combine the cone and rear-leakage responses to produce the system frequency response. What you call the "system response" is not straightforward because of the DSL (diffraction spreading loss) effect. Most design software and reported system responses ignore the DSL. Thus, they are accurate only if you use the enclosure built into, or pushed right up against, the rear wall.

Used away from the wall, the system's true on-axis response takes a low-frequency dip of up to 6dB and the true f_3 value is much higher than reported. Experts say you should consider the on-axis response and not the half-space anechoic response so often reported. We will examine the subwoofer's on-axis response out in the room and up near the rear wall.

We used special software to compute the system response from the measured cone and rear-leakage responses. This software includes the effects of

PHOTO 1: Front view of completed subwoofer.

box depth and box width and what they do in terms of the DSL[1]. The software will also compute the response ignoring the DSL for comparison with other speakers.

Figure 17 shows the subwoofer on-axis response located out in the room; i.e., including the full 6dB DSL effect. The true on-axis f_3 below 40Hz is a good result. This response shows that you could use this driver as a woofer up to about 400Hz in a three-way IB system with a full 6dB of DSL compensation. Note, however, we consider 6dB of DSL compensation a bit high for a floor-standing box with a low-mounted woofer.

You don't really wish to push an IB tight against the rear wall, because it blocks the rear openings, but if you push the subwoofer close to the rear wall the on-axis response will resemble *Fig. 18* with a lot of peaking and a reported f_3 below 30Hz. As with the CB (closed box) and VB, the peaking built into the driver clearly shows in the system response.

It is clear the system has more than the optimum amount of peaking to compensate for DSL. Is it possible to reduce the peaking without designing a new, bigger IB? During the development of the IB concept, with smaller drivers, we noted that lining the DA volume walls with fiberglass had two effects.

First, it reduced the rear leakage at high frequency, and, second, it tended to reduce the peaking around 100Hz by about 1–2dB. We thought that this fiberglass lining was not needed with the subwoofer so we omitted it during construction, but now we added it to learn what effect it has.

We cut batts of nominal 2″-thick fiberglass to fit between the bracing blocks. The box requires four batts 9½″ × 6″ and four batts 9½″ × 7″, installed with rubber cement, but actually wedged between the bracing blocks.

The input impedance was virtually unchanged by the addition of the fiberglass. The peak impedance changed from 15.3 to 15.9Ω, probably a slight change in box sealing due to removing and remounting the driver. *Figure 19* shows the cone and rear-leakage magnitude responses with this lining. Compared to no lining (*Fig. 15*), there is not much change in cone response, but the rear leakage is reduced around 70 to 100Hz.

The cone and rear-leakage phase-shift responses (*Fig. 20*) exhibit slight changes from what was shown in *Fig. 16*. The complex combining of these responses produces the on-axis response with 6dB DSL correction (*Fig. 21*). This shows perhaps 1dB less peaking than without the lining (*Fig. 17*). *Figure 22* shows the on-axis response without DSL correction, nearly the same as with no lining (*Fig. 18*). All detailed listening to the subwoofer was with the fiberglass lining.

Now if you plan to use the system only as a subwoofer up to about 100Hz, you get entirely different values for the –3dB cutoff frequencies. One way to define the passband is to designate the peak of the response as 0dB and see at what frequencies the response hits –3dB. This is a very tough requirement, and a better approach would be to set the peak to +3dB and give the frequency range for a ±3dB response. *Table 6* shows these frequency ranges for both ways of defining the passband. The values for the IB (*Fig. 22*), the 2.85ft^3 CB (Fig. 3) and the 2.85ft^3 VB (Fig. 6) are shown.

Whatever way you choose to view the passband, the VB is superior in terms of the lower f_3 value. The IB will be a more highly damped system with a superior transient response over the VB. Also note the CB and VB have a net internal volume of 2.85ft^3, while the gross internal volume (from the front panel to the back board) of the IB is only 1.97ft^3.

You could extend the lower f_3 value for all these box types downward by using one of those subwoofer (plate)

TABLE 6
PASSBANDS FOR VARIOUS BOXES WITHOUT DSL CORRECTION

BOX	0 TO –3dB RANGE	±3dB RANGE	SEE
Closed Box	39 to 140Hz	30 to >300Hz	Fig. 3
Vented Box	28 to 120Hz	22 to 180Hz	Fig. 6
Infinite Box	46 to 130Hz	38 to 200Hz	*Fig. 22*

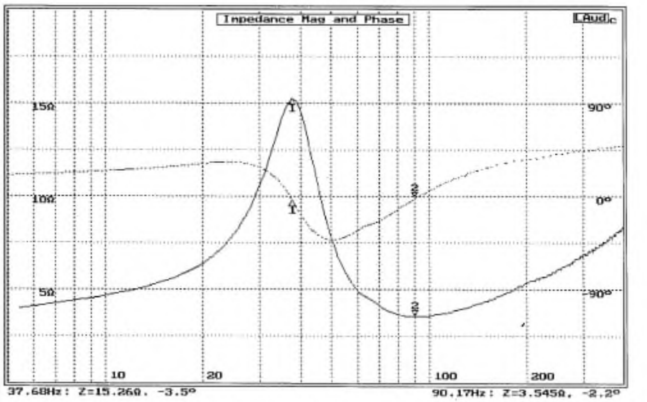


FIGURE 14: Input impedance to subwoofer. B-2110-14

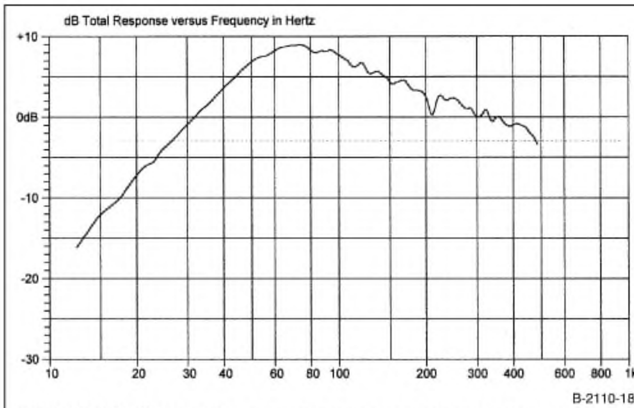
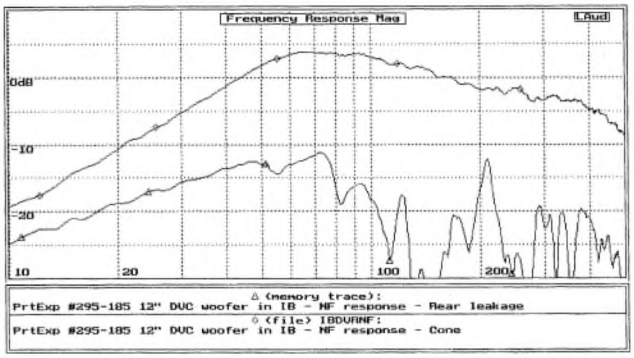


FIGURE 18: Subwoofer on-axis response close to rear wall.



B-2110-15

FIGURE 15: Magnitude of cone and rear-leakage responses.

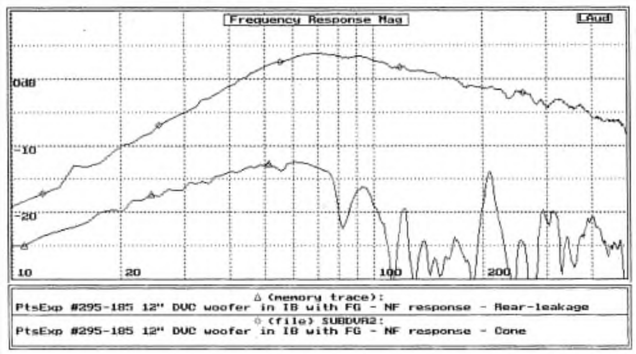
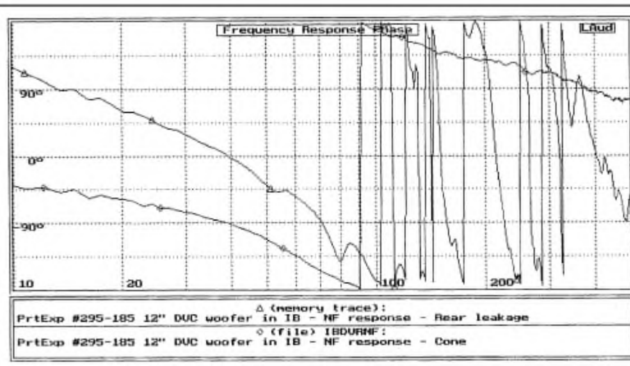


FIGURE 19: Magnitude of cone and rear-leakage responses with lining. B-2110-19



B-2110-16

FIGURE 16: Phase shift of cone and rear-leakage responses.

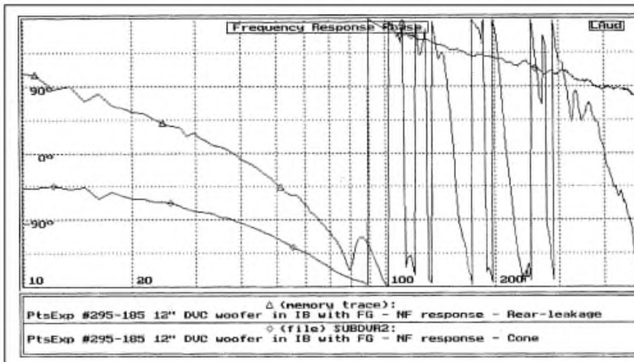


FIGURE 20: Phase shift of cone and rear-leakage responses with lining. B-2110-20

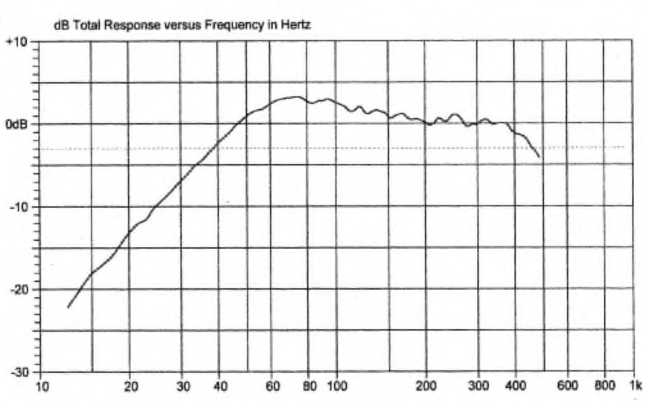


FIGURE 17: Subwoofer on-axis response out in room. B-2110-17

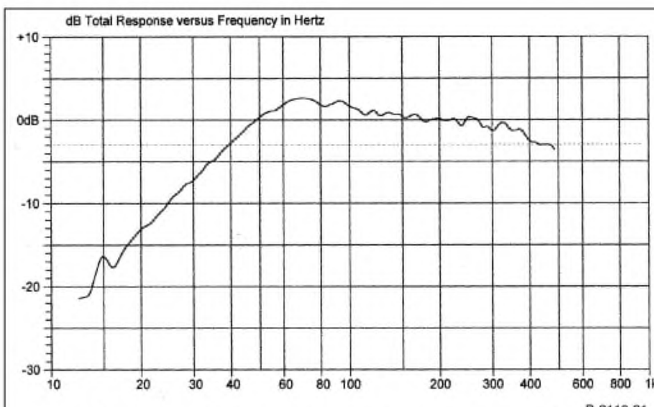


FIGURE 21: Subwoofer on-axis response out in room with lining.

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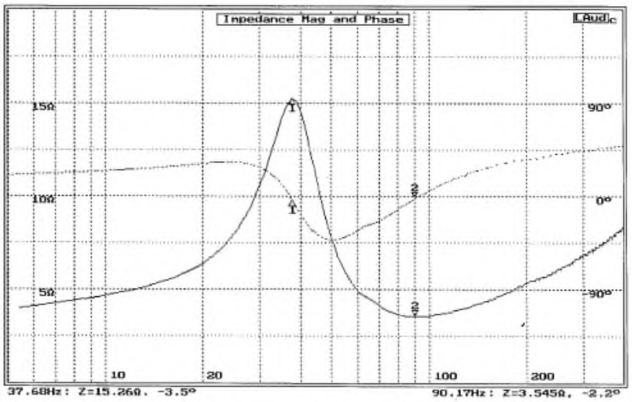
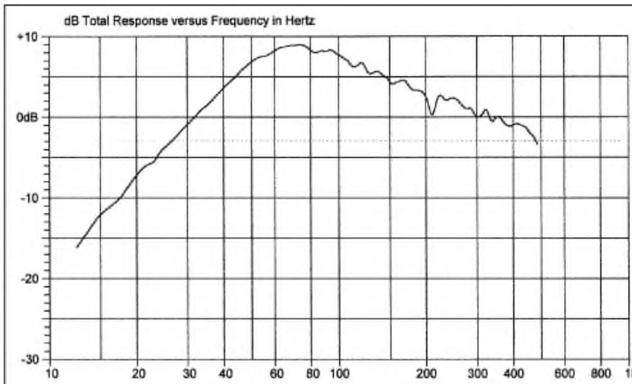
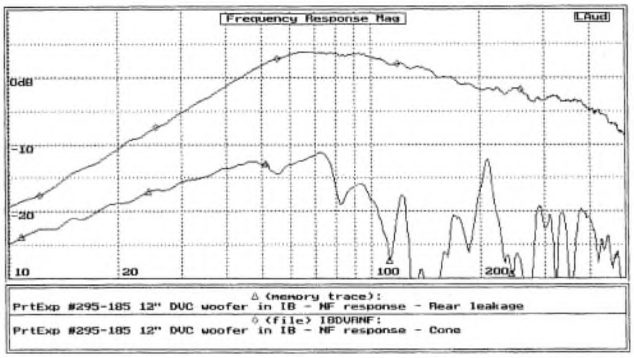


FIGURE 14: Input impedance to subwoofer. B-2110-14



B-2110-18

FIGURE 18: Subwoofer on-axis response close to rear wall.



B-2110-15

FIGURE 15: Magnitude of cone and rear-leakage responses.

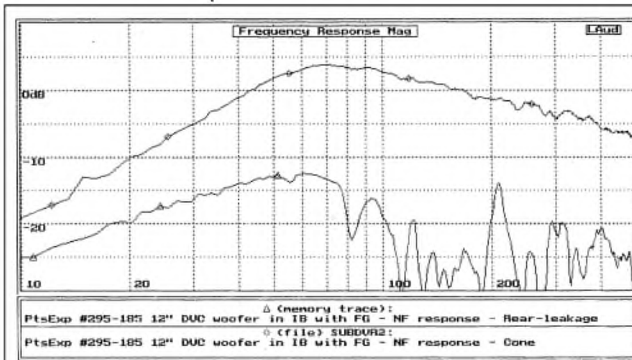
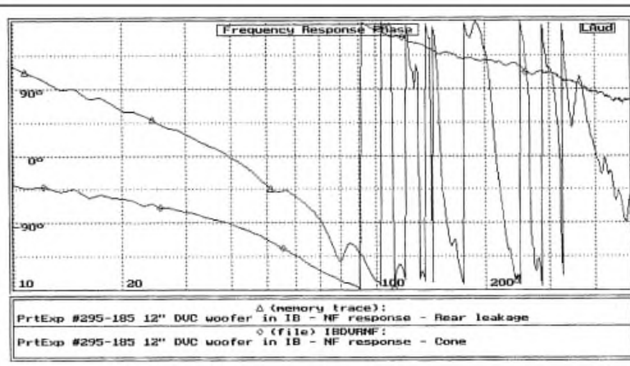


FIGURE 19: Magnitude of cone and rear-leakage responses with lining. B-2110-19



B-2110-16

FIGURE 16: Phase shift of cone and rear-leakage responses.

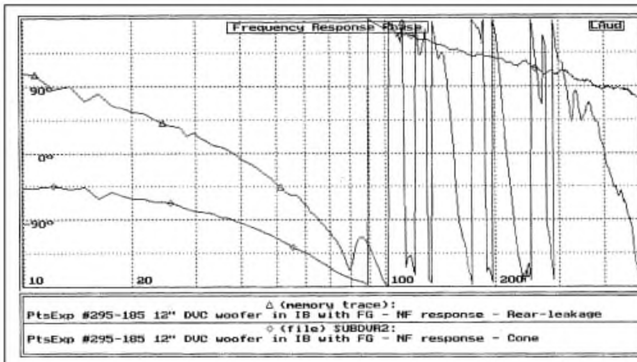


FIGURE 20: Phase shift of cone and rear-leakage responses with lining. B-2110-20

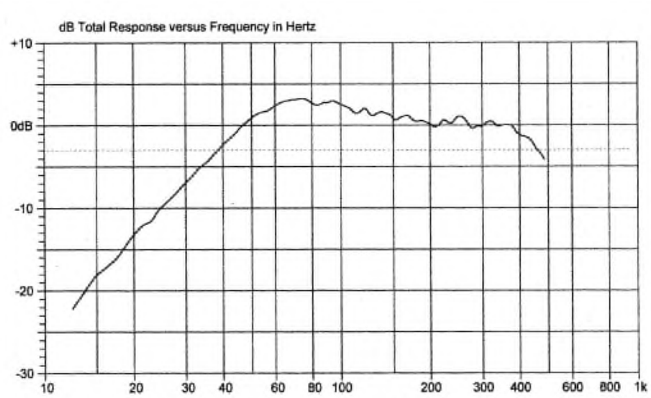
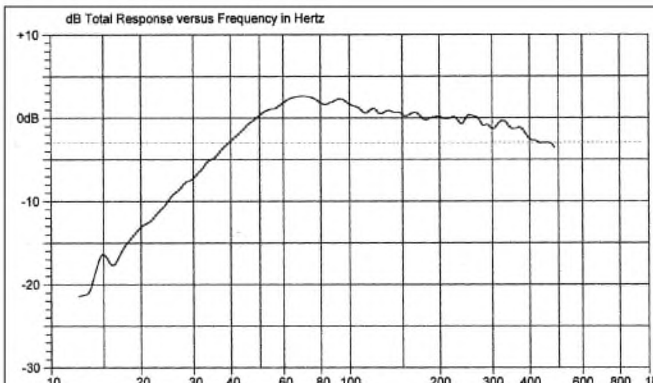


FIGURE 17: Subwoofer on-axis response out in room. B-2110-17



B-2110-21

FIGURE 21: Subwoofer on-axis response out in room with lining.

amplifiers that peaks in the 30–35Hz range[4,5]. You can improve the IB passband by using the proper crossover shape as shown later in the section entitled "An Inexpensive Crossover Approach."

GROUP DELAY

The measured group delay for the subwoofer before adding the fiberglass lining is shown in *Fig. 23*. Ignoring the wiggles caused by noise below 20Hz and the spike at 100Hz where the software failed to sort out the rapid phase rotations, the group delay stays below 8ms. This is an impressive result for a large woofer, an attribute we have learned to expect from the IB. The low, smoothly decaying group delay is an indicator of a good transient response. The fiberglass lining had virtually no effect on the system group delay.

LISTENING TO THE SUBWOOFER

By G. R. Koonce

The subwoofer, with fiberglass wall lining, was used in listening tests in a two-channel stereo system. The "main" speakers were the small CSB computer speakers covered in reference 6. This CB system provides a natural second-order high-pass function with f_3 at about 90Hz.

The main channel outputs were also fed to a passive network (covered later), which summed them and provided an equalized second-order low-pass function, making the subwoofer's useful output range 25 to 100Hz. The network's output drives both channels of a stereo amplifier driving the subwoofer's voice coils. This provided a subwoofer response that mated well with the main speakers and gave lots of power for driving the subwoofer.

The main speakers were driven with inverted polarity after listening to both options. Driving them at normal polarity produced a major response "hole" from 120Hz to down below 100Hz. It is impossible to predict which polarity will be better because of a possible inversion in the subwoofer's amplifier. All speakers were well out from the rear wall with the subwoofer centered between the main speakers.

It quickly became evident that music on FM radio and most popular-music CDs contain little deep bass. We used no movie soundtracks or demonstration CDs. Since the small CSB boxes were driven full bandwidth they set the power limit, so I set the subwoofer level a few dB higher to exercise it.

Where I mainly liked the subwoofer advantage was the depth it gave orchestral music. Strange as it may seem, I believe the subwoofer would allow you to enjoy classical music played at a lower level than without the subwoofer. I'm not sure neighbors in an apartment house would agree your playing level was reduced! I truly enjoyed the marches on *World Famous Marches* (NAXOS Int'l 8.990010) because of the depth the subwoofer added.

I found a couple of CDs that deserve mention. As I listened to them, I viewed the electrical signal spectrum via ⅓ octave RTA.

Back when working on the low-frequency waveguide, I found that Jennifer Warnes' *The Hunter* CD (01005-82089-2-2) had some cuts with 50% of the power below 100Hz. This is a good CD to exercise a subwoofer, because many cuts have 31 to 63Hz content along with good music. Cut #8 ("Way Down Low") has a strong signal that really makes the cone of a subwoofer move about, but the RTA showed it to be in the 63Hz band. The winner on this CD is cut #9 ("The Hunter"), which has good 31 to 63Hz content and transients to verify a subwoofer's damping.

The killer CD, Mickey Hart's *Planet Drum* (Rykodisk RCD 10206), was sent to me by a friend. On the "Temple Caves" cut is something called an "Earth drum." The subwoofer cone

TABLE 7
IB INFORMATION FOR USING 15″ DAYTON #295-190 WOOFER

Design CB-prototype Q_{TC} was 0.8 (will add some response peaking)
Dead-air volume = 2.1ft^3 (3,629 in^3)
Damping layer thickness—6 layers for 6″ total thickness
Damping layer tunnel dimensions—16 11/16″ by 15 5/8″ by 6¼″ deep
Sheets of Owens-Corning #705 material required—two 2′ × 4′ sheets

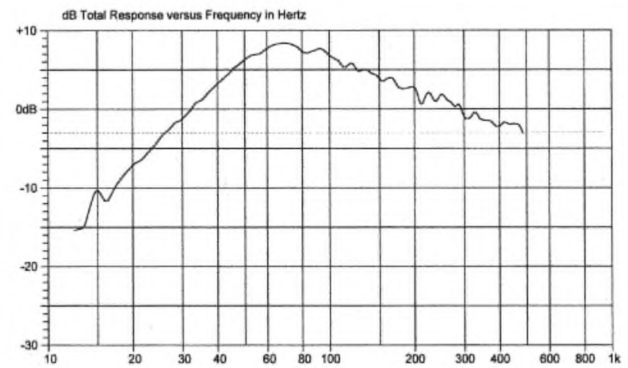


FIGURE 22: Subwoofer on-axis response close to rear wall with lining. B-2110-22

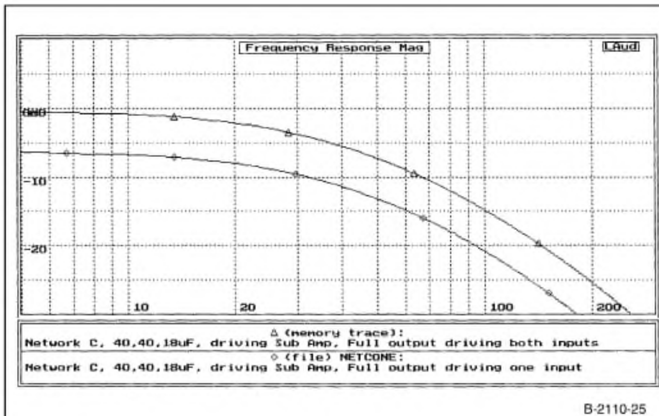


FIGURE 25: Measured electrical response of network.

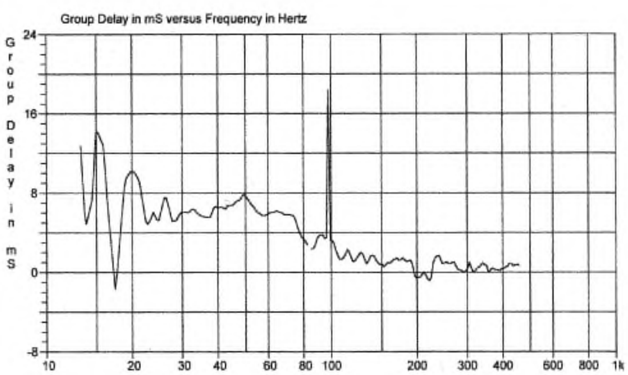


FIGURE 23: Measured group delay of subwoofer. B-2110-23

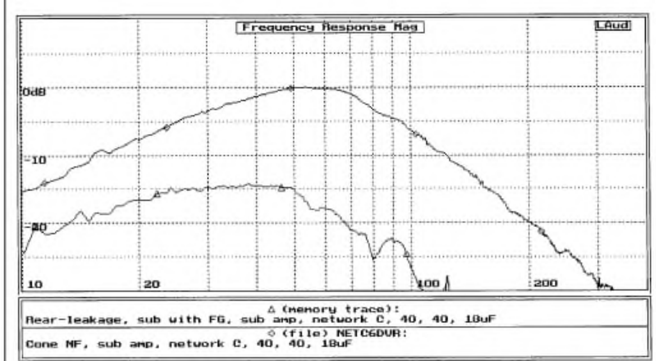


FIGURE 26: Measured cone and rear-leakage magnitudes with network. B-2110-26

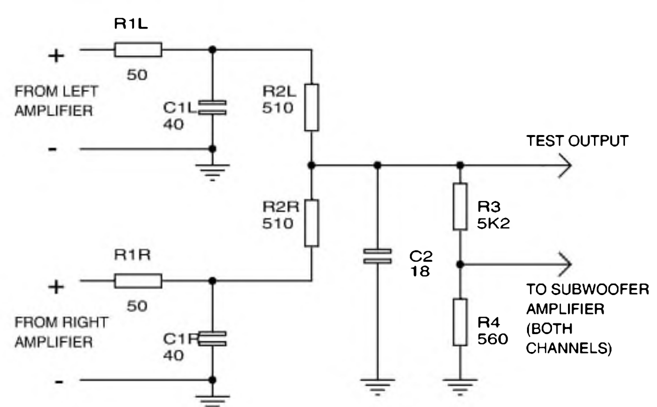


FIGURE 24: Schematic for simple crossover/summer network.

showed great excursion at a frequency so low you could almost count the cycles. The lowest band (31Hz) on the RTA jumped way up, but I believe the frequency may be lower because it is the kind of thing you feel as much as hear. Not a safe cut to play on a VB with a small woofer.

I found the subwoofer highly damped via watching the cone and listening, and it works as well as the smaller IB systems constructed. The excessive response peaking—much of it built into the driver—is best corrected via a simple equalizer as used in my listening session. The simple passive network used worked well, but since the small main speakers see full bandwidth they will limit the playing level below the subwoofer's limit. This might be a good thing!

USING A SUBWOOFER

By R. O. Wright, Jr.

To properly integrate a subwoofer into a two-channel stereo system requires the proper crossovers and often an additional amplifier. Normally the crossover to a subwoofer occurs in the 50 to 150Hz frequency range, with mounting evidence that crossover at or below 80Hz will mask the subwoofer's location. Using a single subwoofer requires that the left- and right-channel signals be summed in some way. This can be accomplished by using a dual-voice-coil (DVC) driver in the subwoofer or electrically summing the two signals.

There are amplifiers built just for subwoofers that in many cases contain the needed summing and crossover networks. The crossover options available are electronic crossovers, passive crossovers at speaker signal level, or passive crossovers at line signal level.

1. Passive crossover at speaker level—It is possible to treat the subwoofer just like the other drivers in your system and use passive crossovers directly at speaker level. This approach requires a subwoofer with a DVC driver or two subwoofers. You need two of each network type: a low-pass (LP) network for each subwoofer voice coil and a high-pass (HP) network for each of the main speakers.

These represent a major design problem for the reader because of the low frequency involved. Inductor values for low-frequency crossovers are quite high and can result in high insertion losses due to the fact they are series components. You can minimize insertion losses by using larger wire gauges, but be prepared to pay for more copper.

Parts Express offers individual HP and LP crossovers at 80 or 100Hz. They additionally offer two units (#260-220, 8Ω, 150Hz; and #260-221, 4Ω, 130Hz) that contain a second-order LP for the subwoofer and a first-order HP for the main speaker. Again, you would need two of these units.

2. Passive crossover at line level—The ideal place to use a passive crossover is at line level ahead of the amplifier. Parts Express offers a line of F MOD crossovers by Harrison that simply plug in ahead of the amplifier. These units are sold in pairs and the purchase of HP and LP units would do the job with one stereo amplifier driving your main channels and a second stereo amplifier driving a DVC subwoofer or pair of subwoofers. Working with one single-voice-coil subwoofer would still leave the problem of summing the left and right channels.

3. Electronic crossovers—There is a variety of electronic crossovers available, many of which will provide summing of the left and right channels for the subwoofer. The crossover would drive one amplifier for the main speakers and a second amplifier for the subwoofer. This is the ideal way to go, especially with a single subwoofer, but generally these units are expensive.

Amplifiers for subwoofers: You can drive the subwoofer by a conventional amplifier if desired. With dual subwoofers or a DVC subwoofer, you can employ both channels of the amplifier for greater power. A special group of amplifiers is built just for subwoofers and generally referred to as "plate amplifiers," which can be built right into the subwoofer cabinet. References 4 and 5 give information on these amplifiers, many of which offer response peaking to extend the subwoofer's lower-frequency limit.

In many cases these amplifiers contain channel summation and all the crossover function that is needed, but read the catalog information carefully. Parts Express offers a full line of subwoofer amplifiers, including a new 250W unit with remote control (#300-793). Call 1-800-338-0531 for information. See sources section for other ways to contact Parts Express.

I would like to thank Karl Keyes of Parts Express for supplying the 12″ woofer used in our subwoofer. Also, I

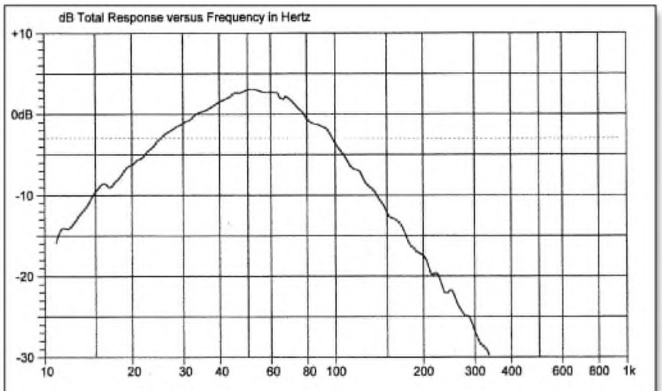


FIGURE 27: Subwoofer on-axis response near rear wall with network.

B-2110-27

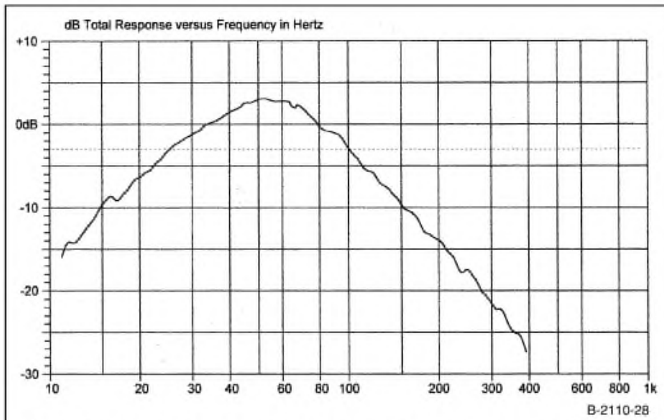


FIGURE 28: Subwoofer on-axis response out in room with network.

would like to thank Tom James of Eminence Speaker for his expert technical input to this section.

AN INEXPENSIVE CROSSOVER APPROACH

To properly apply a subwoofer, you need the right crossovers for both the subwoofer and the main (left and right) speakers. However, if you would just like to try a subwoofer in a stereo system, there is a simple circuit for the IB subwoofer crossover, providing your system meets the following requirements:

1. You will drive the main speakers with full bandwidth; that is, no crossovers are used in the main channels. This will limit playing level to what you are currently using.

2. The low sides of both main speakers return to the same common ground. This excludes any bridged amplifier or other type amplifier that does not return the low side of the speaker to ground.

3. You have an extra stereo amplifier on hand—ideally, one with volume control.

4. Your main amplifier driving the main speakers can tolerate the additional load of a 50Ω resistor into a large capacitor without problem. This "Zobel like" circuit should generally not be a problem.

Figure 24 shows the schematic of the tested circuit. The 50Ω resistors are driven by your main amplifier's left- and right-channel outputs. The circuit provides a second-order low-pass function on these signals and sums them to drive the subwoofer. The network drives both channels of your extra amplifier, and one channel is used to drive each voice coil of the subwoofer, but be careful to keep them in phase.

You may need to vary the values of R3 and R4 to get the signal within the range of the extra amplifier's volume control, keeping the sum near 5kΩ. Our extra amplifier had no volume control, so for listening we made R3 a 2.5kΩ resistor and R4 a 2.5kΩ potentiometer to allow setting the subwoofer level. An amplifier with very low input impedance could modify the response if allowed to pull the network's load much below 5kΩ.

Figure 25 shows the measured electrical response of this network when one and then both channels are driven with the same signal verifying it sums properly. This is the "test output" level of *Fig. 24*. *Figure 26* shows the cone and rear-leakage response magnitudes for the 12″ IB subwoofer, with fiberglass lining, driven by the network and additional amplifier with both network inputs driven. Both the cone and rear-leakage responses are rolled off at high frequency at about 12dB/octave.

The subwoofer on-axis response with the network and without DSL correction is shown in *Fig. 27*. The response is flat within ±3dB from about 25 to 100Hz and falls at about 12dB/octave above this. This is the response with the subwoofer near the rear wall. The on-axis response with 6dB DSL correction (*Fig. 28*) shows the same result!

Thus the network produces about the same on-axis response shape near the wall or out in the room; however, in theory you get about a 6dB increase in on-axis output as you move the subwoofer close to the wall. That is, close to the rear wall you can get several dB more on-axis acoustic output for the same power input than well away from any wall.

This network shows considerable loss, but this occurs ahead of the subwoofer amplifier, so it does not dissipate subwoofer power or limit the subwoofer's maximum output level. Remember with this circuit to experiment with the polarity of the wiring to your main speakers (but not to the network) to see which polarity works best.

The network will clearly work best with small main speakers that do not go much below 100Hz. You can adjust the subwoofer upper-frequency limit by playing with network capacitor C2. You should not use this simple network with other subwoofers without testing to verify response shape.

An electronic crossover using a second-order Butterworth low-pass function cannot really match the equalization shape of this simple passive circuit. The best approximation would be to set the breakpoint frequency at about 50Hz and use the IB subwoofer to about 90Hz.

THE 15″ DRIVER SUBWOOFER

A friend, J. R. Farrand, decided to build an IB subwoofer to compare with his existing VB subwoofer, which uses a non-Dayton 12″ woofer. He used the 15″ version of the Dayton driver (Parts Express #295-190) and included the same internal amplifier (Parts Express #300-792) used in his 12″ VB subwoofer. This amplifier has 5dB peaking at 35Hz built into its response along with a tunable crossover for the subwoofer.

JRF built his 15″ driver IB into an existing enclosure (about 3ft^3 net volume), so no construction details are given. Since the amplifier had peaking, the IB design was for moderate response peaking. For those who would like to try an IB with the 15″ driver, *Table 7* shows the information pertaining to the IB aspects. No testing was performed on the 15″ driver or on the finished 15″ subwoofer.

Initial listening showed good sound with a highly damped response at low volume, but bad "cracking" sounds when the box was pushed. JRF added additional internal bracing to the box to cure this problem. When working with these very powerful high-displacement drivers, you want to build a very strong box and solidly mount the driver. You will also learn to keep your fingers away from these drivers when operating, as we learned trying to see whether the bad sound was due to a loose dust cap!

JRF and GRK compared the 15″ IB subwoofer with the 12″ VB subwoofer in a stereo system. Both can move lots of air and both sound good. However, both listeners agreed that they sound different—it is just difficult to explain exactly what is different.

GRK, a biased listener, thought the IB sub was better on transient material, while the VB subwoofer may have had better bass extension. It is very difficult to discern such differences via listening in this frequency range. Which you liked better was a matter of personal preference, and there was no clear winner. The two main channel speakers were two-way dynamic-driver systems, and it may be that you would obtain different results with other types.

LOW-FREQUENCY OVERLOAD

Most people are aware that a CB offers air stiffness to help protect a woofer from excessive cone excursion with very low-frequency input. Also that the VB does not provide this with only the driver's suspension available to limit excursion below the box's tuned frequency. The IB falls somewhere in between. When you get down near 0Hz, the damping material offers virtually no resistance to cone motion. As frequency increases, the damping material will start to help limit cone motion.

With any type subwoofer enclosure, you want to avoid excessive infrasonic power input and keep alert for excessive cone motion below the system passband during initial testing. We had no problem in this area while playing with the IB subwoofer. However, we used a system with third-order infrasonic filters below about 18Hz.

WHAT WOULD WE DO DIFFERENTLY

Based on this experience, we would do some things differently next time. Increasing the particleboard thickness to 1″ would be first. The back board did not seem stiff enough. The box depth would need to increase to account for the three thicker vertical panels. The internal bracing blocks would also be made of the 1″ material.

We missed on our measurement of the damping material thickness and did not get the ⅛″ compression when we installed the back. This may account for the back board not seeming stiff, because its center was not compressing the damping material. The measurement was based on a single sheet of Owens-Corning material, whose thickness seems to vary a small amount from sheet to sheet. If in doubt, we recommend over-compressing.

We built the corner bracing blocks to offset to one side of the corner. This made them easier to fabricate, because the edge facing the wall required only a single cut at a 45° angle. We found it difficult to position these blocks against the wall to assure full contact during installation. Doing it again, we would double the 45° angle cuts of the blocks and install them directly in the corners.

We recommend feet or spikes on the bottom of the box if you plan to use it on a bare floor. We set the subwoofer in GRK's garage on a board to protect it in case rain came in under the garage door. Music with heavy bass content caused the subwoofer to move quietly about on this board. A small amount of masking tape was sufficient to fix this problem.

SUMMARY

The intent of this exercise to develop a subwoofer using the IB technique was to learn whether the IB approach works with large drivers and thus large dead-air volumes as it had worked with drivers up to 8″ diameter. Happily it does.

The IB's advantages are a very well-damped system and a relatively small box size with big drivers. The IB system can also have a response peak, although the peaking built into our selected driver wasted this advantage. The IB's disadvantage is the measured f_3 may not be as low as with a bigger VB. The measured half-space f_3 does not always tell you how the system will sound.

You can improve the on-axis bass performance of a speaker by building peaking into the frequency response to correct for diffraction spreading loss (DSL). The optimum location of this peak is based on the enclosure dimensions. The CB and VB tend to sound "underdamped" when you design them with a peaked response. The IB does not show this problem. Apparently the various manufacturers have recognized the advantage of DSL compensation, because we found peaking built into the response of our 12″ driver and available on many of the subwoofer amplifiers sold.

Be careful in applying peaking in the box design when using such drivers or amplifiers. Remember that DSL applies only to boxes sitting out away from the rear wall. Amplifiers that have their peaking very low in frequency can be useful in improving the low-frequency extension of your subwoofer.

The high-displacement drivers intended for subwoofer application are heavy and move a lot of air. The box must be solidly constructed and you must mount the driver securely to avoid unwanted noises and assure proper performance. Any vented box should use a huge port—preferably flared—that is rear-mounted to limit and mask air-flow-induced noise.

The IB sounds different from other box types, less noticeable on a subwoofer than a woofer. Whether you consider this "different" sound better or worse is a matter of personal taste. If you are looking for the lowest f_3 value, then don't use an IB. If you are looking for very tight bass, then the IB is a great approach.

The Owens-Corning damping material is a fiberglass-based product, so take the proper safety precautions while working with it. You should also use grille cloth on the back board to keep all the material inside the box. ❖

REFERENCES

1. Koonce, G.R. and Wright, Jr., R.O., "The Infinite Box Concept," *aX* January 2002, p. 8, and *aX* February 2002, p. 38.

4. Perazella, Thomas, "Low Down Power, Part 1," *aX* May 2001, p. 65.

5. Perazella, Thomas, "Low Down Power, Part 2," *aX* June 2001, p. 52.

6. Koonce, G.R., "A Pair of Computer Speaker System Designs, Part 1," *aX* March 2001, p. 8.

SOURCES

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A Sensible McIntosh 225 Refurbish

This author's solution to outfit a sidelined McIntosh 225 amp involves minimal intrusion, a 6L6/5881 swap for 7591s, and a soft-on, soft-off outboard heater power supply. **By T.D. Yeago**

What to do about the Mac 225? Here's a cute little basic amp, the junior model of the McIntosh line of basic amps. Which means it employs the unique Mac output circuit to deliver 25W/channel, first-rate build quality, and that oddly enigmatic and attractive "Mac look."

Another part of belonging to the Mac line is desirability; MC 225s in good shape fetch $600 or so. The downside is that the 225 used the 7591 pentodes for output tubes, and decent 7591s have been getting very scarce since the tube became extinct about ten years ago. (As luck would have it, Sovtek is now issuing a new "7591XYZ" (?) just after publisher Ed Dell gave me the green light for this piece.)

I don't normally concern myself with such audio arcana, except that my little brother was given a 225 some years ago, complete with a set of Sylvania 7591s that looked virtually new. But he's on the cautious side, so he asked me to look at the amp to see whether something readily available could be put to work in the outputs.

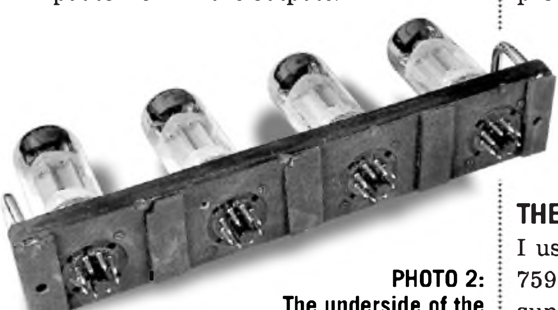

PHOTO 2: The underside of the adapter plate, showing socket hardware salvaged from some old relays. The neoprene cushioning strips prevent expensive cosmetic damage to the Mac's chassis.

But there are a couple of conflicting concerns here. Practically speaking, I know someone with a 225 who has used the original 7591s for more than 30 years. But looking at the value side of the question, it would be nice to use something else.

Which brings up the other nettlesome aspect: alternative tubes are out there (see *GA* 4/96, Gary McClellan's article "Yes, You Can Substitute for the 7591 Tube!"), but that means rewiring the sockets and changing the Mac circuit to adjust the bias. But because people willing to pay big bucks for old Macs insist on strict, bare-bones stock units, adapting means taking a bath on resale value. So I shelved the 225 until a reasonable idea popped into my head.

I suppose it doesn't say much for my creative impetus to allow that all this started about six years ago, but I admit it. I'd hit on something that might serve to address part of the problem, run into trouble, and back on the shelf the amp would go. Well, I've finally got it done and run in, and I wound up with a very good little tube amp, which should last for a long, long time.

THE FINAL PRODUCT

I used the Sovtek 6L6/5881 in lieu of 7591s, and built an outboard heater supply (DC, but no active regulation) and soft-on, soft-off circuit, as you can see in *Photo 1*. The only modifications to the original amp are to 1) unsolder the 6.3V AC windings from the power supply transformer and insulate the leads, and 2) solder some 16 AWG zip cord at the same points I'd unsoldered the stock leads, and route the new heater supply cord out through the keyed hole in the octal hi-Z output socket (*Photo 2*). Instead of rewiring the Mac's sockets for the 5881s, I fashioned an adapter plate.

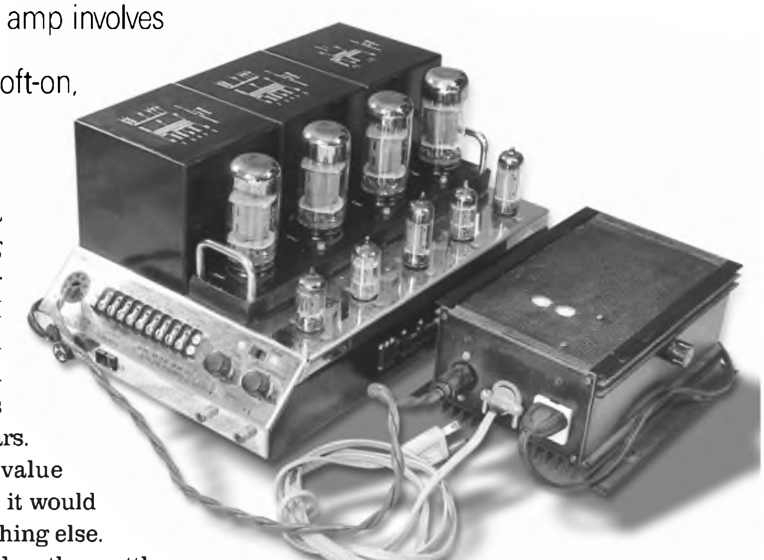

PHOTO 1: The outfitted MC-225 ready for service. Four 5881s and adapter plate are a good fit both electrically and cosmetically. The relays for the soft-on/soft-off feature and DC heater supply for the heaters reside in the black box at right.

USING 5881S FOR 7591S

I refer you to Mr. McClellan's article, referenced previously. Basically, you can get Russky 5881s for less than $10 per. And although they are down on transconductance, have different pinouts, and require re-biasing, they're a sensible substitution. They feature more plate and screen dissipation size-wise and ask for only 0.1A more heater current.

Now, making an adapter plate or adapter sockets isn't rocket science. I used plug hardware from the bases of

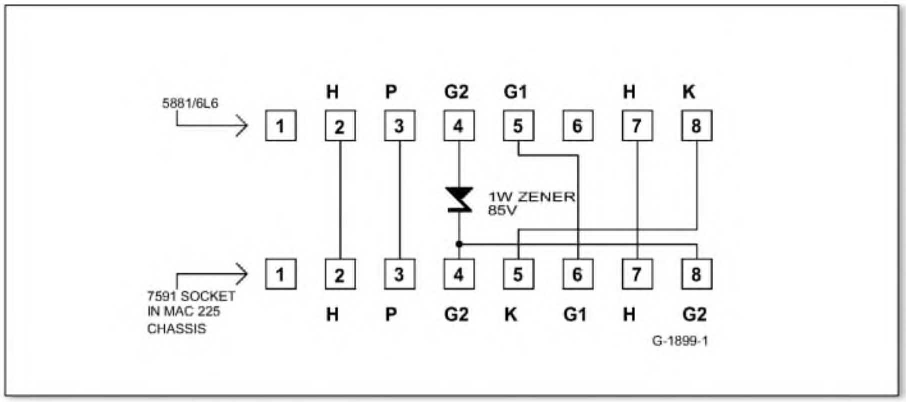

FIGURE 1: Here's the wiring to adapt 5881/6L6s for the stock 7591s. Keep the adapter thickness to a minimum to prevent the 5881s from protruding above the transformers. The zener drops the screen voltage to rebias the new tubes, avoiding the need to change the 225's stock circuitry.

relays and garden-variety sockets. I screwed them into a plate built up of laminating Masonite and Plexiglas. I recommend Teflon-insulated wire, because space is tight. See *Fig. 1* for the pin-to-pin to translate 7591 to 5881.

What's left is the question of rebiasing for the 5881s. In this circuit the 7591s idle at about 14mA with plate voltage of about 330V. Without rebiasing, 5881s would draw (or try to) about 80mA each, which is way, way too much.

I was extremely reluctant to meddle with the Mac bias circuit, so I was flummoxed until I hit upon the expedient of reducing the screen voltage to get the 5881s idling at a sane level. I'd not seen this little trick used elsewhere, but the curves are in the tube manuals. They suggest that for every volt dropped on the screen, the current will drop about 1mA. Why not? So I began substituting zener diodes in the adapter plate to drop the screen voltage, and it worked just fine.

I settled on 85V zeners—1W is plenty. The 5881s allow 35mA of current each at idle, which seems to be a good range for a 6L6, and the plate voltage at this level drops to about 285V, which means only about 10W of plate and screen power dissipated. I call this conservative operation, as this type is rated at 26W (according to RCA) with the added benefit of more class-A operating time.

WHAT, ME WORRY?

Now, why am I so cavalier about installing 6L6/5881s in this venerable little amp? Easy. Just look at the McIntosh output circuit itself. I won't go into detail, but the vast majority of output circuits are basically two plate-followers configured in push-pull. This makes the characteristics of the output tubes an important consideration.

Macs are different. The output circuit used in all the basic amps is most easily thought of as two split-load phase inverters in push-pull. That is, from the B+ the current goes through (let us say) X turns on the primary, then comes out and goes through the output tube, and then back into the transformer to go through another X turns on the primary, then finally to ground. The plate sees the same load as the cathode.

And since the circuit's behavior is controlled by the cathode-follower part of the arrangement, you have an unusual situation. The plate-to-cathode gain has a limit of two; all that cathode feedback means low output impedance, high tolerance for tube variations, and low distortion.

It also provides a handy way to measure current at idle. Just measure the DC resistance of the "bottom" sections of each output transformer; that is, from cathode to ground. On the 225 it's about 41Ω. To figure idle current, measure the standing DC voltage across this 41Ω (comes to about 1.45V DC in my case). Apply Ohm's law. Getting to one of these pins can be tricky, but I found an easy way (*Fig. 2*).

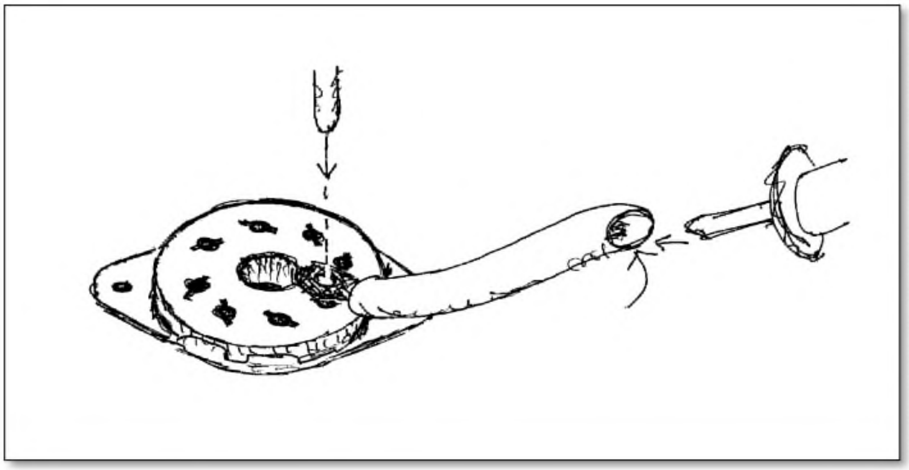

FIGURE 2: **With the tubes plugged in, getting a probe in to measure the voltage and figure bias can be tricky. Take a length of braided shielding a couple inches long from some coax. Tin one end, snip off the excess strays, and poke a hole in it. Next take some insulation from the same coax and slide the braid in. You now have a safety lead to measure the voltage at the cathode. Stick pin 8 through the hole in the braid and plug the tube into the socket. To measure pin 8 voltage, just poke the probe inside to touch the braid.**

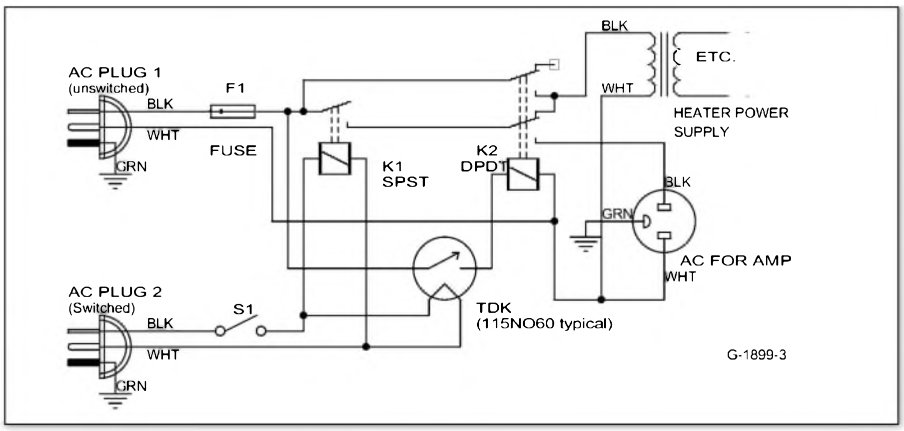

FIGURE 3: **A straightforward soft-on/soft-off circuit using an outboard heater supply and a second line cord to allow remote turn-on of the amp.**

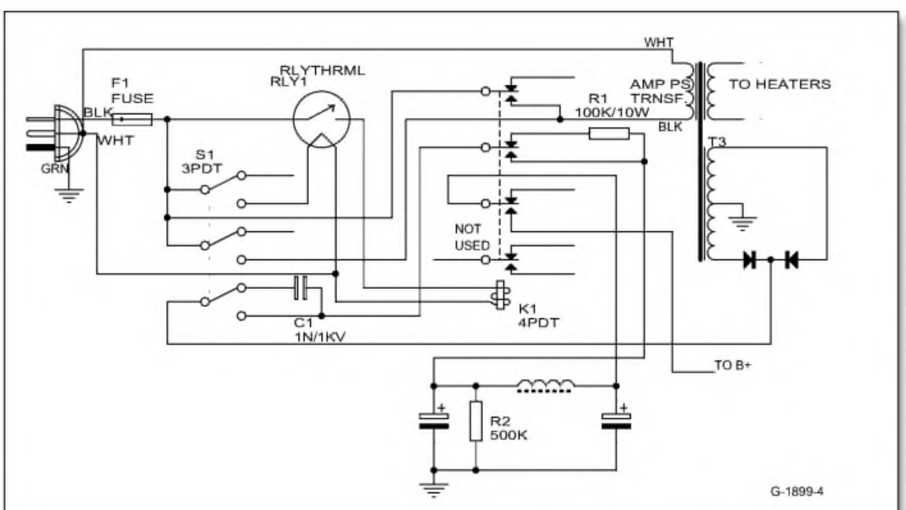

FIGURE 4: **Another incarnation of the soft-on/soft-off. This one supposes everything will be built inside the amp chassis and use the amp's heater supply instead of an outboard supply. Because the B+ windings are energized as well as the heater windings, B+ switching is called for. R1 slowly brings the pi-type power supply filter up to voltage; R2 is a typical bleeder.**

SOFT-ON/OFF

That funny black chassis contains a conventional DC power supply for all the heaters (not just the signal tubes) and a couple relays that make for soft turn-on and (unusual, as far as I can tell) soft turn-off of the amp. Everybody knows about soft-on's virtues. If there's high voltage on the plate before the cathode is hot and spewing electrons, a bad thing called "cathode-stripping" happens.

I don't see anything to prevent cathode-stripping in the turn-off cycle, if the cathodes cool down before the B+ is drained down to harmless levels. I know, I know: old units such as the 225 don't have huge filter caps, so the output tubes do a pretty good job of draining the caps before they cool. But I was building a soft-on circuit, so I included a soft-off feature, too.

Look at *Fig. 3*, the schematic for a basic circuit. I use thermal relays because they provide a time delay before they close, but because they take a while to cool off, there's a time delay before they open, too.

Most of you will use little solid-state timer circuits. But a few words about the old thermal time-delay relays (TDKs) might be of interest. They inhabit vacuum tube packages, both miniature 9-pin and octal, and are simplicity itself. A heater warms a bimetal strip which bends and closes (or opens) a pair of contacts. Amperites seem most common and designations are transparent.

A 115NO60 means the heater is intended for 115V, the contacts are normally open (NO), and it should take 60 seconds for the strip to bend sufficiently to close the contacts. Because we're talking about a simple heater here, manipulation is easy. If a relay is too fast for your taste—30 seconds, say—just wire a diode in series with the heater (or drop the voltage another way), and you'll slow the heating process and increase the elapsed time.

BASIC CIRCUIT

But enough about the details; go back to the first schematic (*Fig. 3*). If you're using a heater supply independent of the amp's power transformer, things are simplified considerably. Using relays that run on 115V AC for simplicity's sake, here's what happens:

The power switch (S1) closes, warming the thermal relay and closing K1, which sends AC through K2 to turn on the heater power supply. The heaters in the amp warm for, let us say, 60 seconds, the time it takes for the TDK to close.

TDK closes, tripping K2, a 2P, 2T unit. The bottom set of contacts switches AC away from the heater power supply and to the AC plug, where the amp is plugged in, energizing the amp's B+ power supply. Simultaneously, K2's other set of contacts closes, bypassing K1 and maintaining AC to the heater power supply.

On turn-off this is the sequence: S1 opens and immediately K1 opens, but K2 is still closed, so AC to the amp is switched off and B+ is drained down by the still-hot cathodes. K2 is still closed because TDK, which is a thermal relay, is still closed, and even though AC to its heater is switched off by S1, it takes 30 seconds or so for the unit to cool enough to open.

When TDK finally opens, K2 opens and the heater power supply shuts down. I use two line cords and AC plugs so I can plug P1, the fused line, into the wall or an unswitched outlet. P2, the switched line, supplies only the TDK and K2, and so draws very little current. You can turn the amp on with S1 or close it and plug P2 into the switched outlet on the back of your preamp. The fuse in the unswitched line should be the same as in the amp itself, by the way.

ANOTHER VERSION

Figure 4 assumes you need to use the amp's power transformer to supply power for the heaters. Again, I assume a TDK and K1 that use 115V AC. All the switching and bypassing require a 3P, DT on-off switch (S1) and a 3P, DT relay (K1). And because you need AC to the primary to energize the heaters, you must switch the B+.

The top two sections of S1 and top section of K1 deal with the AC. The bottom section of S1 and bottom two sections of K1 handle the B+ end of things. The TDK and K1 need their own pole in S1, or else the top section of K1 would bypass S1, and throwing S1 wouldn't shut down the TDK heater and K1 on turn-off.

Concerning handling B+, things become a little touchy. There are three sets of contacts in series, ultimately. Bypassing with "spark-killer" caps or RCs is important.

At turn-on, S1 closes, energizing the primary and all secondaries. But K1 isolates the pi-type power supply filter. The filter slowly charges via R1 (100k, 10W, typical), but is isolated from the output stage by the bottom section of K1 to let the tubes warm up in peace. When the TDK closes after 60 seconds or so, power is applied and everything operates as normal.

On turn-off S1 is opened, shutting off AC to the TDK's heater and disconnecting the pi filter's input from the power supply diodes. K1 stays closed as the TDK cools. The power supply caps discharge through the various tubes. When TDK finally opens, AC to the primary is finally shut off and the cathodes cool. If you go this route, it is important to have an adequate rating for S1 and to keep the hot and neutral sides of the AC line straight.

COMPONENTS

Figure 5 shows what I actually built into that black box. I used salvaged parts as a generality, and used separate 6.3V AC relays because I had them and they fit inside the cramped enclosure. The TDK is a 90-second unit and takes about 30 seconds to cool. The rectifiers are half of a 20A block, and thanks to all that obsolete digital stuff, big, low-voltage caps are pretty common. I also wired in some LEDs as pilot lights over the heater and AC sockets to show what's going on.

Running the output tubes' heaters on DC is fun for one-upmanship, but there's a downside, as usual. The secondary is only 6.3V AC, and considering that the heater drain is almost 6A at 6.3V, drops in the diodes and winding R losses limit this unit to 5.7V for the heaters instead of the 6 to 6.3V specified.

But this is not a problem. The output

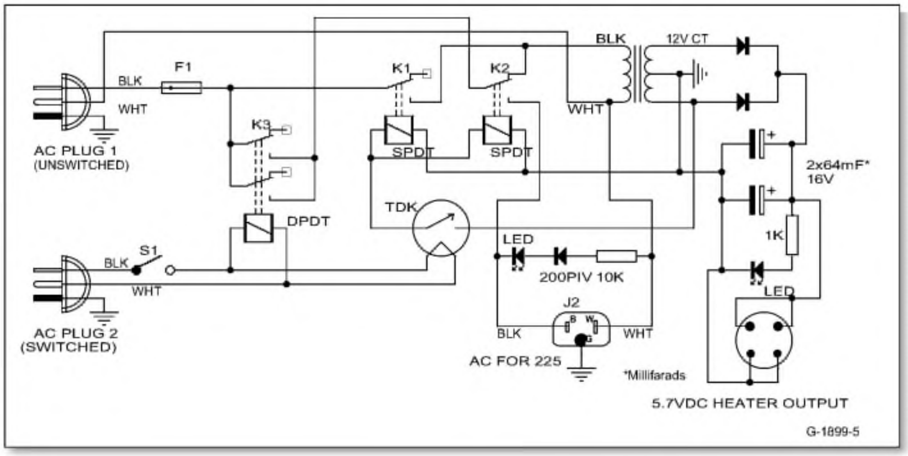

FIGURE 5: This is what I actually managed to stuff into that black box. There's no particular reason I used these parts, other than they were what I had on hand. Under load the DC heater supply sags to 5.7V, but the amp seems oblivious to the shortfall.

tubes flow the 35mA each at idle and the operation of the signal tubes does not seem to be impaired. That is, the amp works absolutely fine. And my memory vaguely recalls some literature about heater voltage and tube life, indicating that voltage in this range leads to longer life—on signal tubes anyway. There will, I expect, be comments from the learned readership about this.

Aside from making the switching easier, using a separate power supply for the heaters has another benefit: it relieves the stock power supply transformer of supplying 35W to its heater winding. Which means more flux in the core available to the B+ winding, which means better B+ regulation.

USER COMMENTS

I wired some 20Ω dummy loads across the 16Ω taps and brought the unit up to operating voltage. Although the Russky 5881s aren't in any way matched or selected, they're all within 1mA of 35mA at idle.

I have some relatively high impedance speakers (14Ω over the power spectrum, dropping to 10Ω in the treble), so I went ahead and hooked them up to the 8Ω taps. An easy load. There was slight hum in one channel, cured by installing another 12AU7 which wasn't, evidently, microphonic. And the chassis of the 225 itself hummed a little. Inverting the 225's AC plug fixed that.

I did no testing other than the voltage measurements mentioned previously and about a month's worth of listening. I like this amp very, very much. The speakers I used (much-modified ADVENT two-ways, mid-80s vintage) are of moderate sensitivity, but on lots of varied program material I never felt cramped on the headroom front. I won't repeat all the usual (and suspect) adjectives here. I'll just say that I have no criticisms, and if limited to one adjective, I'd say this little amp is charming.

The amp does hum through the speakers for about 1 second when the B+ is switched on, and there's a mild thump when the B+ is shut off. And the Sovtek 5881s show almost no cathode glow, but there is a strange blue glow within. Also, for a tube amp, this unit generates surprisingly little heat.

My brother has had the amp about a month, now, but hasn't hooked it up as yet, so he has nothing to report. Too bad. I managed to obtain some old Tung-Sol 5881s for him to try out. Unless some of the caps go south, I see no reason why this amp shouldn't see use past 2020. ❖

Moving-Coil Phono Cartridge Transformers

If you're a fan of analog sound, this easy project can save you a few hundred dollars. **By Pete Millett**

Face it, folks—vinyl is here to stay. Whether you'd just like to keep listening to your old record collection, or you believe that no digital reproduction can match the fidelity of a good analog recording, for many of us LPs live on.

I've just started getting back into vinyl after nearly 20 years of listening to CDs almost exclusively. Records, especially used, are still a pretty good bargain, if you can find a good source. And after doing some direct A/B comparisons between CD and vinyl using headphones, I really do believe that a good analog recording can't be beat.

In my quest for great analog sound at a low price, I ran into an obstacle—I got a great deal on a new moving-coil phono cartridge, but my tube preamp wasn't designed to accept its low-level output. What to do?

A number of products on the market adapt a "normal" phono input to the low output of a moving-coil cartridge. I'm sure that they are all great products—but I had a problem with their price tags. After all, I figured, all these units contain are a pair of tiny step-up audio transformers; why do they sell for between $450 and $2,000?

To make a long story short, though they are not inexpensive, you can buy excellent transformers for a lot less money than a finished commercial product. With an hour or so of effort, a power drill, and a soldering iron, you can make your own moving-coil step-up transformer setup that will rival their performance for half the cost.

MOVING-COIL VERSUS MOVING-MAGNET CARTRIDGES

I'm no cartridge expert, but here's a simplified review of the different types of phono cartridges.

Moving-magnet phono cartridges are the "standard" type of cartridges that have been in use since the introduction of stereo LPs. They work by mechanically coupling a tiny magnet to the stylus (needle) that rests in the groove of the record. Vibrations picked up by the stylus vibrate the magnet, which induces a voltage into coils of wire. This voltage is then amplified by a phono preamp.

Typical moving-magnet cartridges have a voltage output of about 3-4mV. For historical reasons, the standard load impedance (resistance of the preamp input) is 47,000Ω, and all moving-magnet cartridges are designed to drive this load.

Moving-coil cartridges reverse the positions of the magnet and coil—the coil moves and the magnet is stationary. Most moving-coil cartridges have much lower output voltages—between 100μV and 500μV—as well as a much lower impedance, being designed to drive loads between 20 and 500Ω. (Though not common, there are also "high-output" moving-coil cartridges that have a high enough output to directly drive a standard phono input without a step-up transformer).

To drive a phono preamp with a "normal" moving-magnet input with a low-output moving-coil cartridge, you need to boost the voltage by a factor of between 10 and 50. Since the impedance of a moving-coil cartridge is very low, this can be done with a simple step-up transformer, without an active amplifi-

PHOTO 1: A Sowter moving-coil transformer (courtesy Sowter Company).

PHOTO 2: A Jensen moving-coil transformer (courtesy Jensen Transformers).

PHOTO 3: A moving-coil transformer shown outside of its metal case (courtesy Sowter Company).

er. Though audio transformers are not without their problems, at this very low signal level, their noise and distortion characteristics are hard to beat with an amplifier stage.

Photos 1 and *2* show two typical moving-coil transformers. They both are enclosed inside a metal shielding can, which is about 1¼″ in diameter and 1¼″ to 1½″ tall. Inside the can is a small transformer, similar to that shown in *Photo 3*.

HOW THE STEP-UP TRANSFORMER WORKS

Figuring out which transformer type is right for your cartridge can be a bit confusing. So, let's start with a bit of simple transformer theory.

A step-up transformer works by having two windings with different numbers of turns. Grossly oversimplified, if you apply a voltage of 1V to a winding that has one turn, it will induce a voltage of 10V in a winding that has ten turns. In other words, the voltage ratio is the same as the turns ratio. A transformer with a turns ratio of 1:10 will provide an output voltage ten times that of the input voltage.

The impedance of an audio transformer is a little trickier. Transformers don't have specific impedances themselves; they are designed to accommodate a specific impedance, at a certain level and frequency. That's way beyond the scope of this article, so I'll confine the discussion to the subject at hand, which is more concerned with the impedance ratio between primary and secondary.

The impedance ratio of a transformer is equal to the turns ratio squared. So, in our transformer with a 1:10 turns ratio, the impedance ratio will be 1:100. So if you load the secondary with 47,000Ω, the primary will present a 470Ω load to the cartridge (47,000/100 = 470).

It is difficult, and expensive, to manufacture a good moving-coil transformer. Typically, very thin laminations of special materials are used in the core to get very low distortion of the small signals coming from the cartridge. *Photo 4* shows the laminations used in the Sowter 6495, which are only 0.004″ thick!

PHOTO 4: Thin laminations of special alloys are usually used to make moving-coil transformers (courtesy Sowter Company).

PHOTO 5: The finished unit with the cover on.

SELECTING THE RIGHT TRANSFORMER

To select the correct transformer, you need to know a couple of things about the cartridge that you intend to use: First, the output voltage, and second, either the cartridge's impedance, or (preferably) the recommended load impedance. You always want to provide a load of greater than the minimum recommended load; if the minimum load is not specified, figure it should be a minimum of ten times the cartridge's impedance.

The goal is to take the output of the cartridge and step it up to somewhere between 3 and 5mV, and provide a load impedance to the cartridge that is greater than the recommended load. To illustrate, let's use an example:

You have a cartridge that has an output of 130µV, with a minimum recommended load impedance of 20Ω. What kind of transformer do you need?

First, calculate the voltage ratio (which, you remember, is also the turns ratio). If you want a 4mV output, you would need a ratio of 130µV/4mV = 1:30. A 1:30 turns ratio gives a 1:900 impedance ratio, so the cartridge would see a load of 47,000/900 = 52Ω. This is greater than the manufacturer's recommended minimum—so everything looks great!

But, of course, a 1:30 ratio transformer doesn't exist. Knowing what you'd ideally like, now you must find the best fit among the available off-the-shelf moving-coil transformers. The most commonly available transformers are wound with 1:10 and 1:37 turns ratios, although there are others. Given our example, which would be best?

With the 1:10 transformer, you will get only 1.3mV of output (10 × 130µV = 1.3mV) and a reflected load impedance of 470Ω (47,000/100 = 470). The 1.3mV value is a bit on the low side—you really want 4–5mV of signal to drive the preamp—so the 1:10 transformer might not be a very good choice.

With a 1:37 ratio, you would get 4.81mV and a reflected load of 34Ω. This 1:37 ratio transformer would work fine, since the reflected load is still above the manufacturer's recommendation, and 4.81mV is about the right level to drive a preamp.

If all that seems too complicated, just follow these guidelines suggested by Jensen Transformers:

For a 1:37 ratio transformer:

A) Your cartridge should have an output voltage of at least 130µV.

B) Your cartridge should have a rated impedance of 5Ω or less and/or a rec-

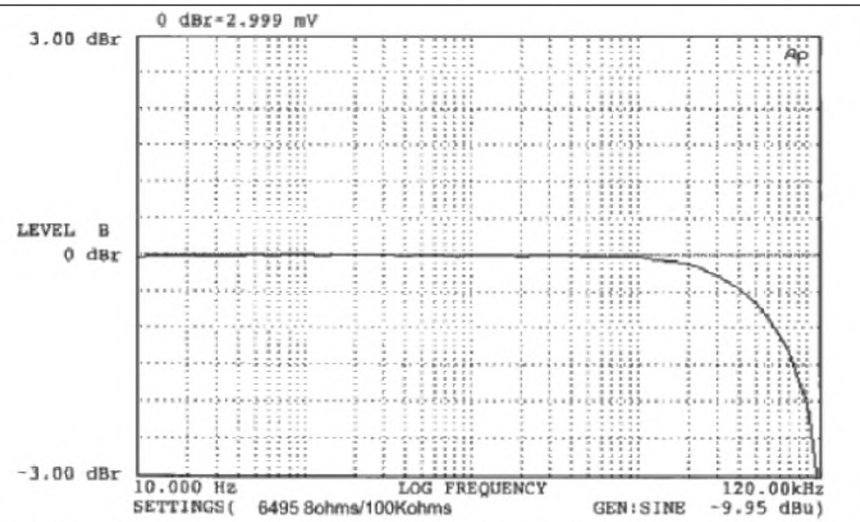

FIGURE 1: Frequency response graph of the Sowter 6495 (courtesy Sowter Co.). A-2035-1

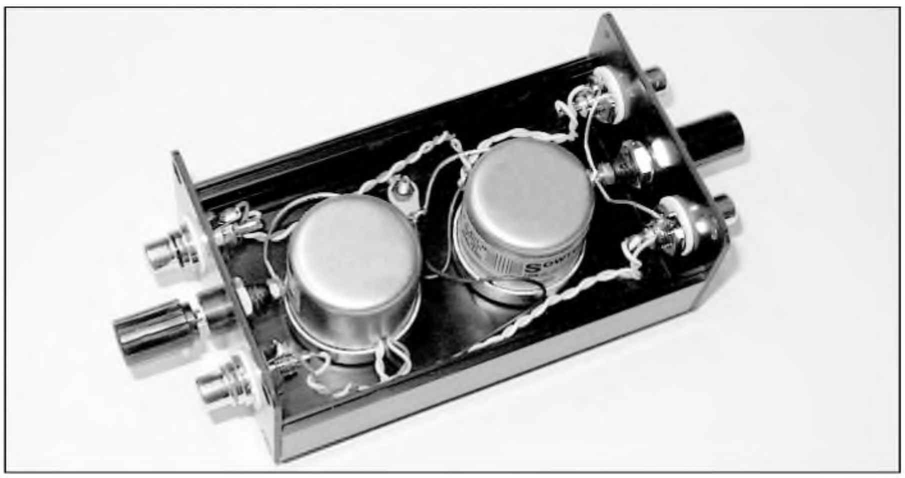

PHOTO 6: The transformers mounted into an aluminum enclosure.

PHOTO 7: Close-up of the input jacks. Note the extra ground wire run to the RCA jacks (see text).

ommended load impedance of 32Ω or less.

For a 1:10 ratio transformer:
A) Your cartridge should have an output voltage of at least 500μV.
B) Your cartridge should have a rated impedance of 20 to 50Ω and/or a recommended load impedance of 430Ω or less.

Note that there are some cartridges that just don't work right within these limitations, but most will. And it's important to remember that you will need two transformers, one for the right channel, and one for the left.

TRANSFORMER SUPPLIERS AND PRICES

I found several companies that sell unpackaged moving-coil transformers to the hobbyist. *Table 1* shows a few of the possibilities. Note that this is not an all-encompassing list—there are more manufacturers, and certainly the ones represented here have many more products than what I've shown—but this is a good place to start.

Costs shown were those published on the supplier's websites (in the UK, in the case of Audio Note and Sowter) and have been converted to US dollars. Actual costs in the US might be different, so contact the manufacturers directly to get an accurate quote.

For my project, I bought a pair of transformers from Sowter Co. in England, partly because they were the least expensive, and partly because I've had good luck with their products in the past. The transformer I chose was their model 6495 (*Photo 1*).

Even though (by comparison) it's an inexpensive transformer, the 6495's specs are quite good. Its frequency response (*Fig. 1*) is within a couple tenths of a dB from 10Hz to 100kHz, and has less than 0.5% distortion at 50Hz. Note that low-frequency fidelity is one area in which paying more money will get you better specs—the Jensen 1:10 transformer (*Photo 2*), a bit more expensive than the Sowter, is specified at 0.1% THD at 20Hz, and has low-frequency response that extends down to 2Hz.

MECHANICAL ASSEMBLY

So, now you've figured out which transformer to use. All you need to do is mount it in a box, wire it up to some phono jacks, and you're done!

For the unit that I built, I purchased an extruded aluminum enclosure made by LMB-Heeger (model number EAS-100) at a local electronics store, along with two pairs of nice gold-plated RCA jacks with insulating washers and two binding posts for ground connections. The LMB extruded box is just the right size for mounting two transformers and looks quite professional (*Photo 5*).

Assembly work is straightforward. First, you need to drill holes in the enclosure to mount the transformers (the transformers I used required two screws in the bottom of the transformer), and holes to mount the RCA jacks, grounding binding posts, and a hole for a screw to provide a ground connection to the case (*Photos 6* and *7*). *Photo 8* shows a close-up of the jacks and ground post on one end of the enclosure; the other end is identical. The exact placement of the parts inside the enclosure is not critical.

WIRING

Wiring the unit is simple as well. Refer to the schematic diagram (*Fig. 2*), which shows the unit that I built using the Sowter transformers (other transformers will have different wire colors, but the same general idea applies). The primary winding has two wires—one that connects to the center conductor of the RCA jack, the other to the outer shell—on the input side (connected to the turntable).

The secondary is wired the same way to the RCA jack on the output side (to the preamp). Both channels are wired in an identical fashion. Note that the outer shells of the RCA jacks are insulated from the case with nylon shoulder washers (these came included with the jacks). It's a good idea to twist the two wires from each winding

TABLE 1
SOME AVAILABLE MOVING-COIL TRANSFORMERS
(N/A = INFO NOT AVAILABLE)

TRANSFORMER	COST (APPROX.)	RATIO*	FREQUENCY RESPONSE	THD	NOTES
Sowter 6495	$57.56	1:10	5Hz–100kHz	0.5% @ 50Hz	
Sowter 7136	$57.56	1:20	10Hz–100kHz	0.5% @ 50Hz	
Sowter 6778	$57.56	1:37	10Hz–20kHz	n/a	
Jensen JT-34K-DPC	$78.51	1:37	4Hz–180kHz	0.12% @ 20Hz	PCB mount
Jensen JT-34K-DX	$104.76	1:37	4Hz–180kHz	0.12% @ 20Hz	Double magnetic shield
Jensen JT-44K- DX	$104.76	1:10	2Hz–180kHz	0.10% @ 20Hz	Double magnetic shield
Audio Note TRANS-465	$73.15	3/12Ω	n/a	n/a	
Audio Note TRANS-470	$73.15	15/60Ω	n/a	n/a	
Audio Note TRANS-460	$111.07	3/12Ω	n/a	n/a	
Audio Note TRANS-455	$111.07	15/60Ω	n/a	n/a	
Audio Note TRANS-481	$365	1/4/9Ω	n/a	n/a	"Ultra-high quality"
Audio Note TRANS-492	$1,086	3/12Ω	n/a	n/a	Silver wire

*Audio Note does not seem to specify their winding ratios directly; instead, they rate the transformer with a recommended cartridge impedance.

tightly together, to help reject any hum that might become coupled into the wires.

Most transformers will have an extra wire, connected to an electrostatic shield inside the transformer. This wire is connected to ground, as are the ground binding posts at both the input and output. These connections are all made at one single point, which is a solder lug that is screwed to the metal case. In any case, refer to the datasheet for the transformer that you are using, and follow the manufacturer's instructions.

If your turntable does not have a separate ground wire, but is instead grounded through the RCA plugs, you will probably need to connect a wire from your ground lug to the outer shells of both the left and right RCA input connectors to reduce hum pickup from the turntable and tonearm. This wiring is visible in *Photo 7*, but is not shown on the schematic diagram.

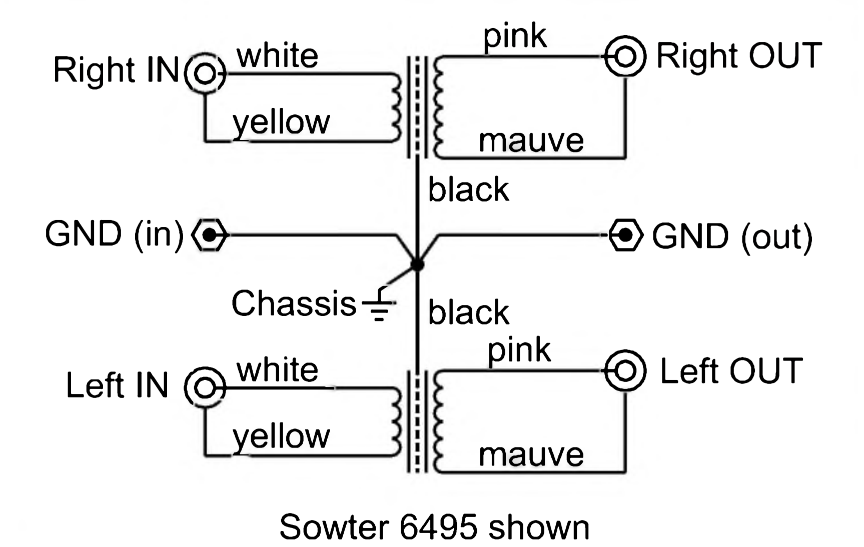

FIGURE 2: Schematic diagram of the complete unit.

PHOTO 8: The RCA jacks and ground binding posts.

THE FINISHED PROJECT

Photo 5 shows what the entire unit looks like when assembled. To me, it looks almost as nice as some commercial products, and you have the satisfaction of having built it yourself. And with the money you save, you can buy more records!

To connect your transformer into your system, plug the cables from your turntable directly into the input RCA connectors (connected to the primary windings of the transformers) and connect the ground wire (if equipped) to one of the binding posts. On the output jacks, make sure you use good, shielded, low-capacitance interconnect cables between your transformer setup and preamp—the shorter the better. Some audiophile interconnects are not shielded, and just consist of two wires inside a jacket. Do not use this type of cable for a phono input, because it will pick up lots of noise. Connect a ground wire between one of the binding posts and the preamp. You're ready to listen!

Since I haven't had the opportunity to listen to any other moving-coil step-up transformers, I can't really give an objective opinion about how these transformers sound as compared to others that are available. I can say that my system, using an Audio-Technica OC-9 cartridge on an NAD 533 table, is quiet and produces wonderfully clear and detailed sound from a good record. ❖

SUPPLIER CONTACTS

Audio Note North America
2395 Cawthra Rd., Unit 15
Mississauga, Ontario, L5A 2W8
(905) 306-1677
FAX (905) 306-9689
www.audionote.on.ca
www.audionote.co.uk

Jensen Transformers
7135 Hayvenhurst Ave.
Van Nuys, CA 91406
(818) 374-5857
FAX (818) 374-5856
www.jensen-transformers.com

LMB-Heeger
6446 Flotilla
Commerce, CA 90040
(323)-728-5108 or (323)-728-4729
FAX (323)-728-4740
www.lmbheeger.com

Sowter Transformers
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FAX +44 1473 236188
www.sowter.co.uk

Product Review

GW Labs DSP

Reviewed by Gary Galo

GW Labs DSP (Digital Signal Processor). Centasound International, Inc., 484 23rd Ave., PO Box 210337, San Francisco, CA 94121, 415-668-9003, FAX 415-668-9638, www.gw-labs.com. DSP–$399.

PHOTO 1: Front view of the GW Labs DSP. Two switches select line or opto inputs, and an output sampling frequency of 44.1 or 96kHz.

GW Labs is probably unknown to most *audioXpress* readers. This small California-based company specializes in vacuum tube audio equipment. Their product line includes the MM-1 two-channel multimedia power amplifiers, the Cyclops two-channel integrated amplifier, and the model 270 two-channel power amplifier. A vacuum tube D/A converter is forthcoming.

The DSP, for Digital Signal Processor, is their most recent product and is being offered at a special introductory price of $399 (*Photo 1*). The DSP is a digital-to-digital upsampler that will accept input sampling frequencies from 32kHz to as high as 96kHz, and output a user-selectable frequency of either 44.1kHz or 96kHz.

DSP FEATURES

Several design choices remind me of the Monarchy digital audio products. The DSP accepts S/PDIF, AES/EBU, and Toslink optical inputs (*Photo 2*). Like the Monarchy DIPs and D/A converters, the S/PDIF and AES/EBU inputs are an "either/or" situation. You can have only one of these inputs connected; otherwise, the DSP won't function. The front panel switch selects either "line" or "opto." The DSP has both S/PDIF and AES/EBU outputs.

GW Labs uses the same high-quality pulse transformers for input and output coupling as Monarchy, and they cite the advantage of component isolation using transformer coupling. Like the Monarchy DIP 96/24 (see review, Sept. 2001 *aX*, p. 66), the DSP boosts the level of the S/PDIF or AES/EBU signal—by a factor of ten according to GW Labs. Actually, the S/PDIF output is held at a constant 3V peak-to-peak regardless of the input level, an increase by a factor of six for nominal S/PDIF inputs of 0.5V peak-to-peak. Finally, they have employed Monarchy's "pull-up" resistor biasing arrangement for the Toslink input receiver (see my article "Optimizing the S/PDIF" in *AE* 4/98).

At the heart of the DSP is the Cirrus Logic CS8420 Sample Rate Converter chip (*Photo 3*). The excellent CS8420 has become a de facto industry standard in the 96kHz/24-bit world, and is used in Parts Connexion's D2D-1 Sample Rate Converter, as well as Perpetual Technologies' P-1A Digital Correction Engine and P-3A D/A converter (*aX* April 2002, p. 52). One of the handy features of the CS8420 is the inclusion of an AES/EBU input receiver and transmitter, which can substantially simplify design. GW Labs has taken advantage of this, using the CS8420 as a stand-alone device. In its default configuration, the CS8420 adds output dither, and automatically adjusts the dither to match the output word length. GW Labs uses the CS8420 in this default setting, as recommended by Cirrus Logic.

AC power connects to the DSP via a standard IEC power connector, but without the usual built-in power-line filter. Instead, GW Labs has designed their own common-mode line filter and placed it on the main PC board. The power transformer is a dual-bobbin type, for additional rejection of power-line noise. Two secondary windings feed a pair of Shindengen LN2SB low-noise rectifier bridges. Each rectifier output is L/C filtered with a series inductor and two 10,000µF capacitors in a "pi" configuration.

Four 7805-type regulators provide regulation of the +5V lines. Seven ferrite beads are used to reduce high-frequency noise on the regulated lines, and to provide a certain degree of high-frequency isolation between active components. One of the +5V regulators appears dedicated to the CS8420's analog supply pin, which is the critical PLL supply.

Both the AES/EBU XLR connectors and S/PDIF RCA jacks have gold-plated pins, and the RCA connectors are Teflon®-insulated. I am not familiar with the type of RCA connectors used in the DSP, but they are PC-mounted and extremely well-made.

ASYNCHRONOUS OPERATION

Upsamplers offer three performance advantages: suppression of clock jitter, sample rate conversion to 96kHz, and

PHOTO 2: The rear panel features Toslink Optical plus AES/EBU and S/PDIF inputs, along with AES/EBU and S/PDIF outputs.

word length conversion to 24-bit. The sample rate and word length conversion will allow the digital filter and D/A converter chips in a 96kHz/24-bit DAC to perform to their fullest potential when playing conventional compact discs. The manufacturer also notes that the DSP generates its own precise clock, and does not use the recovered clock from the CD or DVD transport. This advantage is inherent in asynchronous sampling rate converter chips, but won't necessarily apply to any product using the CS8420.

Cirrus Logic offers the option of either using the recovered clock or deriving a new output clock from a stable crystal oscillator. GW Labs has wisely chosen the second option.

The DSP is extremely simple to install and operate. Just insert it between your CD or DVD transport and an outboard D/A converter with some high-quality cables (I use D.H. Labs' Silver Sonic D-75), and you're in business. There is no power switch—the DSP is designed to be left on at all times, or connected with a power-line filter offering sequential switching of audio components (such as those made by Adcom and Parasound).

Operation involves only two switches. One selects either the line (AES/EBU or S/PDIF) or opto inputs. The other switch selects the output sampling frequency—either 44.1kHz or 96kHz. A digital lock LED is included on the front panel, and a pair of LEDs indicate the selected digital input. On CDs recorded with high-frequency preemphasis, the DSP passes the deemphasis flag on to the D/A converter for proper decoding.

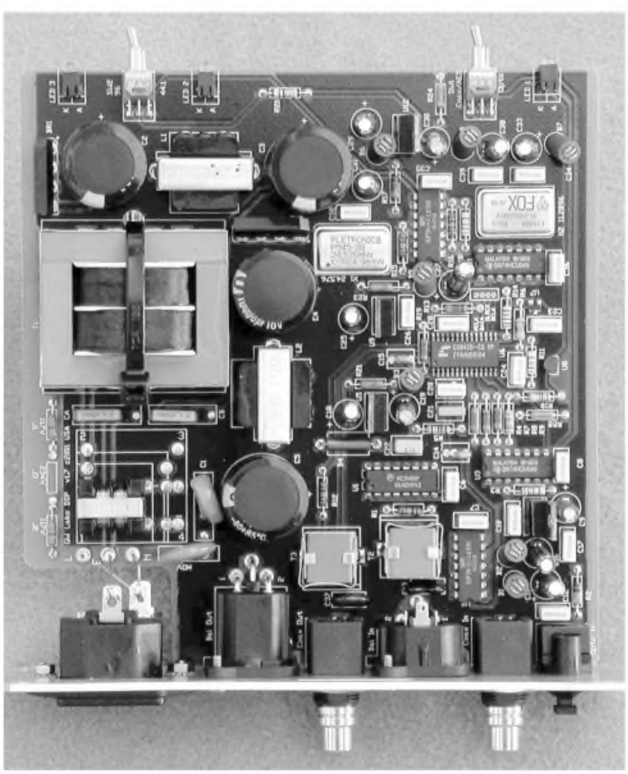

PHOTO 3: All DSP circuitry is contained on one PC board. The heart of the DSP is Cirrus Logic's excellent CS8420 Sample Rate Converter chip in a 28-pin SOIC package, shown center right. A common-mode line filter and dual-bobbin power transformer, shown on the left, help attenuate power-line noise.

SONIC IMPROVEMENTS

In my review of the Monarchy's DIP 96/24, I noted that it was the first jitter-suppression device that actually improved the performance of my Parts Connexion DAC 3.0. For only $150 more, the DSP takes these improvements to a new level. Soundstage width and depth are improved, with the image stretching closer to the edges of my loudspeakers.

On Dorati's Mercury recording of Schoenberg's *Five Fieces for Orchestra*, the delineation of the various sections of the orchestra is enhanced by the precise positioning rendered by the DSP. Clarity and definition are excellent with the DSP. Charles Munch's reference-quality recording of *The Sorcerer's Apprentice* was reproduced with excellent clarity and definition, and the sense of air and space in the treble region was most impressive. The delicate, clean reproduction of the orchestra bells was particularly noteworthy.

The DSP offers a warm, musical sonic presentation. Massed low strings, such as those at the beginning of the fourth movement of the Reiner *Scheherazade*, are rich and gutsy, palpably more realistic than before. The treble region with the DSP inserted takes on added smoothness. The DSP is quite euphonic in the treble, mellowing the hardness and glare on some of my not-so-well-recorded CDs.

Decca's 1984 digital transfer of the Solti *Ring*, even as refurbished for the 1997 re-release, is showing signs of age. In particular, there is a bit of digital hardness in the treble region that is not found in the best of today's reissues of material from the analog era. Played through the DSP, my reference passages in this recording sounded smoother and more musical than before.

The DSP makes a noteworthy improvement in the bass. In Ansermet's recording of Ravel's *Alborada del Gracioso*, and the "Gnomus" section of Reiner's *Pictures*, the bass drums are weightier, but at the same time tighter and better defined.

Classic Records' superb 96kHz/24-bit DVD transfer of Rachmaninoff's *Symphonic Dances* exhibits reference-quality bass, as captured on the original tapes by engineer David Hancock. The

DSP acquitted itself admirably on this recording, offering a more powerful, and at the same time, more articulate, low end. The clarity and definition in the midrange and upper frequencies on this recording were very good, but I was left with a sense that the re-writing of the data put a slight veil over this recording (more on this later).

I should add that the DSP's excellent jitter suppression makes it a good choice for use with older-generation D/A converters. With the output sampling rate set at 44.1kHz, the DSP becomes fully compatible with earlier DACs that don't operate with 96kHz/24-bit datastreams. Furthermore, the DSP also makes DVD transports compatible with older DACs.

If you own a DVD player with a 96kHz/24-bit digital output (such as one of the Pioneer models), the DSP will allow you to play 96kHz DVDs through an older D/A converter. You'll get better sonic performance—letting the DSP alone do the downsampling to 44.1kHz—than you will if the DVD player's digital output is set to a lower sampling frequency (the lower choice is invariably 48kHz). This is due to the fact that only one sample rate conversion takes place. If the DVD player downsamples the digital output to 48kHz, the DSP will still make yet another conversion to 44.1kHz. In this case, two sample rate conversions are not better than one!

SUMMARY

Strictly as a jitter suppressor and upsampler, the DSP is hands-down superior to Perpetual Technologies' P-1A, and at a fraction of the cost. I have only two minor quibbles with the DSP. First, I'd like to see 75Ω BNC connectors offered—perhaps as an option—on the S/PDIF input and output. On a product this good, the difference is likely to be audible.

Second, the CS8420 supports internal bypassing of the sample rate converter, while retaining the jitter suppression this chip offers. Adding a "bypass" mode to the output sample rate selector would be beneficial when a DVD transport is already supplying a 96kHz datastream to the DSP. There's no need to rewrite the data when the sampling rate is already 96kHz. Indeed, my observations on the reproduction of Classic Records' 96/24 DVD of the Rachmaninoff *Symphonic Dances* leads me to believe that re-writing the data may be detrimental when playing 96/24 source material. The lack of a bypass mode also makes the DSP incompatible with HDCD discs, which may be of concern to some users.

But, these are minor points. The DSP is an excellent product, at a very fair price, and is highly recommended.

RECORDINGS USED FOR THIS REVIEW

Ravel: *Alborada del Gracioso*. L'Orchestre de la Suisse Romande conducted by Ernest Ansermet. London 433 717-2.

Dukas: *The Sorcerer's Apprentice*. Boston Symphony Orchestra conducted by Charles Munch. RCA Victor Living Stereo 68978-2 ("The French Touch").

Rimsky-Korsakov: *Scheherezade*, Op. 35. Chicago Symphony Orchestra conducted by Fritz Reiner. RCA Victor Living Stereo 68568-2 (UV22-Encoded Limited Edition Gold CD version of 68168-2).

Mussorgsky/Ravel: *Fictures at an Exhibition*, especially track 2, "Gnomus." Chicago Symphony Orchestra conducted by Fritz Reiner. RCA Victor Living Stereo 68571-1 (UV22-Encoded Limited Edition Gold CD version of 61958-2).

Schoenberg: *Five Fieces for Orchestra*, Op. 16. London Symphony Orchestra conducted by Antal Dorati. Mercury Living Presence 432 006-2).

Wagner: *Der Ring des Nibelungen*, especially "Siegfried's Death and Funeral March" from *Götterdämmerung* (CD 4, Tr. 10-11), and the "Forging Scene" from *Siegfried* (CD 2, Tr. 3-5). Birgit Nilsson, Wolfgang Windgassen, et al. Vienna Philharmonic Orchestra conducted by Georg Solti. Decca 455 555-2.

Rachmaninoff: *Symphonic Dances*, Op. 45. Dallas Symphony Orchestra conducted by Donald Johanos. Classic Records 96kHz/24-bit DVD (played on a Pioneer DV-525 DVD player with a 96kHz/24-bit S/PDIF digital output). ❖

Xpress Mail

DISTORTION, MEASURED AND PERCEIVED

Thanks for publishing Jean Hiraga's article "Harmony and Distortion" (*aX* Feb./Mar. 2002); we need to continue the research into tonal perception probably started by Helmholtz (the classic "On the Sensation of Tone").

I'd like to address Figs. 11 and 12 (pp. 50–51, March). Figure 12 shows a monotonically decreasing harmonic series, with the second 22dB below the fundamental; this spectrum is said to sound like a pure tone. Interestingly, the ear itself is said to have about 1% second harmonic distortion (–40dB) at 100dB SPL, for lower frequencies at least; loud live music peaks are well above 100dB SPL, causing comparable ear harmonics to Fig. 12. (No mention of SPL range is shown for this study.) A well-known effect among flutists is the low-frequency IM difference "wolf tone"; with two flutes playing close intervals, an inharmonically related "phantom bass flute" is heard, due to the ear's distortion.

But what's most interesting to me is Fig. 11 in light of this. I doubt that the sonic characteristics of the eight amplifiers can be attributed to the harmonic distortion spectra shown, because nearly all the harmonics are at least 80dB down, about 60dB lower than those in the pure-tone-sounding Fig. 12.

The 400W amp, said to be very harsh, has a maximum harmonic (third) of about -85dB (but at what power level?); the ear itself has much more third harmonic at high SPL! Now I realize that Fig. 12 may be a specific "formula" that either just sounds pure or compensates the ear's nonlinearity (the latter would require specific phases); the ear with over 120dB range (trillion to one power) could still be sensitive to much lower distortion than its own.

But I propose a different explanation: An amp tested at a constant level into the usual resistive test-load is a long way secluded from the real world. Musical peaks often reach clipping even at moderate levels, so the whole power range must be considered.

However, the most serious flaw in normal amp testing may be the resistive load. We all know how reactive most speakers are, but consider how profoundly this exacerbates amp distortion: you have a wildly-gyrating elliptical load-line plunging the output circuitry into all kinds of normally-uncharted regions (sounds figurative but is literal here). The "wild gyrations" are from the vast multitude of simultaneous musical tones exciting the usual speaker impedance variations (magnitude and phase) across the audio band.

So it would be at least a step in the right direction to use a small number of standardized reactive loads (for example, 30° and 60°, inductive and capacitive) at several frequencies, while measuring distortion.

And to "exercise" the amp, a good choice could be 11 pure sines at semitone spacing (1:2 1/12 = 1.059463...). This 11/12 of an octave span contains no integral, or harmonic, relationships, so all amp distortion would be distinguishable from the original. (I've tried this on a Hammond electric organ; while the tone generators, magnetic wheels, are separate, the amp/speaker IM is very audible.) This is, of course, an extension of the standard two-tone IM signal, but with much higher peak/average ratio. *Table 1* shows a wider-band test with amplitude weighting.

I believe this test signal with reactive load would be so revealing that (1) amplifier manufacturers would never publish the very unpretty results, and (2) more important, the audiophile would begin to see strong correlation between a measurement and the sound.

A much easier and perceptual test is to feed two separate sine sources into the L and R inputs of a preamp with a stereo/mono switch, then into the stereo amp(s) and speakers. First listen in stereo (each of the two sines has its own separate path to your ears; no IM except for the ears). With frequencies of 600Hz and 680Hz, for example, at a high enough SPL your ears will generate an IM difference tone ("wolf tone") at 80Hz, clearly audible.

Then switch to mono; now both sines coexist to cause amp distortion (perhaps speakers also). Chances are the distortion increase will be dramatic, plus a multitude of additional tones may rear their ugly heads–but beware–chances are you may not like your amplifier any more!

One other factor should be considered: the often-quoted possibility that certain harmonic "additions" (a nicer word than "distortions") in audio equipment may (1) simply sound pleasing,

TABLE 1
PROPOSED 9-TONE IM TEST

INTERVAL = 13 SEMITONES = 2 13/12 = 2.11892618872

FREQ. (HZ)	PITCH	REL. LEVEL	RMS VOLTS
41.203	E	–6dB	0.2200
87.307	F	–3dB	0.3107
185.000	F#	0dB	0.4389
392.000	G	0dB	0.4389
830.61	G#	0dB	0.4389
1760.00	A	–2dB	0.3486
3729.31	A#	–4dB	0.2769
7902.13	B	–6dB	0.2200
1674.40	C	–8dB	0.1747

9-Tone Combination:
RMS = 1.0000V
Peak = ±4.0554V
Peak/RMS = 4.0554 = 12.16dB
Lowest 2nd-order IM product = 46.104Hz

An attenuator should be adjusted so that when fed a single sine wave of ±4.0554V peak (2.8676V RMS), the amp just reaches full-rated power output. Then, with the above 9-tone signal, peak output will only occasionally be reached, but never exceeded. Average output power will be 12.16dB lower, or 6.08% of full-rated power. This and the spectral amplitude weighting shown are fairly representative of complex wideband music signals.
CAUTION: This test may be hazardous to your amplifier's self-esteem!

and/or (2) restore some live overtones lost or not adequately registered in the recording process, or not heard due to the absence of many 3-D spatial cues in 2- (or 5.1) channel stereo. I think this may be the mechanism in Fig. 13, where feedback reduces the higher harmonics.

You must realize that these are *not* attenuated *source* harmonics; the amp is simply not producing as much distortion! But again, this is only the response to one pure tone; it can't be telling the whole story by any means. It may simply be a matter of what *type* of distortion is the least unpleasant or most euphonic, or most compensatory of recording deficiencies.

None of this is a criticism of this article; even though I disagree with some conclusions about the data, there is a wealth of detailed information presented. And I couldn't agree more with the statement that the most difficult research is yet to be done.

Dennis Colin
dcolin@worldpath.net

SCHEMATIC GLITCH

In the "Filtered Audio Oscillator" article (*Audio Electronics* 4/99), the power-supply connections of the LTC1063 filters appear to be incorrect. According to the Linear Tech datasheet, pin 3 is V–, pin 6 is V+. Otherwise, a good article.

Jack Walton
Short Hills, N.J.

Dick Crawford responds:

Thank you for your recent letter describing the error in the power supply connections. Indeed, the Linear Technology 1063 should have V- connected to pin 3 and V+ connected to pin 6.

POWER SUPPLY UPDATE

In your November 2001 issue Stuart Rubin wrote an article on a mike preamp power supply ("A Switching Power Supply for a Mike Preamp," p. 32). I built the 12V section of the supply and found two problems. One is that the 2N2169 transistor isn't heavy enough to carry the load of two 12AX7s, let alone four, as he suggests. It crosses to an nte126a. The other is that Figure 2 shows a 100n cap from the 50k res to ground and Figure 4 shows a 1n cap to ground.

What is the correct part for that application? Which cap should be used?

Larry Epperson
Glen Burnie, Md.

Stuart Rubin responds:

Larry, thank you very much for pointing out these two errors in my article. First, in the 12.6V step-down supply, you apparently did a fair amount of research to cross-reference the incorrect part, then determine it was under-rated for this application. The correct transistor should be 2N6109 (not 2N2169), available from On Semiconductors and probably others. This bipolar transistor is rated at 7A, 50V, which is more than adequate for this application.

Second, you are correct that the value of C6 in Figure 2 should be 1nF (not 100nF). It is correct in the full schematic, Figure 4.

I hope that this has not caused you or anyone any serious problems. I would appreciate hearing about your results in a future letter to the magazine.

I think there is a design flaw in Stuart Rubin's article. There is an inverse relationship between the required inductor and the current drawn by a switcher as he describes. Using National Semiconductor's guidelines, the 150V DC/2mA supply should have a value of about 8mH, not 220µH. The formula is:

$L = 2.5 \times ((Vin)^2) \times (Vo - Vin) / (f \times Io \times (Vo^2))$.

(See the Nat Semi website for applications relating to the LM3524n; Nat Semi and Linear Technologies also have great tutorials on their sites.)

Not being an EE by trade, I don't want to go throwing bricks into anyone's design. I have built several (many) switchmode power supplies and can attest to the fact that there aren't many ways to go insane more quickly.

Jack Walton
Short Hills, N.J.

Stuart Rubin responds:

Jack, thank you very much for your interest *in my article. It is very rewarding to know that someone with so much switching power-supply experience has taken the time to analyze my work. Your assertion about the inductor and current relationship is absolutely correct, as is the equation for the case of the step-up architecture.*

I originally designed the circuit to run just at the edge of the audio-frequency range, 20kHz. The value of 220µH was derived for the case of the 12.6V step-down circuit. Using the equation for the step-down regulator found in the National Semiconductor application note, I derived the value of 210µH:

$$L = (2.5 \times Vo \times (Vin - Vo)) / (Io \times Vin \times f)$$

and Vo = 12.6V, Vin = 15V, Io = 1.2A, and f = 20kHz is derived.

The closest value I had available to me when I built the prototype was 220µH. This worked, so I decided to move the frequency a full order of magnitude higher, which ended up about 300kHz. The circuit still worked fine at this frequency without changing the inductor, so I left it. The other two supplies in the circuit (150V and 48V step-up) both worked with 220µH, so I chose to keep the values the same for all three supplies.

This opens the possibility for further optimization of the supply. Changing the inductor values could definitely improve the efficiency of the design, but could also make the noise figures better or worse, add physical size and cost, and who knows what else. I encourage anyone building this circuit to take the time and tailor the values to his or her specific application.

NOT MY CUP OF TEA

An article by Andy Nehan entitled "No-Feedback Voltage Regulators" appeared in the October 2001 issue. The proposition that a regulator could operate without feedback intrigued me. I had heard of feedforward regulators, which are currently items undergoing some research, but the regulator described in the article actually works with negative feedback.

If you examine the basic configuration shown in Figure 9 of the article, you can quickly establish how it works. You start with a fixed input voltage applied across the pass transistor's collector and the ground terminal and a load attached between the emitter and ground terminal. A current will be drawn out of the emitter terminal and the difference between the base and emitter terminals determines the current flow to the load.

In fact, I_c, the collector current, is

(to page 72)

Henry Kloss

from page 6

next things—including what became, at Advent, the first two cassette decks with Dolby B noise reduction.

That, to my mind, was Henry at his best—cutting to the chase, and making elegance serve a much greater number of people. He did a lot of that. When I called Andy Kotsatsos of Boston Acoustics recently to check on a couple of my recollections, Andy, who had worked so closely with Henry on speaker design at Advent, reported that when he reviewed his own memories, he was stunned to realize just how many major home electronics developments since World War II had come in one way or another out of Henry's work.

THREE PIECES

But not all of them are what you would think. Chances are that when you look at shelves full of mass-market compact stereo systems at Circuit City, you'd have trouble identifying Henry with them. But in many ways his proudest accomplishment inadvertently led to them. It was his advocacy of the simple, direct idea of "three piecedness." That is, Henry took the whole phenomenon of component audio of the time and got it down to the essence: "What one wants is three pieces—the music source and controls in one box where one can get at them easily, and the two speakers placed separately across the room where they sound best."

The KLH Models Eleven and Fifteen were the first of these, but the Models Twenty and later the Twenty-Four were what made the idea take off. These two products, at $400 and $300, respectively, offered so much sound of such high quality that Henry's reduction of the component idea to three boxes actually *became* the idea of components to countless young people of the late 1960s and early '70s.

The very success of these products, though, was what helped usher in the mass-market audio-commodity era, dominated by the giant manufacturers. The trigger was partially that the 1960s had made music systems proliferate enough by the mid-1970s to reach the magic 10-percent-of-the-market point. That was the point at which they were visible enough to jump dramatically into the remaining 90%.

But the impetus was also Henry's relentless reduction of the component idea of separate pieces to the three that made sense for the greatest number of people. That was something the big guys could readily understand, and Henry respected them for getting the point (eventually) and running efficiently with it.

CONSUMER ADVOCATE

One thing that happened along the way, however, as we watched the first imitations of our systems, was that we became increasingly aware—with increasing annoyance on all our parts, but especially Henry's—of the difference between designing equipment for the home, which we did, and designing for the audio showroom, which we believed the others did. Among other things, our music systems used deliberately understated, semi-translucent pearl-gray control panels with simple white, knurled knobs that came with walnut inserts. The imitations, the "compact stereos," on the other hand, had flashy chrome or faux gold looks, and lots of flashing lights.

Henry muttered about this for a while, but although he was one of the world's most pragmatic people, who thought nothing of taking shortcuts that were anathema to audio purists, he wouldn't give in on the difference between designing for the home and for the showroom. (That applied even to some very advanced components that were venerated by audio purists, including the KLH Model Sixteen amplifier and the Advent Receiver.) The point was to make things that people would use with pleasure, not just buy.

And Henry took almost a little-boy kind of delight in doing things that would grab a listener from the moment he got a product home. All our music systems, for instance, were shipped with the volume set at a moderate position, the source set to FM, and the vernier tuning dial set to the most frequently used FM frequency in the US. As soon as you plugged the set in, there—*voila* and to your shock—was *music*.

I think Tom DeVesto, who now runs Tivoli but then was President of Cambridge Soundworks, deserves tremendous credit for managing to solve the whole home-vs-showroom problem for Henry, many years after our frustrations with audio dealers at KLH. After some rebuffs by dealers at the outset, Tom built Cambridge Soundworks' beautifully executed direct-to-the-consumer way of selling audio/video products, and then opened the company's own stores. Tom made it possible for Henry to have a company that not only "did it right" in audio for the ultimate consumer, but that probably also became one of the preeminent "customer-advocate" companies of our time.

Not surprisingly to those who knew Henry, this kind of victory wasn't enough to overcome other frustrations and keep him from eventually developing the same seven-year itch that had made him move on earlier from AR to KLH to Advent to Kloss Video, before landing at Cambridge Soundworks. But it did give him the chance to conduct what turned out to be his most successful pursuit of the irreducible minimum in a whole series of products. (It also gave him a chance during the last few years at Cambridge to give himself a business card on which his self-designated position in the company was "*Eminence Grise*." The carefully restrained glee with which he handed you that card was vintage Henry.)

Strangely enough, Advent's highly prepossessing seven-foot-screen projection TV system, which was about as far as you can get from today's little Tivoli radio, also represented a kind of irreducible minimum to Henry. "If one wants to watch television," he was certain, "one wants to see it." And that first seven-foot epicenter of a true home theater was nothing if not visible.

Watching it, I became aware that the hypnotic effect of conventional television was tied in part to the small screen, which was almost as perfect as the hypnotist's pendulum or shiny dime for inducing a kind of trance state. With Advent's VideoBeam, on the other hand, your eyes weren't locked into the little rectangle of conventional television, and the experience was vastly different. (The locked-in fascination that often developed anyway was for other reasons than trance.)

That first seven-foot Advent screen actually was the smallest screen size Henry ever believed was just right for full involvement of the viewer. The six- and five-foot Advent and Kloss Video screens that came later were not as satisfying to his mind—let alone today's relatively small-screen home theater systems.

PUBLIC RECOGNITION

Getting an "Emmy" for his work on projection television vastly delighted Henry. (His *persona's* combination of shyness and assertiveness, which had been waiting for that level of public recognition, was good evidence for the contention that being shy comes from fear of not being appreciated as one really is.) But what I think Henry really deserves an award—and abundant appreciation—for is his unremitting advocacy of *rightness* in products.

He wasn't always right himself. All of us who had to market them can remember frustrations such as the way the Advent cassette deck had to be held in rewind or fast-forward by the user, and the way you had to turn on the automatic turntable in the KLH Model Eleven portable (and the wood-cased Eleven-W) to get the "Aux" inputs to work. But no one else I know of in the audio-video world came anywhere near Henry's ability to get product design down to its essence. And I think the only non-audio electronic product currently that carries on Henry's full notion of elegance—of doing everything needed without an ounce of excess or some penalty of use—is the laptop computer.

It seems fitting to have written this appreciation of Henry on one. Every time I use it, there's reason to think of him. As there is whenever one hears satisfying sound—especially from small, modest boxes. ❖

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given approximately by an equation:

$$I_c = I_s \exp (V_{be}/kT).$$

If the so-called saturation current, V_{be}, is the base-emitter voltage, k is Boltzmann's constant and T is absolute temperature.

If you reduce the load's resistance, the base-emitter voltage difference will increase slightly and the transistor will then provide more collector current to fulfill the demand for increased load current. In other words, the transistor is sensing an error signal V_{be}, making sure that V_{out} is always $V_{ref} + V_{be}$. This is clearly the action you would expect to see in a negative feedback loop.

This is not a "no feedback" regulator in any sense.

David Bourner
Staff Engineer Field Applications
National Semiconductor
Columbia, Md.

Andy Nehan responds:

I thank David Bourner for taking the time and trouble to pen his comments, but I think that he and I differ fundamentally in our understanding of the word "feedback." There is no feedback error amplifier in my circuit—whereas a "normal" regulator—e.g. 78XX series—has an error amplifier within it, whose purpose is to keep the output voltage constant. This is usually achieved by the mechanism set out crudely in Figure 1 of the article.

The output of my circuit is not as constant under differing load conditions as, say, a 78XX series device—since there is no feedback to apply a corrective signal to the pass element. As the load current changes, so the load voltage will change. This can be expressed another way by saying that my regulator has a high output resistance—in the region of tenths of an ohm rather than milliohms for a "feedback" regulator.

Look at it in another way. Imagine a perfect regulator with an absolutely fixed output voltage independent of load current, with—for the sake of argument—no feedback within it. Then place a resistor in series with the output. The voltage at the load end of this resistor would vary with the load current—but there would be no feedback involved, merely the application of Ohm's law. The pass device is similar: perfect transistor with an inherent emitter resistor but no feedback.

To my understanding, describing the output voltage of the regulator as being V_{ref} plus V_{be} is a tautology.

However, this is all rather irrelevant. What actually matters is that this regulator sounds different than feedback regulators. I placed it forward as a proven design for those who want regulation but without the use of three-legged feedback devices and who wish to be able to tailor regulators to their needs. You may like the sound or you may not—but that's a personal preference. I like my tea black, but people whose opinions I value actually put milk in it—bizarre!

GREAT ARTICLE

I just finished Stuart Rubin's article entitled "A Switching Power Supply for a Mike Preamp" (Nov. 2001, p. 32) and am fascinated by it. I've been working on a linear power supply for a headphone amp I built from HeadWise, (Szekeres Class-A MOSFET headphone driver) and have been trying to get the hum out of it. Everybody at HeadWise advises against using switching supplies, but your article makes a lot of sense.

I am very interested in your project. I have wished to build a pair of amps for some time and just don't know where to start. Have you built your preamp yet and have you written anything about it?

I belong to an Irish band and do all the recording for them. If I had a quiet, good, mike preamp and a good set of mikes, I could do a much better job.

Steve Everett
Hollidaysburg, Pa.

Stuart Rubin responds:

Thank you very much for taking the time to respond to the article. As a first-time author, it's pretty exciting to get such a positive response. I think this little switcher could do a nice job on a headphone amp. My thought is that if it's quiet at about 80dB of gain for a microphone, it should be quiet enough for unity gain (?) for headphones.

I've built three of the mike preamps and all of them have been working pretty well. Two of them are very happy in local studios, and the third is being demo'd in various places. One of the mikes was recently used on an album co-produced by Jimmy Johnson of Muscle Shoals fame. I haven't written about them because I was hoping to sell more of them before I gave away the design! But, since this is not my full-time job, I have not been able to devote too much time to marketing them. I suppose that under the right circumstances, I could publish the design, sell a kit, blank PCBs, or something like that.

If you're interested in purchasing one of the preamps, or parts to build one, let me know!

HELP WANTED

I play bass guitar and want to make a vented box. I have two drivers—Electro-Voice EVI series 150W, 8Ω, 10″ and 15″ (EVI-10 and EVI-15). Where can I find detailed technical support for those drivers? I've checked the Electro-Voice official website, but I couldn't find what I needed. ❖

Vladimir Cukic
vcukic@tesla.rcub.bg.ac.yu

Readers with information on this topic are encouraged to respond directly to the letter writer at the address provided. –Eds