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Editorial

Building Beauty

By Edward T. Dell, Jr.

I once had the good fortune of standing in Meijii Park in Tokyo on Meijii Day, a national celebration throughout Japan. One of the many ceremonies taking place was a demonstration whose name I never discovered. A large crowd gathered before a rough, raised platform, with a roof of straw with a railing surrounding the floor area, about waist height for a dozen or so boys dressed in formal kimonos.

An old gentleman was master for the ceremony, which was, apparently, a competition among the participating boys. The master called each of the boys in order. For each of them he placed on the railing a bundle of straw, approximately 3″ in diameter.

In turn, each was handed what appeared to be a very sharp, two-edged sword about three feet in length. In a clearly defined ceremonial set of motions, the boys moved the sword from behind, up over their heads, and down into the straw bundle.

The object of the exercise, I was told, was to cut the bundle into two equal pieces, leaving *one* straw uncut. Nearly everywhere you look in Japan, there are examples of what, to my eye, can only be evidence of some national habit of a consistent, unremitting care for precision and for beauty. I doubt anyone will discover an aesthetic gene in the current classification of the human genome, but if someone does I will not be surprised. I would also not be surprised if it turned out to be a national trait in France as well as Japan.

One of my pleasures as editor of this publication is the magazine exchange with an extensive list of audio periodicals worldwide. In many avocations this would not be very useful, because of the language barrier. Happily, in electronics, there are schematics whose symbology is universal—or nearly so. My monthly mail includes two periodicals of especial interest.

One is *Revue du Son, et du Home Cinema* edited by the legendary Jean Hiraga. What is so surprising in this periodical is the style and design of the equipment, of which the magazine reviews an impressive quantity each issue. Hiraga keeps track of the major moves in the home audio area, with especial attention to tubes, and each month features a home theater installation, but also publishes at least one article per issue on amateur construction. This is doubtless a holdover from his days as the editor of *L'Audiophile*, renowned for innovative articles and ideas on vacuum tubes long before the present revival took place.

The French undoubtedly hold the title for unusual design in the audio field, especially as regards loudspeakers. A great many are fairly pedestrian boxes, but when a show opens, the most eye-opening surprises are usually in the speaker category.

The other magazine of note is *MJ* (*Musen to Jikken* translates roughly as Audio Technology). This 240-page monthly periodical has been around since the end of WWII and is predominately dedicated to DIY audio in all of its forms. It pays much attention to new products as well, but also devotes two pages monthly to a tube museum.

Most startling about the content of *MJ* are the extensive construction features. They are not only of great variety—a recent issue included ten DIY projects—but the majority are also indistinguishable from commercial products. A good example graces the pages of the first issue of *audioXpress* from the pen and hands of Satoru Kobayashi. This is *not* a junk box project and the author will have absolutely no spouse acceptance problem, I would guess.

Beauty is not an accident—and especially not in audio equipment we build with our hands. The Japanese do not have an exclusive claim on component beauty, however. In that same issue, note Tom Perazella's beautiful ribbon speaker. Beauty and aesthetics are not cheap, however.

I do not know what these projects may have cost their authors, but the primary ingredient is a commitment to how the finished device looks. We fortunately live in a time when special services abound in metal cutting, finishing, and engraving, not to mention exotic woods and high-tech construction aids. Such things are not cheap, but they can lend a quality to the finished product which can add a lot to the value and satisfaction we derive from the undertaking.

The aesthetic dimension requires a commitment to the idea, to begin with. Then follows the time required to research what might be possible in building any audio device. The computer is also a powerful adjunct to audio construction, where the web offers accessibility to almost any materials or services the builder might want. The CAD capabilities of even inexpensive programs to aid physical design are miles beyond what was possible even a decade ago.

Many of you reading this page have specialized knowledge about such services, materials, tools, and the like. Your shared ideas can enrich the work of every amateur in our group. I hope 2001 can be a year when we do a lot of thinking about the aesthetics of what we build. Those of you who have built a project you are proud of, but perhaps don't wish to write extensively about it, are encouraged to take some photos, write a caption for each, and send them along for publication in these magazine pages.—**E.T.D.**

audioXPRESS

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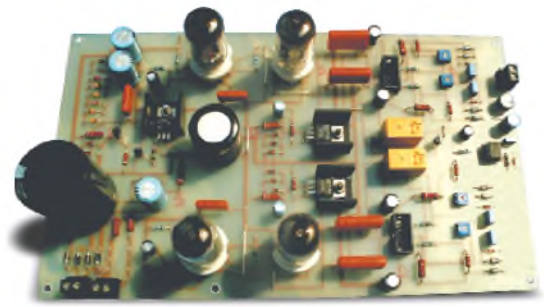

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CODE

Articles and letters identified with the following symbols:

solid state tubes speakers

THE STAFF

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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Audio News

➲ IKONOKLAST™

Ikonoklast™ is a small (35″H × 8¼″W × 12″D) 2-way floor-standing system made to reproduce powerful deep bass using only a 5¼″ midrange driver. The Ikonoklast has a sensitivity of 95dB/2.83V/m for use with very low powered amplifiers, such as vacuum tube single-ended triodes. It is finished in 2-ply hardwood veneer. Warren Gregoire & Associates LLC, 229 El Pueblo Place, Clayton, CA 94517, 1-800-634-0094 or 925-673-9393 (international), FAX 925-673-0538, website: www.warrengregoire.com.

■ OCTAL 6

Mapletree Audio Design is offering a new line-level preamp kit called the Octal 6. It is the first in the Hollow State Club Series, which features vacuum tube audio components with unique layouts and features, designed by Dr. Lloyd Peppard, a retired Professor of Electrical Engineering. The Octal 6 is a dual-circuit stereo line-level preamplifier employing six octal tubes, permitting any of three input sources to be switched to a shunt-regulated push-pull (SSRP), a common-cathode/cathode follower circuit, or a passive path. The kit includes all parts, hardware, wire, solder, NOS tubes, a punched and painted chassis with brass trim, and a 40-page illustrated step-by-step instruction manual. Mapletree Audio Design, RR1, Seeley's Bay, Ontario, Canada K0H 2N0, (613) 387-3830, http://hollowstate.netfirms.com, e-mail: peppardl@post.queensu.ca.

■ MK 10 MICROPHONE

Gold Line introduces the multi-purpose MK 10 omnidirectional measurement/recording microphone, which is designed for ultra-flat frequency response. The MK 10 features a ½″-diameter pre-polarized condenser capsule with an omnidirectional polar pattern, so it is ideal for certain studio and live recording applications such as music ensembles, pianos, and choirs. It has also been included in Gold Line's "Prokit" Measurement System.

The MK 10 has a frequency range of 20Hz-10kHz ±1dB or 10Hz–20kHz ±2dB, a maximum SPL of 128dB, and low self-noise. Each mike comes with a serial number, individual frequency plot, foam padded carrying case, and mike clip. The housing is machined brass with matte silver finish. Gold Line/TEF, Box 500, West Redding, CT 06896, (203) 938-2588, FAX (203) 938-8740.

■ PARTS EXPRESS CATALOG

Parts Express, a full-line distributor of electronic parts and accessories for the consumer electronics industry, the technical hobbyist, and the commercial or residential installer, announces the release of their comprehensive, 308-page catalog. They stock a selection of raw loudspeaker drivers for home and automotive applications, home theater and home automation products, alarm systems for home and auto, mobile video, tools and technical aids, test equipment, computer accessories, chemicals, telephone products, wire, connectors, instructional books and videotapes, speaker design software, cellular phone accessories, stage lighting, and pro sound equipment. Parts Express, 725 Pleasant Valley Drive, Springboro, OH 45066-0611, 1-800-338-0531, www.partsexpress.com.

■ TRANSCENDENT SOUND PREAMP KIT

Transcendent Sound announces its entry into the kit market with the Grounded Grid Preamp kit. The unit is based on a simple yet sophisticated wide band gain stage, which is featured in the book *Audio Reality* by Bruce Rozenblit. Performance is flat from 5Hz to 300kHz with harmonic distortion less than .02%. The preamp encompasses a "purist" approach with a three position selector, volume control, and power switch. The cabinet includes a brushed stainless steel lid with hand grained and etched faceplate. Transcendent Sound, PO Box 22547, Kansas City, MO 64113, (816) 333-7358, FAX (816) 822-2318, www.transcendentsound.com.

■ KT-88 AMPLIFIER

Accurate! Sound Labs (a Division of Accurate Electronics) announces a new line of high-quality imported tube audio amplifiers. One model features the KT-88, driven as a Class-"A" triode mode, monoblock amplifier. Very detailed highs and balanced lows, as well as extensive quality control and testing facilities ensure hassle-free performance from every unit. Accurate! Sound Labs, 6692 Brockton Ave., Riverside, CA 92506, (909) 686-6550, FAX (909) 686-6536, www.accuratesoundlabs.com.

■ VIRTUAL ELECTRONICS LAB™

Protel International Ltd. announces the release of Circuitmaker 2000-The Virtual Electronics Lab. This new version of Circuitmaker™ software now provides complete schematic design, simulation, PCB layout, and autorouting in one box. New features include a new waveform viewer, a new device browser, and a streamlined user interface for easier access to various features.

Circuitmaker 2000's Professional Edition PCB layout editor (formerly TraxMaker PRO) also includes a new rip-up and retry maze autorouter. Circuitmaker 2000 allows a designer to capture a design, perform advanced circuit simulation, analyze the operation of a circuit, and complete the design and testing with full-featured PCB design and layout capabilities. It is available in both Standard and Professional editions. Protel International, Level 3, 12a Rodborough Rd, Frenchs Forest NSW Australia 2086, +61 2 9975 7710, FAX +61 2 9975 7720, website: http://www.protel.com.

A Modular Hybrid Amp System

Tubes or transistors? Whatever your prejudices, this article may make you change your mind about how you view hybrid designs.

By Marco Ferretti

Forget for the moment tubes, resistors, capacitors, and output transformers, but think about what your system is asked to do. Yes, you know the answer: your equipment should reproduce music as closely as possible to the way you'd hear it in a live performance. All the instruments should sound real, and you must receive the impression that they are being played right in front of you. How is this done? Simple: you need a source, a preamp, a power amp, and a pair of speakers.

DESIGN APPROACH

Your equipment should be designed to offer the best musical performance possible depending on costs and on the technology you are willing to use. It is often said that tubes sound better than transistors. I think this is not completely true, since the quality of sound does not depend only on the devices you are using, but more on how they are used.

The approach when designing audio equipment is first of all to obtain the best electrical performance from the components you choose, and then to optimize the circuit for the best sound. If you are not completely satisfied, abandon the first approach and look for a different design.

Now, looking at a generic transformer-coupled tube power stage, you find that it has great handicaps: it is unable to supply large amounts of current to the load; its damping factor is very low; it phase-rotates in the low-frequency region; it is quite difficult to achieve an output power over 50–60W while still having good musical performance; finally, all of these and other parameters vary greatly depending on the regularity and nature of the load itself.

The list is long, but the greatest handicap is that tubes are unable to supply enough current to the load. This means that with some speakers you'll never obtain good driving (and listening) performance from a tube amp.

Now that the problem is clear, you can look for a solution. I am convinced that tubes do sound better than transistors, but the problem is how to achieve an exciting tube-sound driving performance from the power stage while avoiding the solid-state devices I need to use to affect the sound.

VOLTAGE-AMPLIFIER STAGE

I can affirm that what makes good sound in an amp is mostly the first stage, the voltage amplifier. It adds gain, which means distortion, nonlinearity, and noise, but if carefully designed to operate with all the rest of the amp circuitry, it can really make a difference. The most suitable device to use here is a tube; I chose an SRPP design because of its low distortion, considerable gain, low output resistance, striking dynamics, and good slew rate.

The power stage is less critical; after all, it exists only to move current and drive the speakers. My choice here was a complementary solid-state-follower configuration, since it has no voltage gain and almost no distortion; it can move a great amount of current and deliver high peaks of it to the load; it has a very high damping factor and low output resistance; and, finally, it's very reliable.

Now the question is: bipolar or MOSFET? Transistors are current driven. If you really wish to use them in your design, you need at least two current amplifiers to drive their bases, but this means more active devices that the precious signal must pass through, and

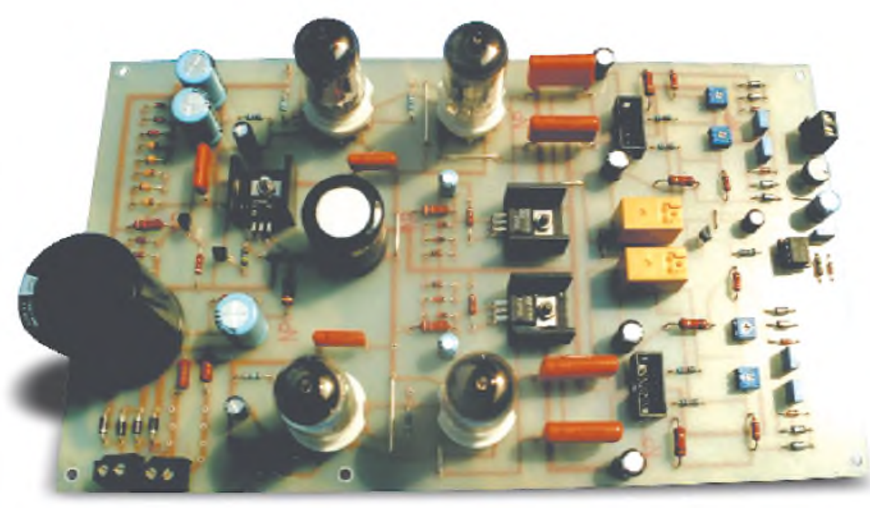


PHOTO 1: Top view of driver board with all tubes mounted.

ABOUT THE AUTHOR

Marco Ferretti is currently a student of electronic engineering at the Turin Polytechnic in Italy. He became involved with tube amplification in high school, and then was attracted to audio reproduction. In the summer of 1994 he built his first power amplifier, a KT88 push-pull based on the famous Leak TL50. Since then, he has designed and built a number of amps, including single-ended designs.

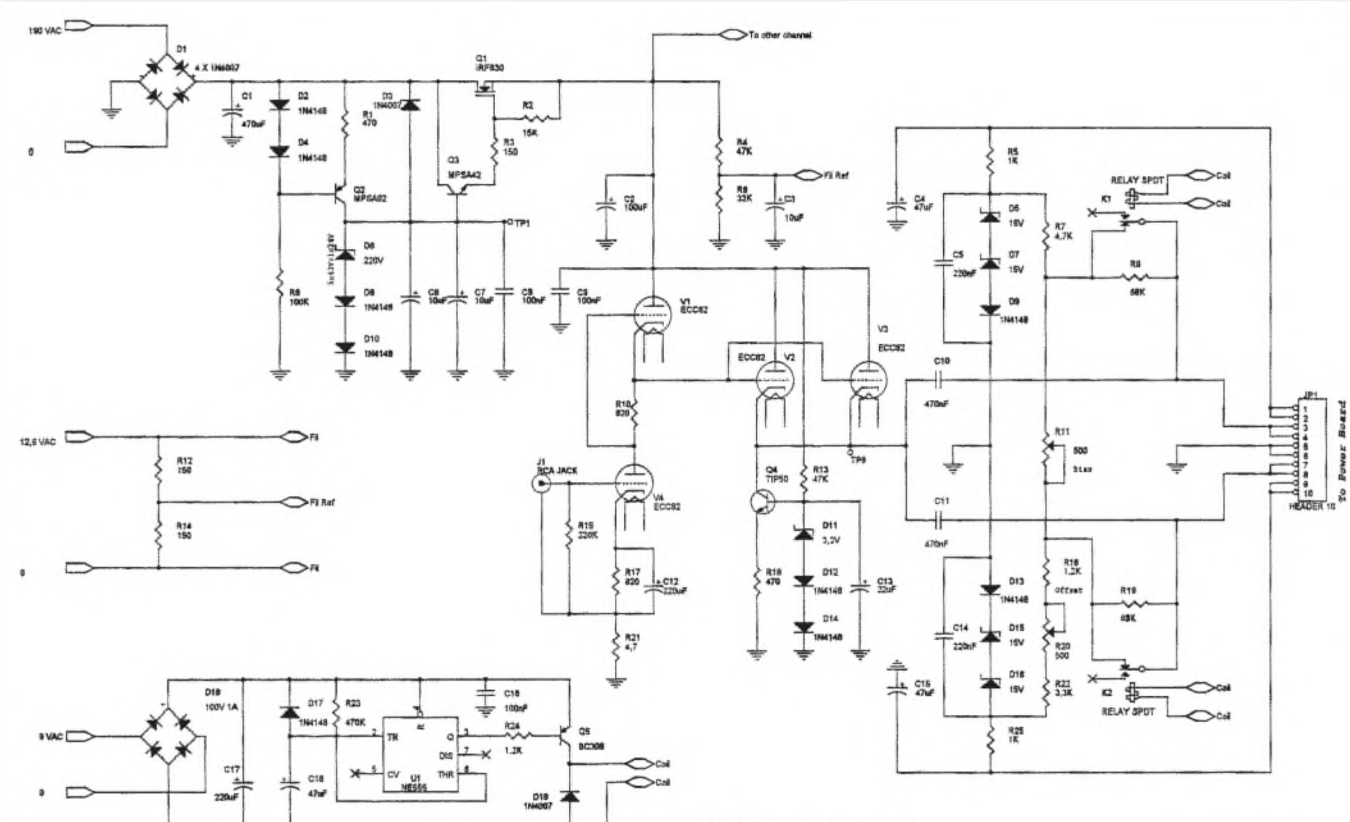


FIGURE 1: Driver board.

G-1554-1

this is not good for sound!

Instead, MOSFETs are voltage driven, so you can derive the signal directly from the gain stage and apply it to their gates. Easy? Not at all! MOSFETs usually have huge source-to-gate and gate-to-drain capacitance, which means their gates will absorb current. The input stages must be able to supply this current; otherwise, frequency response and dynamics will be seriously compromised.

I have focused on the main points of the design: a tube gain stage, a tough driver, and a MOSFET power stage; and if what I've said is right, you'll have an amp with superb driving performance that sounds very like tube equipment.

The driver stage is AC-coupled to the power stage, which offers an enormous advantage in that you can choose the number of devices your power stage will contain without changing anything in either the driver stage or bias network, since MOSFETs are voltage driven. Now you are dealing not just with a single amplifier, but with a complete amplification system that allows you an output power of from 50–200W. All ver-

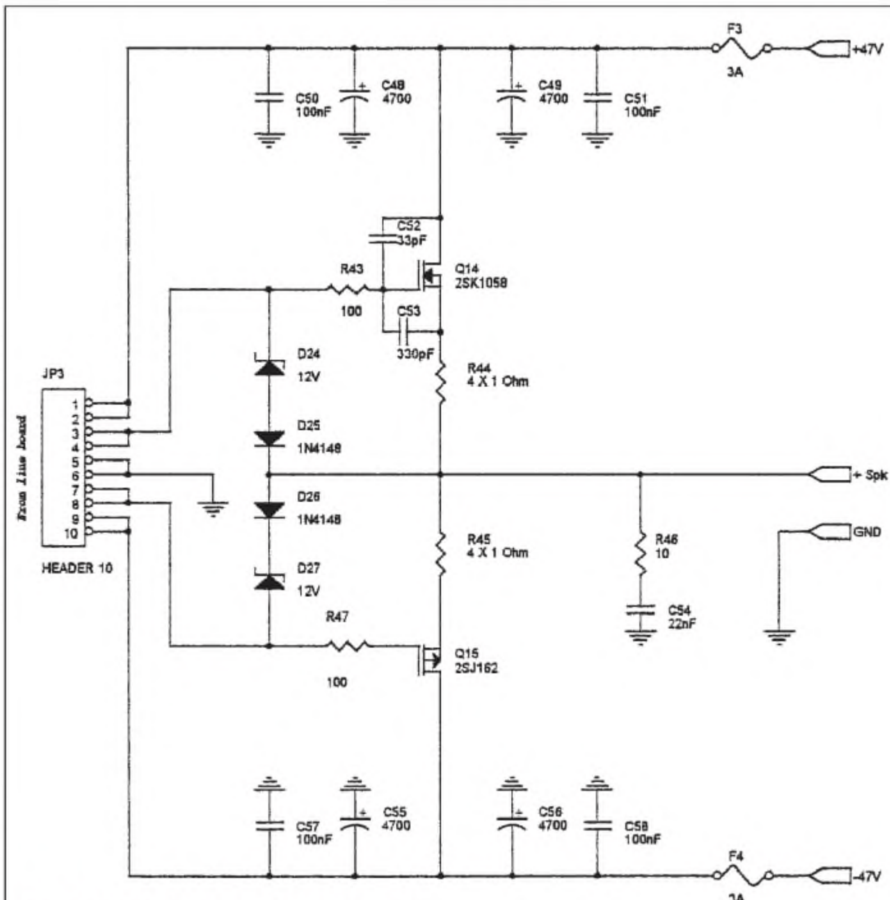


FIGURE 2: 50W power-board schematic.

G-1554-2

sions will have the same timbre, the same sound, but different power, depending on how many devices you mount on the power stage.

This is exactly what I was looking for: an audiophile amplification system for my home-theater room. The 200W version powers the main speakers, and less powerful versions drive the other channels that are provided by a Dolby digital decoder.

AMPLIFIER DESIGN

One of my goals when I began to design this amp was not to use overall feedback; in other words, this needed to be an open-loop design. I think the only way to achieve a stable open-loop amplifier is to set a relatively low gain and let the preamp drive the power amp at full power.

There is only one voltage amplifier here (*Fig. 1*). It's a shunted regulated push-pull (SRPP) stage, which I think is the best solution, since it has a high gain and a good slew rate; it can deliver a large voltage swing, and it has low output resistance. It is based on an ECC82/12AU7, which sounds much better than the ECC83/12AX7 in an SRPP configuration and has a lower, but sufficient, gain.

The power stage (*Figs. 2–5*) uses from two to eight complementary power MOSFETs in parallel. I have chosen the Hitachi 2SK1058 and 2SJ162 because of the great amount of current they can handle and for their S = 1 transconductance. I have already used these devices in a solid-state power amp, and was impressed by their reliability and very good musical performance.

It is true that the power stage is symmetrical like a tube push-pull, but it needs no phase splitter, since this power stage uses complementary devices.

Notice that these power MOSFETs are used in a follower configuration (the N-channel loads the P-channel and vice versa). They therefore have no voltage gain, which means no added distortion and nonlinearity; they will only move current and supply it to the load.

GREAT ELECTRICAL PERFORMANCE

The electrical performance of such a power stage is remarkable: almost no added distortion, a very high damping factor, a large amount of current delivered to the load, and extremely low output resistance. Now the question is how to drive it? It's true that MOSFETs are voltage driven, but they have a considerably high source-to-gate and gate-to-drain capacitance, which means they do absorb current from the driver when a signal is applied to the gates. Even if the SRPP has relatively low output resistance, it would not be able to drive four pairs of MOSFETs.

Now imagine that all the input ca-

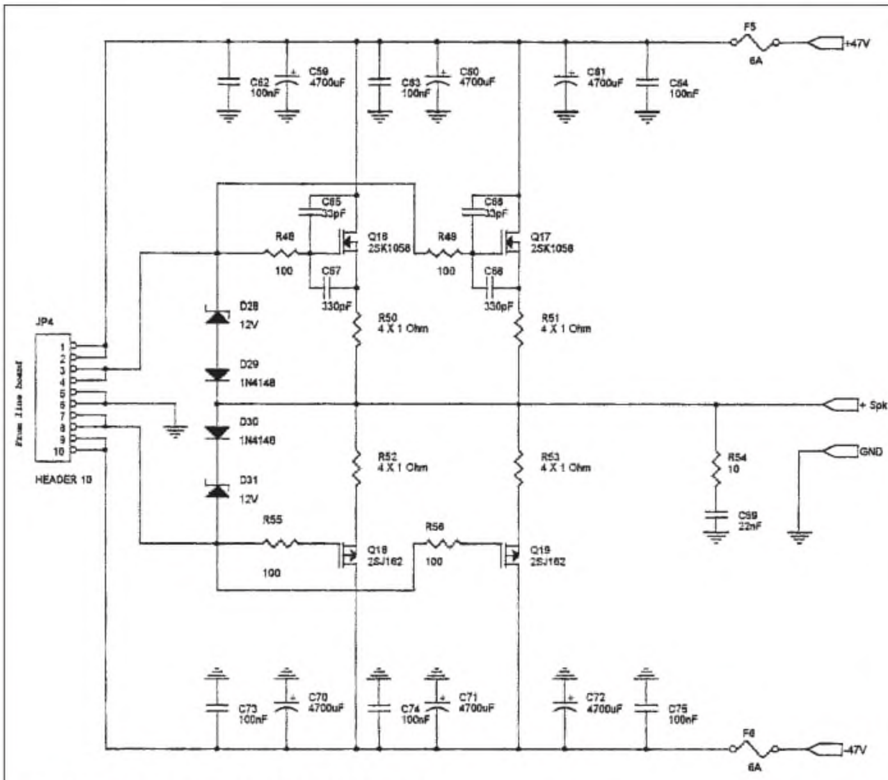


FIGURE 3: 100W power-board schematic. G-1554-3

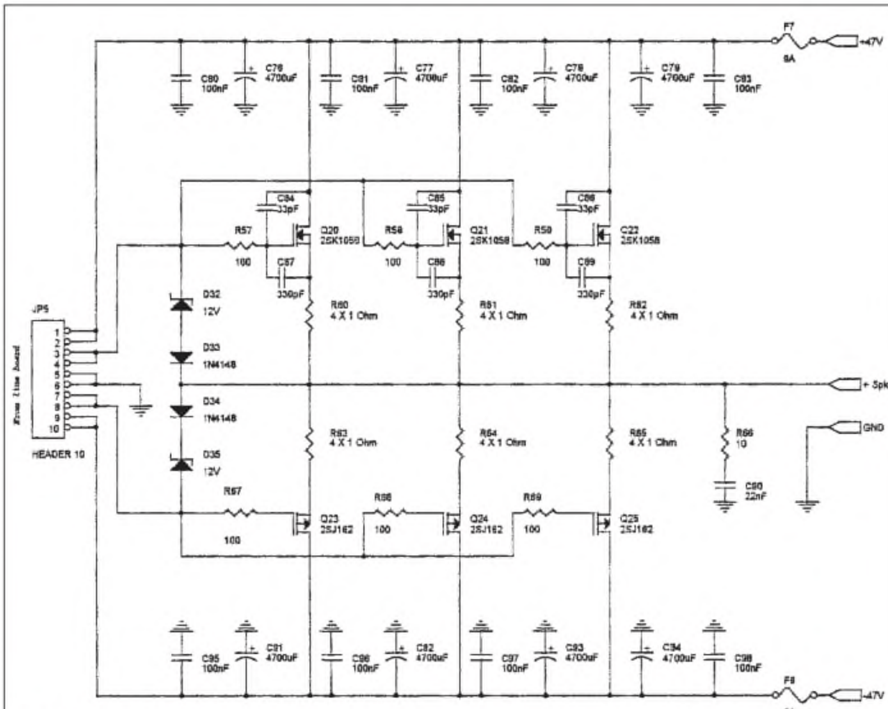


FIGURE 4: 150W power-board schematic. G-1554-4

pacitance of eight devices is paralleled. This requires an extra driver stage, based on a tube so it can be directly coupled to the SRPP and loaded by a solid-state constant-current generator in order to enhance its performance. I paralleled two sections of an ECC82/12AU7 to increase the current flow and reduce the output resistance.

The power stage is biased through a resistor network, where you will find two trimmers, one to set the bias current, and the other to correct and eliminate the DC offset.

What about the relay? During tube warm-up, the voltage on the driver's cathode increases slowly until it is about half the supply voltage. Now this voltage is not constant, but variable, so it will pass through the coupling capacitors and the power stage will amplify it until the driver's cathode is at its fixed potential. As a result, you have a huge DC voltage with almost no current limitation on your speakers. You can imagine what will happen.

During tube warm-up, the relay will short R9 and R19, connecting the gates directly to the biasing network, which has low resistance. In this way, the voltage swing on the driver's cathodes passing through the coupling caps will be absorbed by the biasing network and will not drive the power stage. The speakers are safe now! This is how the amp works. Now let's examine the reasons for some tricks you will probably notice while looking at the schematics.

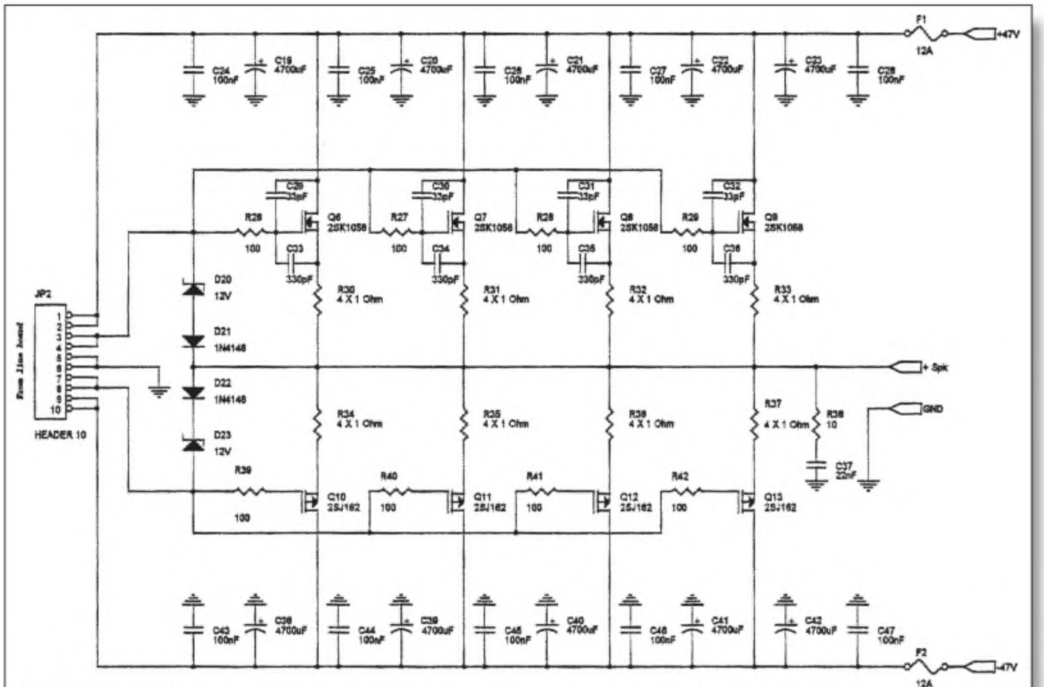


FIGURE 5: 200W power-board schematic. G-1554-5

CIRCUIT TRICKS

Consider the input plug. The negative lead is not grounded, but is connected to the R17/R21 node. R21 is not there to provide local feedback to the input stage since its value is too low, but it acts as a ground decoupler. I had some trouble with noise and hum, and this was the easiest way to make the amp mute.

This trick, often used in preamplifiers and solid-state amps, is very simple and works well, but if your amp is still noisy due to a ground-loop caused by the input connections between the amp and preamp, try increasing R21 up to 15Ω. Bear in mind, however, that there may be other causes that make the amp noisy; in any case, check your layout first.

On the power boards, there are a couple of zeners, called "clippers," between the gates and the speaker outputs that limit the current flow to 12A through each device, since a higher current might cause them to fail.

I said before that MOSFETs have high interelectrode capacitance, but P devices have higher capacitance than do N devices, and since the power-stage design is electrically symmetrical, all N devices must be compensated. To do this, increase their internal capacitance with two capacitors mounted as close as possible to each MOSFET so they will all have identical capacitance. This will make the power stage more reliable by decreasing intermodulation distortion at high frequencies and increasing bandwidth.

TABLE 1
DRIVER-BOARD PARTS LIST

R1	470Ω, 1W	C10, C11	470nF 630V MKP Solen, Wima, or others
R2	15kΩ, ¼W 1% metal film	C12, C17	220µF 25V
R3	150Ω, ¼W 1% metal film	C13	22µF 25V
R4, R13	47kΩ, 2W	C16	100nF 603V MKT
R5, R25	1kΩ, 2W	C18	47µF 25V
R6	33kΩ, 2W	D1	4 × 1N4007 bridge
R7	4.7kΩ, 1W	D2, D4, D8–D10, D12–D14, D17	1N4148
R8	100kΩ, 2W	D3, D19	1N4007
R9, R19	82k, ¼W 1% metal film	D5, D7, D15, D16	Zener 15V 1W
R10, R17	820Ω, ½W 1% metal film	D6	5 × zener 43V + 12V zener, 1W in series
R11, R20	500Ω trimmer cermet	D11	Zener 3.3V 1W
R12, R14	150Ω, 2W	D18	100V 1° bridge
R15	220kΩ, ¼W 1%	Q1	IRF830
R16	1.2kΩ, 1W	Q2	MPSA92
R18	470Ω, 1W	Q3	MPSA42
R21	4.7Ω, ¼W 1%	Q4	TIP50
R22	3.3kΩ, 1W	Q5	BC308
R23	470kΩ, ¼W 1% metal film	U1	NE555
R24	1.5k 1% metal film	V1, V2, V3, V4	ECC82
C1	470µF 400V	JP1	10-pole connector
C2	100µF 400V	K1, K2	12V relay
C3, C6, C7	10µF 400V		
C4, C15	47µF 63V electrolytic		
C5, C14	220nF 63V MKT		
C8, C9	100nF 400V MKT		

On all the MOSFETs' source pins, four 1Ω resistors are paralleled (on the schematics, only one resistor is shown). This is because power resistors have a little leakage inductance due to the way they are constructed (they are usually made of a wire wound on a ceramic base).

The advantage of having four resistors paralleled is that you can use 1Ω carbon or metal-film resistors that have no leakage inductance, which is very important in maintaining a stable and reliable operation.

To make this amp work, you will need the driver board (*Fig. 1*) and one of the four proposed power stages (*Figs. 2-5*), which you just plug into the driver stage to enjoy your favorite music! The driver-board parts list is in *Table 1*, and the parts lists for the power stages are in *Table 2*.

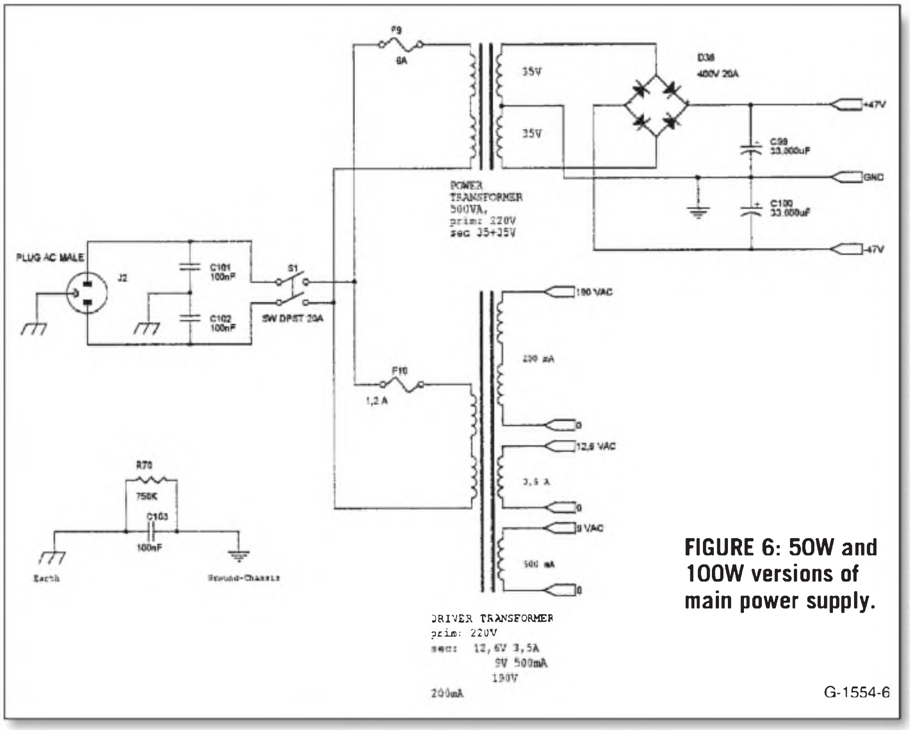


FIGURE 6: 50W and 100W versions of main power supply.

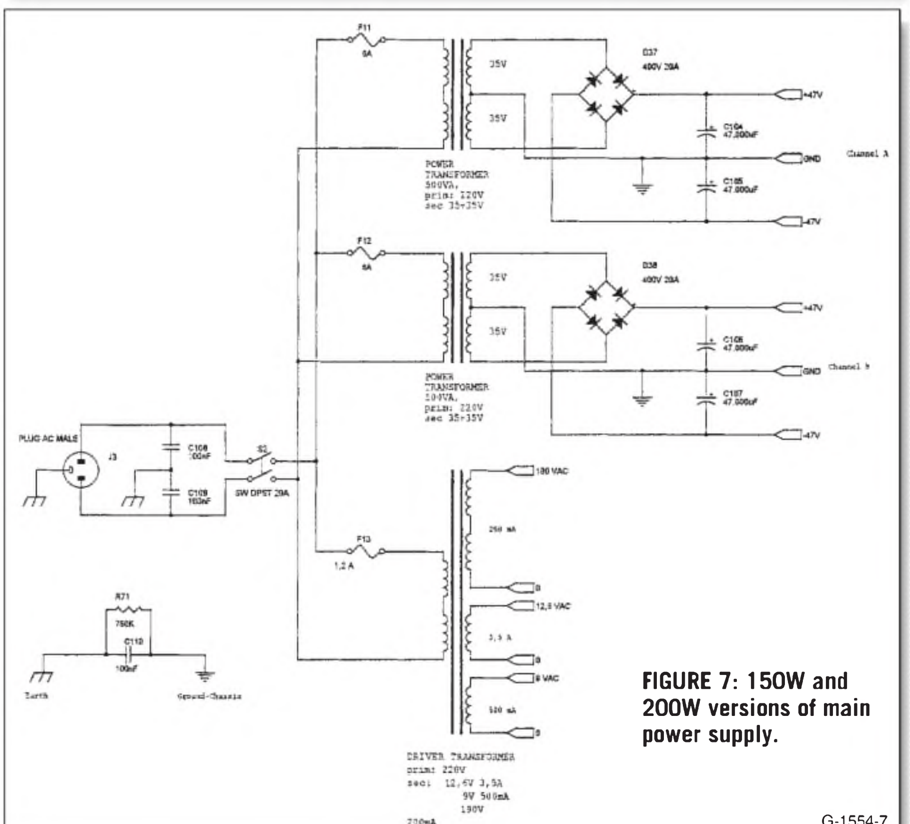
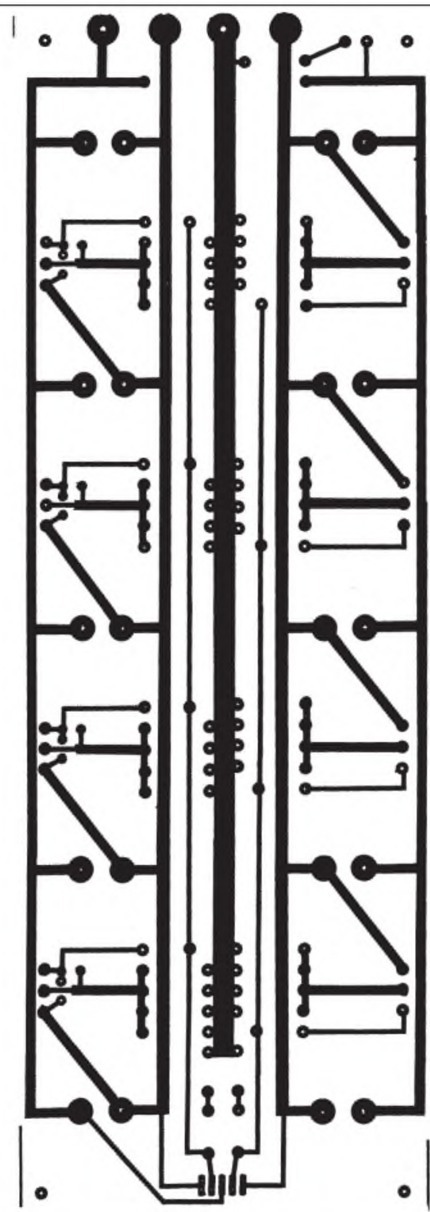


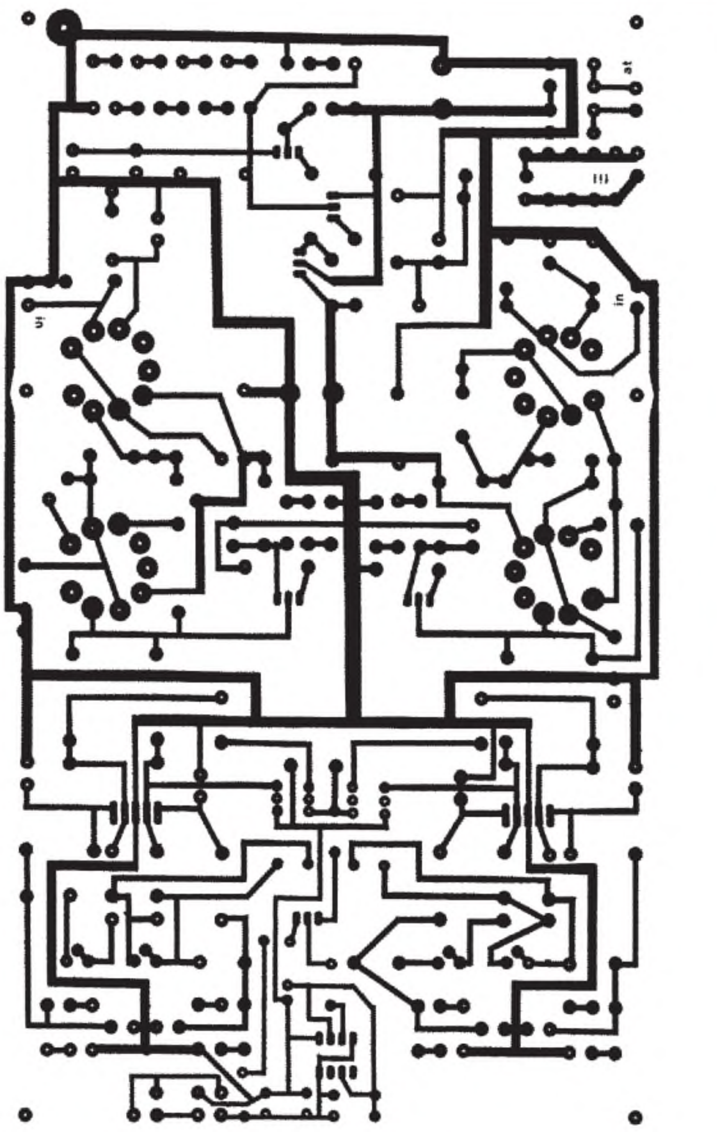
FIGURE 7: 150W and 200W versions of main power supply.

TABLE 2
POWER-STAGE PARTS LISTS

TWO-MOSFET:	
R43, R47	100Ω, ¼W 1% metal film
R44, R45	4 × 1Ω, 1W paralleled
R46	10Ω, 4W
C48, C49, C55, C56	4700μF 63V
C50, C51, C57, C58	100nF 100V MKT
C52	33pF
C53	330pF
C54	22nF 100V MKT
D24, D27	Zener 12V ½W
D25, D26	1N4148
Q14	2SK1058
Q15	2SJ162
F3, F4	Fuse 3A
JP3	10-pole connector
FOUR-MOSFET:	
R48, R49, R55, R56	100Ω, ¼W 1% metal film
R50-R53	4 × 1Ω, 1W paralleled
R54	10Ω, 4W
C59–C61, C70–C72	4700μF 63V
C62-C64, C73–C75	100nF 100V MKT
C65, C66	33pF
C67, C68	330pF
C69	22nF 100V MKT
D28, D31	Zener 12V ½W
D29, D30	1N4148
Q16, Q17	2SK1058
Q18, Q19	2SJ162
F5, F6	Fuse 6A
JP4	10-pole connector
SIX-MOSFET:	
R57–R59, R67–R69	100Ω, ¼W 1% metal film
R60-R65	4 × 1Ω, 1W paralleled
R66	10Ω, 4W
C76–C79, C91–C94	4700μF 63V
C80–C83, C95–C98	100nF 100V MKT
C84–C86	33pF
C87–C89	330pF
C90	22nF 100V MKT
D32, D35	Zener 12V ½W
D33, D34	1N4148
Q20–Q22	2SK1058
Q23–Q25	2SJ162
F5, F6	Fuse 9A
JP5	10-pole connector
EIGHT-MOSFET:	
R26–R29, R39–R42	100Ω, ¼W 1% metal film
R30–R37	4 × 1Ω, 1W paralleled
R38	10Ω, 4W
C19–C23, C38–C42	4700μF 63V
C24–C28, C43–C47	100nF 100V MKT
C29–C32	33pF
C33–C36	330pF
C37	22nF 100V MKT
D20, D23	Zener 12V ½W
D21, D22	1N4148
Q6–Q9	2SK1058
Q10–Q13	2SJ162
F1, F2	Fuse 12A
JP2	10-pole connector



◀ **FIGURE 8: Power-stage board.**



▲ **FIGURE 9: Driver-stage board.**

POWER SUPPLIES

The input stage's power supply is regulated; the circuit is the same as one I have used on the SV572-10 amplifier. A constant-current generator fixes a current through the zener series, and an npn transistor drives a power MOSFET used as a series regulator. This power supply, housed on the driver board, is the same for all versions.

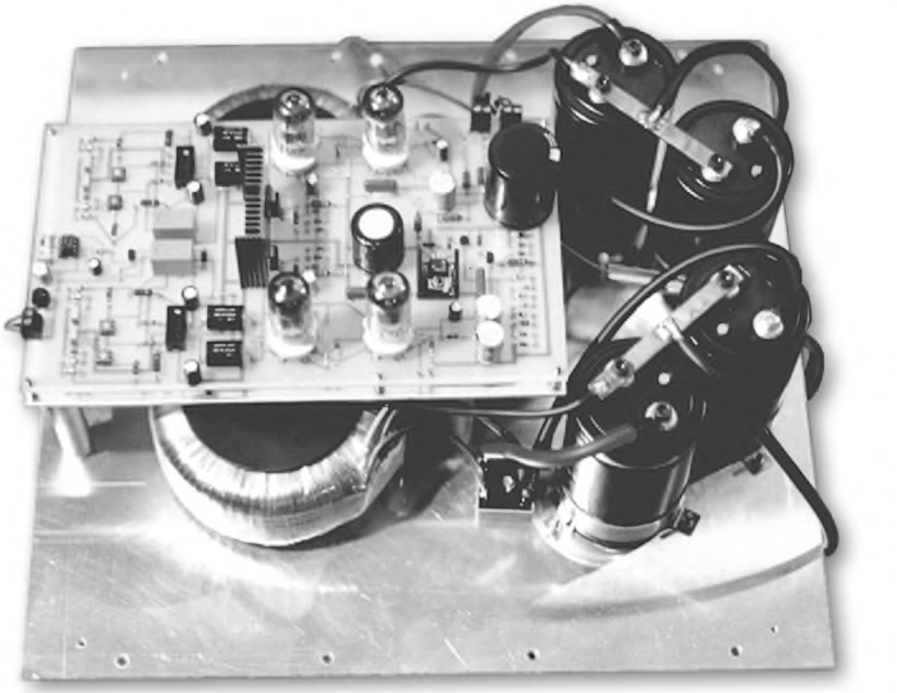
I mentioned before that this amp was designed to have a very high-performing power stage; as a consequence, its power supply must be very tough. Each version has its dedicated power supply (*Figs. 6 and 7*) giving current to the power stage. As you can see, these supplies are somewhat oversized because they must act as great energy reservoirs capable of delivering high peaks of current.

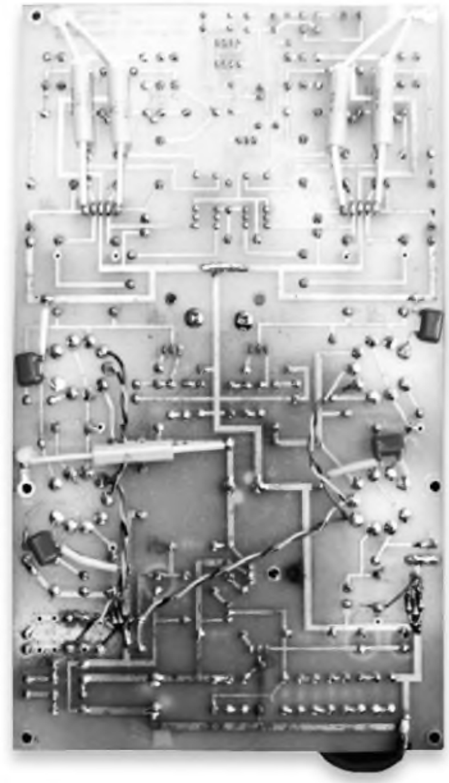
On the 8-MOSFET version of the power-stage board (*Fig. 8*), you will find ten capacitor cans mounted and paralleled five by five. This is still part of the power-supply design: five small capacitors paralleled have a lower R_{esr} than an equivalent single capacitor, and finally, each power MOSFET is connected directly to the closest capacitor, allowing a very short connection between the power device and its power supply.

Unfortunately, having the filtering capacitance directly on the power board is very expensive. It also takes a lot of space, which necessitates having very long routes on the PCB—not a healthy situation. So additional capacitors are mounted on the amplifier's chassis to allow you to reach the de-

TABLE 3
MAIN POWER-SUPPLY PARTS LIST

(SIX- AND EIGHT-MOSFET VERSIONS)	
R70, R71	750kΩ, 1W
C99, C100	33,000µF 63V
C101–C102 C108–C109	100nF 630V
C104–C107	47,000µF 63V
D36–D38	Bridge 400V 20A
S1, S2	20A switch
F9, F11, F12	6A fuse
F10, F13	1.2A fuse
T1a, T1b	500VA toroidal transformer, primary 220V, secondary 35 + 35V
T2	Toroidal transformer, primary 220V, Secondary: 190V 200mA, 12.6V 3.5A, 9V 500mA





▲ **PHOTO 2: The driver board mounted above the power transformers.**

▶ **PHOTO 3: Copper-side view of the driver board.**

sired capacitance value. Refer to *Table 4* for the appropriate values for each power-stage supply.

The 6 and 8 MOSFET versions have a double power-stage power supply, with separated transformers and filtering units for each channel, while all other versions have a common power supply for both channels. Finally, the driver board is supplied by its dedicated power transformer.

Whatever version you choose to build will have a two-channel driver board with its power transformer and two power boards with their power supplies. Just plug one of the power stages into the driver board, and your amp will start working.

MOUNTING THE DRIVER BOARD

First of all you must make your PCB (*Fig. 9, Photo 1*) and have someone cut a 3mm-thick aluminum board with the PCB's dimensions to shield the driver stage from the power transformers. In my layout, the power transformers are mounted just under the driver board, which I know is not the optimum solution, but I lacked enough space in my standard four-unit 19″ rack.

On the driver board, all connections between parts are routed except for the filament supply. Don't forget that filaments are 12.6V operated, so you must connect the 12.6V to pins 5 and 6, leaving pin 9 unconnected on all tube sockets. Once you have your PCB finished, mount it on the aluminum board, leaving a 15–20mm space between them (*Photo 2*).

In order to reduce hum, connect a 100nF 400V capacitor between pin 9 of each tube socket and the aluminum board (*Photo 3*). The same must be done for each DC supply: the high-voltage drivers' supply and both positive and negative legs of the power stage's supply coming from the power boards to the 10-pin connectors you find on your driver board. All this is not strictly necessary, but it will help in reducing hum.

The driver-board layout (*Fig. 9*) is not updated with the ground decoupling network. You will need to cut the ground routes next to the inputs and

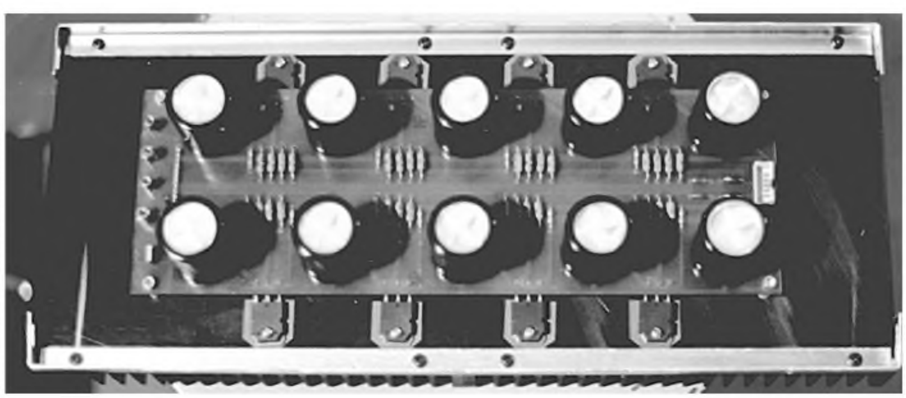


PHOTO 4: Power board mounted on the heatsink.

TABLE 4
POWER-STAGE POWER SUPPLY

VERSION	TRANSFORMER RATINGS	CHASSIS CAPACITANCE VALUE
2 MOSFETs per channel	One 500VA, sec 35 + 35V AC	33,000µF per leg
4 MOSFETs per channel	One 500VA, sec 35 + 35V AC	33,000µF per leg
6 MOSFETs per channel	Two 500VA, sec 35 + 35V AC	47,000µF per leg
8 MOSFETs per channel	Two 500VA, sec 35 + 35V AC	47,000µF per leg

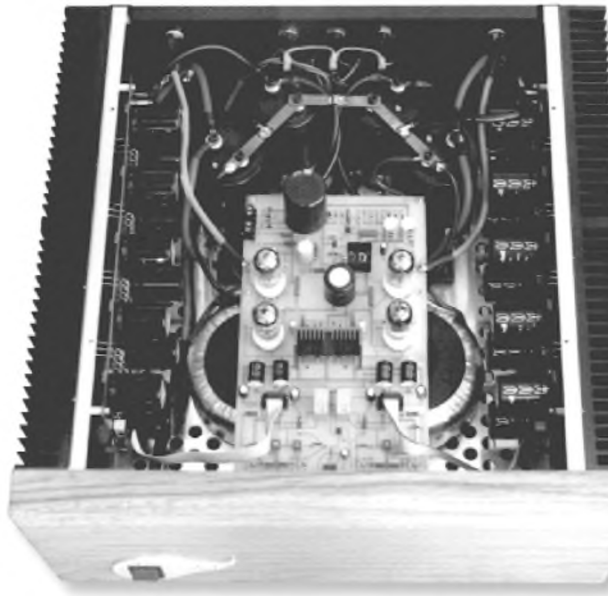


PHOTO 5: Top view of the 8-MOSFET version.

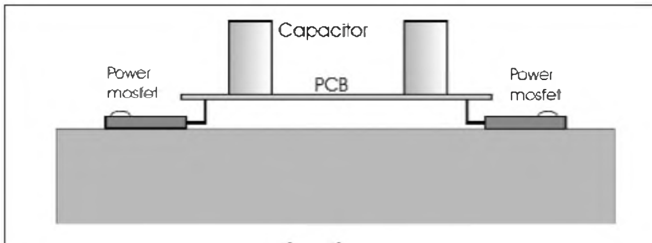


FIGURE 10: MOSFET mounting diagram.

solder R21 under the PCB.

Next to the zener series there is a large ground connection that must be made directly to the power-stage power supply's ground on the capacitors leads. This is the main ground connection between all power supplies.

MOUNTING THE POWER BOARDS

I have mounted and tested a prototype of each version, but I have finished only the most powerful one (8-MOSFET version), so everything I say from now on refers to this amp's layout and construction. But don't worry if you intend to build one of the smaller versions, because they all have many things in common, from the design to the final layout and construction.

Mounting these boards won't cause any trouble (*Photo 4*). By the way, pay attention when you mount the capacitors: the power supply has a positive and a negative lead with respect to the ground, so make certain the polarities of all electrolytic caps are correct. The positive lead supplies all the 2SK1058 MOSFETs, while the negative lead is connected to the 2SJ162 devices.

On the other side of the board, you may mount a 100nF 100V MKT capacitor under each 4700µF can; just solder each film cap directly to the 4700µF capacitor pins. There was not enough space on the component side of the PCB.

All MOSFET pins must be bent at 90° and soldered to the copper side of the board so that you can mount the power devices on the heatsink. *Figure 10* and *Photo 4* may explain much better than words how you do this job. Don't forget that you must mount all MOSFETs on the heatsink with TO3P insulation kits.

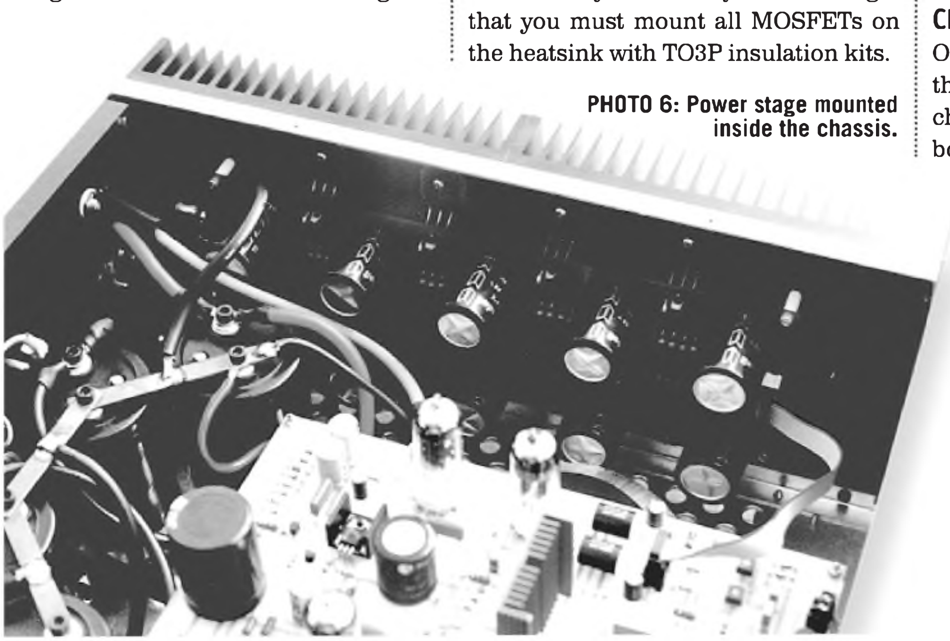


PHOTO 6: Power stage mounted inside the chassis.

DISSIPATING HEAT

All solid-state devices produce much heat that must be dissipated. You will need a couple of large heatsinks. The best is one of those 19″ racks with heatsinks on its sides. A four-unit and 400mm-deep rack is required for the 8-MOSFET version (*Photos 5* and *6*). The 6- and 4-MOSFET versions will fit into three-unit racks, and I think the two-unit rack is sufficient for the smallest version of the system.

There are precise formulas and procedures for calculating the optimal heatsink dimensions according to the amount of power you must dissipate, but it would take too much space to describe all this. A little suggestion, however: when you mount a power device on a heatsink with its insulation kit, the rule of thumb is to be sure the heatsink is large enough so the running temperature never rises above 50–55°C. If you can still keep your hands on it without being burned, it means the temperature is 55°C, or less. But if the heatsink becomes too hot, you will need to mount a larger one.

CHASSIS LAYOUT

Once you have finished mounting the three boards, begin to think about the chassis layout. In my rack, the bottom board was not strong enough, so I replaced it with a 3mm-thick aluminum board. If you have the same problem, you will need to drill large holes in the board so that air may flow inside the box.

As you can see from *Photo 7*, I

TABLE 5
BIAS TABLE

VERSION	BIAS CURRENT
2 MOSFETs per channel	120mA
4 MOSFETs per channel	210mA
6 MOSFETs per channel	310mA
8 MOSFETs per channel	400mA

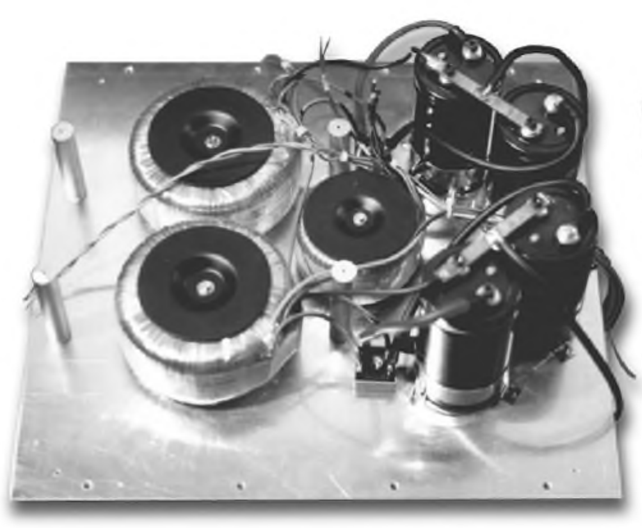


PHOTO 7: Power transformers and capacitor mounted on the bottom board.

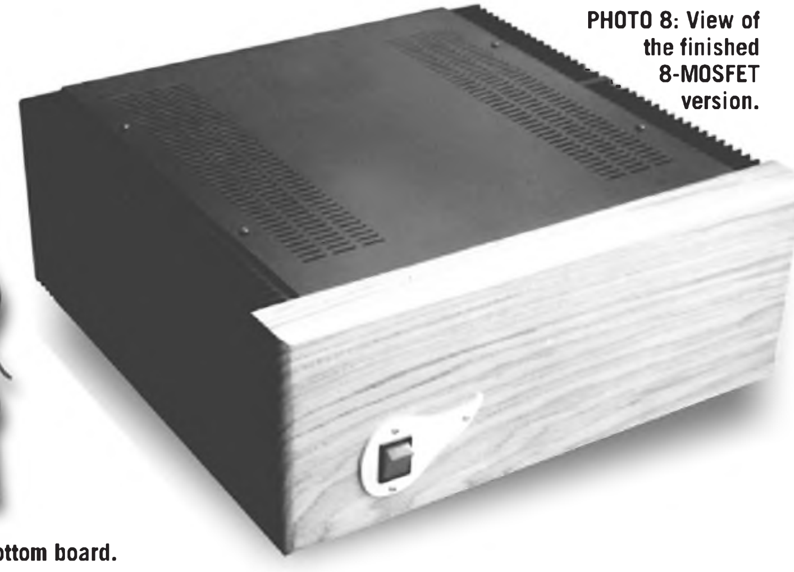


PHOTO 8: View of the finished 8-MOSFET version.

mounted the three power transformers in the mid-front side of the chassis, the capacitor cans in the rear, and the bridges between them.

The driver board is elevated above the power transformers with four aluminum bars (*Photo 2*), and all secondaries of the driver-stage transformer are connected to the driver board. Both center taps of the two 500VA power transformers are wired to the main ground connection, which is directly on the capacitor leads. From there you must ground the chassis, the driver board, and both power PCBs. You must wire the positive and negative legs of the power-stage power supply from each bridge to the capacitor cans (watch out for the polarity!), and from the capacitors to the power boards.

The power-stage power supplies must be wired with 4mm wires. I have used 6mm wires, which you can do as well, but consider that they are not easy to solder. You will need a very big and powerful soldering iron.

I made the ground connections between the capacitor cans with a 10×2mm copper bar, giving a very clean look to the layout (*Photo 5*). But, again, it's really hard to solder a 6mm wire on it!

Don't forget that the input plugs must be isolated from the chassis electrical ground. Finally, you will need a ten-wire flat cable to connect the power PCBs to the driver board.

FIRST START-UP AND TUNING

Once you have finished mounting, check your layout carefully. Disconnect the power stage's power supplies, turn the amp on, and be sure that the input stages do their job correctly (if you have a scope, use it!). Check the delay circuits 15 to 20 seconds after power is on.

TABLE 6

MEASUREMENT TABLE

OUTPUT POWER	
2-MOSFET version	45W
4-MOSFET version	92W
6-MOSFET version	138W
8-MOSFET version	185W
Frequency response	15–60,000Hz –3dB
THD	0.18% 1W@ 1kHz

All measurements are referred to the 8-MOSFET version except for output power.

The relay shorts R9 and R19. After a few seconds' delay, the relay contact opens, unshorting R9 and R19.

Turn the bias trimmer R11 until it shorts and the offset trimmer R20 half way. You must tune one channel at a time, so leave one of the two power supplies disconnected and connect a DC milliammeter in series with the positive leg of the power stage's power supply of the channel you are currently tuning. Then connect to the output a DC voltmeter paralleled to a 10Ω resistor. Switch the amp on and turn the offset trimmer until you read 0V on your millivoltmeter; then set the proper bias current. Refer to *Table 5* (the bias table) and look for the current required by the power stage you have built.

Wait until the heatsink has reached its running temperature and make the corrections needed to both bias current and offset. Be sure that the ammeter leads do not become disconnected while you are tuning the amp, or you will blow all the power devices! Repeat these few steps on the other channel, but before doing so, you must wait a couple of minutes until the capacitors are discharged; or you may discharge them yourself with a 1k 10W resistor connected between the positive and negative legs of the power supply.

Be very precise with the offset tuning. A 20–30mV offset is still permissible, but a 100–200mV voltage is not good for your speakers. Remember that all tunings on the power stage must be done with a 10Ω 10W resistor load connected to the outputs.

MEASUREMENTS

At low levels, the THD is mainly due to the input stage, thanks to the power-stage complementary follower configuration, which does not affect the signal in a significant way. THD increases at higher levels, since the power stage gradually switches from Class A to Class AB operation.

The graph in *Fig. 11* reveals what the 8-MOSFET version is able to do, showing how the power stage reacts with different load impedances. The measurement has been done at full power (soft clipping) in both continu-

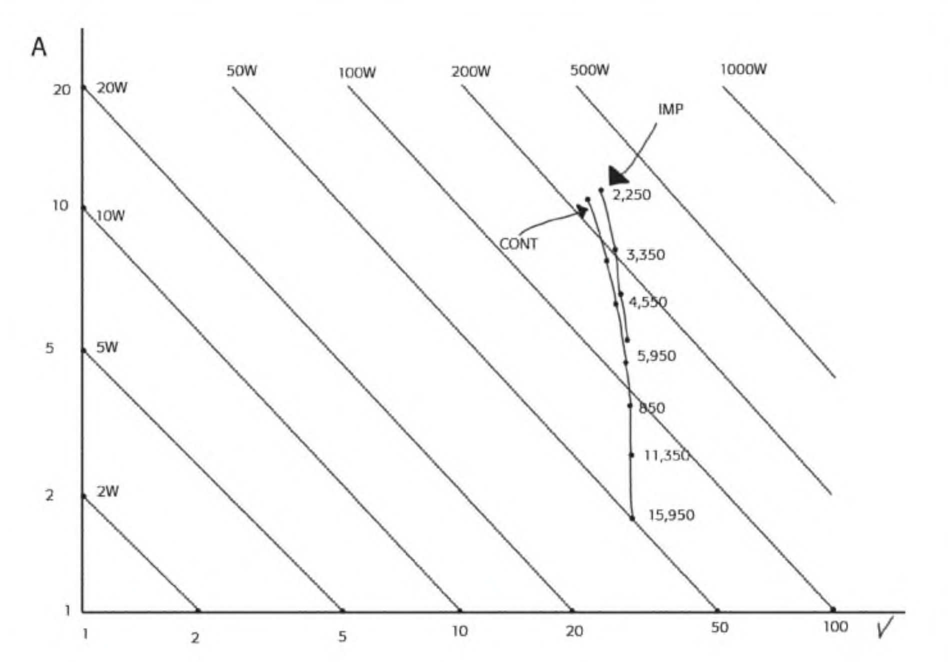


FIGURE 11: Driving performance of the 8-MOSFET power stage at different load values.

ous and intermittent range (the test signal is a pulse signal). You may notice that it looks almost like a straight line, which indicates there is almost no voltage drop when a huge amount of current is required.

As a result, you can see the benefits of the choices in both power-stage and power-supply design. Refer to *Table 6* for further details.

LISTENING IMPRESSIONS

I conducted all listening tests using the following equipment: a Pioneer PD507 CD player with custom output stage, a 6SN7 single-ended buffered by a cathode-follower preamplifier, and Magneplanar MG1.6 speakers.

In all versions, the timbre at low levels was substantially the same. What I said previously was correct: the input stage is the one that forms the sound of the amp. The timbre is very bright, but well balanced, with every detail clear. The high range is very precise, the bass is powerful and well damped, and the mid-frequencies are present and well linked with the upper and lower range.

Sometimes I sense that it is a bit rough in the midband, but less so than in push-pull tube amps. Consider, however, that I am currently influenced by the sound of the SV572-10 single-ended amp.

The virtual image is well focused. Voices and instruments are precisely defined and their images are very stable inside the virtual scene, giving a great sense of depth.

The 2-MOSFET version has very good driving performance, but when high power is required, the image becomes a little confused. I think this is due to the fact that the power stage switches to Class AB at a relatively low output power. This does not happen with the 8-MOSFET version, since Class A is guaranteed up to 50W.

The 200W version has extraordinary driving performances and damping factor. While mid- and high-range performances are substantially identical for all versions, the 8-MOSFET-per-channel amp (*Photo 8*) has an excellent bass response and striking dynamic range, allowing the best reproduction of every kind of music.

I love playing music with this amp at very low levels, because every instrument is so present and detailed, and the dynamic range is so wide, even when the speakers are just whispering. ❖

A Quick Bookshelf Pair

Here is another in our series of basic speaker projects, designed to provide a weekend of learning and fun. **By Lester Mertz**

All too often we speaker builders undertake some huge project, only to be overcome by it, especially if things don't go the way we intend. Maybe this occurs because we haven't warmed up enough, like a pianist running the scales to limber up before a special performance. Or sometimes we just procrastinate and never begin. But here's a fun project just to tackle and enjoy.

This project is a simple bookshelf-sized speaker. I designed it to be about the size of a regular hard-cover book; in fact, it can fit on virtually any bookshelf. It uses a single full-range 5″ speaker, without a crossover or any other special construction techniques.

A FEW OLD PARTS

I desired a speaker that would fit right on the shelf next to my receiver—nothing pretentious, for I listen to the receiver infrequently. I had some simple 5″ speakers on hand—just basic stamped frames with treated paper cones. I had picked them up with the idea of using them as midranges in another project, so I thought I might as well see how they would sound in a small closed box.

I cut some scrap pieces of ¾″ MDF into enough sections to make almost all the sides of the box, but there wasn't enough for the two back panels. That didn't stop me; I just glued together everything I had at the moment, figuring that I'd pick up a small piece of MDF and cut some back-panel sections another time.

The external measurements are 8.75″ tall × 6.5″ wide × 6.75″ deep (without the back on). Since I have a ¾″ round-over bit for my router, I rounded over everything but the incomplete back. It looked a little bit like a Chicklet® (candy-gum), so I stained it with some whitewash.

An old tweeter-baffle template, which is nothing but a masonite board with an appropriately sized hole for guiding a router around the inside to make a perfect circle, was just about right for the front-speaker opening. You can use a saber saw as well; just draw a penciled circle, drill a "start-hole" big enough for the blade to slip through, and carefully cut out the circle.

Check to see that the speaker frame drops down flush with the front surface. I used a half-round file to produce the exact fit. The speaker must be tightly seated against the wood so that no air can leak past the frame. Any unintentional leakage will degrade the sound quality.

SPEAKER MOUNTING

Solder some wire onto the terminals of the speaker, red to the "plus," and black to the other. Mount the speaker and mark the face with a pencil for drilling the pilot-guide holes for the retaining screws. Remove the speaker again and drill the four guide holes, then brush everything clean. Now mount the speaker and carefully install four fine-thread sheetrock screws to hold it firmly. Do not over-tighten, or the MDF will break out, and you'll need to turn the speaker and start again after cleaning and repairing the MDF.

A speaker box needs something inside to damp the sound that's bouncing around, and there are many materials being used today. If you have some polyester batting, hobby-store quilting, or pillow stuffing, use two big handfuls and loosely fill each box.

The project was close to completion, but I wasn't willing to wait. Since I did not have a back, but did have some Masonite® on hand, I quickly cut two pieces to size using this material. It would just be a temporary fix, right? So, to seal the backs, I used some silicon tub and tile sealer I had already opened. Some small brads held the back flat while the sealer set up.

The very next day, I hooked up the new speakers to the receiver, and they stayed there for a good while—all summer, in fact. The sound was satisfying, especially the voices of newscasters and Garrison Kiellor. But when recorded music came on, I would turn the set off, go into my listening room, and get my music fix on the big system.

After "Pairie Home Companion" one evening, a local jazz show started. I was feeling lazy just sitting there listening to a piano and bass duet, when it stuck me that the sound was just not right. It was compressed, if you know what I mean; it just didn't have any life. I played with the remote, trying to adjust the tonal balance and anything else the receiver offered, but after minutes of fussing, I turned it off and headed for the listening room.

DAWNS THE LIGHT

The next morning I awoke with an idea as to how to fix those little speakers. I took them over to the bench and ripped off the Masonite backs. Then I cut some wire screen, the half-inch-square stuff

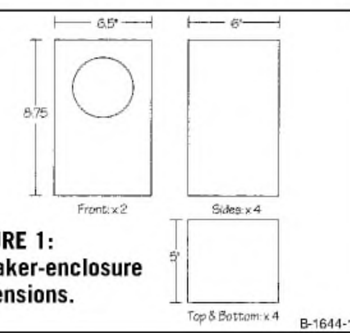


FIGURE 1: Speaker-enclosure dimensions.

that you use around plants, to fit over the back. I found a small sheet of fiberglass padding and, cutting out one piece for the back of each box, I placed it behind the existing polyester stuffing and stapled the wire screen over that.

Several days later, I bought two small pieces of white felt to cover the wire screen, and attached them with spray adhesive. I used scissors to cut the felt to size and shape while it was right on the speaker.

I took the units back to the book shelf and reconnected everything. Wow! The compression was gone, and the little babies were singing music. The polyester fill plus the fiberglass was damping the backwave, but not completely. If I stood close, a couple of feet away, I could pick up some of the reflected wall sound. But seven or eight feet away in the easy chair, it was just great.

The male voices lost any chestiness that was there, and the sound of the small speakers filled the house as never before. They were now much louder, by almost ten clicks of the remote's volume control—a transformation! Now I can really pick out the recorded promos from the live news, and poor taping techniques are very discernible. But the music was alive.

THE OPEN-BACK SOLUTION

There are many old radios still in use that utilize an open-backed cabinet as the speaker enclosure. They all sound pretty much the same, and not bad at that, considering what's there. Usually it's a little stamped-frame paper unit that comes in different sizes—3, 4, 5, or even up to 6 × 9 inches. You can usually find these at a discount electronics shop for a few bucks.

The principle is simplicity itself—the sealed enclosure was restricting the inexpensive unit's ability to move. And it's not that this speaker didn't have a decent magnet; it does—almost as large as the frame—but it wasn't designed for what I was attempting to do with it. Trying to turn a cheap speaker into an acoustic-suspension design (another name for a closed box) didn't work out. But that was initially; you should hear them now! As for you purists, remember that the open-back cabinet was John Dalquist's claim to speaker fame. ❖

More on TL Speaker Design

Here's an update to the article "Another Look at TL Design" (*SB* 3/99), in which the author described a two-way system featuring TL designs for both bass and treble speakers. **By John Mattern**

In my original article, the bass unit covered the bass and midrange, while the treble unit covered the midrange and high frequencies. The treble unit was originally a cone type with a wizzer, but I replaced it with a coaxial speaker of the same size. The 3½″ coaxial speaker does not require external crossover components and therefore has little impact on the electrical diagram.

Because of my wish to use a simple 6dB crossover, I needed a very large overlap of the bass and treble drivers in the midrange. The new treble unit in its TL enclosure responds down to 128Hz, providing more than three octaves of overlap below the 1.5kHz crossover, while the bass speaker provides roughly two octaves above the crossover. This bass-speaker design was based on a computer simulation that resulted in a highly satisfactory unit.

A newly revised computer simulation uses Olson's[1] finite conical horn equation (modified to include losses) rather than Ramo and Whinnery's[2] general electrical transmission-line equation to calculate the load at the throat of the horn. Both approaches gave virtually the same result. Another change to the simulation involved a new model for the velocity of sound that accounts for the dependency of velocity on frequency. I will discuss these changes in more detail later.

BASS-UNIT CHANGES

The bass unit received a coat of semi-gloss black and a full-length grille with narrow black trim on the edges (*Photo 1*). A speaker stand containing a redesigned crossover provided additional height to place the treble unit at ear level. I built a second complete system except that the bass driver I used was of necessity the Audax woven-carbon-fiber HM130C0 that replaces the discontinued HM130X0. Frequency-response measurements showed no significant differences between the two systems.

TREBLE-UNIT CHANGES

Changes to the treble unit are shown in the mechanical drawing of *Fig. 1*. I cut a 3″ port opening to extend the bottom end of the treble unit's response, but this resulted in a sizable dip at 512Hz. When I reduced the opening to 1.75″, the dip was more reasonable. To do this, I inserted a hardboard washer with a 3″ outer diameter and a 1.75″ inner diameter into the 3″ port. The dip was apparently caused by out-of-phase sound from the port. I also reduced the amount of fiberglass stuffing, which now has a 2″ × 3″ cross section instead of the original 3″ × 3″.

I also changed the driver to a Boston Acoustics 3.5″ coaxial driver that has a more extended high end. I can't hear it anyway (11kHz is my current limit). The box was turned on its end with the driver down, and I added side blocks with 45° corners to reduce the diffraction effects above the step frequency. Measurements showed only a slight improvement, suggesting that the fluctuations are largely dependent on the driver design rather than on the enclosure.

Fortunately, no life-threatening modifications of the treble box were needed, which was not true in the case of the crossover. I painted the treble box to match the repainted bass box.

PHOTO 1: The revised left channel complete with absorbers. The treble unit is placed vertically and is flanked by 1.75″-thick blocks with 45° ends.

CROSSOVER CHANGES

The new speaker stand with an internal crossover is shown in the mechanical drawing of *Fig. 2*. I located the capacitors and resistors on a PC board I attached to the inside top of the stand (*Photo 2*). I placed the inductors and Lpad on the inner side walls of the stand. The circuit schematic of the speaker system is shown in *Fig. 3*, and the parts list is in *Table 1*. The speaker stand brings the treble unit to the correct height for listening when seated.

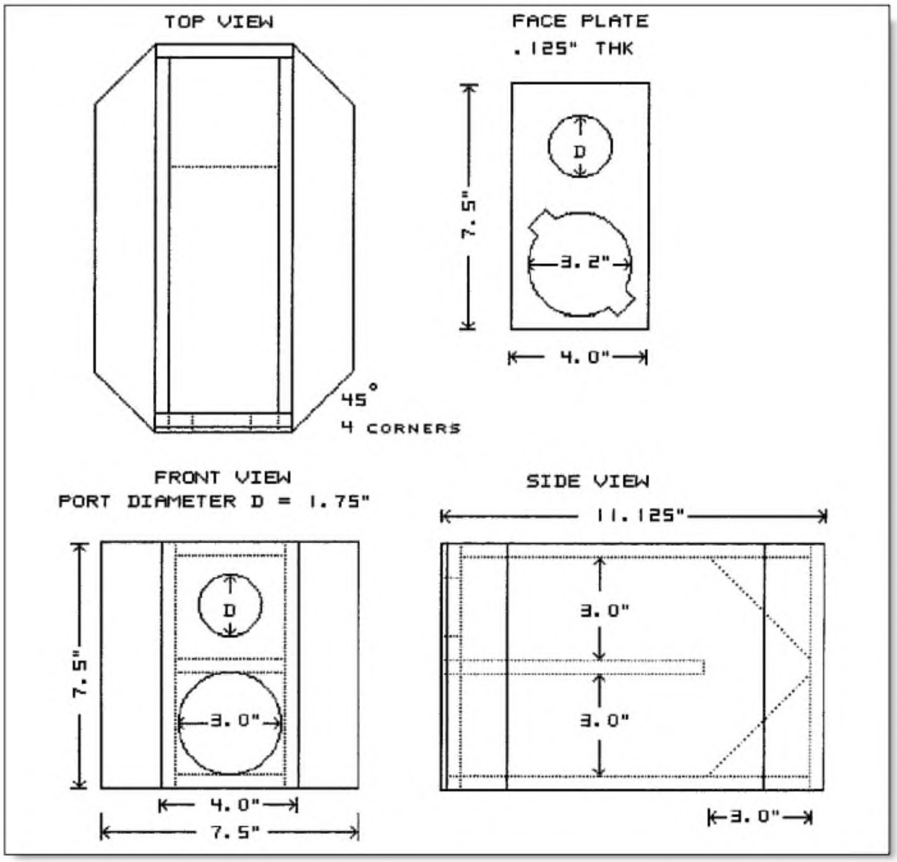


FIGURE 1: Modified treble enclosure.

I added the parallel combination L1 and R3 to correct the bass-unit diffraction step. The original high-pass filter had two paralleled 8.2mF capacitors that I replaced with a single 33mF one to adjust the diffraction step for the treble speaker. Note that the Lpad setting must be reduced to complete the correction. Since the new driver is a 4Ω unit, I placed a 4Ω resistor in series with it.

PHOTO 2: Component side of crossover board showing resistors, capacitors, and posts for connections to external components. (The series-connected resistors are a substitute for the specific parts, which were out of stock.)

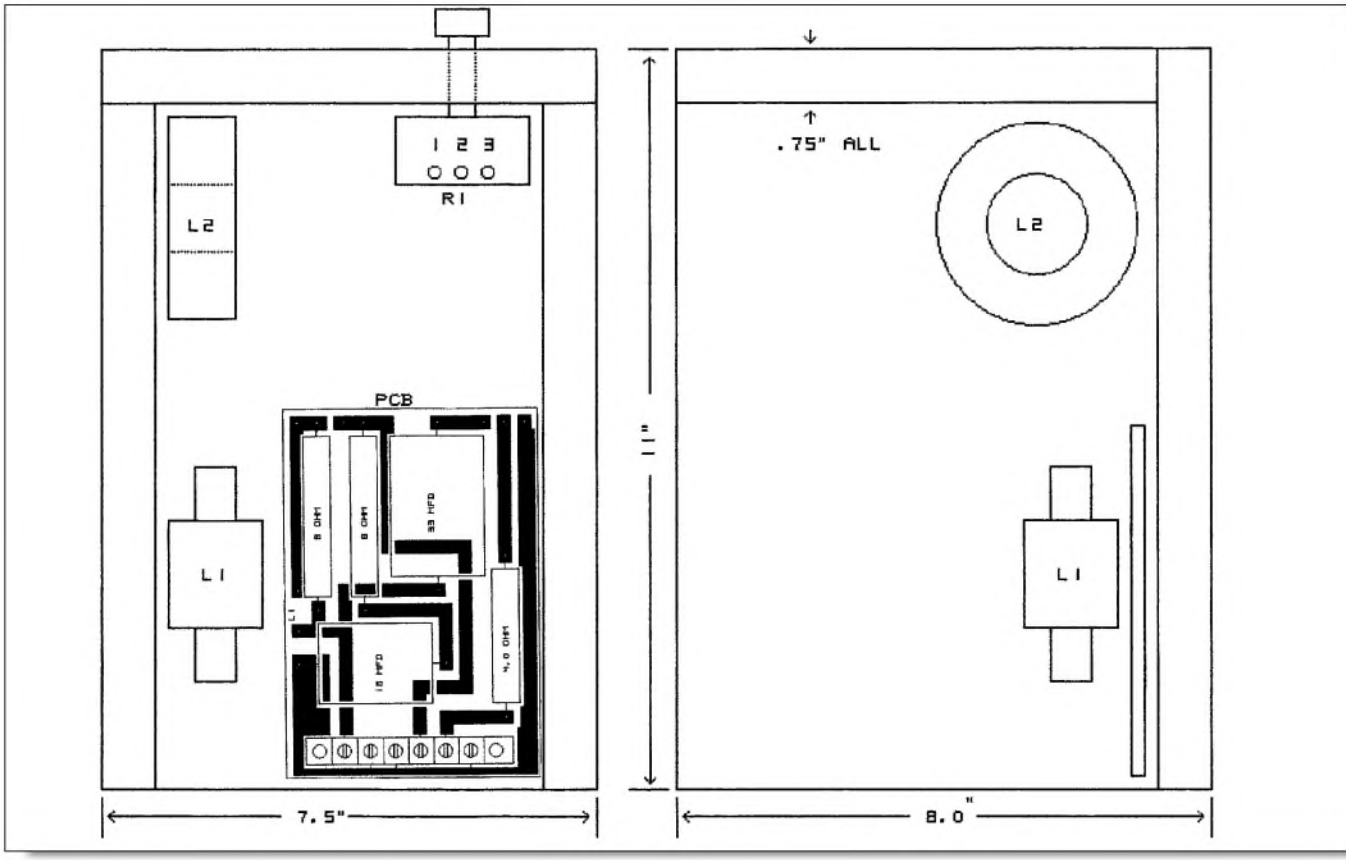


FIGURE 2: Bottom and side views of crossover inside speaker stand. (The stand's right leg is omitted in the side view.)

BASS-UNIT DIFFRACTION STEP

The near-field frequency response for the bass unit is shown in *Fig. 4* for the system with the HM130XO drivers. The top curve shows the near-field response of the bass with a diffraction step taking place between 256 and 512Hz.

At first I used the receiver's tone control to balance the output from the bass unit. However, I soon resolved to correct this external acoustic problem in the crossover rather than in the amplifier. The approach I used was suggested by Ralph Gonzalez and reported by Weems.[3] Gonzalez's solution to increasing bass response for a single speaker with only one voice-coil winding puts an inductor shunted by a resistance equal to the speaker impedance in series with the bass driver.

The values I used were based on the data in *Fig. 4*. I added a 3.0mH inductor, L1, with a 8Ω parallel resistor, R3, in series with the bass speaker. This centers a 6dB downward step over the upward diffraction step with the results shown in the lower curve of *Fig. 4*. Note that when the L1 and R3 are placed in

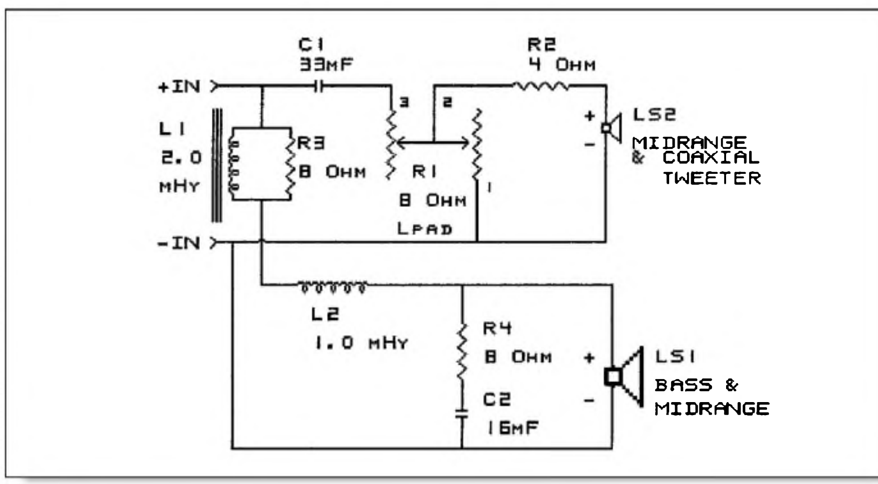


FIGURE 3: Speaker-system electrical schematic.

TABLE 1
SPEAKER SYSTEM ELECTRICAL PARTS LIST

PART	VALUE	DESCRIPTION	SOURCE	PART #
C1	33.0mF	400V polypropylene	Parts Express—Solen	027-586
C2	16.0mF	400V polypropylene	Parts Express—Solen	027-578
R1	8.0Ω	50W Lpad	Parts Express	260-255
R2	4.0Ω	20W wirewound	Parts Express	017-4
R3, R4	8.0Ω	20W noninductive	Radio Shack	271-120
L1	3.0mH	I core .33Ω	Parts Express	266-558
L2	1.0mH	Air core .21Ω	Parts Express	266-350
LS1	8.0Ω	Carbon fiber midbass woofer	Parts Express	296-063(HT 130CO)
LS2	4.0Ω	Midbass & coaxial tweeter	Boston Acoustics	CX3
TB1	6 Term	PC mount barrier strip	Radio Shack	RSU 11673175

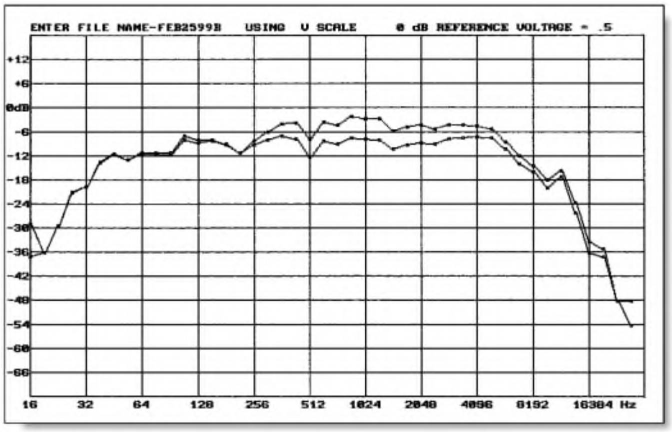


FIGURE 4: Near-field response of bass speaker. Top curve shows diffraction step starting near 250Hz. Bottom curve shows response with RL circuit in series with speaker.

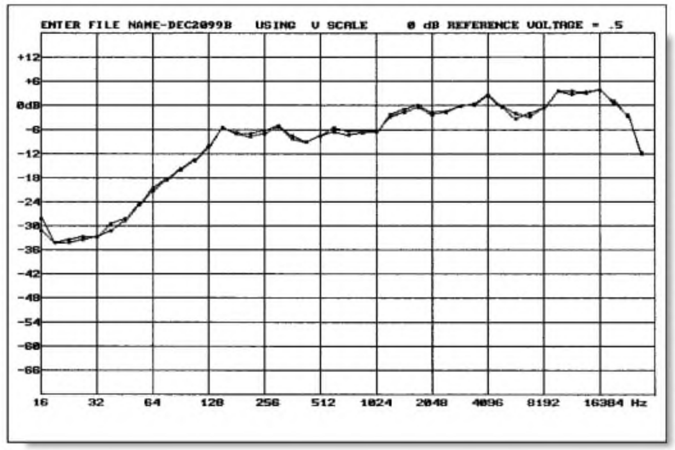


FIGURE 5: Near-field response of treble speaker with and without side pieces.

series with L2, the correct value of L1 becomes 1.8mH.

The 3.0mH Parts Express inductor specified in *Table 1* is satisfactory, but puts the total resistance in series with the speaker a little on the high side, so I tried winding my own inductors. I wound several with iron cores, trying both I and H cores. Each used many large laminations, which I obtained by scavenging two heavy-filter chokes.

Because of the known problems with harmonics, I used an RLC bridge to evaluate the distortion. The I core's unbalanced harmonic residue was better than 60dB down, while the H-core inductor was only about 50dB down. The respective resistances were .4 and .25Ω. I am currently using the I-core inductors. A computer plot of the low-pass CO branch using a diffraction-step model based on Olson's data for a spherical baffle showed the optimum value for L1 to be 1.8mH. The model is quite accurate for a spherical baffle, but will not model the fluctuation at frequencies just above the step in the case of an angular baffle.

TREBLE-UNIT DIFFRACTION-STEP CORRECTION

In the case of the treble unit, I used a different approach because of the high

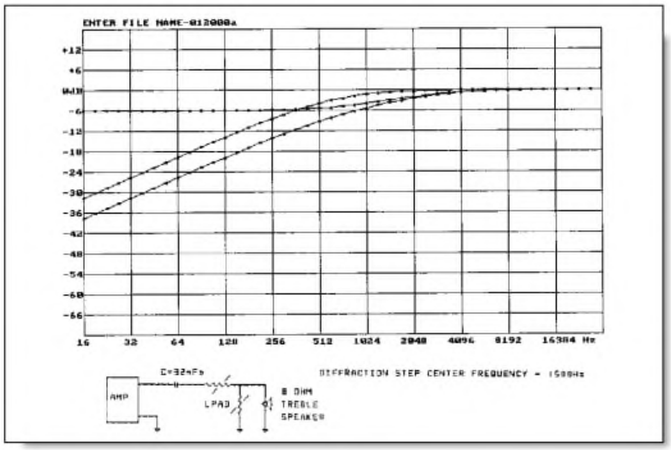


FIGURE 6: High-pass filter diagram and response with diffraction correction.

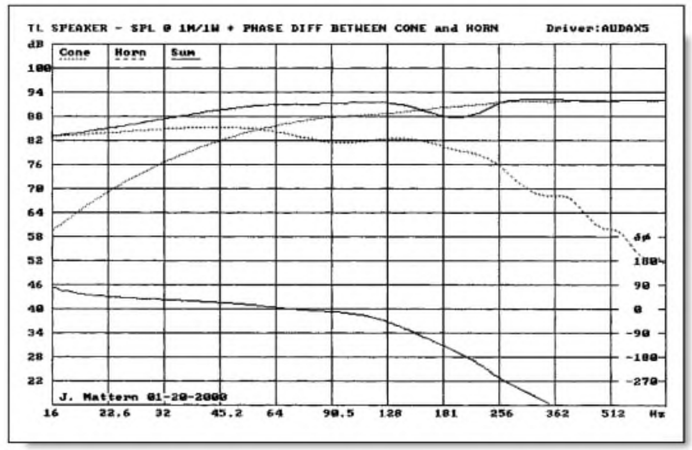


FIGURE 7: Simulation of TL speaker using alternate software.

impedance of the treble branch of the crossover. The approach I chose used the diffraction step as part of the treble-unit rolloff, which meant using the center of the step as the design crossover and locating the LF corner frequency an octave below what it would normally be. The new crossover is shown in the speaker-system diagram (*Fig. 3*). *Figure 5* shows that the diffraction step occurs between 1 and 2kHz, at approximately 1.5kHz. I have used 1.5k as the center frequency in *Fig. 6* to determine the value of capacitor C1.

The diffraction-step response (top curve of *Fig. 6*) was plotted from a model based on Olson's data[4]. The center curve is the filter response of a single-pole high-pass stage taken alone, while the bottom curve is the effective response when the dB values of the two curves are added. The corner frequency of the high-pass filter was adjusted until the 3dB point of the sum curve was at 1.5kHz. Listening tests confirmed that this was a reasonable approach. Of course, that does not deal with the fluctuations that occur above the step frequency. This would require that the treble enclosure be an isolated sphere. The latest treble enclosure is a compromise in that direction.

CHANGES TO THE SIMULATION SOFTWARE

I would like to report that my measured response of the bass unit agreed with the simulated response, but this is not the case. A comparison of my measured response with my simulated response shows a significant discrepancy below 38Hz. A possible cause is the approximation used in modeling the horn taper.

Olson's equations[1] for the cylindrical and conical horns do not provide for losses, so instead I used the electrical transmission-line equation given by Ramo and Whinnery.[2] They give two forms of this equation: an ideal form that does not provide for losses, and a general form that does. Their ideal equation is mathematically identical to Olson's cylindrical horn equation, so I was reasonably safe in using their general equation that includes losses. I accounted for the taper by treating the line as a quarter-wave transformer whose throat-to-port area ratio determined a velocity correction to the untapered line. I believe the result to be valid as long as the line length is a quarter wavelength or longer.

The new simulation software (*Fig. 7*) uses Olson's equation[1] for the throat impedance of a finite conical horn instead of the equation for a finite cylindrical horn. This also is an ideal equation for a horn without losses. Expanding the equation revealed the terms that should be lossy. These I replaced with appropriate hyperbolic trig functions that contain both phase and loss information.

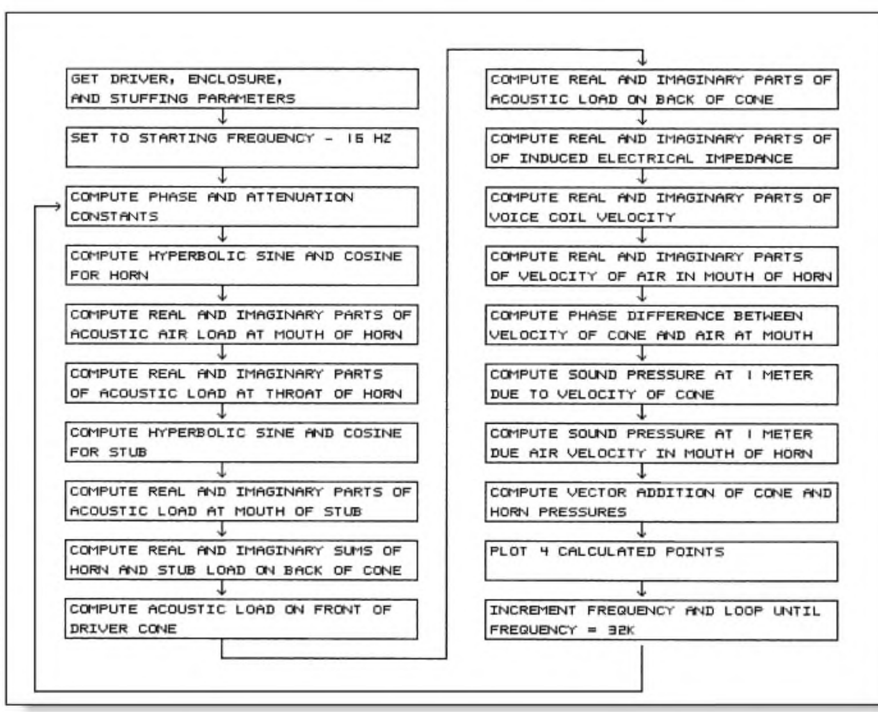


FIGURE 8: Simulation-program flowchart.

When this awful mess was included in the software program, the bottom end dropped off slightly faster than with the previous approximation, but there were no startling differences. In this case, no velocity correction for the area ratios was required. I have included a program flowchart (*Fig. 8*) for the sake of completeness.

CONCLUSION

My simulation calculations are based on an assumption that both the driver cone and the horn's mouth are flush with a large baffle. Neither of these assumptions is fully satisfied in the present design. The assumption is satisfied at the high end of the spectrum, but not at the low end, where the cone and port radiate both forward and backward due to diffraction around the enclosure. Here it causes a 6dB loss. The changes to the crossover deal with the step, but not the fluctuations that accompany it. Those require a rounded enclosure without sharp edges.

I must comment on the discrepancy between the simulated and the measured low-frequency rolloff. The simulation in *Fig. 7* shows a gradual fall-off below 38Hz, while the measured response fell off rapidly below 38Hz. The rapid drop in the real speaker could be caused by air leaks. An obvious case in system #1 is the leakage path between the voice coil and the pole piece. The Audax drivers used in system #2 lacked the phase plug, having instead a cap over the voice-coil former, thus closing off this leakage path. However, system #2 had virtually the same low-end response as system #1, even though there was no leakage path around the voice coil!

Of course, the microphone will contribute to the measured fall-off. I have been using the same Panasonic electret element as does the Mitey Mike for my frequency-response measurements. I have several of these inexpensive elements from Radio Shack, and they all give the same result when used with the Radio Shack digital multimeter and my NEC portable computer. I replaced the .47mF coupling capacitor in the battery pack with a 10mF tantalum capacitor. My mike amplifier is very wide-band.

I would like to point out that it is not unreasonable to achieve high levels of power from the horn. Predicted and measured pressure on the backside of the cone at low frequencies far exceeds the pressure on the front side when the line's cross section is equal to the cone area. I chose this area because Bailey[5] reported that smaller areas were undesirable.

The simulated frequency-response curve of *Fig. 6* shows a dip in the simulated response at 180Hz, which I believe is the result of cancellation by out-of-phase radiation from the port. When the measured attenuation in a straight transmission line is used in the simulation, the predicted dip is larger than the measured one. This suggests that the attenuation in the folded horn is increased at the higher frequencies by its four bends. Accordingly, I have increased the attenuation coefficient in the simulation plot of *Fig. 7* by 3dB at 180Hz to equalize the two dips.

I am surprised by the large change in velocity of sound with frequency caused by movement of the fiberglass tangle. The rapid drop of velocity with frequency appears to hold the phase difference at the port to less than 90° down to very low frequencies. This seems to me an advantage.

Even if you assign all the discrepancy to the simulation model, it is still reasonable to consider the bass-unit design a success. However, I will continue to seek an explanation for the differences. Also, the new crossover is probably close to optimum for the design approach I chose. I am not quite satisfied with the treble-unit enclosure, and will probably try a rounded enclosure next. As it now stands, the sound is balanced, very clean, and natural. The last two adjectives are not mine. These changes to my TL speakers have resolved some but not all of my questions. ❖

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3. Weems D. B., *Great Sound Stereo Speaker Manual*, Tab Books, p. 181, 1990.
4. Olson, H.F., op. cit., p. 20.
5. Bailey A. R., "A Non-resonant Loudspeaker Enclosure Design," *Wireless World*, p. 486, October 1965.

A Phono Pre-Preamplifier for the CD Era

Part 2

In the second part of this preamp project, you'll discover that paying close attention to good power-supply design and construction techniques offers its own rewards. **Norman E. Thagard**

I have favored on-board regulation since my digital design days. However, my amplifier designs have had discrete regulators located on their own separate PC boards. This was partly because I used voltage-doubling techniques that added enough bulk to the doubler/regulator that it was not practical to place this circuitry on the preamplifier's PC board. Here, I returned to my roots with small IC regulators located on the preamp circuit board.

TRANSFORMER SELECTION

The current requirement for each preamp channel is only 20mA/rail with another few required by the monolithic voltage regulator. Thus, almost any transformer with sufficient secondary voltage should work. I had a 48V CT, 150mA transformer in my parts bin. I also purchased from Digi-Key a small 44V CT at 73mA toroidal transformer in case I decided to mount the power supply in the same enclosure as the preamp.

In the end, I decided to take the conservative step of placing the power supply (*Fig. 3*) in its own enclosure with the unregulated DC output supplied to the preamp through a connecting cable. So, I used the old parts-bin transformer. If you wish two true monaural channels, I would recommend using two of the aforementioned toroids and duplicating everything. As for me, I opted for the dual mono configuration where the transformer secondary is the last common component with everything downstream electrically and electronically separate.

A given power transformer can radiate at a level sufficient to induce hum in the preamp.[24] In theory, a toroidal transformer would restrict its field to the torus. Even so, it is probably better to locate the AC power portions of the system away from the signal portions. If that is undesirable or impractical, I recommend a small toroidal power transformer located as far from the preamp boards as possible.

This is more than a theoretical consideration, since the hum induced in this preamp when it was located immediately above the power amplifier's power transformer was intolerably loud. This occurred even though the prototype phono preamplifier is in a ferrous metal (steel) enclosure, and the power amplifier used two toroidal power transformers.

Electrostatic shielding with aluminum will clearly not prevent such hum induction. Simply locating the preamp away from intense alternating magnetic fields is the most cost-effective solution. Moving the preamp just 6″ away from the amplifier reduced the hum below the audible level.

FUSING

For the dual-mono configuration shown herein, the requirement from the power supply should be P_{supply} = 3.4W (two channels, 0.025A/channel, 48V RMS, $\sqrt{2}$ VDC/V RMS). This means a current draw from the power line of 3.4W/117V RMS = 29mA. I used a 63mA fuse for F1 in the prototype because I had one in the parts bin, but a 0.05A rating would be more conservative. I do not recommend using a fuse rated more than 0.1A. If it blows, check for a wiring error or component failure before proceeding.

FILTER DESIGN

With such a small current requirement, it makes sense to try for additional ripple reduction by using two stages of filtering. This takes advantage of very large, but still very compact electrolytic capacitors, where the usual capacitor-

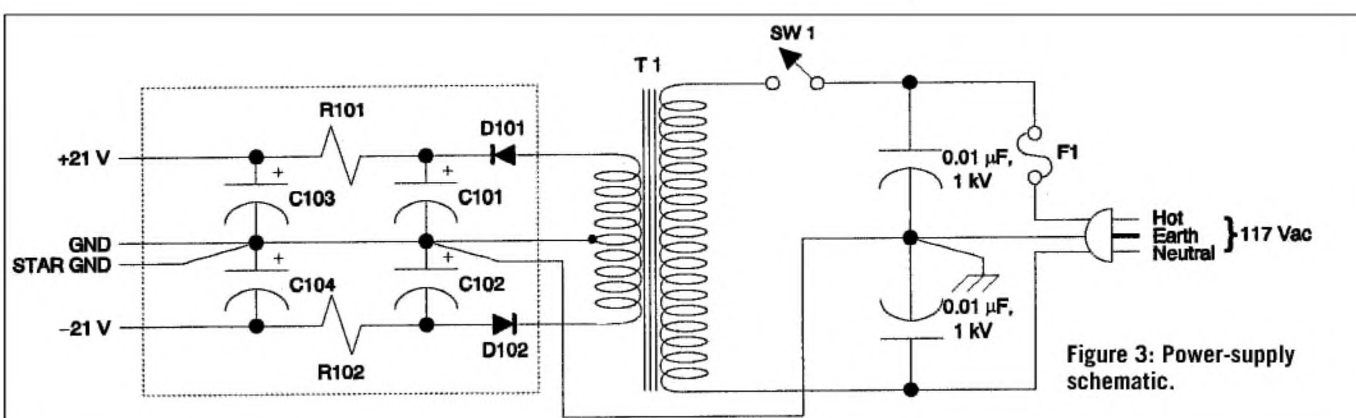

Figure 3: Power-supply schematic.

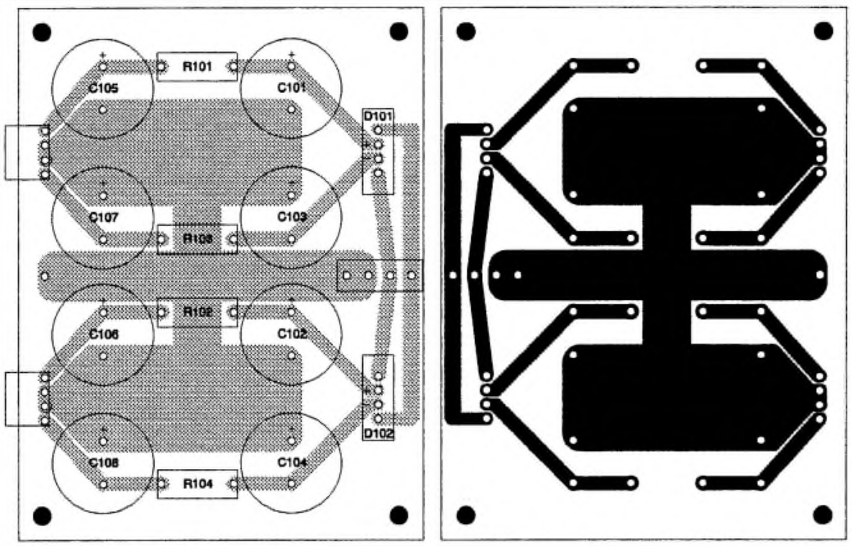

Figure 4: Power-supply board parts placement (left) and PCB pattern (right).

input power-supply filter is followed by an RC low-pass filter.

My criterion for the capacitor values was the size of the capacitors in my parts bin. I had on hand four 2,200μF, 50V and four 4,700μF, 35V capacitors that were perfect for the same kind of Radio Shack prototyping boards that had served for the preamplifier channels. The former units are positioned as conventional capacitor-input filters, with one each in the positive and negative power supplies for each channel. Four 220Ω resistors connect the four 2,200μF capacitors to the four 4,700μF capacitors.

In this way, each positive and negative power supply of each channel follows the raw DC output across the 2,200μF capacitors with an RC filter of 1.034s time constant. This corresponds to a filter cutoff frequency of $(2\pi RC)^{-1}$ = 0.154Hz. Capacitors of 470μF would be perfectly acceptable.

The total amount of ripple attenuation provided by the filter circuitry depends upon whether you use half- or full-wave rectification. I know that this is almost always full-wave in high-end equipment. However, the amount of ripple attenuation is so high that either type is acceptable in this application, and it may be that the 60Hz hum from residual ripple in a half-wave supply is less objectionable than any 120Hz hum from full-wave rectification.[25]

The idea is, of course, that no audible hum be present no matter which rectification scheme you use. There is no audible hum produced by residual ripple in this preamp with the power supply described here, a supply that uses half-wave rectifiers.

Since ripple is attenuated by a factor of ten (20dB) for each decade that the ripple frequency lies above the filter cutoff frequency, the second stage of filtering here provides 20dB/decade × 2.6 decades = 52dB of additional reduction where:

$$60\text{Hz}/0.154\text{Hz} = 390$$

number of decades = x, where $10^x = 390$;

$$\log 10^x = x = \log 390 \cong 2.6$$

I based the selection of 220Ω as the value in this application on a good comfort level that the unregulated, but heavily filtered output voltage delivered to the preamplifier circuit board would be no less than about 18V under worst-case conditions. The reason for that criterion will become evident. The above relationships show that if you select the full-wave rectifier, you will obtain 6dB more ripple attenuation from the RC filter.

RIPPLE BLIPS

The ripple on an oscilloscope at 2mV/division is barely visible, appearing almost as a pulse train of small "blips." The blips are probably due to the heavy current flow during the short period in which charging current flows through the rectifier diodes to the 2,200μF capacitors of the capacitor-input filter. A small resistor between the diodes and the capacitors would reduce the amplitude of the blips, but there is already a heavy overkill situation here.

As a matter of interest, if ripple were to be reduced to 10mV with a capacitor input filter alone, the charging current through the rectifier diode would be about 5A, even though load current is a mere 25mA. Charge is the product of current and time. The charge removed from the filter capacitor during the 16.7ms cycle time T (T = 8.3ms in a full-wave rectified supply) must be restored to the capacitor during the short (in a well-designed supply) recharge period, Δt.

If recharge (diode) current is considered constant (it is not, but the approximate answer so obtained is sufficient), then $I_{LOAD} \times T = I_{RECHARGE} \times \Delta t$. Solving this expression for charging current yields $I_{RECHARGE} = (T/\Delta t) \times I_{LOAD}$. This formula already suggests that charging current will be much greater than load current because the small charging interval Δt is so short in comparison to the relatively long cycle period T.[26]

Although diode conduction actually continues past the peak capacitor voltage V_{peak}, if you assume that conduction begins at time Δt before it ceases at V_{peak}, then voltage magnitude at the onset of conduction is $V_{peak}\cos\omega\Delta t$. The quantity $\omega\Delta t$ is the diode conduction angle, and for a half-wave rectifier the ripple frequency is $\omega = 377\ s^{-1}$. The peak-to-peak ripple voltage is therefore $V_{ripple} = V_{peak} - V_{peak}\cos\omega\Delta t$.

Remember that $\omega\Delta t$ is intentionally made small to reduce ripple so that the trigonometric approximation $\cos\omega\Delta t \cong 1 - ½(\omega\Delta t)^2$ is valid. If the output voltage of the supply is to be about 21V, then you can solve for $\Delta t \cong [2V_{ripple}/V_{peak}]^{½}/\omega = [2(0.01)/21]^{½}/377 \cong 82\mu s$. From this, it follows immediately that $I_{RECHARGE} \cong (0.0167/0.000082)(.025A) = 5.09A$, a pretty impressive number.

I_{LOAD} is constant at about 25mA, and the capacitor discharges for almost the entire 16.7ms period T. The formula for the discharge of a capacitor at a constant current leads to $V_{ripple} \cong I_{LOAD}T/C$. For the 2,200μF filter capacitors, then, $V_{ripple} \cong (0.025)(0.0167)/(0.0022) \cong 0.189V$.

Surge current through the diode in the preamp power supply is on the order of 1.2A, which is still a pretty startling number, but the surge ratings on even small rectifier diodes are well in excess of this. For example, a 1N4002 diode has a 30A surge rating, but is sold as a 1A rectifier. It should be clear why surge ratings need to be so much higher than the average DC load-current rating.

A capacitor of almost 42,000µF would be required to reduce the ripple to 10mV. Using the additional RC filter reduces the almost 200mV ripple to less than ½mV with far smaller capacitors. It is enlightening, even eye opening, to look at things like this. Many designers treat power supply design very casually. I used to, too, until it bit me.

VOLTAGE REGULATORS

Unregulated output voltage from the rectifier-filter will vary, depending upon the power transformer secondary voltage under load. Measured voltage was 24V at a current draw of 25mA with the particular transformer that I used. Once you've selected the actual transformer, you could simply increase the resistor value in the RC filter or, better, add yet another stage of RC filter so that approximately 15V would be presented to the preamplifier under load. In either case, it would require no voltage regulator.

While you are certainly free to use other schemes, the one chosen here was to use 78L15 and 79L15 IC voltage regulators on the preamplifier circuit boards to supply the needed ±15V power rails. These voltages were limited to a 15V magnitude because of the 30V drain-to-source voltage limitation of the 2SJ109 JFETs. Some features of on-card regulation are discussed in National Semiconductor's *Voltage Regulator Handbook.*[27] This publication is a good general reference for power-supply design.

As always, there are constraints you must consider. If the unregulated voltage is too high, the voltage and/or power-dissipation limits of the regulators will be exceeded. If this voltage is too low, then the regulator will "drop out," that is, it will cease to regulate because some active device within it is no longer in its active operating region.

Although these limitations are slightly different for the 78L15 and 79L15 regulators, the worst-case limits require an unregulated voltage magnitude in the range $17.5V \leq V_{in\text{-}unreg} \leq 28V$. These regulators are widely available from several manufacturers, including the replacement series manufacturers such as ECG, NTE, and RCA.

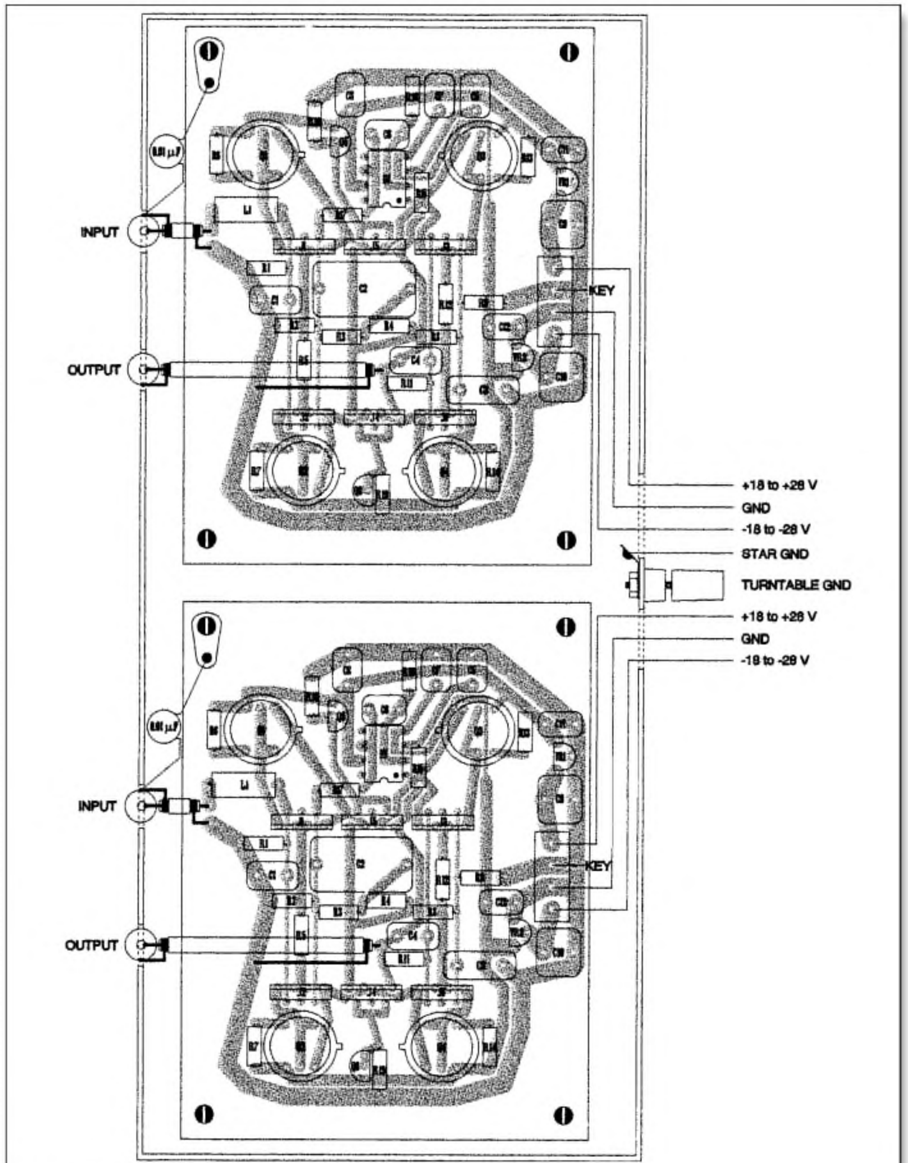

Figure 5. Preamp-enclosure construction pictorial.

Therefore, since parts numbers will vary, be careful to select regulators whose output-voltage tolerances are guaranteed to be within ±5%. There are even tighter-tolerance devices available if you wish to go to the trouble and expense of finding them, but it really isn't necessary to do so.

Each regulator will, for stability, require the input capacitor shown, unless you place the regulator immediately at the output of the unregulated supply. With the recommended value for the 79L15's input capacitor, there was an intermittent, low-amplitude, high-frequency oscillation. This I tamed by increasing the capacitor value to 1µF. For consistency, I also used 1µF input capacitors for the 78L15 regulators.

A case could be made for deriving first-stage supplies from the output of the 78L15/79L15 regulators by using yet another RC filter, or perhaps a zener. This would mean that the first stage would operate at some lower voltage. This should be no problem, however, given that the output swing demanded from the first stage is significantly less than that required from the second.

CONSTRUCTION

The layout and construction details are reasonably well described in *Figs. 5*

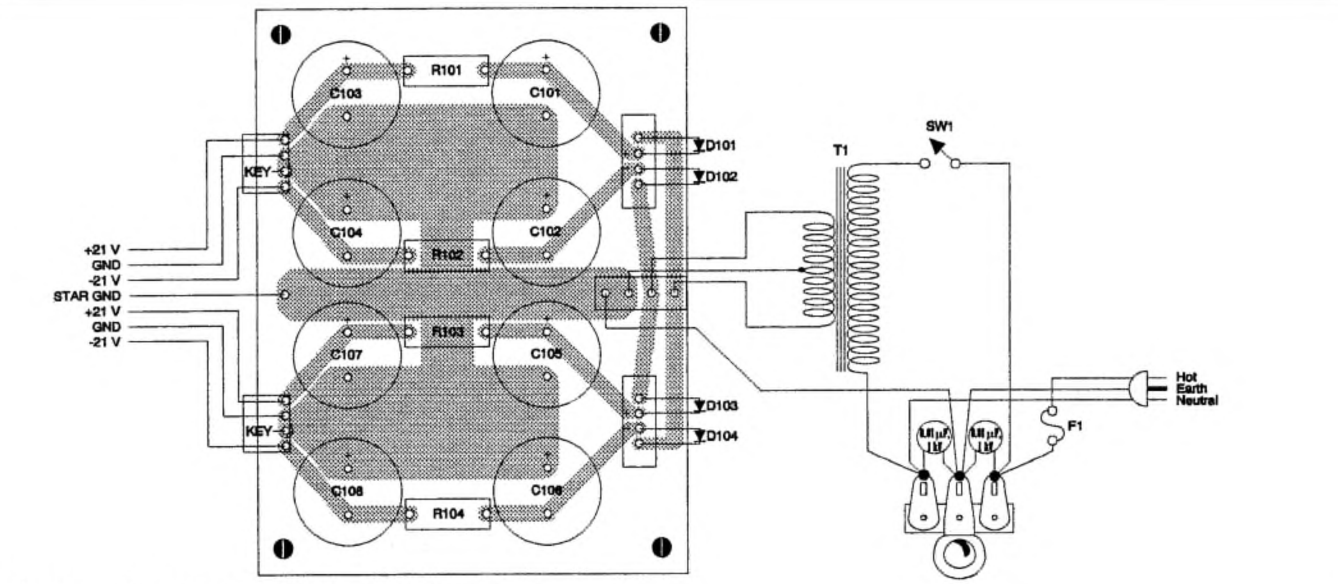

Figure 6: Power-supply-enclosure construction pictorial.

and *6*. Nonetheless, I should elaborate on a few of the details.

Pay attention to the connection of the three-wire AC power cord. There is potential danger if it is not wired as shown. For the two-phase, 117V AC house line, the color convention for wiring, including the AC line cord, is green = ground, white = neutral, and black = hot.[28]

As a designer, I always think about the possibility of modification or repair of the device, especially for the prototype version. For that reason, I liberally employ quick-disconnects (QDs) as well as transistor and IC sockets. The JFETs are not socketed here because of their unusual pinout. The power cable between power supply and preamplifier enclosure has QDs at both ends. If you use QDs on this cable, be sure that it has male connectors at the power-supply end and female connectors at the preamp end. Thus, even if you use Molex header-type connectors, the "hot" connector pins are not exposed. I have serious reservations about claims of sonic effects of QD metals. Even so, I favor gold connectors because of past experience with other types' reliability problems due to corrosion.

I mounted chassis-mount RCA jacks to the edge of the PC board, and used uninsulated wire to both strap and ground the sleeve (outer portion) of each jack to the board, with the mating portion of the jacks extending out beyond the board's edge. This arrangement allows unimpeded connection to the jack of the RCA plug from the tonearm. I soldered a 0.01µF disc ceramic capacitor to the ground trace immediately adjacent to the ground trace connection of the input jack, and soldered the other end of the capacitor to a solder lug, which, in turn, I grounded to the chassis via the screw attaching the PC board to the nearby metal standoff.

A standoff under each corner of the board provides its secure mounting to the chassis while ensuring that the bottom of the board does not contact the chassis. All four standoffs should obviously be of the same height. You can make them either of metal or insulating material, except that, as described earlier, the one to which the solder lug is attached must be made of conducting material.

I made the connection between the inner-conductor (center) connector lug of the input jack and the board trace to inductor L1 with a very short piece of uninsulated wire, since the jack was mounted immediately adjacent to L1. I similarly strapped and grounded the output RCA jack at the board's edge, but made the connection between its inner-conductor connector lug and the appropriate board trace with small-gauge shielded cable because of the distance involved. The shield of this cable is grounded only at the phono-jack end.

With this mounting arrangement of the jacks, it was necessary to drill holes only in the front panel of the preamp enclosure, allowing the RCA jacks to protrude almost completely outside the enclosure. To do this, the front edge of the PC board must contact the inner side of the front panel of the enclosure. The front-panel holes for the jacks must be sufficiently large that neither the jack nor the shield connector portion of any connected RCA plug contacts the front panel. Also be sure that no other PC board component or trace contacts the front panel. This should not be a problem with the PC-board pattern provided.

You could achieve the same effect more easily by using PC-mount RCA jacks, but I had none in my parts bin, and so devised the scheme just described. If you use PC-mount jacks, you must modify the board's trace pattern accordingly.

GROUNDING

Some attention to the grounding scheme is required. For one thing, there are differences among various turntables, tonearms, and even cartridges in the way grounding is handled. For that reason, the actual interconnections may be different, depending upon the builder's specific system and its components. It will probably be true that most builders will be very familiar with the schemes that work best in their systems.

I emphasize again that the RCA input and output connectors are insulated from the enclosure. It is important that the input connectors be grounded to chassis only through the 10n capaci-

TABLE 2: PARTS LIST, POWER SUPPLY

RESISTORS	
R101, R102	220Ω, 2W, 2% metal
CAPACITORS	
C101, C102	2,200µF, 50V aluminum electrolytic
C103, C104	4,700µF, 35V aluminum electrolytic
TRANSFORMER	
T1	48Vct at 150mA (see text)
FUSE	
F1	0.05A (see text)

tors as shown. While I understand that isolating the input connector ground from the chassis avoids a ground loop, I have never seen a reason given for the capacitors. I assume they are for RF bypass, given the small capacitance value.

These capacitors may be disc-ceramic types. Voltage rating is probably not critical, but since 10n, 1kV disc-ceramic capacitors are used as AC-line filters for the power supply, it is convenient to make all of these capacitors the same and perhaps save on quantity purchases.

A ground loop can occur anyway if the cartridge or tonearm cable connects both channel grounds at the turntable end. If a continuity check indicates this situation, then break the potential loop by opening the shield connection at the turntable end for one channel only.[29] If that is not possible, you can try omitting one shield connection to the PC board at the preamplifier end for one channel only.

I am not sure whether this, of itself, will give satisfactory results, because signal ground for that channel then comes circuitously from the other channel to the power supply and back to the first channel. I have seen amplifiers oscillate with such circuitous ground paths. As an alternative, after breaking one shield connection for the PC-board RCA input jack, connect the ground trace of both channels' PC boards together and break the ground return to the power supply for one channel only.

Some turntable/tonearm combinations offer an optional ground lead for connection to the preamplifier chassis. Accommodation of this option is the reason for the binding post labeled "TURNTABLE GND" on *Fig. 5*. The decision whether or not to connect this optional ground lead is based empirically on the situation that results in the least hum.

For my system, the SME 3009 tonearm cable provides this optional ground lead, which, at the tonearm end, is connected to the turntable platform of a Thorens TD 125. Connection of the optional ground to the preamplifier binding post is the preferred configuration in my case, since this reduces hum below the audible level.

At the power supply, only one chassis interconnection point should exist. This is nicely illustrated in *Fig. 6*, which also indicates what is meant by such terms as "STAR GND."

LAYOUT

Some of the layout considerations have already been mentioned: for example, the importance of keeping the power transformer and other possible sources of hum induction as far away from the preamp boards as possible. Otherwise, the layout is not very critical.

If you use my PC-board patterns, much of the layout is predetermined. The power-supply rectifier diode patterns marked D101 and D102 are for the particular monolithic full-wave diode package used. The pattern will accommodate discrete diodes of the 1N4002 type. Just connect the anode of one diode to the point marked "–" and the cathode of the second diode to the point marked "+." You should place the other ends of the two diodes in the adjacent unmarked AC input points. This is shown in *Fig. 6*. For the placement of any other components not specified in the parts list (*Table 2*), my assumption is that if you are smart enough to select an appropriate substitute, you are smart enough to determine its proper placement.

SETUP AND ADJUSTMENT

No adjustments are required. It is a good idea, as mentioned in the text, to verify the differential amplifier currents by measuring the voltage across current source resistors R5 and R12. Unless you used severely mismatched BJTs in the current mirrors, cascode current should be OK.

I always check the power-supply voltages before I connect the rails to the circuit. I also use current-limited bench power supplies for breadboard work, and a Variac® for initial power up and checkout of the prototype version of a new design or after repairs. The presence of the 220Ω resistors in the power supply affords short-term protection against short circuits downstream, but in the long term, their resulting power dissipation would exceed their power rating. The IC voltage regulators are internally protected against short circuits.

CONCLUSION

I think that this pretty well covers the whys and wherefores of this project. It is a satisfying approach from the standpoint of precision in a relatively simple discrete design. The THD was only about 0.006% up to several kHz, rising to 0.026% at 20kHz with a 0.5V RMS output.

Obviously, it is possible to achieve significantly lower levels of THD with higher open-loop gain and therefore more feedback. It is difficult to do so, however, with the precise active-passive equalization scheme realized through just two one-stage op amps as I did here. The proponents of "less is better," and especially those skeptical of the benefits of negative feedback, will appreciate this trade-off.

As for listening attributes, the sound is open and dynamic. You hear no stridency, even on massed strings. The bass is awesome, and I have no explanation for this. I do not know about the Adcom or Marantz preamps, but otherwise it is true that there are no coupling capacitors in the signal path, even at the input of the power amplifier currently in my system. This was possible because the servo limited DC offset at around 600µV. Still, there should be no perceptible difference in bass, even with coupling capacitors, as long as the low-frequency cutoff is well below the lower limit of hearing.

It is simply amazing how good many recordings were in the days of vinyl. The ambience in the few Mercury Living Presence records that I have is remarkable. I am skeptical of many "golden ear" claims of magical qualities of amplifiers and preamplifiers and I attribute none to this design. I simply assert that this phono preamp is good enough to accurately reproduce the information that is in the recording medium.

I believe you will thoroughly enjoy its

use, provided that other components in the stereo system are equally good. Happily, I continue to be impressed with the sound even after several weeks. The more usual case is to be very impressed initially, with the enthusiasm fading after the first few listenings.

I am listening at higher volume levels than before. This may be a sign of lower apparent distortion, since there is a tendency to adjust volume to a level just below that at which distortion begins to be objectionable. Finally, it's impossible to overstate the enjoyment derived from "rolling your own." ❖

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Testing the B&W 602 S2 Loudspeaker

Reviewed by Joseph D'Appolito

I ran a series of impedance, frequency response, and distortion tests on the B&W 602 S2 loudspeaker. *Figure 1* is a plot of system impedance magnitude. At low frequencies the plot displays the double-peaked curve of a vented system.

The impedance minimum of 5.55Ω at 42Hz indicates the vented-box tuning frequency. There is a second local impedance minimum of 5.1Ω at 162Hz. Impedance phase lies between +47° and −55° over the full audio range. Fortunately, these rather large phase angles occur at relatively high impedance values. With minima in the range of 5Ω, B&W's 8Ω rating for this system seems a bit high.

FREQUENCY RESPONSE

Figure 2 shows the 602's full-range frequency response, which I obtained as a combination of the far-field quasi-anechoic response and properly summed near-field woofer and port responses. I placed the microphone along the tweeter centerline at a distance of 1.25m to produce the far-field response. I then spliced together the near- and far-field responses at 200Hz to produce the full-range response[1].

The response shown in *Fig. 2* has been normalized to 1m to obtain system sensitivity. Sensitivity averages 87.9dB in the two octaves between 500Hz and 2kHz. This is about 2dB less than the figure quoted in B&W's specs.

Relative to this level, the low-frequency −3dB point is 48Hz. Response shelves up above 7kHz by 3dB. There is also a broad response peak of about 2dB centered on 100Hz and a sharp response dip of 3–5dB centered on 6kHz. Finally, an ultrasonic peak of 9dB at 24.8kHz (not shown) is due to the tweeter's "oil can" resonance.

The 602 has two pairs of binding posts for bi-wiring. This allowed me to measure the response of the individual drivers. The result is plotted in *Fig. 3*. The crossover frequency is seen to be 3500Hz.

Notice that the tweeter response is quite smooth. The woofer that is out of phase with the tweeter at this point causes the sharp system response dip at 6kHz. B&W claims that the crossover is fourth-order, but it is clear that woofer response is not falling off anywhere near that fast in the first octave above crossover. I suspect that the woofer response peaks in this frequency region.

CUMULATIVE SPECTRAL DECAY

The 602's cumulative spectral decay (CSD) response (*Fig. 4*) shows the frequency content of the system response following a sharp impulsive input at time zero. On the CSD plot, frequency increases from left to right and time moves forward from the rear. Each slice represents a 0.05ms increment of time. The total vertical scale covers a dynamic 32dB range.

Ideally the response should decay to zero instantaneously. Inertia and stored energy that take a finite amount of time to die away, however, characterize real loudspeakers. A prominent ridge parallel to the time axis would indicate the presence of a strong system resonance.

The first time slice in *Fig. 4* (0.00ms) represents the system frequency response. Tweeter decay time is good, but there is a strong woofer resonance mode at 7kHz that rises out of the background beginning at about 0.44ms. This ridge indicates the presence of a woofer response peak, which in turn, prevents the woofer from rolling off rapidly just above the crossover frequency and leads to the response dip.

WOOFER/TWEETER TIMING

The sharp first positive peak in the 602's step response (*Fig. 5*) indicates the tweeter arrival at the test mike. The second (more slowly rising) peak indicates the woofer arrival. This plot tells us that tweeter and woofer are wired in phase, but it also shows that the 602 is not time-coherent. The excess group-delay plot (not shown) reveals that the woofer arrives at the listening position 200μs later than the tweeter.

POLAR RESPONSE

Figure 6 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

ABOUT THE AUTHOR

Joseph D'Appolito, regular contributor to *SB* and author of many papers on loudspeaker system design, holds four degrees in electrical and systems engineering, including a Ph.D. Previously, he developed acoustic propagation models and advanced sonar signal processing techniques at an analytical services company. He now runs his own consulting firm specializing in audio, acoustics, and loudspeaker system design. A long time audio enthusiast, he now designs loudspeaker systems for several small companies in the US and Europe.

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.

With the exception of the off-axis peak at 6kHz, horizontal response is quite uniform out to ±30°. The on-axis response dip at 6kHz tends to disappear as you move off-axis. This causes an off-axis peak in the polar response curves that are plotted relative to the on-axis response. An alternate view of horizontal polar response (*Fig. 7*) plots responses on-axis and at 10° and 20° off-axis. Note the smaller response dip at 6kHz and reduced upward shelving above 7kHz at the off-axis angles.

The average response over a 60° horizontal angle (±30°) in the forward direction is shown in *Fig. 8*. The dip at 6kHz is greatly reduced. Shelving of the average forward response above 7kHz is also reduced somewhat relative to the on-axis response. This may make the audible effect of the shelving less apparent, since the human ear integrates direct and reflected sound when judging the overall spectral balance of a loudspeaker.

On balance, however, the frequency range between 500Hz and 6kHz is depressed relative to the response at the frequency extremes. This might make the 602 sound somewhat recessed.

Figure 9 is a waterfall plot of vertical polar response, with responses shown in 5° increments from 20° below (−20°) the tweeter axis to 20° above it. Response changes very rapidly as you move off-axis. Here again, off-axis response is smoother. *Figure 10* shows response on-axis and at 5° and 10° below horizontal; the −5 and −10° are much smoother than the on-axis response.

HARMONIC DISTORTION

I ran harmonic-distortion tests at an average level of 90dB SPL. Ideally, harmonic-distortion tests should be run in an anechoic environment. In practice, it is important to minimize reflections at the microphone during these tests. Out-of-phase reflections can produce false readings by reducing the level of the fundamental while boosting the amplitude of a harmonic. In order to reduce the impact of reflections, I placed the microphone at 0.5m from the loudspeaker. Second-harmonic distortion was below 1% over most of the audible frequency range. Below 100Hz second-harmonic distortion did rise to 1.3%, but this is still a very low figure. Third-harmonic distortion was 0.7% or less. This is an excellent result.

A note on testing: The B&W 602 S2 was tested in the laboratories of Audio and Acoustics, Ltd. Measurements were made with the MLSSA and CLIO PC-based acoustic data-acquisition and analysis systems using an ACO 7012 ½″ laboratory-grade condenser microphone with a custom-designed wideband, low-noise preamp. The polar response tests were performed with a computer-controlled OUTLINE turntable on loan from the Old Colony Division of Audio Amateur Corporation.

DM™ 602 S2

Technical Features	Nautilus™ tweeter tube Flat ring tweeter suspension Woven Kevlar® brand fiber cone Bullet dust cap
Description	2-way 4th-order vented-box system
Drive Units	1 × 180mm (7 in) woven Kevlar® cone bass/mid 1 × 26mm (1 in)metal dome high-frequency
Frequency range	−6dB at 43Hz and 30kHz
Frequency response	52Hz – 20kHz ±3dB on reference axis
Dispersion	Within 2dB of response on reference axis Horizontal: over 40° arc Vertical: over 10° arc
Sensitivity	90dB spl (2.83V, 1m)
Harmonic Distortion	2nd and 3rd harmonics <1% 60Hz – 20kHz (90dB spl, 1m)
Nominal Impedance	8Ω (minimum 4.3Ω)
Crossover Frequency	4kHz
Power handling	25W – 120W continuous into 8Ω on unclipped program
Max recommended cable impedance	0.1Ω
Dimensions	Height: 490mm (19.3 in) Width: 236mm (9.3 in) Depth: 306mm (12 in)
Net Weight	9.8kg (21.6 lb)
Finishes	Cabinet: Cherry or Black Ash Vinyl Grille: Black Cloth

B&W Loudspeakers, 54 Concord St., North Reading, MA 01864, (978) 664-2870, FAX (978) 664-4109, e-mail: marketing@bwaudio.com, website:www. BWSPEAKERS.COM

INTERMODULATION DISTORTION

Next I measured intermodulation distortion. In this test two frequencies are input to the speaker. Intermodulation distortion produces output frequencies that are not harmonically related to the input. These frequencies are much more audible and annoying than harmonic distortion.

Let the symbols f_1 and f_2 represent the two frequencies used in

(to page 48)

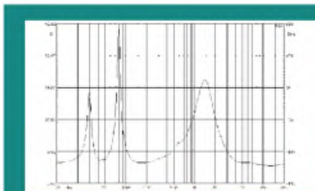

FIGURE 1: B&W 602 impedance magnitude.

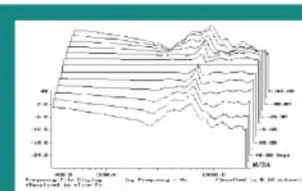

FIGURE 6: Horizontal polar response.

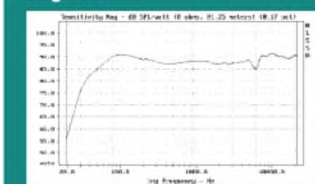

FIGURE 2: System frequency response.

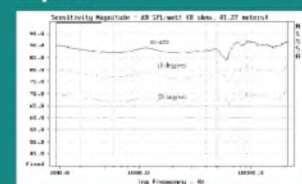

FIGURE 7: B&W 602 at 0, 10, and 20° off-axis.

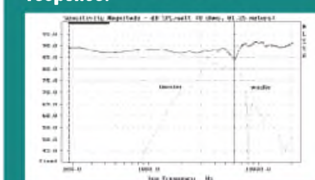

FIGURE 3: System and driver responses.

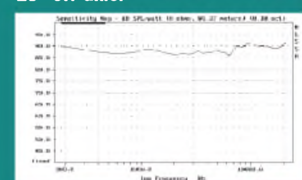

FIGURE 8: B&W average horizontal response over 60° window.

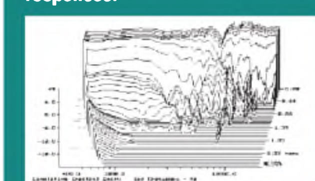

FIGURE 4: Cumulative spectral decay.

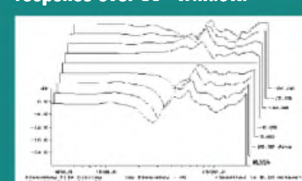

FIGURE 9: Vertical polar response.

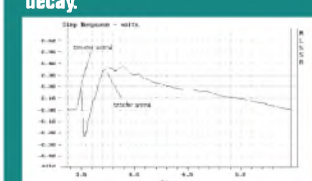

FIGURE 5: Step response.

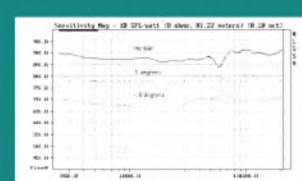

FIGURE 10: Vertical response at 0, −5, and −10°.

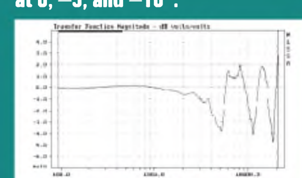

FIGURE 11: Effect of grille on the 602 frequency response.

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Reviewed by Tom Perazella

I must have a thing with word associations. When someone mentions the word Nautilus, I first think of a submarine or a speaker, not the chambered sea animal from which they both derive their names. The sea animal has certainly been around longer, but alas, it has not had the marketing efforts of the former, possibly explaining the difference in recognition.

The B&W Nautilus speaker, however, was a real attention-getter when introduced. The radical multiple-tapered chamber design not only looked striking, but it also had significant sonic benefits. B&W has taken the knowledge gained in producing the Nautilus and transferred it to a more affordable line of speakers. The DM602 S2 is one of a family of speakers derived from the Nautilus that boasts a friendly combination of small space and low price.

Much more conventional in looks than the Nautilus, the design is a two-way utilizing a 7″ woven Kevlar bass/midrange and a 1″ metal dome tweeter. The tweeter, similar to that in the Nautilus, is loaded by a closed tapered tube to absorb rear radiation. The enclosure is a vented box 19.3″ high × 9.3″ wide × 12″ deep. Weight is specified as 22 lbs. A knuckle-rap test showed that the construction was fairly solid, with some resonance from the back panel and upper area of the side panels.

Connections are made on the rear through two sets of gold-plated binding posts, one for the woofer and one for the tweeter. Gold-plated jumper strips are provided for normal connections with the ability to remove them for bi-wire applications. As is typical of most UK and European products, the accursed positive-to-negative terminal spacing does not match standard ¾″ plugs. A pox on the safety fanatics responsible for trying to save the world from self-immolation by making nonstandard speaker post spacing.

The posts themselves have small plastic plugs to prevent insertion of banana plugs. You can remove these plastic inserts by unscrewing the caps of the posts, and then use standard individual banana plugs. The good news is that the posts have very large holes for wire, making insertion of 12-gauge speaker cables easy.

TEST SETUP

I auditioned the units in my loft that is 21′ × 16′ with vaulted ceiling. The room has only three walls; it is open to a lower family room along the long dimension. I placed the speakers along the 21′ wall facing both the listening position and the opening to the family room.

I positioned them 6′ apart with the right speaker 6′ from the right wall and the left 9′ from the left wall. Both were 5′ from the back wall and angled toward the listening position that was 8′ from each. The stands used raised the tweeter 40.5″ from the floor and positioned the woofer 34″ from the floor. I conducted all of the tests with the grilles removed.

The signal source was a Sony 707ESD CD player feeding a custom pre-amp. I used two power amps. Since B&W specified a range of 25-120W as a suitable drive level, most of the listening was done with an AudioSource Amp Three that I have measured at 168W per channel at clipping into 8Ω and 267W per channel into 4Ω. This amp is very clean and is one of the best values on the market.

For higher power tests where I wished to eliminate the amplifier as a potential limiting factor, I used a Crown Macro Reference amplifier. At 760W/channel into 8Ω and 1160W/channel into 4Ω, it was more than sufficient to handle any levels likely to be encountered by a speaker of this size. I made connections from the amplifier to the speakers with 12′ of 12-gauge stranded zip cord.

SOURCE MATERIALS

To provide consistency for all tests that appear in this publication, the editor has asked all contributors to include one reference CD as part of any audition. That CD, the *Hi-Fi News & Record Review* Test CD III, number HFN020, contains both musical and test passages. This is a tough test disk because, in my opinion, it is recorded a little on the bright side. It is clean, but if you have a forward-sounding speaker, this disk will immediately point out the brightness.

For this review, I concentrated on the musical test passages, except for the garage door track. The other CDs were:

Telarc	CD-80126	Britten—*Young Person's Guide to the Orchestra*
Columbia	CK57424	Tony Bennett—*Steppin' Out*
Columbia	C2K68519	Pink Floyd—*The Wall*
Proprius	PRCD7778	*Jazz at the Pawnshop*
Chesky	JD49	Clark Terry—*Live at the Village Gate*
Mapleshade	06932	Blue Rider Trio—*Harp, Steel & Guts*
CBS	MK37793	Vollenweider—*Behind the Garden*
Tristar Music	WK35862	Kodo—*Ibuki*

LISTENING TESTS

The first series of tests was done with the AudioSource Amp Three.

TRACK 1—La Rejouissance—Handel

Like most of the initial tracks on the *Hi-Fi News* CD, the recording was made outside. It is a good test of imaging and definition, with some fireworks thrown in for dynamic flavor. It is by no means a killer cut, but there is enough energy in some of the fireworks to provide a sample of what you can expect in dynamic range from most commercial music CDs.

From the first few minutes of this first piece, the strongest point of the DM602s was readily apparent. The midrange was very good. As the old saying goes, if you are going to get something right, make it the midrange.

Also apparent were the very clean and detailed highs. For my taste they were just a little too apparent. Never did I get the feeling that at any sane volume level the problem was distortion. We're not talking about fatiguing here. The sound was just a little more forward than I would prefer.

The bass was also quite good compared to many of the small speakers I have auditioned. It didn't come close to shaking anything off the walls, but the fireworks sounded like fireworks, although with a light impact. There seemed to be a slight rise in the upper bass with most of the instruments, but thankfully, the high level of boom that is often present in small speakers trying to make an impact on a prospective buyer was absent. The sins in the bass range were mostly venial rather than mortal, being those of omission, rather than commission.

The location and separation of the instruments was very good, and the decay of the fireworks was very realistic. The oohs and aahs from the audience as the fireworks went off were rather convincing.

TRACK 2—Jerusalem—Parry

This cut has a choral work that is especially revealing of the midrange and treble detail. The definition of the voices was excellent, but again, the voices seemed to be just a tad forward. The sibilants were a little too noticeable, and the balance seemed just a bit light. Otherwise, they provided a very good accounting of the piece.

At this point, I did a stand-up test and noticed quite a bit of change in the tonal balance. You definitely should listen somewhere around the tweeter axis.

TRACK 3—Henry V—Doyle

This piece has two tough tests, a high level of cymbals and a solo male voice. The cymbals at the lead-in sounded bright, but the crowd sounds were good. The voice and voice echos were very clean and detailed. Positioning of the voice was excellent, separated quite nicely from the instruments. On this piece, I did a left-to-right head movement to check for tonal changes. They proved to be minimal.

TRACK 4—Trumpet Concerto in C—Vivaldi

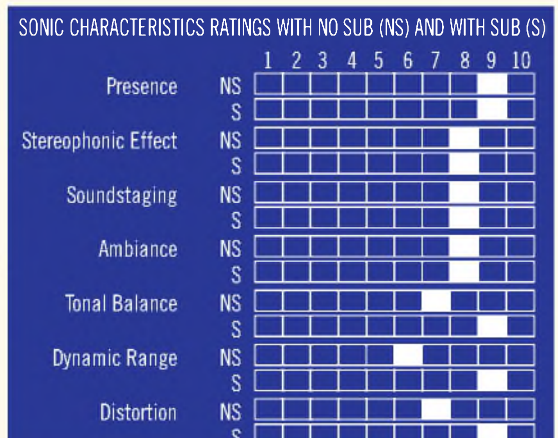
The sound on this cut was very clean with good separation of the instruments, but with just a bit too much bite on the brass.

TRACK 7—Welcome, Welcome—Purcell

The vocals on this track had excellent separation and definition of the different vocal characteristics. The harpsichord was very distinct.

TRACK 14—Rio Napa RSS Demo

This track is a good test of image movement, and there certainly was a lot of movement with these



ABOUT THE AUTHOR

Thomas Perazella is the Director of Information Systems for a national retailer of professional photographic equipment headquartered in the midwest. His prior experience includes work as a Criminalist in the San Diego and Long Beach California Crime Labs and Director of Marketing for a photographic wholesale distributor. In addition to speaker design, Mr. Perazella has designed commercial high-powered electronic flash equipment, as well as numerous pieces of audio electronics for his own use. Other leisure activities include cooking, golf, scuba diving, and motorcycles.

speakers. The bass was just a little loose on this track, but the percussion was otherwise very good.

TRACK 17—Return of the Garage Door
This was my one concession to non-music, but I could not resist the sound of a garage door closing and someone banging on it. One of my friends once said that I would stop to listen to a garbage can crashing down a concrete stairwell. What can I say? Dynamic range and, especially, transient dynamics provide a sense of realism that's hard to beat.

On this cut there was a great sense of depth, but the dynamics came up a little compressed. If you are not a garbage-can junkie, you probably won't notice it.

Britten—*Young Person's Guide to the Orchestra*
I listened to quite a few instruments, so I'll just list them and give you short snippets.

Woodwinds—Piccolo a little bright, but flute good.
Brass—Good midrange, especially the lower mid.
Strings—Good lower mids and mids with highs a little forward.
Percussion—OK at lower levels, but some loss of definition at higher levels.
Clarinet—Very good. Delicate with good air.
Oboe—Good reed sounds.
Bassoon—Smooth with good room sounds.
Violins—A little bright.
Violas—Smooth and silky.
Cellos—Good string sound but a little light.
Double basses—Very good string sounds but a little light in the lower registers.
Harp—Excellent definition but a little light on the lower strings.
Horn—Clean with good definition.
Trumpet—Excellent definition.
Trombone and tuba—Trombone very smooth, but tuba a little light.
Bass drums and cymbals—No weight to drums. Short on definition.
Castanets and gong—Excellent air and shimmer.

Tony Bennett
TRACK 2—Who Cares
The piano, a Bosendorfer, had excellent definition of strings and hammer blows. The lower registers were a little light. Voice was very good. Intonation was very clear, and the gravely texture in Tony's voice was presented clearly without being raw. The dynamics were also very good.

Pink Floyd—*The Wall*
TRACK 3—Another Brick in the Wall
There are a lot of things going on in this album, and it takes a speaker with good resolution to really produce the right mood. Voices were very clear. Guitar definition was excellent. The bass lines were just a little weak.

The best part was that the sound was smooth and did not offend, even at relatively high levels. Separation of the different sounds was very good. Dynamics were quite good for a speaker of this size.

Jazz at the Pawnshop
TRACK 1—Limehouse Blues
I've listened to this piece so many times I feel like I own the club where it was recorded. The room sounds during the intro and at several places during the cut were excellent. This level of micro detail establishes a "you are there" effect.

The brushes on the drums were very clear. The clarinet had good tonal range and was very smooth with just slight bite at times. The vibes were very good with the right combination of attack and reverberation. Stick sounds on the cymbals were very realistic.

Clark Terry—*Live at The Village Gate*
TRACK 8—Hey Mr. Mumbles
Voices and room sounds are the key to this track. The audience sounds were very realistic, including some background noises in the room. Terry's voice was the right level and reminded me of one of his live performances I attended.

Bass was a little fat and not very deep. Sax was very good. The mute had good definition and bite where appropriate. Drums lost a little in the impact department, but again, not bad for a speaker of this size.

Blue Rider Trio
TRACK 8—Stagolee
If you haven't heard any of the Mapleshade recordings, you should. So far I haven't found any Grammy material, but the sense of having Pierre Sprey produce a transparent window to the artists is fascinating. On this piece, the guitar is just great. The string picking and harmonics are exceptional. Intonation on the voice is very clear, and the roughness in Ben Andrews' voice is distinct.

Let me try to explain the term distinct roughness. I think the real test in resolving power of any speaker is to separate sounds such as a rough human voice until you can almost count the different tones making up the voice. If it can do that, you have character instead of offensive noise. These speakers did that in spades, with only a slight prominence of sibilants. Placement of sounds was also excellent, with the ability to almost see the position of Andrews' head as he plays. The sound was totally removed from the speakers.

Vollenweider
TRACK 1—Behind the Gardens
During the early part of this piece, a woman laughs from a position that seems well to the right of the right speaker, roughly 45° to the right of center. This is not quite as far as my reference system, but farther than many speakers I have auditioned. When you first hear it, it can be quite surprising, as you don't expect sound to come from that direction in a two-channel system. There are also some bird songs in this piece, and my dog Cinder, who is a very experienced listener by now, perked up his ears when he heard them.

Kodo—*Ibuki*
TRACK 3—Akabanah
This is a very difficult piece for any speaker to reproduce. Some high levels of low-frequency drum sounds are superimposed on top of some delicate mid- and high-frequency sounds. Want some intermodulation distortion? Just crank up this piece. Two-way systems are particularly susceptible as the woofer extends well into the midrange, in this case 4kHz. At high levels, there was almost no appreciable reproduction of the drum fundamentals, and quite a bit of distortion in the midrange.

TRACK 5—The Hunted
Impact is the name of the game with this track. Although not as deep as Akabanah, the drums in this piece have tremendous sock. In addition, there are again midrange details that should not become trampled in the fray. The results were better than with Akabanah, but the drum impact was moderate.

TESTING WITH THE CROWN
When trying to determine the limitations of a speaker, it's best to make sure that the power amp is not the cause of any problems. To do that, I use a Crown Macro Reference amplifier as mentioned earlier. I repeated the previous tracks until I got a clear understanding what the speaker and prior amp limitations were. There were no big surprises.

TRACK 1—La Rejouissance
With the Crown, the overall tonal balance was the same as with the Amp Three. The sound was still a little on the light side, but the fireworks had a little more impact at very high levels.

TRACK 2—Jerusalem
The results were the same as before.

TRACK 3—Henry V
The results were the same as before.

Britten—*Young Person's Guide to the Orchestra*
On the bass drum track, there was still no deep bass, but the sound was cleaner than before.

Tony Bennett
TRACK 2—Who Cares
The results were the same as before.

Kodo—*Ibuki*
TRACK 3—Akabanah
The very low bass was still absent, but the midrange had not quite as much distortion.

TRACK 5—The Hunted
There was much more impact from the drums than before, but still not as deep as it could be. At very high levels, the speakers surprised me. Even though they were 8′ away from the listening position, on major hits of the drums, the puffs of air from the ports actually hit me in the face and the shirt sleeves. Mind you, this was at levels that were for testing only and not representative of normal listening. However, it was very distracting.

The results showed that at sane listening levels with most musical sources, a competent 150W/channel amp should be a good complement to these speakers. Substitute a super amp costing in excess of $4K, and you'll get some improvement, but hardly worth the expenditure.

FINAL SOLUTION
I'll admit it, I'm biased. I think a good subwoofer is one of the best improvements you can make in any sound system. There are lots of reasons, but one of the most important is that a sub will unload the most difficult tasks from your main speakers.

To see how this philosophy would work with the 602s, I used the high-pass function of my Orban parametric equalizers to act as a 12dB/octave high-pass filter to feed them signals only above 80Hz. Then, I passed all frequencies below 80Hz through a 12dB/octave summed mono low-pass filter to one half of my reference sub. Power for the 602s was again provided by the Amp Three. The results were dramatic. Regardless of how many times I do this, I'm always surprised at the improvement this makes. Here are the results.

TRACK 1—La Rejouissance
Most of this piece sounded the same, but the fireworks had much more impact, especially toward the end where there is a reasonable amount of low-frequency energy.

Britten—*Young Person's Guide to the Orchestra*
On the cut with the bass drum, there was a total transformation. At high levels, the sound at all frequencies was still clean, with the required room shake from the drum.

Tony Bennett
TRACK 2—Who Cares
The piano now had the required weight you would *(to page 48)*

Perazella Critique

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expect from a Bosendorfer. The bass was also stronger. As a matter of fact, the bass is almost too heavy on this recording, and that was apparent in this configuration.

Kodo—*Ibuki*
TRACK 3—Akabanah
Wow! Holy cow! The drums sound like drums again. The lows were very strong and clean. The mids lost the IM distortion that was evident when the seven-incher was chugging its little heart out trying to act like a big mean old drum.

TRACK 5—The Hunted
Put on your flak vests for this one. Floor-pounding, chest-thumping bass was abundant, again coupled with a very clean midrange. The annoying puffs in the face were also gone. As a matter of fact, putting my hand directly in front of the port revealed very little air movement, confirming the effectiveness of the high-pass filter.

CONCLUSIONS
Again, for consistency, reviewers present a subjective rating of components being tested using four categories: presence, stereophonic effect, soundstaging, and ambiance. However, for testing speakers I'm also including three other characteristics that I believe are very critical: tonal balance, dynamic range, and distortion. Ratings range from 0 to 10, with 10 being heavenly and 0 being unlistenable.

These speakers are a great example of good engineering. Given their size and price, they get the basics right without trying to reach for the sky. By doing so, the addition of a sub to extend the frequency and dynamic ranges and also lighten the burden for the bass/mid driver yields a truly enjoyable system. Good speakers allow you to move into the recording venue and feel the music as a participant rather than an outsider. These speakers provide that to a high degree. Lesser approaches sacrifice midband performance to try to extend reach, making improvements more difficult to achieve.

If price is an issue, there are other speakers that have a slightly more extended reach into the bass, but not with the clarity of the DM602s. If you never plan to add a subwoofer, some of those may be better choices. But, if you are planning to use any speaker in this size and price range without a sub, I think you are missing the mark. There is no other addition you can make that will have as significant an effect as using a sub. Look at the results of changing from a $350 amplifier to a kilobuck amp compared to using the small amp and the sub. The difference is huge.

Save your pennies and buy or build your own sub. Never before has there been a time when you have had so many choices of great drivers to build a sub. Companies such as ACI, HSU Research, Adire Audio, Madisound, and Parts Express, to name just a few, have drivers you would have killed for just a few years ago and at prices that will make people think you were holding a gun to someone's head when you bought them. News Flash: Parts Express has just come out with a pair of dual voice-coil drivers with X_{MAX} of around 16mm at ridiculous prices. The 15″ driver has a suggested price of $132 in single quantities and $125 for four or more. This is probably the new champ for volume displacement per dollar. So give the DM602s a listen and then go build a killer sub for a great-sounding system.

D'Appolito Review

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the test. Then a second-order nonlinearity will produce intermods at frequencies of $f_1 \pm f_2$. A third-order nonlinearity generates intermods at $2f_1 \pm f_2$ and $f_1 \pm 2f_2$.

I first examined woofer intermods by inputting 900Hz and 1kHz signals at equal levels. These frequencies should appear predominantly in the woofer output. Total SPL with the two signals was adjusted to 86dB at 1m. Because steady tones are used in the IM test, I thought it safer to use a lower power level to prevent possible tweeter damage. Principal woofer IM products occurred at 800, 1100, 2800, and 2900Hz. However, the overall level was only 0.47%, an excellent result.

I measured tweeter intermods with a 10kHz and 11kHz input pair also adjusted to produce 86dB SPL at 1m, observing IM products at 8, 9, 12, and 13kHz. Total distortion was 0.18%. Again, this is a very low figure.

The last IM test examines cross-intermodulation distortion between the woofer and tweeter using frequencies of 900Hz and 10kHz. (A 1kHz signal would produce intermods that fall on harmonic-distortion lines, confusing the results.) Ideally, the crossover should prevent high-frequency energy from entering the woofer and low-frequency energy from entering the tweeter. IM products appeared at 6.4, 9.1, and 10.4kHz at a level of 0.07%, the lowest figure I have measured so far in the series of tests.

ADDITIONAL TESTS

I conducted all of the above tests with the grille off. *Figure 11* shows the 602's system response with the grille on, but referenced to the response with the grille off; that is, it plots the change in response under the two conditions. Below 3kHz the grille has little effect. Above 3kHz, however, the grille causes ragged response deviations of +2 to –4.5dB. As usual the grille has only cosmetic value.

Two samples of the 602 system were available for testing. I conducted all of the tests described so far on one sample. Frequency response of the second sample matched the first to within –0.5 and +1dB from 100Hz – 16kHz. Tweeter response of the second sample rose to 1.8dB above the first at 20kHz, but this should be of little consequence. ❖

Product Review

NAD T550 DVD Player

Reviewed by Gary A. Galo

PHOTO 1: Front view of the NAD T550 DVD player and remote control.

NAD T550 DVD Player. NAD Electronics International, 633 Granite Court, Pickering Ontario, Canada L1W 3K1, 905-831-0799 (Worldwide), 800-263-4641 (NA), nad@NADelectronics.com, www.nadelectronics.com. Price: $799.

The T550 (*Photo 1*) is NAD's first entry into the DVD-player market. NAD calls their T550 a "Music First" DVD player, and their goal was to build a player with the same level of musical performance as their best stereo audio products. The T550 is a completely self-contained product, with built-in Dolby 5.1 and DTS surround decoding.

The video performance of the T550 has been designed to compete with the best DVD players on the market. Video processing and D/A conversion are accomplished with a state-of-the-art, fourth-generation chip. Like all current-generation DVD players, the T550 has 24-bit/96kHz digital audio circuitry.

You can also use the T550 as a transport, with external audio decoding. Both RCA coaxial and Toslink optical digital outputs are included. You can feed these outputs to an external two-channel D/A converter, Dolby 5.1 decoder, or a DTS decoder. Composite, component (Y/Cb/Cr), and S-Video outputs are also included, allowing optimum interfacing with any monitor or projector (*Photo 2*).

In the 230V version of the T550, the component outputs are replaced with SCART RGB outputs. The 230V version also has built-in MPEG decoding. The T550 has a single-laser-beam pickup, so it won't play CD-R discs (dual wavelength laser pickups are required).

The T550 is a full-featured DVD player, supporting Multi-Angle, Multi-Sound, and Multi-Subtitle capabilities on suitably provided software. Frame-by-frame viewing is possible on all DVD discs, as is chapter and title repeat. The T550 also allows you to repeat sections between two points of your choice.

You can also store markers in the player's memory, allowing you to return to it at a later time. You must leave the DVD in the player, however. Opening the drawer erases the marker memory. A zoom feature is also included, allowing you to "zoom in" by enlarging the picture to either 4× or 16× its normal size.

The T550's menu system includes all of the standard choices offered these days by DVD players, and is quite intuitive in operation. The menus allow selection of the disc audio language, subtitle language, menu language, aspect ratio, TV mode, FLT brightness, and audio output. The remote control is easy to operate, though somewhat directional. The T550 is labeled as manufactured in South Korea.

CIRCUIT OVERVIEW

NAD did not provide me with a service manual or a schematic diagram; however, from their product literature and a look inside the player (*Photo 3*), I was able to make some observations. On the video end, the heart of the T550 is a 10-bit video D/A converter, the Thomson Omega-DVD STi5505AVB-X, one of the best chips available today. NAD's product literature claims that a Burr-Brown PCM1716, a 96kHz/24-bit digital filter/DAC combination chip, is used for the two-channel stereo/downmix outputs.

My player did not have a PCM1716, however. The review sample has a Burr-Brown PCM1600, a six-channel 96kHz/24-bit Delta-Sigma DAC, with on-board digital filters. Since the PCM1600 is the only DAC chip on the T550's PC board, I assume that two channels of this chip are also used for stereo/downmix operation.

The PC board in the review sample contains an empty surface-mount footprint labeled "PCM1716," but the chip is not installed. I suspect that the PC board was designed so that it could be used in players with and without internal Dolby 5.1 decoding. A player without internal Dolby 5.1 decoding would probably employ a PCM1716 for the two-channel-only internal D/A conversion.

Most DVD and CD players made in the Far East contain mediocre analog circuitry. 4560 op-amps are commonplace, and many players still incorporate the 5532. NAD has paid careful attention to the analog circuitry in the T550, selecting the excellent Burr-Brown OPA2134 dual op-amp for all audio amplification.

The OPA2134 is essentially a dual version of the OPA134, a high-performance op-amp, and part of Burr-Brown's *Sound Plus*™ series. The chip has a slew rate of 20V/μs, a bandwidth of 8MHz, and a high open-loop gain of

PHOTO 2: Rear panel of the T550. The player has built-in Dolby 5.1 decoding and the six required outputs. Separate stereo/mixdown outputs are provided for two-channel operation. Component, composite, and S-Video outputs are also included.

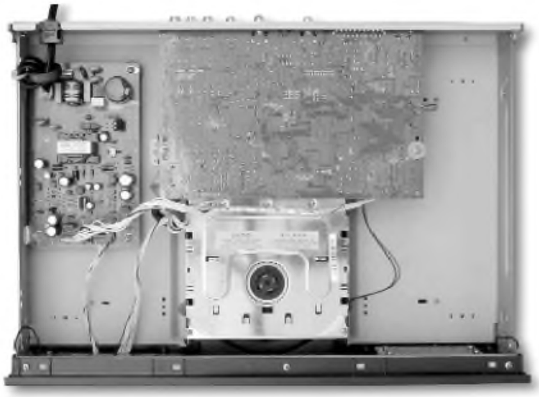

PHOTO 3: Inside the T550. The main PC board is mounted upside down. The switching power-supply board is on the left.

120dB. A true FET input results in an input bias current of only 5pA, yet the voltage noise is only 8nV/√Hz. A total of four OPA2134 chips are used, one for the stereo/mixdown outputs, and three for the six Dolby 5.1 outputs.

NAD has also employed metal-film resistors and polypropylene capacitors in critical locations. As *Photo 4* shows, these appear to be in the form of surface-mount, chip-type components. NAD deserves kudos for paying attention to the quality of passive components in the T550. The audio outputs of the T550 appear to be coupled with electrolytic capacitors, however.

POWER SUPPLY

All DVD players, regardless of manufacturer, incorporate switching power supplies. The T550 is no exception. Audiophiles may cringe at the thought of a "computer" power supply operating high-end audio equipment, but the entire DVD player industry appears to have embraced switching supplies.

The switching-supply board outputs several voltages to the main PC board. 5V lines labeled A5V and D5V are, in fact, the same supply line fed to two pins on the PC connector. There is also a separate 5V pin labeled V5V. Two 3.3V pins are from the same source, and a pair of +12V pins are also ganged together. There is also a single –5V pin.

The main T550 PC board has three three-terminal IC regulators. A 7808 and 7908 pair is used to regulate the positive and negative supplies for the analog circuitry.

I made some measurements in order to learn more about the power-supply scheme, and stumbled upon a curious design situation. The 7808/7908 IC regulators are fed from the switching supply's ±12V lines through a pair of series inductors which, along with a pair of bypass capacitors, reduce noise from the switching supply. The series inductors seem to have been an afterthought—traces have been cut on the PC board in order to accommodate them (the traces would bypass the inductors, otherwise). The series resistance of these inductors is high enough to drop several volts, to the point where neither regulator has sufficient input voltage to operate properly.

The input to the –8V 7908 regulator is at –7.75V, *less than the actual regulator voltage.* This means that the 7908 is not regulating at all. The output from the 7908 is –6.96V. The input to the 7808 sees +9V, and it outputs 7.75V.

Although the output is within the tolerance of a 7808 regulator, I was suspicious: an input/output difference of only 1.25V is insufficient for a regulator of this type. To confirm my suspicions, I temporarily soldered jumpers across both series inductors. With ±12V at the regulator inputs, the regulated voltages now measured a full ±8V.

As configured by NAD, both analog supply regulators are in a constant state of dropout: the high-performance OPA2134 op-amps are powered by unregulated supplies. With the series inductors jumpered, the output noise from the regulators, viewed on a wideband oscilloscope, is essentially the same as with the inductors in the circuit. When the regulators are functioning properly, their own line rejection has the same effectiveness as the filtering from the series inductors. When in a state of dropout, they have no line rejection.

I am completely baffled by the design rationale for this power supply. There is a second 7808 regulator on the main PC board, but its function is unclear. This 7808 is fed directly from the +12V line and is operating properly with an output of +8V.

PERFORMANCE

It is hard to find a DVD player with a bad picture these days. I found the picture quality on the T550 to be as good as any DVD player I have seen: sharp, detailed, with brilliant, life-like color. It is easily the equal of Sony's flagship DVP-S7700, which retails for $999. You'll be hard-pressed to better the picture quality of the T550. But, I was even more interested in the sound quality of the T550, so I'll cover that in greater detail.

I do not own a surround-sound setup, so I took the T550 to a local audio-video dealer to evaluate its Dolby 5.1 surround performance. The player on hand for comparison was a Sony DVP-S560D. Although this player retails for only $399 (half the price of the T550), it is Sony's only single-disc DVD player with built-in Dolby 5.1 decoding.

In the film *Armageddon* (The Criterion Collection #40), the NAD T550's

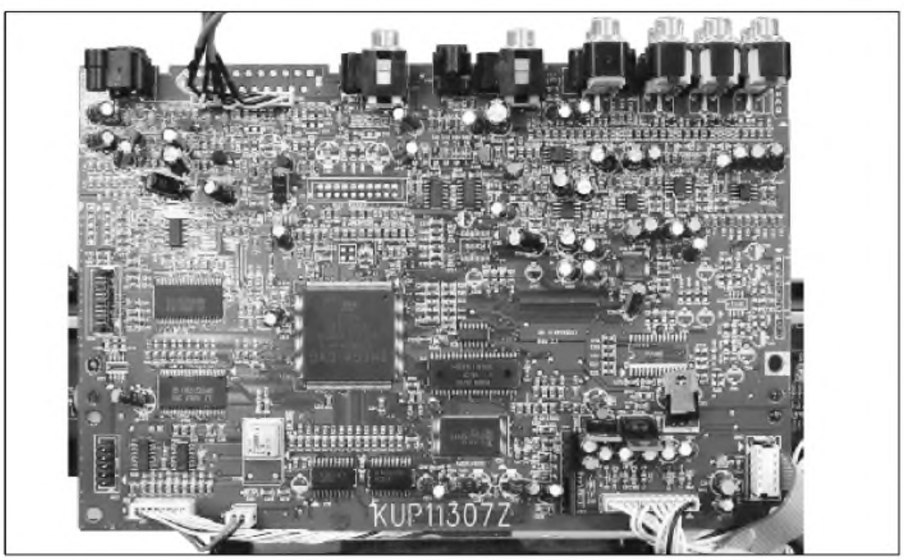

PHOTO 4: The T550's main PC board. You can see the four Burr-Brown OPA2134 dual op-amps in the upper right, near the output RCA jacks.

Dolby surround performance was clearly superior to the Sony player. The surround effects were reproduced with a substantially larger soundstage and more precise localization. Detail and treble smoothness were also superior to the Sony.

On Herbert von Karajan's 1983 film of Beethoven's Symphony No. 9 with the Berlin Philharmonic (Sony Classical SVD 46364), there was a greater sense of hall ambience in the Dolby 5.1 surround mode. The string tone was sweeter and more natural, and the orchestra was presented with greater weight and dynamic impact.

CD CRITIQUE

Unlike most DVD players, the T550—price considered—gave a reasonable account of itself as a stand-alone two-channel CD player. In particular, the T550 outshone the Marantz PMD340 player (which I review in an upcoming issue of *aX*). The T550 has a smoother, more musical, more articulate sonic presentation, and more precise soundstaging than the Marantz. The Marantz has a superior power-supply regulation scheme, but its DAC and analog circuitry are well behind those used in the T550.

Even though the T550's power-supply topology is inferior to the Marantz, the excellent Burr-Brown DAC and op-amps used in the T550 put the balance in NAD's favor. Overall, the sonic presentation of the T550, used as a CD player, errs toward the warm side, and thus offers musical performance superior to the more analytical Marantz.

I also evaluated the T550 as a CD transport, using my reference Parts Connection DAC 3.0 as the outboard D/A converter. Here, the T550 performed extremely well. In comparisons I made with Sony's top-of-the-line DVP-S7700 DVD player (retailing at $999), the NAD T550 consistently outperformed the Sony player. The NAD was simply more musical, with a larger soundstage, greater inner detail, and a smoother and warmer sonic presentation.

The sonic presentation of the Sony, as a transport, was more analytical. This despite the fact that the Sony is physically much more robust than the NAD. As a stand-alone CD player, I also found the Sony inferior to the NAD.

I also evaluated the T550 using one of the DAD, audio-only DVD discs from Classic Records, their 96kHz/24-bit PCM transfer of the Rachmaninoff *Symphonic Dances* performed by the Dallas Symphony Orchestra conducted by Donald Johanos. This superb recording was engineered by David Hancock in 1967, and released on LP by Vox/Turnabout. Played using my Pioneer DV-525 DVD player, which has a 96kHz/24-bit digital output, and my outboard DAC 3.0, this disc is one of the most impressive recordings I have ever heard—dynamic, airy, with instrumental timbres that are uncanny in their realism, and tremendous bass impact.

The T550's digital output has been downsampled to 48kHz, and the downsampling removes some of the subtleties which make this recording so special. On the T550, used as a transport, this recording is less airy, and slightly "closed down" when compared to the sonic presentation on my Pioneer DV-525 (if I change the output

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Save the Tubes

Class A audio is among the least stressful uses for a tube if you take precautions to avoid conditions known to lead to premature failures. This article shows you how. **By Larry Lisle**

Vacuum tubes have an undeserved reputation as being short-lived, unreliable devices. They're not. When operated with care, tubes can last thousands of hours.

RECONDITIONING NOS TUBES

Many tubes used today are 40, 50, or more years old! Even though they're NOS (new, old stock) they've been sitting a long time, and you should take some precautions before firing them up, especially the big costly ones!

First, hold the tube upright and gently tap the envelope to let tiny particles of whatever may have been left inside the envelope settle to the bottom before they cause arcing. Next, reactivate the getter to remove any gas that may have invaded the tube through the seals during its decades on the shelf. The easiest way to do this is to build a simple test stand with an appropriate socket (*Fig. 1*).

Wire the filament pins to a source of the correct voltage; the grid should be connected to the cathode through a 1kΩ resistor; the plate and screen are tied together and connected to a source of 200V or so through a milliammeter. The cathode is connected to ground through a variable resistor of a value and wattage that will limit the current to about one-third of normal. (Consult a tube manual and use Ohm's law.) Apply the filament voltage gradually, and then apply the B+, also gradually, watching the milliammeter for signs of runaway. If all seems well, let the tube idle for a few hours, and it should be ready.

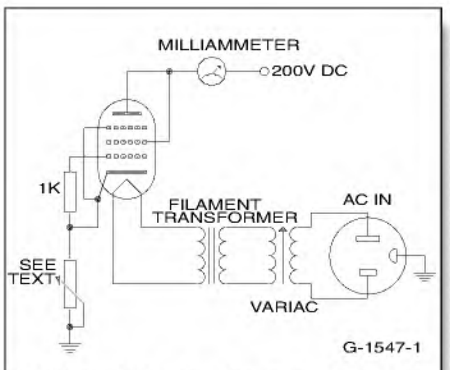

FIGURE 1: Schematic of a simple test stand.

PHYSICAL CONSIDERATIONS

Putting a tube, especially one in which the pins go directly through the glass envelope, into a socket can be very hard on the tube. It's easy to start a crack that may eventually lead to tube failure. Be careful in removing and replacing tubes. I recommend a pin straightener. Remember, sockets should have some "give" around the pins. Be careful when soldering that each of the socket holes has a little wiggle room.

Filamentary tubes should be oriented properly so the filament doesn't sag near another element—vertical is best. Sometimes manufacturers will say horizontal mounting is permitted if certain socket pins are in a vertical plane. This is true if the elements are in alignment with those pins. Sometimes they're a little off, and a rectifier, for instance, working at near maximum ratings may arc when it wouldn't normally be expected to.

Be cautious, also, about working with filamentary tubes on a chassis tipped on its side for servicing or testing. It's better to turn it completely upside down.

Any metal that comes in contact with the glass envelope of a tube can cause the glass to heat or cool unevenly, which may eventually lead to failure.

Speaking of heat, this is probably the worst enemy of tubes and other components. Don't crowd your chassis; allow for ventilation. Tube coolers and forced-air cooling may be worth looking at. Use shields only when absolutely necessary, and then use only those with blackened interiors.

CIRCUIT CONSIDERATIONS

Filament voltages in excess of the center value given in the manuals should be avoided. It's far better to let the filaments run a few percent low than too high. Also, try to limit the strains caused by turning on filaments. Possible methods include letting the filaments run all the time, even when the high voltage and bias are turned off, or letting them run at 50% voltage when the other voltages are turned off. These methods are more suitable for studios. For the home, use a surge-protection system, which can be as simple as a resistor and a switch.

Never apply the plate or screen voltage until the cathode is at operating temperature. (*Tube manuals usually list filament heating times.—Ed.)* Try to avoid having more than 90V between the filament and cathode, no matter what the manual says.

Fixed grid or screen voltages can lead to tube failure in the event of the demise of another component, but sonic considerations may conflict with this. If there's a choice between two tubes to do the same job, other things being equal, the one with the bigger elements and envelope will probably last longer.

Be careful to avoid unwittingly using

ABOUT THE AUTHOR
Larry Lisle is a teacher in Rockford, Ill. He has been writing articles, generally of a technical nature, since 1968. His hobbies include building and restoring audio and radio equipment, amateur radio, and coaching basketball.

a "select" tube in a circuit you're designing. That is, if at all possible, try several tubes of the same type in the socket of your prototype to make sure it works satisfactorily with all of them. If you design your circuit around one tube and it happens to have unusually high transconductance or low microphonics or hum or whatever, you may have difficulty in the future understanding why your circuit stopped working correctly and a new tube won't fix it.

If your design includes filamentary and heater-type tubes, take into consideration the different warm-up rates and be sure all tubes are properly biased when the high voltage is applied.

Finally, check your line voltage from time to time. Some areas frequently run much higher than the standard 117V. This can shorten tube and component life. Also, remember that many years ago the standard was 110V. If you're running some antique equipment, you could really be over voltage!

TUBE TESTING

I'm not very fond of tube testers for two reasons: first, they usually don't test the tubes under the conditions of the circuit they're used in (an exception is the simple tester I described in "An Audio Tube Tester," *GA* 1/98, p. 12).

The second reason is that they can actually harm the tube they're testing. For example, some testers connect all the elements except the filament and cathode together and apply AC voltage through a milliammeter between the cathode and the other elements. The idea is to test the emission of the cathode. But the grid is now many volts positive compared to the cathode, and in high-transconductance types it can easily draw excessive current and be overheated. I much prefer to "test" a suspect tube by replacing it with a known good one.

For routine checks on tubes, as in a studio, include in the circuitry a provision for easily measuring the plate current of each tube in an amplifier while the circuit is functioning normally. This will be more enlightening than a tube-tester reading, and will avoid the need to remove and replace tubes in their sockets, which can lead to eventual failure. It will also tell much about how the whole stage is doing, not just the tube.

I hope some of this material will prove useful. It's fortunate that some tubes continue to be made and some excellent new types have been introduced. Other less popular tubes are definitely a nonrenewable resource, but with care they can last a long, long time. ❖

TABLE 1
FURTHER READING

Tomer, Robert, *Getting The Most Out of Vacuum Tubes*, Howard Sams, 1960. An excellent book on tube reliability, hard to find. (The book is available from Old Colony.)

Smith, Langford, *Radiotron Designers Handbook, 4th Ed.*, Part 1. Available from Old Colony Sound Lab, 888-924-9465, www.audioXpress.com.

Kleronomos, Bill, "Don't be Sorry! Condition Those Tubes," *Electric Radio Magazine*, Nov. 1993, Electric Radio Press Inc., 14643 Counth Rd. G, Cortez, CO 81321-9575. Excellent article on de-gassing old tubes.

Kleronomos, Bill, "Electron Tube Survival Primer," *Electric Radio Magazine,* Oct. 1994. Tube shields compared.

Osterwald, Ray, "Thermionic Mysteries," *Electric Radio Magazine*, Oct. 1993. A look at cathode interface resistance.

THD Is Meaningless, Part 2

Are intermodulation distortion figures far more relevant than those of THD in measuring amplifier sound quality? This author, with his explanation of what IMD is and how to measure it, believes so. **By Anthony New**

(Reprinted with permission from Electronics World.*)*

Although IMD has not been discussed as much in the design of audio amplifiers compared with THD, there is considerable literature on IMD in general and its application to RF amplifiers. For this reason, I will give a simple overview of IMD and point out a few implications for its use in audio design.

WHAT IS IMD?

In general, IMD is produced whenever two or more signals with distinct frequencies F_1 and F_2 pass through a device—be it an amplifier, filter, or other circuit—that possesses an amplitude nonlinearity of the form

$$Y = A_1X + A_2X^2 + A_3X^3 + A_4X^4 + A_5X^5 + \ldots$$

where A_1 is the nominal gain of the amplifier and the higher powers of X correspond to the various nonlinearities that may be present. I have ignored phase nonlinearities here for simplicity.

This nonlinearity produces IMD products at the following frequencies

$$nF_1 + mF_2$$

where n and m are non-zero integers. Note that if you put n.m = 0, you get purely harmonic distortion rather than IMD, which indicates that harmonic distortion is a special case of a more general phenomenon.

The order of the IMD products is defined as

$$k = |n| + |m|$$

so that the "third-order" products that often dominate are of the form

$$F_1 \pm 2 \times F_2,\ F_2 \pm 2 \times F_1 \text{ and } 3 \times F_1,\ 3 \times F_2$$

In RF amplifiers a further restriction often applies. Even in many "wideband" products the overall bandwidth of the amplifier is less than an octave, and so only those odd-order products with

$$|n| - |m| = \pm 1$$

are "in-band" and of concern. Consequently, even-order nonlinearities—second, fourth, and so on—which produce no odd-order products are usually ignored.

However, in a multi-octave device such as an audio amplifier, this restriction will not apply. The most common and usually most important nonlinearity is, however, still a third-order nonlinearity of the form:

$$Y = A_0X + A_3X^3$$

which will result in IMD products of third order only, namely ($2F_1 - F_2$, $2F_2 - F_1$, $2F_1 + F_2$, and $2F_2 + F_1$). The first two represent the classical IMD products and the other two are higher-frequency IMD products, at roughly three times the fundamental frequencies when these are close together.

The spectrum of Fig. 6 in Part 1 shows these third-order products of a two-tone signal, in addition to higher-level, second-order products. Note that only the third-order products are close in frequency to the input tones.

Figure 7 shows an idealized spectrum of four-tone test sometimes used with RF amplifiers. If the power input to the device is varied, the output levels of the IMD products will vary, too. This

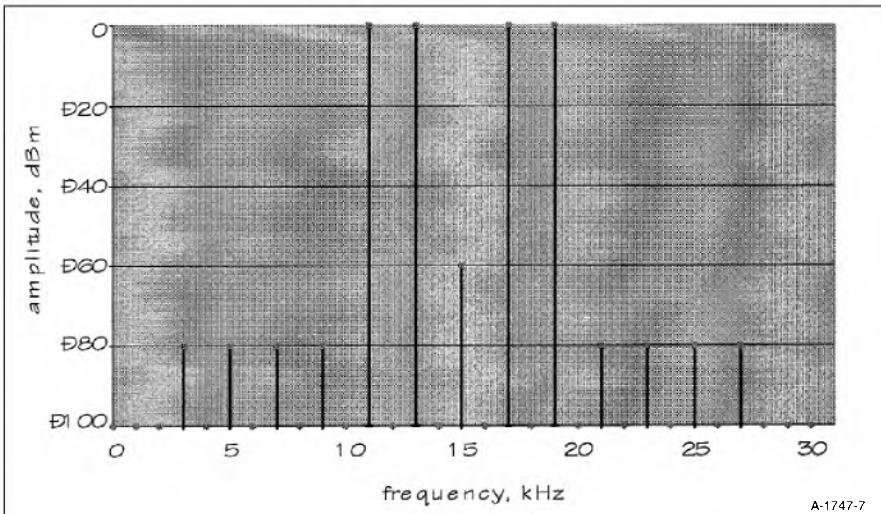

FIGURE 7: Idealized spectrum of four-tone test sometimes used with RF amplifiers. The four main tones are harmonically related, phase-locked and phase-peaked to maximize the peak value of the signal envelope in the time domain. This—in an RF amplifier, at least—is likely to maximize the visible third-order IMD products, particularly the central one between the two pairs of tones which thus makes an easy frequency component to check. With zero IMD this component would be completely absent.

is shown in *Fig. 8*, from which you can see that if the input level increases by 10dB, the third-order IMD products increase in absolute level by 30dB. Their level relative to the wanted output signals also increases by 20dB.

If the straight lines are extended to the right, you will see that they all meet at a single point. For obvious reasons, this is known as the "output intercept point." Strictly in this instance, it is the "two-tone, third-order output IMD intercept point," or IP3.

For any signal below this point, you can estimate the level of IMD products by subtracting the output signal level from the IP3 to give a figure in decibels, and doubling this to give a figure in dBc. This represents the IMD relative to the "carrier," i.e., wanted signals, assuming them to be similar in level.

REAL SIGNALS

It is highly unlikely that a real device could be operated anywhere near its IP3 point, which is only useful for calculation and reference. Also, a real device is likely to show IMD at other or-

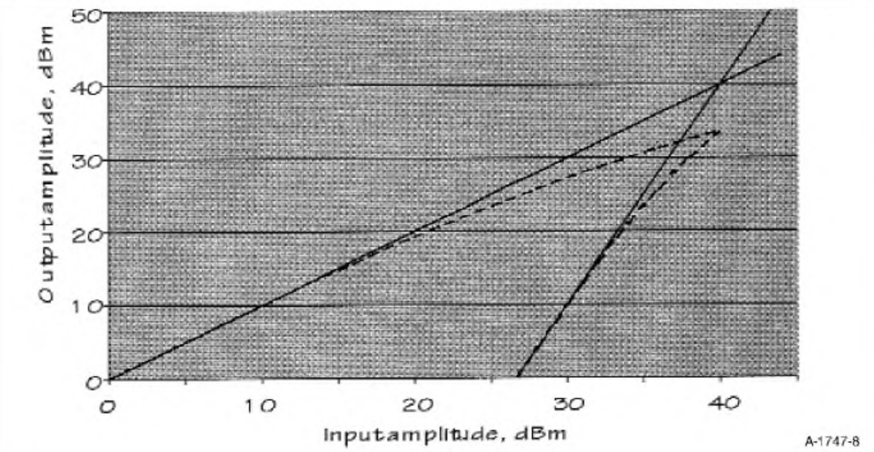

FIGURE 8: Amplitude response of amplifier displaying "classical" third-order IMD. Amplifier input signal level is displayed on the X-axis, and output level on the Y-axis; both axes are logarithmic. The straight line through the origin represents ideal linear response. The straight line at a steeper angle shows the theoretical level of IMD products, which change three times as quickly with input amplitude as the signal itself. The point where the straight lines meet is the IP3. The curved lines show the likely real characteristics as the amplifier begins to clip. However, for sensible operating points well below the IP3, the straight lines are a fairly good match for a single-stage class-A RF amplifier without any special linearization techniques. The IP3 concept is also useful for other devices such as mixers, which also display IMD. For any input signal level on the X-axis, the upper line will show the nominal output level, and the vertical separation between the two straight lines will show the expected linearity in dBc. When high-level multi-tone signals are concerned, this figure—rather than the noise figure—usually represents the dynamic range of the signal, since it indicates the relative level of interfering products.

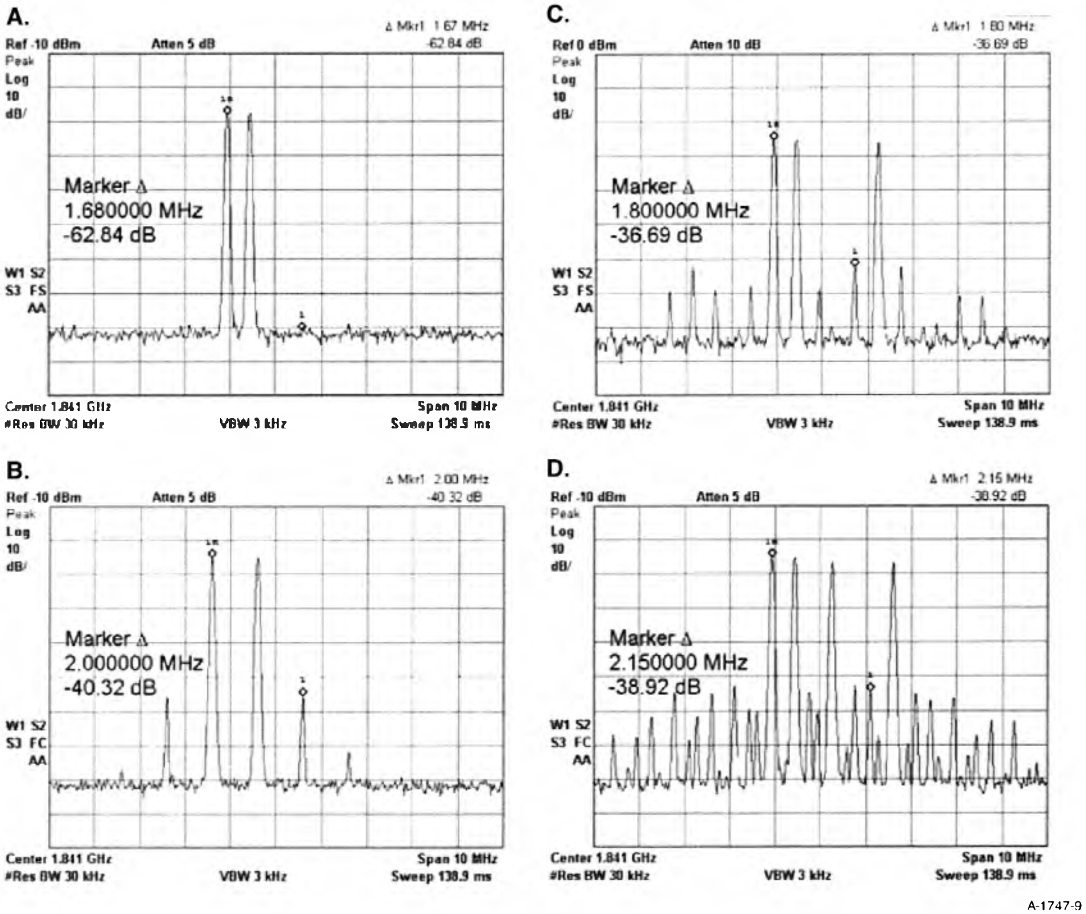

FIGURE 9: Spectrograms of real signals. For convenience these have been taken at RF, though similar spectra could be observed at audio. (a) two-tone signal, no visible distortion; (b) two-tone signal plus obvious third-order IMD; (c) three-tone signal; and (d) four-tone signal. Note that each extra main tone produces many extra IMD products. In (d) the tone frequencies have been deliberately chosen to make as many products visible as possible; usually several would either overlap or appear to do so within the resolution of the spectrum analyzer. However, in a complex musical signal the large number of signal tones would cause the hundreds of IMD products to merge into a noise-like background which reduces the clarity of the signal.

ders, particularly fifth, when it is driven at all hard. As these will reduce by the fifth power of the signal level instead of the third, though, they are likely to be lower in level.

However, a fifth-order nonlinearity will also produce some third-order IMD. This may even cancel out some or all of the third-order IMD produced by the third-order nonlinearity, resulting in the fifth-order IMD product dominating at some output power. A similar situation exists for higher-order IMD products, but the actual levels are generally both lower than third- and fifth-order IMD products and rather less predictable.

When more than two large signals are sent through the same amplifier at the same time, the number of IMD products grows rapidly. *Figure 9* shows spectra of a real, albeit RF, amplifier with real multi-tone signals.

Two tones produce two close-in IMD products, in addition to the other distant ones shown in Fig. 6, but nine products are visible with three tones (*Fig. 9c*).

With four tones the number increases again (*Fig. 9d*), though in practice some of these may be co-incident. When a complex modulated signal is used rather than a set of CW tones, the IMD products occupy a bandwidth rather like a noise spectrum.

Since these IMD products are not harmonically related to their causative signals, they behave like noise, too, reducing the intelligibility of speech or data transmissions to a measurable degree. It is not possible to filter them out, since although the bandwidth they occupy increases with their order of distortion (*Fig. 10*), their bandwidth always includes the original signal.

MEASURING

As I commented, one of the benefits of IMD over THD is the relative ease of measurement due to the distorted products being not harmonically related to the original signals. A typical setup will consist of a pair of signal generators—or, often, a dual-output generator—a linear combiner, possibly resistive, and a spectrum analyzer (Fig. 4 in part 1). The analyzer display will then look something like Fig. 6 if the analyzer span is sufficiently wide, or like *Fig. 9b* in the more usual narrowband case.

The IMD products may be much lower in level, but are easily seen, provided the analyzer has enough dynamic range. If the analyzer has appropriate delta markers, you can simply read the relative distortion off the screen display. If not, a little mental arithmetic is required.

Note that with IMD tests it is not necessary to use especially low-distortion oscillators, since the harmonics produced will not normally interfere with the measurement process. However, the linearity of commercial spectrum analyzers is rarely much better than 80dBc—or 0.01% in voltage terms—and may be poorer. For the best amplifiers some additional filtering may be needed to notch out the pure signals, or a coherent subtraction method (Fig. 5).

Another test commonly used is the four-tone test illustrated in *Fig. 7*. Here, four tones at, say, 3, 4, 6, and 7kHz are produced by four phase-locked generators and the analyzer tuned to look for the missing 5kHz component, which can only arise from a nonlinearity.

FIGURE-OF-MERIT

A figure-of-merit commonly used in RF amplifier design is the intercept point in dBm (*Fig. 8*). The higher this is for a given power level required from the amplifier, the lower will be the distortion produced, and from this IP3 figure it is quite easy to calculate how much distortion is likely at any given power level.

However, at this point I should comment that one of the many differences between RF and audio amplifiers is that RF amplifiers are usually operated somewhere near their continuous peak power rating. Alternatively, they are at least backed off from this by a consistent amount. Also, they do not often use feedback to achieve good linearity. Consequently, they may have an IMD response which approximates to a classical curve over most of their useful power range and for which a single IP3 specification is a useful measure of linearity in any application.

Where real audio amplifiers are concerned, I believe that typical responses are unlikely to be so simple over the wider range of signal levels encountered, particularly in a "blameless amplifier," where all of the distortion mechanisms have been separately identified and reduced to a low level by various means. Furthermore, audio music signals can have a very high peak-to-mean ratio. It is common practice to specify amplifiers with a power handling greatly in excess of what is normally required. As a result, much of the time they will be operating at a very small fraction of their nominal power output, where the real distortion produced is somewhat different from the "classical" third-order model.

It is likely, therefore, that rather than a single calculated IP3 figure, a curve of measured IMD levels versus signal level is more appropriate, rather like the waterfall spectrographs sometimes used. It would also give a far better indication of the order of distortion produced than a single figure, even an IP3 figure.

Nevertheless, the level of intermodulation produced by an amplifier is, as I have shown, absolutely critical to its quality as an audio device. Any useful specification for its linearity should reference this. I therefore propose that the specification should be something like this:

"Two-tone third-order IMD performance: better than –70dBc over 0.1W to 30W and 50Hz to 20kHz"

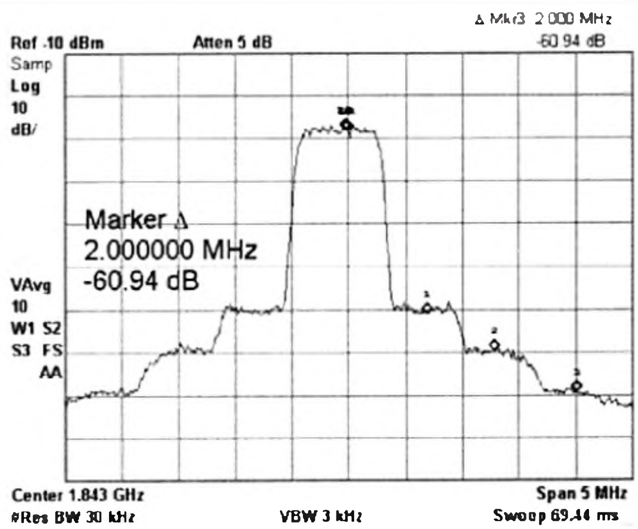

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FIGURE 10: Example of a real modulated signal with IMD—here the IMD products are not discernible individually but serve effectively to raise the noise level in steps—each step corresponds to a particular order of IMD. The first step on each side is produced by third-order IMD, the next by 5th, the next by 7th, and so on. Note that the IMD products occupy more bandwidth than the original signals; the higher the order, the more bandwidth occupied. The step pattern might, however, not be visible with the less-ordered spectrum of a typical music signal.

DECIBELS, DBM, DBW, AND DBC

It is common to specify amplifier distortion in terms of percentage, with the understanding that voltage ratios are intended. However, where loudspeakers have to be driven, it is power that is more relevant.

For constant-impedance systems with a wide signal range, a convenient logarithmic measure is the decibel, or dB. This is strictly a ratio of two quantities with the convenient feature that 10dB corresponds to an increase in signal power by a factor of ten, and 20dB corresponds to an increase of ten in voltage and ten in current, making 100 in power. Specifying an increase of 20dB is then unambiguous, regardless of whether the speaker is thinking in terms of power or voltage.

Where an absolute level is needed, the terms dBm, i.e., dB relative to one milliwatt, and dBW, i.e., relative to 1 watt, are commonly used. In specifying levels of distortion, a further measure is useful, namely dBc. This refers not to a noise-suppression scheme but to decibels relative to the carrier, i.e., the main signal.

Where multi-tone signals are present, there is, however, a further possible confusion between dBm/tone, dBm mean, and dBm peak.–AN or something similar. In practice, it may be necessary to limit the tests to a set of standard test tones, for example, 3.5kHz and 4.5kHz. Intermodulation products would be looked for at 1kHz, 2.5, 5.5, 11.5, and 12.5kHz.

Harmonic products at 7kHz and 9kHz might also be present but could be due to the signals sources themselves. It would also be possible to repeat the test at additional low and high frequencies to test IMD performance there, such as 350/450Hz and 15/16kHz.

Of course, modern lab equipment is capable of performing swept measurements and downloading the results to a PC for analysis and printing, so a swept measurement may be acceptable as a standard.

SECOND-ORDER NONLINEARITY

The third-order function discussed earlier was selected to represent a typical amplifier nonlinearity. What would happen with an amplifier having a second-order nonlinearity? This is a rather interesting case study, as it helps to explain the difference between "valve sound" and "transistor sound," which used to convey such emotion many years ago...and in some circles still does.

A second-order nonlinearity such as that often found in a thermionic valve produces second-harmonic distortion—which is not unpleasant in moderation. And it produces only even-order IMD, namely zeroth and second-order, at low level. It produces no odd-order IMD products of the form ($F_1 \pm 2 \times F_2$).

A narrowband amplifier produces no in-band IMD at all. Thus the absence of any second-harmonic cancellation in a class-A configuration has no impact on the IMD present, as suspected by those who prefer this configuration.

For much of the music program, the loudest frequencies present in the signal will often be harmonically related. Many of these extra distortion terms, of the form $F_1 \pm F_2$, will fall on, or close to, existing signal frequencies at much higher level and may be reasonably effectively masked.

Provided the levels of distortion are not excessive, the result will probably not be particularly unpleasant, and may give the effect of a warm coloration to which you can become accustomed. Note that this form of distortion reduces markedly as output level drops, so that soft passages may be portrayed quite realistically. Loud passages are likely, in any case, to have a richer harmonic texture which hides the IMD more effectively.

In contrast, the chief distortion mechanism of early transistor amplifiers was not large-signal output device nonlinearity, but crossover distortion. This generally becomes increasingly noticeable at low volume settings. Large amounts of feedback were often added to cure this and other problems, though the designers perhaps did not always appreciate how much the loop gain dropped in the crossover region. Consequently, transistor amplifiers tended to suffer from less high-amplitude low-order nonlinearity and more low-amplitude high-order distortion.

The effect of this on the reproduced sound was quite distinctive. Gone was the warm colored but fairly clean sound familiar to many who hadn't perhaps experienced the best valve amplifiers.

Even by the 1960s, these could boast less than 0.1% THD, most of that being the relatively benign low orders. In its place was a cold, muzzy (and sometimes hissy, but that's another issue) sound that could be particularly noticeable in solo piano works. I think the worst commercial design I ever heard was the "Sinclair 2000," which was pretty, but built down to a price.

It has been said that the unthinking application of negative feedback around an amplifier can often merely transform large amounts of low-order distortion into small amounts of high-order distortion. It is also true that any crossover-induced IMD remaining after the application of negative feedback is still present at low signal levels rather than diminishing with volume.

It would have been nice had designers appreciated the futility of their policy of measuring distortion in terms of THD at full output. Instead they concentrated in reducing it, albeit with some success. To those concerned with measurable THD, the trade-off seems worthwhile, but the high-order IMD products were usually spread far away in frequency from any masking tones in the signal, and were thus very audible.

On complex music containing many strong frequency components, the large number of high-order products degenerate into a background noise. This noise is signal-dependent and hides any subtle details from the ear (*Fig. 10*). It is no accident that the "clarity," often regarded as the highest accolade in audio, is the direct result of an absence of IM products.

TIM AND OTHER FACTORS

Distortion in phase can also occur in an amplifier that causes changes in pulse response. Real amplifiers also usually display some amplitude-to-pulse modulation and pulse-to-amplitude modulation conversion, too. I have deliberately avoided discussion of these since there is considerable doubt whether modest phase effects are audible at all. However, it is less contentious to say that over some of the audible range at least, differences in phase response between channels will at the very least degrade or alter the stereo image presentation and are therefore undesirable.

There is another form of intermodulation distortion that has been discussed in audio design, namely, transient IMD, or TIM. This is the distortion said to occur when a part of an amplifier suffers slew-rate limiting. For a brief period of time, the amplifier is unable to follow the input signal at all. During this time the amplifier gain is zero.

The usual remedy is to ensure effective low-pass filtering prior to any stage that suffers slew-rate limiting. But since the event is transitory, the distortion may not show up in steady-state measurements—particularly the continuous-sine waveforms generally used in total-harmonic distortion measurements. With a suitable input signal, though (one with a high peak-to-mean ratio, perhaps), this should show up in an intermodulation distortion test.

Other test waveforms are often used with amplifiers; for example, square waves to show load stability. These may well continue to be necessary, though it may be sufficient—and perhaps preferable—instead to measure the IMD performance with a range of realistic load impedances, since it is the distortion we are primarily interested in.

SUMMARY

I have shown here that current testing methodology fails to test the performance of audio amplifiers adequately in a manner that relates to audible performance. It fails to measure what it purports to do, namely, the difference between a representative complex and time-varying signal input to an amplifier and the actual output from it.

Instead, testing concentrates on an extremely narrow interpretation of "distortion" that the ear doesn't actually hear as distortion at all. It makes no attempt to measure important types of distortion that certainly are audible. And it does not apply the same rigor to frequency-response issues that it does to THD.

Tests for load stability are also generally done separately to other tests. This is presumably done on the assumption that variations in loads can't possibly affect other aspects!

In my view, any real test of an amplifier should apply a representative complex signal and compare this with the actual output of the amplifier under a range of likely loads. This could be done in many ways, with real or artificial sources and measured over frequency or time.

A suitable and relatively simple means exists which is already used in other fields, namely, IMD measurement under multi-tone conditions. The exact format of these tests could, and should, be adjusted to maximize their relevance to the particular case of wideband ultra-low distortion audio amplifiers. These or other measurements should also be capable of measuring frequency response to a far greater level of accuracy than is current.

When appropriate and psychoacoustically relevant tests are available, then perhaps we can better assess audio amplifiers objectively and better relate their objective performance to subjective tests. ❖

On Angels' Wings, Part 2

Completed speaker.

This second part details the speaker construction and listening trials by one audiophile smitten by the clarity of ribbons.

By Tom Perazella

Construction of the box was very simple. I cut four pieces of walnut for the sides, and one piece for the top. I glued and clamped the four side pieces and the top. After the top dried, I rounded the edges of the box with a disc sander and drilled mounting holes. I routed the back sides of the holes to allow attachment of the nuts. Looking back, I should have gotten the longer versions of the binding posts, BP1.5, which would have eliminated the routing.

Photo 20 is a top view of the box. To provide an exit path for the wire, I cut a slot in the back side of the box, which is the side that would face the driver. Using a slot eliminated threading wires through holes. I also drilled pilot holes for mounting in the bottom edges of the box using the base template. *Photo 21* shows the wire slot.

FINAL PREP

Veneering the baffle was the next step. The walnut veneer I chose was quite a bit heavier than the paper-based types used in some applications. The greater thickness provided a smooth, ripple-free surface when used over the Kerfkore. However, it was still sufficiently flexibile to wrap around the half-round on one side of the mounting plate.

At this point, I got cold feet; the thought of working with large, expensive sheets of veneer on top of two large, expensive sheets of Kerfkore®—into which many hours of work had gone—led me to look in Dave's direction with worried, pleading eyes. Dave came to the rescue. He not only applied the veneer, but also routed it to match the driver opening in the mounting plate. Thank you very, very much Dave!

The last stages of the assembly before finishing were to paint the unveneered, non-hardwood pieces with flat black paint, add foam to the interior cavities of the baffle, and mount carpet to the back plate. *Photo 22* shows the baffle masked before painting.

In order to prevent internal resonances in the cavities formed by the ribs and outer covering, Rudi suggested filling them with insulating foam. I decided to use the canned version available at the local hardware store (used to seal around window frames and elec-

PHOTO 20: Terminal mounting block.

PHOTO 21: Slot for lead wires.

PHOTO 22: Masking before painting.

trical outlet boxes). This material is easy to use and readily available.

To apply the foam, I inserted the tubing supplied into holes previously drilled into the bottom corner of each cavity. I then released the foam into the cavities. After using more than two cases of the foam, I realized that, were I to do this again, I would use an alternative approach.

The first would be to find someone who had an "Insta-Pak" machine, used by professional packers and is essentially a high-volume version of the home spray cans. They are refillable, and foaming this way is much faster and more economical. Another approach would be to buy blocks of foam and insert them into the cavities before applying the Kerfkore.

Before applying the carpet, I first cut it to size and then test-fitted it on the baffle. I purchased the carpet from Parts Express; it has a latex backing, making application easier. After I completed the sizing, I sprayed the back plate and carpet backing with 3M #77 adhesive. Once the adhesive was tacky, I applied the carpet and allowed it to dry. *Photo 23* shows the carpet in position.

THE FINISH

The last operation before final assembly was the finishing of the veneered surfaces and the top and bottom plates. Regardless of the finish applied, the most important step in any finishing operation is to properly prepare the wood. Translated, that means to sand, sand, and sand again with progressively finer grits of sandpaper. I did the preliminary sanding on the solid pieces with a belt sander. Following that I

PHOTO 23: Carpeted back plate.

PHOTO 24: Finished top plate.

used an inline sander, and finally, hand sanding. For the veneer, I eliminated the belt sanding.

Picking a finish is a matter of personal choice. When using high-quality hardwoods, I've always preferred a straight oil finish. Good hardwood has natural colors and shadings that can often stand on their own without help from a stain. Using a stain, may, in some cases, hide the natural beauty of the wood. In addition, oil provides a low gloss shine with a deep luster that says "real wood"; plastic need not apply here.

For this project, I decided to use one of my old favorites, Watco Danish Oil. Before proceeding, I took scraps of the veneer and solids, sanded them, and then applied the oil. It never ceases to amaze me how wood transforms as you apply the first coat of oil. The grain and natural color of the wood seem to leap out at you. Properly finished real wood is, as Martha would say, "a good thing." Compare *Fhoto 24* (showing a finished top piece) with Photo 19 (which shows an unfinished piece).

After several sandings and coatings of oil on the sample pieces, I decided the tone I preferred could be obtained with the oil alone. For the solids, three coats were necessary. The veneer required four coats to achieve the same effect. For some reason, the veneer seemed to absorb more oil than the solids.

If you have never done an oiled finish, it is actually uncomplicated, but rather labor intensive. After you properly sand the surfaces, vacuum them and finish by wiping with a tack cloth. Next, liberally apply the oil to the wood using a clean rag, and wait around 30 minutes for the oil to soak in. Do not be stingy with the oil, since the wood will absorb quite a bit.

PHOTO 25: Finished terminal block.

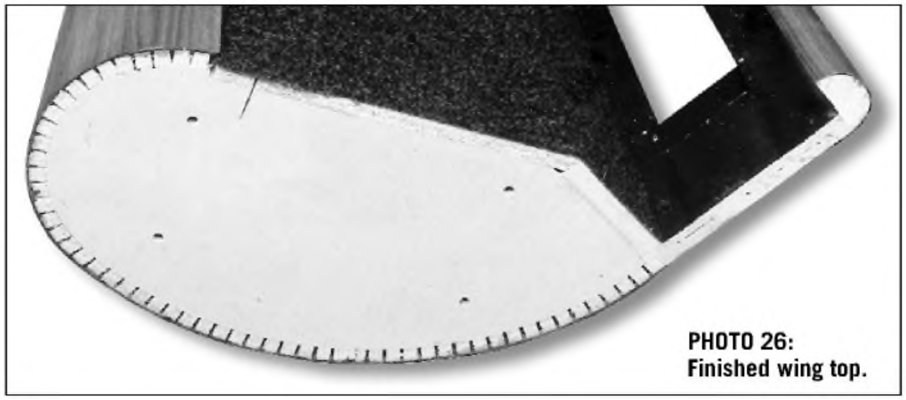

PHOTO 26: Finished wing top.

Reapply the oil and wait about 15 minutes for it to soak in. Then wipe off the excess oil with a clean rag. I usually wait two days before applying subsequent coats.

Before I apply additional coats, I sand again, using 400 wet or dry paper. Don't skimp on the sanding; all the stains, oils, and polyurethanes in the world won't make up for improperly prepared wood. If done correctly, the resulting presentation of the wood color and grain make it all worthwhile. When doing any work with oils, another very important consideration is proper disposal of oil-soaked rags; place them in a water-filled metal container before disposal. If you do not, spontaneous combustion is a very real possibility. It would be a shame to burn down your house with your newly built speakers inside.

FINAL ASSEMBLY

I began the final assembly by mounting the Vampire binding posts in the terminal blocks. I inserted the blocks into the mounting bases and then through the holes in the blocks. I then fastened them, using the nuts supplied. *Photo 25* shows the posts mounted on a block.

Before attaching the top and bottom plates and the drivers, I moved the baffles to my listening area in the loft where I planned to use them. This reduced the weight and bulk of the baffles and made it a two-person job. The drivers would have added another 35 pounds to the baffle, making the job quite a bit harder.

PHOTO 27: Mounting the top plate.

PHOTO 28: Mounting the bottom plate.

PHOTO 29: Positioning the driver.

PHOTO 30: Hanger bolt.

PHOTO 31: Mounting a hanger bolt.

PHOTO 33: Terminal block mounted.

PHOTO 32: Attaching speaker leads.

Once in the loft, I added the tops. *Photo 26* shows the finished baffle on the floor of the loft in preparation for the addition of the top. Note the four holes for the ¼″ × 20 T-nuts. The pilot holes for the wood screws are not as visible. For the tops, I used four ¼″ × 20 flat head Phillips machine screws to mate the T-nuts, and I used four #8 flat head Phillips particleboard screws to mate with the driver mount and half round.

I began the fastening by aligning the holes in the top with the pilot holes, and inserting the four ¼″ × 20 screws loosely into the T-nuts. Next, I inserted the four #8 screws through the top into the pilot holes and tightened them. Finally, I tightened the machine screws. *Photo 27* shows a top being installed.

I used four ¼″ × 20 furniture bolts and four #8 flat-head Phillips particleboard screws to fasten the bottoms to the baffles. Furniture bolts are different from ordinary bolts in that they have

PHOTO 34:
Factory passive filter.

wide, relatively thin heads. This allows the clamping force to be spread over a relatively large area, while maintaining a low profile. The heads are also round so that they won't snag carpeting if you slide the bases around for positioning. To tighten them, a recessed hex opening is provided in the center.

Photo 28 shows a bottom being fastened to a baffle. Also visible in the photo are the furniture glides I installed around the perimeter of the base. The assembly sequence is the same as for the top, except for the substitution of the furniture bolts for the machine screws.

CONNECTIONS

The lead wires of the RD75s were a little too short to reach the terminal blocks once I mounted the blocks on the base. I made extensions using #12 speaker wire and added them to the speaker leads. To enhance the appearance of the cables, I purchased a length of the split black corrugated plastic sheathing used to route wires underneath a car hood from my local auto parts store. Since it is split its entire length, it is an easy task to slip over existing wires and provides a much more professional look. I added ring lugs to the leads for later attachment to the binding posts.

A word about crimping is in order. I've heard others mention that they solder their connections instead of crimping them, since that results in a better connection. Wrong! To see really good connections, look to the power distribution business, which must handle currents in the range of thousands of amps.

At a company where I once worked, we used 500 MCM wire (that's big) to distribute power in TV studio applications. These connections required such low resistance that the main distribution bus bars were silver-plated at the points where the connectors were bolted to them. All connectors were crimped to these wires using a hydraulic crimper.

One day, just for grins, I decided to cut apart one of the crimped connectors to see whether I could find any voids. Not only were there no voids, but the joint looked like one solid piece of copper. A good crimp actually causes the connector and wire to flow together, resulting in much lower resistance than a solder joint. You are more likely to wind up with a cold solder joint than a bad crimp if you use a good crimper.

My advice is to spend money on a good compound action, ratcheted crimper and a few sets of the proper dies and crimp your connections. You'll probably spend around $100 and have a lifetime of good connections with no melted insulation.

DRIVER MOUNTING

Next, I mounted the RD75s to the baffle. I positioned the drivers on the rear of the mounting plates and made marks through the mounting holes for drilling pilot holes. I removed the drivers and

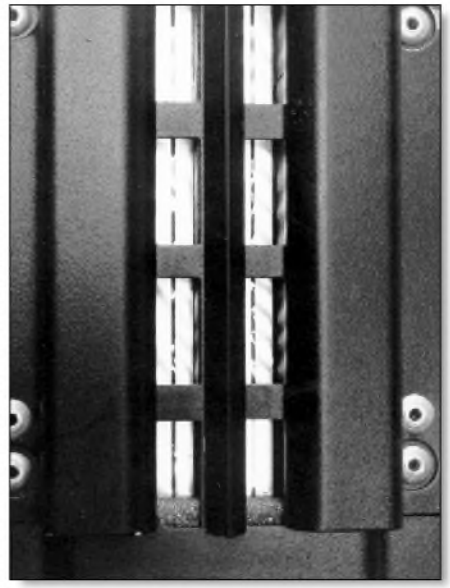

PHOTO 35: Diaphragm ripples.

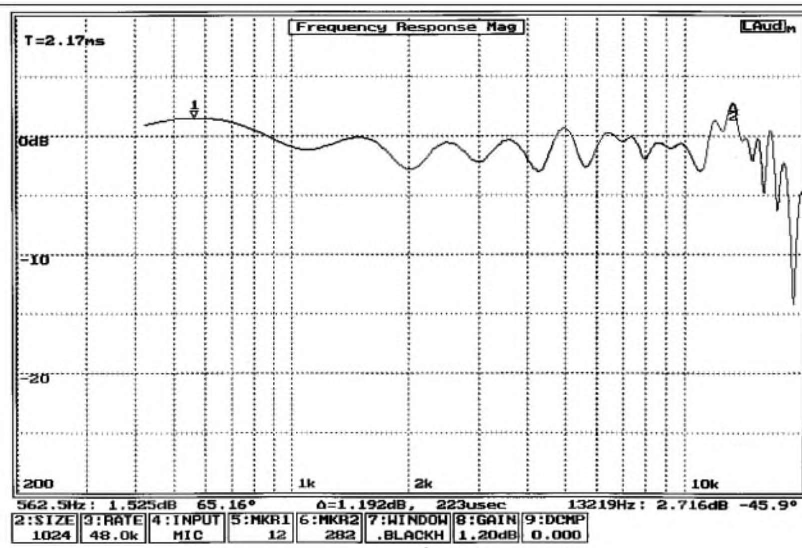

FIGURE 1: The unsmoothed 1m frequency response curve. B-1648-1

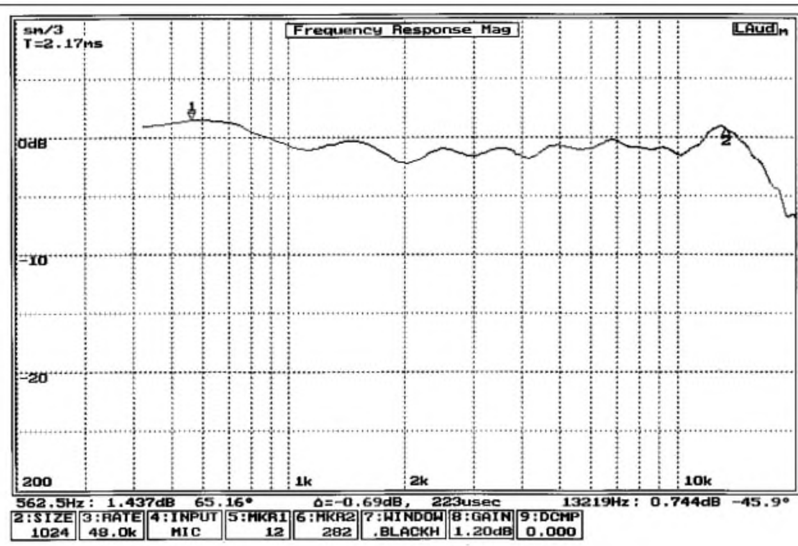

FIGURE 2: The ⅓ octave smooth response. B-1648-2

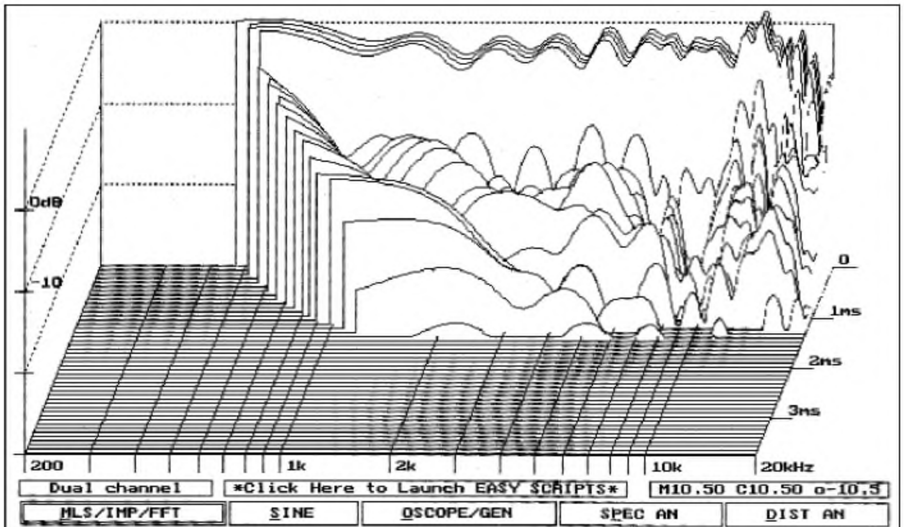

FIGURE 3: Cumulative spectral decay plot. B-1648-3

made pilot holes using a drill stop. Failure to use a drill stop could result in ugly holes through the veneer if you drill too deeply. *Photo 29* shows a driver in position for marking.

The driver mounting hardware I chose included 10 × 24 hanger bolts, #10 flat washers, #10 lock washers, and 10 × 24 nuts. Hanger bolts have a wood screw thread on one end and a machine screw thread on the other (*Photo 30*). The advantage of a hanger bolt is that you can insert a machine screw stud into a blind hole in a wood substrate. There is no need to drill a through hole as you would with a screw or bolt. Once the hanger bolt is in place, you have the advantage of a machine screw in a wooden piece; that is, you can fasten or unfasten the device without damaging the wood.

To insert the hanger bolts into the mounting plate on the machine screw end, apply a nut, flat washer, and then another nut. Tighten the nuts. Then you can insert the wood screw end of the bolt assembly into the pilot hole and tighten with a wrench. *Photo 31* shows a hanger bolt in the process of insertion.

Do not drive the bolt too far into the wood or you will have a dimple in the veneer on the other side of the hole. Once the bolt is seated, you can remove the nuts and washer by placing a wrench on the lower nut and then loosening the upper nut with another wrench. Remove the two nuts and washer. Repeat the procedure for all the bolts.

After all the bolts were in place, I carefully positioned the mounting holes in the drivers over the hanger bolts and lowered the drivers into place. The drivers are heavy. If you do this, be careful not to drop the driver on a hanger bolt. Then fasten the drivers in place.

Once I secured the drivers to the mounting plates, I fastened the driver leads to the binding posts in the terminal blocks. *Photo 32* shows the bottom of a terminal block with the wires attached. The pilot holes for mounting the block, as well as the black corrugated cable sheathing, are also visible.

The last assembly step was to fasten the mounting blocks to the base plates. For this, I used #6 flat-head Phillips wood screws. *Photo 33* shows a terminal block mounted to a base. The only thing left to do was to raise the baffles into an upright position and slide them into place.

POWER-UP

The RD75s are supplied with passive filters (*Photo 34*), consisting of an air-core choke, a film capacitor, and a resistor, that introduce a notch in the response where the drivers exhibit a cavity resonance. Since I was using active equalization and a separate amplifier to directly drive the RD75s, I did not wish to introduce any unnecessary impedance in the circuit and decided not to use the passive notch filter. Instead, I used a pair of Orban parametric equalizers to precisely correct the cavity resonance.

To do that, I used an ACO Pacific 7012 mike feeding LAUD measurement software, with LAUD used in the repeated or cycling FFT measuring mode. The center frequency, Q, and level parameters of the Orbans were adjusted until LAUD produced a flat frequency plot. Parametric equalizers and computer measuring systems can really make life easy.

BG claims a response down to 150 Hz for the RD75. To ensure high power handling (without overload due to displacement limitations), I decided to set the crossover for the RD75s at 250Hz using a 12dB/octave slope. I was already using a custom four-way electronic crossover for my then-current system, which used 7″ Eton midranges and Heil AMT tweeters to cover the range from 400Hz up.

I modified the high-pass section to provide the 250Hz point for the RD 75s and dropped the high end of the bass bandpass section from 400Hz to 250Hz. The old midrange bandpass section was not used, effectively making it a three-way crossover. This also freed the tweeter amplifier. The RD75s are driven by a modified Hafler DH500, formerly used for the Etons.

LISTENING GLITCH

The moment we all wait for had arrived. I was ready to listen. For the first selection, I used a CD that I knew would test the BGs, "Steppin' Out" with Tony Bennett. Ralph Sharon plays some great piano sections on a Bosendorfer. The intro to the second track, "Who Cares?" is especially good.

From the first note, I knew that I was in for something special. The sound of the piano was warm, dynamic, and detailed. I had never heard anything as good, short of live sound.

I was in heaven. A few minutes into the cut, however, I heard something rough on a few of the louder passages. It must have been my imagination; the rest of the piece was just great.

The next piece was Pink Floyd, "The Wall." This is a very complex album, and the BGs revealed detail I had never heard before. The sense of room space in the intro of the cut "One of My Turns" was the most realistic I had ever heard. While listening through the various cuts, however, I thought I heard occasional harsh spots.

I kept thinking to myself that it could not be. I spent all this time, money, and sweat to produce these masterpieces; there couldn't be a problem. Not being one to stick my head in the sand for extended lengths of time, I decided to do some quick tests. I ran some separate sine-wave sweeps on each driver.

This can be a very revealing test, and reveal it did; both drivers had rather strong resonances at multiple frequencies starting at 110Hz in one sample and 470Hz in the other. I noticed that both drivers had what appeared to be ripples in the diaphragms (*Photo 35*). I wasn't sure whether these ripples were the problem, but it was time to call Rudi. He was surprised as he had never encountered this problem before, but immediately agreed to replace them. BG sent out replacements, and I packed up the originals in the same carton and returned them.

The new drivers showed no traces of the ripples that were in the first samples. I ran sweeps again and noticed no resonances (other than the expected cavity resonance characteristic of the driver). After mounting the drivers in the baffles and putting them back in position, I put the Tony Bennett CD back in, sat down, steadied my shaking hands enough to hit the play button and waited the hours (read two seconds) for the piece to begin playing.

Getting through each note was agony. Everything seemed like a slow-motion nightmare as I waited for the bad sounds that I had heard before to re-appear.

There was nothing except glorious sound! The occasional bad sounds I heard before were gone. What was left was the most detailed, dynamic sound that I had ever heard, at whatever level I tried. The sound stayed clean up to the point that the custom speaker protection network that I use disconnected the drivers, and that was in excess of 110dB.

CRITIQUE

Normally, at this point I would describe the characteristics of the sound that I heard with various types of music. However, thinking about it, that doesn't make sense in this case. Regardless of the music I listened to—be it rock, classical, folk, blues, jazz, new wave, or whatever—a common set of characteristics came through.

First, there was a sense that you were in the presence of the performers. Subtle details that were masked by my (very good) previous system were now present. The effect was quite different than the sense of forwardness you get when the frequency balance is skewed toward the highs. Rather than parts of the sound hitting you in the face, there was a sense of texture coupled with smoothness that presented a more complex, yet more understandable, rendition of the performance. Whether it was a flute, trumpet, harp, violin, piano, voice, synthesizer, or effects recording, the effect was the same—greater realization of the subtleties of the acoustic output being produced by the source.

Second, that transparency was available at all levels, not just at middle levels, as often happens with some drivers. Whether listening at whisper levels or full rock blast, the sound never became congested. The ability to have this kind of dynamic range does wonders for establishing a lifelike listening experience.

Mini monitors need not apply in this category. This has to do with the volume displacement issues that we often think of when trying to get good bass. However, volume displacement is an issue at any frequency.

The volume displacements required for a given SPL decrease with increasing frequency, but as the SPL requirements increase, so do the volume displacement requirements at all frequencies. Small area mid- and high-frequency drivers need more displacement to meet the demands of high SPLs. This can place them out of their linear operating range. The large surface area of the RD75s means less displacement for a given SPL, or conversely, the capability to reproduce higher SPLs before reaching a nonlinear operating range.

Third, the sense of space and placement was exceptional. I've often heard that dipoles can't image. Hogwash! Place a dipole close to a rear wall where the back wave output is perceived as early reflections and you'll have problems. However, placing any speaker near a wall, or for that matter near the ceiling or floor, is looking for trouble.

Dipoles have an advantage—there are nulls to the sides, reducing problem reflections from walls. The RD75 also behaves as a line array because it is so long with respect to the wavelengths being reproduced. This results in a cylindrical wave launch instead of the spherical wave launch that you get from a small dynamic driver. The severe cutoff of output above and below the long axis of the RD75 very effectively reduces floor and ceiling interactions.

I took measurements of the finished RD75/baffle combination with LAUD. The un-smoothed 1m frequency response curve is shown in *Fig. 1.* The response is very smooth up to 13kHz, but beyond that, there are some ripples with a decrease in response beyond 16kHz. The commonly reported ⅓ octave smoothed response is shown in *Fig. 2.* Except for the 13kHz bump, the response is generally within ±1.5dB out to about 15kHz.

The cumulative spectral decay plot is very interesting (*Fig. 3*). I have measured a fair number of panel drivers, and most do not show a very good spectral decay plot, although most sounded quite detailed. This one is actually especially good, with almost nothing beyond 1ms.

If the spectral decay plot of a panel driver is not indicative of the detailed sound generally attributed to such drivers, what is the reason for this sound? My guess is that the controlled directivity caused by the large size, and often dipole configuration, result in fewer early reflections from nearby surfaces and this is interpreted as greater detail. Certainly, I have had other conventional drivers that had spectral decay plots that have looked as good as the RD75s, but did not have the apparent detail.

COST

What was the final cost of these speakers, excluding my labor? Just over \$1700. That may seem like a lot, but considering the RD75 forms the basis for speakers in the \$15,000 to \$125,000 range, I consider it a bargain. The costs break down as follows:

1. Two RD75 drivers from Audio-X-Stream including shipping	$895.00
2. One 4 × 8′ sheet of ¾″ MDF	30.00
3. Two 4 × 8′ sheets of Kerfkore	257.08
4. Two 4 × 8′ sheets of walnut veneer	210.30
5. Solid 1″ walnut planks	131.16
6. Latex-backed carpet	21.05
7. Miscellaneous glue, hardware, tape, oil, foam, etc	95.00
8. Two pairs of Vampire BPHEX binding posts	63.00
Total	$1,702.59

Having lived with the RD75s for quite some time now, I can honestly say that they have come closer to my idea of a "perfect" speaker than anything else I've ever had. So where did the title of this article come from? If you heard them you would think that the music was so true and pure that it came directly from heaven on angels' wings. In this case, the wings are man-made, but the results are the same. ❖

Designing Your Own Amplifier Part 6b: Special Output Circuits

By Norman H. Crowhurst

This article originally appeared in Audiocraft, *March 1957. ©1957 by Audiocom, Inc.*

A variant of the pentode cathode-follower circuit is the unity-coupled circuit used by McIntosh, and shown schematically in *Fig. 11.* Half the output load is put in the plate circuit and half in the cathode circuit, using an equal number of turns in each. By cross-coupling the screens, the screen-cathode potential is maintained constant during the audio cycle, as with the straight cathode-follower pentode.

It is important that the screen be tightly coupled to its corresponding cathode. For this reason the screen-and-plate winding is wound bifilar with the cathode winding; that is, the turns of both are put on side by side at the same time. Good insulation is needed between these individual turns, since the voltage between them is full B+. DC voltage on the screen will be the same as the plate voltage, as in the ultralinear method of operation.

The drive swing will be a little more than half that required for the pentode cathode follower, since it will be made up of half the plate plus the full grid swing.

The reduction in harmonic distortion would also be about half that obtained in the cathode follower, resulting in about 0.5% without any overall feedback. Source resistance will also be about double that of the full cathode-follower pentode circuit.

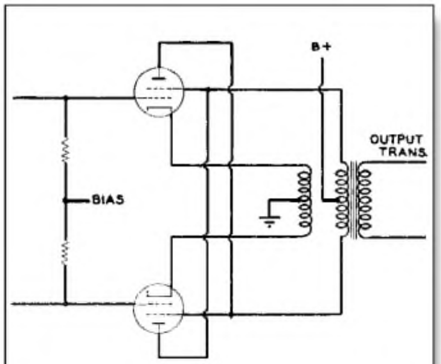

FIGURE 11: Schematic of the so-called unity-coupling circuit. Here the tubes are operating as pentodes or tetrodes, with half the amount of feedback achieved by the cathode-follower circuit, *Fig. 7.*

CATHODE-COUPLED ULTRALINEAR

Another variant may be regarded as an ultralinear circuit with partial cathode feedback. This is shown schematically in *Fig. 12.* The proper fraction of swing for the screens is obtained by this cathode winding, and the screens themselves are coupled to a B+ point. This enables a lower B+ to be used on the screens than on the plates, without the necessity for a separate screen winding. The circuit is used in the Bell 2200-C amplifier and also several in the Bogen line.

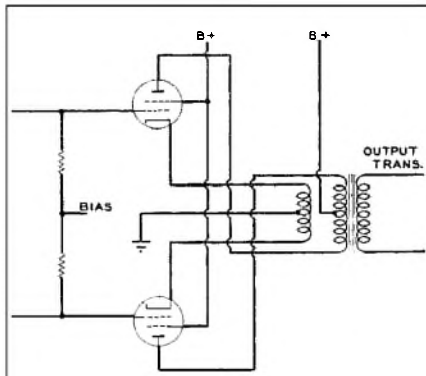

FIGURE 12: Another variant is this virtual ultralinear arrangement, employing partial cathode coupling. For 43% equivalent tapping, the cathode winding should have 43% of the total equivalent primary turns, while the plate winding should have the remaining 57%. This arrangement enables the plates and screen to operate at different DC potentials for increased power, if desired.

CIRCLOTRON

The last circuit we shall consider in this article is the Electro-Voice Circlotron circuit, shown in *Fig. 13.* Batteries are shown for the two separate B+ supplies necessary with this arrangement.

This circuit is named for the loop (or circle) consisting of the two output tubes and the two B+ supplies. When the bias on each output tube is identical, which occurs at the quiescent condition, the current around the entire loop is uniform. But when one grid is driven positive and the other negative, the currents passed by the two output tubes differ, as do the proportions of total voltage drop across each tube. Then the transformer connected between cathodes (which previously were both at the same potential) carries a load current.

The tubes here are acting as pentodes because cathode and screen are separated in each case by a constant potential. They are virtually in parallel, each cathode being connected to the other plate, with the output transformer being connected across the whole combination. This is where this circuit differs from the normal push-pull arrangement, in which the two tubes virtually feed the load *in series.* The result is that the plate-to-plate (or cathode-to-cathode—whichever you prefer to call it) load has a value one quarter of that proper for the normal push-pull output.

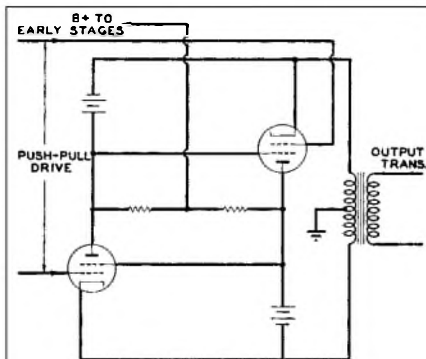

FIGURE 13: A simplified schematic of the Circlotron circuit. Batteries have been shown in place of the two separate B+ supplies necessary for the system, both of which are isolated from ground.

The B+ supply for the earlier part of the amplifier is achieved by putting two resistors in series between the two B+ points and taking the supply from the center point. Ground is provided by the center tap of the output transformer.

If the output-tube grids were returned to ground, the arrangement would be somewhat analogous to a cathode-follower circuit, because the drive at the grids would have to provide the grid swing in addition to the output swing on the cathodes. This is partially offset in the Circlotron circuit by returning the B+ for the push-pull drive stage to the positive voltage from the opposite output tube in each case. It is difficult to know whether this should be called a positive feedback or a reduction in negative feedback.

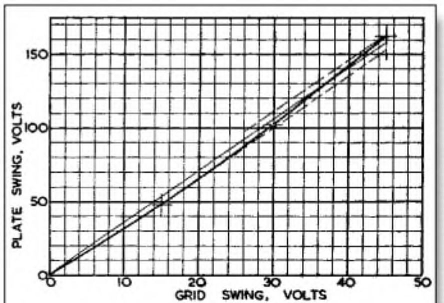

FIGURE 14: Estimating the distortion of 5881 tubes as triodes from the transfer characteristic: the peak fundamental is about 158V, while the peak harmonic is about 5.5V; for this the peak ratio represents 3.5% harmonic.

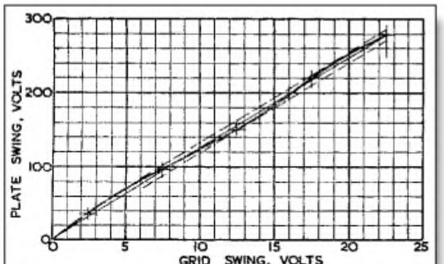

FIGURE 15: The distortion estimated for the pentode (tetrode) operation: peak fundamental is 280V, and peak harmonic 8V, for about 2.85% total.

ESTIMATING DISTORTION

So much for a number of popular varieties of output circuit. One thing more is needed to complete this article: a graphical way to estimate the distortion. This is illustrated in *Figs. 14, 15,* and *16* for the triode, pentode, and ultralinear operation of the 5881 shown in Figs. 2, 6, and 9 of Part 6a.

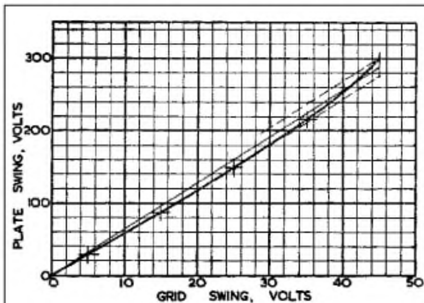

FIGURE 16: Estimating distortion for ultralinear operation, from the transfer characteristics: peak fundamental 288V, peak harmonic 11V, and total harmonic distortion, 3.8%. These figures give peak summation in each case, and so differ from the RMS text analysis.

In each case the transfer characteristic is plotted, plate swing vertically against grid swing horizontally. Only one half of the characteristic is plotted because, being push-pull, the two halves are symmetrical. The negative swing would be exactly similar, but to the left and downward from the zero points.

The curve is plotted and then a straight line is drawn from the origin to pass through the curve in such a way that the excursions above and below the straight line (as determined by further dotted lines parallel with it) are equal on both sides.

This method of measurement does not determine the relationship between RMS harmonic and RMS fundamental, but between peak harmonic and peak fundamental. If only one harmonic is present, the two figures will, of course, be identical, but invariably more than one harmonic is present. Then the usual trend is that the peak relationship gives a higher percentage harmonic than the RMS relationship.

Although this means that the figure obtained from the construction will not agree with that obtained by the usual forms of measurement, it does mean that it gives a better practical indication of the annoyance value of the distortion, because multiple harmonics—especially high-order ones—are much more annoying than a single lower-order harmonic.

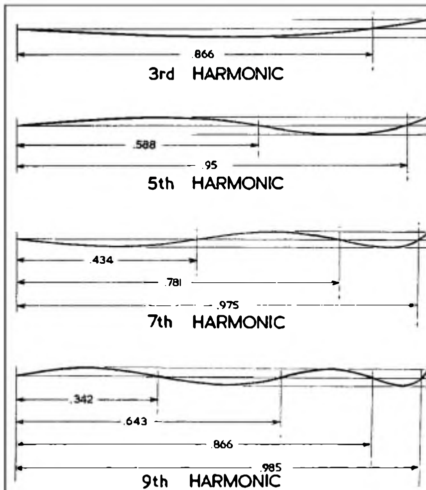

FIGURE 17: This chart is to aid in identifying the various odd-order harmonics present from the shape of the transfer characteristic. The fractions give the points at which the curve should cross the straight fundamental line, when only that harmonic is present in the signal.

Figure 17 is given to aid in identifying various harmonics from the third to the ninth, assuming they are present as separate entities. Here the fundamental is represented as being removed and the deviation shown horizontally.

Compare *Fig. 16* with *Fig. 17*: the point where the curve crosses the center line should be 0.866 of the total excursion for the harmonic to be pure third. In *Fig. 16* the crossover point is a little nearer the top than this, meaning that there are some higher-order harmonics, although quite small in amplitude.

On the other hand, the crossover point shown for triode characteristics in *Fig. 14* is considerably lower than 0.866, indicating in this case that there are probably components of fifth, seventh, and ninth.

The multiple curvature of the transfer characteristic in *Fig. 15* can only be explained by the presence of harmonics at least up to the seventh.

CONCLUSIONS

Now, having studied different kinds of output circuits somewhat, are we in a better position to assess which kind of output—triode, pentode, or ultralinear—is the best one to use? Perhaps it would be more accurate to say that now we can see why there is so much confusion on the issue and why different individuals prefer different circuits.

From our previous discussion of amplifier design we learned that the circuit should be designed to give the lowest possible distortion before any feedback is applied (and if possible this distortion should be of low *order*). Comparing the three methods of operation for the 5881:

The pentode seemed to show the lowest distortion, but the dominant components were third and seventh. Additionally, the fact that the load line goes across crooked characteristics means that any load deviation from a straight line of specific value will cause considerable variation in the distortion produced.

The ultralinear circuit, with 3.3% almost pure third-harmonic distortion, and characteristics whose shape will not cause appreciably increased distortion when the load line deviates from its particular resistance value or goes reactive, is the most practical circuit of the three.

The triode-connected arrangement strikes an intermediate position, with 3.3% third- and 1.25% fifth-harmonic distortion. Variation of resistance value over a limited range will not seriously affect the distortion percentage, but reactive components are a little more likely to do so.

The unity-coupled arrangement achieves a high output from a pentode without the same loss of gain that would be necessary when cathode coupling. Because of the extremely tight coupling between cathode and plate-screen circuits, it is possible to operate this amplifier into the grid-current region and still maintain a low order of distortion.

The partially cathode-coupled ultralinear circuit is another good variant, but requires careful attention to transformer design.

The Circlotron obviates the necessity for highly critical design of the output transformer. What could prove to be a further liability in this arrangement is the fact that the whole of the two B+ supplies constitute part of the output circuit. This means they can contribute to the frequency response or stability characteristics of the entire amplifier. ❖

Xpress Mail

PHONO PREAMP

As a regular reader of *GA*, I have enjoyed your magazine and built several projects straight from the articles, including my own phono preamp. My goals were: two stages, no global feedback, and low output impedance. The starting point was a textbook two-stage, passive-equalization circuit that uses no electrolytics.

I replaced the usual cathode-follower output stage with Rickard Berglund's asymmetrical mu follower (*GA* 1/93, 1/94). The low output impedance allows the use of a 10kΩ volume pot. I optimized network resistors R6 and R7 for correct response using Norman Koren's SPICE techniques (*GA* 2/97, 4/97). The overall gain is 36dB at 1kHz.

The supply voltages in *Fig. 1* are not critical and represent the working voltage after soft-regulating 450V down to 400V followed by 5kΩ pi filters. The 6SN7 draws 6mA and the 12AX7 sections draw 1.5mA each.

Many possibilities for variation exist. Bypassing the cathodes on V1 will increase gain by 4dB. In the first stage, you can use other low-noise, high-gain tubes, such as the 6SL7 for an all-octal preamp. I encourage other builders to incorporate their favorite tubes.

I chose the 6SN7 for its combination of low noise and high plate dissipation. Rickard Berglund's articles have many more examples. Be careful not to exceed the heater-to-cathode limit on the upper section of the asymmetrical mu follower. SPICE makes it easy to check this and other "what if" scenarios quickly.

Jim Ryan
Baltimore, Md.

GROUND LOOP FIX

I just finished reading the Jensen ISO-MAX product review (*AE* 6/00). Jensen undoubtedly makes fine products, but it's possible to cure CATV/antenna ground loop (hum) problems at far lower cost than the $50 price of Jensen's ISO-MAX CATV Ground Isolator.

I've done this many times by simply connecting a couple of 10nF or 4N7 ceramic disc capacitors in series with the shield and center conductor of the coaxial cable to the antenna input connector of an FM tuner or receiver. It's important to keep capacitor lead lengths near zero, and choose capacitors marked 500V or higher for the sake of reliability.

For a portable capacitive isolation device, I mount a pair of female F fittings in opposite walls of a small plastic box and bridge them together with the capacitors. Lead length for the shield connection can be excessive in this case, so I use long flat ground lugs to reach the capacitor body with minimum lead inductance.

Mike Hardwick
Decade Engineering
Turner, Oreg.

PREAMP DESIGN

Robbert ter Haar's interesting phono preamp project ("The True Realism Preamplifier," *GA* 4/00, p. 18) is one that I may well chose to build, but there is one critical issue that was not addressed. What is the gain? To the experts in electronics, I'm sure it is obvious. It isn't to me. I need upwards of 60dB for my cartridge.

G. Max Carter
Cascade, Colo.

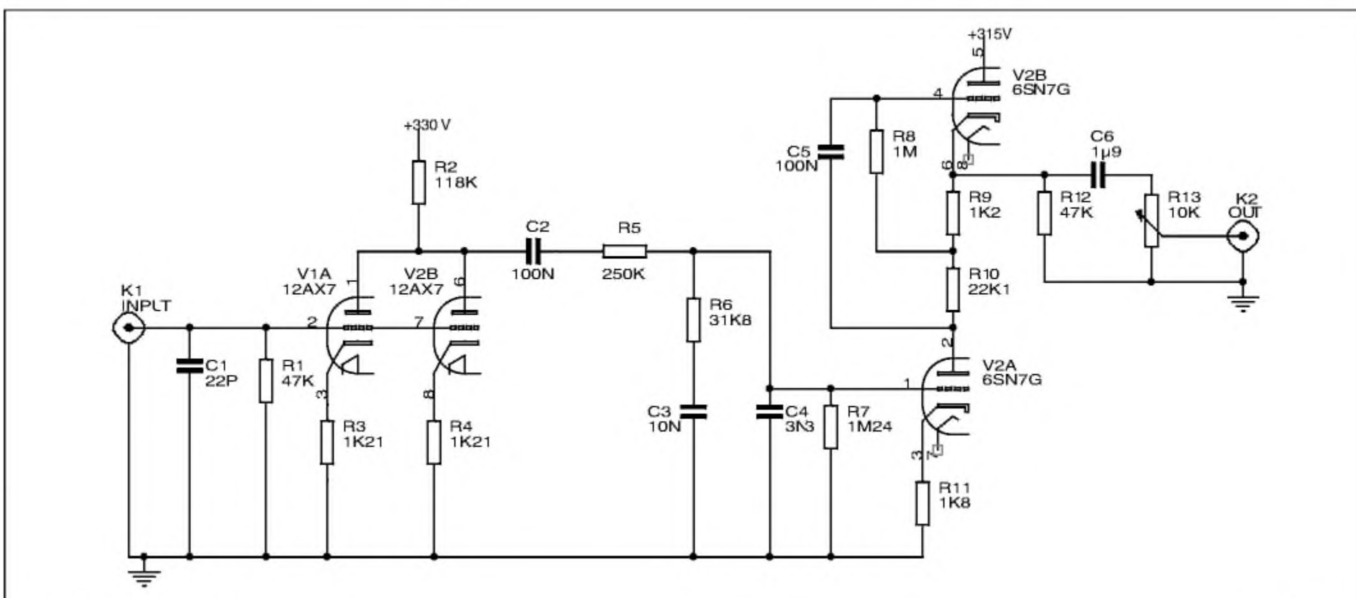

FIGURE 1: Phono preamp circuit.

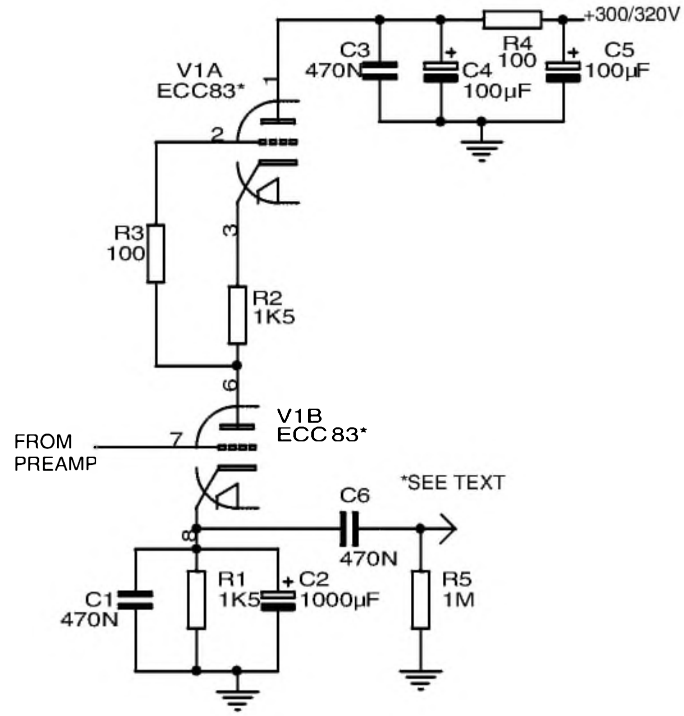

FIGURE 2: Line amp circuit.

A-terha-2

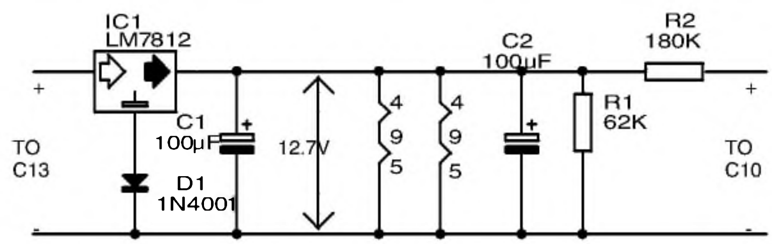

FIGURE 3: Stabilization section.

A-terha-3

Robbert H. ter Haar responds:

Thank you for your interest in my unorthodox preamp design. The gain of the amplifier depends on the tubes you use. With the ECC83 you can expect a total gain of around 46dB; for the ECC81, you will have a gain of approximately 41dB.

I am somewhat surprised about your need for a 60dB gain in the preamp. I am using the DENON DL-103LC II, which has an output voltage of approximately 0.25mV (1kHz 50mm/sec). The output voltage of the preamp is now almost similar to my CD player.

To make an extra gain of approximately 20dB, I suggest putting the line amplifier in Fig. 2 *after your preamp. The tubes for this design are the ECC82s. As you see, you can connect this section on the same power supply as the preamp. For the filaments, you need to add a third stabilization section (*Fig. 3*).*

It is also advisable to increase the filament current transformer from 2.5–3.5 amp to reduce the heat of the transformer and therefore extend the life of all components around this transformer. Please also refer to the power supply schematic (Fig. 5) in my preamp article.

Mr. ter Haar has written a very interesting article. However, he has no voltage ratings on the capacitors. For example, can I assume C3 and C4 to be low voltage? My caps are 470µF/35V. Is this okay?

Hans Olofsson
Skelleftea, Sweden

Robbert ter Haar responds:

Thank you for your interest in my unorthodox preamp design. Unfortunately, due to my eagerness to publish this article and schemes so other DIYers could also enjoy the nice sound, I forgot to mention the voltages of the capacitors. They are:

For the amplifier—400V rating for C1, C4, C5, C6, C7, and C11; 250V rating for C8, C9, C10, and C12; and 35V rating for C2, C3, C13, and C14. Ci does not have a voltage rating, since the total voltage applied will be the one from your cartridge or pre-preamplifier.

For the power supply, the voltages are: 400V rating for C5, C6, C7, C8, C9, and C10; 125V rating for C11 and C12; and 35V rating for C13, C14, and C15.

Readers should also note the following: the parts list for the power supply is incorrect. R3 is mentioned twice. The values noted in the Fig. 3 schematic are the right ones. Although clear in the drawing, do not ground the 12V filament power supply, since this one is connected with the high-voltage power supply with the resistor bridge 180k and 62k.

TUBE MAKEUP

I would like to add to the discussion regarding electron tube getters in a letter from John Caruso and a response from Eric Barbour (*GA* 3/00, p. 56).

The most common getter is the evaporative-type containing barium in an iron ring. The barium content may be 30%–50% and is stabilized with aluminum and sintered to form an agglomeration. The ring is formed with a trough to hold from 1–5 mg, and a nickel wire post is usually attached for mounting. The final two positions of the pump station activate the barium: first to degas by RF heating at 700°C, and the last to vaporize the barium for 5–8 seconds at 900°C.

During vaporization, O_2, H_2, CO, CO_2, H_2O, and CH_4 are absorbed. After condensing on the glass, more gas is absorbed, and this action will continue during tube life. During flashing, there is a significant drop in pressure which is maintained by tipping off within seconds; otherwise, it would rise back to the pump's manifold pressure. Initial gas from glass walls, cathode/filament coating, and metal parts are removed by heating and pumping before flashing.

Getters flashed slowly will have a uniform bright silver color. Those flashed rapidly will diffuse and show several colors. A darker flash has more absorption capacity at high pressure (1mm Hg), but at low pressure (10^{-6}mm) color is not a factor and a uniform silver color on a clean, relatively cool glass surface is desirable for appearance. A flash in argon gas is black due to light absorption and is not an indication of contamination.

Getters do not absorb the noble gasses—argon, neon, krypton, xenon, and helium. A barium flash tends to darken slightly after many hours of use and the rough feathered edges will become smooth and more sharply defined. Quite often, in a used tube, residual barium is left in the trough and can be reflashed.

Overheating may burn the ring and spew metal particles.

The u-shaped ring shields and directs the vaporized barium away from tube elements, since a deposit on the mica is conductive and may cause inter-electrode leakage and capacitance changes. A large flash may reduce power dissipation by reflecting heat back to the plate. A barium flash in a bad tube that has lost vacuum will turn milky white and peel from the glass. A getter touching the bulb will check the glass when heated and cause a slow leaker.

A tube with no flash may test okay since surfaces inside—properly degreased and heat-treated in hydrogen—will absorb, and a getter may not even be needed. However, the getter is better at absorbing "deep" gasses that evolve slowly during long life and will protect the cathode/filament coating from emission poisoning. The life of a tube made with high-purity nickel parts was considered to be 40k to 100k hours or more depending on operating temperature. The main killers of a tube are improper voltages, oxygen, sulphur, heat, arcing, and stem leakers.

Zirconium hydride, a reactive absorber, can be sprayed as a coating on the anode. The Batalum getter (barium beryliate) produced a barium coating by chemical reduction. The Kic getter was metal-clad and produced a reactive coating. The pan getter was a dimple on a thin metal flag filled with magnesium and produced a white silver flash. Magnesium was also used in mercury vapor tubes since barium exposed to mercury released unwanted gasses. Barium for gettering is gray metallic, while barium for cathode/filament emission is a white powder carbonate.

Kemet and SAES brands were popular for barium ring getters. They were packaged in vacuum-sealed cans and, after opening, were stored in a desiccator to prevent rust.

Bernard Magers
Lees Summit, Mo.

John Caruso responds:

Bernard Magers' comments on the subject of radio tube getters are noteworthy and provide deep insight to the physics and chemical aspects of this subject. I am inclined to think that he has probably been very much involved in getter technology and practice during tube manufacturing.

My opinion and past experience as a layman in this field is that a tube's health cannot be estimated based on the appearance of getter flash except if it is milky white. The final judge, then, must be the electrical results of a good tube tester.

I would also like to thank Eric Barbour for his comments in response to my letter in GA 3/00 regarding "Old Wives' Tale" about getters. Mr. Magers and Mr. Barbour have both put the true facts of this old art into prominence.

SYNCHRONOUS RECTIFICATION

I wish to comment about the article by Michael Kornacker in *AE* 5/00, p. 26 ("Try Synchronous Rectification"). When simulating Fig. 2 using Analog Workbench (Cadence), I get large RMS currents in the filter cap as the MOSFETs conduct both ways. Using a 5000µF filter cap, 19V peak in T1 and T2, and IRF610 MOSFETs, I get 3.3A RMS in the MOSFETs and 4.7A RMS in the filter cap. As this could probably go unnoticed, I thought I had better warn you about it.

If you make a supply suitable for a power amplifier (2000µF, T2 = 19V peak, T1 = 50V peak, and IRF150 MOSFETs), it will blow up, 47 and 67A RMS.

Lars Sodergren
Goteborg, Sweden

Michael Kornacker responds:

Thank you, Mr. Sodergren, for responding to my article. Unfortunately, I cannot talk very well about the problems you raised, since I'm not at all familiar with the simulation package you are using. I would like to suggest that you double-check that you are observing the correct phase relationship for the different transformers, as this is the fundamental requirement for successful operation of a synchronous rectifier.

Make sure that AC voltages to the source and gate occur at the same time or phase as referenced from the center-tap ground, and that the gate voltages are greater than the source voltages at the same phase by at least 10V peak to less than 20V peak.

I also noticed that the schematics of some of the MOSFETs seem to be drawn backwards in the figures but the G, S, and D labels are correct, so go by them.

After saying that, all I can say is to get the literature I used as a reference that is listed at the end of the article. You should find a wealth of information and compelling evidence for the validity of the SR concept. And then, build the circuit for yourself and see whether it works or not. That's the ultimate test. I think we should always rely on an actual circuit instead of a virtual circuit. Simulation software might have its place but it cannot replace breadboarding.

Since the article was published, there is one other thing I *have realized that I would like to pass along. Ideally, in theory, the MOSFETs would rectify more efficiently if the gate voltage was more like a square wave than a sine wave. This way, the gates would turn on and off instantaneously like a switch as the source voltage increased, decreased, and reversed direction, in the sine-wave manner.*

One way of making the gate voltages from 12a and b look like a square wave is to use transformers with higher voltage secondaries (greater than 48V peak) and then use appropriate dropping resistors and 12V zener diodes to get it back down to acceptable levels for the gates. The gate voltages would then look trapezoidal (pretty close to square) and would effectively turn the MOSFETs on and off quicker as the source AC voltages rose, fell, and changed direction. This way the MOSFETs would go into and out of saturation quicker and no part of the AC voltage would be cut off.

AVAC DEBATE

I am really impressed with the dispute between Mr. Hoolhorst and Mr. Wheeler in *SB* 7/00 ("*SB* Mailbox," p. 42) on the speaker air volume-acceleration subject. For my part I will not dive deep into formulas or close scrutiny.

As a speaker builder for more than 20 years, I do trust my ears and experience, and in so doing I would follow the arguments of Mr. Wheeler. In comparing a "slow" Dynaudio 15W75 with a "fast" Focal Audiom 7K (both indeed excellent drivers for their special application), Mr. Hoolhorst did not mention the cone excursion. To catch up to 9dB of sensitivity, the Dynaudio must go to a higher level of cone excursion.

To reproduce complex music, a driver's cone needs to follow extremely fast signals. Lesser cone excursion (forward and backward) at a given sound level means faster reaction to the music signals. Remember the excellent resolution of big electrostatic panels. Therefore, I cannot appreciate the idea of recent driver development which attempts to persuade a 7″ or 8″ driver to reproduce 20Hz from a small cabinet at the cost of heavy cones, extremely low resonance frequency, and unbelievably increased cone excursion.

To get the highest levels of volume displacement mainly due to cone excursion is the wrong path. That is why I intended to go back to the roots: horn systems!

Modern spherical horn shapes and the new age of digital control gives you the opportunity to eliminate the old disadvantages of horn systems (coloration, distortion, poor time coherence) and gain impressively dynamic performance. An audio analyzer measuring a 30Hz sine-wave signal does not know how the signal was reproduced, but our ears have the capability to appreciate the differences.

It's no wonder that in Germany's hi-fi publications big horn systems are at the top of the ranking, not because of their smooth sound, their adjusted linearity or deep bass, but because of their explosive dynamic. They call it "better than life."

Although I cannot prove it by physical formulas, it seems to me logical that this is caused by the acceleration behavior coupled with the least possible cone motion.

Heinz Jaskolka
Velbert, Germany

Otto S. Hoolhorst responds:

Many professional, semiprofessional, and amateur speaker designers attempt to correlate measurable driver characteristics with their perceptions of the acoustic pressure waves that result when music signals are applied to those drivers. While valuable insights can be gleaned from such attempts, I am convinced that most of the plethora of variables involved in the process chain from electric signal input to human auditory perception are largely misunderstood. This is especially so when you take into account the complexities of listening environments.

Most audiophiles have a "reference," which is a practical ideal that they refer to for comparison. My reference is my pair of Sennheiser HD580 headphones. To date, regardless of all the money I have spent on my amateur speaker designs, the Sennheisers sound better than any of my speaker systems do. This does not disappoint me much, because the Sennheisers sound better than any commercial system I have listened to (this excludes products in the "looney" price ranges).

At home I listen loud, for many hours at a time. I have at times listened to more than ten CDs in a row. As a former professional rock guitarist and singer, music is my major recreational pastime, and my taste ranges across jazz, blues, rock, classical, ethnic, and so on. That does not mean that I know more about music and speakers than most others do.

I am simply confirming that the experience of music is my purpose. In that context, I see theory as no more than one of a range of tools that can facilitate (albeit limited) understanding of the complex mechanisms involved in reproducing recorded sound.

Messrs. Jaskolka and Wheeler obviously both share my passion for better music-reproduction systems. In response to Mr. Jaskolka's claims about cone excursion, of course, the Dynaudio will have greater cone excursion than the Focal at any reason-

able, specified acoustic output, simply because the Focal has a greater effective cone area (Sc). That does not mean that the Focal will provide a more accurate rendition of the music signals applied to it.

Yes, the Dynaudio is much less efficient than the Focal, but it is not struggling to "catch up" with the Focal. This is because the Dynaudio is not that much less efficient than most speakers are, and within its bandwidth, can relatively effortlessly produce an average acoustic output of over 100dB on a continuous basis, without compression. Such output is more than sufficient for most people in domestic listening situations.

On the point of cone excursion, lesser cone excursion does not mean faster reaction to music signals. The nature of a driver's impedance, in particular the reactive (inductance) component, as well as mass and suspension stiffness and damping, all have an effect on moving assembly acceleration. Furthermore, acceleration capability alone does not guarantee accuracy. Mr. Jaskolka will surely know that the greatest enemies of accuracy in speaker systems are:

1. resonances of driver components and driver enclosures
2. moving system undershoot and overshoot
3. unevenness of diaphragm driving force capability along the axis of diaphragm travel.

Acceleration potential of a driver moving system does not mitigate the detrimental effects of these enemies of accuracy in reproducing recorded music. After all, acceleration equals force divided by mass:

$$a = \frac{F}{m} \quad .$$

Distance travelled is not part of the equation. If you choose to explore the effect of distance, you must look at the equation for velocity. Velocity equals distance divided by time:

$$v = \frac{d}{t} \ .$$

With a speaker driver moving system, lower excursion within a given period of time requires a lower peak velocity, which requires lower force, and lower acceleration occurs. However, having a much higher acceleration capability than will be utilized does not give better acoustic performance. The "speed" of a driver moving system reproducing music signals within its design bandwidth is not determined by its air volume acceleration capability (AVAC), but by the nature of the input waveform.

AVAC cannot be a limiting factor within the design power bandwidth because that bandwidth is partly a function of the AVAC. In other words, the designer has provided sufficient AVAC to guarantee the power bandwidth. Only when the given period of time for diaphragm travel reduces dramatically at the upper end of a driver's practical bandwidth (the "extremely fast signals" that Mr. Jaskolka refers to) is a driver's AVAC likely to approach full utility. This is because as frequency rises the need for higher peak velocities is not sufficiently compensated for by the diminishing cone excursion requirements. This point needs to be clearly understood.

Moving system excursion at the upper extreme of a driver's bandwidth, even at its full rated power, is vanishingly small compared to the excursion at the lower extreme of its bandwidth. Yet AVAC can only practically become the limiting factor at the upper end. Ultimately, AVAC is only one of the factors, along with moving system mass, damping, inductance, diaphragm geometry, and so forth, which combine in various ways, depending on the design, to limit a driver's frequency response. It must also be understood that high air volume acceleration capability is wasted if strategies are not in place to keep moving system undershoot and overshoot within acceptable limits.

On the subject of electrostatic panels, once again, short diaphragm travel is not the reason for the cleaner sound of good electrostatic speaker systems. Instead, a combination of relatively extremely low diaphragm mass and the damping effect of the relatively much more significant air load prevents electrostatic panel diaphragms from undershooting and overshooting to the degree that electrodynamic driver diaphragms do. The short excursion is simply a function of the dual (and conflicting) requirements of keeping the diaphragm very close to the polarizing stators in order to have sufficient driving force and to avoid "sparking" when the electrically highly charged diaphragm stretches too close to the polarizing stators.

The small excursion capability is part of the reason that such huge electrostatic panels are required for reproducing bass frequencies. I wouldn't use them below 100Hz, but I'm certainly a fan of electrostatic panels. In fact, one of Australia's most experienced panel designers and manufacturers has been experimenting for nearly a year now with the aim of producing a custom mid/high panel for my home theater system.

My requirements were a panel for a crossover frequency of between 250Hz and 600Hz, similar sensitivity, dispersion and maximum output as the best dynamic speakers but using the lightest practical Mylar diaphragm film available. He's getting close to meeting my requirements, but so far, not close enough. I may need to wait much longer, but I am confident I will get what I wish.

On the subject of cone excursions at bass frequencies with smaller bass drivers, most people don't realize that almost all such drivers grossly exceed their linear travel at common low bass tones down to just above 40Hz (even worse if you go lower). Many 12″ drivers also suffer from this problem. This means that the peaks of the waveforms of these low tones and the higher frequencies superimposed on those peaks are distorted.

To avoid this problem, do a computer simulation of cone excursion at bass frequencies, using "Bassbox," for example, so you can see the actual cone excursion (up to 200Hz) at various power inputs. Check that cone excursion remains within the range of the driver's specified linear cone excursion for power inputs up to its rated nominal power, or better still, its IEC short-term maximum power.

Personally, I wouldn't use the Dynaudio 15W75 for full-range bass, though Dynaudio aficionados have disagreed with me, citing that the 15W75 magnetic circuit design is such that the voice coil is effectively underhung and distortion does not rise dramatically as it exceeds its specified linear excursion. Nevertheless, I would cross it over to a big woofer no lower than 80Hz. It is not that smaller drivers cannot reproduce the lowest audible frequencies, it is just that most of them do not have the linear excursion *to enable them to do so without very high levels of distortion.*

On the subject of horn systems, once again, higher efficiency and lower diaphragm excursion do not an accurate reproducer make. Granted, there are some excellent horn systems available (check out Nicholas McKinney's innovative UNITY HORN design at www.lambdacoustics.com.), but I do not accept that they reproduce transients "faster" or "better" than other speaker system types. Obviously, their extremely high efficiency means that with normal amplifier power, clipping will be dramatically reduced or eliminated in domestic listening situations, and this is more likely to be the explanation for "cleaner" dynamics with horn systems.

On the other hand, if you use any of the ultra low powered single-ended, looney-priced, valve amplifiers, well...(oops, I better not get into that argument).

POLYDAX TWEETERS

I have a pair of Polydax tweeters that from time to time blow the coil. I used to get replacement coils (it also contains the soft dome surface) from Just Speakers of San Francisco. I can't find them anymore and would like to know from where I can order replacements.

insearchofbrock@hotmail.com

Polydax was, for years, the U.S. designation for Audax, a French manufacturer. Query www.audax.com regarding replacements.

AM TUNER

I would really like to see articles on other items besides amps; for example, AM tuners. They are a part of audio and I would like to build a good AM tuner for DXing. And I would be especially pleased if it had an "eye" tube (GES) in it for tuning. You could start off with a simple set and continue into more complicated sets—a good challenge and something different than the old run-of-the-mill this and that amp. I'm sure some people out there would love to submit articles on them!

Vince Roberts
Beth, Pa.

HELP WANTED

I originally used your advice on upgrading/modifying my Hafler power and preamps (*The Audio Amateur*, 1980/1981). I did the recommended upgrades primarily on the 101, which had suffered some "system noise." Your mod made a great improvement to the preamp.

Are there any more mods that I can do to further improve my equipment? Here in Australia, I have had some difficulties in locating help on Haflers. I am under the impression that I can quite easily make a mod to my 200 amp and remove some of the noise that exists in the high end of its power output.

Phil Lock
philbert_lock@primus.com.au

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

NAD T550 DVD Player

from page 51

sampling rate on the Pioneer to 48kHz, there is a similar degradation).

I also tried the Classic Records DAD using the T550 as a stand-alone player. This was an area where I found the performance of the T550 disappointing. Although the T550 has on-board 96kHz/24-bit converters, it was unable to offer a musical sonic presentation on the DAD PCM disc. I found the sound somewhat harsh and aggressive, revealing few of the refinements for which this recording is justly famous.

The T550 performs far better as a stand-alone CD player than it does as a 96kHz/24-bit DAD player. I suspect that this is where the limitations of the analog power-supply regulators really manifest themselves.

CONCLUSION

The NAD T550 is one of the better DVD players on the market today. Its video performance leaves nothing to be desired, and NAD has paid more careful attention to their analog circuitry than most other DVD player manufacturers. As a result, the T550 outperforms typical Far East DVD players, including some commanding higher prices.

The primary disappointments in the T550 are the misconfigured analog power-supply regulators. Had the regulation circuitry been designed to function properly, NAD could have extracted even more from the excellent Burr-Brown chips. Yet, as a stand-alone Dolby 5.1 DVD player, the T550 will still outperform most of the competition. ❖

Book Review

High-Power Audio Amplifier Construction Manual

Reviewed by Charles Hansen

High-Power Audio Amplifier Construction Manual, *G. Randy Slone. McGraw Hill, 1996, 476 pages. ISBN 0-07-134119-6, $34.95.*

The main theme of the book is the construction of 12 solid-state audio amplifier designs, so it is not a design manual per se, but Mr. Slone references *Audio Power Amplifier Design Handbook* by Douglas Self (Newnes, 1996, ISBN 0-7506-2788-3) for that purpose. You may remember Mr. Self's series of articles, "Distortion in Audio Power Amps," *AE* 2/00, pp. 14–21; 3/99, pp. 30–38; 4/99, pp. 26–33.

Mr. Slone defines high-power as 50W or more, at reasonable expense (25–50 cents/watt). The author's analysis focuses on all major solid-state power amplifier architectures, beginning with some historical ones. He progresses to his optimum design in each case. He assumes that the reader, while perhaps a complete audio novice, is experienced in electronics construction, fundamentals, and safety.

In this publication, Mr. Slone describes modifications for compatibility with pro audio, musical instrument, home high-fidelity, and budget applications. All the designs are discrete (no op amps, ICs, or hybrid modules), and he presents "new research" in push-pull voltage amplifier stages, crossover distortion, and stability.

CHAPTER ONE—FUNDAMENTALS OF HIGH POWER AUDIO AMPLIFIERS

Chapter One addresses performance, reliability, and construction. Mr. Slone's performance criteria are low-distortion, wide-bandwidth, low-noise, and stability. He incorporates self-protection circuits, analyzes the use of thermal cyclic curves for the output devices, and mandates input-coupling capacitors to protect speakers from DC voltage ("DC-coupled designs are accidents waiting to happen"). He advocates strong enclosure designs that protect the electronics against contamination, and are safe for the user.

The book uses only 13 different active devices—small-signal JFETs and BJTs (bipolar-junction transistors), and power MOSFETs and BJTs—in all his designs.

CHAPTER TWO—MISINFORMATION IN AUDIO

Mr. Slone devotes an entire chapter to "subjectivism and its corruptive influence upon the audio industry," jumping squarely into the golden-ear controversy with "Science versus Subjectivism," stating that "subjectivism is often in direct conflict with known scientific fact," and that "the consumer audio industry is severely compromised by subjectivism."

He presents his argument that transient intermodulation distortion (TID) evolved as a misunderstanding of inadequate input-stage design, where current-starving is combined with a loss of linearizing negative feedback (NFB) at high frequencies.

He takes subjectivists to task for their love of vacuum-tube-amplifier distortion, directional cables, and exotic components and materials; and for their disdain of NFB, tone controls, protective relays, and fuses. He lists eight disadvantages of tube amplifiers as compared with solid-state. He doesn't believe metal film resistors or low-noise active devices are necessary in amplifier input stages. Indeed, the list of small-signal input devices he uses does not include any of the classic low-noise JFETs or BJTs. Some of his lower-power designs use back-to-back tantalum input-coupling capacitors.[1]

Mr. Slone does not say that all amplifiers sound the same, or that no further improvements are possible—only that good design is predictable and repeatable. He defends his designs in a way that may further agitate the pure subjectivists—with SPICE simulations, mathematical analysis, and distortion measurements that back his case. His viewpoint shows that very low levels of distortion cannot be heard, only detected with instruments. But he does concede, "one of the really positive aspects of subjectivism [is that] it has forced us to go back to the basics in search of a better sonic experience."

CHAPTER THREE—BASICS OF AUDIO POWER AMPLIFIERS

Chapter Three begins with a pertinent discussion on safety and hearing protection, presenting a basic list of audio power amplifier design specifications:

- Input impedance: 10k minimum;
- Output loading: 4Ω or greater (avoid 2Ω loads);
- Frequency response: 3Hz to 80–200kHz;
- Distortion: less than 0.03% at 1kHz, for 0.3% maximum at 20kHz;
- Damping factor: 0.04 to 0.02;
- Noise: –90dBr or better;
- Maximum reserve power: 2–3 times continuous rating; and
- Parallel output devices above 80W.

There is a discussion here on statistical and predictable reliability, and the areas of design, especially as used in output devices where reliability can be improved.

His fundamental architecture is the Lin three-stage design by RCA in 1956 with global NFB. Here, he includes a section on NFB terminology.

CHAPTER FOUR—INPUT STAGE CONFIGURATIONS AND ANALYSIS

This chapter covers a number of input stage configurations:

- Current sources (zener, transistor-diode/LED, dual-transistor, JFET);
- Current mirrors;
- Input-stage fundamentals (transconductance amplifiers, pole splitting);
- Evolution of modern input-stage designs:
  - Simple BJT differential pair,
  - BJT differential pair with current mirror and current source,
  - Cascode JFET with current mirror and current source,
  - Mirror-image input stage;
- DC offset; and
- Input signal conditioning.

Mr. Slone uses BJT input stages in each of his designs, except for cascoded JFETs in the Versatile 1980s Vintage Amplifier. This is followed by a description of the "dubious benefits of JFETs and cascode stages" in power-amplifier differential input stages. There is no mention of MOSFET input stages (good examples might be those Nelson Pass uses in his designs). Each input-stage design is presented as a SPICE analysis, with second- and third-harmonic distortion simulation curves.

CHAPTER FIVE— VOLTAGE AMPLIFIER STAGE: CONFIGURATIONS AND ANALYSIS

The voltage-amplifier stage is responsible for all the voltage gain in a Lin power amplifier. Mr. Slone presents "common misconceptions" in the areas of linearity, NFB, and compensation. The typical VA configurations are current source loaded, cascode, Darlington (beta-enhanced), differential, bootstrapped, buffered, and push-pull.

"Determining the Dominant-pole Compensation Cap" value, and "Understanding two-pole Compensation" are some of the sections. In "Optimizing the VA Stage," Slone makes an excellent case for incorporating current-limiting in the VA stage to protect the VA-stage amplifier transistor during output-stage short-circuit protection activation. This, Slone has found, almost universally involves shorting the VA input signal to the predriver transistors. The last section in this chapter is a discussion of modern VA-stage design, from mediocre (differential JFET) to ultra high end.

CHAPTER SIX—OUTPUT STAGES: CONFIGURATIONS, CLASSES, AND DEVICE TYPES

The salient topics are:

- General principles of output stages (OPS);
- Purpose and function of the output stage; and
- Output-stage classes

While the designs in the book are limited to Class B (with the exception of one Class A), all the output-class stage topologies are discussed. Slone gives them the following definitions: Class AB, "a poor marriage of both A and B characteristics," and a form of crossover distortion called "gm doubling." Class C and E are mainly used in RF circuits, and Class D is a pulse-width modulation (PWM) technique. Class G uses two power-supply rails, while Class H uses dynamic rail voltages, both designed to improve efficiency. Class S is designed for vacuum-tube amplifiers, which are not included in the book.

The Class-A output-stage designs discussed in detail are single-ended (SE) resistor loaded, SE active loaded, and push-pull. Class-A biasing methods covered in this section are the balanced current regulator and SE current regulator.

While Mr. Slone believes that Class-A designs are not practical because of their low efficiency and their tendency toward transient distortion due to lack of headroom, he does present one push-pull "High Performance Class-A Cookbook Design." He believes however,

"enthusiasm for Class A is often supported by a variety of misunderstandings and myths."

Next we get to the methodology to which the book is really dedicated, Class-B designs. A number of Class-B BJT output stages are discussed in detail:

- Emitter follower (EF) or complementary symmetrical;
- Complementary feedback (CF);
- Quasi-complementary (QC) historical perspective;
- Output triples (another historical perspective);
- Paralleled BJT output stages;
- Summary of Class-B OPS characteristics.

He also covers Class-B MOSFET output stages, which he uses in a number of designs:

- MOSFET versus BJT characteristics;
- MOSFET OPS configurations:
  - Source follower (SF),
  - Quasi-complementary (historical perspective),
  - Complementary feedback; and
- Paralleled MOSFET output stages.

Finally, Slone touches on insulated gate bipolar transistors (IGBT), which are optimized for switching power supplies, rather than linear audio applications. There are photos of two Mark V amplifiers in this chapter, the TA-800 MkII and the AF-3. Perhaps Mr. Slone served as a design consultant for Mark V Electronics.

CHAPTER SEVEN—STABILITY, DISTORTION, AND PERFORMANCE

Mr. Slone presents a detailed analysis of stability criteria, covering:

- Bias generator stability (Vbe multiplier, thermal coupling, and delays);
- Output-stage stability (internal, external, Zobel network, damped inductor);
- Overall amplifier stability (Nyquist stability, dominant-pole compensation, phase margin, overall NFB); and
- General summary of stability concerns and temperature compensation.

In the section on distortion mechanisms, he covers a wide variety of topics:

- Crossover distortion (the most tenacious in Class B, decreasing with decreasing power output and the effect of global NFB);
- Large-signal nonlinearities (beta-droop in BJTs; EF, CF, QC solutions);
- Switching distortion (cross-conduction)
- Differential-imbalance distortion;
- VA stage-loading distortion;
- High-power distortion mechanisms (IR drops, electric fields) and their solutions (star-grounding, separation and shielding, critical reference points);
- Electrolytic-capacitor distortion (DC blocking, input coupling); and
- Output fuse distortion (there are no recommended output/speaker fuses).

The final section is devoted to slew-rate requirements and calculations, and improving the amplifier slew rate.

CHAPTER EIGHT— AMPLIFIER AND LOUDSPEAKER PROTECTION METHODS

The chapter starts with safety (mains and power-supply fusing, power cords, insulation and enclosure issues). "Electronic Overload Protection for Semiconductor Devices" deals separately with BJT and MOSFET output devices. MOSFETs are easier to protect since they have integral protection diodes, higher maximum junction temperatures, self-limiting channel current, and are immune to secondary breakdown effects.

Mr. Slone presents various electronic protection circuits, from simple source/emitter current limiting to single-slope, dual-slope, and multislope VI protection. There is also a discussion on various "radical" methods of OPS protection, such as speaker-load monitoring and microprocessor-based detection.

The author's speaker-protection circuits are all relay-based. He develops circuits for turn-on delay muting, DC offset protection, and HF oscillation protection (which he believes is not needed with proper stability design).

CHAPTER NINE— AUDIO AMPLIFIER POWER SUPPLIES AND CONSTRUCTION

Three main types of power-supply topology are investigated in this chapter.

Dual-polarity unregulated supplies are used in all these designs. He deals with ripple rejection by proper power-supply rejection ratio (PSRR) design. Signal injection onto the rails is prevented with bypass capacitors, and rectifier-bridge diode switching EMI is minimized with snubbing capacitors.

Linear regulator power supplies are a poor choice, according to Slone. They eliminate ripple at the expense of reduced efficiency, reliability, headroom, and perhaps reduced slew rate. His position is that a well-designed amplifier will reject ripple as well as the series-pass active devices in a linear regulator.

Switching power supplies are discussed in "serious performance shortcomings in audio circuits," where there is focus on the areas of lower cost, size, weight, and the opportunity for power-factor correction. EMI emission remains a serious drawback, according to Slone.

The remainder of the chapter deals with the design of a dual-polarity unregulated audio power-amplifier power supply. The topics are:

- Power-transformer considerations (ratings, EMI, and flux leakage);
- Rectification and snubbing capacitors;
- Power-supply fusing;
- Reservoir caps (1000µF for each 10W), voltage derating, bleeder resistors, rail-decoupling caps;
- Power-supply calculations;
- Rail rejection in power amplifiers;
- Power-supply wiring techniques (twisted wiring, high-quality star grounding, wire gauge requirements); and
- Crosstalk and DC supplies in stereo amplifiers.

CHAPTER TEN—BUILDING THE OPTIMUM AUDIO POWER AMPLIFIER

This is the culmination of all the previous chapters, providing a single practical construction guide with four-step planning:

Step 1: establish the construction goal;
Step 2: design the power supply;
Step 3: lay out the basic topology;
Step 4: determine application-dependent variables.

CHAPTER 11— AUDIO AMPLIFIER COOKBOOK DESIGNS AND DISCUSSIONS

Only amplifier schematics are shown in this chapter. Supporting circuits such as the power supply, amplifier and speaker protection, parallel output devices, and exact power ratings, are omitted for clarity. Mr. Slone presents 12 designs for various musical-instrument, pro audio, and home high-fidelity applications:

- Basic MOSFET design;
- Versatile 1980s-vintage BJT amplifier design;
- Optimum low-distortion medium-power BJT audio amp;
- Low-distortion L-MOSFET audio amplifier;
- Low-distortion L-MOSFET professional audio amplifier;
- Mirror-image professional audio amplifier;
- High-power professional MOSFET audio amplifier;
- Two high-quality, general-purpose BJT amplifier designs;

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- Low-distortion EF amplifier design;
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- The optimum power amplifier design.

CHAPTER 12—CONSTRUCTION TECHNIQUES AND CONSIDERATIONS

The topics covered in this chapter are:

- Mechanical considerations of thermal dynamics;
- Thermal resistance;
- Thermal dynamics and heatsink evaluation;
- Thermal dynamics and OPS evaluation;
- Thermal overload protection;
- Thermal dynamics and thermal delays;
- PC board construction and generating CAD artwork;
- Mechanical layout of a completed audio amplifier; and
- Wiring methods for audio power amplifiers.

CHAPTER 13—POWER-AMPLIFIER DIAGNOSTIC EQUIPMENT AND TESTING PROCEDURES

After another reminder and discussion about safety, the author presents three areas of test and analysis for finished power-amplifier projects.

"Practical Considerations of Required Test Equipment" discusses the Variac™ variable AC transformer, semiconductor testers, LCR meters, DC bench power supplies (including a schematic for building your own), and dummy loads.

"Computerized Analysis Equipment" discusses computerized design and layout programs, SPICE simulation, and schematic capture for PC board layout.

"Testing Procedures for Audio Power Amplifiers" is broken down into five testing categories:

- Evaluation testing (including distortion and spectrum analysis);
- Functional testing;
- Symptomatic troubleshooting (personal designs and commercial units);
- Setup procedures (VA stage calibration and DC output level); and
- Load testing.

Appendix A is a 28-page glossary of audio power-amplifier terminology. Appendix B is a section on electronics units and abbreviations.

Appendix C is the PC board artwork for the 12 power-amplifier designs that were presented in Chapter 11. All the author's designs use single-sided boards; etched and drilled boards are available from his company, Seal Electronics. Appendix D is a list of suppliers of information and materials.

CONCLUSION

I found this to be a fascinating and technically valuable book. The author built and tested all twelve designs. He provides PC board artwork and helpful hints on buying the expensive items, such as power transformers, heatsinks, reservoir capacitors, chassis, and so on. He gives a detailed yet clear and concise description of the functions of every component in every design in the book. There are many sidebars called "QUICK>>>TIP" and "TRADE-SECRET" throughout the book, and his writing style is very engaging.

I think this is an excellent addition to anyone's audio technical library, even the pure subjectivist. Slone's philosophy is that it is okay to disagree on some issues and pursue your own heartfelt insights into building a better audio amplifier. ❖

REFERENCES

1. W. Jung and R. Marsh, "Selection of Capacitors for Optimum Performance," *Audio* 2/80 and 3/80. The authors found that a polarized tantalum capacitor acts as a cap shunted by an imperfect diode. The success of a back-to-back connection of tantalum caps depends strongly on the match of the two units. Since commercial caps are rare finds at less than ±10% tolerance, and are often ±20%, they become difficult to match. (Expensive military MIL-C-39003 types are available with ±5% tolerance.) Although the impedance and ESR curves are flat out to MHz frequencies, the tracking of solid tantalum caps is not all that good at elevated temperatures. Tantalum pentoxide is a rather high-k dielectric material (26, or three times higher than aluminum oxide). At high temperatures, capacitance can increase as much as 8–12%, DC leakage increases 20 times, and the dissipation factor doubles, although not as much as aluminum, which also has higher ESL. I prefer to use polypropylene film input-coupling caps in every instance. Even though they are more expensive, you need only one film cap of half the value of the two tantalums.

Glass Shard

Bias: Fixed or Not Fixed

As most people know, the output of a Class AB1 amplifier with an automatic bias by means of a resistor in the cathode is quite a bit less than with a fixed bias. The increase in average current when delivering power increases the voltage drop in the bias resistor and thus the bias voltage. Changing from auto bias to fixed bias is not always easy to do, surely not in existing equipment.

But auto bias also has some advantages, as it compensates a bit for any mismatch between the tubes. That's why some recent tube power amps use an automatic control for the nominal current in the tubes using a semiconductor control circuit. You can more or less combine the advantages of both auto and fixed bias with the circuit in *Fig. 1.*

To stop the increase of the bias voltage when delivering power, you can shunt the bias resistor with a power transistor that provides a bypass for the increase in current. The power transistor will start to conduct as soon as the bias voltage rises above the zener voltage. Since the bias voltage in this configuration has almost no effect on the maximum power output, you can easily increase the bias by increasing RB so far that the zener just starts to conduct but not enough to switch on TR, which can be a cheap power Darlington.

But bear in mind that the average current can be 100 to 250mA, so please consider the use of heatsinks. The higher bias will also run the tube a bit cooler as well and not lower the maximum output at all. To prevent distortion you must include capacitor CB; the transistor is only there to stabilize the bias.

However, don't expect any miracles if the power supply and the output transformer are already running at their limits. A new output transformer with lower plate-to-plate resistance will surely deliver more power. But as most speaker impedances are not really 4 or 8Ω, the impedance seen by the transformer primary might be low enough to give you the extra watts anyway. ❖

A. J. van Doorn
Amersfoort, the Netherlands

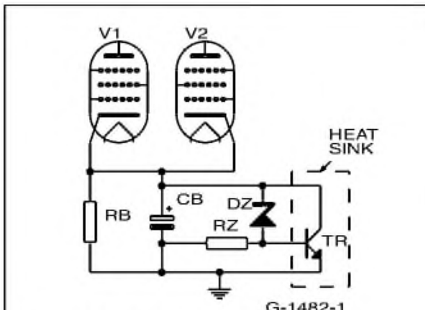

FIGURE 1: Floating tube bias.

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New Chips on the Block

AKM Semiconductor AK4112A

By Charles Hansen

AKM Semiconductor (www.akm.com) introduced the AK4112A, an S/PDIF or AES/EBU digital audio interface receiver (DIR) compliant with 24-bit, 96kHz data or non-PCM data stream such as AC-3.

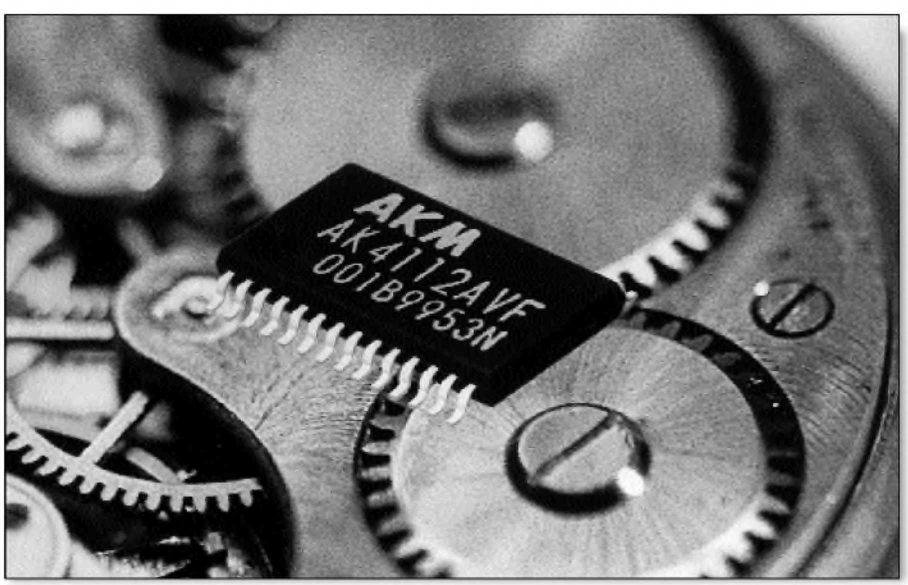
Although conventional digital audio interface standards (IEC958 and others) for high-quality audio playback in DVD and home-theater systems are capable of transmitting multi-channel audio signals (AC-3/MPEG and others) while data is in compressed form, conventional DIRs are unable to differentiate between non-PCM and existing PCM data.

The AK4112A is able to cope not only with non-PCM audio data, but can detect compressed formats as well. Its low jitter clock recovery circuit, supported by a newly developed high-performance phase-locked-loop (PLL), delivers a master clock output that produces an optimal audio decoder front end for PCM data up to the DVD standard of 24-bit, 96kHz (24/96). A de-emphasis filter enables automatic de-emphasis of PCM data, reducing the complexity in digital signal processing (DSP) required for pre-emphasized signals.

The AK4112A has a channel selector that enables a 4-channel input suitable for current consumer-device applications as well as the direct input of unbalanced signals to the built-in amplifier. The 3V or 3V3 power supply makes the AK4112A ideal for DSP interfacing and portable applications.

The AK4112A is compliant to consumer and professional audio standards, with performance suitable for interface with studio devices such as mixers or effects processors.

The $3.15 (10k) AK4112A uses a 28-pin VSOP package. Contact Richard Kulavik toll-free at 888-256-7364 for engineering samples. ❖



Music Corner:

What Happened to the Magic?

By Don Monroe

I have a number of recordings of classical music that I particularly enjoy. Some are the first recordings of the particular piece that I've heard, or renditions or interpretations that I know really well: works by Tchaikovsky, Rachmaninoff, Rimsky-Korsakov, George Gershwin, and a few other composers.

When I hear a different recording of these works, it sometimes sounds so different, I wonder if it's played from the same score. I'm referring to music with a lot of interplay between the various sections of the orchestra—e.g., the brass vs. the woodwinds. In the recordings with which I am familiar, this interplay between the various sections makes wonderful sense. But the new version sounds as though the various sections aren't even on the same page of the score.

The magic I am accustomed to hearing just isn't there. I don't know whether the conductor just doesn't know the score (not likely), or his vision of the piece is so different, or he is of a different school, or the recording engineer goofed up, or what. Or is it that I know the old recording so well, that *any* difference will sound wrong to me?

In the field of computing, it has been said that the text editor you learn to work with first will become the standard by which you measure all others. You learn how to do things with the first editor that will define, for you, how those things should be done. You adopt a new editor only grudgingly.

The same with operating systems. If you start out on a Vax, and then must switch to a Unix platform, you do it with great reluctance, kicking and screaming all the way. This phenomenon probably happens in many different areas of life.

Maybe the same thing is happening when I listen to music. This could well be the case when one conductor plays a piece faster or slower than I am accustomed to. But I think I am talking about something more than differences in tempo. In the new recording, I just don't hear the *music* I heard before. Do others ever have that experience?

Reprinted, with permission, from The Arizona Audiophile, *5321 E. Elmwood St., Mesa, AZ 85205. Don may be reached by e-mail: dmonroe@amug.org. The club sponsors a lively schedule of events and publishes an interesting and informative newsletter.*

Don Monroe raises two interesting questions about listening which I have often thought about. First, does our first hearing of a particular piece imprint some kind of preference standard in our minds for how it should be performed? Second, what are the variables that make performances seem so different to our interpreting minds?

I think these questions are of sufficient interest that we might have a worthwhile series of interchanges on them, perhaps as an ongoing feature in this magazine. After all, music listening is, at bottom, what this periodical is about.

I share Don's belief that somehow a first hearing of a particular performance can often set up a sort of musical yardstick for subsequent hearings. I always listen very carefully to the second movement of performances of Prokofiev's "Classical" symphony—a schoolboy exercise that launched his career. I first heard the St. Louis Symphony's performance of this work—on 78rpm disks. Very rarely does any subsequent performance of the second movement seem lyrically limpid enough to match my memory of something I heard in my 20s.

Harold Shoenberg wrote a wonderful commentary in the late, lamented Fi*'s pages about performance length. He observed that contemporary style seems to dictate much slower, more deliberate performances, comparing earlier and later renditions of Mahler in which certain symphonies were up to 17 minutes longer. Most of us who listen to classical music recordings are clearly aware that both the conductor and the orchestra make major differences in how a performance is executed.*

Most experienced listeners develop preferences for certain favorite conductors—and sometimes a set of choices of a conductor or conductors for a specific composer. If you are able to compare six performances of Shostakovich's fifth symphony, I will wager that no two will be very much alike. Fortunately, the willingness of some record companies to re-issue performances of early recordings offers a chance to compare some much older styles with those of the present day.

Given our common interest in music, it seems to me we could spare some space in this magazine for a conversation about our listening habits. I hope I am not opening a can of worms here, but the response to my invitation to share reader preferences for selections which test equipment performance, Test Tracks*, has been very gratifying. In like manner, I hope many others will contribute.*

If you have something to offer for what we will call "Music Corner," please feel welcome. I will ask you to keep your offerings to no more than 500–750 words—about two or three typed, double-spaced pages, please. Attachments to e-mail would be helpful, or clean typed pages that can be scanned. I look forward to hearing from you.—E.T.D. ❖

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Test Tracks

Reader-submitted selections to test audio systems

My seven favorite test CDs, by category, are as follows.

1. Electric Bass—Brian Bromberg, track 13 "My Bass," Nova Records 9351-2 (Santa Monica, Calif.).
Incredible bass playing and fidelity. This is a great test of inter-driver tonal balance and "voicing." There is a lot of depth and speed at the same time.

2. Deep Bass—*The Organ Works of Cesar Franck* Track 3, Disk 1, Dorian Recordings DOR-90135I (Troy, N.Y.). Track 3 is "Fantasy in A Major."
Within the liner notes is a statement about not rolling off the bass anywhere in the recording chain. The 32′ pipe comes through loud and clear on subwoofers that can accurately reproduce 16Hz. Since these "infrasonic sounds" are closer to pure tones than not, harmonics are not that rich.

Some lower-voiced recordings of instruments such as contrabassoon, bass drum, and so on are rather rich in harmonics and can easily fool the ear into thinking you are hearing the fundamentals. With the above-mentioned recording, this is not the case; if you don't have subwoofers that can provide adequate acoustic energy below 20Hz, the big pipes simply will not be heard (or rather "felt").

3. Piano—*The Wonderful Sound of Three Blind Mice*, Track 2 "Midnight Sugar," TBM CD-2530. Also available from Acoustic Sounds as JVC "XRCD"; "Midnight Sugar" is on CJVC 0023 and there is another: *The Famous Sound of Three Blind Mice* which is CJVC 0991.
This is a Japanese CD that was often mentioned within the pages of *Audio Magazine.* I believe it is available from a distributor in Salina, Kans. (I no longer have the contact information). This is the most realistic-sounding piano recording I have ever heard (*Audio* always said the same thing, and they were right). The piano is reproduced with uncanny dynamics and authority; very clean and refreshing.

4. Female Voice—Nnenna Freelon, Track 6, "Future News Blues," Columbia, CK 48981 (New York).
Powerful, smooth voice; SSSSs and ZZZZs of many female voices and/or microphones are not present. Purposely having a tweeter level set too high will artificially raise the "S and Z" levels. If the speakers under evaluation appear to have this characteristic on this recording, it is my opinion that the tweeter level is either too high or has "peaking" in the high presence/low treble range.

I also use this recording as a check of tonal balance between all drivers. Later in the track there is a very forward-sounding sax part which is well recorded and also powerful yet well balanced throughout the range.

5. Violin—Midori, *Live at Carnegie Hall*, Track 10, "Tzigane," Maurice Ravel, Sony Classical SK 46742.
The solo violin is very "wooden, natural and open." This piece does not sound strained or screechy like many violin recordings. To my ears, this is what a violin really sounds like at a live performance. This recording really brings out the subtle nuances of Midori's playing. Lesser speakers will show their weakness(es), and better speakers will absolutely bloom when set up correctly.

6. Full Orchestra—Gustav Holst, *The Planets*, Chicago Symphony Orchestra and Chorus, James Levine. Deutsche Grammophon 429 730-2 (Hamburg). Track 1, "Mars, the Bringer of War."
The opening measures have the strings playing "spicatto" (the bow lightly and repeatedly striking the strings rather than bowing). I have heard other recordings and performances of this same piece in which this effect is either muffled or totally missing. This section should sound light and delicate yet still be obvious.

Later in this track the snare drum comes in; the sound of the drum is of equal intensity to the sound of the snare. Again, this is useful for determining the relative levels of the midrange and tweeter. (If only a two-way speaker, this is a good way to tell whether the upper range of the woofer really reaches into the "midrange.") The upper and lower brass sections are also very well recorded, as the overtones (harmonics) are all preserved in their proper proportions. This is a good test to determine whether there are any major peaks or dips in the "presence band."

7. Transient Response—Pink Floyd, *Dark Side of the Moon*, Track 3, "On the Run." This is the Mobile Fidelity Sound Lab "ULTRADISC" UDCD517 gold CD. The liner shows the original as Harvest Records/EMI. I'm not sure whether the original CD will have the same effect, as I only have the Ultradisc version.
This track has some very sharp clicks and ticks. The midrange and tweeter both must have extremely fast transient responses in order to appreciate this effect. The best setup on which I've ever heard this used an Eton 5¼″ midbass and a Philips "Isophase Leaf" tweeter (ribbon). I used these ticks to time adjust the drivers.

With the arrival structure set correctly, these ticks and clicks just jump right out at you to some point in space well in front of the speaker box. This is a very uncanny phenomenon and definitely three-dimensional. Speakers not having these combined abilities will smear the signal, and maybe bad enough to be missed altogether. ❖

Dan Martin
Greensboro, N.C.

Let's hear from you. Simply describe your seven favorite pieces (not to exceed 1000 words); include the names of the music, composer, manufacturer and manufacturer's number; and send to: "Test Tracks," Audio Amateur, Inc., Box 876, Peterborough, NH 03458, FAX (603) 924-9467, e-mail editorial@audioXpress.com. We will pay readers whose submissions are chosen for publication a modest stipend.