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MEET NOTED AUTHOR DAVID WEEMS

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OBITUARIES

Reginald Williamson 1925—2009

Audio Amateur's very first contributing author passed away quietly in his sleep this past June after a yearlong battle with cancer. He was not only a fine engineer and author, but over the years became a great personal friend. He made many influential friends over his lifetime and performed the painful duty of reporting the passing of most of them. These included Peter Walker, famous for his Quad line of preamps and power amplifiers, but more so for his Electrostatic loudspeakers; and Peter Baxandall, designer of the Baxandall tone control topology, as well as superior test equipment. Not so well known was another fellow British Telcom engineer, Alan Watling, who converted Reggie's designs into working hardware.

Most Britons who have a special interest manage to make the knowledge useful in some way which benefits society. Reggie's love for music took him into positions of leadership and responsibility for music societies such as the one honoring Elgar, or a local Norwich audio club which built a wired-wireless radio facility for entertaining patients in a local hospital.

Reggie was a popular, highly demanding teacher in local community and university colleges. His standards were high but practical as well. He understood the art of the possible and was very good at what he thought were obvious improvements in the designs of others. His simple upgrades to Heath's IG-18 audio generator became one of his most popular and widely used articles in early *Audio Amateur* issues. He was one of the first of many British authors giving permission for their articles to be reprinted in the pages of our magazines. He opened doors to his wide range of technical friends who enriched many of our publications over decades.

He leaves behind a son by his first marriage and his most enthusiastic and supportive wife Marjorie who relays one of Reggie's characteristic quotes, "On the day I refuse my glass of red wine, call the undertaker." We will all miss our lively, roisterous and beloved friend.—ETD

Ronald Wagner 1932—2009

Ronald Wagner, the author of *Electrostatic Loudspeaker Design and Construction*, died of cardiac arrest on April 5, 2009. He was 77. Ron's love for electronics and audio systems started early in life. After graduating high school, he attended Milwaukee School of Engineering. In the 1960s, he and a few coworkers at Ampex Corporation in California started to experiment with electrostatic loudspeakers. Through all of his years of working at various high tech companies in the San Francisco Bay Area, he always was busy in the garage on weekends working on his electronic equipment and designs. Even after he retired, he did some consulting work and was writing a second electrostatic loudspeaker book. In 2003 he moved to Oregon to be closer to his daughters since his wife passed away from melanoma cancer five years prior. While living in Oregon, he enjoyed doing a lot of golfing and even had his own putting green in his backyard. He battled Parkinson's the last several years of his life, which ultimately led to his death in April. He will be missed by his two daughters, one son-in-law, four grandchildren, and many dear friends.

Justin Gordon Holt 1930—2009

Justin Gordon Holt left us the other day, on Monday, July 20, 2009, after losing a long battle with lung cancer. Born in Australia in 1930 to parents who were both killed in an auto accident, he was adopted by American parents whose home was in Wallingford, Penn. After two years as Tape Recording editor at *High Fidelity* magazine, he departed the company, along with Roy Allison, when *High Fidelity* was purchased by *Billboard* magazine. As a new owner, *Billboard* killed *HF*'s younger sibling *Audiocraft*, which was barely out of its swaddling clothes at almost three years old. They claimed that the DIY magazine was not profitable. The real reason: *Audiocraft* had cheaper ad rates and its elimination increased linage for the larger *HF*. Allison went on to do world famous work for Acoustic Research, and Gordon worked for two years with Paul Weathers in New Jersey.

Photo courtesy of *Stereophile*.

Gordon and Roy had other fundamental reasons for leaving Great Barrington (Mass.). *High Fidelity* under Charles Fowler had begun as a well-rounded journal treating both audio equipment topics and very authoritative recorded music reviews as monthly fare of the highest quality. Commercial interest in satisfying advertisers led to more and more "positive" reviews, which bothered Gordon and Roy.

After leaving Weathers, Gordon moved to his widowed mother's home in Wallingford, where he founded *Stereophile* magazine in 1962. Its early issues were quarterly, slender, and quickly became erratic in schedule. Gordon was not a businessman. However, he had other characteristics which fully equipped him as a great editor: he was a superb writer, he had a mordant wit combined with curiosity and a relentless insistence on the truth about things. He loved sound, but cared very little for equipment. He became a tinkerer, not by inclination, but only by necessity if something needed fixing and if he could figure out a way to do it.

The Ampex 601 had a chatter problem, being otherwise a remarkably good tape machine. He repaired his unit and made stunningly realistic recordings. He regularly taped rehearsals of the West

Continued on p.16

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Chester College's orchestra, which were so accurate you could hear every sour note any member played. He made recordings of all manner of mundane sounds, some very distinctive ones heard only in a toilet.

He was also a fine artist, a fine cartoonist, and a master with an airbrush. The earliest covers of *Stereophile* are collector's items. He moved his wife Mary and the family to Santa Fe, falling in love with the beauties of the southwest. In the 70s, he sold *Stereophile* to a master manager, Larry Archibald, who turned the magazine around financially, hired new people, and set it on the road to success.

Gordon's most important achievement was, however, a change in the character of audio equipment reviews in the periodicals of the United States. As several have pointed out in their appreciations of his influence, his reviews were based on listening rather than measurement only. This was not because Gordon did not know the basics of engineering and physics, but because what you heard seemed far more important to him than the few measurements possible at the time.

One of his funniest covers was a spoof of the front panel of Heath's distortion meter, which he had carefully renamed the "Beastkit Audio Analyzer," with switch choices for "Acuity, Timbre, Test and % Fi." The "Beast Company" was located, of course, in "Henton Barber, Mich." The meter markings were "Lousy, Fair, and Whoopee."

Gordon was one of the few reviewers to notice that English has the smallest number of adjectives for the characterization of sound than any of the human senses. One of his major efforts was a glossary/dictionary of audio, published in 1990 (Audio Amateur Press). Peter Mitchell, in his foreword to the book, said, "Holt has always been The Great Explainer—using plain English to tell readers what is happening behind the front panel and how new processes operate to produce a sonic benefit. So this is two dictionaries in one: a glossary of subjective audio and a comprehensive plain-English guide to nearly two thousand technical terms."

Since those early days, most of the audio publications in the US have disappeared. The survivors avow their loyalty to Gordon's methodology and philosophy with varying levels of success. His perception was that reviews were too often affected by periodical management's pressures on reviewers. Responding to this criticism, reviewers countered with the claim that, "No one wants to read bad reviews," or the belief that advertisers are the primary financial support for magazines. This latter ignores the more than evident fact that the money manufacturers spend on promoting their products comes from their paying customers. And the purchaser of a disappointing piece of gear will be sorry he could not read a more candid and revealing review.

Physically, J. Gordon was small of stature, a quirky, friendly, formidable adversary, and a great human being. The only time I saw him clad in conventional shoes was at his wedding. He found old-fashioned sneakers more comfortable. It was always fun to join him for an evening, and deeply interesting. He was unique, one of a kind. We are much poorer for his departure.—ETD

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Build The "Groove Jockey"

Add versatility to your playback of those good old vinyl records!

At the risk of being laughed off the pages of *audioXpress* magazine, I nonetheless feel compelled to confess that, until relatively recently, the turntable I used for listening to treasured vinyl was a 1960s-vintage rim-drive broadcast transcription turntable. This perverse choice was prompted in part by nostalgia from bygone radio station experience, as well as a refusal to accept anything less than a 10 lb platter driven by a washing-machine motor as an adequate means of spinning a record.

But after watching woofer cones undulate at their near-maximum excursion during violin solos, and supposing that a good deal of Doppler distortion was being generated and amplifier power wasted, I finally retired the "Rumble-Master Pro" and replaced it with one of those direct-drive turntables favored by dance club DJs. What a nice surprise to find that the new turntable not only runs circles around the old one in terms of noise and startup time, but its integral tonearm appears to be well made and nicely balanced. What's more, the ability to "scratch" could help open doors to new employment opportunities in these troubled economic times.

BACKGROUND

Buyers' guides in old hi-fi magazines used to list audiophile-grade turntables that ranged from ultra lightweight stamped-sheetmetal designs to heavy lathe-turned platters, most of them rim-driven by an AC motor through a rubber-tire idler wheel. These seemed adequate for monaural recordings, but suffered to a greater or lesser extent from motor vibration and other mechanical noise in the vertical plane. This effectively signed their death warrant when stereo records came onto the scene in the late 1950s. Subsequent generations of turntables targeted the specific playback needs of stereo records and employed belt-drive and direct-drive technologies with quiet, servo-controlled DC motors.

It was the British genius Alan Blumlein who invented the "45/45" stereo disc recording system in 1931 (see *The Inventor of Stereo: The Life and Works of Alan Dower Blumlein*, available from Old Colony Sound Lab, www.audioXpress.com). Stereo records did not achieve commercial success, however, until the Westrex company in the US refined the "StereoDisk" process and brought it into the mainstream in 1958.

The left and right channels of a stereo record are recorded at a 45° angle on opposing groove walls, but an analysis of the recording process generally considers the vector sum and difference of the two channels. The L+R stereo sum represents the lateral (side-to-side) component of groove modulation. Monaural "LP" records had always been lateral recordings, as were almost all the 78s that came before them. This meant that stereo records were "backward compatible" with mono players, and the older monaural records could be played just fine on stereo systems.

The L-R "stereo difference" information is the vertical "hill-and-dale" modulation component of the record groove. Typical stereo music has far less energy in the L-R domain, so distortion mechanisms fundamental to the vertical province of the recording process are of less concern in a stereo disc recording. Furthermore, it has long been standard practice in stereo disc mastering to blend low frequencies to monaural, thus saving the cutterhead and playback cartridge from having to undergo high-amplitude vertical excursions.

Nevertheless, good stereo phono cartridges have high compliance (their ability to follow the groove) in all planes, so even though the lower music frequencies are purposely restricted to lateral groove modulation, the playback system remains sensitive to vertical turntable rumble. Ironically, this situation may actually be less of a problem today with the popular trend toward the use of subwoofers, which respond to the L+R stereo sum common to both channels, fundamentally ignoring stereo-difference (L-R) low frequencies.

HOW THE PROJECT CAME ABOUT

My old broadcast turntable was wired to an 80s-vintage phono preamp that had a pair of very useful front-panel switches. One switch summed the left and right channels to mono, giving a substantial reduction in turntable rumble for mono records and making certain surface noises and distortion less audible with old 78s and early LPs. The other switch inverted the phase in one channel. With both switches engaged, the system played vertically modulated "hill-and-dale" records, such as Edison Diamond Discs, Pathé 78s, and some 1940s-vintage broadcast transcriptions.

My new direct-drive turntable has an extremely quiet built-in preamplifier, but I really missed the versatility of playback mode selection. This prompted a look into just what it would take to restore features of the old preamp and maybe even add another one or two.

THE "GROOVE JOCKEY"

The project evolved into a line-level, unity-gain signal-conditioning module (**Photo 1**) that goes between the output of any phono preamp and the input of the listening system. Construction is a simple panel and circuit board "sandwich" (**Photo 2**). The device allows you to choose among six playback modes:

1) Normal stereo playback
2) A second stereo playback mode in which low frequencies are matrixed into monaural
3) Monaural "lateral" playback with left

and right channels summed
4) Monaural "vertical" playback using the L-R difference signal
5) Playback of the right channel through both speakers
6) Playback of the left channel through both speakers

These modes are explained in more detail with the aid of **Fig. 1**. Incidentally, the seemingly random assignment of component reference designations on the schematic actually reflects their location on the printed circuit board, in a logical top-to-bottom, left-to-right sequence (**Photo 3**).

STEREO I (NORMAL)

The left- and right-channel inputs are fed to buffer stages IC1A and IC1B, which also are configured as second-order high-pass filters. A good deal of turntable rumble is subsonic, better observed as woofer movement than heard as a rumbling noise. These buffer stages impart a low-end rolloff starting at 30Hz. The response of this "rumble filter" is graphed in **Fig. 2**. 20Hz is attenuated by only about 1dB, but 10Hz is down 10dB, and 5Hz more than 20dB.

The gain of these buffer stages is actually just a bit over unity, and you can adjust it using the channel balance control, R2. You can reach this trimpot with a small screwdriver through a hole in the front panel to equalize levels between the two channels. In this first switch position the left and right channels are delivered directly to the output jacks.

STEREO II (MONO BELOW 200HZ)

This playback mode gives the greatest attenuation of turntable rumble and can also greatly reduce the large vertical signal component that's encountered when playing badly warped records. Old 78s often have a nasty "once-around" bump, but heat-distorted LPs can assume the appearance of corrugated-metal roofing. As long as the player can still track the disc, this playback mode will greatly

PHOTO 1: The Groove Jockey as seen from the front.

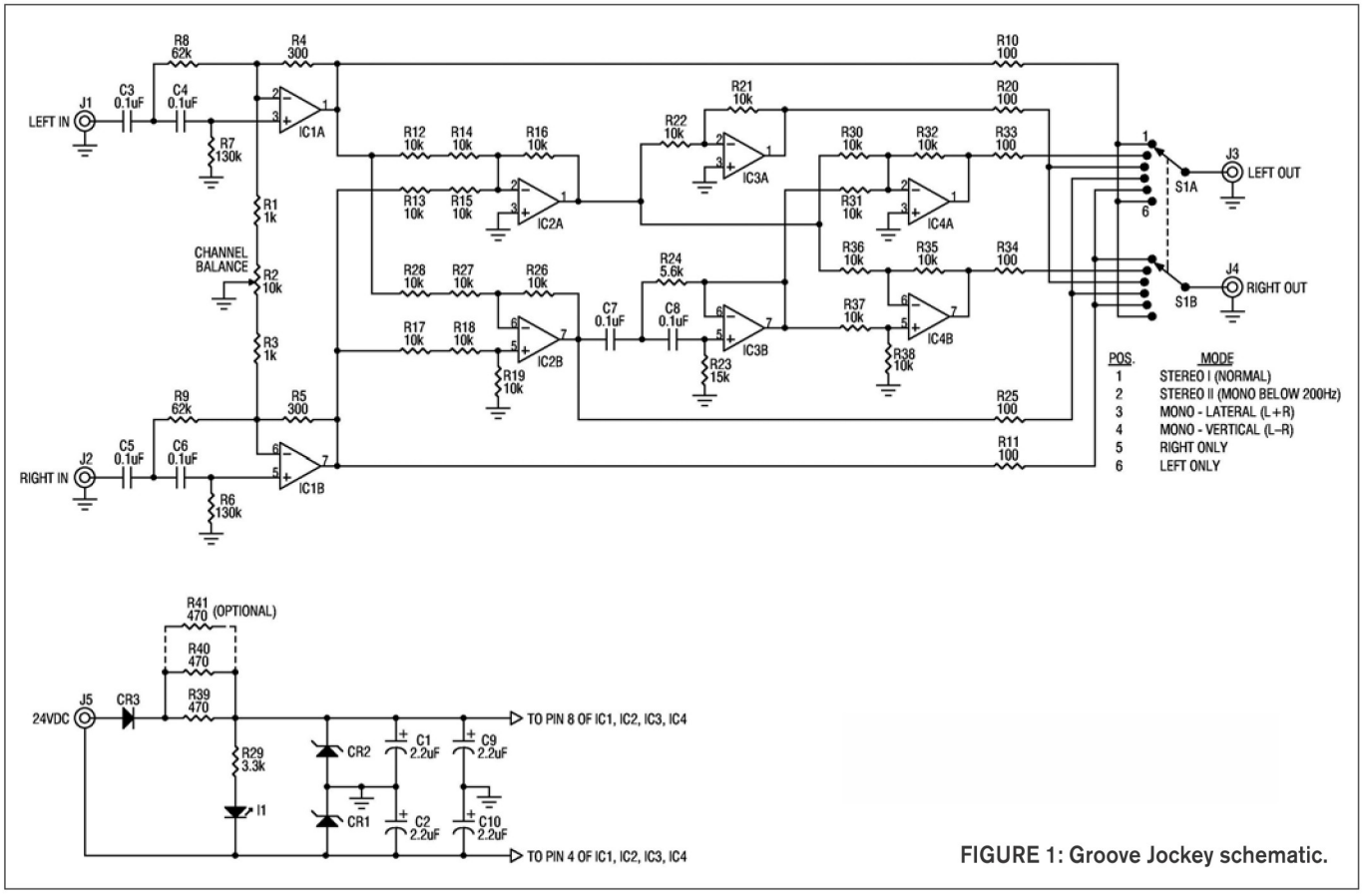

FIGURE 1: Groove Jockey schematic.

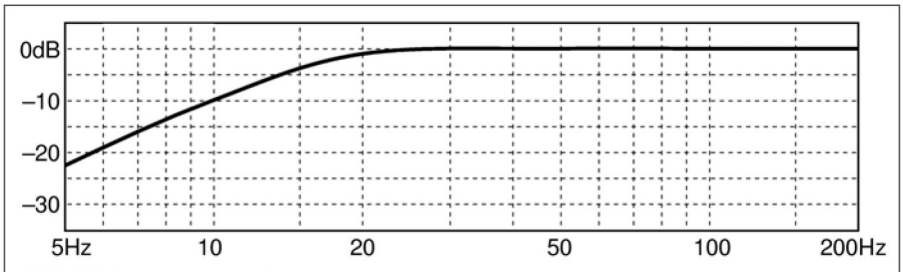

FIGURE 2: Input stage high-pass response.

reduce low-frequency offsets caused by warped records.

The two sections of IC2 are configured as a "matrixing" amplifier. IC2A sums the left and right channels to L+R mono, and IC2B derives an L-R "stereo-difference" signal. The gain of these stages is -6dB to maintain the sum and difference at the same level as either stereo channel.

The four instances of connecting two 10k resistors in series to make 20k resistors may seem odd, but as all 10k resistors in the matrixing networks should have a 1% tolerance, doubling-up on the one value just makes the parts procurement process a bit easier. All other resistor values may be the more common 5% carbon-film type.

The L-R stereo-difference signal (the vertical component of disc playback) is fed to low-pass filter stage IC3B. Starting at about 200Hz, low frequencies are rolled off at a rate of 12dB-per-octave. Again, properly mastered stereo records should have very little vertical component energy at low frequencies to begin with. Following the filter, the L+R and L-R components are de-matrixed back to discrete left- and right-channel signals by IC4, and then sent to the output jacks.

Figure 3 shows the effect of this filter for both lateral and vertical modulation with test signals applied to the input jacks. The two curves represent either a left- or a right-channel output when the inputs are driven in-phase (lateral) and out-of-phase (vertical). For left-only or right-only music components under the filter's influence, the left and right outputs will be complementary; that is, the two channels will add as acoustical energy in an ideal (anechoic) listening environment to give flat overall single-channel response. In a typical listening situation this means that the directionality of the sound source will be somewhat nebulous from 200Hz downward, but this is pretty much the case anyway, and of course there shouldn't be a lot of single-channel, low-frequency energy from the record to begin with.

MONO - LATERAL (L+R)

This third switch position is useful for playing monaural records with a stereo cartridge. Vertical rumble and certain types of disc surface noise are substantially reduced by summing the left and right channels. Also, the pops and clicks from mono discs sound less annoying when they come from the same virtual-center position as the mono music, rather than bouncing between left and right as the stylus encounters dirt and scratches on one groove wall or the other.

In the mono-lateral playback mode, the L+R summing stage—IC2A—feeds both outputs. But because IC2A is itself an inverting amplifier, another phase inverter—IC3A—is included in the signal path to keep the absolute phase of the monaural signal consistent with the stereo playback mode.

MONO - VERTICAL (L-R)

Collectors of old records may own some early "hill-and-dale" recordings; these include Edison, Pathé, and a few other labels from the very early days of the phonograph. The World library of broadcast transcription pressings also employed vertical modulation. The Groove Jockey plays these nicely in this mode, because a stereo cartridge responds equally in both the vertical and lateral axes.

Keep in mind, however, that these old records have wider grooves than LPs. Your fragile hyperelliptical stylus is not going to fit the groove properly and may be sheared off by a piece of grit whizzing along at 78 rpm groove velocities. Stanton, Shure, and some aftermarket stylus manufacturers can supply the proper wide-groove playback stylus for select stereo cartridges.

Listening to a contemporary stereo recording in the L-R playback mode is both interesting and useful. You can immediately hear that there is actually very little energy in that component of a stereo signal (L-R) that actually makes it stereo. All lateral (mono) modulation is effectively cancelled in this mode, and what you will hear is lots of reverb in pop music and very little of anything below 200Hz. The quite audible distortion that you will be surprised to hear in the L-R "difference" signal will make you wonder how the record can sound as good as it does in stereo!

The best way to set the channel balance control, R2, is by listening to a monaural record in the L-R playback mode. Simply tweak R2 for minimum audio out of the speakers. You can also make this adjustment with a stereo pop-music record. Just turn R2 until the lead vocalist, almost always panned to the center position, disappears from the mix.

You can also fine-tune tonearm lateral bias, the "antiskating" adjustment, when listening in the L-R mode. Audible distor-

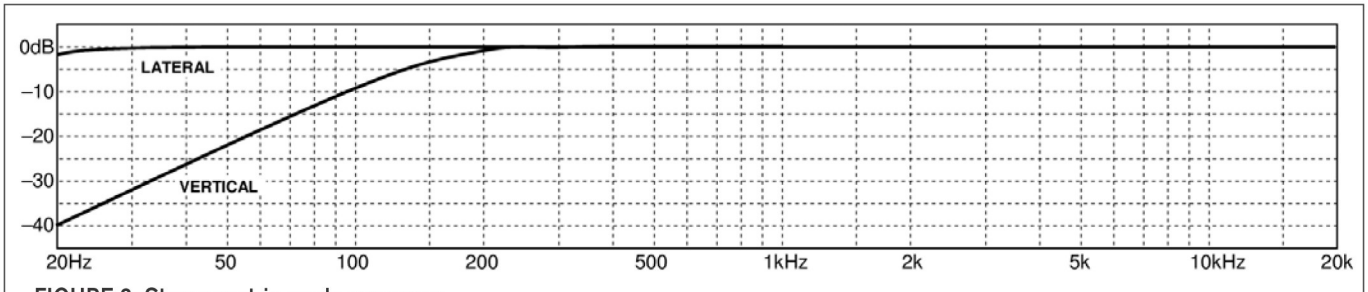

FIGURE 3: Stereo matrix mode response.

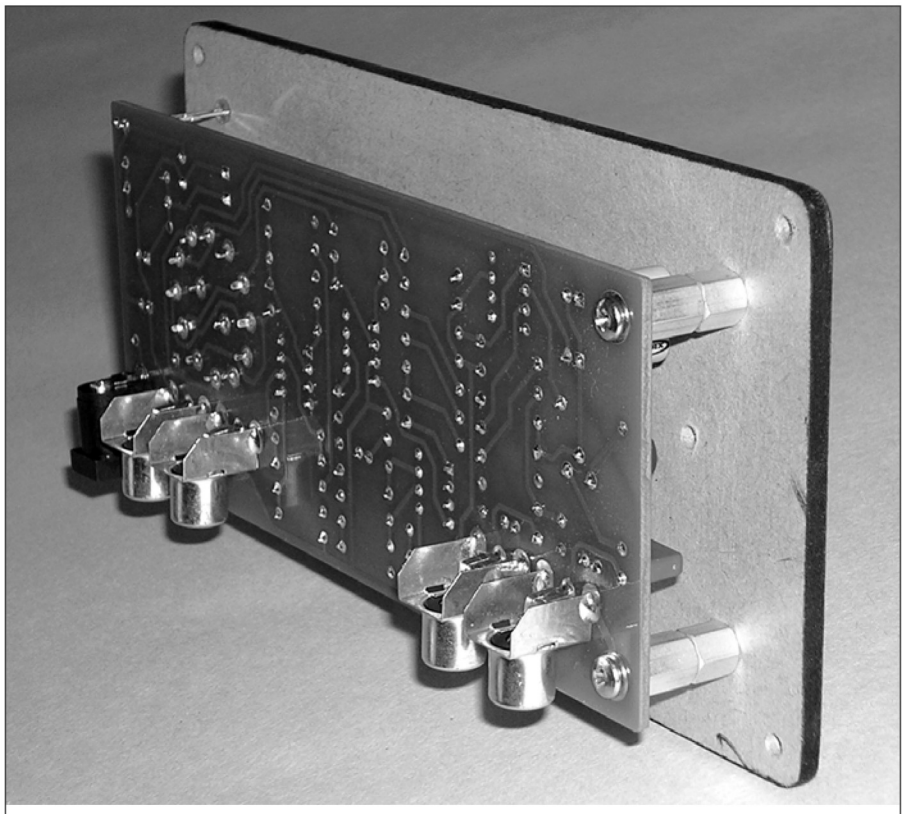

PHOTO 2: Rear view of the circuit assembly "sandwich." Note that the connectors are mounted on the back of the circuit board.

tion will drop to its minimum when the stylus bears equally on both groove walls.

The vertical mode routes the output of IC2B, the L-R difference stage, to both the left and the right output jacks. The perceptive reader will be quick to note that IC2B actually provides R-L at the outputs, and that another inverting stage, such as IC3A, would be required to correct this labeling error. But another stage wasn't handy, nor was an extra switch section to utilize IC3A for this purpose. This oversight doesn't compromise the ability to listen to old Edison records, however.

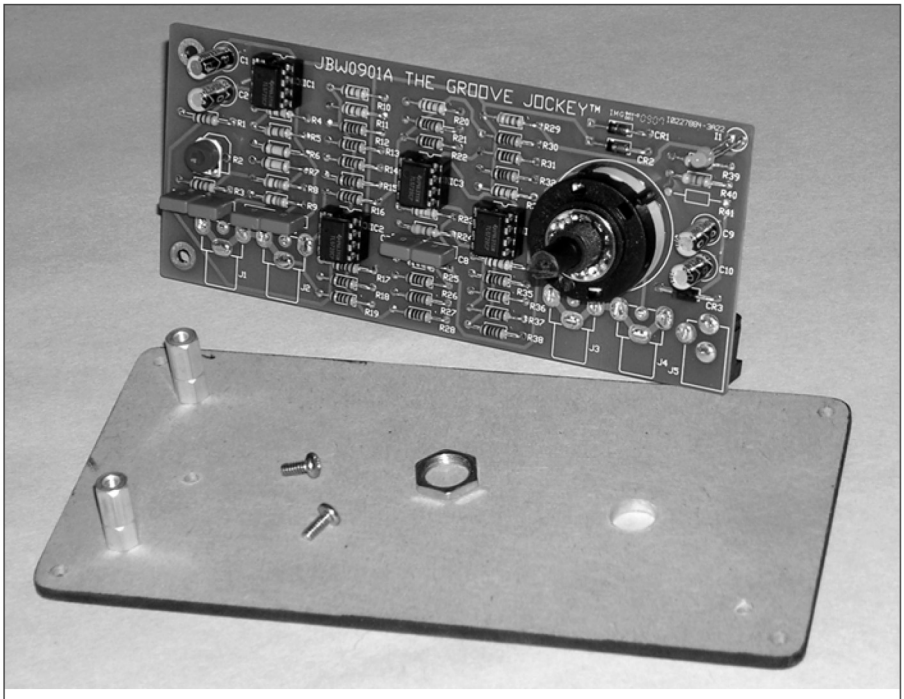

PHOTO 3: Circuit board and front panel prior to final assembly.

RIGHT ONLY AND LEFT ONLY

You can route a single stereo channel—either the left or the right—to both outputs in these two playback modes. This may not seem a terribly useful feature, but it does come in handy for some listening tests and also allows optimum use of a monaural cartridge.

Despite the utility for playing monaural records that the Groove Jockey provides in the mono-lateral (L+R) mode, some record collectors believe that mono records are best played with a monaural cartridge. Indeed, the high compliance of most stereo cartridges, especially in the vertical plane, seems inconsistent with the groove of an old 78 hurtling past at 3 or 4′ per second. Any warp in the record can send the stylus to the limits of its travel. I have found that an old General Electric VR-II cartridge from the 50s frequently does a better job of playing 78s than my Stanton 681-EEE with a 3-mil stylus.

Simply wiring a mono cartridge to both inputs of the phono preamp would double-terminate the cartridge in two 47k load resistors, the common input characteristic for most phono preamps. Instead, simply connect the mono cartridge to one input—the left, for instance—and set the Groove Jockey switch to left-only to get music from both speakers.

CONSTRUCTION NOTES

The Groove Jockey can use "jellybean" (garden-variety) op amp integrated circuits, but installing these in sockets allows you to substitute any of the many different types available with this common dual-amp pinout. The power supply is bipolar ±10V, giving more than 20dB headroom for the nominal -10dBu output level from most phono preamps. The TL072 chips specified in the parts list draw very little quiescent current from the supplies, but other types may have a higher idling current. If the power supply voltage measures below ±10V, additional current is required, so install the optional 470Ω resistor, R41, on the board.

The Groove Jockey is powered by a 24V "wall wart" AC adapter. The bipolar voltage for the op amps is derived from a couple of 10V zener diodes in an "artificial ground" configuration. Op amp ICs

exhibit very good rejection of power supply ripple, so additional regulation and filtering is overkill for this simple device.

All components specified in the parts list are relatively inexpensive and include their corresponding Mouser Electronics ordering codes.

Photos 2 and **3** show how I mounted the finished circuit board on the front panel, which I cut from a piece of 1/8″ Masonite. A panel drilling template has markings for all the holes, plus the black border line that defines the panel dimensions. You can inkjet-print this template onto plain paper and then stick it to the Masonite with 3M "77" spray adhesive. Drill the six 1/8″ holes and the one 3/8″ hole through the paper template, and sand the panel to the outer boundary of the black border line.

You can easily remove the template from the panel by brushing it with paint thinner and lifting it off. Then you can countersink the two holes for the threaded standoffs from the front, and install two 4-40 × 5/8″ standoffs on the rear of the panel with flathead screws (**Photo 3**). I didn't have 5/8″ standoffs, so I used a combination of ¼″ and 3/8″.

Print the panel artwork overlay on good-quality paper; I used a fine letter paper that resembles parchment. When the ink is dry, it's a good idea to cover the paper overlay with laminating film or clear Con-Tact brand shelf paper before finally sticking it down with the 3M spray adhesive and trimming around the edge with a razor blade. You can easily open up the holes with an X-Acto knife.

Screw the assembled circuit board (with the RCA and power connectors on the back of the board) to the panel standoffs and further secure it with the rotary switch bushing. The switch is just the right height with the supplied lock washer, but you must break off the plastic locating pin so that it doesn't bear against the back of the panel. Be careful not to change the position of the switch stop mechanism! The 3/8″ switch nut secures it to the panel.

The Groove Jockey circuit assembly slips into a cutout just a bit larger than the 2¼″ × 5¼″ circuit board. I mounted mine with #4 woodscrews in the turntable console cabinet (**Photo 4**).

You can download a PDF file containing the drilling template, panel artwork, and circuit board layouts at www.inovon.com/jim/groove.pdf. Simply print the panel art with a laser or inkjet printer. You can also print and photograph the circuit board artwork, or take it to a film-output shop where a true film negative can be produced directly from the PDF. *aX*

PHOTO 4: The Groove Jockey installed in the author's turntable console cabinet.

PARTS LIST

REFERENCE NO.	DESCRIPTION	MOUSER NO.
C1, 2, 9, 10	Capacitor, radial electrolytic; 2.2µF, 50V	140-XRL50V2.2-RC
C3-C8	Capacitor, radial polyester film; 0.1µF, 5%, 63V	505-MKS2.1/63/5
CR1, CR2	Diode, zener; 10V, 1W: 1N4740A	512-1N4740A
CR3	Diode, rectifier; 1A, 50PIV: 1N4001	512-1N4001
I1	LED, T-1, yellow	638-204YD
IC1-4	Dual op amp integrated circuit; TL072	595-TL072CP
J1-4	RCA jack, right angle PC-mount	161-4216
J5	DC power jack, PC-Mount, 2.5 × 5.5mm	163-7625E
R1, R3	Resistor, ¼W carbon film, 1k	291-1k-RC
R2	Trim potentiometer, 6mm, 1k	652-3306P-1-103
R4, R5	Resistor, ¼W, 5% carbon film, 300	291-300-RC
R6, R7	Resistor, ¼W, 5% carbon film, 130k	291-130k-RC
R8, R9	Resistor, ¼W, 5% carbon film, 62k	291-62k-RC
R10, 11, 20, 25, 33, 34	Resistor, ¼W, 5% carbon film, 100	291-100-RC
R12-19, 21, 22, 26-28, 30-32, 35-38	Resistor, ¼W, 1% metal film, 10k	271-10k-RC
R23	Resistor, ¼W, 5% carbon film, 15k	291-15k-RC
R24	Resistor, ¼W, 5% carbon film, 5.6k	291-5.6k-RC
R29	Resistor, ¼W, 5% carbon film, 3.3k	291-3.3k-RC
R39, R40	Resistor, ¼W, 5% carbon film, 470	291-470-RC
S1	Switch, 2 pole, 6 position rotary	105-SR2612F-26-38S
(misc.)	4 8-pin-DIP IC sockets	571-1390261-2
(misc.)	Knob	450-1017-GRX
(misc.)	Power supply, 24V DC	412-124013
(misc.)	2 threaded hex standoffs, 4-40 × 5/8″	534-1808
(misc.)	2 4-40 × ¼″ flat head screws	
(misc.)	2 4-40 × ¼″ pan head screws	

A Visit with **Dave Weems**

Meet the man (right), who, for over 50 years has shared his knowledge of, and experiments with, speakers in many periodicals and books.

For five decades, the name David B. Weems has been known to the amateur speaker builder. From his first article in *Audio* in 1954, to his last book in 2000, Dave Weems has made major contributions to the hobby, and inspired generations of speaker enthusiasts to try his designs or originate their own. I had the privilege of interviewing Mr. Weems in his home in Neosho, Mo., in March 2009, and what follows is a distillation of that interview.

Bird: You are originally from this area of Missouri. When were you born?
Weems: I was born in 1922, in the country near Granby, Missouri, and after a year my parents moved to Newtonia, Mo., which is just east of Neosho.

Bird: So you were raised in Newtonia.
Weems: Yes, my father was a farmer near there and I went to school at Newtonia.

Bird: That was until WWII?
Weems: No. I first attended Drury College in Springfield, Mo., where I studied chemistry, physics, math, and German, as I was interested in a degree in physical science.

Bird: Did you serve in WWII?
Weems: After Pearl Harbor a friend and I were selected to go to St. Louis to be tested for the NAVY V5 flyer program. They rejected me due to an old ankle injury (it had once been run over by a truck). However, when I was called up, the Army did not find it a hindrance for 25-mile hikes. I entered the US Army at Camp Crowder, near Newtonia in 1943, and served in Europe with Patton's Third Army. They discovered I had studied German in college, so they made me an interpreter.

Bird: What did you do after the war?
Weems: After being discharged in February of 1946, I eventually ended up in southern California, and enrolled at UCLA, to complete my bachelor's degree. I got a job working as a lab technician for the UCLA College of Agriculture, doing soil physics. I later worked temporarily for the Engineering College at UCLA on a defense contract and that job ended when the funding stopped. Other than that, most of my jobs were teaching—first general science and later physics and chemistry. I also taught a photography class for a few years. Most of my writing has been freelance except for writing a manual for a water and wastewater school.

I met my wife, Charys, at UCLA. She was from Beverly Hills, and we were married shortly after I graduated.

Bird: Do you have any children?
Weems: I have a daughter Helen, who studied music at UCLA and worked for the musicologist Mendel Hood, who was also our best man. I think her interest in music was inspired by hearing a number of good speaker systems under development as she grew up. Helen now lives in Columbia, Md., where she teaches music.

Bird: How did you get interested in audio?
Weems: I had an old five-tube radio with a 4″ PM speaker, and did not like the sound quality. I built a larger baffle and used a 6″ speaker. The improvement in sound was dramatic to me, and I was hooked.

One of my first projects involved some 12″ speakers I found at a surplus dealer in Hollywood. They were very cheap at $2 apiece. I experimented with them, and found the low end was lacking. They had very stiff paper cones and high resonant frequency. I had just enough knowledge of physics, and from reading *Audio,* to know that a lower resonance would produce better bass.

I experimented with various modification methods and finally settled on replacing the surround with chamois material. I cut the original outer edge suspension away and glued the chamois in its place. It produced a very low resonant speaker and was the basis for my first article in the Sept. issue of *Audio* in 1954. An interesting aspect was that I used a clothes sprinkler head for the dust cap and it worked quite well. (**KB**: For the article, see *Audio Anthology, Volume Three,* p. 13, Audio Amateur Publications, 1990.)

It developed into somewhat of a business, as I had requests from local audiophiles for the modified speakers. I sold a number of them, and on some weekends my apartment was besieged with customers. In those days, there was not too much literature available on speaker building, and there was a lot of disagreement on

tuning a speaker cabinet. My chamois-modified driver taught me a lot about housing speakers.

I remember one customer, a lady from Pasadena, who had bought one, and it was mounted in a sealed or infinite baffle. She called and complained that the speaker was now distorting at higher levels. I took the train up there, and while onboard I thought about her problem. She had a small 1 or 2W amplifier in her system at the time. A few months later she got a larger amp, I believe it was 25W, and I thought it may be overloading the speaker.

That's when the thought of reducing the cabinet volume hit me. When I arrived, I fired up her new amplifier and, sure enough, the speaker was being driven beyond the magnetic field. After some thought, I decided to try and reduce the volume of the box. I asked her whether she had any loose bricks on the property. She said no, but after looking around I found the wood box for the fireplace. The logs were cut small enough to fit into the box. I added wood to the cabinet until it was reduced one half or better, closed up the back, and it reduced the distortion, even at higher volumes. I had discovered what air loading of the speaker could do.

Bird: What did you do after UCLA?
Weems: My experience in agriculture at UCLA had led me to running a small plant business, and to an ad in a plant industry magazine about a firm in Boerne, Texas, looking for someone with an interest in the business. We moved to Texas but it did not work out too well, so I quit. I worked in a radio-TV repair shop for a while, and then we returned to California, where I taught high school science in San Diego.

Bird: Did you keep up the speaker interest?
Weems: I did, and my second article was accepted by Oliver P. Ferrel at *Popular Electronics*. I used a 4 × 4 sheet of Celotex® to produce a labyrinth type corner enclosure using a cheap 8″ speaker.

Bird: I know that article (I then produced a copy of the November 1955 issue of *Popular Electronics*). Dave, that was *my* first speaker building project, and I got in trouble with my mother for cutting up that Celotex in her kitchen.
Weems: That was a very popular article that generated a lot of mail for the magazine. It was featured at the L.A. Audio Fair in 1955, with a blow-up photo of the enclosure. It was followed up with a 12″ version called the $3 baffle and later a 15″ version, using two 8″ and two 2″ tweeters. It was called the Hi Five.

Editor Ferrel and I became friends, and I wrote for *PE* for about 15 years.

Bird: When did you come back to Missouri?
Weems: In 1957 after my father suffered a stroke and needed help. I taught school in Newtonia for ten years and later we had a goat farm on land we purchased before we left California. We later sold the land to the Newtonia Battlefields Protective Association. There were two Civil War battles at Newtonia, including the last battle west of the Mississippi.

I once received a good job offer with Bendix in Kansas City, but my wife liked Newtonia so much she asked to stay here.

Bird: Did you keep up with speaker building?
Weems: I continued experimenting and writing articles for *Popular Electronics* and other magazines. I wrote one for *Radio Electronics* in the 1970s. The editor had sat on it for five years and in the meantime the Thiele/Small data was taking hold in the hobby. I got letters saying my article was obsolete in view of the new data. I called the editor and told him not to hold my articles that long.

I wrote my first *Speaker Builder* article on a double chamber system in the mid '80s and followed it up with another on a small transmission line enclosure using a 4″ woofer. I also built a number of tube amplifiers but never wrote any articles about them, and I stopped doing tubes when solid-state came along.

Bird: Where did you obtain most of your drivers?
Weems: In the early years by mail order from McGee Radio, Burnstein-Applebee, and other catalog suppliers. Later it was Radio Shack, MCM, Meniscus, and Madisound. Radio Shack kept changing suppliers and some of their speakers, and while having the same part number, they had much different specifications that could affect the design.

Bird: Did you ever do any consulting for a speaker manufacturer?
Weems: I never did any consulting for manufacturers. I received probes, via publishers, with the possibility of jobs, but in all cases I would have had to relocate and so I was not interested. In 1970 I attended CES in Chicago with Perry Ferrell and the technical editor of *Popular Electronics*. After the show we made a tour of some manufacturing plants in Michigan, including Electro-Voice and The Heath Company.

In 1980 I again attended CES and met with representatives from TAB books and Radio Shack. We planned the book that Radio Shack subsequently published for a number of years.

Bird: Some critics have said that you never provided usable measurements with your projects.
Weems: During my early days of working with speakers, I had little confidence in measurements because I found some designs developed that way did not work out. Perry Ferrell was asked why we did not supply measurements and he gave the opinion they weren't reliable for comparing speaker systems. That is, the kind of measurements I could do. One difference in the 2000 book I wrote with G. R. Koonce (*Great Sound Stereo Speaker Manual*, TAB), is that he did extensive testing to provide driver files and to check the complete projects that we published. However, I did the final "voicing" of the projects. I could not do that now because of my hearing loss at high frequencies.

Bird: When did you write your first book?
Weems: In the late 1970s, an editor for Tab Books called me and said that the editors were interested in a book on building speakers. That resulted in a number of Tab books which were very popular, including one that was based on Radio Shack drivers. My last book was with G. R. Koonce, and it included a CD with a number of system design programs. It was not as successful as I thought it would be, even though Koonce added a lot of value to the book, especially with his CMP (Crossover Modeling Program).

I also did a car restoration book for Tab with a co-author, Mike Murphy. I had been interested in cars in my younger days, but Murphy knew more about cars than I did. The book was poorly produced by Tab, with low-grade photos.

I also wrote a book about goats, *Raising Goats—The Back Yard Alternative*. It also wasn't a great success, because Tab Books probably was not a logical choice for such a title. I think it would do much better today, because there is more interest in goats now (the book is now selling for over $70 on the used book market).

Bird: Your last speaker book may have been a victim of the times. We have been going through an audio cultural change, due to the computer, the Internet, home theater technology, and the iPod. Most of today's audio is being heard through earbuds or in automobiles, and the mainstream young people don't want to, or can't, build their own speakers. Up to a few years ago, prospective builders could go into Radio Shack and get a pretty good driver. Today, Radio Shack, having deserted the electronics hobby market, no longer sells raw speakers, and phone or Internet ordering is the only source now.

Weems: You may be right. G. R. Koonce sent me an e-mail recently in which he stated we speaker builders have become "dinosaurs."

Bird: Of all your projects, did you have a favorite you personally liked the sound of?

Weems: I think the system I liked best was the "Totem Pole" system in the *Popular Electronics* January 1966 issue. They were two columns with 16 4″ speakers in each line and a University dome super tweeter. A separate enclosure housed two 8″ woofers, making it a three enclosure system. I had an accident with this system and damaged the towers, so I went on to other projects.

Stereo was fairly new then and it may have colored my thinking. We liked them for massed voices. I would place some of the "pipes" I did as favorites. Maybe because of the column enclosure they had good imaging. I produced several line arrays for small PA systems that always seemed to work well. I did one for a church in Newtonia, and the people who used it said it gave better performance for voice than anything they had tried.

When G.R. and I were working on my last book, I did two small pipes with 4″ woofers. I think I liked the sound of the SEAS-SP-RCY woofer better than the project we published, but I had noticed a habit of SEAS woofers to change fs with time, while the AUDAX model did not. I just didn't know how much difference that would make in the sound of a pipe.

Bird: The use of 16 4″ speakers was similar to what Jim Kyle did with his "Sweet Sixteen" speaker in the early 60s. [The Sweet Sixteen wasn't a column and sounded terrible.—ETD]

Weems: Mine was a column arrangement, and some people still incorrectly credit me with the original "Sweet Sixteen," even though my design was years later.

Bird: Do you still build speakers?

Weems: When my wife developed lymphoma in 2002, I stopped my speaker work to take care of her. My wife was a great woman who put up with a lot of my crazy speaker building and projects. She passed away in 2006 and I have yet to really get started again. My hearing is not too good any more, and that makes tuning difficult. I still read *audioXpress* and enjoy the articles on speakers.

KB: Thanks, David, for sharing your speaker building years with us. *aX*

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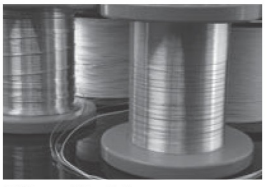

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The Jack 10, Part 2

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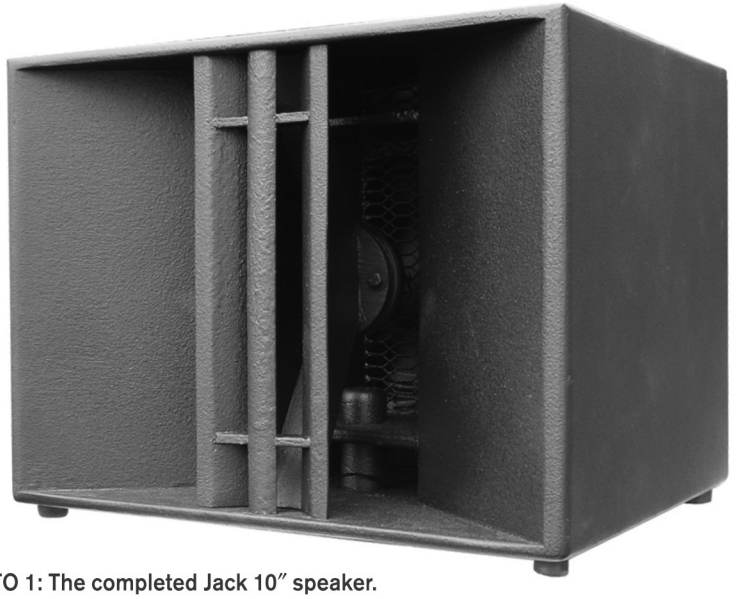

PHOTO 1: The completed Jack 10″ speaker.

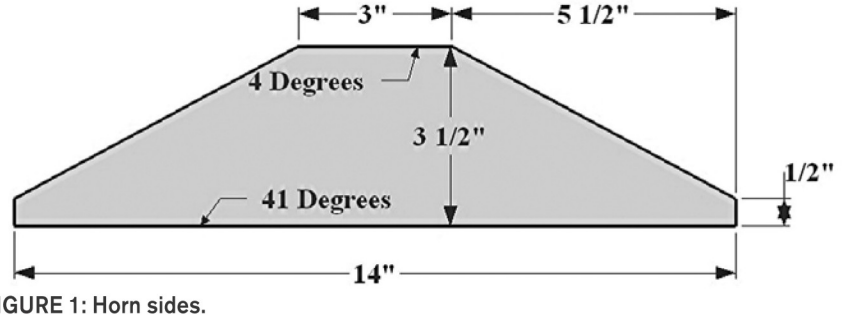

FIGURE 1: Horn sides.

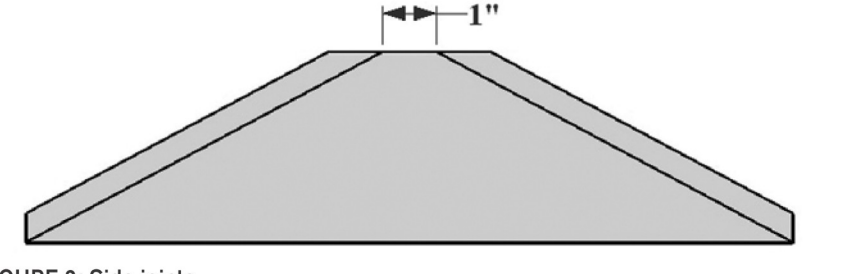

FIGURE 2: Side joints.

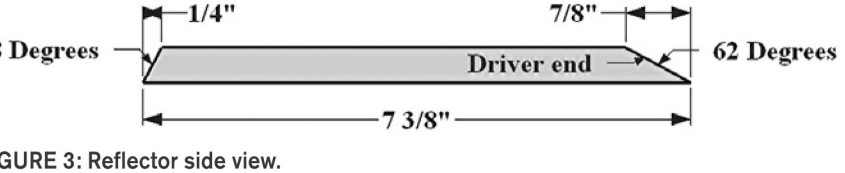

FIGURE 3: Reflector side view.

Last month I showed the woofer horn section of the Jack 10 (**Photo 1**), which you can leave tweeter-less for electric bass use. For PA or instrument use where a full-range response is desired, a high-frequency section is required. A vertical horn loaded by an Eminence ASD 1001 compression driver allows you to stack multiple Jack 10s in a line array, for superior cab-to-cab integration and system pattern control.

CONSTRUCTION

The parts are small, requiring great care to cut safely; a zero-clearance panel cutting jig on a table saw or a band saw is highly recommended. From quarter-inch or half-inch plywood, cut out two horn sides (**Fig. 1**). Note the angles, facing opposite directions. The diagram shows the inner face, which is the wider part.

Draw on the inner faces the joint lines with the reflectors (**Fig. 2**). Cut two reflectors from half-inch plywood. Start with pieces 2 to 3″ wide and cut to length per **Fig. 3**. Cut the reflectors into trapezoids, with the sides angled at 4° and the resulting wider face on the inside of the horn, as shown in **Fig. 4**.

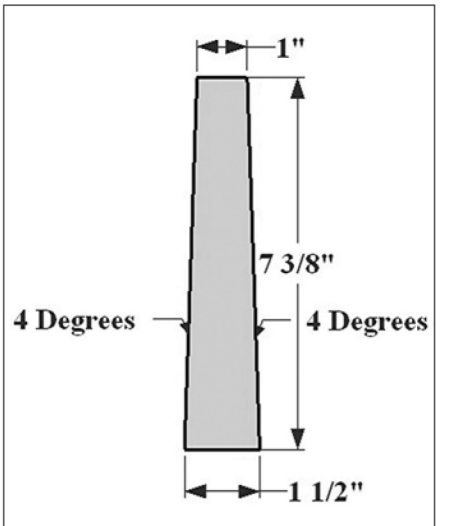

FIGURE 4: Reflector front view.

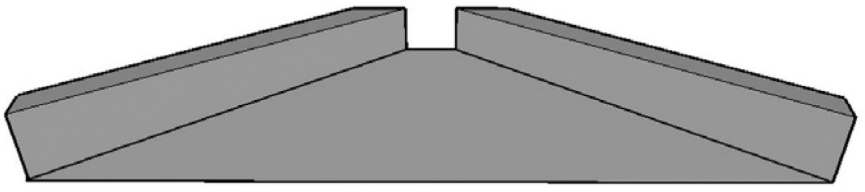

FIGURE 5: Reflector positioning on side.

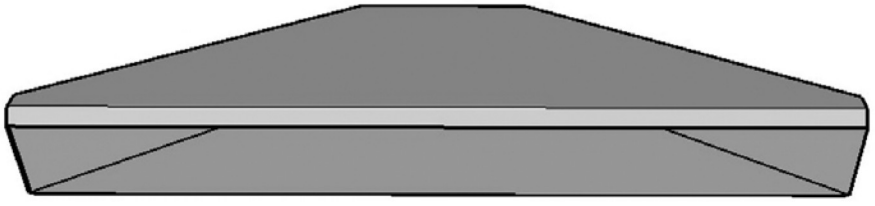

FIGURE 6: Second side attached.

The parts are too small to be easily fastened, so the best way to begin assembling the horn is with traditional woodworking glue, leaving the parts in place until the glue sets. Alternately, you may use quick-setting urethane glue, holding the parts in place while it sets. Sand/trim the reflectors for a good fit, coat one edge with glue, and set them in place on one side (**Fig. 5**). When the glue has set, the assembly will be stable enough so that you may use both glue and fasteners to attach the second side (**Fig. 6**).

Cut the baffle from quarter-inch plywood per **Fig. 7**. Attach it to the horn (**Fig. 8**). If you use a bolt-on driver, you may bolt it directly to the baffle. With a screw-on driver, you'll need a screw-on to bolt-on adapter.

From half-inch plywood cut two mounting flanges, 14¼″ long, with the cross-section shown in **Fig. 9**. From quarter-inch or half-inch plywood, cut two diffuser brackets per **Fig. 10**. Glue the brackets to the flanges per **Fig. 11**, with the brackets 2″ from either end of the flanges. This job also is easiest using woodworking glue.

Place the flange/bracket assembly face down on the workbench to set, putting waxed paper between the assembly and bench so that they won't stick together. Dry-fit the horn assembly to the flange assembly to be sure everything aligns, then glue the brackets to the flanges. When the glue has set, attach the horn to the flanges (**Fig. 12**). Use urethane adhesive here to fill any gaps in the horn to flange joints.

When the adhesive has set, attach the diffuser, a piece of ¾″ quarter-round molding, cut 14″ long (**Fig. 13**). On a tablesaw, or with a circular saw and sled, trim the ends of the assembly to finished height to match the cabinet. Cut a little from each end until the horn will just slide into place. The horn mounts centered in the cab, ½″ back from the front of the cab. Drill pilot holes for mounting screws through the cabinet top and bottom into the flanges.

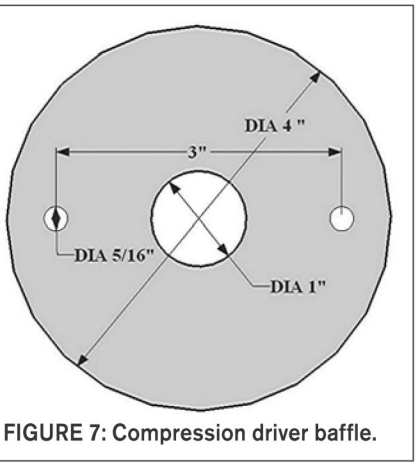

FIGURE 7: Compression driver baffle.

HOOK-UPS

The cab may be bi-amped, or passively crossed over. **Figure 14** is the schematic for an 8Ω 2kHz 4th order high-pass filter. The capacitors are 200V or better poly caps. The inductors are air core, of the smallest gauge available. A value tolerance of ±10% is acceptable.

Figure 15 shows a 2nd order 2kHz low-pass filter to be used on the woofer.

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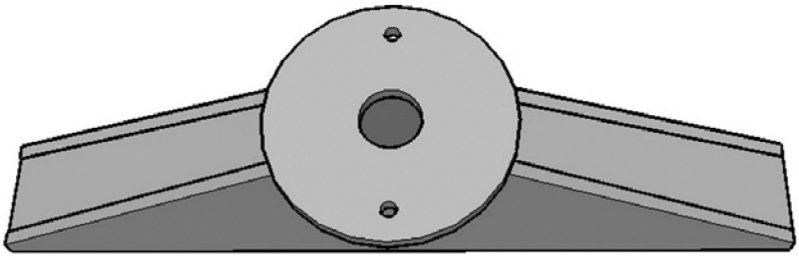


FIGURE 8: Baffle installed.

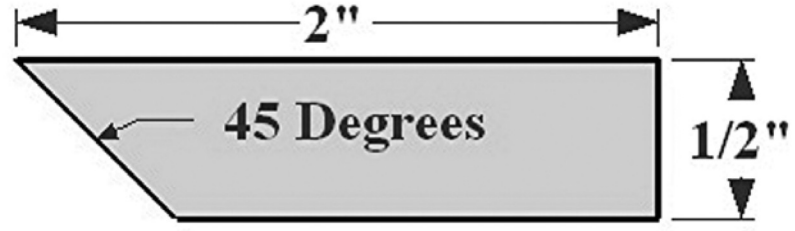


FIGURE 9: Mounting flange cross-section.

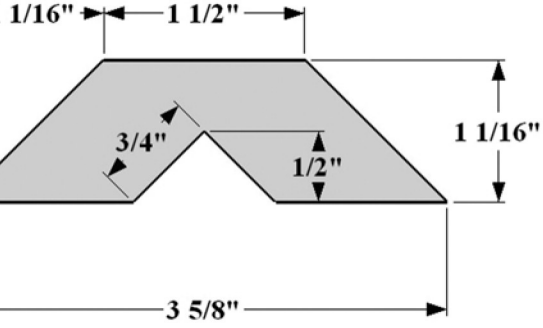


FIGURE 10: Diffuser brackets.

If you are bi-amping, omit the high-pass filter but still use a 200V 20µF polypropylene crossover capacitor in series with the HF driver for turn-on transient protection. You should still use the low-pass filter, because it's a Chebyshev configuration that increases woofer sensitivity near the top of its passband.

The inductor should be at least 16 ga., with a DCR of less than 0.4Ω. Because the capacitor is a shunt, a 100V rated non-polar electrolytic is adequate.

Figure 16 shows axial response with this configuration, compared with that of the similar size Peavey Versarray 112, which at $1200 is one of the least ex-

FIGURE 11: Diffuser/ flange assembly.

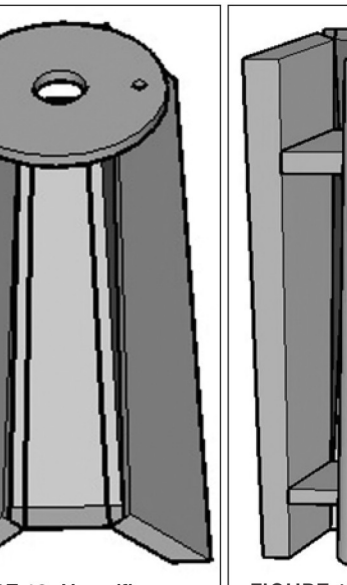


FIGURE 12: Horn/flange assembly.

FIGURE 13: Completed HF horn.

pensive commercial line array speakers available. **Figure 17** shows the Jack 10 response on-axis and 45° off-axis, which is no more than 6dB down all the way to 15kHz.

Whether bi-amping or using a passive crossover, before sealing the cab, test it using a sine wave generator or test CD signal at the crossover frequency. Try it with the woofer only, and then attach the leads to the HF driver. Then reverse-wire the HF leads. Whichever way sounds louder is the correct polarity. If you don't have a meter to test the level, have another person listen to the cab from a few feet away while you switch the leads. Also test the woofer using a 50Hz test tone at a fairly high level.

If you hear any buzzing, it might be caused by the cone hitting the throat filler, or by the suspension hitting the spacer. Sanding the filler will fix the first problem, while adding a layer of cardboard to the frame gasket will fix the second. As you build more cabs, stack the newer one atop the older. Run a 100Hz test tone through the bottom cab, and then plug in the top cab. The volume should go up. If it goes down, the polarity of the new cab is backwards.

You can add a top hat for pole mounting, installed forward of the baffle. Use adhesive to seal it airtight. It's OK to have the top hat extend through the duct plate. *aX*

FIGURE 14: High-pass filter.

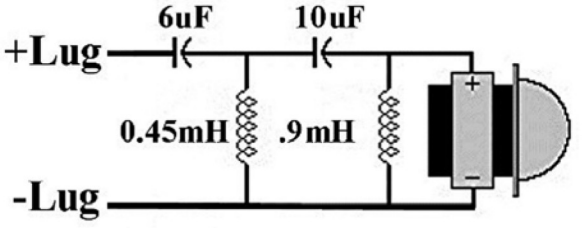
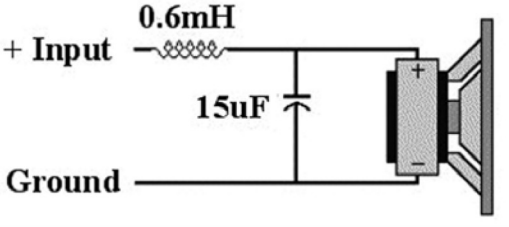


FIGURE 15: Low-pass filter.



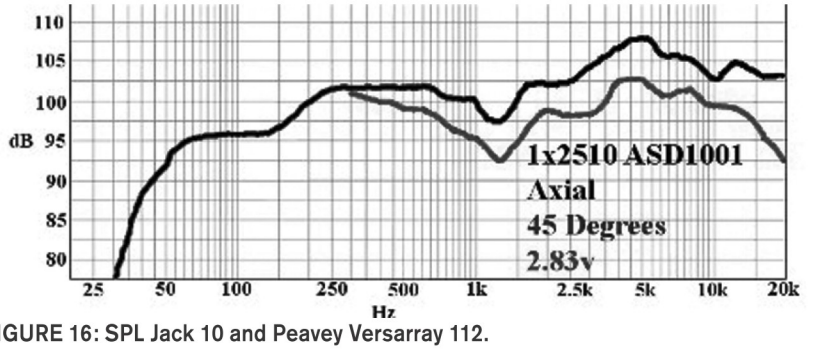


FIGURE 16: SPL Jack 10 and Peavey Versarray 112.

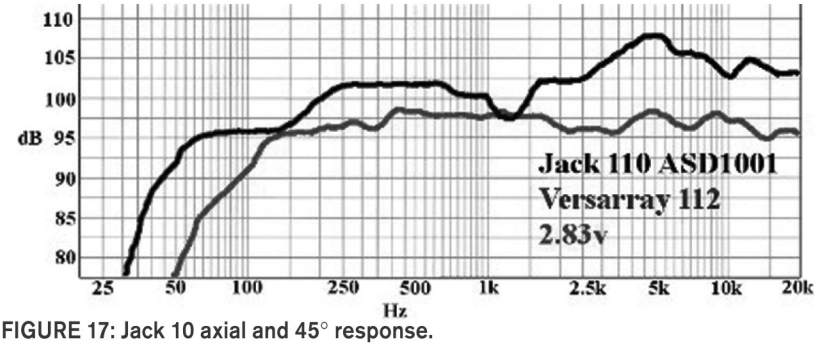


FIGURE 17: Jack 10 axial and 45° response.

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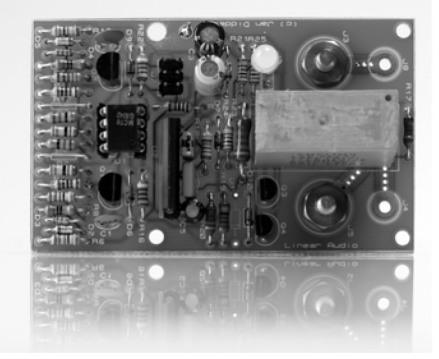

PHOTO 1: The PAX protection system.

In Parts I and II of this series about the paX amplifier, I developed a complete error-correction power amplifier. This article describes the protection system, which, constructed on a separate small circuit board, you can mount directly on the speaker binding posts (**Photo 1**). You can connect the system to the paX amplifier through a six-way flat cable, or use it with almost any amplifier provided you connect the appropriate signals. I will also discuss the Safe Operation Area (SOA) for the output devices and how it is modeled in the protection system.

A good protection system for an audio amplifier has two important tasks: speaker protection and amplifier overload protection. Speaker protection concerns possible excessive DC voltage at the amp output due to either excessive very low frequency signals or a circuit failure inside the amp. (Excessive DC output can also occur if you have a DC coupled power amp and your preamp gives out DC. If you go on a tour of your amp, it would be wise to insert—even temporarily—some input coupling caps.)

There are many convoluted circuits to disconnect the speaker from the amp through a high-power relay in case of significant DC offset at the output. These circuits often do double duty to delay the closing of the relay until after the turn-on transient, as well as immediately opening the relay when power is switched off—to avoid transient on/off thumps in the speaker.

I used a low-cost, but not well-known, IC: the uPC1237 from—among others—NEC. The NTE7100, from NTE Electronics, is pin- and function-compatible. It's an 8-pin Single-In-Line chip with all the functions mentioned above. It can handle two channels for a stereo amp, but I used one chip per channel to make it easier to construct the amp as dual mono.

For the speaker relay I used a little-known device specially designed for this duty and sold by the Dutch company Amplimo. The relay has two parallel internal contacts, one heavy-duty tungsten 100A contact and a smaller gold contact. When the relay closes, the tungsten contact closes first, and is then bridged by the gold contact. The reverse takes place when opening the relay.

CIRCUIT DIAGRAMS

The overall protection system is shown in **Fig. 1**. (The blocks for the Indicator and the Safe Operation Area protection are described later.) The uPC1237 has an internal 3.4V shunt regulator at pin 8, and the supply current is provided by R19 and R20. The relay switching output (pin 6) drives the relay through R18.

Jumper J6 sets the circuit operation. If jumper J6 is placed, the circuit becomes "re-entrant": if there is an error input, the relay falls off, and, after a short delay, is activated again. If, at that point, the error has disappeared, the relay stays activated; if not, it falls off again and the cycle repeats. If J6 is not placed, the relay falls off when an error is present and you must switch the power off and on again to reset the circuit.

DC OFFSET SHUTOFF

Pin 2 of U2 is the DC offset input. AC on the amplifier output signal is removed by R17-C3 and the DC offset is sensed. Although the uPC1237 has only a unipolar supply (the amplifier's VCC+), pin 2 does accept bipolar signals. The calculations are a bit complex because the internal positive and negative thresholds are not equal. It is all explained in the datasheet. With some care you can set the threshold for positive and negative offset to the same value, at 0.6V DC. This value is set by R17.

SWITCH-ON DELAY

You will also see that a sample of the AC from the supply transformer is connected to the uPC via R21 and D8 to C6. This part of the circuit delays the speaker switch-on after a power-up to make sure the amp has stabilized and there are no switch-on thumps. When the power is turned off, this signal will disappear immediately, before the supply capacitors have a chance to discharge. Doing it this way, the speaker relay will be opened before the turn-off thumps occur.

SAFE OPERATION AREA

The second task of the protection system is to guard the output devices against excessive low impedance loads and shorts. Michael Kiwanuka has written a very informative article on Safe Operation Protection in audio power amplifiers[1]. My own protection circuit is different, but was inspired by Mike's design and has many things in common with it. **Figure 2** shows the circuit, which receives two types of information from the power amp. The first is the I_c in the output devices through VprotP and VprotN (via connector J7 in **Fig. 1**). This actually is the voltage across the output stage emitter resistors. Second, through the components connected to the supply and Vout, a voltage is developed that represents V_{ce} of the output devices. (Note that the circuit values in **Fig. 2** are illustrative only. This article later explains the circuit in detail and how to calculate the various component values in a specific case.)

The purpose of the SOA protection is

to limit the combinations of V_{ce} and I_c for the power devices to a safe value. Normally, such a protection circuit would be arranged to take away the drive from the output driver transistors to limit the available output current. The protection transistor collector (Q1 or Q2, **Fig. 2**) would normally be connected to the base of the (pre) driver transistor in the output stage. Q1/Q2 turns on when its V_{be} exceeds 0.6V, shunting the driver base signal away. But in my experience, even with a V_{be} of only a few 100mV, the protection transistor starts to influence the driver base signal, resulting in increased distortion.

For this reason I decided to make this an on/off protection circuit: the protection transistors drive the "overload" pin (pin 1) of the uPC1237 through the dual optocoupler U1, an MCT6. If overload occurs, pin 1 of the uPC is set high via R1 (**Fig. 1**) and the uPC opens the speaker relay. I have the uPC set for auto re-entry by jumpering J6. In this mode, the speaker relay will be opened again if the overload is no longer present.

You can argue that the protection system should work by limiting the output in case of overload and not by disconnecting the speakers. I agree that for PA duty this would be preferred, but for home use it is a very minor nuisance, if at all. But doing it this way prevents influencing the music signal by the circuit when there is no need for protection activation.

In using this amp for about a year now, I have never seen the protection system disconnect the speakers, except for one case in which I had a problem with a balanced cable which put a signal with 2.5V DC on the amplifier input. I immediately heard the off cycling of the amplifier, so I switched it off and managed to find the problem. After I solved the problem, all was as it should be. So, for me, this works quite well.

STATUS INDICATOR

Then, last, there is a small bi-stable circuit that takes the relay drive voltage to drive a bi-color LED, which you can place on the front panel. The LED will be one color when the amplifier is in turn-on delay or in protection and another color if the amplifier is operating normally. This circuit is shown in **Fig. 3**. All components shown in **Figs. 1-3** are on the output/protection (o/p) board, shown in **Fig. 4**.

CONSTRUCTION

Before building the o/p board, you should first complete and test the amplifier you want to use. You need the amplifier to test the o/p board, which you can populate first with the DC and delay circuitry around U2. Start with the components shown in **Fig. 1**.

Next, construct the six-wire flat cable as shown in Fig. A in the Construction Guide (audioXpress.com). Be sure to orient the notches in the six-pin headers exactly as shown! (Consult **Fig. 5** for connections to a different amp than the paX.) Connect the two boards with the flat cable, again noting the header notch

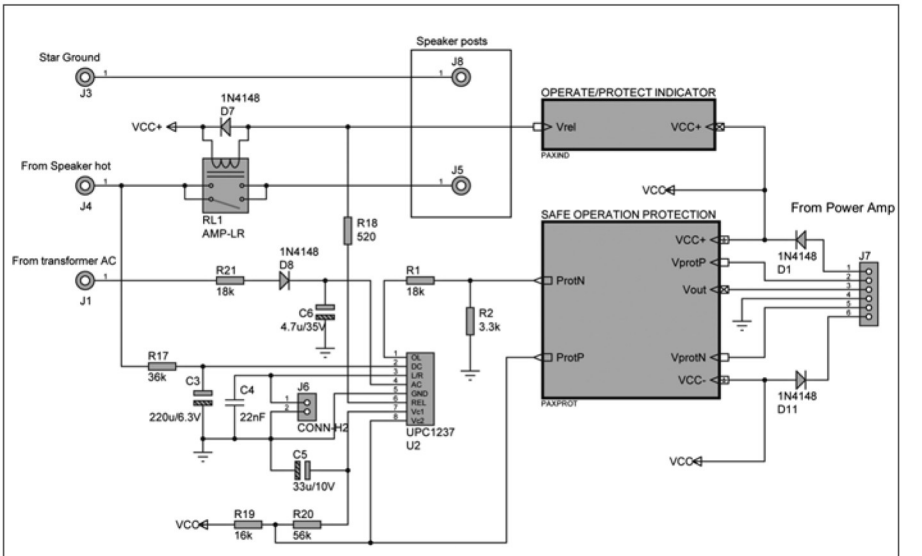

FIGURE 1: Overall protection system.

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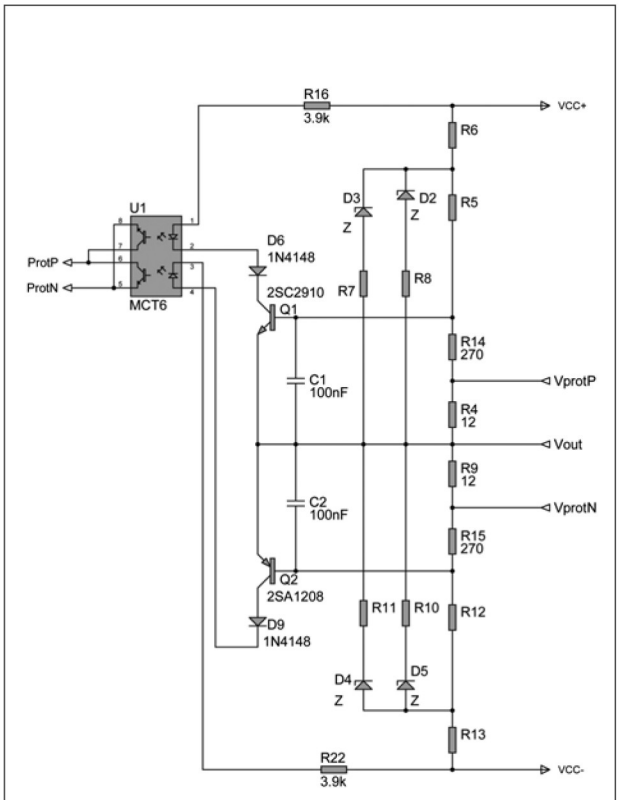

FIGURE 2: Safe Area Protection circuit.

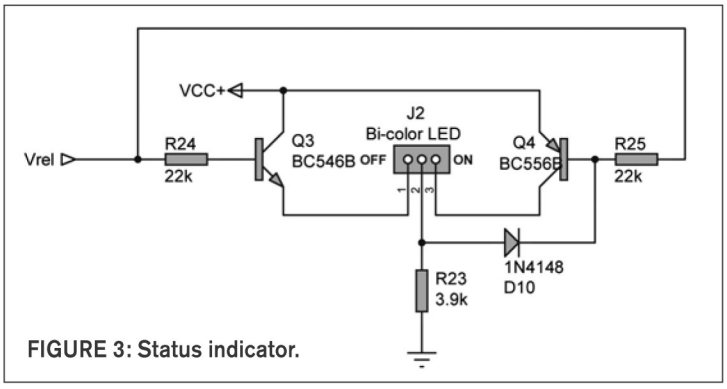

FIGURE 3: Status indicator.

position as indicated on the PCB silkscreen. Also connect one side of the power transformer secondary (the AC point) to the protection board at J1 as shown. Don't connect a speaker yet, nor the speaker wire from the amp to the protection circuit.

When you now switch on the amplifier, you should hear the relay pull in after a few seconds delay. If you find it too fast, you can increase C6 for a longer delay. If you switch off the amp, the relay should immediately open.

Next check the DC offset protection. Temporarily connect a resistor of 100kΩ from the positive or the negative supply to the speaker input point on the protection board (J4). The relay should fall off. Check this with both J6 set and removed to verify correct operation. Then, install all components for the status indicator in **Fig. 3**.

The PCB has labels "ON" and "OFF" at J2. That indicates which pin will be active at which status. If you use a red/green LED, you would connect the RED pin to J2/OFF and the GREEN pin to J2/ON. You may choose to repeat one of the tests described above to verify the correct operation of the status indicator.

The last part deals with the SOA protection components. Put all remaining parts on the protection board. Connect the flat cable, and connect the speaker output from the amplifier (amplifier J3 to o/p board J4), as well as the star ground wire (o/p J3 to supply star ground).

Switch on the amplifier and verify that it works normally. Then, to simulate an overload situation, connect a resistor of 10kΩ between the positive supply and the node Q1(base)/R14/R5/C1 and verify that the relay falls off. Do the same with the negative supply and the node Q2(base)/R12/R15/C2.

Note: If you don't mount the o/p board directly on the speaker output posts, you can connect the speaker return wire directly from the output post to the supply star ground without going through the o/p board. The speaker "hot" connection then goes from the o/p board J8 to the output post.

STAND-ALONE!

This article focuses on using this o/p board with my paX amplifier. But you can use it with any amplifier, provided you adjust the SOA circuit values as re-

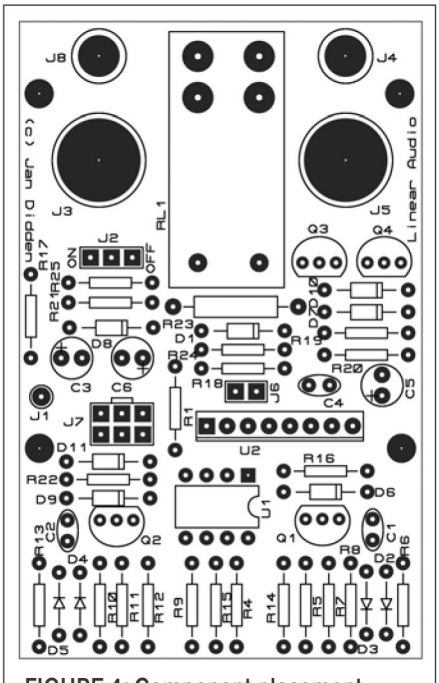

FIGURE 4: Component placement of the double-sided PCB for the o/p board.

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quired, as described in the sidebar. **Figure 5** shows the connections that you need to make from your amplifier to the o/p board through J7. *aX*

REFERENCE

1. Michael Kiwanuka: Transparent V-I Protection in Audio Power Amplifiers, *Electronics World*, Oct. 2002.

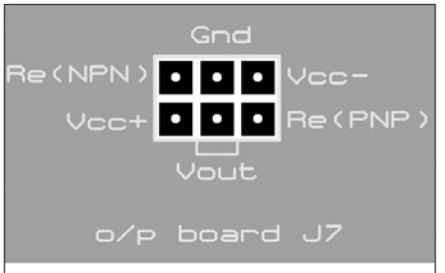

FIGURE 5: Connecting the o/p board to other amplifiers.

Components List Resistors

(0.25W metal film)
R1 = 18kΩ
R2 = 3k3Ω
R4, R9 = 12Ω
R5, R12 = 9k1Ω*
R6, R13 = 9k1Ω*
R7, R11 = 630Ω*
R8, R10 = 780Ω*
R14, R15 = 270Ω
R16, R22 = 3kΩ9
R17 = 36kΩ
R18 = 520Ω
R19 = 16kΩ
R20 = 56kΩ
R21 = 18kΩ
R24, R25 = 22kΩ
R23 = 3.9kΩ 1W

Capacitors

C1, C2 = 100nF ceramic
C3 = 220µF 6.3V
C4 = 22nF ceramic
C5 = 33µF 10V
C6 = 4.7µF 25V

Semiconductors

D1, D6-D11 = 1N4148
D2, D5 = zener diode 18V 0.5W*
D3,D4 = zener diode 15V 0.5W*
J2 = Dual color LED
Q1 = 2SC2910
Q2 = 2SA1208
Q3 = BC546B
Q4 = BC556B
U1 = MCT6 (Fairchild)
U2 = UPC1237 (NEC) or NTE7100 (NTE Electronics)
*: These are nominal values. See sidebar for calculations and adaptions to other amplifiers.

Miscellaneous

J6 = header 2 × 1 pin
J7 = 2 × 3 pin DIL flatcable header
RL1 = Amplimo -LS relay
DIL socket 2 × 4
SIL socket 1 × 8 pin
(See www.pilghamaudio.com for kits)

CALCULATING AND SETTING UP THE PROTECTION SYSTEM

The amplifier protection circuitry uses Safe Operation Area (SOA) protection. In this section, I address the specific design equations and how to develop the circuit values for the SOA part.

Figure A shows the various curves that you need to develop the SOA protection for two pairs of output devices. I_c (A) is on the vertical scale, V_{ce} (V) is on the horizontal scale. When V_{ce} is zero, it means that V_{out} is equal to V_{supply}. In practice, there will always be a few volts V_{ce} remaining at maximum output, but I will disregard that for the moment.

Curve B shows the safe combinations of DC I_c and V_{ce} for two parallel STD03 Sanken Darlingtons used in the paX amplifier. Curve C shows this for loads lasting no longer than 100ms. This means that combinations of I_c and V_{ce} that fall below and to the left of the curve are safe, but as soon as the curve is crossed, the protection system should activate. Because music is impulsive and not DC, you could argue that protection for curve C would be sufficient. If you use curve B you have an additional safety factor, but you are not using the amplifier to its maximum capability for music reproduction. I will, therefore, use some intermediate values.

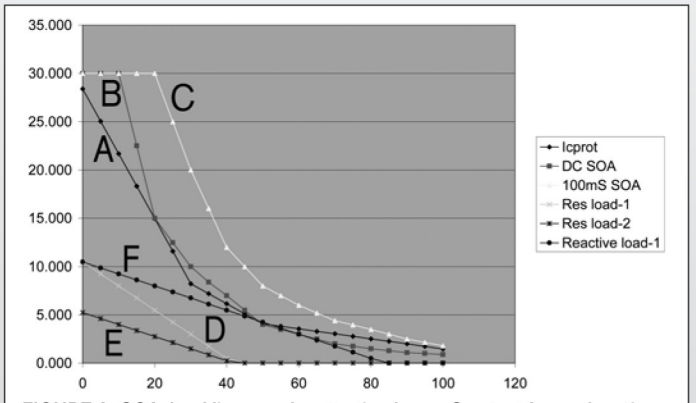

FIGURE A: SOA, load lines, and protection locus. See text for explanations.

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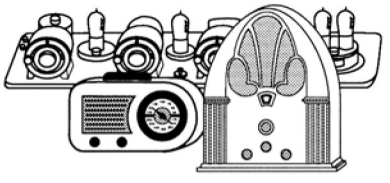

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Figure A also shows some load lines, which are combinations of I_c and V_{ce} required to drive a (resistive) load, assuming ±44V supplies. The lowest one, E, is an 8Ω load; Curve D is 4Ω. For example, in the 4Ω case, you see that when V_{ce} = 0, the output voltage is the supply voltage (44V in this case) and thus I_{out} in 4Ω is 11A. Curve F is also for a 4Ω load, but now with a maximum reactive component of 90° phase shift. This is a particularly mean load.

With V_{ce} = 60V, V_{out} is actually -16V negative, but I_c is still positive (about 3A). That means that the upper transistor (N device), with a 60V $V_{ce,}$ sources current even with a negative V_{out}! Under these circumstances (which can happen easily with real speaker/crossover loads), quite high V_{ce} to I_c combinations can occur although the device dissipation is very much limited as shown in the SOA cure. For instance, in this example, at V_{ce} = 60V and I_c = 3A, allowed dissipation (per device!) is 90W, although the datasheet calls this a 160W device.

So, you can put in a simple current limiter that follows, say, the 4Ω complex load line. But then you cannot use the large safe area, especially at low V_{ce} and I_c between 10A and 30A, which may come in handy with speaker loads that have low resistive impedance dips. It is clear from **Fig. A** that a simple current limiter is not enough.

You want to "model" the Curve B DC or Curve C 100mS in the circuit with some nonlinear circuit elements, and my target SOA protection curve is labeled A. You can see that at higher V_{ce} I allow more dissipation than the DC curve. The reason is that these high V_{ce} excursions only take place with reactive loads and therefore are by definition not DC loads but short-term. The target curve has three inflexion points at V_{ce} = 30V and V_{ce} = 50V.

The way I implemented this is shown in **Fig. B**, which is the positive half of the full circuit given in **Fig. 2** about the protection circuit. (Again, part values are illustrative only; you can calculate the exact values for your amp later. The negative part works similarly, so I will not discuss it.)

There are four distinct points on the target curve (A). The first and last points are defined by V_{ce} = 0 and I_c = 0, respectively. The other breakpoints are at V_{ce} = 30V and V_{ce} = 50V. These two breakpoints are caused by diodes D3 (for 30V) and D2 (for 50V). Remember: all currents are for two parallel devices! At each breakpoint, the diode(s) start to conduct and shunt current away from R5, so that Q1 conducts later.

To calculate the component values for this rather involved circuit, start with the simple part: if V_{ce} = 0, the maximum allowed I_c is 28A for two output pairs. Also at this point, V_{out} = VCC so there is no current coming from R6/R5. The voltage to turn on Q1 comes exclusively from the voltage drop across the output devices' emitter resistors, through Rs. You need to ensure that at 28A the voltage at Q1b is approximately 0.6V. Although there are two emitter resistors (0.22Ω) and two Rs resistors (100Ω) in the circuit, they are in effect parallel, so from now on use Re = 0.11Ω and Rs = 50Ω. With 28A through the parallel value of Re, V_{re} = 3.1V, which is attenuated through Rs and R4: $12/(50 + 12) \times 3.1 = 0.6$V, which is close enough.

Now it gets interesting. Move to the breakpoint where V_{ce} = 30V. The allowed I_c is about 8A (curve A). Set R14 to 270Ω and then calculate the rest. Disregard D2/R8 and D3/R7 for the moment; you just need to be sure later that that was justified (that the diodes don't conduct at up to V_{ce} = 30V).

V_{ce} = 30V and allowed $I_{c\,max}$ = 8A gives an effective Vs of $Vs = R4/(Rs + R4) \times (Re \times I_c)$, which, with the given values, is 0.17V. You need V_b = 0.6V, so the current from R5 should generate the remaining 0.43V across R14. So this current needs to be I_{R5} = 0.43/270, or 1.6mA, which also flows through R6, of course. With a V_{ce} of 30V (disregard the small V_{ce} loss across Re), that would make R5 + R6 = (30 - 0.6)/1.6 = 18.4kΩ. It doesn't really matter how you split up the total value (it does matter for the other values, of course, but not for the functioning), so make them equal and round them off: R5 = 9kΩ and R6 = 9kΩ.

At this breakpoint (V_{ce} = 30V), diode D3 should be on the verge of conduction, so D3 should be a zener equal to the value of Vm. Because you have 1.6mA through R6, Vm = $30 - (1.6 \times 9) = 15.6$V, making D3 a nominally 15.6V zener.

Now move to the next point, 50V/5A. Analogous to the 30V point, D2 should be on the verge of conducting so you can disregard both it and R8 for now.

As before, calculate the contribution of I_c at Vs: $Vs = 12/(12 + 50) \times (0.11 \times 5)$, which is 106mV, so the V_{ce}-derived current from R5

and R6 should add 0.6 - 0.106 = 0.494V at Vb; therefore, I_{R14} = 0.494/270 = 1.8mA. This 1.8mA also flows through R5 and R6, so Vm = 0.6 + (1.8 × 9) = 16.8V. This, of course, means that D2 should be a nominally 16.8V zener.

Still at the V_{ce} = 50V point, with Vm = 16.8V and D3 a 15.6V zener as calculated, you have 1.2V across R7. The voltage across R6 now is V_{R6} = 50 - 16.8 = 33.2V, so I_{R6} = 33.2/9k = 3.7mA. Because 1.8mA flows through R5, 1.9mA must flow through R7 with 1.2V (Vm - 15.6V from D3) across R7, so R7 = 632Ω.

Moving on now to 100V/2A, it becomes pretty boring: I_c = 2A, so the contribution from Re to Vs is Vs = 12/(12 + 50) × (0.11 × 2), which is only 43mV, and the remaining V_{be} = 0.6 - 0.043 = 0.557V must be generated by the current through R5 and R14, so I_{R14} = 0.557/270 = 2mA. This current also flows through R5, so Vm = (2 × 9) + 0.6 = 18.6V.

The current through R6 is: I_{R6} = (100 - 18.6)/9 = 9mA, so the excess of 9mA - 2mA = 7mA needs to be shunted away through D2/R8 and D3/R7. $V_{R7} = V_m - V_{D3}$ = (18.6 - 15.6) = 3V with R7 at 632Ω, so I_{R7} = 4.7mA, which sets I_{R8} to 7 - 4.7 = 2.3mA. Thus, V_{R8} = 18.6 - 16.8 = 1.8V, so R8 = 783Ω.

Phew! There you have it, but it would be nice if there was some way to verify this before committing solder (and parts). I have disregarded several considerations, such as the loss of V_{Re} for V_{ce}, the base current for Q_1 and some roundings in the calculations. I have developed a spreadsheet to try to double-check all this (available on www.linearaudio.nl and www.audioxpress.com). It works backwards from what I just did: using the component values in the circuit, it calculates at each V_{ce} the Vb contribution from the current through R5, and then finds the I_c that would add just enough voltage across Re to make V_{be} = 0.6V.

Plotting those pairs of V_{ce} and I_c, then, should give the actual SOA curve (A) I wanted in the first place. You see that the spreadsheet shows reasonable correspondence to our values, but there are some deviations. The reason is that once I had the spreadsheet, I "played" with the component values to get the best fit to the SOA curve, or to get "nice" values for parts. I used a Vcc of 44V and a Vce(min) of 2V in the spreadsheet. Also, manipulating the R5 and R6 ratio lets you home in on standard zener values.

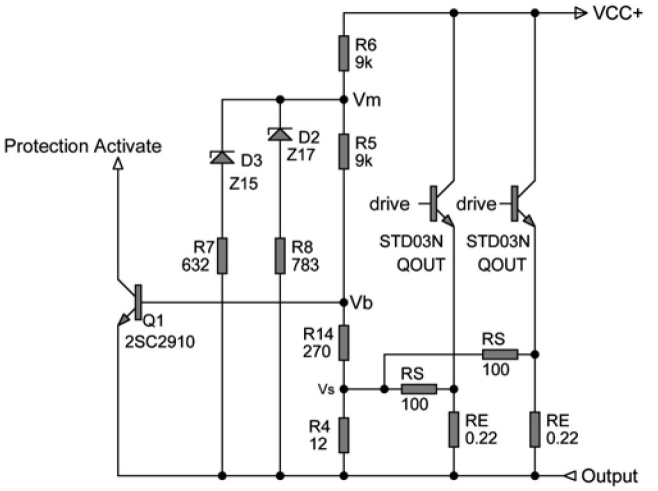

FIGURE B: Positive half of the SOA protection circuit.

The way to do that is to use Excel's solver to find the ratio of R5 to R6 that results in, say, D3 = 15V. Some more playing, then, gets you to a value close to a standard value for D2 as well. In the spreadsheet version on the website, I used the solver to adjust the R5/R6 ratio (cell B12) for a D3 (cell J3) value of 15V and just by luck the D2 value came out to 18V, both standard zener values. The R5/R6 ratio is now 0.51 instead of 0.5. Just try it, you can't break anything, but don't mess with the formulas unless you know exactly what you're doing!

I developed the spreadsheet for this particular application and did a fair amount of checking, but I cannot guarantee that it is without errors. Remember that this spreadsheet gives you an approximation of the real hardware only, but since I stayed on the safe side with the protection curve (A), small deviations won't cause disaster. If you find any errors, please let me know and I will update it, giving credit where it is due, of course.

It is not easy to test the accuracy of such a protection system in the real world, and mistakes could be expensive. So this spreadsheet that double-checks the calculation of the values and then works backwards to verify them is important gaining confidence with this circuit. ■

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Burson Audio Modules

By Ed Simon

For this review I received two clear light anodized aluminum (6000 alloy family, 6061 or 6105, I think) extruded tubing cutoffs 1.5″ in diameter by 6.30″ long with both ends plugged by a closed cell foam disc and covered in shrink wrap. Both were labeled "Burson Audio Melbourne, Burson sounds good like solid state should." One also claimed "Burson Audio Opamps, Dual, matched pair," while the other differed by claiming to be a "Single, matched pair."

FEATURES

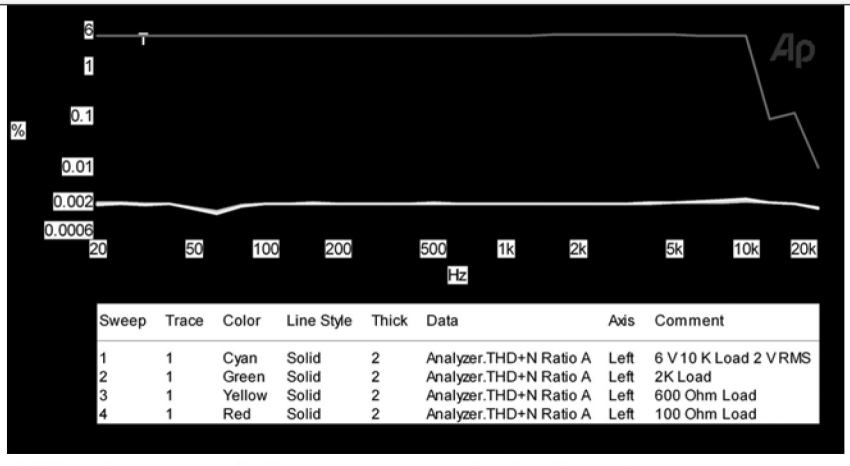
There also was a piece of hollow flexible tubing, spaghetti as it used to be called. For some reason, I expected Teflon, but this seemed to be silicone over fiberglass.

Opening the "Dual, matched pair" tube first, I was surprised to find no moisture-absorbing package—just two modules composed of two almost mirrored PC cards. Each card had 16 TO92 cased semiconductors, 15 fixed resistors, a single silver mica capacitor, and a Bourns multiturn cermet trimpot. All parts were of "audio" types.

Some of the transistor markings could be read—2SA970, 2SC2240, 2SJ74, and 2SK170. The same gain code (blue) was marked on all devices. Three jumpers connected the two cards. Four 20 gauge clad copper leads from each card came out to match an 8-pin DIP mounting.

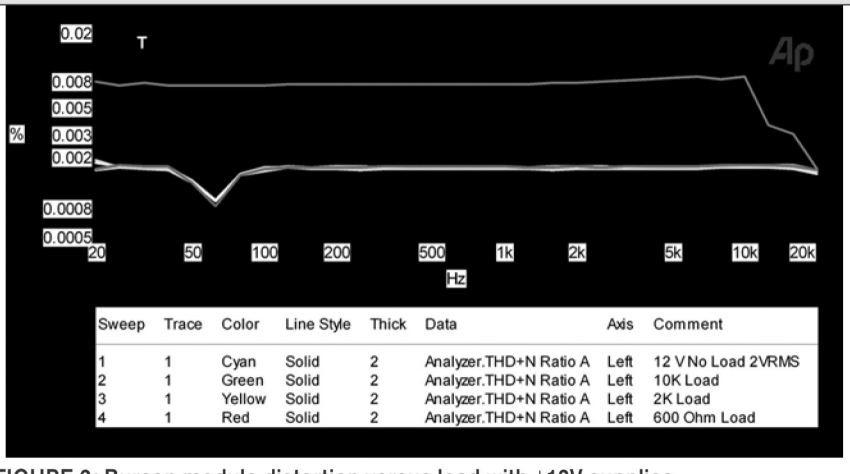
There are some people who believe that FETs (2SJ and 2SK) are the superior audio semiconductor for many reasons, including less RFI susceptibility. Overall, the build quality was good except for the three inter-card jumpers. They had been trimmed so that the cutting action also trimmed the solder joint. According to IPC standards, you should cut the lead and not the fillet. When the fillet is damaged it should be reflowed.

The single-page directions packed in the tube contained only the briefest of mechanical and electrical specifications. The dimensions for the dual op amp were given as 48mm × 22mm × 24mm. The modules measured 39mm × 23mm × 29mm, excluding the 37mm long leads. "Weight" was listed as 28 grams including leads. The mass was really 18.77 grams.



Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment
1	1	Cyan	Solid	2	Analyzer.THD+N Ratio A	Left	6 V 10 K Load 2 VRMS
2	1	Green	Solid	2	Analyzer.THD+N Ratio A	Left	2K Load
3	1	Yellow	Solid	2	Analyzer.THD+N Ratio A	Left	600 Ohm Load
4	1	Red	Solid	2	Analyzer.THD+N Ratio A	Left	100 Ohm Load

FIGURE 1: Burson module distortion versus load with ±6V supplies.



Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment
1	1	Cyan	Solid	2	Analyzer.THD+N Ratio A	Left	12 V No Load 2VRMS
2	1	Green	Solid	2	Analyzer.THD+N Ratio A	Left	10K Load
3	1	Yellow	Solid	2	Analyzer.THD+N Ratio A	Left	2K Load
4	1	Red	Solid	2	Analyzer.THD+N Ratio A	Left	600 Ohm Load

FIGURE 2: Burson module distortion versus load with ±12V supplies.

CD UPGRADE

Rather than see whether the electrical specs were as inaccurate, I decided to try out one of the dual op amps. A CD player seemed to be called for because they are often modified and there is internal noise (RFI) to worry about. I had two Sony SCD-CE595 Super Audio CD players left over from another project. One still sat on a shelf, the other had gravitated to the office sound system, and, when compared to the other gear, it screamed for an upgrade. A quick Google search showed these had been modified by others. It seemed that IC400 on the bottom of the one PC card in the CD player was all I had to change. It was a surface-mount small outline package.

Four sheet metal screws and three machine screws later, I had the cover off the player. Five more screws, and the PC board was loose. On the bottom, as advertised, was IC400. The website had mentioned some folks had difficulty removing this chip and lifted pads. The box

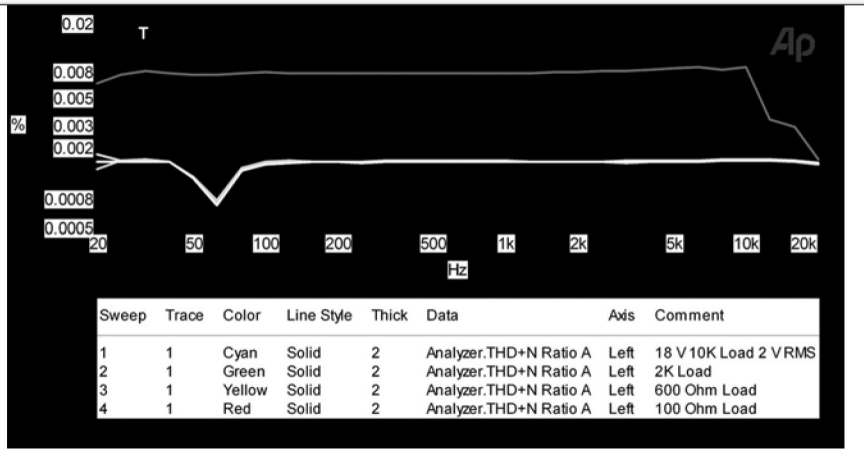
did not have the CD mark on it, but I still assumed it would have lead-free solder.

Using braid wick and a 1.5mm tip, I made the solder under the chip disappear. The chip did not come loose. I noticed several very small hardened glue blobs on the edge of the PC card. The IC had been cemented into place before soldering. Cutting all four leads on one side allowed me to pry up the chip with a small knife tip. Then using solder wick, I picked up the first four loose leads, then the other four with the chip all at once. I now had a nice clean set of pads.

To confirm that the chip was a dual op amp, I tried to Google the part number marked on the chip—no luck. I then checked for continuity between pin eight of my chip and the same pin on two other similar chips. All were connected. Then I tried pin 7, no continuity. A single op amp in a package gets the V+ on pin seven, a dual gets in on pin eight, and this unit could not be a quad op amp because there were only eight leads.

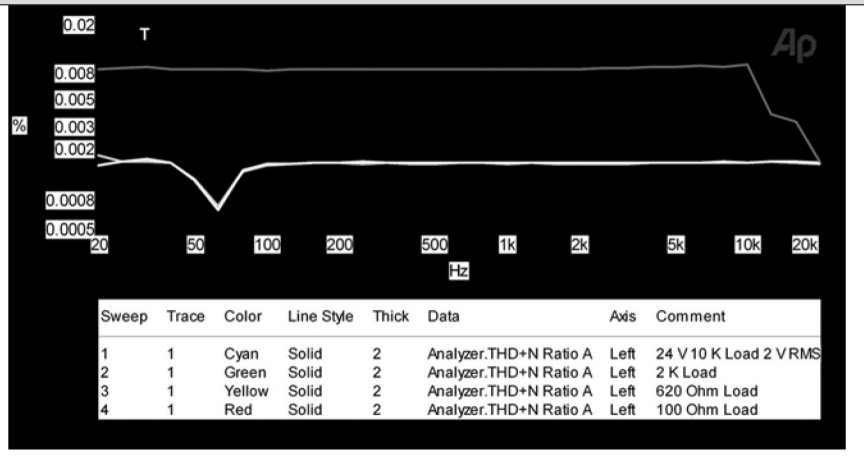
The next problem was that the Burson dual op amp would not fit under the PC card; it was too big. The matched singles looked better, but they still were too thick and had too many leads. The singles also had two PC cards, with the op amp on one and the other a lead holder and ground plane. So it was time to change the second PC card to a piece of 0.002″ brass foil covered by insulation, and change the PC board jumpers to leads.

Now I had a single inline version of the op amp. Wiring the two units side by side underneath the CD players' PC card almost worked. The CD's PC card fit back in, but I had to tilt it slightly. Worse, the 20 gauge wire leads tore



Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment
1	1	Cyan	Solid	2	Analyzer.THD+N Ratio A	Left	18 V 10K Load 2 V RMS
2	1	Green	Solid	2	Analyzer.THD+N Ratio A	Left	2K Load
3	1	Yellow	Solid	2	Analyzer.THD+N Ratio A	Left	600 Ohm Load
4	1	Red	Solid	2	Analyzer.THD+N Ratio A	Left	100 Ohm Load

FIGURE 3: Burson module distortion versus load with ±18V supplies.



Sweep	Trace	Color	Line Style	Thick	Data	Axis	Comment
1	1	Cyan	Solid	2	Analyzer.THD+N Ratio A	Left	24 V 10 K Load 2 V RMS
2	1	Green	Solid	2	Analyzer.THD+N Ratio A	Left	2 K Load
3	1	Yellow	Solid	2	Analyzer.THD+N Ratio A	Left	620 Ohm Load
4	1	Red	Solid	2	Analyzer.THD+N Ratio A	Left	100 Ohm Load

FIGURE 4: Burson module distortion versus load with ±24V supplies.

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off the small pads.

Fortunately, under where the IC400 had been were two vias from the other side that carried inputs. The other two inputs had vias just outside where the IC had been. The outputs had vias to the coupling capacitors. The V- supply had a large pad and, with a bit of poking, I found a via for the V+.

I took a piece of stranded wire, split the stranding, and used each individual conductor to fit into a via hold. I then connected these short jumpers to the Burson SIP modules. A small spacer added to the back made sure the modules would not short.

COMPARISON

The datasheet suggests doubling the value of the "nearby power decoupling capacitors" and replacing them with better units such as Elna or Blackgate. I wanted to know what the Burson did by itself, so I left them alone.

I then powered up the modified unit and the stock one. I burned matched CDs for each. The output of one channel of each went through a stepped attenuator (gold-plated Electroswitch rotary and Holco resistors) to my workbench amplifier (*aX* 4/2006) into an old Spica TC1 loudspeaker.

At first I just listened to be sure both worked. Then I left them running because the workbench amplifier actually needs to warm up a bit. Burson recommends a 100-hour burn-in for their modules!

I went back to the two CD players and took a quick listen and measurement. Another of the Burson meager specifications failed. The Sony CD player had the V- terminal connected to ground and the V+ was only 12V or so. This is half the voltage the Burson op amp is supposed to require.

So to do a fair comparison, I would now need to add a power supply to the CD player. Before I did that, I decided to do the deferred electrical tests to see at which supply voltages and loads the module worked best, because the datasheet gave a range which was already shown to be inaccurate.

MEASUREMENTS

The first thing I measured was the current consumption: 19mA per amp on each of the duals, not the 25mA specified. A spec that at least was close! So I measured THD at unity gain for loads of 10,000, 2000, 620, and 100Ω.

Notice that even at power supplies of ±6V the op amp is capable of working into loads as low as 600Ω at 2V signal, the maximum expected from a CD player. Turns out I did not need to change the power supply.

The distortion drops a bit at 12V and above. Even more interesting is the drop in distortion into the 100Ω load. This discrete op amp can drive loads as low as 600Ω easily and is usable even lower, but the distortion does rise.

Figures 5 and **6** present another difference from the instruction sheet. The distortion is really 0.0007%. Burson just claims less than 0.001%. Not only is the distortion low, but the higher-order products—the ones that sound bad—are almost immeasurable. This test is done at unity gain, which would cause the nasty harmonics to be larger. Using the op amp with a gain of 20 or so would be even better.

The Burson discrete op amps cost $90 for a dual, $180 for a matched pair of duals, and $115 for a matched pair of singles. If you think this is expensive, compare it to the better-known John Hardy 990 op amp. On the other hand, if you prefer to use an IC that can at best come close but not match the Burson, remember that most of them are not "purpose built" for audio.

One final short comment is that Burson expects you to use this discrete op amp as an upgrade to existing prod-

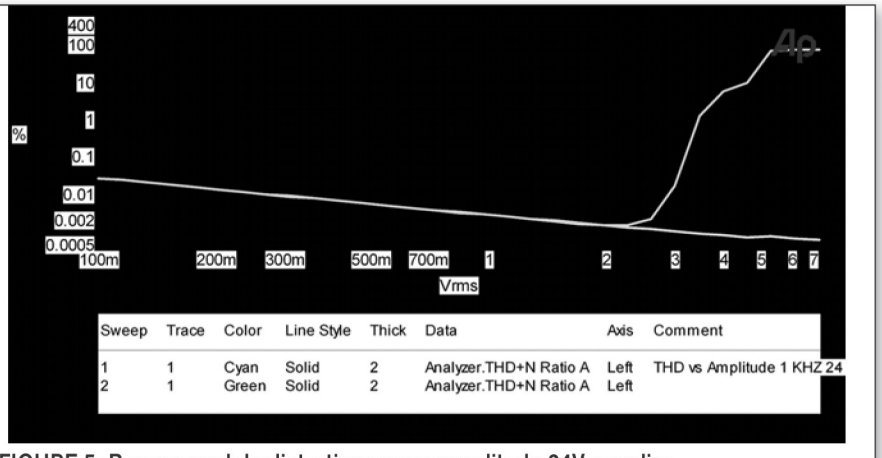

FIGURE 5: Burson module distortion versus amplitude 24V supplies.

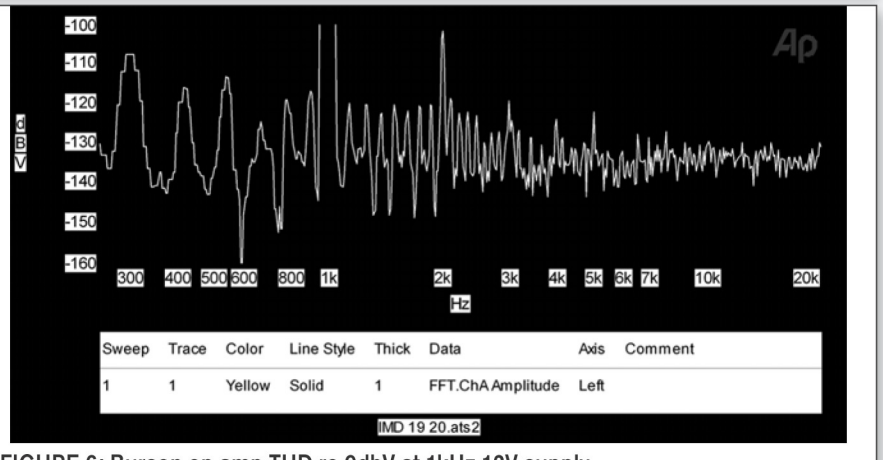

FIGURE 6: Burson op amp THD re 0dbV at 1kHz 12V supply.

ucts. Of course, you can use this in new projects such as a power amplifier and eliminate more than half of the circuitry with something that is significantly better. This is truly a great DIY piece of gear. But then they probably want to sell you their version of amplifiers, preamps, and so on.

But if you are a really maniacal DIYer, you could buy a few hundred of each transistor type and match them, lay out a few different versions of the PC card to see how things interact, and even manage to select really good-sounding resistors instead, but it is your choice. You may find that to buy all the parts actually costs about the same as the complete module.

In conclusion Burson has produced a poor datasheet that does not live up to their product. A comparison of the stock amp to the Burson Pro version is smoother, sweeter, reaches higher without straining, and has depth that others only dream of.

So since I knew what the Burson discrete op amp is capable of doing on its own, I added a few more tweaks to the CD player, 1µF poly bypasses at the modules, and similar giant film caps instead of the miniature electrolytics, and Deoxit on all the connectors. It does not work miracles, but is a noticeable improvement.

The next step will be to build an amplifier using the Burson discrete op amp as a front end. I may even upgrade my preamp with a pair of modules. In pro use these should be a great microphone preamplifier.

Manufacturer's response:

We would like to thank the good people at *audioXpress* for this in-depth evaluation of our Burson Audio Opamp.

The incorrect dimension measurements relate to an older generation of Burson Opamp. Since your review, they have been corrected accordingly. Thank you for also pointing out that our new op amp is 33% lighter than its predecessor.

The various working voltage and power consumption measurements listed on our instructions were recommended to ensure the most satisfactory outcome for our customers. They are not the minimal working requirements for the Burson opamp.

For five years we have worked tirelessly to alert fellow audio designers about the shortcomings of integrated circuitries (ICs) in any audio design, and to inform fellow audiophiles of the supremacy of the Burson opamp over any IC op amps. It is therefore most satisfying to finally receive formal recognition for our hard work by one of the most authoritative audio magazines in the English speaking world.

John Delmo
Burson Audio
aX

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PrimaLuna ProLogue Six Tube Monoblock Power Amplifiers

By Tom Lyle

	Upscale Audio
	2504 Spring Terrace
	Upland, CA 91784
	(909) 931-9686
	Fax: (909) 985-6968
	www.primaluna-usa.com
	Price: $2,599 a pair

There are many options when it comes to powering two-channel systems when using separates. However, the preponderance of options involves solid-state amplifiers, and if you choose a tube amp and are at all concerned about how much green you need to lay out, you aren't going to be able to procure that much wattage. Thanks to the Dutch company PrimaLuna, their Chinese-built ProLogue and DiaLogue series amplifiers make acquiring a relatively high-powered tube amp a reality. Not only do these amps have more than enough wattage to drive most speakers,

PHOTO 1: PrimaLuna's ProLogue tube amp.

FIRST IMPRESSIONS AND BASICS

Each amp arrived at my doorstep triple-boxed and packaged with the tubes installed. The large main power tubes (four EL-34s in each) were wrapped in soft Styrofoam, and white cloth gloves were supplied to check the seating of the tubes and handle the glossy finished, nearly 38 lb units. My samples had silver faceplates,

> "Soldering work and layout [were] designed as though the owner of these amps was going to use them upside down with the covers removed and their innards displayed for all to see."

but also they are of extremely high build quality, are easy to maintain, and have exceptional sound quality (**Photo 1**).

Most of PrimaLuna's integrated and power amps have been more than positively reviewed in the mainstream audio magazines. I wondered whether a pair of their 70W ProLogue Six monoblocks could drive a pair of power hungry, low impedance hybrid electrostats. Of course, I was willing to start out using them in a smaller system using stand-mounted two-way speakers. It would be a better representation of how a good number of audiophiles will use these amps.

which are also available in black. You can easily remove the tube cage, which is held in place with pegs that fit the cage snugly into place corresponding to the exemplary external fit and finish of the amps. The internal fit and finish was also exemplary, with soldering work and layout designed as though the owner of these amps was going to use them upside down with the covers removed and their innards displayed for all to see (**Photo 2**).

On the rear of each amp are a gold-plated RCA single-ended input, four heavy-duty speaker terminals, one common, and terminals for 2, 4, and 8Ω loads. I tried both the 2 and 4Ω outputs when powering the electrostatics, and really didn't hear any difference between the two, but ended up using the 2Ω tap recommended in the manual. I used the 4Ω taps for the small speakers. There is an IEC AC power socket for the detachable power cord and a place for the 5A fuse. The large power switch is on the left side of the amp near the front.

In addition to the quartet of EL-34 output tubes, each amp has a pair of trusty small 12AX7s and 12AU7s in its input section (PrimaLuna buys their tubes from a Chinese supplier, but rejects about 40% of these because they don't meet their standards). PrimaLuna claims that tube life is extended because of their soft-start circuit, which powers up the tubes slowly when the power switch is turned on. The manual recommends turning the amp off when not in use, with no need to run up your electric bill to keep the amps warm when you are not listening to them.

PrimaLuna also claims that these amps are of the "plug and play" variety—an "adaptive autobias" circuit monitors and adjusts the tube's bias automatically. This makes it possible to use unmatched sets of tubes, possibly saving you some money when eventually you need to replace the tubes. This circuit also makes it possible to experiment with different types of tubes. On the

distributor's website, they claim you can use 6550, KT88, KT90, 6L6GC, KT66, 7581, or even 6V6GT tubes instead of the EL-34s. [Manufacturer's note: Though the PrimaLuna is fine with 6V6GT tubes, we are no longer recommending those because there are some low quality ones out there, and instead, we recommend EL-37—Kevin Deal, PrimaLuna USA].

The custom output transformers were designed by an American engineer, then the final design was done by Marcel Croese, who supposedly is a very renowned transformer designer. A good transformer design should lower the noise floor and extend the frequency response.

PrimaLuna says that the ProLogue Sixes have a "dual-feedback topology" which is a cross-coupled current feedback for superior impulse behavior and gain stability. A slight amount of negative feedback is used to achieve exact gain setting, low distortion, and low output impedance. PrimaLuna claims that this effectively cancels the negative side effects associated with both feedback circuits. The end result utilizes all the benefits of feedback without the drawbacks.

The ProLogue Six is made with many quality parts. For example, no plastic tube sockets here, only ceramic sockets are mounted on the chassis. Internally, PrimaLuna uses Nichicon and Realcap capacitors, a toroidal transformer, and the amp is handmade with non-cost-effective point-to-point wiring throughout.

LISTENING—SYSTEM ONE

The smaller of the two systems consists of a digital front end (Arcam) and an analog front end of an old FM tuner (AR), either a solid-state (Jeff Rowland Designs) or tube (Balance Audio Technologies) preamp. The speakers I used were Dynaudio Focus 110.

If you are interested, the preamp-to-amp interconnects are Cardas, the CD to preamp is Monster Cable, and the tuner is connected with out-of-the-box-type cable. The speaker cable is sometimes Kimber, sometimes Monster. I use the stock power cords and no power conditioner. The PSB speaker stands are filled with marble gravel, and all the other equipment sits on the hardwood floor atop and separated by antique Audio-Technica pucks.

While the amps were in either system, I heard no complaints—there were no buzzes, hiccups, static, or any extraneous noises at any time. I could hear no background noise other than a slight hiss when I turned up the volume on the preamp and put my ears up to the speakers. Other components could have contributed to this background noise.

After they were just about fully broken in, which took about two weeks of on and off use, the PrimaLuna ProLogue Sixes sounded magnificent in this system—even when listening to the FM tuner, which was usually set to the local jazz station WBGO in Newark, N.J. (terrestrial out-and-out jazz stations are very scarce these days). On one occasion there was a solo tenor sax playing (I think it was a recent Sonny Rollins release). Its sound entered the room with such a pure tone it was disquieting.

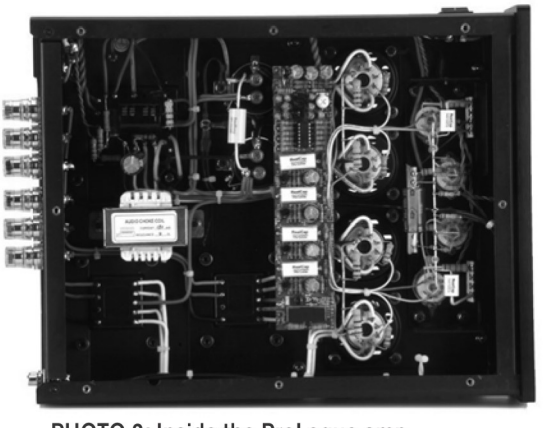

PHOTO 2: Inside the ProLogue amp.

When using the CD player, of course, things became better. One high point was Wilco's *Yankee Hotel Foxtrot*. When playing this CD the system sounded *alive*: the treble had a natural timbre, and both the mids and highs had a rightness that made this disc sound as good as it ever has—with a tube sweetness that was addictive. This tube sweetness was not a euphonic coloration, but it included as one of its features a less forward sound that diminished listening fatigue to a point that made for some very long listening sessions.

When I played orchestral discs such as Sibelius' symphonies with Colin Davis conducting the LSO on RCA, the string sound was luscious in large part because midrange was so accurately portrayed. There was no loss in resolution anywhere in any frequency, and there was an abundance of what I like to call "dynamic distance" between sounds—that is, the separation of two or more instruments playing simultaneously but at different volumes.

I could kick out the jams, too, playing "Wasted Years '99" from Iron Maiden's *Best of the B-Sides*, which was included in the Iron Maiden box set (packaged in a metal casket, by the way). There was plenty of power displayed in the crunching guitars and thunderous rhythm section. The non-pumped-up midbass shook the air, which says a lot for the superb speakers, at least from 40Hz on up.

When I came back to earth, I played the Kronos Quartet's album *Pieces of Africa*, and reveled in the sound of the individual strings and percussion instruments clearly portrayed in the space of the proportionally scaled soundstage. I'm not going to say that the instruments sounded indistinguishable from the real thing—that's ridiculous—but it sure was impressive how lifelike this CD sounded in this relatively modest system. The PrimaLunas and the Dynaudios were a perfect match.

LISTENING—SYSTEM TWO

This system is in my main listening room and consists of analog (Basis/Tri-planar/Lyra turntable setup and an NAD FM tuner) and digital (Arcam and Oppo) front ends, a tube preamp (Balance Audio Technologies), and Sound Lab Dynastat hybrid electrostatic speakers, sometimes augmented by a Velodyne sub. If you care, the interconnects are either MIT or Virtual Dynamics, the speaker cable MIT, power cables Virtual Dynamics and MIT, and all the equipment is plugged into either PS Audio Power Plants or a Chang Lightspeed conditioner. Then these are plugged into Virtual Dynamics wall receptacles with two

dedicated 15A lines. Most of the equipment rests on an Arcici Suspense equipment rack, and the room is treated with Echo Busters acoustical panels.

When I first had the amps in this system, I was careful. I didn't want to judge the amps too harshly with complex material, so along with some other small combo recordings, I played an LP of Julliard String Quartet's version of Bartok's *String Quartet No. 6*. The amps sounded good, but they had a more burnished, laid-back sound than I was used to. But they were reasonably neutral sounding, a little rolled off in the highs (which as a benefit reduced the background noise on the old record), and a little less defined in the midrange and mid-bass than absolute, but the overall sound was very good.

As far as the decrease in treble energy, nobody is going to describe the Sound Lab speakers I use as bright sounding; some might even say they are a little rolled off. I disagree. I find them to be extremely transparent in the treble region, but I've used them as my reference for years, so I'm probably used to them.

Nevertheless, when I played the SACD layer of Stokowski's *Rhapsodies* album, a Living Stereo re-issued by BMG, the sound of the solo instruments on an excerpt from Wagner's *Tristan and Isolde* sounded rather lifelike—especially the English horn—and I could clearly sense the ambience of NYC's Manhattan Center. The amps were able to sort out more complex fare—even when I threw on some convoluted rock (or whatever you want to call it) such as Frank Zappa's *Weasels Ripped My Flesh*. The amps kept their composure (even when the musicians did not) while playing this thorny music.

But, even with all their positive traits, I wasn't totally blown away. They were just too rolled off in the highs for my taste, and the midrange was a tad too recessed. I could only chalk that up to a combination of the speaker's relatively soft highs (although, again, I'm prepared to debate that fact), perhaps their demand for high power, and their challenging impedance characteristics.

TUBE SWITCH

When speaking to the US distributor of the PrimaLuna, Upscale Audio's Kevin Deal, I think he sensed my lack of enthusiasm for the combination of the ProLogue Sixes and the Sound Labs. He said he'd send me a set of PrimaLuna KT88 power tubes to try out instead of the stock EL-34s. They arrived about five days later, and I installed them forthwith.

The improvement in the sound of the amps was extraordinary. The midrange reached a more natural level and was much more defined, and the highs were restored to a point where their extension nearly rivaled a solid-state amp, yet instruments within both frequencies were rendered with what I can only describe as tube sweetness. It was a very lifelike sound, which included excellent separa-

tion of individual instruments and groups of instruments. There was ample power in the bass, and the overall sound of the amps was extremely inviting.

I played a new favorite of mine—Nine Inch Nails' *Year Zero* CD. The album came across as aggressive and relentless, which was as it should be. Turned up nice and loud, the bass shook the walls. Synths and the hammering percussion sounds were upfront and distinct, yet incorporated into the dense mix to form the intense wall of sound. Wonderful stuff.

The sometimes raucous CD of Prokofiev's complete soundtrack of *Alexander Nevsky* with Uri Temirkanov conducting the St. Petersburg PO was superb. Like the NIN, this is a dense recording (*Romeo and Juliet*, it ain't), yet the ProLogue Sixes had no problem sorting out the orchestra, chorus, and soloists. But in contrast to the NIN the participants and the space were real, and it was like I had a tenth row seat during the recording session. The highs were sweet and credible and the mids were well sorted out in the very intricate, large soundstage.

I loaded the Zappa disc again, and was reminded that the recording was a bit bright. But despite this, it was almost as though I was listening to a live mike feed in the studio rather than a CD. The drums, in particular, were outstanding, in part because the transient response was surprisingly good; there was a snap to the sound that I wasn't expecting from a tube amp.

I also spun the best pressing I have of Jimi Hendrix's *Axis Bold as Love* LP, a mono pressing reissued by Classic Records (vinyl, tubes, mono—please help me). This version, of course, disposes of the psychedelic hard left-right panning and presents the album in a much more listenable arrangement. I was able to follow and dissect all the overdubbed guitar and effects. But especially good was "Little Wing," which wasn't laden with overdubs and let me hear all that engineer Eddie Kramer dialed into the mix, particularly the goose-bump-raising flanged guitar. I admit that the recording quality isn't the greatest, and the PrimaLunas did not fail to disclose that fact.

Because I was stuck in the 1960s for much of my listening of late, I played "India"—the first cut on Coltrane's *Impressions* album on a Japanese Impulse pressing, recorded

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C'mon...You know you want to!

at the Village Vanguard in 1961. The track takes up most of side one (John Coltrane and Eric Dolphy together—it doesn't get much better than this).

Even though the highs were a smidge softer than I'm used to, they were very revealing, and there was lots of air around the horns. The low end and mid-bass were also fine; the dual basses of Reggie Workman and Jimmy Garrison weren't too meshed together, so it was possible to identify each instrument. The recording quality is very good; I just wish McCoy Tyner's piano wasn't mixed so low. The vinyl is silent, with the only background noise being the tape hiss which the PrimaLunas made clearly evident. (Depending on how you look at it this was or was not a good thing.)

As I continued to live with the amps, I realized there wasn't any type of music that could trip them up, and the more I used them, the more I appreciated their sound. CD after CD, record after record, the midrange had an impressive ability to separate out all the instruments in any mix, and both the mids and the highs exhibited the tube magic that I've come to expect from a great valve amp. This "tube magic" wasn't the old-school-type of tube sound—rolled off in the bass and highs, and their only strength the midrange—it was more that instruments had that dynamic distance I mentioned when using them in the smaller system, combined with less in-your-face forwardness than the majority of affordable solid-state amps.

I was skeptical; I've had other tube amps paired with these speakers before, all with less power than the PrimaLunas and with mixed results. But still, even with 70W per channel, I didn't think the Sixes would have enough power to drive a pair of these rather large electrostatic/dynamic speakers. I was wrong.

I needed to turn up the volume on the preamp higher than I did with a 250W per channel solid-state power amp, but this did not in any way increase the background noise. I had no trouble reaching the volume limits of the speakers, which was quite loud—loud enough for me to back off a little to prevent possible hearing damage. What I can't explain is why I didn't *need* to turn up the volume so loud. I could listen to music at lower dBs with the Sixes than I was accustomed to, and still consider it to be an acceptable level.

CONCLUSION

I think many audiophiles will like the fact that simply swapping tubes can have such a dramatic effect on the sound; I could only imagine what a field day a tube roller would have swapping different brands and vintages. Others will not want to mess with such stuff, and I guess they'll just have to stick with the stock tubes or use a solid-state amp. I can appreciate both worlds. But I must admit that after replacing the KT88s with the stock EL-34 tubes and bringing the amps downstairs again to use with smaller speakers, it was a better match. They spent a lot of time in that system, and I could live with these amps forever.

Ultimately, I preferred the control and power of the solid-state muscle amp in the big system; I think that's a matter of taste more than which one was "better." There is a chance that you might prefer the PrimaLunas with your large electrostatic speakers, even with the warmer-sounding stock EL-34s installed. I preferred the tubed PrimaLunas by a wide margin in the smaller system.

But wait—if you have the money, the next step up is the PrimaLuna ProLogue *Seven* mono amps that come standard with KT88s and upgraded internal parts, also at 70W per channel, for $3149. Loading up a pair of ProLogue Sixes with PrimaLuna KT88 tubes will cost you about $280, so the price comes within shouting distance of the ProLogue Sevens. I'm not totally sure, but it seems that the Sevens might be the more sensible choice when used with large speakers.

Still, the majority of customers will use them with more efficient speakers, and to them I give the PrimaLuna ProLogue Sixes my highest recommendation. From my evaluations, using them with large speakers is not out of the question, and quite sensible if you have a hankering for tubes and don't want to take out a second mortgage to do so. I'm not going to pretend that $2600 is pocket change, but the PrimaLuna ProLogue Six monoblocks are worth every penny.

Manufacturer's response:

All of us associated with PrimaLuna wish to graciously thank Tom Lyle and *audioXpress* for the thorough and thoughtful review of the ProLogue Six Monoblock Power Amplifiers. When Tom writes that the amps "sounded magnificent," that ". . . there wasn't any type of music that could trip them up," and that his experience was ". . . like I had a tenth row seat during the recording session," we all can only smile because in a nutshell, he articulates exactly what we're all about.

We're especially pleased that Tom spent the time and effort to evaluate the amps in two systems. It comes as no surprise to us that the ProLogue Sixes performed magnificently in both.

We'd like to highlight one of those evaluations—his experience with the Dynastat electrostatic speakers—because it illustrates a wonderful point. Tom states that initially, he was not particularly thrilled with the EL34-based monoblocks pushing the Dynastats—the highs were "a little too rolled off" and the mid "a tad too recessed." He speculated that perhaps it was the speaker's demand for high power or maybe their challenging impedance characteristics. We suggested switching tubes and shipped him a set of stock PrimaLuna KT88s. And night turned to day. Tom writes that the improvement in sound "was extraordinary" and then spent the next 10 paragraphs raving about the amps' characteristics.

The point to be taken from this is that you never know what kind of speakers you may wind up with. All PrimaLuna products can be tweaked—nudged, if you will, in a particular direction and that gentle nudge can be exactly what turns a mundane experience into an extraordinary one.

We sincerely appreciate Tom's willingness to recommend these monoblocks for use with power-hungry speakers like the Dynastats; I have yet to find a speaker the ProLogue Sixes can't drive. I've paired it beautifully with my Sonus faber Guarneris and have heard raves from people who use it to drive difficult loads such as Martin Logan Vantages (impedance—4Ω, 1 at 20kHz).

Thanks again for the glowing evaluation. When a reviewer is moved from non-plussed to ecstatic by simply switching tubes, we know we've created something special.

Kevin Deal
PrimaLuna, USA
rr

Pioneer X-Z9 Compact Stereo System, Pt. 2

By David A. Rich

MEASUREMENTS

The midrange sounded smooth, but lacked detail. Annoying brightness at the top-end could be moderated with the treble control. The coax did its job in stand-up tests, where the upper midrange was unchanged. Frankly, I was expecting more, so I brought out the test equipment.

Figure 1 is the on-axis response of the speaker, the mike 1m from the speaker using the Acoustisoft ETF quasi-anechoic measurement system. I applied one-sixth octave smoothing to the graph. I achieved the best results with the microphone 8″ below the tweeter, slightly below the woofer center. This graph also plots the horizontal radiation pattern of the speaker.

The overall response is smooth except for the giant, puzzling dip starting at 2kHz and declining by 4dB. Baseline returns at 6kHz. The dip hardly changes even out to 60° off-axis. There is some reduction in amplitude at wider angles above 4kHz, but the amplitude reduction at 3kHz is nearly identical at 12kHz. Clearly, the handoff from the midrange to the tweeter is fine.

The tweeter response is remarkably smooth given the speaker's coax design and the fact that the graph is smoothed to only one-sixth octave. There are no significant ripples typical of a coax as the tweeter interacts with the midrange cone. The ripples are larger relative to the KEF Reference 205, but the bookshelf speaker in the KEF Reference line is $6000. No bookshelf or free-standing KEF speaker exhibits the upper midrange dip, but some of the lower priced units show more ripple at the high-end from the coax design.

Take a look at the vertical radiation patterns. **Figure 2** is the response starting at the woofer axis and moving down to 25° off-axis. There are no crossover effects! A slight reduction at the high-end is an expected off-axis effect at the tweeter. This is an excellent result in the absence of the aforementioned dip. Again, this graph is shown with one-sixth octave smoothing applied to the graph.

A rise (**Fig. 3**) in the vertical direction highlights woofer/midrange interaction at the 800Hz crossover, typical of a well-executed 12dB octave crossover for non-coincident drivers. At the midrange–tweeter crossover, there is no change out to 30° off-axis. The dip cannot be a crossover artifact because it persists as the vertical and horizontal angles change. Indeed, it seems to be some sort of electronic filter in the crossover path.

My in-room speaker measurement (**Fig. 4**) is an average of nine measurements taken around the listening point using the Acoustisoft RplusD software. The one-second gate time is equivalent to an RTA measurement but with 100 band because I used tenth octave smoothing. Ignoring the room modes and the speaker room interactions (Allison effects) below 150Hz, the speaker is admirably flat with the same dip unchanged in magnitude or bandwidth. Remember, this graph is smoothed to a tenth of an octave, not the traditional third octave smoothing.

The high-end is a little rolled off—as expected—from an in-room measurement that is off-axis to the speaker.

I also made an average of 20 measurements with the microphone deployed at a variety of positions throughout the room (not shown) to gauge the speaker's power response. The curve was similar to **Fig. 4**, with a similar smooth response except for the dip unchanged in magnitude or bandwidth. Measurements with the speaker in other placements in the room or in other rooms did not show any significant change in the dip.

DIP EFFECTS

Figure 4 confirms the hypothesis that the dip is not driver- or crossover-related. It looks as though a filter was deliberately placed in-line with the speaker because the dip is invariant to the type of acoustic measurement. The dip looks to be an intentional voicing, often called a BBC or Gundry dip. The BBC observed the dip's ability to produce an artificial sense of depth and spaciousness.

Pioneer may have applied the dip to the S-Z9 expecting that it would either be pushed against a wall and the pair spaced close together (4′ or less), which is the appli-

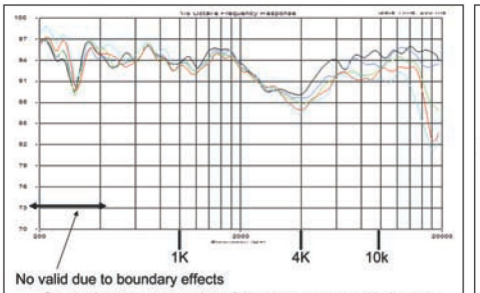

FIGURE 1: Horizontal radiation pattern with the vertical axis centered at slightly below the woofer's center.

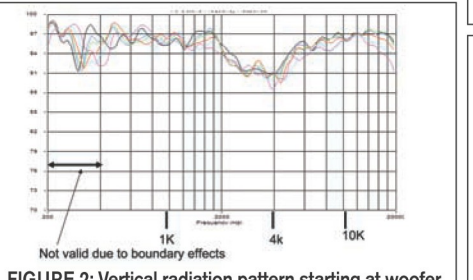

FIGURE 2: Vertical radiation pattern starting at woofer axis and going down by 5° to 25° below the speaker.

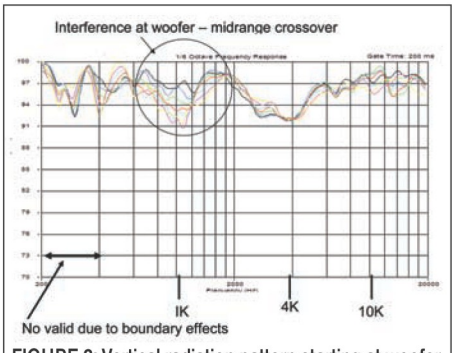

FIGURE 3: Vertical radiation pattern starting at woofer axis and going up by 5° to 30° above the speaker.

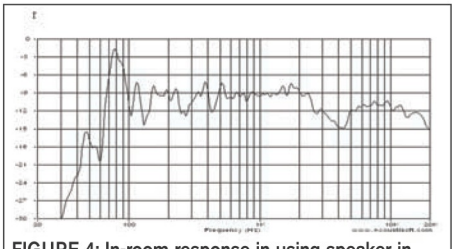

FIGURE 4: In-room response in using speaker in position 1.

cation demonstrated on the Japanese website. Larger room sizes in the States should provide more latitude for speaker positioning. The instruction manual says absolutely nothing about speaker placement.

The dip in this speaker is not a signature Pioneer effect; it is absent in the top-of-the-line Pioneer S-2EX bookshelf speaker and the larger members of the family, which have been independently measured by *Stereophile* and *Ultimate AV*. The Japanese website pays much attention to the X-Z9's "AIR Studios™ Sound Tuning." The British Air Studio is well known for producing highly marketable albums. Could the "Sound Tuning" introduce the dip? I do not know, and Pioneer America could not shed any light on the situation.

I find the dip in this speaker to be a defect that should be fixed. This is readily accomplished by modifying the components in the crossover that introduce the dip. Ignoring the dip (**Figs. 1–4**), the speaker is exceptionally flat on-axis, off-axis, and in the room.

To simulate how the speaker sounds without the dip, I applied the pricey ($4000) Lyngdorf RP-1 room equalizer to neutralize it. The equalization curve did little above 300Hz aside from filling the dip. Not surprising, because the speaker is otherwise flat in anechoic measurement and in a room, so it does not need to be equalized outside the vicinity of the dip in the speaker-dominated frequency spectrum.

The change was remarkable. The high-end brightness disappeared with the tone control set to flat. I was clearly using the 3-4kHz minimum dip level as a reference and preserved the increase in level above 4kHz (as the dip recedes) as an out-of-balance high-end. John Atkinson of *Stereophile* reports other reactions are possible for a speaker with an upper midrange suckout.

While the S-Z9 sounds pleasant enough in the upper midrange with the dip and the treble control engaged, the speaker blossoms with rich harmonic information restored in the absence of the dip. The effect was notable in the strings; all instruments with harmonics above 1kHz and vocals also benefited.

The resulting sound could challenge mini-monitor speakers at more than twice the price of the entire X-Z9 complex. This is a function of the properly executed crossover design packaged in a genuine three-way speaker with a highly innovative coaxial driver. The execution leads to an extremely flat response of the speaker, where the drivers and enclosure are free of resonances. Most speaker manufacturers could not pull this off because they source their drivers from third parties, opting not to customize their drivers given the huge expense. Even the few speaker manufacturers who develop drivers would be challenged to execute a speaker as flat and free of radiation pattern abnormalities as the S-Z9.

Figure 5 is an in-room measurement with the speaker placed at a different

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position. The high-end is strengthened by room effects and a variation in the resonance of the tweeter cone for this speaker sample. These manufacturing tolerance changes are common for less expensive metallic drivers and are amplified in a coax design by variations in the midrange cone that forms the tweeter's waveguide. Pioneer was kind enough to send two more S-Z9s to test, and they matched within 1dB out to 6kHz. All had identical dips.

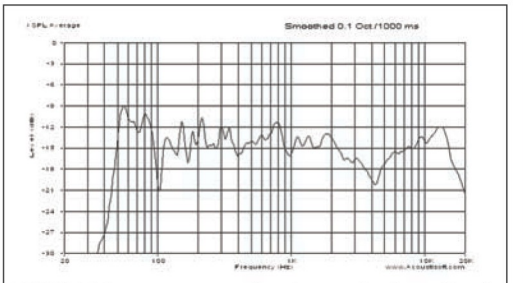


FIGURE 5: In-room response in using speaker in position 2.

The frequencies at which the bass responses in **Figs. 4** and **5** peak are not correlated, indicating the room effects are not from a port tuning that bumps up specific bass frequencies in the speaker. There is an honest 45Hz low-end, admirable for a small speaker with a tiny 5″ woofer. Pioneer gives a low-end response specification of 36Hz, but is silent on its depth. You can't fool Mother Nature, but you can cheat by lowering the efficiency of the speaker. Again, this is not an issue with an integrated system such as the X-Z9, which includes the electronics and speakers and is true to its mission statement: produce great sound in a small- to medium-size room.

From the room response, you wish for the removal of the dip and a smoother bass. Room effects can only be managed with an electronic room equalizer or expensive passive room treatment. I think the XC-Z9 has enough digital signal processing power to perform room equalization just the way it is. Pioneer uses similar Freescale DSPs in 7.1 channel audio/video receivers with room correction. If they can make it work with eight channels, it would be a snap with two. All they need to do is include a microphone and route the signal to the ADC.

Obviously, the cost of software development would need to be recovered, but I estimate the total cost of adding an effective room correction system from 40Hz to 300Hz is no more than $100. You do not need any equalization above 300Hz because this is the area that the speaker and not the room response will dominate. This speaker is so flat (with the dip removed), no equalization is required. Additional reasons equalization would not be required above 300Hz is that the exceptionally uniform off-axis response resulting from the coax designs ensures early reflection responses will be similar to the on-axis response.

To find out what would happen, I used the Lyngdorf as a proxy. Given the Lyngdorf's sophisticated algorithm for flattening the bass (±1dB ignoring room nodes that cannot be filled), I could not expect the same performance from the simpler Pioneer system, but it might be able to hold things to ±3dB below 300Hz, especially because the room correction algorithm would have knowledge of the S-Z9 characteristics. Nonetheless, with the Lyngdorf in front of the S-Z9, the bass became magical provided the speaker was not pushed past its limits (about 85dB). True, some of the low-end was missing, but not that many instruments reside below 45Hz.

WRAP-UP

In summary, the S-Z9 has the right stuff, but somebody painted a mustache on it in the form of a filter that makes a dip starting at 2kHz. In my opinion, this is a defect and Pioneer should eliminate the filter, because the resulting system would be superior to a more costly assemblage of components (SACD player, networked media system, stereo receiver, and three-way speakers) from disparate manufacturers. My thought experiment of adding room correction to a next generation XC-Z9 would take the system to another plane of excellence. Pioneer can bring it all together. I do not know of any other manufacturer that is equally well-positioned given their first mover advantage with the current X-Z9 and the unit's low price.

My message to Pioneer: Please remove that dip, allowing the excellence of the X-Z9 to shine and be known to all. Given the high level of engineering and technology that was needed to produce the electronics and the speakers' performance over nearly all their operating range, I have no doubt that the dip could easily be remedied. ***rr***

seas Loki
Speaker Kit

Madisound and Seas are pleased to offer the Loki kit. This small magnetically shielded system is the perfect choice for many applications. The Loki, with its coincidentally mounted tweeter, sounds fantastic from any position in your listening room.

The Loki uses an 18cm Seas coaxial speaker. The tweeter is mounted at the base of the woofer cone, coincidental with the woofer voice coil. This type of mounting allows you to hear the woofer and the tweeter at the same "time" and from any direction, nearly eliminating the confines of a "sweet spot". This configuration also allows you to use the speaker on its side without any change is sound.

Cabinets are available in Maple, Natural Cherry, and Black Painted Ash. The cabinet dimensions are 8" wide by 15" tall by 11" deep. The cabinets are constructed from 3/4" MDF with a 1" MDF front baffle. The cabinets are veneered with real wood on all sides. Cabinets are fully assembled.

Crossover are constructed from air coil inductors, polypropylene film capacitors and film resistors.

Frequency response is from 46Hz to 25kHz, with an average efficiency of 88db. The speaker impedance is 8 ohm.

Consider the Loki all the way around in your home theater system; as book shelf speakers; or as computer monitors.

Price Each $281.00

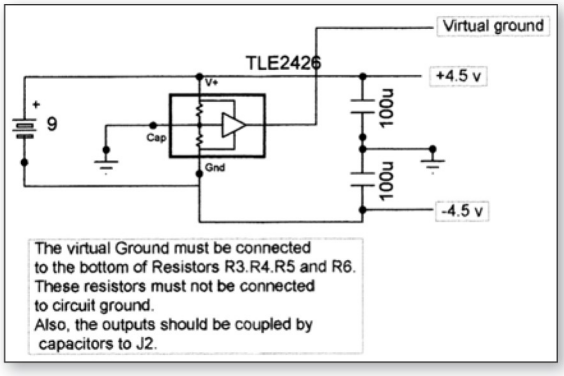
Madisound Speaker Components, Inc.
P.O. Box 44283
Madison, WI 53744-4283 USA
Tel:608-831-3433 Fax:608-831-3771
madisound.com
info@madisound.com

DIY AUDIO EVENT

The third international Burning Amplifier Festival will be held October 18 at the Presidio Yacht Club in San Francisco/Sausalito. An international group of audio DIY-ers will gather to show their projects and discuss audio matters. Presentations will include a talk on audio op amps by the designer of the AD797, Scott Wurcer, as well as a presentation of a new DIY project by Nelson Pass of Pass Labs. This event promises great equipment, great sound, and plenty of new ideas. Be there and bring your stuff! Go to www.burningamp.com for more information and to register.

CORRECTION

I would like to point out an error in your June 2009 issue. Ian Hilton (Xpress Mail, p. 45) refers to a revised design in Fig. 1 on p. 44. The enclosed schematic drawing shows the corrections with suitable notes.



Keep up the good work! I really enjoy your magazine.

Ed Sioma
sioma@erols.com

AMP PROJECT

Congratulations to Arto Terho for his very impressive redesign and the improved modifications he made to the 20W $260 amplifier (Feb. '09 *aX*, p. 38). I'm glad you enjoy the performance of the amp. I, too, was impressed with the sound. Your design using a larger chassis is indeed commendable. This amp, as I'm sure you'll agree, is ideal for a DIY project. Thank you for building it!

Joseph Still
Mt. Holly Springs, Penn.

HISTORIC GR DATA

The General Radio Historical Society is selling a series of CDs containing General Radio instruction manuals, handbooks, documents, photos, bulletins, parts catalogs, issues of the *Experimenter* publication, application notes, and technical papers. Proceeds from sales of this treasure trove of information will benefit the museum dedicated to General Radio, one of the giants in the instrumentation field. This effort is the work of Henry Hall, longtime instrument designer at GR, from his home in Concord, MA. For a listing of available CDs and pricing information, contact hphall@ieee.org.

TUNER MEASUREMENTS

In the Sony XDR-F1HD review (*aX* 8/09, p. 34), David Rich said Brian Beezley is the only reviewer still making measurements of FM tuners. Not so. I have made measurements for a number of FM tuners:

Blaupunkt Munchen RDM 169 Car Audio Head Unit, Hansen, C., *audioXpress* 9/01, pp. 62-65.

FM Tuners: Adcom, NAD, Parasound, Hansen, C. *audioXpress* 6/02, pp. 60-64.

NAD L53 DVD/CD Stereo Receiver, Hansen, C., *audioXpress* 12/05, pp. 38-53.

My FM test equipment is still available should you ever need FM tuner measurements.

Chuck Hansen
Ocean, N.J.

MORE IMPEDANCE TESTS

I welcome the opportunity to respond to Messrs. Blencowe and Mallory ("Impedance Dispute," Xpress Mail, 8/09, p. 45). It's at least 60 years past the time to clarify and settle this matter.

Regarding Mr. Mallory's letter, I'm with him through the paragraph where he points out that if voltage sources in *two* separate circuits were loaded identically, and their output voltages "changed equivalently" (the ratios of loaded to unloaded voltages were equal), then their output impedances would be equal. But then he applies this observation to a *single* circuit with loads on *two* sides of *one* source!

Because of that unsupported leap of logic, he is not entitled to his conclusion. I invite Mr. Mallory instead to derive expressions for the impedances (not voltages) of the plate (P) and cathode (K). Surely this is possible to do, and it gets right to the heart of the matter.

Regarding Mr. Blencowe: first, I never made the false claim that, ". . . mutual feedback between each output and the grid causes the output impedances to become identical. . ." Second, Mr. Blencowe needs to read all of his sources more carefully. The page 330 RDHB4 expression he cited is described by a sentence he incompletely quoted which states in full, "The equivalent source impedance that determines the high frequency attenuation for each channel due to shunt capacitance is given by (equation 34a)." But this "equivalent source impedance" is *not* the output resistance of P or K, which can be found earlier on the same page: "The effective output resistance is different for the two output channels, since P operates with current feedback and K operates with voltage feedback:"

Channel P:	rp` = rp + u Rk	where Rk = RL	(30)
Channel K:	rp` = (rp + RL) / (u + 1)		(31)

The Priesman paper Mr. Blencowe cites reprises the above equations as (1) and (2), with the exception that the u in RDHB4 equation (30) is replaced with u + 1. And in the Jones paper Mr. Blencowe cites, the second paragraph states, "The cathode source impedance, being that of an amplifier with degenerative voltage feedback, is low. At the plate terminal an amplifier with degenerative current feedback is seen, and here the source impedance is high." **All three authors specifically deny the claim of equal electrode impedances (EEI)!** RDHB4 goes on to explain that the currents out of the P and K are equal, and when they flow into equal loads, they produce equal voltages. This fact neither implies nor is implied by EEI.

Mr. Blencowe agrees that if one measures the output impedances one at a time, they do indeed "appear" different. But he opines that this is "pointless,"

since more than one output is used at a time, and so therefore this definition of output impedance is also pointless! (It is telling that he offers no alternative definition.) And although no proof of this sweeping assertion is supplied, it's supposed to be so "obvious" that the texts wouldn't even mention it! Allow me to observe that the absence from the texts of an error is even more likely than the absence of the obvious.

But you must pick your battles, so let's make sure that we respect Mr. Blencowe's admonition of "using the circuit as a phase inverter" when deriving its output impedances. From this I infer that he requires the P and K loads to remain identical and the voltages equal and opposite—the circuit must stay "balanced" at all times. Unfortunately, I don't know how to measure circuit impedances by adding a standard test load while keeping this circuit balanced. So I'll need to use a less common technique.

Impedances can always be expressed as a test current-induced change of voltage divided by that test current. The test current must flow between the circuit node under test and some point outside of the circuit under test. The following method is in accord with this principle, and I'd be glad to provide a generalized, algebraic proof of its equivalence to the more traditional ones if anyone is interested.

Referring to **Fig. 1**, drive opposing but equal AC test currents into the two cathodyne outputs. This maintains balance without adding any load impedances to the circuit being tested. Now write conservation of current equations at the plate and cathode:

$$\frac{Vp}{Rp} - ip + \frac{Vp - Vk(\mu + 1) + \mu \cdot Vg}{rp} = 0$$

$$\frac{(\mu + 1)\cdot Vk - \mu \cdot Vg - Vp}{rp} + \frac{Vk}{Rk} - ik = 0$$

You can solve the first equation for Vk, substitute the expression for Vk into the second, and then solve for Vp. The process is reversed for Vk, first solving the second equation for Vp, and so on. Sparing readers the algebra (which I'd supply if this weren't a letter), you can rearrange these equations to form two others more suited to our purposes:

$$Vp = \frac{[rp + Rk\cdot(\mu + 1)]\cdot Rp}{Rp + rp + Rk\cdot(\mu + 1)}\cdot ip + \frac{(1 + \mu)\cdot Rk\cdot Rp}{Rp + rp + Rk\cdot(\mu + 1)}\cdot ik - \frac{\mu\cdot Rp}{Rp + rp + Rk\cdot(\mu + 1)}\cdot Vg$$

$$Vk = \frac{Rk\cdot\frac{Rp + rp}{\mu + 1}}{Rk + \frac{Rp + rp}{\mu + 1}}\cdot ik + \frac{\frac{Rk\cdot Rp}{\mu + 1}}{Rk + \frac{Rp + rp}{1 + \mu}}\cdot ip + \frac{\mu\cdot Rk}{Rp + rp + Rk\cdot(\mu + 1)}\cdot Vg$$

There is a simple way to gain confidence that this second pair of equations is correctly derived from the first. Choose arbitrary values for all terms in the first pair except for ip and ik and calculate, solving for ip and ik. Now substitute all of these values into the second pair. The second pair will equate if they were correctly derived from the first. You can repeat this for more sets of values to gain even more confidence. (I routinely do this to check my derivations.) Or, you can continue the derivation process I described and arrive at the second equation pair yourself.

Let's look at the third equation. The portion of Vp that is due to the test current flowing between P and ground outside the cathodyne is expressed by the first (ip) term on the right-hand side. Dividing that term by ip leaves us with

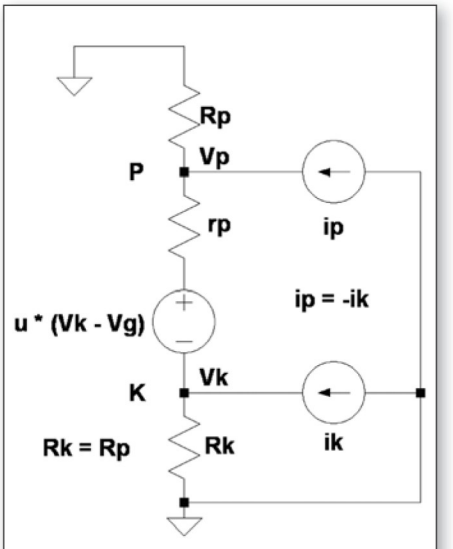


FIGURE 1: Balanced cathodyne phase splitter test circuit.

a result that can be expressed as Rp || rp + (1+u)Rk. But this is simply the plate resistor Rp in parallel with Priesman's result for the impedance looking into the plate, which as noted earlier is virtually identical to the RDHB4 result.

A similar approach is taken with the fourth equation, for Vk. Now the ik term is the relevant one. Dividing that by ik leaves an expression equivalent to the parallel combination of Rk and (Rp + rp)/(1 + u). This is the cathode resistor in parallel with the impedance looking into the cathode, as arrived at by both Priesman and RDHB4.

So now we have proven that the output impedances of a perfectly balanced cathodyne operating as a phase splitter are quite different. You can also see that these results are identical to what we would have obtained if we had applied the test currents one at a time. You can demonstrate this by turning on only the current of interest while removing (zeroing) the other. You can even set Vg = 0. All of this works because of the principle of superposition. Anyone who is surprised by this result may wish to investigate that principle. It can also be shown that the traditional method of adding test loads to the plate and cathode one at a time, unbalancing the circuit, will produce the same results.

To sum up, anyone still contending EEI will need to successfully refute RDHB4, Jones, Priesman, and the algebra above, to say nothing of providing his/her own algebraic proof—the gold standard of any such argument—for EEI. Anyone still asserting that the cathodyne P & K output impedances cannot be tested one at a time will need to explain why the results of doing so are identical to those of Priesman and the algebra above, and virtually identical to those of RDHB4.

To just ignore these results would obviously be pointless.

Chris Paul
cpaul@suffolk.lib.ny.us
aX

CONTRIBUTORS

Jim Wood ("Build The "Groove Jockey", p. 8) is the founder and CEO of Inovonics, Inc., a longtime California manufacturer of professional audio and RF support equipment for radio broadcasting. Jim has been an avid audio and RF circuit experimenter since the mid-1950s.

Kenneth L. Bird ("A Visit With Dave Weems," p. 17) has been building speakers since 1954, and was installing 12" speakers in the rear decks of 1958 Chevys and Cadillacs long before the word "autosound" was coined. Armed with a BSEE and much later with an MBA, he has spent most of his career in audio related pursuits with firms such as GTE, Rockwell International, and Shure Brothers. Currently, he is a Product Manager with MAXTEC International, makers of the B+K line of test equipment.

Bill Fitzmaurice ("The Jack 10, Pt. 2," p. 20) has been a professional musician since 1966 and has been constructing instruments, amplifiers, and speakers for just as long. Vice president of DeltaSounds Loudspeakers Inc., Fitzmaurice is the author of *Speaker Builder's Loudspeakers for Musicians* and over 30 magazine articles dealing with speakers and electric instruments, as well as the action-adventure novel *Operation: Sergeant York*. Bill and his wife of over 30 years reside in Laconia, N.H.

Jan Didden ("paX, Part 3," p. 24) built his first OTL amp with 807 tubes 35 years ago. He has built speakers, preamps, and tape recorders, but is most interested in power amps, especially using error correction as discussed by Hawksford. Many of his projects have been published by *audioXpress*. Now retired from a career with the Netherlands Airforce and NATO, he tries to complete all those half-finished projects accumulated for lack of time. He now also has the time to travel to interesting audio events and interview audio luminaries. His projects are documented on his Linear Audio website.

Ed Simon (Review: Burson Audio Modules, p. 30) received his B.S.E.E. at Carnegie-Mellon University. He has installed over 500 sound systems at venues including Jacob's Field, Cleveland, Ohio; MCI Center, Washington D.C.; Museum of Modern Art Restaurants, New York; The Coliseum, Nashville, Tenn.; The Forum, Los Angeles; Fisher Cats Stadium, Manchester, N.H.

Tom Lyle (Review: PrimaLuna ProLogue Six Tube Monoblock Power Amplifiers, p. 34) has written for the web magazine Enjoy the Music, as well as *The Sensible Sound*.

David A. Rich (Review: Pioneer X-Z9 Compact Stereo System, Pt. 2, p. 39) received his MSEE from Columbia University and his Ph.D. from Polytechnic University of NYU. He specializes in the design of analog and mixed-signal integrated circuits and has taught graduate and undergraduate courses in integrated electronics and electroacoustics. Student work under his guidance, including a novel high-efficiency mixed-signal integrated power amplifier, has won numerous awards. His industrial positions include Technical Manager at Bell Laboratories. His portfolio has spanned the design of audio ICs for Air Force One to RF ICs for wireless cell phones, and his innovations have earned 14 patents. He is a Senior Member of the IEEE and has frequently served as chairperson for technical and panel sessions at IEEE conferences. He has been a member of the AES signal processing technical committee and has been Technical Editor for *Audio Critic* and *Sensible Sound*. He is the head of the music committee of the Bethlehem Chamber Music Society.

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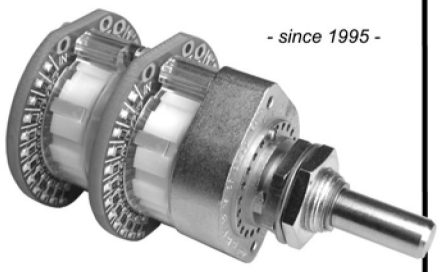
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