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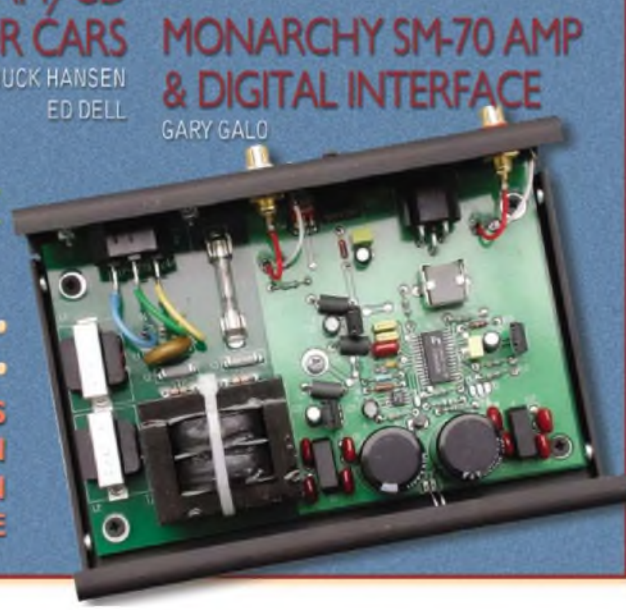

Reviews:

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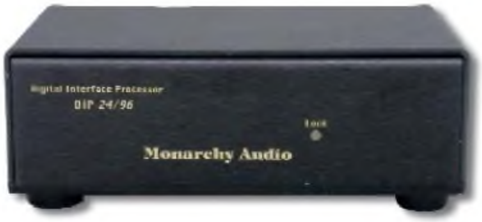

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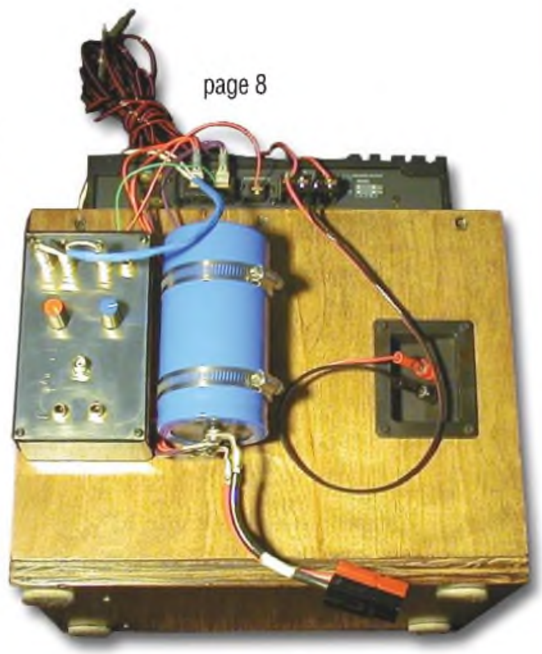

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THE STAFF

Editor and Publisher
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Regular Contributors
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Kelly Bennett Customer Service
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Advertising Department
Alden Sales, Inc.
PO Box 271
Rindge, NH 03461
603-899-3010
Fax: 603-899-2343
E-mail: advertising@audioXpress.com

Rich Alden
Advertising Sales

Louise O'Sullivan
National Sales Manager
603-899-5068

Nancy Vernazzaro
Advertising/Account Coordinator

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

JOHN STUART MILL

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CLASSIFIEDS & WEB LISTINGS
Contact Nancy Vernazzaro, Advertising Department, *audioXpress*, PO Box 876, Peterborough, NH 03458, 603-924-7292, FAX 603-924-6230, **E-mail nancy@audioXpress.com.**

Audio News

NEW SILVER SONIC PRODUCTS FROM DH LABS

DH Labs added several new products to its Silver Sonic line of cables. The Q-10 speaker cable features a 4-conductor, multiple-gauge design, and the Air Matrix interconnect features DH Labs' proprietary new air-teflon matrix dielectric (which can provide stable dielectric performance to 17GHz). Other new products include the Deluxe Toslink optical cable, the Subsonic subwoofer cable, and composite and component video cables. For more, contact DH Labs, 612 N. Orange Ave. #A-2, Jupiter, FL 33458, phone/FAX 561-745-6406, website:www.silversonic.com.

SOUNDELUX MICROPHONES

Soundelux Microphones announced that the ELUX 251 microphone is now shipping to dealers nationwide. Modeled after the classic Telefunken ELAM 251, the ELUX offers the same performance as vintage models, but at a significantly lower cost. The ELUX 251 construction features hand-built, hand-tuned, and strictly tested European capsules, with time-proven tube and transformer circuitry; all electronic components are point-to-point wired, with each individually prescreened for sonic quality. For more, contact Livewire Studios, 323-603-2131, FAX 323-603-3287, e-mail: enusbaum@soundelux.com.

QUAD II VALVE AMP

The QUAD II valve amp is available again. You can order your pair of serial-refurbished QUAD II amplifiers, in excellent condition, directly from QUAD Musikwiedergabe GmbH, Rheinstrasse 30, 56068 Koblenz, Germany, +49 (0)261 38824, FAX +49 (0)261 38172, e-mail: quad-gmbh@t-online.de, website: www.Quad-Musik.de.

SOUNDSMITH MODEL CDT-4

Soundsmith introduces the Model CDT-4, a non-invasive tester for "catching" intermittent conditions anywhere an analog signal can go. The CDT-4 tests for such conditions as signal level change, "noise distortion," or signal shut-down. It features a multi-segmented display on the front of the instrument to indicate the number of tests, number and type of failures, and when during the test they occurred, logging errors for up to five days and displaying them on large numeric LEDs. The CDT-4 comes with a specially designed audio CD, for automated integrity testing of any analog signal path, as well as a 13V AC transformer, and a computer interface to allow you to view, store, or produce a printout of the results. For more, contact Soundsmith, 8 John Walsh Boulevard, Suite 417, Peekskill, NY 10566, 800-942-8009, FAX 914-739-5204, website: www.sound-smith.com.

PS110 CORNER SUB

Tannoy offers a new subwoofer, the PS110 Corner Sub, designed with space and décor issues in mind. This subwoofer features a 10" down-firing bass unit powered by a 110W MOSFET amplifier, is unobtrusive, and is designed to take advantage of the additional loading offered from use in the corner of a room. Tannoy, 519-745-1158, e-mail: litplease@tgina.com.

U-VOLA™ SPEAKERS

SYN FACTORY SRL is producing an innovative loudspeaker called U-VOLA™, designed by Alessandro Copetti. U-VOLA™ is a hanging-suspended, elliptically-shaped loudspeaker based on a mass cabinet without joints and made of a mix of mineral aggregates (KCF®). The homogeneity of the structure and the particular frame reduce reflections and reverberations, and the suspension (a steel wire from the ceiling, a wall-bracket, or a tripod) helps to cancel resonances and/or vibrations by avoiding contact with surrounding surfaces. The simple and fashionable shape of U-VOLA™, as well as its wide range of colors and finishes, makes it suitable for many listening environments. For more, contact SYN FACTORY SRL, 33010 Gemona del Friuli (UD)-ITALY, Zona Art Campagnola, 1, +39 (0432) 972362, FAX +39 (0432) 972684, e-mail: info@synfactory.com, website: www.synfactory.com.

CHORD COMPANY CABLES

Bluebird Music announces the immediate availability in North America of the Chord Company's range of high-performance audio and A/V cables. Chord offers eight interconnects, three speaker cables, a variety of digital interconnects, video cables, and in-wall installation speaker wire. To achieve superior performance, they use only Polyethelyne and Teflon® insulators in the Chord cables; they never use PVC. Chord also offers a variety of RCA, XLR, DIN, BNC, Optical, and other plugs. Bluebird Music is now looking to set up a limited network of authorized dealers throughout North America. For more information, contact Bluebird Music Limited, 620 Wilson Avenue, Suite 360, Toronto, Ontario M3K 1Z3, 416-638-8207, FAX 416-638-8115, e-mail: sales@bluebirdmusic.ca, website: www.bluebirdmusic.ca.

SPATIAL AUDIO

Focal Press announces the upcoming publication of *Spatial Audio* by Francis Rumsey and *The Microphone Book* by John Eargle. *Spatial Audio* explores the principles and practical considerations of spatial sound recording and reproduction; the increasing importance of multichannel surround sound and 3D audio, including binaural approaches, is emphasized, without ignoring conventional stereo. *The Microphone Book* is a comprehensive guide to the latest in microphone technology, application, and technique. For more, contact Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458-0876, 603-924-6371, 888-924-9465 (US/Canada), FAX 603-924-9467, e-mail: custserv@audioXpress.com, website: www.audioXpress.com.

VSAC SHOW

The third annual VSAC Show will take place September 7, 8, and 9, 2001, at the West Coast Silverdale Hotel in Silverdale, Wash. Sponsored by the Bottlehead Corporation, this show is geared toward the DIY audiophile. It will offer original designs in tube equipment, a Craftsman's Room (showcasing designs and demonstrations by amateur builders), a Seminar Series, and a Vintage Audio Room (displaying antique radios and pre-transistor items). The show is open to the public, and the press is very welcome and encouraged to attend. For more, please contact Eileen Schmalle, Bottlehead Corporation, 360-662-1386, e-mail: eileen@bottlehead.com.

ALTEC HEADSETS AND HEADPHONES

Altec Lansing Technologies, Inc., announces the company's entry into the headphone and headset markets with a new line of products designed for personal entertainment and voice-entry. Designed for users on the go, the headphones (AHP) are perfect for a variety of portable audio devices including MP3, CD, and DVD players. For use in the office and at home, the headsets (AHS) provide accurate voice capture and sound quality from desktop and personal computers. Altec Lansing Technologies, Inc., Milford, PA 18337-0277, 800-258-3288, website: www. altecmm.com.

Build an Automotive Sub-Satellite Speaker System

Here's an ambitious car-installation project to help make that daily commute or vacation trip that much more enjoyable. **By Jules Ryckebusch**

I have always been an audio enthusiast. Having successfully designed and built several home speaker systems, I decided to tackle an automotive sound system. Typically, I have not been enthusiastic about "stock" car stereos and found the aftermarket full of hype and, of course, the ever-rampant "Bass-Thumpers." I prefer a complete audio frequency response from a stereo. My musical tastes run the gamut from classical to hip-hop, so good low-end response is still a requirement.

I have put a lot of thought into the best method for stereo reproduction in a vehicle and kept returning to the concept of the subwoofer/satellite speaker system that features physically small satellites and a "hidden" subwoofer. I also chose to keep the cost down and not use esoteric components. To achieve the system I envisioned, an active crossover and bi-amping are required (*Photo 1*).

PHOTO 1: Completed system.

COMPONENT SELECTION

After some extensive research on car audio amplifiers, I settled on a KAC-848 Kenwood four-channel amp capable of supplying 50W RMS a channel to a 4Ω load at an acceptable distortion level of .1%. I could then bridge two channels and send 100W to the subwoofer. Be careful when selecting amplifiers because there are misleading specifications on many "car audio" products. Some three- and four-channel amplifiers I investigated were, in reality, two-channel amps that had extra connections or had the subwoofer tap off the right speaker (+) lead and the left speaker (–). Hmmm...This just didn't seem right!

In fact, I mail-ordered one amplifier only to find out it wasn't really what the ad led me to believe. For comparison purposes, the Bose subwoofer satellite system with two satellites provides 20W RMS to the satellites and 90W RMS to the subwoofer.

After selecting the amplifier, it was time to choose the drivers. For the subwoofer portion I settled on 8″ drivers used in an isobaric acoustic-suspension enclosure. I am a big fan of acoustic-suspension enclosures, even though they are not as efficient as ported enclosures. They do provide a smoother frequency response and an extended low end below the cut-off frequency of the driver/enclosure.

Breaking out my Parts Express catalog (800-338-0531, www.partsexpress.com), I searched for potential drivers. I settled on a treated paper cone driver made by Dayton Loudspeaker (part #295-310), which was suitable for sealed enclosure use and cost about $20 (*Photo 2*).

Then I took the driver parameters and consulted the *Loudspeaker Design Cookbook* (Fifth edition, Vance Dickason, Audio Amateur Press) to refresh myself on design equations for sealed enclosures. I set Q_{TC} = .707, my preferred Q for a sealed box, and set the equations up in an Excel spreadsheet. The results were an initial box volume of .83ft^3 for one woofer.

I then cut that in half because I was using an isobaric configuration. I decided not to use the "golden rule" for box dimensions and square off two dimensions for the box and settled on enclosure dimensions of 10″ × 10″ × 7″, which was very close to the theoretical box volume.

Then it was time to select a high-frequency driver. After much thought concerning size and cost, I settled on an Audax HT080MO 3″ woofer (Parts Express number 296-015). It features a frequency response of 80Hz–18kHz. Even though it is listed as a woofer, it is more of a full-range driver (*Photo 3*).

PHOTO 2: Subwoofer.

I decided to use two drivers for each satellite for a total of four.

PHOTO 3: Satellites.

Since I wasn't trying to obtain a low-frequency response from the drivers, I wasn't worried about box size and put the Audax units in as small an enclosure as I could fit them.

CROSSOVER DESIGN

I had previous experience with an active third-order crossover design, which I use for my PA system and DJ work. It is derived from Walt Jung's *Audio Applications of Operational Amplifiers* (1987). Walt's original circuit was set for a 500Hz cutoff. Several car audio installers suggested I select it as low as possible, i.e., under 100Hz.

The built-in adjustable crossover allowed me to do some initial experimentation with crossover frequency. I discovered that too low a frequency caused problems with cone excursion in the satellites. Additionally, the subwoofer response easily extended to around 1kHz. I settled on 250Hz as an optimal balance (*Photo 4*).

Along with performing the active crossover function, I needed the circuitry to perform a couple of other functions, such as additional gain to adjust the levels of my speaker amplifier system to augment the installed radio/cassette/CD changer. Optimally, both the installed stereo and the sub/satellite system should reach maximum volume prior to distortion from both the installed stereo and the sub/satellites at the same volume setting. Not knowing the efficiencies I would achieve with either the subwoofer or the satellites, I needed to be able to independently adjust the signal level for the subwoofer and the mid/high frequency feed. I also only needed a mono subwoofer signal.

I do most of my design work in Circuit Maker, which I really like. It allows for SPICE simulation and provides frequency-response graphs that accurately reflect real-world circuit response. It also allows for experimentation with component values to optimize a design prior to building a prototype. I ended up with a final cutoff frequency of just above 250Hz. *Figure 1* shows my final schematic.

The right and left line-level signals are first buffered with a voltage gain of about 10dB. The right and left channels are sent to third-order high-pass filters and also summed and sent to the low-pass filter section.

I am using OPA2134 dual operational amplifiers from Burr-Brown, which I have used in several other audio projects. They feature .00008% distortion—

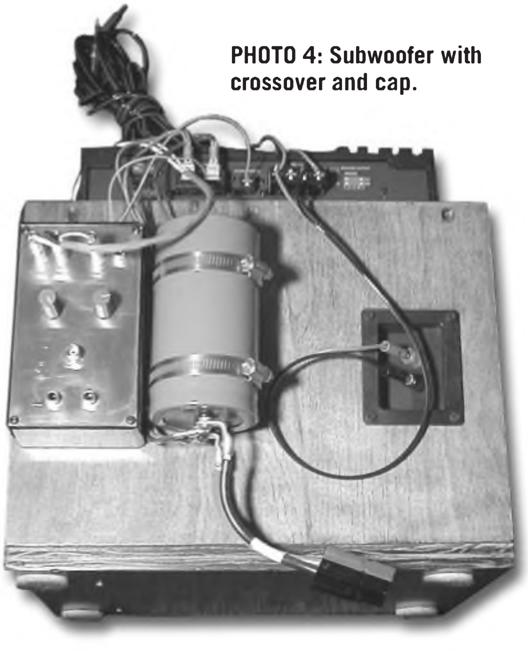

PHOTO 4: Subwoofer with crossover and cap.

yes, that is less than 1PPM! They also feature 20V/µs slew rate and very low noise. I recommend downloading the data sheet from www.burr-brown.com. Small quantities are relatively inexpensive from Digi-Key. All resistors are 1% metal film. All Panasonic capacitors (also available from Digi-Key) are 2% polypropylene.

Figure 2 shows the frequency-response graph of the low-pass section with a –3dB frequency of around 260Hz. The frequency response of the high-pass section is shown in *Fig. 3*. Combined response of the low-pass and high-pass sections is shown in *Fig. 4*.

I achieved just under a 1dB dip at the crossover frequency using standard component values. You may have noticed that there is a phase inversion occurring in the low-pass section that is not in the high-pass sections due to the inverting summer I am using. I believe that due to the physically different locations between the sub and the satellites that this won't be an issue.

CONSTRUCTION

The subwoofer enclosure is made from ¾″ thick birch plywood. All the dimensions are given in *Fig. 5*. Construction is standard butt joint with the exception of the sides. Glue and countersunk sheetrock screws hold all the pieces together.

To allow access to the enclosure for stuffing and potential experimentation, I left one side unglued, held together only with screws. I used a router to cut out the holes and mounted the two drivers with the supplied gasket material. You need to drill two small holes in the inner drivers' mounting board to allow the wires from the front driver to pass through to the second. Both are wired in parallel providing a 4Ω load to the power amp.

Stuff the enclosure with half a bag of fiberglass insulation from Radio Shack. I used 12-gauge type "E" Teflon-insulated wire for the internal wiring to the external banana-jack plate. I then finished the entire enclosure with stain and semi-gloss polyurethane.

The two satellite speakers are housed in ½″ birch plywood enclosures. As previously mentioned, size was the determining factor in their design. This, coupled with such a high crossover frequency, ensured that enclosure volume wouldn't affect their performance. Dimensions are given in *Fig. 6*. Based on some other potential uses for my system, I chose to use recessed ¼″ phone

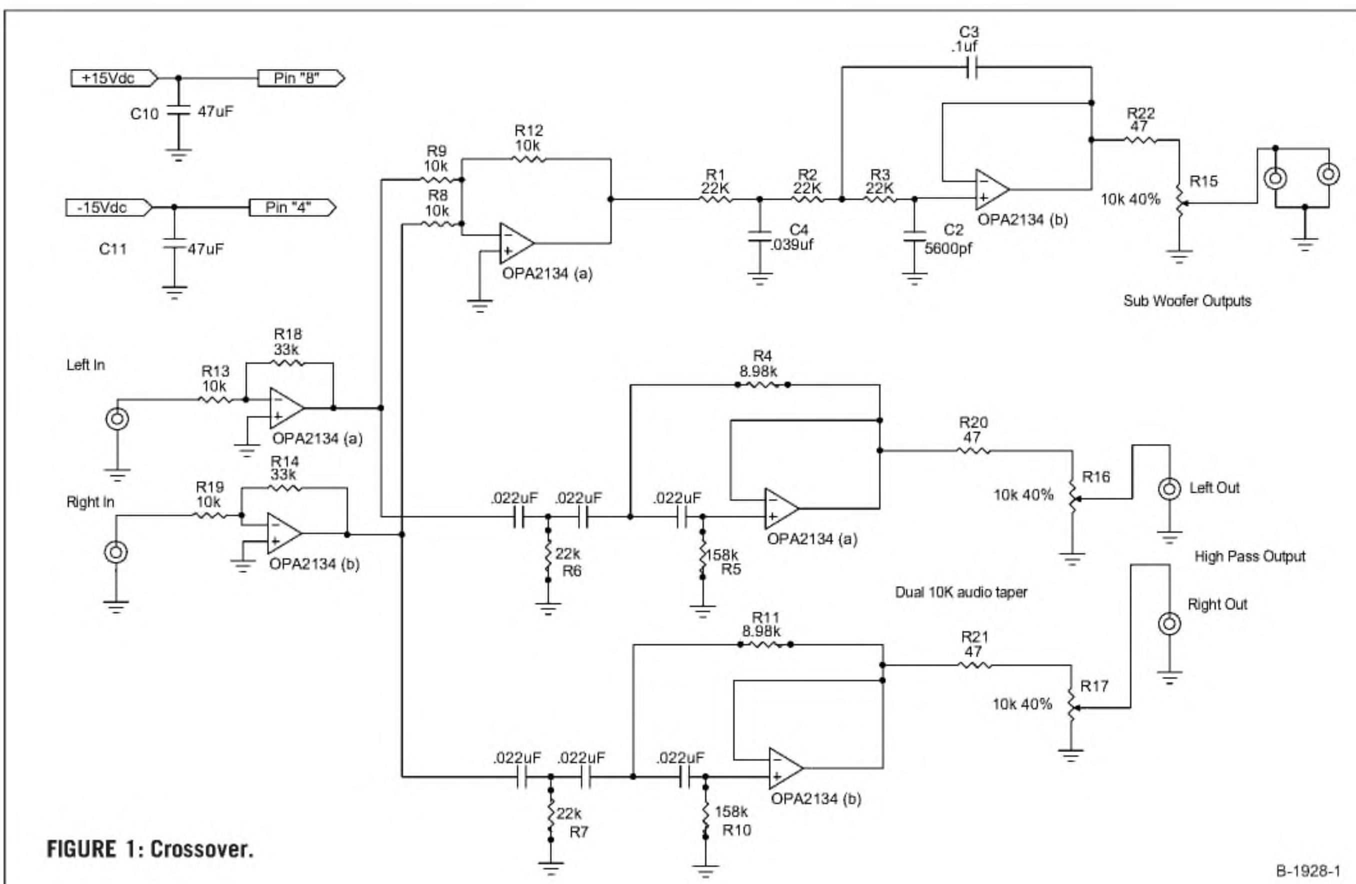

FIGURE 1: Crossover.

TABLE 1
PARTS LIST

ACTIVE CROSSOVER	
DESIGNATION	**DESCRIPTION**
C1, C5–C9	.022µF
C2	5600pF
C3	.1µF
C4	.039µF
C10, C11	47µF
J1–J5, J7	RCA jacks
R1–R3, R6, R7	22k
R4, R11	8.98k
R5, R10	158k
R8, R9, R12, R13, R19	10k
R14, R18	33k
R15–R17	10k var
R20–R22	47
U1–U3, U5–U7	OPA2134
"±15V DC SUPPLY"	
C1	.1µF
C2	47µF
C3, C4	10µF
U1	7812
U2	580-NMH1215S
SPEAKER/DRIVERS	
Dayton 8″ woofer, PE #295-310	
Audax HT080MO 3″ driver, PE #296-015	
MISCELLANEOUS	
6 male RCA plugs	
wire	
screws and hardware	
chassis box for crossover	

jacks to connect the satellites to the power amplifier.

I used glue and butt joints for assembly, clamping the unit until the glue dried. I assembled the boxes without the faceplates and then finished the boxes and the faceplates using the same stain as the subwoofer enclosure. I installed and wired the drivers to the recessed jack in the back of the enclosure. Once again I used 12-gauge wire. (Probably a little overkill!)

After stuffing enough fiberglass in the enclosure to still allow the faceplate to set comfortably, and testing the wiring with an ohmmeter, I glued the faceplate directly to the case of the enclosure. This may not be the most desirable assembly method. I figured that if I needed to tweak these enclosures, I was starting completely from scratch anyway, so, in one sense, they were disposable.

ELECTRONICS ASSEMBLY

I built the active electronics portion on a Radio Shack experimenters PC board 278-168. These work great for single designs. The only remaining problem was where to get the ±15V DC to run the circuit.

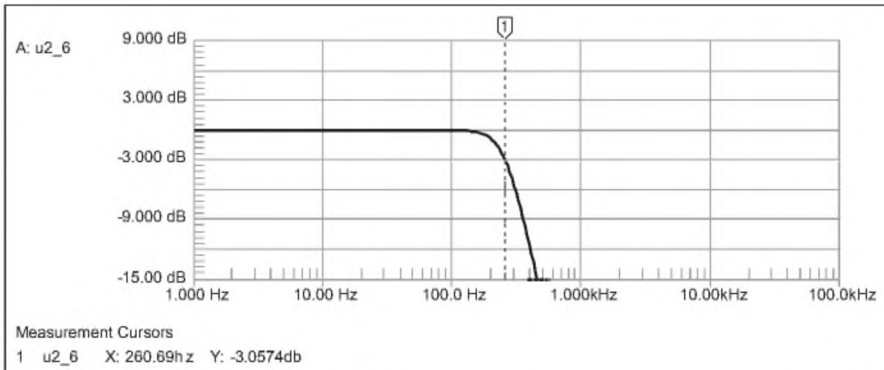

FIGURE 2: Low-pass frequency response. B-1928-2

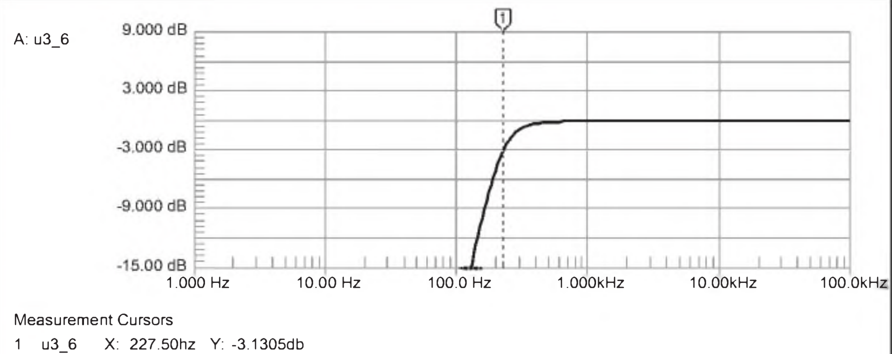

FIGURE 3: High-pass frequency response. B-1928-3

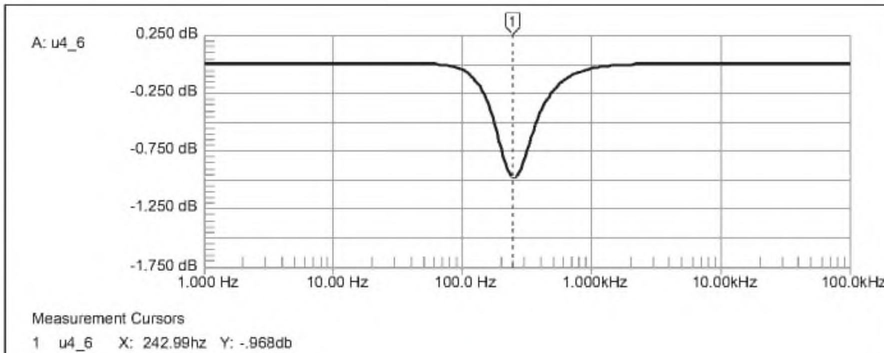

FIGURE 4: Combined response. B-1928-4

I had previously noted many small (under 5W) DC-to-DC converters that were appearing on the market and ordered a 2W 12V DC to ±15V DC model from Mouser Electronics (part #580-NMH1215S) made by Newport Components. I was very interested in how well it would perform.

Once it arrived, it was time for a little bench testing. I hooked it up to a variable supply and loaded it with a couple of resistors that would draw 50mA per supply. I then connected the outputs to my DMM. With 12V DC input I got +14.98 and −15.03V DC. Excellent!

Now, for the acid test, I started varying my supply voltage and the output tracked the input almost perfectly. This wouldn't work well.

The 12V DC in an automobile is actually 13.6–14.4V DC, depending on your alternator. The minimum to keep a float charge on the car's lead acid battery is 13.6V DC. So worst case with 14.4V in, the output became ±18V DC. The op amp I am using is rated at ±18V DC, and I thought I needed to be a little more cautious.

I ended up pre-regulating the car's "12V DC" to 12V with a 7812 three-terminal regulator. This sounds strange but it works. The nominal auto voltage of 13.6V DC is high enough to ensure the 7812 has enough "head room" to function properly. In the real world, with less than 12V into the regulator, there is an output less than 12V but high enough for the DC-to-DC converter to function, although less than ±15V DC out. The power-supply schematic is shown in *Fig. 7*.

I brought out the +15V DC to a BNC jack so that I could easily connect it to my oscilloscope and really see what was going on. My concern was noise getting into the op-amp circuitry. To my surprise, there was very little ripple to worry about.

You need to keep track of grounds. The 7812 and the DC-to-DC converter input are tied to the vehicle's normal chassis ground. The output of the DC-to-DC converter is isolated from the vehicle's chassis ground and ties in at only one point, which is where the outputs of the crossover connect to the power amplifier.

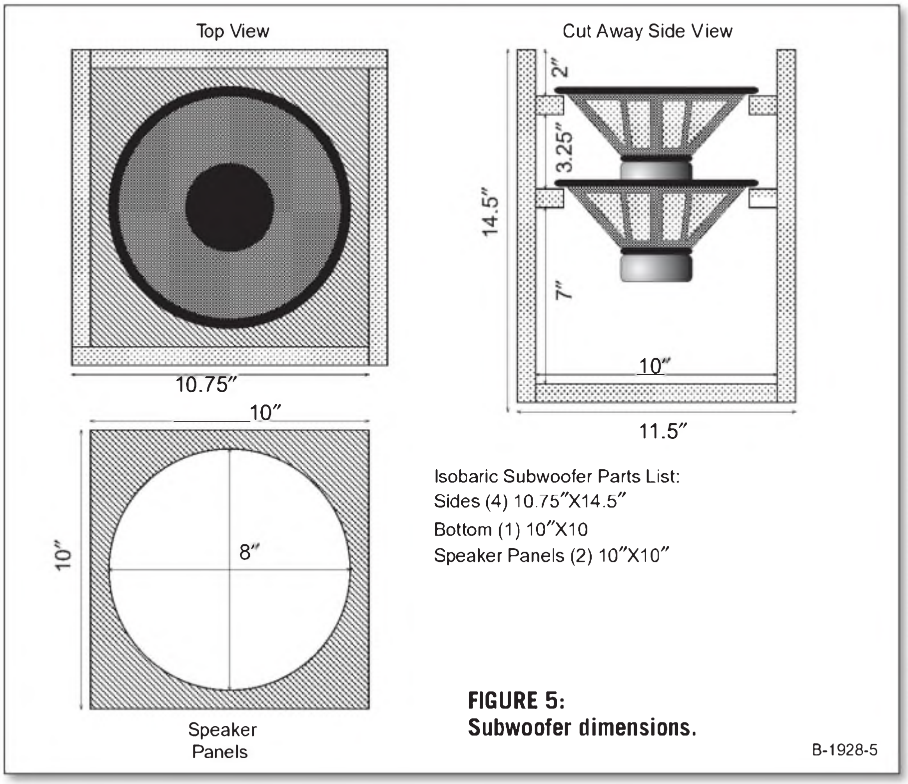

FIGURE 5: Subwoofer dimensions.

SYSTEM ASSEMBLY AND TESTING

I have a bench supply capable of supplying 12V DC at 20A, so I was able to perform all my testing and circuit tweaking on the bench in my garage. As previously mentioned, I initially attempted to use the built-in adjustable 12dB/octave crossovers in the Kenwood amplifier. These worked well enough to prove my idea of a sub/satellite system. They did not merit direct use in the vehicle.

My other discovery was that when the amplifier was set up to use the built-in crossovers, individual level controls did not function; both amplifier sections were fed by section "A" gain controls. This prevented independent level controls that I needed to balance out the differences in efficiencies of the subwoofer and the satellites.

The electronics assembly all mounts into a small Bakelite chassis box with an aluminum cover that facilitates input and output jacks. Note that I am using two jacks for the bass amplifier. I needed only one with the amplifier I finally selected. The Kenwood allows two of the channels to be bridged with a dipswitch setting.

I mounted the crossover chassis and the Kenwood amplifier directly to the subwoofer box. I also included a large filter capacitor (110,000µF) rated at 20V DC mounted on the subwoofer box. I added a quick disconnect rated for 50A to the capacitor via 6″ or so of wire to facilitate simple removal of the entire system.

For my particular installation, I ran a wire harness that contains the power leads, the line-level left and right audio, and a wire that when energized from 12V turned on the amplifier and supplies power to the crossover circuitry. For power leads I connected directly to the vehicle's battery posts with 8-gauge wire.

There is a 50A fuse mounted to the firewall in the engine compartment for protection. I mounted a small toggle switch in my dashboard for turning on the amplifier and crossover. This keeps the system separate from the main stereo and allows me to turn it on independently.

FINAL INSTALLATION AND LISTENING

After spending a couple hours running

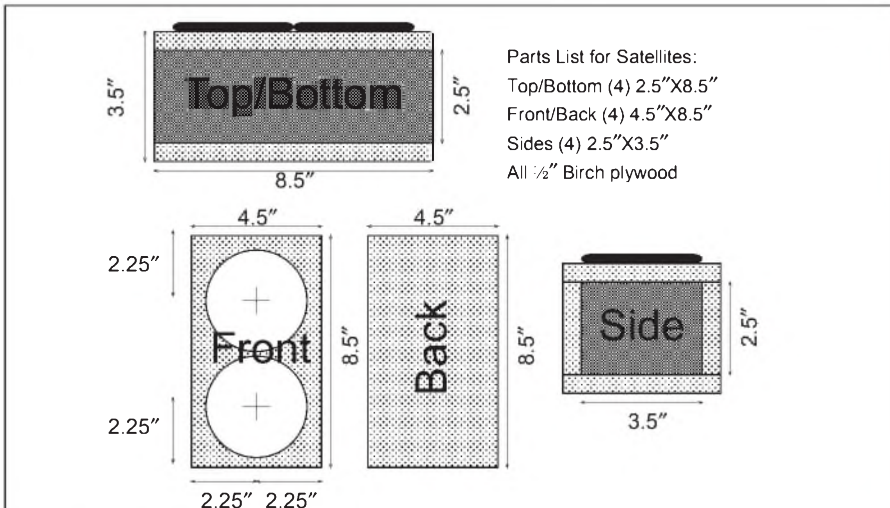

FIGURE 6: Satellite dimensions.

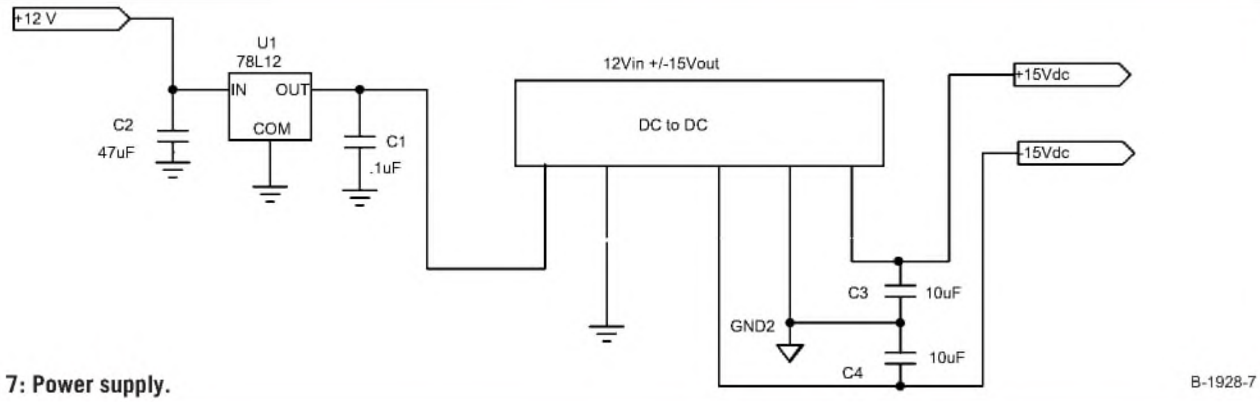

FIGURE 7: Power supply.

wires, removing and reinstalling the dashboard stereo head unit, I was ready to put the system in my van and check it out. I initially turned both level controls on the crossover all the way down and powered up the amplifier and crossover. Then I brought up the level controls and was very happy with what I heard.

There were still two problems: first, a small amount of alternator-type buzzing that was evident at low volume levels; and, second, how to mount the satellites. Where I had initially thought of putting them wasn't going to work. Initially I placed them on the rear-most seats facing up to allow me to set levels for the sub/satellites. This worked well and I found that setting the level of the satellites fully up and the sub about 75% full volume worked best. Remember that the head unit in the dash controls total system volume.

Now, where was the buzz coming from? I found a ground loop, which I eliminated. I had initially wired the 7812 incorrectly and actually floated its ground. Once this was resolved, there was no more buzz.

Now all I needed to do was figure out how to better mount the satellite speakers! Actually, there are many options here, but I wanted to be able to easily remove the entire system, so I am quite happy to leave them on the far-back seats facing upward. All in all, I am quite pleased with the entire system, which makes road trips much more enjoyable. I am thinking about redoing the satellites in an MTM configuration with a dome tweeter, but that is for another day! ❖

The AC Power Line and Audio Equipment, Part 1

Here's a look at the many factors affecting your power line and some tips on how to protect your audio equipment. **By Charles Hansen**

Audio/video equipment connected to the AC mains can handle more than the 50–60Hz power necessary to operate it. The alternating current at your AC receptacle is laden with power-line harmonics, voltage fluctuations, transients, radio interference, and other disturbances that, in all probability, will reduce your listening pleasure.

THE ELECTRICAL GENERATING STATION

The utilities (or increasingly in these days of deregulation, holding companies) generate our 60Hz AC line power primarily with synchronous AC generators[1], or more properly, alternators. Electronic power inverters operating from solar power, windmills, or fuel cells may produce a small fraction of the power, often storing it in batteries for peak usage.

Alternators are energy-conversion devices, or *transducers*, that convert mechanical to electrical energy. Transducers of all types operate through the physical laws that relate magnetic and electrical fields to mechanical force and motion:

1. A mechanical force is exerted on a current-carrying conductor in a magnetic field. Conversely, voltage is induced in a conductor moving through a magnetic field.

2. A mechanical force is exerted on a magnetic material that tends to align it with the highest magnetic flux density path. Conversely, moving a permanent magnet or electromagnet past a stationary conductor causes a change in the magnetic flux linking the conductor and will induce a voltage in the conductor.

3. The magnetic properties of a magnetic material are changed when it is mechanically strained. Conversely, magnetic materials are slightly deformed when placed in a magnetic field. This is called magnetostriction.

4. When certain crystals are mechanically strained, they form an electric charge. Conversely, when voltages are applied to them, these crystals are deformed in certain defined directions. This is called the piezoelectric effect.

5. A mechanical force is exerted on the plates and insulators of a charged capacitor in an electric field. Conversely, relative motion between capacitor plates and insulators causes a change in the charge or voltage across that capacitor.

APPLICATION OF THE LAWS

You can also see practical applications of these laws in audio transducers: loudspeaker and headphone drivers, relays, microphones, phono cartridges, crystal oscillators, guitar pickups, and magnetic tape heads.

The rotor of an alternator (*Fig. 1*) consists of coils, wound around iron poles, that carry a DC current whose magnitude is controlled by a voltage regulator. This rotor and its windings are called the field, and it produces alternating north and south electromagnetic poles. The DC field current is applied to the rotor in one of two ways: through carbon brushes and slip ring conductors, or by means of rotating rectifiers following a smaller exciter AC generator on the same shaft as the main alternator.

As the rotor turns inside the stator or *armature* of the alternator, the time-varying magnetic flux resulting from the rotation of the fixed magnetic field induces current into the stator windings, which are distributed in a number of slots inside the stator iron. A very small field current (maybe 3% of the rated alternator output current) from the regulator controls the much higher stator current.

The frequency of the AC voltage is determined by the number of poles on the rotor and its rotational speed (rpm). A four-pole machine turns at 1,800 rpm to produce 60Hz. (Countries outside North America generate 50Hz power, while the aircraft industry uses 400Hz power.)

The weight of the alternator for a

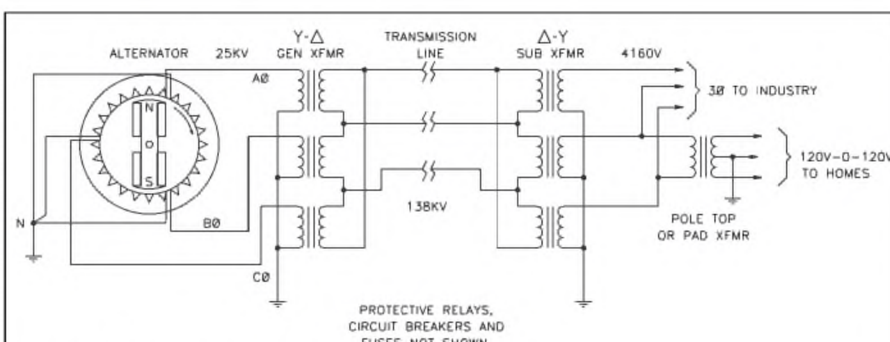

FIGURE 1: Electric generation and distribution system.

given output power is directly proportional to the rotating speed of the alternator. Fewer poles require higher rpm, but also result in a lighter machine. However, the rotor must then be designed for the higher centrifugal forces that occur at higher speeds. The field poles are always wound to produce alternate north and south polarity, so only an even number of poles are possible. An alternator must have at least two poles (one north, one south), so 3,600 rpm is the maximum speed for 60Hz.

HARMONICS

It is just not feasible to generate a pure 60Hz sine wave, so cost-effective electric alternators produce a number of low-order 60Hz *harmonics*. These are sinusoidal currents or voltages with frequencies that are integral multiples of the fundamental power-line frequency, and they distort a pure sine wave.

Each of these harmonic sinusoids has a different magnitude from the others. Typically, each harmonic is represented as a percentage of, or dB below, the root mean square (RMS) value of the fundamental. Total harmonic distortion (THD) equals the RMS value of all harmonics divided by the RMS value of the fundamental, usually converted to a percentage.

Even-order harmonics (second, fourth, and so on) are vanishingly small or nonexistent at the alternator terminals, since a single stator winding circuit produces both the positive and negative AC half-cycles as the alternate magnetic rotor poles pass by.

Alternator designers can take advantage of physics to reduce 60Hz harmonics. In three-phase machines with balanced loads (equal current in all three phases), the *triplens*—the third harmonic and its odd multiples—can easily be cancelled in one of two ways. If power is taken directly from the alternator terminals, the stator winding can be made with a 120° *phase belt*, which fully cancels triplens. Now the winding chord angle, or *pitch*, can be designed to cancel any two other adjacent odd harmonics—the fifth and seventh are usually the highest.

A 60° phase belt results in a slightly smaller machine, but the winding pitch must then be fixed at ⅔ (0.667) to cancel triplens, and you must accept the higher harmonics as they come. Machines with 60° phase belts are sometimes used in aircraft, where weight is of primary concern.

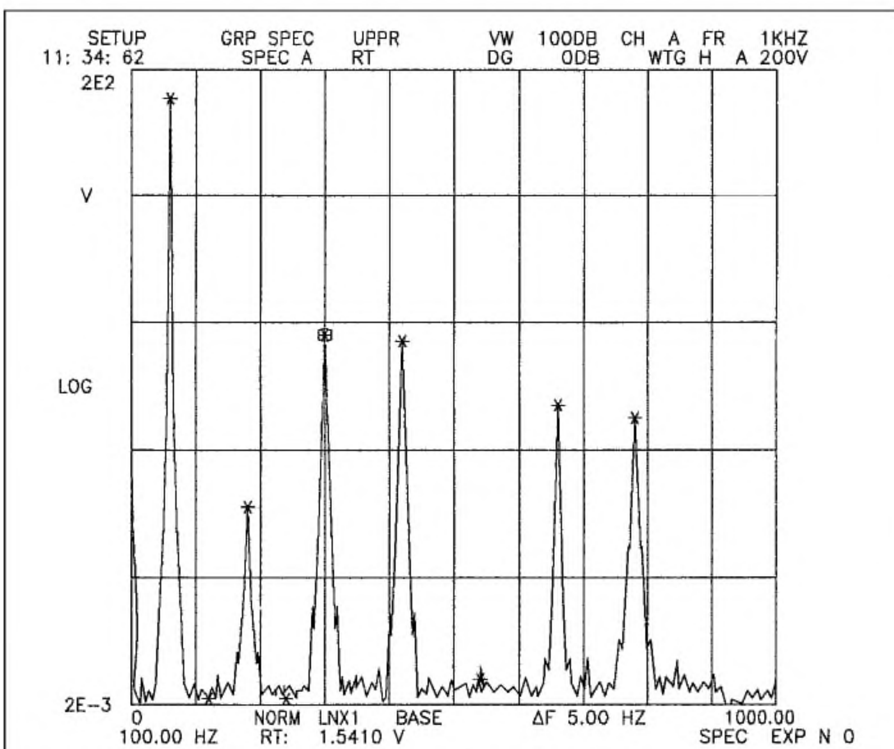

FIGURE 2: Spectrum of 60Hz harmonics at generating station.

The curved profile (pole arc) at the face of the rotor pole also has some effect on the higher harmonics, and optimizing the shape of the pole tips that overhang the windings can help reduce higher odd harmonics.

SLOT HARMONICS

An alternator also generates higher harmonics because its stator laminations do not have smooth surfaces. The stator windings reside in slots in the iron that cause flux pulses (called *slot harmonics*) as the rotor passes. By making a slight twist in either the rotor-pole axis or the stator slots, the magnetic axis is skewed by one stator slot-pitch, thus reducing the space harmonics introduced by the winding slots.

Since the utilities must generate huge amounts of power (millions of kilowatts) and transport it over long transmission lines, they do not send power directly from the alternator terminals. Instead, they step the voltage up with large transformers, say from 25kV at the alternator to 138kV or more. This lessens the current in direct proportion, reducing the transmission losses (I^2R).

It is desirable to use a three-phase Y connection in the alternator so its neutral can be grounded—either directly or through a neutral reactor, a device used to limit the unbalanced fault current in a three-phase system. This provides lightning and overload protection. The engineers can then use a wye-delta (Y-Δ) transformer (*Fig. 1* again) to remove the third harmonic, since in a current-balanced three-phase delta transformer winding it cancels out.

With the third harmonic cancelled in the transformer, the alternator can use a 60° phase belt, and a winding pitch of ⅚ (0.833) will cancel the fifth and seventh harmonics. As a result, the 60Hz sine wave at the step-up transformer secondary has a typical unity-power-factor distortion of 1% to 1.5%, mostly due to nontriplen odd harmonics. You can see this in the spectrum in *Fig. 2*. The third harmonic is not completely cancelled, but it is a low 0.06%.

LOAD-INDUCED ALTERNATOR HARMONIC DISTORTION

Power factor is the ratio of real power (watts) to total volt-amperes (VA). VA is the product of RMS voltage times RMS current. Unity power factor results in the current being in phase with the voltage.

In rotating AC machines, the power factor for any given mechanical load is fixed and continuous. Harmonics are introduced whenever the power factor departs from unity. If the load becomes inductive (motors are the most prominent inductive loads), the phase current lags the voltage. This is called *lagging power factor* (capacitive loads produce a *leading* power factor).

Back at the alternator, the inductive phase current causes the flux between the rotor and stator, which is perpendicular to the rotor pole face with purely resistive loads, to change angle. This alters the relative orientation of the pole and slot flux lines, generating higher order harmonics. The third harmonic is introduced if magnetic loads begin to saturate, as happens when motors or transformers are first energized, or are subject to overload.

All electromagnetic devices (alternators, transformers, motors, and so on) are subject to mechanical forces from the varying magnetic flux. The alternator field is a powerful electromagnet that tries to deflect the stator core iron toward the field poles. Even the slightest relative motion between the windings or their iron cores, as well as vibration in the structure, will produce harmonics.

Under normal conditions, the force acting on the windings is much smaller than that acting on the iron core. However, if a short circuit should occur in a large alternator or transformer, the force on the coils could be extreme. I once saw the neutral-reactor coil in a 32MW gas turbine generator completely straighten itself out when a short occurred in the main generator step-up transformer. Protective relays saved the generator, turbine, and switchgear, but the neutral reactor and transformer were scrap!

High-voltage three-phase power is sent over transmission lines and through substations, whose Δ-Y transformers step the voltage back down to intermediate levels, and then on to industry and homes (see *Fig. 1* again). Industry often makes direct use of three-phase voltages between 440 and 4160V, but homes require single-phase 120V-0-120V (the "220V" service entrance line into your home).

This is provided by pole-top or pad step-down transformers, and along the way the balanced three-phase loading is locally upset, and some third harmonic is introduced. The 120V-0-120V AC line is not balanced, so differences in loading on each side of neutral can introduce even harmonics into the 60Hz waveform. As a result, the typical 60Hz AC line distortion at your home will increase to 3% to 3.5%.

POWER QUALITY PROBLEMS

The conveniences of modern society employ a large variety of nonlinear loads, where the load current is not proportional to the instantaneous line voltage. Examples are motor controllers, switching power supplies, and rectifier-filters. The filter capacitors in a rectified linear DC supply may draw current only at the peaks of the AC line voltage. Here, the instantaneous power factor varies over each half cycle of the 60Hz AC line.

These loads cause adverse effects on the line voltage that can, in turn, damage capacitors, relays, motors, and transformers exposed to the harmonic distortion. The harmonics can also cause excessive neutral currents and higher reactive currents that impose power-factor penalties, thus limiting the total output from utility alternators.

Current distortion interacts with the wiring impedance, producing voltage distortion that can interfere with the operation of other equipment. Conducted noise can enter your electrical system through the utility ground. If the ground wire contains noise or harmonics, they will travel the path of least resistance and can return to their point of origin through equipment plugged into your receptacles.

Motors are sensitive to voltage distortion. High frequency harmonic currents cause higher operating temperatures in the windings due to eddy-current losses. Harmonic voltages can produce excessive vibration, leading to abnormal bearing wear and reduced reliability. Harmonic currents in transformer windings generate excessive heat even if the transformer is not carrying its full-rated electrical load.

The normal steady-state voltage range at the "pole top" is 112 to 123V AC (224 to 246 for a nominal "220V" line). It can be much worse in remote rural areas. Most consumer electronic equipment is designed to operate between 110 to 125V AC. Power quality problems can manifest themselves in a number of ways[2].

STEADY-STATE PROBLEMS

A power failure or outage is a zero-voltage condition lasting for more than one cycle (1/60 second), while a blackout is a total power failure lasting from seconds to hours or more. A brownout is a region-wide reduction in the steady-state voltage imposed by the utility in response to high electrical consumption, such as peak air-conditioner demands in the summer. If things become really bad, utilities may be forced to use rolling blackouts to further protect the power grid from excess demand or insufficient power generation reserve.

TRANSIENT PROBLEMS

All power-line disturbances are transient by definition. A *sag* is a cycle-to-cycle decrease in the line voltage below 105V AC for 0.5 seconds (30 cycles of the 60Hz line frequency) or more, which then returns to the normal level. A *dip* is a fast sag.

A *surge* is a cycle-to-cycle increase in the line voltage above 127V AC for 0.5 seconds or more, which then returns to the normal level. If the voltage fails to return to normal, overvoltage-protection relays open the alternator or distribution circuit breakers to protect the customers.

An *impulse* is a short-term (subcycle) disturbance superimposed on the AC sine wave that typically lasts between 0.5 and 100μs. In-phase impulses ranging from 400V to 5600V (the latter from lightning strikes) that increase the instantaneous line voltage are called *spikes*. Spikes in excess of 600V are potentially very damaging, since this is the insulation voltage that residential electrical wiring and fixtures are generally designed to withstand.

Out-of-phase impulses that decrease the instantaneous voltage are called *notches*. These typically last as long as

(to page 85)

A Single-Ended 6550 Amplifier

Here's another side of the dependable, powerful, classic 6550 audio tube. **By Rick Spencer**

PHOTO 1: The finished amp.

Many articles have appeared lately regarding projects using one of the most sturdy, versatile, and trustworthy audio tubes ever made—the 6550. Most of them show these audio beauties used in the push-pull configuration in order to obtain the greatest efficiency and the most power output. In a recent article, "100W Triode Amplifier" (*GA* 3/00), Joseph Still even showed how to construct a very powerful triode amp using the 6550 to produce a total output of 100W. If you need a high-power amp, this one is great!

CLEAN SE DESIGN

You will see here that the 6550 can also be used in a clean, sweet-sounding, single-ended design. The amplifier has very few parts, is easy to build and repair, and the cost of the project is nominal when you consider the kind of sound that is produced. The transformers used in this project are available from a couple of sources; the brands of tubes you use are, of course, your choice.

I chose transformers from Hammond, which makes a great single-ended output transformer—the 1628SE—that was perfect for this amplifier. See the parts list in *Table 1*. The power transformer Hammond manufactures, the 270HX, supplies all the voltages needed for the power supply. Antique Electronic Supply (AES) stocks Hammond products, including chokes, and is very pleasant and professional to deal with.

AES is also one of the few suppliers to carry my most favorite of all the 6550s ever made—Tung Sol. This wonderful beam-pentode power-amplifier tube has a sound unmatched by all the others, regardless of their ad claims. I used these when I built the 70W Mac project that Bruce Rozenblit contributed to the *GA* 1/90 issue. I tried other brands at first, but it wasn't until I put in the Tung Sol tubes that the amp really came alive.

I've been hooked on the sound of these tubes ever since, and have searched high and low to find more of them. They cost more than the other brands, but if you can find them, please give them a try—you won't regret it. The highs are very smooth and clean, the mid-range has magic, and the bass is more solid and articulate than I've ever experienced with any other type or brand of power tube.

STRAIGHTFORWARD CIRCUIT

This article will contain no complex math or equations, but only the basic information you will need to complete the amp. The circuit is very straightforward and should not give you any problems if you follow the schematic diagram (*Fig. 1*). (Note: The diagram shows the pinout for the 6J5, not the 6SN7.)

Tube V1 is a 6J5 triode, actually one-half of a 6SN7, so if you wish to save on space and cost, you can use a 6SN7 and just wire in each triode section to the proper channel. I used the 6J5 because it improved the looks of the completed amp and because the chassis I used had plenty of spare holes. I obtained the 6J5s from AES and they are the

ABOUT THE AUTHOR

After receiving his technical training in the Air Force in the mid-1960s, Rick Spencer has worked in the air conditioning, heating, and refrigeration systems field ever since. He has built over 30 audio components in the last 41 years and is currently working on at least two amplifiers that will be low in cost but high in sound quality (and they are geared toward the novice who is looking to get started in this wonderful hobby).

TABLE 1
PARTS LIST

REFERENCE	PART
B1	50V/25A
C1, C4	100µF
C2, C5	100N
C3	220N
C6, C10, C12	330N
C7, C11	250µF
C8, C9	900µF
C13	100m
C14, C15, C16	10N/1kV
D1, D2	1N4004
F1	3A
F2	0.5A
J1	AC IN
K1	RCA Jack
L1	5H, 200mA
RE1	Relay SPST
R1	100k
R2	6k8
R3	470k
R4	20k, 1W
R5, R10, R11	270k
R6	1k
R7	330, 25W
R8	15k
R9	100
R12	330k/2W
R13	1R/25W
S1, S3	SPST
S2	SPST/10A
T1	Hammond 270HX
T2	Hammond 1628SE
V1	6J5 or ½ 6SN7
V2	6550

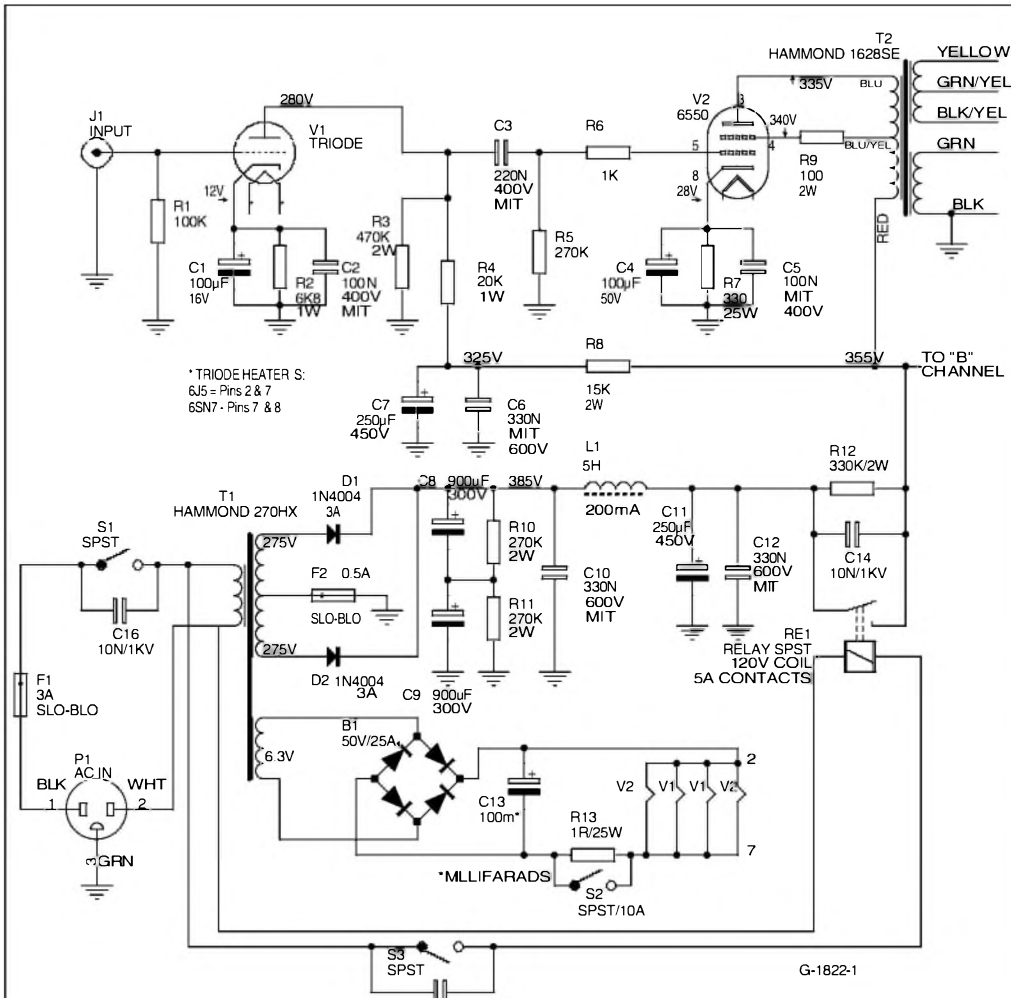

FIGURE 1: Amplifier schematic.

Raytheon brand, a beautiful tube with one triode section standing alone in the center of the glass envelope.

I also purchased the Tung Sol 6550s from AES. According to the factory specs, the 6550 in single-ended mode is capable of a fairly good power output. The spec sheet shows almost 20W of Class A1 output for a single tube, with a plate voltage of 400V and G2 voltage at 225V. The output on this amp is probably a little lower, but it certainly doesn't sound like it. When it's driven by my VTL PR-1 preamp, I can't move the volume control past the 10 o'clock position without being driven from the room!

Hammond includes a 40% tap in the primary winding of its 1628SE transformer. This is left unused if you wish to try the amp in the triode mode, with a 120Ω 5W resistor tying G2 to the plate, and with some power reduction, of course. I think the amp sounds great either way. I used Holco resistors in the amp circuit. The power supply uses metal-film, and the large power resistors are wirewound types from NTE. The coupling capacitors are from MIT, and all the power-supply and cathode caps are bypassed with small-value MITs.

BYPASS CAPS WORK!

I don't like using electrolytics when they can affect the signal, so I took Eric Barbour's advice (*GA* 1/93) and used the bypass caps to help smooth out the high frequency. It works! Because this circuit is so simple, you can experiment with it all you like. You can try a differ-

CONSTRUCTION HINTS

If you love this hobby as much as I do, then whenever you see a new and different diagram or picture of a piece of audio equipment, you probably can't help but wonder what it would be like to build your own version of it and see just how well it will work and how good it might sound. In total, I've built over 30 projects consisting of preamps, amps, and speakers, and I still feel the same excitement every time I start something new. That's how I felt when I decided to build something new and different with the 6550s I had on hand.

The chassis I chose had what seemed to be a vast expanse of open space for just four tubes, but part of the thrill of being a do-it-yourselfer is working out your own topology and circuit layout and being able to put things exactly where you wish them to be. The chassis, all 18″ of it, had lots of holes to fill, so instead of installing the filter capacitors inside, I opted for above-the-top-plate mounting. With only the two 6J5s and the two 6550s on such a wide platform, the caps filled in the holes and the extra space quite nicely.

If I ever need to replace or upgrade a capacitor in the future, it will be an easy task. These capacitors are mounted on regular project breadboard that is attached to the bottom of the chassis plate. To make it easier on the old memory cells, I painted the bottom of the plate with flat white enamel and then labeled the terminal strips and other wiring points with a marker pen. This is a big time-saver, especially if I must stop the project for a few days, and it also helps for any future service or repair needs.

By the way, I used the two 900µF caps in series because I had them in stock. You can use any good-quality capacitor here that is rated at 450V. Just be sure to have adequate capacitance, and your power supply will probably be really quiet like this one. Also try to use good-quality wire when building this or any other project, and use good soldering technique. I find that type THHN and THW copper wire, which is rated for 600V, is fine for the power supply and heater circuits and is available at almost any hardware store.

For the signal circuit, I try to use Cardas or other Teflon® copper or silver wire, but I really can't tell the difference in sound between one good-quality wire and the others. I truly believe that good sonic performance comes from the resistors, capacitors, transformers, and tubes you use.

OTHER HANDY TIPS

The schematic in *Fig. 1* should be fairly easy to follow. On the diagram you will notice in the power supply that the 6.3V DC wiring for the tube heaters is connected to allow for equal voltage at all the tubes, instead of having one of them with a slightly lower voltage at the end of the usual wiring "chain." This technique has always worked well for me, especially when using tubes that pull a lot of amps on their heaters or filaments.

You will also see that the B+ switching relay has a resistor and capacitor across the points. This is to prevent point-flash and to help eliminate the "pop" or "thump" that is always annoying in your speakers when the B+ hits the tubes after the cathodes are warmed up. Please try to power up any new project with a Variac® and an amp meter, also available from AES.

I've found a digital wattmeter is also great for monitoring the power draw when ramping up the AC voltage. Always try to keep a voltmeter on the 6.3V and the HV in case of any low readings or any overshoot. These readings, when done without the tubes in place, will always be a little high, of course; and don't forget to safely discharge the filter caps, thus preventing those unpleasant surprises we have all experienced!

I always test the tubes beforehand, even new ones, to make sure they are up to specs. If you finish your project and power up with a bad tube in place, it may send you searching through the entire circuit you had already tested and were pretty sure was OK up to this point. It is also a good idea to check any resistors and capacitors you will be using, to ensure you have a good match in values, and to catch any problems in advance.

This amplifier seemed to require about 25 hours of break-in time, so be patient and allow everything to settle in. What's the best way to break in your new amplifier? Why, listen to your favorite music, of course! Factory units always need time to break in, and our homemade units are certainly no exception.

ent power supply, transformers, resistors, caps, and any brand of 6550 or KT-88 tube.

For the rectifiers in the B+, I used half of a full-wave bridge I had in stock, rated at 800V and 5A. This made for easy mounting of the rectifiers because this part has its own case, which is also a heatsink. Switch S2 is for soft-starting the heaters, and S3 is to delay the B+ until the cathodes are warmed up to prevent stripping.

You can build the amp without these switches, but I'm going to do all I can to prolong the life of these precious old tubes. The completed amp is quite heavy, so I assembled it with all the parts except for the output transformers, saving them for last. This made moving the chassis around a lot easier and safer. These transformers have their core plates turned at 90° angles to each other.

This amp, like most SE amps, is speaker-sensitive. You should try some different combinations to achieve the results you desire. I found that Focals, at 92dB efficiency, sounded very good, but some Polk speakers, at 90dB, were more open and airy. Over the years, after using many brands and types of speakers, I have found that Legacy produces what I consider to be the king of full-range high-efficiency speakers. I have their Focus model, which sports a 96dB efficiency with a response of 16Hz to 30kHz!

These speakers, when used with tube amplifiers, have the best midrange sound of all that I've heard in my 41 years of working with audio systems. They are currently hooked to the Mac 70s, and the amps drive them without even the slightest indication of being overworked. I have even driven them with an old Harman Kardon push-pull amp using 6V6s and rated at only 12 to 14W.

NATURAL SOUND

The sound of this amplifier is very full and natural, and the way it treats the female voice is amazing. On Jennifer Warnes' album, *The Hunter*, track 2, I felt as though she were almost in the room with me. Enya's delicate voice has never sounded so beautiful. The same goes for Sade on her album, *Stronger Than Pride*, track 9, which has a soft and easy sound as she "sighs" the words.

If you have any friends who think that tubes can't handle bass as well as solid state, just play the album *The Misty Mood of Three Blind Mice*, track 5, and watch their expressions change. This track, and track 8, have over twelve minutes of the best recording I've ever heard of string bass notes. (And, yes, if you notice the bass starting in the left channel only, you're hearing correctly.)

This amplifier has been a great "test bed" for the 6550s that are currently available. With only one tube to change for each sampling and no bias adjustment to worry about, it was very easy to compare tubes. GE's own 6550A was really good, with great highs and a very open midrange. It would be my second choice after the Tung Sol. The Svetlanas also sound good, have a lot of midbass punch, and are very good from sample to sample.

I'm sure that most readers are aware of the possible higher distortion levels in this SE design versus some PP designs, but that's not the end for which I build. I build and modify my circuits to suit the ear, not the meter. I'll "tune" an amp until the sound is pleasing and just right. Using the components I've listed has given me that sound. If you use good construction techniques and good parts, you'll have the same results. ❖

How Good Is Your Port?

This author's extensive look at the problem of port nonlinearity will help designers determine correct port size. **By Bohdan Raczynski**

Ported enclosures are known to extend the low-frequency output of loudspeakers by exploiting the Helmholtz resonator effect produced by the compliance of the air inside the enclosure and inertance of the air in the port. An acoustic transformer produced this way has its own resonant frequency, f_B, at which most (or all, if there were no losses) of the system acoustic output comes from the port.

A fairly obvious implication of this is the significant velocity associated with the air flow through the port. This, in turn, causes all sorts of nonmusical noises to be generated by the port, as well as distortion and acoustic compression. Depending on port geometry and required low-frequency SPL of the system, the issue can be quite significant.

The problem just described belongs to a rather complex field of fluid flow theory. Assuming incompressible flow, some sophisticated FEM programs would be able to model air turbulence and associated vortex shedding in more detail[1], but this is well beyond the scope of this article.

Fortunately, existing research results, design material, and test results enable us to formulate an approximate macro view of the problem and look at the acoustic impedance of the port under high air velocity. I would like to stress that the approach I am taking here is a significant simplification of the physics of the problem. However, the resulting model is quite useful and is confirmed in practical tests[2].

INTUITIVE APPROACH TO PORT NONLINEARITY

You are no doubt familiar with the need for good carpentry skills when building speaker boxes. Accuracy of joints and sealing the box is essential for proper operation of all types of boxes, be they sealed, vented, passive radiators, and so on.

A sealed box means exactly that—the air inside the box is trapped and sealed from the external world. There is no parasitic or accidental leakage from the box, so that the mathematical model developed for the enclosure continues to be accurate. Consequently, the box Q_B factor is controlled by the designer and not by sloppy workmanship.

A vented enclosure also needs to be "air tight"; of course, with the exception of the purposefully introduced opening port. Just as for the sealed box, the cabinet needs to be sealed, so that no accidental air leakage from the box can occur. A properly executed design would include all sorts of seals and gaskets to ensure that connector boards or drivers themselves do not cause air leakage.

Assuming you have built your perfect vented box, you may expect that the SPL curve and input impedance curve will look as in *Fig. 1*. The impedance curve is very familiar and has two characteristic peaks with the dip between them. The dip is located exactly on the box tuning frequency, $f_B = 25Hz$.

My design is a QB3 type with system parameters as shown on *Fig. 1*, and I have also assumed that $Q_B = Q_P = 1000$, so I have a very low-loss design. Port diameter in this example is 15cm (6″) and the input power to the system is 1W.

Now, assume that I start reducing port diameter. Initially, the picture will not change much. However, when the port eventually becomes very small, intuitively there should be very little difference between the vented box with a "very small port" and the sealed box with a "large leakage" problem[3].

In this case, you would expect the SPL curve will resemble that of a leaky sealed box and the input impedance

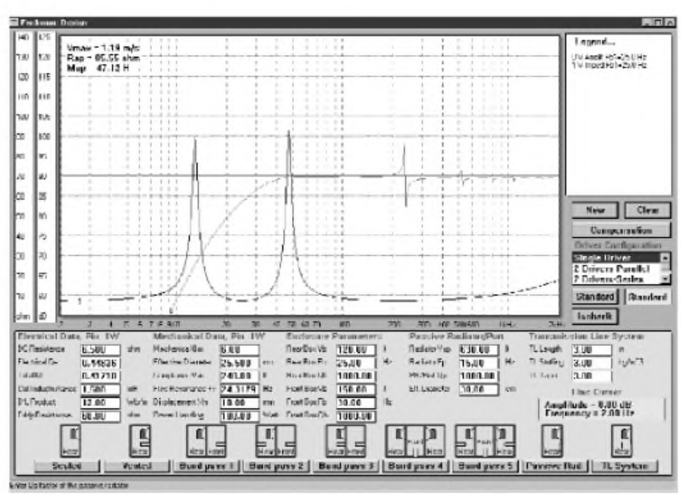

FIGURE 1: SPL and input impedance at 1W, port diameter = 15cm.

ABOUT THE AUTHOR

Bodhan Raczynski has master's degree in Acoustics and is currently a director of a small software company, "Bodzio Software," in Melbourne, Australia. Born in Poland, he developed a passion for audio and electronics in his teens. After graduating from Polytechnik of Gdansk in Poland, he designed a number of valve and solid-state power and high power amplifiers.

He moved to Australia and in 1990, developed his first commercial program, Speaker System Designer (SSD), which became available first in Australia and then US and Europe. In mid '90, he replaced SSD with the now well-recognized SoundEasy. Bohdan lives in the eastern suburbs of Melbourne with his wife and three children.

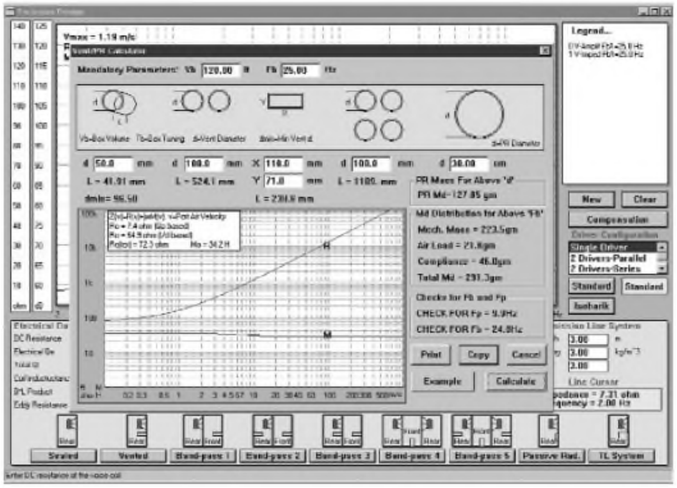

FIGURE 2: Acoustic impedance R+jωM of a short port.

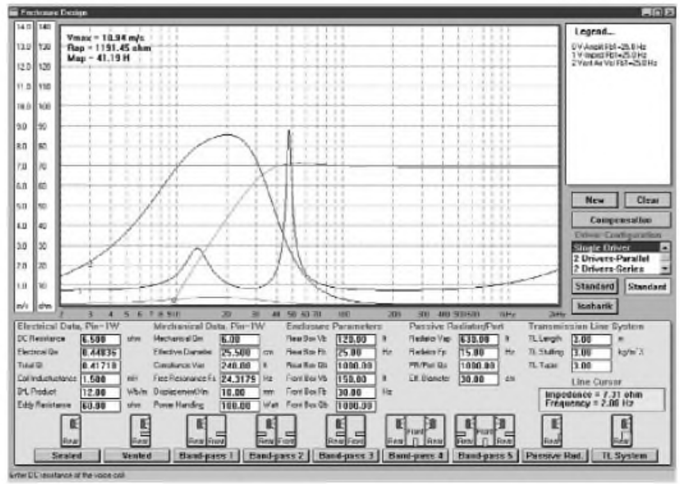

FIGURE 3: Port air velocity (curve 1) for port diameter = 5cm.

curve will lose the lower peak and become a "single peak" curve just like those of the sealed boxes. Between those two extremes, you may expect problems commonly known as "port nonlinearity."

HISTORICAL RESEARCH ON FLUIDS IN TUBES

Bies and Wilson[4] actually experimented with an orifice (more like a decent port) 9cm (3.5″) in diameter and 2.9cm (1.13″) long. They found that the acoustic resistance, R, of the tube varies with the particle velocity in a similar way as shown in *Fig. 2*. Thurston[5], working with fluid flow through circular tubes, found that analogous acoustic resistance, R, and inductance, M, again vary in a similar way as shown in *Fig. 2*.

A landmark paper by Ingard[6] offers an empirical formula for the nonlinear component of the acoustic resistance of a tube. Later on, this work was expanded by Ingard and Ising[7], who offered more complete mathematical treatment of orifice behavior.

Backman[2], experimenting with port nonlinearity, plotted drivers' input impedance for various input voltages and determined that the most sensitive to the amplitude variations is the magni-

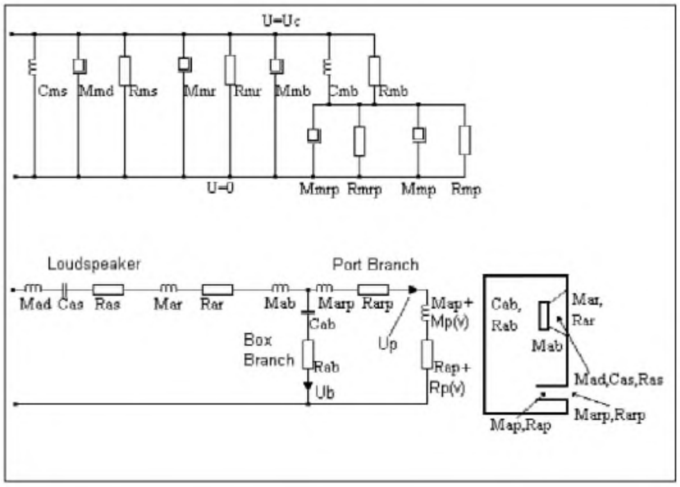

FIGURE 4: Modified model of the vented box including Zp(v) = RP(v)+jωMP(v).

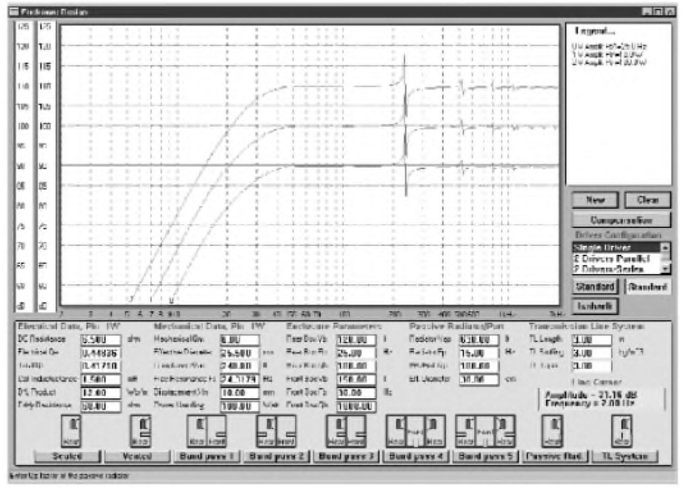

FIGURE 6: SPL for port diameter of 15cm at 1W, 10W, and 100W input power.

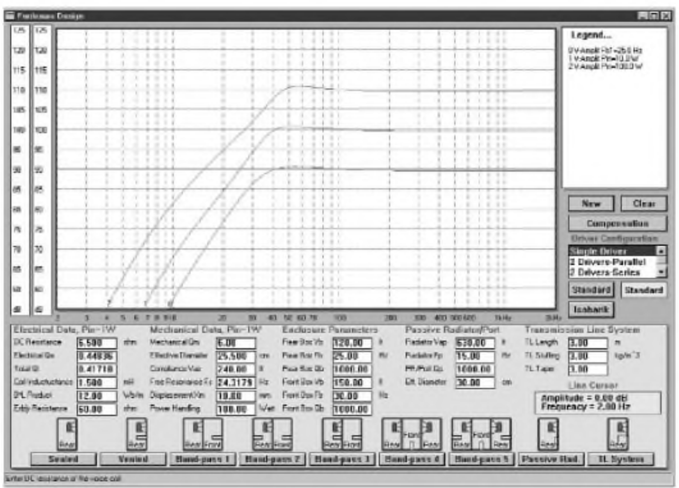

FIGURE 5: SPL for port diameter of 5cm at 1W, 10W, and 100W input power.

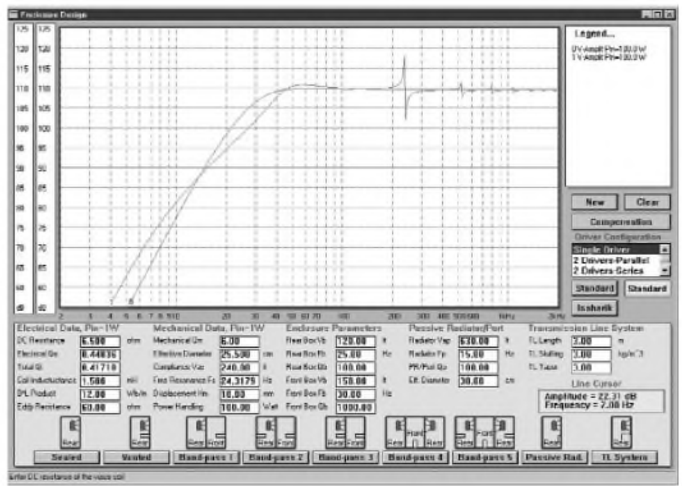

FIGURE 7: SPL for port diameter of 5cm (curve 0) and 15cm (curve 1) at 100W.

tude of lower impedance peak. He pointed out that the behavior of the vented enclosure approaches a sealed enclosure at high levels. The inner diameter of the port was 102mm (≈4″), and the length of the port was 168mm (≈6.6″).

Vanderkooy[8] measured and analyzed port performance at high levels and proposed a simplified nonlinear model of a port. The comprehensive analysis included jet formation, acoustic compression, turbulence noise, and distortion.

SIMPLIFIED VIEW OF THE PROBLEM

Typically, the area of the cone is many times larger than the area of the port. Since the volume of the air pushed via the port needs to be approximately equal to the volume displacement of the cone, the only way to accomplish this is to push the air in the port much faster. At low SPL levels, the volume displacement of the cone is small, and, consequently, the cone velocity is low as well. This, in turn, stipulates low air velocity in the port, or what is known as a laminar air flow.

The process is characterized by a low Reynolds number, Re<2000, and there is no turbulence in the air flow.

$$Re = (r \times V_{max} \times \rho)/\mu$$

where V_{max} is the peak flow rate, ρ is the density, μ is the air viscosity, and r is the port radius. At this point, the acoustic resistance of the port, R_P (port loss), is linear and is relatively low (50–200Ω).

Also, the mass of air in the port, M_P, is constant. Remember that mass of the air in the port resonates at f_B (box tuning frequency) with the compliance of the air inside the box halting the movement of the cone. If there were no losses, at resonance, f_B, all acoustic output of the system would come from the port.

Now, start reducing the port diameter. As you can guess, the volume displacement of the cone remains the same, so the air in the port must move faster and faster. Soon, a turbulence occurs in the air flow, jets of air are forming, and the process is no longer linear.

The Reynolds number corresponding to the onset of turbulence is Re ≈ 20000. When the Reynolds number hits 50000, your vent is compressing. It is now more and more difficult to push the air in the vent.

The acoustic output of the port is below the expected level (if the process continues to be linear), and you begin to experience a "port compression" effect. Port impedance, Zp, now has an

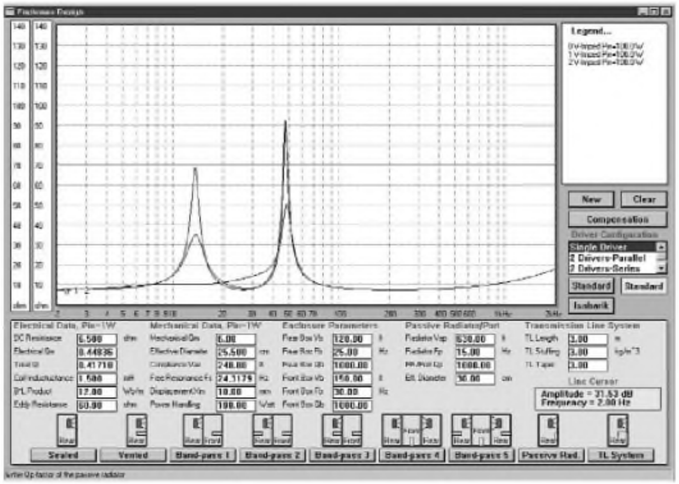

FIGURE 8: Impedance at 100W for port diameter of 15cm, 10cm, and 5cm.

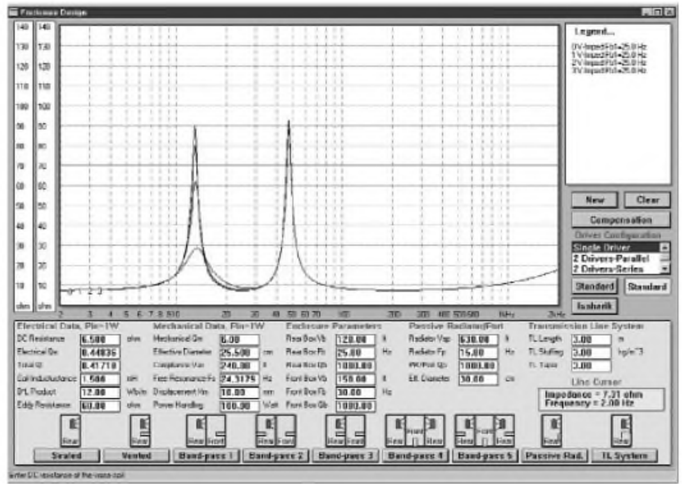

FIGURE 9: Input impedance at 1W for port diameter of 15cm, 10cm, 7.5cm, and 5cm.

additional component relating to turbulent air flow and depending on the air velocity, $R_P(v)$. Also, the M_P has changed as well, drifting towards 60–70% of its original value and becoming velocity dependent.

The $R_P(v)$ is an interesting element. If plotted in the frequency domain, it would resemble the curve depicting port air velocity (*Fig. 3*). This should be no surprise, as $R_P(v)$ is so heavily dependent on the port air velocity. In the frequency domain, the $R_P(v)$ will quickly reach its peak not far from the lower peak of the input impedance curve. Depending on the geometry of the port and the volume displacement of the driver, the air velocity in the port may reach 50–100m/s. Such a high value of air flow results in $R_P(v)$ reaching 5–10kΩ levels.

The "double-peak" impedance curve is a clear result of the port action. The port air velocity curve plotted on *Fig. 3* clearly indicates that the lower impedance peak will be much more affected by the port nonlinearity than the upper impedance peak. Therefore, the $R_P(v)$ needs to be incorporated in the driver's model.

CHANGES IN DRIVER'S MODEL

The vented enclosure shown in *Fig. 4* provides different loading for the back of the diaphragm, as compared to the sealed box. The vibrating system and front loading of the diaphragm are represented on the mechanical mobility circuit the same way as for the sealed enclosure.

Introduction of the vent adds several more components such as: (1) mass of the air in the port (Mmp) and its losses (Rmp), and (2) radiation impedance of the port represented by Rmrp and Mmrp. The air in the port is treated as a mass because of its small volume and, more important, because it is incompressible. Particles of air will move on both sides of the vent with the same velocity. The air compressed in the box by the back side of the diaphragm has only one path for escape—pushing the air mass through the vent. Therefore, the pressure path consists of the series connection of Cmb, representing compliant air in the box and the four elements of the port.

Since the air in the port is incompressible, the immediate layer of air in front of the box (radiation impedance) will be connected to the same velocity line as the entry to the port inside the box. The other end of the masses is connected to the U = 0, or reference velocity as required in mechanical mobility circuits.

The mechanics of the preceding process can be easily demonstrated on a physical model of a vented box. Connecting a small (1.5V) battery across vented box terminals, you can displace the cone in or out of the box. A small air-flow detecting device (candle) positioned in front of the port will show significant air movements in the direction opposite to the diaphragm. The volume of air displaced by the cone should be similar to the volume of air leaving the port. If the difference is significant, then leakage losses are significant.

This experiment clearly shows the pressure (current) path in the mechanical mobility model, so it should now be easy to explain why the compliance of the box is connected in-series with port elements. It is observable that Cmb and Mmp+Mmrp form a series-resonant circuit in the mechanical mobility representation. The circuit will act as a "selective short circuit" for the volume velocity Uc, shorting it to U = 0 (ground) at the circuit resonant frequency. Because of the circuit losses, the short is not perfect, but velocity Uc will be much reduced. In the practical system this situation translates into much reduced cone excursion at the box resonant frequency.

Acoustical impedance representation shows C_{AS} and Map+Marp forming a parallel resonant circuit. Electrical circuit theory advocates that very little energy (current) needs to be fed into the circuit for it to resonate and for the current (volume velocity) in the resonant circuit to be still very high.

Therefore, volume velocity in the "feeding" branch, which contains diaphragm output, will be very small, and volume velocity in the resonant circuit containing port will be high. This effect, although the strongest on the resonant frequency f_B, will extend over some narrow frequency range, and on the low-end side produces extended system output. The enclosure/port resonance effect is being exploited here to augment system output at low frequency. *Figure 4* shows mechanical mobility (top diagram) and acoustical impedance (bottom diagram) representation adopted for the vented enclosure model. The components are:

- C_{AS}, equivalent compliance volume V_{AS} transformed to acoustical side.
- Mad, mass of the vibrating system Mms transformed to acoustical side.
- Ras, vibrating assembly loss Rms transformed to acoustical side.
- Mar+Mab, air radiation of the front side of the diaphragm and air load of the back side of the diaphragm.
- Rar, air radiation of the front side of the diaphragm.
- C_{AB}, enclosure compliance V_{AB} transformed to acoustical side.
- Rab, absorption losses of the enclosure transformed to acoustical side.
- Marp, Rarp port radiation.
- Map, mass of the air in the port.
- Rap, frictional losses in the port.

The nonlinear port impedance was implemented as $Zp(v) = R_P(v) + j\omega M_P(v)$

in the port branch. Please note that $Map+M_P(v)$ will exhibit a slight reduction in value as the air speed increases, and $Rap+R_P(v)$ will exhibit significant increase in value as the air velocity in the port increases.

RESULTING PERFORMANCE

In order to gain some insight into system performance affected by port nonlinearity problems, I plotted the SPL for a port of 5cm in diameter for 1W (curve 0), 10W (curve 1), and 100W (curve 2) input power (*Fig. 5*). As you can see, with the increased input power, there is a sort of "saddle" developing on the SPL curve around the box tuning frequency of 25-30Hz. This is exactly the frequency range where you would expect the port to contribute most to the system SPL. Our small port is clearly not performing as anticipated.

Next, I modeled SPL for the same power levels, but this time I used a larger port, 15cm in diameter. The resulting plots in *Fig. 6* do not exhibit the "saddle" any more for the 100W power level. Clearly, as the input power is increased, the SPL curves go up, maintaining the approximate shape acquired at 1W power level.

It is also easy to observe that the SPL curves now have resonant peaks above 200Hz, not seen in *Fig. 5*. This is the result of enlarging the diameter of the port. You may remember that larger port must also be longer if tuned to the same frequency. The length of the port is such that the self-resonances of the port tube fall into a much lower frequency range—just to be displayed on the screen.

In order to estimate "port compression" effect, I plotted the SPL of the two ports on the same screen (*Fig. 7*). Visual inspection of the graphs shows about 5dB of port compression at f_B = 25Hz. I have just about lost my vented box performance gains if I were to use the smaller port.

In the next step, it would be interesting to compare the input impedance plots to see whether the lower impedance peak, characteristic for the vented enclosure, indeed disappeared from the plots. The answer is clearly evident in *Fig. 8*, where the input impedance for three ports is plotted at 100W power level. The 15cm vent exhibits the expected "double peaks" curve; however, the compression is still registered on the impedance plot. Ideally, the port should still be larger.

The Reynolds number calculated for these conditions is Re = 68000. Therefore, the port is indeed compressing slightly. However, the 10cm port has the lower impedance peak significantly reduced. This is a sure sign that this port is too small for the job.

The severely undersized 5cm port produces a "single peak" impedance curve for 100W input power. This port would also prove to be inadequate for 1W of input power. The Reynolds number calculated for 1W conditions is Re = 20000, which is a clear indication that the port becomes turbulent. Indeed, the corresponding input impedance plot shown in *Fig. 9* is a clear indication of the nonlinearity problem at 1W for this port.

REMEDIES

First and foremost, the problem is related to air velocity in the port. As we know[9], air velocity in the port, Vp, depends on volume velocity Up via the port branch divided by the port's area, Sp.

$$Vp = Up/Sp$$

Since the area $Sp = \pi r^2$, it is easy to observe that air velocity in the port depends on the inverse squared of the port's radius. It seems rather obvious that port radius should be kept as large as possible.

Dickason[10] recommends 15cm (6″) ports for 15″ woofers as a minimum and 10cm (4″) ports as a minimum for 12″ woofers.

Salvatti[3] offers a number of recommendations relating to port geometry, including balancing inlet and exit flows by using different tapers. Roozen[1] advocates port geometry based on 6° diverging contour towards the ends of the port. The inlet and outlet are rounded with a relatively small curvature.

CONCLUSIONS

I have described a simple modification of the acoustic impedance of the port in the vented enclosure. The modification consists of adding extra components

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such as $M_P(v)$ and $R_P(v)$, which are dependent on air velocity inside the port. The modification reflects acoustic compression of the port and shows the changes to the input impedance of the driver under high SPL levels. The changes in input impedance reflect experimental findings[2].

Looking at the performance of the 15cm (6″) vent, I was surprised to see the compression evident at 100W input power. The vent seems quite large, and to tune it to 25Hz in a 120 ltr box, I would need to make it nearly 60cm (23.6″) long. This could be a construction challenge, and besides, this vent is still not completely linear at high SPL.

This brief analysis of the vent performance clearly indicates that nearly all practical size vents will compress at higher SPL levels. Loudspeakers intended for high power stage applications should give very careful consideration regarding port nonlinearity. For this type of application, the port compression problem will tend to further degrade system SPL, already affected by the thermal compression during prolonged stage performance. ❖

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The Ultra Fidelity Computer Sound System, Part 1

This five-part series considers the unique computer acoustic, including the design of high-quality speakers, subwoofers, and amplifiers best suited to the computer environment. **By R.K. Stonjek**

Multimedia computers first came to the public's attention in the early 1980s. They started with squeaking noises from a PC speaker, which was fine for audio cues and alarms. Sound cards soon followed with speakers placed either side of the computer. They, too, sounded "plastic" and "squeaky."

EVOLUTION OF COMPUTER SOUND

OEM speakers have changed little since then, but sound cards have come a long way. The better ones have a dynamic range of over 90dB. Optical output and four-channel sound are becoming popular.

The 1990s provided something to listen to. Games took on a fully spatial, multidimensional, and multichannel sound. Music creation and manipulation software became cheap enough for everyone to use. The MP3 format allowed you to share music files over the web. By the end of the 1990s, the DVD format featuring crystal-clear audio and video became widely available.

The modern computer's speed, memory size, hard disk space, and monitor resolution have all leaped forward over the past few years, but computer speakers have, in general, just become cheaper—both in price and quality. For under a hundred dollars you can get a powered sub and satellite system made of and sounding like so much cheap plastic. Of the better-quality computer sound systems, the majority seem to be derived from hi-fi speakers that are simply shoehorned into the computer environment.

With the advent of inexpensive computer DVD drives, advanced sound cards, readily available MP3 music on the Internet, and games with unbelievable sound effects, the time is more than ripe for a computer sound system of a quality comparable to the rest of the modern computer.

The Ultra Fidelity Computer Sound System is a high-end computer sound system boasting low distortion and coloration, flat frequency response, high output (almost 110dB), and an extremely clear and precise full-range sound. Its sound quality is comparable to that of the picture quality of the modern 17″ LCD monitor.

By analyzing the computer environment, you can optimize the design, making it simple, effective, and reasonably inexpensive. The Ultra Fidelity Computer Sound System has the additional advantage of being easy enough for the average constructor to build over a couple of weekends and costs only around 600 Australian dollars to build (one US dollar = two Australian dollars). It even works with IMac computers.

BASIC COMPUTER ACOUSTICS

First, consider three listening environments—a PA installation, a home hi-fi, and a computer sound system. Typically, you sit between 0.5 and 0.75m from the computer, 3.5 to 4.5m from the hi-fi, and 10 to 20m from the stage at a concert (*Table 1*, row 1). The most striking thing about these three very different environments is the required acoustic output.

Row 2 in *Table 1* lists the sound pressure level (SPL) at the listener position for 1W input and a speaker with an efficiency of 90dB at 1m for 1W. The computer already has a healthy output high enough for most listening requirements. The hi-fi, between 77 and 79dB, is a little low but suitable for background music—at dinner, for instance.

Row 3 examines how much power is required to get 100dB at the listening position using a speaker with 80dB efficiency (at 1W, 1m). The computer environment can manage this with a regular hi-fi power amplifier. The hi-fi environment requires a PA amplifier, and the PA environment requires tens of thousands of watts.

Rows 4 and 5 include more efficient speakers, 90 and 100dB, respectively. You can now see a pattern forming. The standard power rating for a regular PA amp is 1,000W, with speaker efficiency typically between 95 and 100dB. The hi-fi amplifier is typically 100 to 150W, with speaker efficiency between 85 and 90dB. If you follow this pattern to the computer speaker, amplifiers will be around 20–30W and speaker efficiency

PHOTO 1: Round sub/IMac computer.

		Computer		HiFi		P.A.		
1	Typical listening Distance	0.5	0.75	3.5	4.5	10	20	m
2	Sound Pressure at listener for 1w and 90dB speaker	96	92.5	79	77	70	64	dB
3	Power for 100dB at listener using an 80dB Speaker	25	56.2	1,259	1,995	10,000	39,810	W
4	Power for 100dB at listener using an 90dB Speaker	2.5	5.6	126	199.5	1,000	3,910	W
5	Power for 100dB at listener using an 100dB Speaker	0.25	0.56	12.6	19.9	100	391	W
6	Power required for 100dB from 82.5dB Computer, 87.5dB HiFi, & 97.5dB P.A. speakers, all in Stereo	3.5 17.5	8.0 39.8	54.9 36.6	89.1 59.4	44.7 4.5	177.8 17.8	W %
7	Maximum output with efficiency as above and using 20w Computer, 150w HiFi and 1,000w P.A. Amps	107.5	104.0	104.4	102.3	113.5	107.5	dB

TABLE 1: The three listening environments considered.

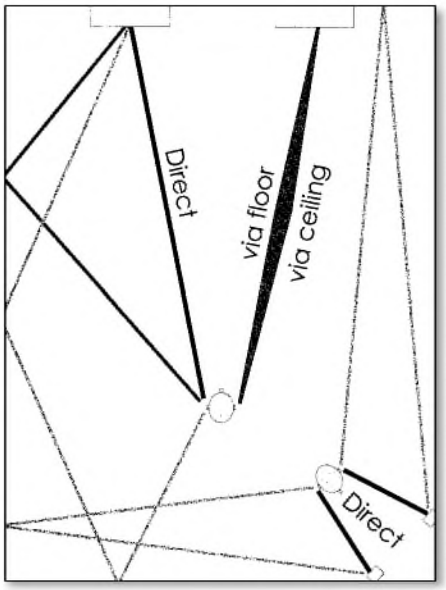


FIGURE 1: Direct and reflected sound paths to listener.

will be around 80–85dB.

Row 6 tests this theory by investigating how much power is required to achieve 100dB from a pair of 82.5dB computer speakers, 87.5dB hi-fi speakers, and 97.5dB PA speakers. The percentage figure below the power reading indicates the percentage of power used for a 20W stereo computer amp, 150W hi-fi, and 1,000W PA amplifiers. Notice how well the computer at 0.5m compares to the PA at 20m.

Row 7 shows the expected maximum output in dB at the listening position for a typical hi-fi amp and speakers, as well as a PA installation and the theoretical computer speakers using typical amplifiers. The computer compares well with the hi-fi, having an extra 3dB at the closer listening distance. The PA is a solid setup for most smaller venues such as pubs, clubs, and smaller halls. Typically, however, you can wheel out extra speakers and accompanying amplifiers as needed.

The home hi-fi format originally borrowed much of its speaker technology from theater speakers. There is plenty of space behind the screen in a theater, so no serious effort was ever made to compact the speakers. Hi-fi speakers were equally enormous. It took companies such as Acoustic Research in the 1950s to think "hi-fi" and develop products designed for the actual listening environment in which the speakers were to be used.

This does not seem to have happened in the computer speaker market. There are plenty of speakers designed to match the computer environment aesthetically, but not acoustically. Powered computer speakers typically have amplifiers that produce less than one genuine watt, often much less.

COMPUTER VERSUS HI-FI— THE ACOUSTICS

The following comparisons concentrate only on the computer and hi-fi environments. *Figure 1* shows the direct and reflected paths for the sound reaching the listener. In the hi-fi environment reflected paths to the listener have such shallow angles, such as via the floor or adjacent walls, that as much as half or more of the sound arriving at the listener has been reflected off at least one surface.

For the computer speaker listener, the situation is quite different. The shortest reflected path is via the floor. For a listener at 0.5m and a speaker at 1m above the floor, the angle to the floor is around 86°. Furthermore, the path is 2m long, four times the direct path.

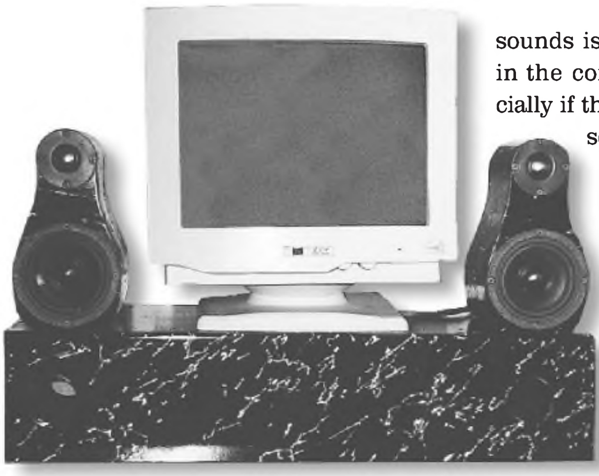


PHOTO 2: Square sub/14″ monitor.

This means a further 12dB attenuation for the reflected path. The overall attenuation is more than 30dB (steep angle plus distance), which means that reflected sound forms almost none of the overall sound (<5%). If you have ever listened to your hi-fi speakers in the open air away from walls, then you know the potential the computer speaker has.

CHANNEL SEPARATION

A further consideration is channel separation. With so much reflected sound, the channel separation of the hi-fi system is far from optimal. *Figure 2* shows why the computer environment approaches that of headphones.

To pick the position of an object from audio cues alone, you must process the relative phase and loudness of the sound that reaches each ear. With so much reflected sound, this is vague for the hi-fi, though vastly improved if speakers are well off the floor. In the example given, the difference in sound level from, say, the left speaker as heard by the right and left ears is just 1.2dB for the computer, but it's impossible to detect 0.12dB for the hi-fi.

The angle of the right speaker to the left ear for the computer is steeper as well: 36.1° as opposed to 21.5° for the hi-fi. Considerably greater rolloff of the higher frequencies from the opposite speaker in the computer environment will occur. From the left ear to the left speaker the computer wins again; 18.3° as opposed to 21.0° means slightly less attenuation of the desired sound.

This means that channel separation and therefore your ability to locate sounds is quite extraordinarily good in the computer environment, especially if the treble comes from a point source (small tweeter dome). Locating the position of players in an orchestra, objects in a computer game, or the action on a DVD movie can be chillingly accurate.

The down side is that these steep angles mean you can't move your head without affecting the location of sonic objects. You also tend to be far more aware of phase differences between midrange and treble speakers, and the delay between bass and subbass. You can address the midrange-treble problem with active crossovers and phase-aligned drivers.

SUBWOOFER PROBLEMS

Current computer sound systems featuring subwoofers all seem to have cube-shaped subwoofer cabinets. There are two problems with these boxes.

First, it is physically impossible to place these large boxes as close to the listener as the main speakers. You can place the subwoofer either behind the monitor or, more commonly, near the listener's feet. On the floor it is at least 1.5m, three times as far from the listener's ears as the main speakers.

Attenuation due to the extra distance is 9.5dB, requiring eight times the power than if level with the main speakers. Placed behind the monitor, the subwoofer is still twice the distance and so requires an additional 6dB, or four times the power, just to keep up.

This is a lose-lose situation—the woofer must be bigger to take all the extra power it needs. Being bigger it must be placed even further away. The direct/reflected ratio changes from almost all direct for the satellites to almost all reflected for the sub.

This problem never occurs with hi-fi. It is usually impossible to place the sub-

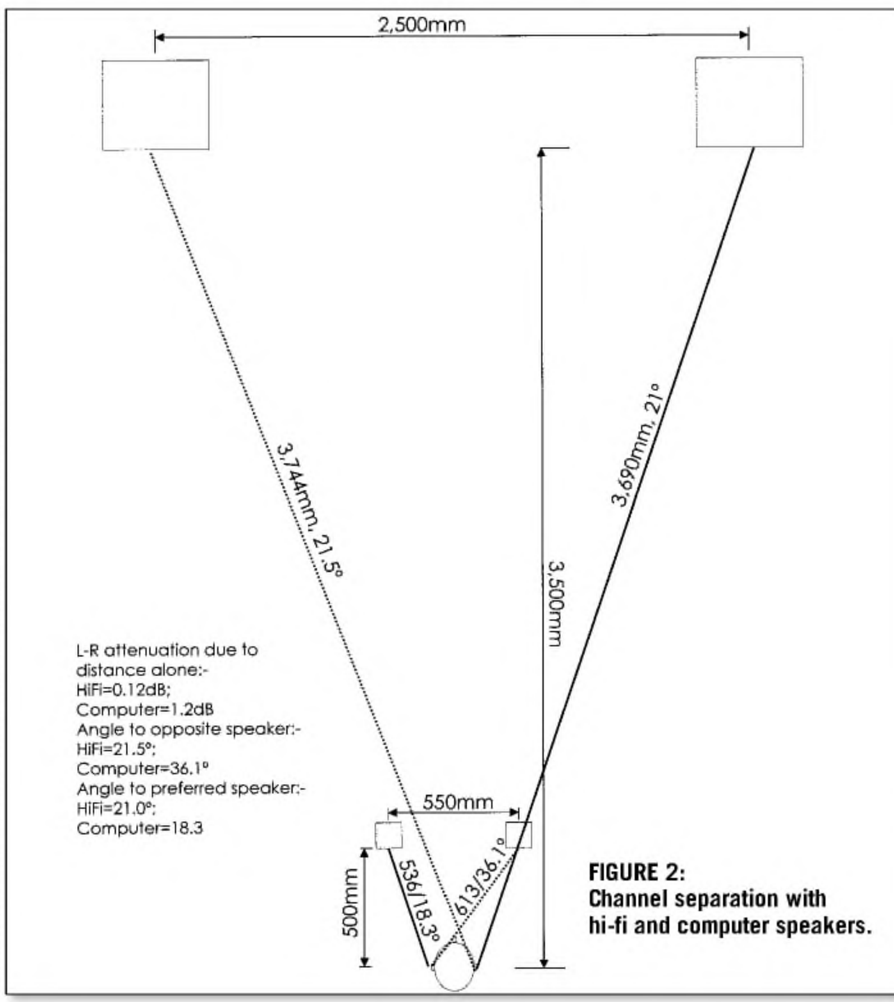


FIGURE 2: Channel separation with hi-fi and computer speakers.

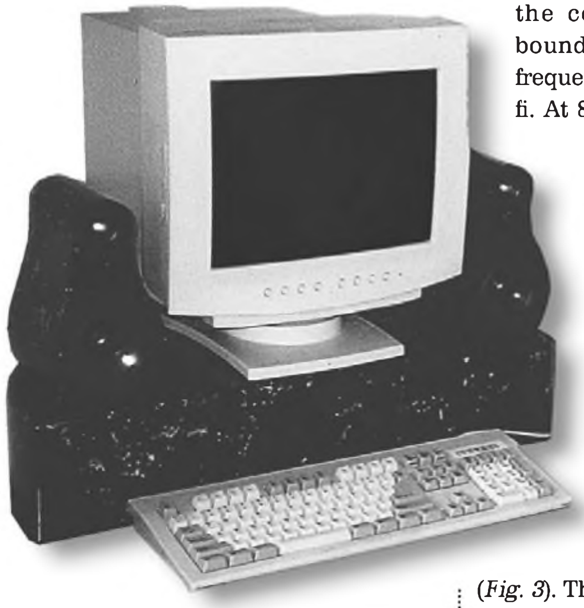


PHOTO 3: Round sub/15″ monitor.

woofer at three times the distance of the main speakers. If the main speakers are at 3.5m, the sub would need to be placed at over 10m! Imagine the delay!

The second problem concerns the crossover point. The main speakers in the computer environment are bound to be small, so their cut-off frequency is not as low as for the hi-fi. At 80Hz crossover you can place the sub just about anywhere. At 100Hz the sub should be placed at the same distance and plane as the main speakers. At 150Hz it should be as close as possible to the main speakers, preferably between them.

The Ultra Fidelity Computer Sound System has a subwoofer that doubles as a monitor stand and is wide enough to accommodate the satellites as well (*Fig. 3*). There is a noticeable difference between the two versions (described in part 3), one has the high frequency port on the far left, the other in the middle. Just 250mm makes quite a difference!

PROBLEMS CLOSE UP

Finally, there are a number of subjective differences between the two environ-

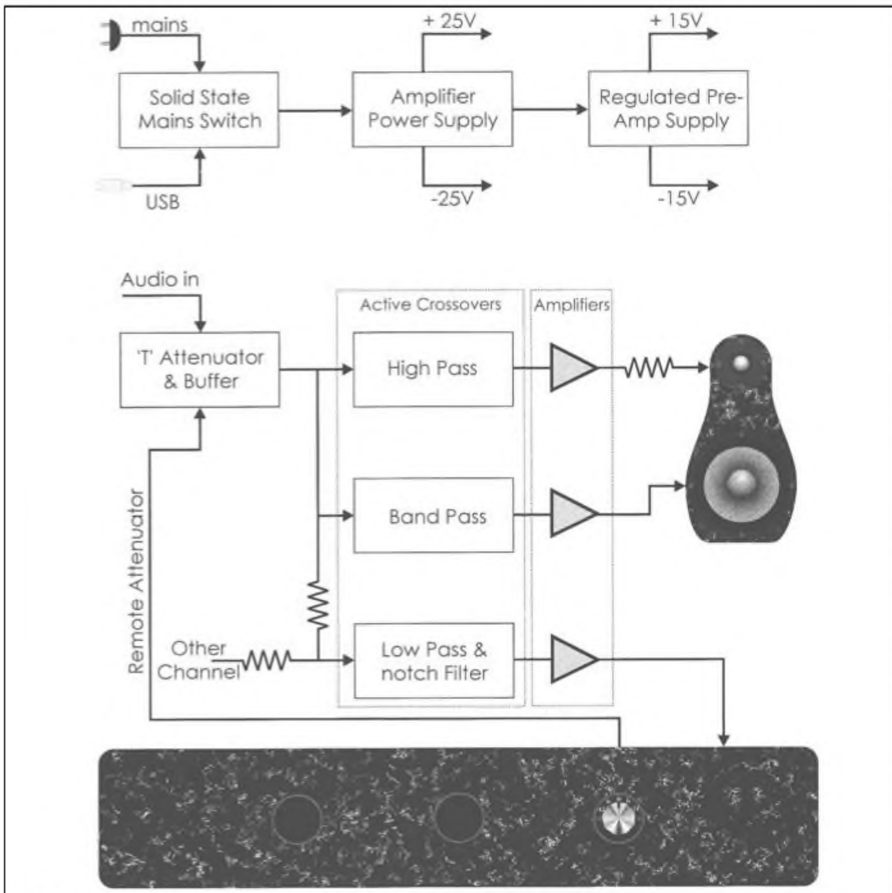


FIGURE 3: Ultra Fidelity Computer Sound System.

ments to take into consideration. If you've ever worked on speaker design—as I have—you will have noticed how different speakers can sound up close. This is partially due to the distance between the various drivers. The further apart they are, the further back you must be before they sound "integrated." The second problem is that you can "hear the cones" when up close.

By this I mean that a 12″ woofer/midrange, for instance, echoes like a bowl (probably because the phase change across the cone is noticeable only up close). You can also hear the sound of various cone materials—cloth dome midranges sound "clothy," polypropylene cones sound "plastic," paper sound "papery," but Aerogel does not seem to suffer this strange effect.

Finally, coloration in general seems to be more noticeable, particularly that sound coming from the cabinet. This may be caused by the fact that almost all sound is coming direct, so extraneous noise is easier to hear. I am relating my first-hand experience. I will leave the critical analysis to others.

DESIGNING THE COMPUTER SOUND SYSTEM

In designing a computer sound system, I also did an informal study of my friend's computers—the sound they were getting and measurements of the output and frequency range of their sound cards. Output was typically between 0.5 and 1.5V, but some that had higher outputs still clipped at around 1.5V. Older sound cards were reliably good, but newer ones often had difficulty with higher frequencies—two brand new cards rolled off at just 5kHz! I couldn't measure the output of my own card above 5kHz without a high-pass filter because of low-frequency noise. I upgraded straightaway.

Looking at pictures of sound cards in magazines, I notice that even the best ones still use cheap and nasty electrolytic capacitors. The problem for designers is the single 5V rail—the best op amps all require a balanced power supply, typically ±15V. The D-A converters are rough and ready compared to those of the better hi-fi CD players.

The Ultra Fidelity Computer Sound System is designed to suit the computer environment. The electronics (*Fig. 3*) feature an automatic mains switch that senses the computer's "on" state via a USB port (or keyboard or PS/2 socket). The signal runs through a "Pull-Down T Attenuator" (June '01 *aX*, p. 42) and then on to a three-way active crossover.

All the electronics fit onto a single PCB, including the power supply, power switch, active crossovers, and the five power amplifiers. I have specified Audax speakers throughout, but the design is flexible enough to allow experimenters to try other brands. The system frequency response has –3dB points at 35Hz and about 25kHz (within a 5dB envelope between 250 and 20kHz).

My approach has been to make the system integrated. In this way you can adjust the active crossovers for a particular set of speakers for instance, but the overall cost of the system is kept low. I have also made it open, so that anyone who thinks they know better can replace, upgrade, or modify any part of the system.

Although the speakers are not efficient by hi-fi standards, just as hi-fi speakers are not efficient by PA standards, the maximum SPL at the listener is as high as hi-fi.

The main speakers have low resonant cabinets featuring a design technique suitable only for computer speakers. The tweeter and bass/mid drivers are phase-aligned.

LISTENING TESTS

As a reference sound system for subjective listening tests, I used the magnificent Ultra Fidelity Concert Mk.4 sound system, a hi-fi system that would have sold for around $20,000 had it made it to market ("The Actipass Crossover System," *SB* 8/00, p. 14).

The computer system had better channel separation and so could place sonic objects more precisely. The sound is very accurate without being "tinny." Midrange is superb. The lack of reflected sound gives clarity, but lacks that all-embracing presence of a big powerful system.

The deep bass extends well down. But the Mk.4 runs all the way down to 26Hz, noticeable on only a few tracks. The deep bass does not hit you from every direction at once as the Mk.4 does, but is still fuller and more accurate than any other computer system I have heard, or heard of.

The sound is somewhere between hi-fi and a really good pair of headphones, but with the high-frequency accuracy that only real tweeters can give, and a deep bass presence that only a real subbass can produce. For spoken voice the Computer System is better than most hi-fis. When it comes to computer speakers, think "information."

The information reaching the listener is theoretically highest with headphones. But headphones do nowhere near as well at placing sonic objects in space as the computer sound system does. A well-designed and -built, high-quality computer sound system offers a new dimension to the experience of sound.

In part two I will describe the satellite speakers; part three will focus on the subbass; and subsequent parts will detail the electronics. ❖

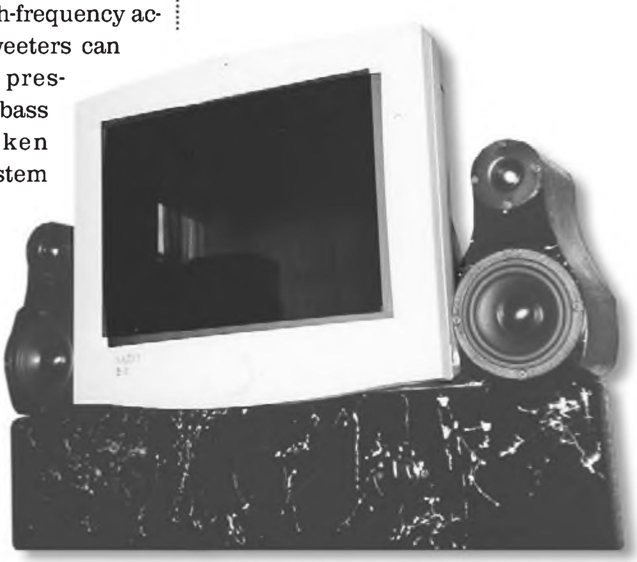


PHOTO 4: Round sub/17″ monitor.

Phase Inverting By Transformers

This author shows how you can use today's transformers—with the right tubes—to drive push-pull amps. **By Bernard Bégaud**

Much as Dr. Jekyll and Mr. Hyde, interstage transformers are presented as both the best and the worst of audio reproduction. In early audio, transformers were the only way to drive push-pull amplifiers. Because of their technical limitations (core saturation, narrow bandwidth, and distortion) they were progressively abandoned when tube phase-inverters became available. From the primitive but excellent split-load inverter to sophisticated circuits developed for the last "monsters" of valve amplification, it was a commonly held opinion that transformers were just good enough for public-address purposes or amateur builders of prehistoric systems.

Interstage transformers regained popularity recently in a restricted circle of audiophiles, mainly in Japan where some manufacturers such as Tamura and Tango developed upgraded transformers based on new core materials and sophisticated winding techniques.

Intrinsically, transformers approach the gold standard for phase inverting through having

- a minimum number of components;
- total symmetry of inverted signals;
- insensibility to tube aging; and
- the capability of delivering very large output signals, e.g., >100V RMS.

The two plagues of inverting transformers remain:

- a limited bandwidth: at low frequencies because of core saturation, and at high frequencies because of phase rotations;
- the near impossibility of using a large amount of global negative feedback without stability problems.

You may in part eliminate these two drawbacks by using a modern performing transformer.

ABOUT THE AUTHOR

Bernard Bégaud is an enthusiastic subscriber to *audioXpress*. He has conceived and built many audio circuits, and has a good audio lab that allows measurements of harmonic-distortion components up to 0.005%. He is particularly interested in articles that start with reliable theories and proceed to actual (not simulated) tested circuits with extensive and relevant measurements of performance without forgetting that the final objective is music reproduction.

METHOD

I started with the Tango NC14 transformer, which is readily available at an acceptable price. Its primary impedance is on the order of 5kΩ, and its bandwidth, according to Tango, is reasonably good: 25Hz to 40kHz at –2dB, although it depends on the tube you use.

Its primary coil is split into two equal parts, which allows them to be used in series (impedance ≈ 5kΩ) or in parallel (≈ 1.25kΩ). I must confess that I never achieved satisfactory performance when using the parallel design, so in the following I will consider only the first type of connection.

In the NC14 package insert, Tango suggests several tubes for feeding the primary: 6V6 and 6F6 triode-connected, 6AH4 GT, 5687, and 6350. In fact, for a primary impedance of 5kΩ, any tube could be a candidate if it has an R_p (plate resistance) ranging between 2.5 and 4kΩ and a plate current ranging between 5 and 30mA. Other considerations are amplification factor, plate dissipation, and performances achieved.

Although suggested by Tango, I rejected triode-connected pentodes or tetrodes and stayed with miniature and octal triodes. After an extensive review of all tubes meeting these criteria, I retained a series of 11 candidates, built an appropriate testing circuit, and compared achieved performances in terms of bandwidth, gain, maximum output level, aspect of square-wave signals, and harmonic distortion.

For the last attribute, I did not consider the total harmonic distortion, but the second and third components separately, which is much more informative. (I would prefer to have 1% distortion of mostly even harmonics, rather than 0.25% of third and fifth harmonics.)

For these measures I used the following material:

- Leader LAG 126 low distortion audio generator;
- Hewlett-Packard 201C audio generator;
- Dumont AN-USM 281E 50MHz dual-trace scope;
- Hewlett-Packard 302A wave analyzer;
- Hewlett-Packard 400GL AC voltmeter;
- Metrix MX55 high-precision multimeter;
- three other multimeters;
- FX3380A multifunction counter; and
- two variable power supplies: 0–1,000V and 1.25–32V, regulated.

The tubes I compared were: Philips ECG 5687WB JAN, Sylvania 6350, GE 6443, Sylvania 6CM7, RCA 6CS7, CBS 6BK7B, Western Electric 417A, Raytheon 6AH4 GT, Raytheon 2C50 JAN, RCA 6BL7GTA, and Tung-Sol 6SN7 WGT JAN.

RESULTS

The first problem was to adopt the best configuration for the NC14. I obtained

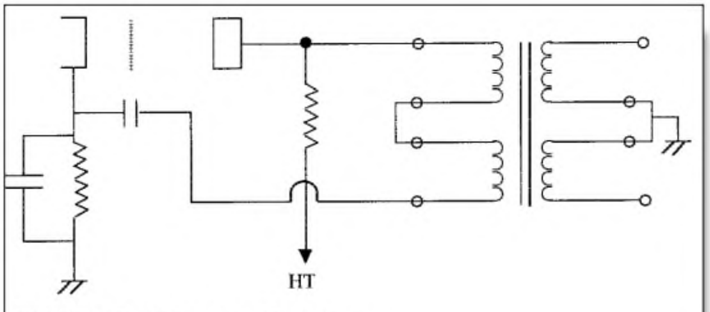


FIGURE 1: Traditional configuration.

satisfactory square-wave signals only when the center tap of the secondary was grounded. This could be a problem if you wish to apply independent negative-bias voltages to the grids of the output tubes through each half-secondary. In this case, it is necessary to ground the corresponding tap through a large capacitor (e.g., 4 to 10µF). All other configurations resulted in poor performance (irregularities on square-wave signals).

I did not see any advantage in having a resistance in parallel with each part of the secondary, nor to the circuit (*Fig. 1*) you classically see in Japanese audiophile books.

This circuit avoids the tube current passing through the primary, thus reducing the core saturation. This I expected to extend low-frequency response and to reduce distortion. With the NC14 and the tested tubes, this was not the case. Moreover, this schematic is far from satisfactory, because at least one of the greatest advantages of transformers from the point of view of sound quality is to avoid having the resistance charge the plate.

Since I am not enthusiastic about having a capacitor in the signal path, I was disappointed to see that it was impossible to avoid the capacitor bypassing the cathode resistor of the driver tube. I never obtained satisfactory signals and performance without it.

After several weeks of troublesome testing (each tube being tested under several values of plate voltage and cathode resistor), I came to the following conclusions:

5687WB

One of the tubes suggested by Tango, and they are right! It is a very well-made tube, easy to find, relatively cheap, with excellent characteristics: plate dissipation of more than 4W, amplification factor of 16.5, R_p = 2,500Ω. Moreover, this tube enjoys an excellent reputation among audiophiles. With the NC14, it provided good square-wave signals and performance (*Table 1*). I kept it as a finalist.

6350

A good performer, although less so than the 5687 on several parameters. Overall, the square waves were not very satisfactory.

6443

Performances were similar or better than 6350, but not as good as the 5687.

6CM7

This tube contains a high-mu and a medium-mu triode, which I used. After testing several options, I was unable to obtain satisfactory results.

6CS7

Like the 6CM7, it is a high- and medium-mu double triode. Performances were better than the 6CM7 (HT 230V, R_k = 900Ω, and I_p = 12.4mA). In fact, it is a good performer for those who wish to use the high-mu section as a preamplifier to obtain a three-valve push-pull amplifier.

6BK7B

The only tube which did not produce any overshoot on square-wave signals. However, in all test situations, the global performance was disappointing.

WE417A

One of the two black ducks of the sample, since this tube did not meet the inclusion criteria because of its R_p of 1,800Ω. However, I retained it because I am a 417A aficionado. Unfortunately, 417A did not like to work with the NC14. It provided a lot of gain (more than 20) and good square-wave signals—close to those of the 5687 with more overshoot (lower R_p)—but I could not obtain an output of more than 30V RMS before clipping. Because of its low R_p, it was tempting to use it with the paralleled primaries, but this was even worse.

6AH4 GT

What impressive heating! Just as the WE417A, this tube was included in the selection despite its low R_p of 1,780Ω. Very good square-wave signals, better than the 5687, but less gain and not as interesting for other performances.

2C50

A very scarce tube (I found only two of different brands after several calls to suppliers). The 2C50 and 2C52 double triode series enjoys an excellent reputation among audiophiles. One of the four best performers of the sample (*Table 1*). Unfortunately, the best performances in terms of harmonic distortion were obtained for a plate dissipation (i.e., 350V and 11.6mA) that exceeded the maximum admissible according to

TABLE 1
PERFORMANCE WITH THE TANGO NC14 AND FOUR DIFFERENT TUBES

	2C50	5687WB	6BL7GTA	6SN7 WGT (PARALLELED)
	R_1 = 2.5kΩ 350V/11.6mA	R_1 = 1.8kΩ 320V/8.2mA	R_1 = 1.1kΩ 400V/20mA	R_1 = 1kΩ 410V/16mA
gain	5	8.3	6.6	12.5
bandwidth (−1dB at 50V RMS)	30Hz–21kHz	31Hz–20kHz	22Hz–25kHz	21Hz–23kHz
maximum output level (V RMS); 1kHz sine waves	100	80	150	150
maximum output level (V RMS) at 25Hz before clipping	30	15	40	50
harmonic distortion (at 50V RMS)	H_2: 1.6% H_3: 0.2%	H_2: 1.25% H_3: 0.1%	H_2: 1.6% H_3: 0.1%	H_2: 0.4% H_3: 0.03%

specs (3.85W). However, considering the rugged construction of this tube, it seems acceptable, and I retained it for the final race.

6BL7GTA

I had several of these atypical medium-mu triodes with an amplification factor of 15 despite a low R_p of 2.15kΩ. It is a rugged tube, very easy to find at a good price. It gave excellent results with the NC14 (*Table 1*), and was one of the best performers of the sample.

6SN7 WGT

The good surprise came from this tube used with the two sections in parallel (resulting in an R_p of 3.5kΩ). Driven under a 15–16mA current, it provided an impressive set of performances: a large bandwidth, the lowest harmonic distortion, the highest gain and output signal, and nondistorted 25Hz sine waves up to 50Vms!

FOUR FINALISTS

Table 1 compares the results obtained with the four finalists (one miniature and three octal tubes, all of them twin medium-mu triodes). The 5687WB was good for harmonic distortion and gain. In fact, the 5687 is probably an excellent choice for those who don't need a very large output signal.

The 6BL7GTA was better for other parameters. Its gain ranges between those of the 2C50 and 5687WB, it had the wider bandwidth, and can provide an output signal as high as 150V RMS per secondary section. Moreover, unlike the 5687WB, low-frequency sine waves were not distorted up to 40V RMS at 25Hz. Distortion, especially third-harmonic, was low. The 2C50, if available, is an excellent compromise, but it is a rather masochist behavior to overdrive

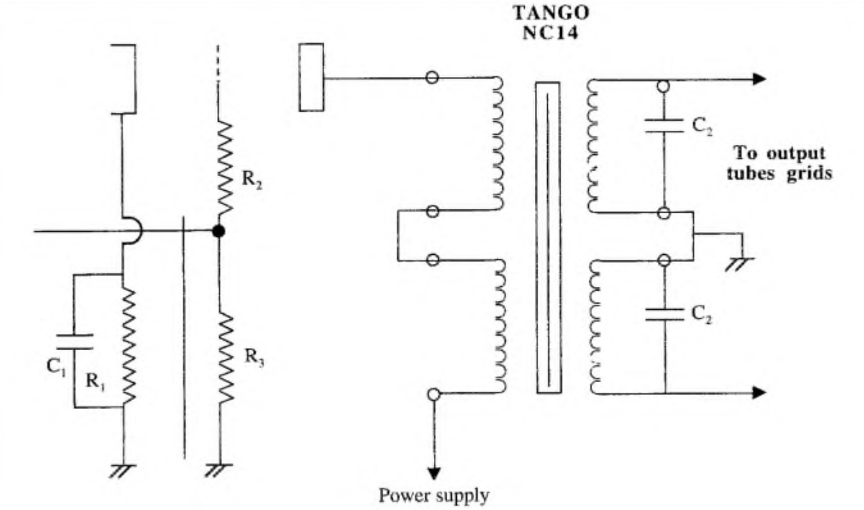


FIGURE 2: The tested NC14 phase inverter. R_2 = 2.2kΩ, R_3 = 220kΩ, C_1 = 100μF 25V, the value of C_2 should be adjusted for best square-wave signals. The value of R_1 and power-supply voltage depend on the tube used (see *Table 1*).

a tube that is almost impossible to find.

The 6SN7 WGT was definitely the winner, i.e., the best for each parameter (gain, distortion, output) except for bandwidth. One problem with the 6BL7GTA and 6SN7 WGT was a marked overshoot on 1kHz square-wave signals corresponding to a peak in high-frequency response (+2.8dB at 43.3kHz). Such an overshoot is common with interstage transformers and is probably of no consequence for sound reproduction, since it corresponds to inaudible frequencies.

After trying several remedies (local feedback, capacitors, snubbers in various places), the best compromise was to parallel each secondary coil with a 1–4nF capacitor. The square waves obtained were more acceptable and the frequency response was flat up to 25kHz (–0.1dB at 20kHz, –0.8dB at 25kHz, –2.6dB at 30kHz, and –11dB at 40kHz).

However, I think these capacitors are unnecessary except for those who prefer to have a scope plugged into their amp rather than a loudspeaker. Many famous audiophile designs suffer from such overshoots that do not seem to have audible consequences. The complete schematic is shown in *Fig. 2*.

With the paralleled 6SN7, the bandwidth, for an output of 50V RMS, was 21Hz–23kHz at –1dB, with an impressive maximum output voltage of more than 150V RMS and a harmonic distortion of 0.9% H_2 and 0.1% H_3 at 100V RMS, 0.4% H_2 and 0.03% H_3 at 50V RMS, and 0.17% H_2 and 0.01% H_3 at 20V RMS. You should keep in mind that I obtained these figures without any feedback and that the third-harmonic component was very low compared to the second. Moreover, 50V RMS corresponds to no less than 140V pk-pk.

For those (and they are right) who don't wish to have a resistor or capacitor on the signal path, it is possible to ground the cathode of the driver tube and to apply an equivalent negative-bias voltage on the grid of the 6SN7. In this case, the grid resistor is not grounded, but is connected to a negative-voltage supply of –25V or, better, to the wiper of a 47kHz 10-turn potentiometer connected between a –40V source and ground. This will allow a precise adjustment of bias for minimal harmonic distortion.

Even if I found no advantage to this solution in terms of harmonic distortion, avoiding resistances and capacitors in the signal path is always preferable from a sound-quality point of view. However, it is necessary to kill any residual hum and RF noise in the negative-voltage supply. For example, during my experiments, I used a power supply regulated by an LM 317T and ad hoc capacitors. Despite that, the sine waves were intermodulated by a very low residual RF noise (very unpleasant, at least on the scope). The only satisfactory solution is to use a perfectly filtered power supply, or 40V batteries, or to return to the cathode-bias solution.

The described phase-inverter reached particularly good performances: a gain of 12.5, an unconditional symmetry, and an unsurpassable output level with very low odd distortion. It would be suitable with many push-pull amplifiers, mainly those that do not require a large amount of negative feedback or no feedback at all, such as triode push-pulls. Not so bad for a prehistoric design! ❖

JBL L300 Summit Renovation

This article describes the search for a pair of JBL L300 speakers, and subsequent renovations and improvements to achieve a truly outstanding level of performance. **By George Brooke**

About 25 years ago I was living in Munich, Germany, which is a country where large loudspeakers are still popular. During a visit to a Munich hi-fi specialist, I had a demonstration of various speakers, including Audiostatic, Quad Electrostatics, and a pair of JBL L300 Summits. The JBLs were out of my price range, but what I heard stayed with me for a long time. They were amazing in terms of breadth, scale, and sheer impact.

Coincidentally, that month they were reviewed by Martin Colloms who concluded, "Quite simply, they delivered the goods"[1]. I agreed with that comment, and then some.

THE QUEST

Since then I have gone through many other speakers, including Kef 105 Monitors, Klipschorns, IMF Transmission Lines, Spendor BCIII, and several "home-brews." The Klipsches were close to what I was seeking, but they seemed to trade detail for clout, and ultimately I found them tiring.

Then there was a local development. A nearby store started trading used high-end audio systems. I quickly went through amplifiers from Accuphase, Krell, Lumley, and finally settled on Nelson Pass' Aleph 5 power amp and matching preamp. This combination provided outstanding detail and focus, and also kept the house warm as a bonus.

The Pass amps continue to be my reference amplifiers. I started to look into selecting speakers. A pair of Spendor BCIII speakers arrived. They provided excellent detail, imaging, and clarity, but insufficient excitement or involvement. (They now reside with my son in Brussels.)

Remembering the sound of the L300s (from a quarter of a century before), I tried to locate a pair in the UK without any luck. JBL over here (outside of professional systems) is now best known for their in-car equipment. Then on holiday in France I happened across a French hi-fi magazine. In the small ads in the back pages there were several L300 ads, with prices at around £1500 a pair.

After an incredible amount of arranging via telephone, fax, and my son in Brussels, I selected a Parisian dealer

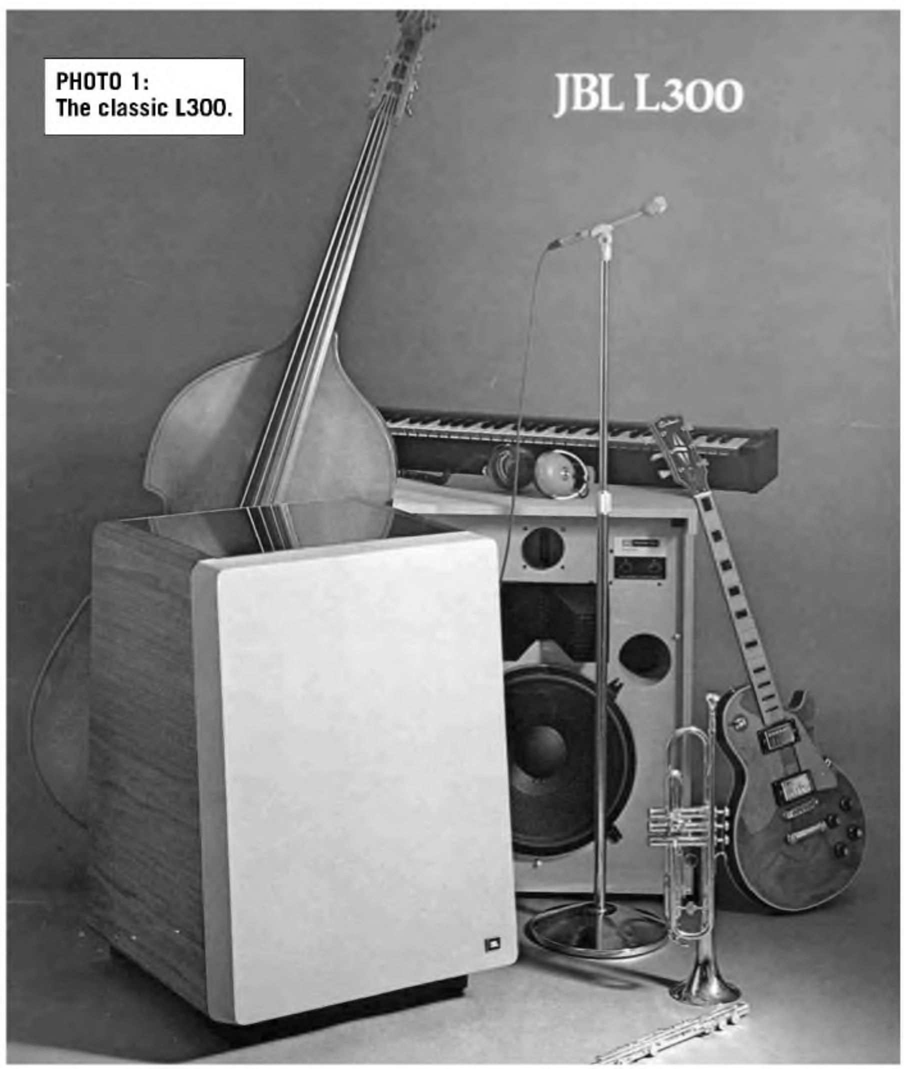


PHOTO 1: The classic L300.

ABOUT THE AUTHOR
George Brooke is an IT consultant and trainer, who works in object-oriented and web technologies for the telecommunications and banking industries. He has been building loudspeakers and amplifiers for over 40 years. He lives in Cambridge UK, where he runs his consultancy Oak-Lodge Consulting Ltd. He can be contacted at george@oak-lodge.com or via his website www.oak-lodge.com.

PHOTO 2: Prior to rebuilding the grille.

PHOTO 4: Bass and treble units.

PHOTO 3: Damaged frame.

advertising a pair of JBL L300 Summits in "perfect condition" and with re-coned bass-units. (The dealer is best kept nameless in this charade). I sent off the check, which was cashed immediately. The speakers finally arrived just over three months later in a home-made packing case. *Photo 1* shows how the speakers looked originally (this is from JBL's original advertising). This article describes what I did to try to recapture that sound of 25 years ago.

THE STARTING POINT

To say that these speakers were not "perfect" is an understatement. The wood was scratched; what should have been tinted glass tops were cheap brown plastic; the wire connectors were split; the wheels were missing.

However, this is all easy stuff to fix.

Worse, though, was the complex front grille arrangement. One grille looked as though someone had stomped all over it. The "repair" made by the previous owner was to glue and staple the wooden frame together. Unfortunately, the glue had dripped through into the brown cloth, staining it a dirty white.

Apart from that, when I connected the speakers they sounded just terrible—sharp, aggressive, and with a massive, boomy, and loose bass. I found that the level settings for the mid and high frequencies were both set on full. Even reducing them to center did not remove the edge from the sound. I wondered about the hearing of the previous owner! The bass units had been reconed, though.

In comparison with my recent Spendors, the sound had taken a great leap backwards. The detail and depth had vanished, and the imaging was very poor. There was simply no comparison, even though they cost three times the price of the Spendors. This was clearly going to be a more extensive project than I had expected.

My initial thought was to sell them, but I doubted if they were saleable (except perhaps to people like me). So, I made a plan to try to get the speakers back to my remembered quality of a quarter of a century previously. It worked.

THE PLAN

I decided to broadly separate the work into the cosmetics and the electricals. The cosmetics would address everything to do with the appearance, and the electricals would address everything to do with the sound-generating components.

Before going too far, I chose to research the speakers, in terms of components, crossover design, cabinet design, and so on. I soon found, via the Internet, like-minded individuals who were happy to provide good background information. For example, Ken Manson (who I contacted via a JBL enthusiast group) even sent me an original JBL color brochure of the speakers. The Audioheritage (www.audioheritage.org) website was a goldmine of information about the speakers, and I managed to get a copy of the crossover design from Wizards Electronics Repair Center for just $3. Website www.audioasylum.com was also very useful.

An exploratory opening up of the speakers was very reassuring. There were no woodscrews for holding removable components. It was all metal bolts into metal sockets. Moreover, the design was beautifully modular. Access to all components was either by removing the bass unit or by removing the access panel at the rear.

Apart from the level controls (which are bolted behind a JBL identification strip and which would be ruined by their removal), I can now strip one of these speakers in about 15 minutes without fear of damaging fittings. This obvious quality of the speakers really cheered me up after a pretty depressing start.

THE COSMETICS

First to be fixed, due to the weight, were the wheels, or rather the absence of wheels. There were holes where they had been fitted, so I needed to avoid these because my castors had a slightly larger base. I really could not handle spikes on 75kg speakers, so I needed wheels, which helped a lot.

Next, I repaired the cabinet with small amounts of plastic wood, and oiled it repeatedly (Danish Oil). The result was a beautiful deep glow from the walnut veneer. I judiciously used matte black paint to repair the various minor scuffmarks.

A visit to the glass specialist produced a perfect-fitting smoked glass top to replace the cheap plastic used by my Parisian friend.

The grille damage was much more difficult to repair. The grille is very visible, and any irregularity looks very bad. I decided that I must rebuild the entire grille of one speaker.

The grille is complex, about 3″ deep, with a front and rear-frame to clear the midrange horn diffuser (*Photo 2*), and tapered from front to back. The grille cloth is extremely stretchy and open-weave. I could not find, and still have not found, a source for this cloth, so I needed to be able to reuse anything I removed.

The original grille frame was made from particleboard, which is fragile for frame purposes. I decided to use MDF instead, and used the frame of the good speaker as a model. I rabetted a recess

PHOTO 5: Working on the crossover.

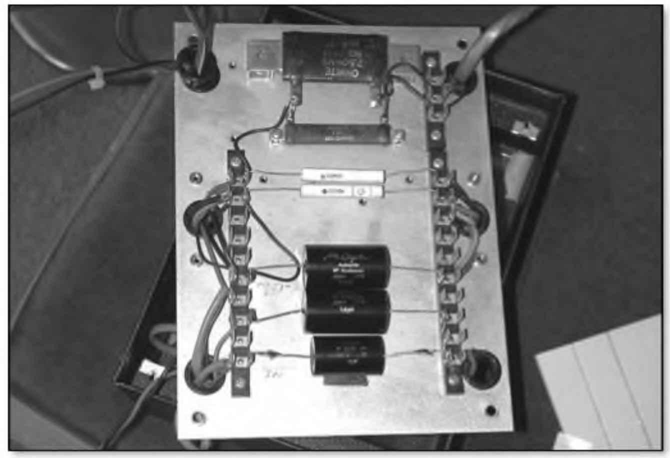


PHOTO 6: Capacitor replacements.

about 1″ wide and 1/8″ deep all around, so I could staple the cloth in place. After removing the cloth from the damaged frame, I attempted to reduce the staining from the dripped glue. I shifted most, but not all of it.

In the process, however, the cloth shrank dramatically. It still fits the new grille, but the tension is too high for comfort and I continue to look for this quality of cloth as a backup. It has been in place for about three months, so perhaps there is no cause for alarm. Fitting took a while, but the rebuilt grille is indistinguishable from the other, and the work was certainly worthwhile.

By far the most critical part of rebuilding the grille is the tapering from front to back. The tapering is controlled by the four corner pieces, which were undamaged and able to be reused. Without these, the work would have been quite complex. You can see the damaged frame on the floor outside of my garage in *Photo 3*. It was totally useless. The tension of the cloth finally caused it to collapse after just a few days.

Finally, I contacted JBL in the UK to get appropriate JBL badges for the speakers. I eventually reached one of their enthusiasts who remembered these speakers and promptly dispatched two large JBL badges. Now the speakers looked (more than) acceptable in a domestic environment. But they still sounded terrible.

THE ELECTRICALS

I had several stages planned for the sound improvement. One was to biwire the speakers. I have used this in the past and found that on high power levels it makes a significant improvement to the sound. On high-efficiency speakers such as the Klipschorns or the L300s, the improvement is unlikely to be much.

However, since I had made a biwire cable (consisting of Monster cable for the bass and solid-silver for the rest) for the Spendors, I chose to use it for the L300s, too. The biwiring, for compatibility reasons, involved the use of Neutrik Speakon four-way connectors, to replace the damaged bare-wire connectors of the L300s.

I would also upgrade the internal wiring, with heavy-duty silver-plated wire for the bass and a continuation of the solid silver for the mid and high units. Obviously, I needed to do all soldering with silver-loaded solder. Finally, I intended to replace the smaller electrolytics with Musicap equivalents and to bypass the larger electrolytics. This did not quite work out as planned.

STRIPPING THE SPEAKER

First, I disconnected the speaker and positioned it on its back and removed the bass unit. The bass unit should have been held with eight Allen bolts, but my speakers had just four loose bolts on each bass unit, possibly accounting for some of the boom. These bolts go into T-sockets behind the front panel (which itself is not removable).

I removed the small rear panel to access the midrange unit. The rubber seal around this panel had dried out and needed to be replaced. I used a double synthetic-rubber seal in its place.

Once this panel is removed, the rear of the midrange unit and its connections are easily accessed. Note that there is a kind of wooden "key" fitted to this backpanel to offer some support for the midrange unit. The result is that there is only one way of fitting the back panel.

The treble unit is held via four external bolts. If you undo all of them, the unit will fall inside. This is not a good

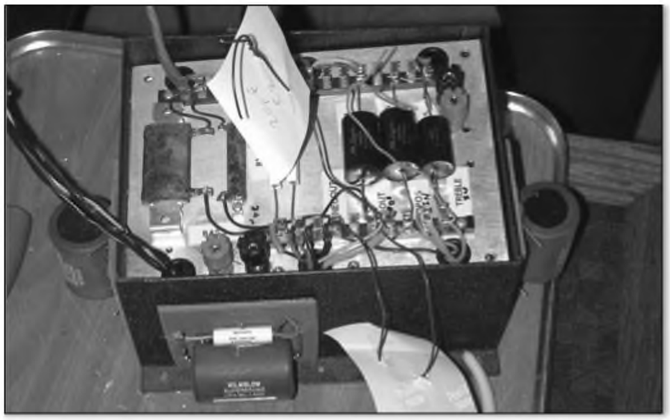


PHOTO 7: Wiring work.

PHOTO 8: The revamped JBL speaker.

idea because it is very heavy.

I found that the best approach is to hold the back of the unit with one hand while undoing the bolts from the front. You must find a way of getting your arm past the middle-frequency horn and up to the rear of the treble-unit. Removing the midrange horn will certainly help, but I never needed to. I have long arms! *Photo 4* shows the bass and treble units after I removed them.

The crossover is located on the floor of the speaker and is held in place by four bolts, the wires to the speaker inputs, and the wire to the variable inductors, which are mounted on the inside front panel. I never did remove these, because it would have caused damage to the JBL model information transfer. The cables connecting the variable inductors to the crossover are conveniently long enough (after their ties have been cut) to allow you to work on the crossover outside of the cabinet. You can see the general working arrangement in *Photo 5*.

At this point I had a speaker cabinet containing just the midrange unit and the inductors. The crossover can be held about a foot away from the cabinet.

I found that the inner floor of the speakers (both of them) had no absorbent covering. However, it is clear from the original JBL brochures that they should have, so I purchased some wadding and added 1½″ around the crossover. I did this only after all the other work was finished, to keep out of the way of all the wiring.

As I took the speaker apart, I checked the connections against those documented in the crossover. They were identical. I mention this because sometimes manufacturers make undocumented alterations. However, here JBL had been completely consistent.

THE SPEAKER CONNECTORS

This was the only disappointing part of the whole construction. The two spindly connectors were mounted on a ¼″ fiberboard panel. The connections on the inside of the panel were made via two deteriorating spade connectors. The wire was of reasonable quality, but nothing special.

My problem was that Speakon sockets need a thicker mount for their screw fittings. I decided to preserve the fiberboard, but to back it with ½″ MDF panel, drilled for the Speakon connectors. I cut the appropriate size hole in the fiberboard and in the MDF panel. Then I aligned them from the inside and used the now vacant holes in the fiberboard from the original connectors to screw the MDF panel to the fiberboard. I kept the Speakon connector in place as a guide for this tricky alignment operation.

GENERAL RE-WIRING

I simply cut my silver wire to the same lengths as the original wires and replaced them one by one, preserving the color coding (as described in the crossover diagram). However, although this was straightforward, you need to read about the crossover itself (following) before undertaking this task.

THE CROSSOVER

Unlike previous crossovers on which I have worked, this was based on point-to-point wiring. So, no printed circuit board! The crossover was approximately 10″ × 4″ × 6″. The top was a removable (four screws) metal grille, mounted on four aluminum standoff posts.

Immediately under the grille were the point-to-point wiring tag-strips, capacitors, and very heavy-duty resistors. But not all components were mounted here! It turned out that there was a further, lower layer that contained the large components such as the inductors and the large-value capacitors. The wires for these were brought through the above layer at several points for connection to the tag-strips.

Unfortunately, the wires had been pulled through to keep them short, so there was no way of getting to the large capacitors without cutting their wires. I could see through the narrow gap that all of the large components had been potted in a pool of heavy resin and were completely immovable. This was bad for me, but better for the sound. Obviously, this reduces the effect of vibration on the crossover, but without taking a chisel to the unit I could not see how to remove these components, and I really wished to reuse the original crossover as much as possible.

My upgrade plan was to replace all capacitors, and to use silver wire wherever possible. There was an intermediate stage, where I reused the existing large capacitors, but added high-speed bypass capacitors to them. I tested this on one speaker, but I found that the full capacitor replacement was superior. (Though not enormously so, so to save money, you can add bypasses to the larger capacitors, rather than replace them. This depends, of course, on their condition.)

I replaced all the smaller capacitors with Musicap equivalents (*Photo 6*). This was very simple because they were all top mounted, as you can see in the picture. The "solution" for the larger capacitors was not very elegant, but worked. I decided to leave the large capacitors embedded in the resin, but disconnected.

I built three fiberboard panels of approximately 3″ × 4″, each holding modern, non-electrolytic equivalent values, all with 5% tolerance. I built the 16.5μF capacitor from three paralleled capacitors. I found that by using very long side cutters, I could cut the wires to the large resin-bound capacitors just above their cases and tease these wires through to the top. I could then replace these wires with silver wiring directly to my fiberboard modules, which I mounted on the outside of the crossover box.

You can see how this looks in *Photo 7*. The Post-it notes on the wires are my notes about where each wire should go. I was also careful to preserve the orientation of my new capacitors with the originals, by keeping them parallel.

The only remaining task was to split the crossover to allow the bi-wiring. At this point the number of wires hanging from the crossover was becoming difficult to manage, and I preferred not to hard-solder four more long wires from the unit through to the back of the loudspeaker cabinet, so I decided to use shorter wires from the back of the Speakon connector—now fitted to the back panel of the speaker—and to mount multiway connection posts directly onto the crossover board. There is plenty of space for this, and apart from drilling the board for these mounting points, you need only trim the ventilation grille to allow the extra entry points for the cables. This gives a more flexible setup, at the expense of a couple of extra metal-to-metal connection points in the loudspeaker-amplifier pathway.

The tag-strips have many spare connection tags, so separating the bass-unit component of the crossover from the rest was quite straightforward. I found that the point-to-point wiring makes things harder to follow than with a printed circuit, so I labeled components and tag points by references to the crossover schematic. I also took digital photographs to help when repeating the work on the second speaker.

RE-ASSEMBLY

Re-assembly is a bit of an anti-climax because it is so easy. Keep in mind that it is (obviously) critical to preserve the connection polarities from the schematic. There were basically six stages:

1. Re-mount the crossover in the cabinet.

2. Connect the tweeter to the crossover before bolting the tweeter inside the cabinet.

3. Put in the (missing) wadding on the speaker floor around the crossover.

4. Connect the midrange unit to the crossover.

5. Replace the rear access panel, replacing the dried-out seals as suggested previously.

6. With the speaker on its back, reconnect the bass unit and drop/bolt it into the cabinet.

Finally, set the speaker upright. This should get you to a working speaker, biwired, and with the latest capacitor and wiring technology. *Photo 8* shows one of the finished speakers near the equipment rack.

RESULTS

In the past when I have made successful equipment upgrades, I have noticed that I listen more often and I hear instruments and voices which were not previously obvious. Well, I heard all of this and more. First, the big problem of "edginess" has completely disappeared. More detail is being revealed than ever before.

The bass boom has also gone, although I would be hard-pressed to say which change actually did it. The integration of bass with the rest of the music is tighter, and overall the system sounds faster and crisper. Compared with the Spendor BCIII, the imaging of the L300 is not as precise, but the excitement and involvement is much better. The bass is solid and detailed, but not at all intrusive. I frequently hear instruments of which I was not previously aware playing away in the deep bass. Similarly, on close harmony, the voices easily separate out instead of blurring together.

I suppose the question arises as to whether I have achieved the sound I heard in Munich over 25 years ago. To be perfectly honest, I have no idea. But I am very happy with the sound that I have, and I hope to stay with this particular setup for a very long time.

This is probably the most significant upgrade I have made, excluding the Pass amps themselves. If you have the chance to work on a pair of these excellent loudspeakers, I recommend that you try these changes for yourself. If you need help or encouragement, contact me at george@oak-lodge.com. ❖

REFERENCES

1. I have been unable to track down this review. I would be grateful if anyone can point me toward a copy of it.

Tannoy S8LR

By Joseph D'Appolito
and Dennis Colin

This speaker is a bit different than other two-way speakers I have tested. First, the S8LR uses Tannoy's trademark "Dual Concentric" driver. The tweeter diaphragm and motor are mounted directly behind the woofer magnetic circuit. The high frequencies are emitted via an acoustic wave-guide passing through the woofer's pole piece. Second, Tannoy supplies a cylindrical foam plug that you can use to close the speaker's port, thereby converting the S8LR from a vented to a closed box system.

I ran a series of impedance, frequency response, and distortion tests on the S8LR loudspeaker under both configurations. *Figure 1A* is a plot of system impedance magnitude with the port open. At low frequencies the plot displays the double-peaked curve of a vented system. The impedance minimum of 5.8Ω at 33.6Hz indicates the vented-box tuning frequency. There is a second local impedance minimum of 4.6Ω at 165Hz.

Impedance phase lies between +59° and −56° over the full audio range. Fortunately, these rather large phase angles occur at relatively high impedance values. With minima in the range of 5Ω, Tannoy's 6Ω rating is appropriate.

The impedance peak of 53Ω at 2600Hz (*Fig. 1A*) is probably caused by the interaction of the woofer and tweeter crossover networks forming a parallel resonance at that frequency. We have seen this phenomenon several times before, but the value of the peak is quite a bit higher than values obtained in previous reports.

Figure 1B is the S8LR impedance with the port plugged. This plot shows a single low-frequency peak characteristic of a closed box speaker. The peak value of 42.1Ω occurs at 60Hz. Above 120Hz the closed port impedance curve is identical to the open port plot.

FREQUENCY RESPONSE

Figure 2 shows the S8LR's full-range frequency response for both open and closed port conditions. This response is obtained as a combination of the far-field quasi-anechoic response and properly summed near-field woofer and, for the open port case, the near-field port responses. I placed the microphone along the common centerline at a distance of 1.25m to produce the far-field response, and then spliced the near- and far-field responses together at 200Hz to produce the full-range response[1].

The response shown in *Fig. 2* has been normalized to 1m to obtain system sensitivity. Sensitivity averages 87.2dB in the four octaves between 250Hz and 4kHz. This is almost 3dB less than the figure quoted in Tannoy's specs. Relative to this level the low-frequency −3dB open port point is 43Hz. With the port closed the corresponding figure is 56Hz.

Response is relatively smooth between 2 and 15kHz. There is a broad response dip between 300Hz and 1.6kHz. This may give the S8LR a somewhat recessed character. Response dips 2.4dB just

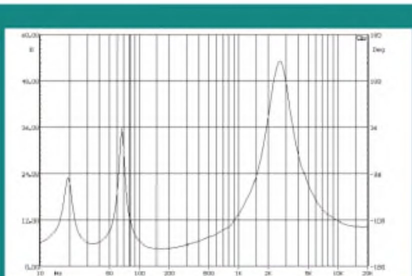


FIGURE 1A: S8LR impedance response with port open.

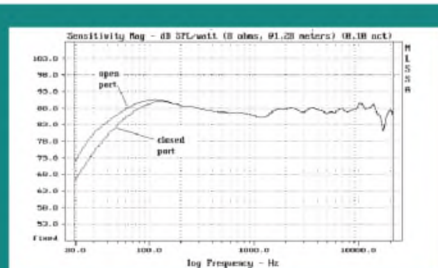


FIGURE 2: S8LR response with open and closed port.

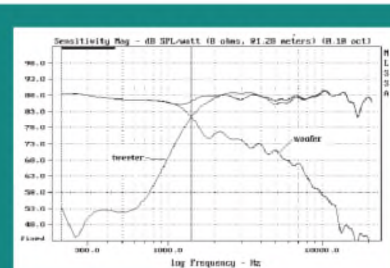


FIGURE 4: System and driver responses.

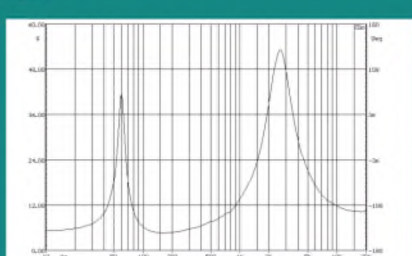


FIGURE 1B: S8LR impedance with plugged port.

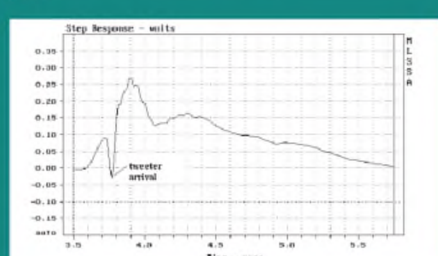


FIGURE 3: Step response with closed port.

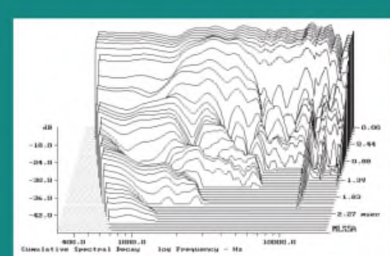


FIGURE 5: Cumulative spectral decay.

above 1kHz and then peaks 2.4dB at 130Hz for the closed port condition. A 3dB peak at 110Hz is obtained with the port open. These response peaks may give a sense of warmth to male voices. The overall closed port response is ±2.4dB from 60Hz to 15kHz.

With the port open I get +3/ –2.4dB from 50Hz to 15kHz. The sharp response dip at 17kHz appears to be a diffraction-induced anomaly caused by interaction between the tweeter waveguide and the woofer cone. More on this later.

WOOFER/TWEETER TIMING

Figure 3 shows the S8LR step response. The rather amazing thing about this plot is that it shows the woofer response at the microphone rising before the tweeter arrives! This occurs because the tweeter diaphragm is at the rear of the woofer. In all of the test reports presented in this series, this is the first system to exhibit this behavior.

Tannoy claims in their literature that the S8LR phase response is linear in frequency, which implies that the speaker is time-coherent. In spite of the concentric mounting, however, the S8LR is not quite time-coherent.

Excess group delay is a very sensitive measure of inter-driver time-coherence. (See chapters 6 and 7 of reference 1 for a detailed discussion of the various properties of group delay.) Excess group delay is relatively constant and small below 1kHz and above 4kHz, indicating that the S8LR is time-coherent in those frequency ranges. However, between 1 and 4kHz, excess group delay changes rapidly, rising to 450ms at 2800Hz. This time offset of drivers appears to be caused by the crossover action discussed later. Notice also that tweeter polarity is reversed relative to the woofer so that the system cannot preserve waveform.

CROSSOVER ACTION

The S8LR has two pairs of binding posts for bi-wiring. This allowed me to measure the response of the individual drivers. The result is plotted in *Fig. 4*. The crossover frequency is seen to be 1410Hz.

In the octave below crossover, tweeter response falls 25dB. Above crossover, woofer response falls off at 6dB/octave out to 7kHz. Beyond this point, woofer response falls at 18dB/octave. The slow woofer rolloff causes some interaction between the woofer and tweeter in the 2–7kHz range.

CUMULATIVE SPECTRAL DECAY

The S8LR's cumulative spectral decay (CSD) response is presented in *Fig. 5*. This waterfall plot shows the frequency content of the system response following a sharp impulsive input at time zero. On the CSD plot, frequency increases from left to right and time moves forward from the rear. Each slice represents a 0.05ms increment of time. The total vertical scale covers a dynamic 35dB range.

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

The first time slice in *Fig. 4* (0.00ms) represents the system frequency response. Tweeter high-frequency decay is relatively good; the bulk of its response has decayed away in about 1ms. However, there is a step "glitch" of 2.6dB in tweeter response at 11.8kHz that does not show up in *Fig. 2* due to the 0.1 octave smoothing applied to the data, but it is there. It causes the sharp ridge seen in the tweeter decay beyond 1ms. It will be interesting to see whether this ridge has any sonic imprint. Low-frequency decay is typical for two-way systems, which is neither very good nor very bad.

POLAR RESPONSE

Polar response is examined in *Figs. 6–8*. *Figure 6* is a waterfall plot of horizontal polar response in 10° increments from 60° right (-60°) to 60° left (+60°) when facing the speaker. All off-axis plots are referenced to the on-axis response, which appears as a straight line at 0.00 degrees. Thus, the plotted curves show the change in response as you move off-axis.

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

In *Fig. 6*, frequency response is limited to 15kHz. Otherwise, the sharp response dip at 17kHz produces a very confusing waterfall display. Notice that response falls off rapidly with increasing angle above 5kHz. At 10kHz the –3dB beam width is only ±10°. Contrast this with a typical 28mm dome tweeter that will have a –3dB beam width of ±25° at 15kHz.

It appears that the woofer cone limits off-axis response at higher frequencies. The restricted high-frequency coverage may limit the sweet spot in stereo listening. It will be interesting to see what the listening tests reveal.

The average response over a 60° horizontal window (±30°) in the forward direction is shown in *Fig. 7*. Average response falls rapidly above 10kHz and is down about 6dB rela-

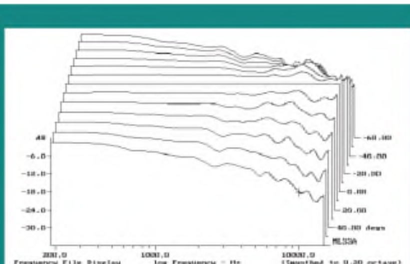


FIGURE 6: Horizontal polar response.

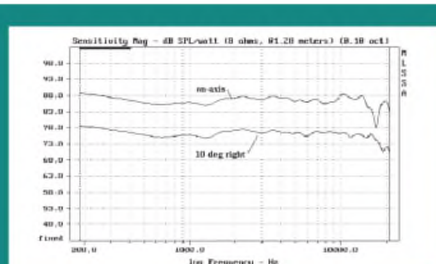


FIGURE 8: Response on-axis and at 10° right.

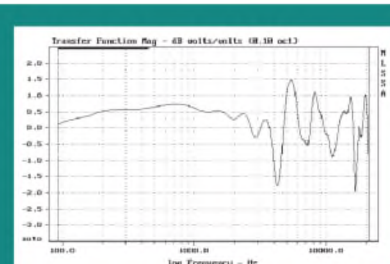


FIGURE 10: Effect of grille on frequency response.

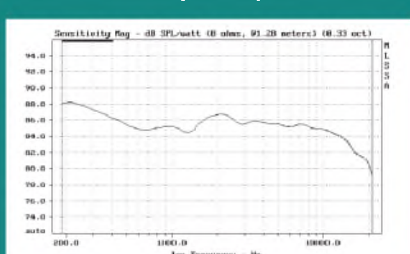


FIGURE 7: Average response over 60° horizontal window.

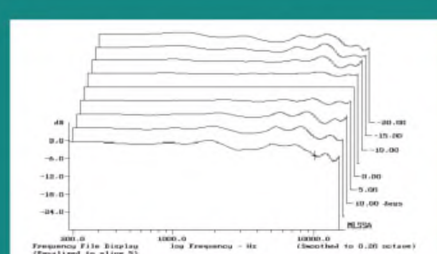


FIGURE 9: Vertical polar response.

■ **Reviewed by Dennis Colin**

With the grilles off and vent damping plugs not used, I first played music from "Carmen" (unknown recording from an FM station). I was immediately impressed with a dramatically natural sense of 3-D spaciousness, with solid image coherence even 45° off the central listening axis. The flute, with its pyrotechnic melody embellishment, sounded extremely natural, as did the piano. (I grew up hearing my father play flute and I played piano.) This recording lacked for nothing.

Next, I tried the *Hi-Fi News and Record Review* Test Disc III CD. In Track 2, the chorus sounded naturally spacious and detailed. I thought the voice timbre was somewhat "midrangey," but this track sounds that way on every system I've played it on. On track 4, the trumpets sounded good, but a little bright.

Tracks 5 and 6—All the "Peter and the Wolf" instruments sounded very natural. Of particular "note" were the horns (lush and spacious) and the drums (very solid and real-sounding). The narrator remained faithful to my impressions on other speakers—that he used a garbage can with a voice coil for a microphone! (I can understand not wanting to rent a forest, but couldn't they afford a better reverb?) I was surprised at the bass clarity and extension (I would guess to 40Hz).

At this point I tried the supplied foam vent plugs, which reduced the bass extension and provided no benefit in my well-damped room, so I took them out. (They could be of use in a room with excessive bass "boom.")

Track 7—Voices were realistic. The harpsichord had very fast, clean, natural transient "bite." I think the Tannoy "Dual Concentric" system (tweeter center coincident with woofer apex) is well-designed: high-frequency transients maintained solid coherence at any position in the room. Absolutely no "phasiness" or image-smearing on- and off-axis.

Track 10—The percussion array (drums and tympani) sounded very real and solid, with no sense of time or space blurring sometimes heard on non-coincident multi-way speakers.

Track 14—While not my kind of music ("space-age"?), I was impressed by the ability of the speakers to keep up with these sonic fireworks. In particular, the bass drum sounded like a bass drum, rather than a pair of small speaker boxes.

I also tried a variety of other music, which sounded very natural in tone and space, with surprisingly deep, clean bass. The only coloration I heard was a slight brightness on some vocals (Barbra Streisand and Linda Ronstadt). It sounded like a mild rise (my guess is 1dB) of the 2kHz area regarding that at 1kHz on down. Similar to the Purcell trumpet on the *HFN* Track 4, the effect was not at all ragged-sounding, but rather like a very small amount of a single gentle resonance. Small enough not to hear on 90% of the selections.

Bass was very natural and deeply extended. Power handling was very good; I could drive the 100W/ch amp to clipping without noticeable speaker distortion. All music—whether classical, jazz, rock, blues, or other—was reproduced with excellent clarity, imaging, ambience, and power.

HIGH-FREQUENCY SMOOTHNESS

Violins are my favorite test. The bow drags the string until it slips, and the cycle repeats, producing a sawtooth waveform. This contains a uniform series of all harmonics to beyond 20kHz, which although shaped by the sound board resonance, also reaches the ear directly from the strings. In addition, the harmonics are sufficiently phase-coherent and evenly distributed to convey the "pulsiness" of the sawtooth waveform, responsible for the rich but velvety "bite."

The Tannoys are one of the very few speakers I've heard that preserve violin tone and detail satisfactorily. Another is the Swans M1 with superb ribbon tweeter (*SB* 3/99). This is still my favorite, but not by much. Both are good enough to allow you to "hear into" the actual instrument; that is, to effortlessly and without distraction picture the original as close enough to touch.

SPACIOUSNESS AND IMAGE COHERENCE

These two characteristics are often assumed to be mutually exclusive. Not so here! Nowhere in the room did the images become de-focused, yet this precision was accompanied by lush spaciousness that sounded "coherent with" the direct sound. That is, the hall ambience sounded seamlessly connected to the direct sounds stimulating it. So much so that when I momentarily switched on some surround speakers, the spatial envelopment naturalness actually decreased!

I'm sure that's because this concentric design radiates as an effective point source with waveform coherence across the whole audio range. Thus all room reflections can be acoustically "ray traced" back to the same source point, maintaining a believable image of both sound sources and 3-D space. You simply must hear a good coincident-driver design such as this Tannoy to realize how solid a reproduced image can be.

COMMENTS ON MEASUREMENT

Frequency response (Joe's *Fig. 2*)—I'm not surprised by the smoothness; that's how they sounded. I didn't hear the elevation around 100Hz—probably because I've measured a mild depression at 100Hz in this room with drivers at ≈3′ off the floor (near the ¼ wave floor-bounce cancelling distance at 100Hz). The rise within 1–2kHz is probably the occasional brightness I heard on the trumpets and some voices.

CSD (*Fig. 5*)—The transparency and detail I heard agree with the good HF decay. The 11.8kHz step glitch and ridge may explain the slight brightness heard on some vocals, and why I rated violin reproduction second to the superb Swan's ribbon tweeter.

Horizontal and vertical dispersion (*Figs. 6* and *9*)—While rolloff is rapid off-axis, it maintains a very smooth, virtually monotonic response (also similar in all directions, due to the concentric symmetry). I think this agrees very well with the superbly solid imaging and maintenance of tonal neutrality regardless of listening position.

Ten degrees off-axis (*Fig. 8*)—Other concentric systems I've measured also have a HF dip that disappears off-axis. The 17kHz dip here is above my hearing (≈15kHz), but those other systems had this waveguide dip around 12kHz. The audible effect, though, was very difficult to detect. At 17kHz, I doubt it will be noticed on the Tannoys.

Distortion—I needed to drive the 100W/channel amp near clipping to notice any distortion, and I can't say it wasn't the amp. Even at fairly loud levels, the speakers sounded unstrained and transparent. It is no surprise that Joe reported excellent low-distortion performance. These speakers can play "loud and clear." I probably reached 107dB pushing the 2 × 100W

ABOUT THE AUTHOR

Dennis P. Colin graduated with a BSEE from the University of Lowell (MA) and is currently an Analog Circuit Design Consultant for microwave radios. Previously a keyboardist and a recording engineer, he has been published in the *Journal* of the Audio Engineering Society. He has demonstrated the audibility of phase distortion at Boston Audio Society, and has designed the "Omni-Focus" speaker (bipolar coincidental with phase-linear first-order crossover), ARP 2600 analog music synthesizer, 1kW bi-amp and PWM supply at A/D/S, and Class D amps.

amp, and my ears probably distorted more than the speakers.

Crossover (*Fig. 4*)—Joe mentioned the slow 6dB/octave woofer rolloff up to 7kHz. Notice that the woofer bump at 2kHz is only 12dB below the summed response. I wonder whether this causes the mild step in *Fig. 2* that I may have been hearing on a few sources.

Joe also pointed out the unusual arrival of woofer before tweeter in *Fig. 3* step response. Well, let me tell you how useful this is in achieving full-band waveform coherence—all crossovers (except extremely complicated ones) delay bass more than treble (second- and higher-order low-pass and high-pass filters do this). So with the unusual greater tweeter delay in this speaker, it would be easy to add woofer delay so the driver arrivals are synchronous, resulting in a near-ideal transient response.

Actually, the Tannoy's inter-driver synchrony is within 0.2ms, which is probably inaudible. But my point is that I would steepen the woofer LP slope, uninvert the tweeter, and tweak for a month. I would hope to eliminate possibly audible woofer resonances and achieve perfectionistic time coherence as a side effect. I know, we *audioXpress* people always think we can improve anything!

CONCLUSION

This is a very good speaker—effortless, transparent, and very clear-sounding. Also very powerful for its size. And the imaging is, I would dare say, audibly perfect—this concentric design, superbly engineered, produces an acoustic image that has absolutely no, as in zero, smearing, wandering, de-focusing, or other spatial distortion. The Tannoy S8LR, with good source material, reproduces a sound field that is holographic in its 3-D lucidity.

EQUIPMENT AND SETUP

I used the same Nakamichi AV-1 receiver (100W/channel) and Yamaha CDC 755 CD changer (plus turntable and cassette player) that I've become very familiar with after three years, and on which I've heard many speakers—some very good and some not.

LISTENING ROOM

Approximately 20′ × 18′ × 8½′ (3000ft^3), the room is moderately damped with stuffed chairs, carpet, and drapes. It is well-dispersed by numerous openings and stepped walls. Room response is smooth (for a room) to below 16Hz. Many other speakers sound excellent in this room, including the Swans M1 (*SB* 3/99, p. 36).

I placed the speakers on stands with tweeters at seated ear height (≈36″), 3′ from the front wall and 4½′ from side walls (11′ apart); the distance to listeners was approximately 12′.

SOURCE MATERIAL

I used the *Hi-Fi News and Record Review* Test Disc III CD (tracks 2, 4, 5, 6, 7, 10, 14), and also played a variety of other material, including music from "Carmen" (flute and piano), Stevie Ray Vaughn, Julio Iglesias ("Tango"), Barbra Streisand, Linda Ronstadt, and "The Blue Danube" on LP (Ormandy/Phila. Orch.).

BREAK-IN

I played the speakers loudly for one hour. I heard none of the sometimes referred-to "roughness" even out of the boxes, and no difference after the break-in hour.

tive to mid-frequencies at 20kHz. Most two-way systems I have tested in the past show fairly uniform average response out to 20kHz.

Figure 8 compares on-axis response with response at 10° off-axis to the right. Notice that the sharp dip at 17kHz is gone at 10°. This is further confirmation of a diffraction-induced interaction between the tweeter and the woofer cone.

Figure 9 presents a waterfall plot of vertical polar response over a range of ±20° in 5° increments. The concentric mounting of the woofer and tweeter produces a plot similar to that of *Fig. 8* over the same angular range.

HARMONIC DISTORTION

I ran harmonic distortion tests at an average level of 90dB SPL. Ideally, harmonic distortion tests should be run in an anechoic environment. In practice, it is important to minimize reflections at the microphone during these tests. Out-of-phase reflections can produce false readings by reducing the level of the fundamental while boosting the amplitude of a harmonic. In order to reduce the impact of reflections, I placed the microphone at 0.5m from the loudspeaker.

With the port open, second- and third-harmonic distortions at 50Hz were 1.8% and 2.4%, respectively—a rather good result. All HD distortion was below 1% above 110Hz, which is also a good result.

With the port closed, second-harmonic distortion at 50Hz doubled to 3.6%, and third-harmonic distortion was unchanged. The reflex action produced with the port reduces second-harmonic distortion, but has little effect on the third-harmonic. At 150Hz the port has little effect and the woofer behaves as though it is in a closed box.

INTERMODULATION DISTORTION

Next I measured intermodulation distortion. In this test two frequencies are input to the speaker. Intermodulation distortion produces output frequencies that are not harmonically related to the input. These frequencies are much more audible and annoying than harmonic distortion. Let the symbols f_1 and f_2 represent the two frequencies used in the test. Then a second-order nonlinearity will produce intermods at frequencies of $f_1 \pm f_2$. A third-order nonlinearity generates intermods at $2f_1 \pm f_2$ and $f_1 \pm 2f_2$.

I first examined woofer intermods by inputting 400Hz and 500Hz signals at equal levels. These frequencies should appear predominantly in the woofer output. I adjusted total SPL with the two signals to 87dB at 1m. Because steady tones are used in the IM test, I believed it was safer to use a lower power level to prevent possible tweeter damage. Principal woofer IM products occurred at 900, 1300, 1400Hz. However, the overall level was only 0.33%, which is an excellent result.

I measured tweeter intermods with a 10kHz and 11kHz input pair also adjusted to produce 87dB SPL at 1m. I observed IM products at 9 and 12kHz. Total distortion was 0.06%, which again, is a very low figure.

The last IM test examines cross-intermodulation distortion between the woofer and tweeter using frequencies of 400Hz and 10kHz. Ideally, the crossover should prevent high-frequency energy from entering the woofer and low-frequency energy from entering the tweeter. IM products appeared at 8.4, 9.2, 9.6, and 10.8kHz at a level of 0.03%, the lowest figure I have measured so far in this series of tests for *audioXpress.*

ADDITIONAL TESTS

I conducted all of the above tests with the grille off. *Figure 10* shows the S8LR's response with the grille on, but referenced to the response with grille off. That is, it plots the change in response under the two conditions.

The grille has little effect below 3kHz. Above 3kHz, however, the grille causes ragged response deviations of +1.5 to −2dB. Although the plot looks bad, this is still a factor of two better than most other speakers I have tested. The perfect grille is still the Holy Grail of speaker design!

Two samples of the S8LR system were available for testing. All of the tests described so far were conducted on one sample. Frequency response of the second sample matched the first to within ±0.2 below 2.5kHz. Above this frequency, matching degraded to ±2dB. Part of the problem here may be due to the difficulty in determining the on-axis position exactly coupled with the restricted high-frequency polar coverage of the S8LR.

A NOTE ON TESTING

The Tannoy S8LR was tested in the laboratories of Audio and Acoustics, Ltd. using the MLSSA and CLIO PC-based acoustic data acquisition and analysis systems. Acoustic data was measured with an ACO 7012 ½″ laboratory grade condenser microphone and a custom-designed wideband, low-noise preamp. Polar response tests were performed with a computer-controlled OUTLINE turntable on loan from the Old Colony Division of the Audio Amateur Corporation. ❖

REFERENCE

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

consumeR PRODUCTS REVIEWED

Blaupunkt München RDM 169 Car Audio Head Unit

Reviewed by Charles Hansen and Edward T. Dell, Jr.

PHOTO 1: RDM 169 front view.

Blaupunkt, Robert Bosch Corp., 2800 South 25th Ave., Broadview, IL 60155, 708-865-5654, www.blaupunkt.com. Dimensions: single DIN, 7.325″W × 7.5″D × 2.25″H. Price: $480 US. Warranty: 12 months.

The Blaupunkt München RDM 169 is a single-DIN head unit with AM/FM tuner/CD/CD changer capabilities. "München" is the German word for what we call Munich, and Blaupunkt means "blue point." Interestingly, the München was manufactured in Portugal, rather than Germany or the Far-East.

The faceplate (*Photo 1*), with its large multi-color display, flips open for access to the CD slot and Blaupunkt's key card security system. A clock display is included as one of its features. The radio is also supplied with a steering-wheel-mounted infrared (IR) remote control.

FEATURES

The München provides four preamp outputs, a line-level subwoofer output, internal amps rated at 4 × 40W, and digital parametric equalization (EQ). Two line inputs accept signals from a compatible remote-mounted CD changer, and the head unit provides bus control input/output and power connections to the changer. The volume-control display indication varies from "0" to "66," and a triangle-shaped horizontal bar graph displays the dynamic audio level.

You can save your own custom EQ settings in the user mode. You access the low- and high-frequency digital parametric EQ with the DPE button on the front face. These DSP filters have a variable center frequency, boost/cut, and Q (bandwidth) adjustments. The low filter is variable from 32Hz to 500Hz in 13 steps (⅓ octave). The high filter is variable from 630Hz to 10kHz, also in ⅓-octave steps.

You can boost or cut the response over a display-indicated range of "+8" to "−8." These parameters change just over 1dB per step. You can vary the filter quality (Q) in three steps, with bandwidths from two octaves down to ¾ octave.

In addition to the user-adjustable EQ settings, there are six factory-presets: Linear, Rock, Disco, Jazz, Classic, and Vocal. The manual includes a page with tips on how to adjust the EQ. If you thoroughly muck up the sound, pressing the DPE button for two seconds will deactivate the EQ function.

The München's user interface is quite extensive, but fairly easy once you master the operating instructions. The volume control is a large rotary knob, not one of those hard-to-see/use sets of up/down buttons on some car radios. A button in the center of the volume control turns the power on ("WELCOME"), puts the unit in mute, and turns it off ("GOODBYE"); and its icy "blue point" glows when the power is on.

You select the tuner, CD, and CD changer by individual push-buttons under the volume control. You can program CD tracks (and CDs with the external changer) in any selected or random order. You can assign names to up to 30 CDs (99 CDs in changer mode). The internal CD slot is lighted with a red bar and securely holds the CD during insertion and ejection so it won't fly out of the München while you are cornering your BMW at 9⁄10ths.

A round four-way rocker on the right side and two groups of four vertical menu-identified soft keys handle the other functions. These become station selector buttons in the tuner mode. FM stations are automatically identified with their call letters using RDS[1], and you can assign your favorite stations one of 23 program types (News, Jazz, Classical, and so on) via the PTY function.

The display is quite readable and offers viewing-angle and brightness adjustments. However, I would not try to operate the menu-driven functions while driving—they require too much attention to the radio display.

You can operate the unit with the ignition off via the "permanent" +12V DC connection. In this mode, the unit shuts down automatically after one hour to prevent battery drain.

INSIDE THE AMPLIFIER

The head unit, which is extremely rugged, is constructed of heavy gauge steel, and the top cover is perforated to enhance cooling of the power amplifier module. A heavy mounting bracket holds the unit securely in place.

The rear panel (not shown) has five receptacles, a grounding stud, antenna jack, and a plug-in 10A fuse. Connector wiring harnesses are supplied for power and ground, adjustable illumination, cellphone mute, antenna motor control, speakers, line outputs, and subwoofer output. The four line output "flying lead" RCA connectors are all gold-plated. Additional receptacle pins are provided for the remote CD changer, so you will need a compatible plug and wiring harness.

The installation instructions are very rudimentary. Unless you are fairly proficient in auto radio in-

TABLE 1
MEASURED PERFORMANCE

PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Line-out level	3.2V, 1% THD+N 4.8V maximum	3.8V, 1% THD+N 4.8V maximum
Output impedance	100Ω	100Ω
Frequency response	±1dB, 20–20kHz	±0.7dB, 8–20kHz
Usable dynamic range	–90dB	
Subwoofer out level	3V, 1% THD+N	3.1V, 1% THD+N
Internal amplifier levels	4 × 25W, 1% THD+N 4 × 40W maximum	4 × 17.5W, 1% THD, 4Ω 4 × 27W maximum, 4Ω
IMD 11kHz + 12kHz	N/S	0.022% 1kHz product
Signal-to-noise	–90dB	–78dB (see text)
Channel error	0.25dB	0.12dB
Frequency response	20–20kHz	±1dB, 8–20kHz
FM mono sensitivity	0.7µV at S/N ratio of 26dB	<10dBf
Usable FM sensitivity	7.8dBf	<10dBf
FM stereo sensitivity	N/S	21dBf
FM frequency response	20–16kHz	±3dB, 20–11kHz
FM stereo separation	N/S	35dB

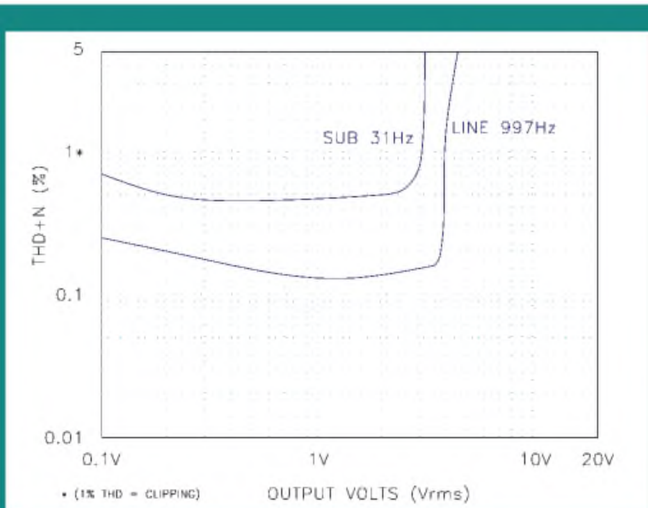


FIGURE 1: Preamp THD+N versus output voltage.

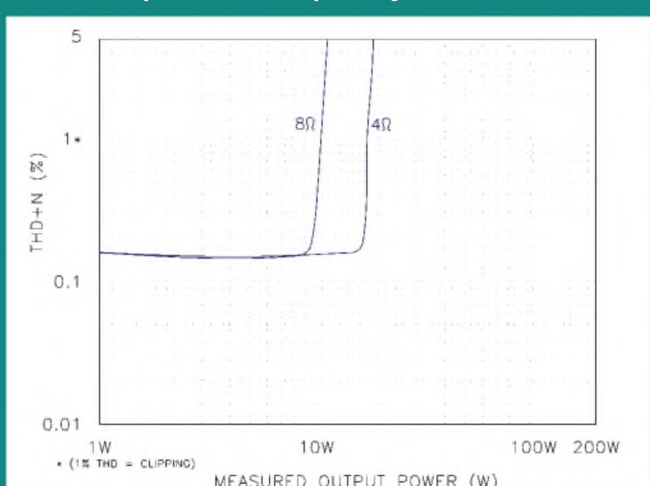


FIGURE 2: Amplifier THD+N versus output power.

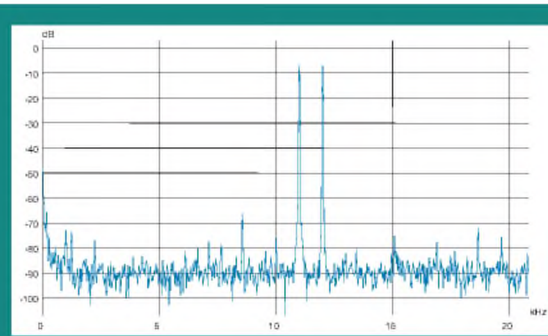


FIGURE 3: Spectrum of 11kHz + 12kHz intermodulation signal.

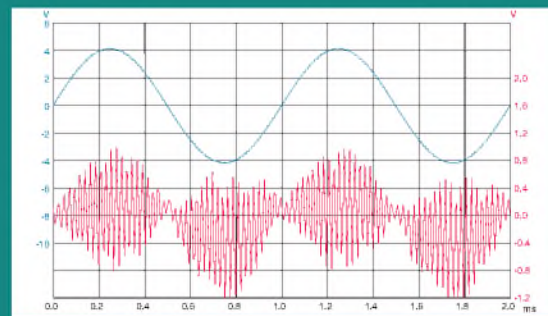


FIGURE 4: Residual distortion.

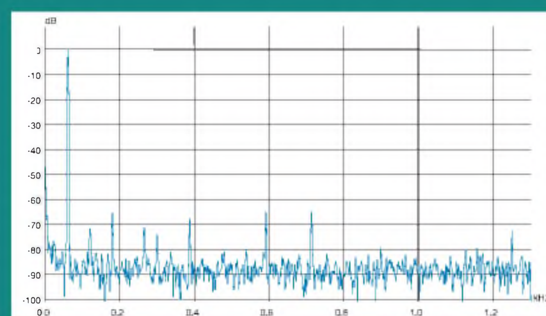


FIGURE 5: Spectrum of 61Hz sine wave.

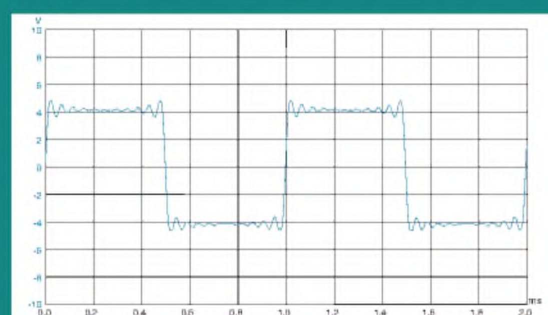


FIGURE 6: 997Hz square wave response.

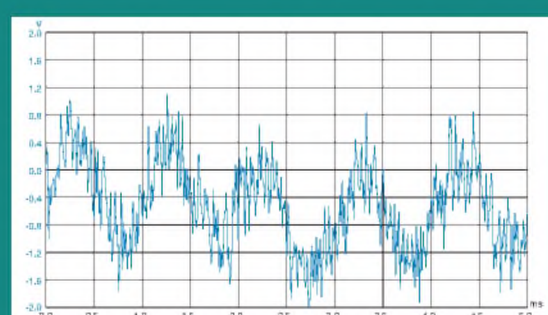


FIGURE 7: Undithered 1kHz sine wave at –59.94dBFS.

■ **Reviewed by Edward T. Dell, Jr.**

The problems associated with reviewing an automotive tuner/amplifier seemed daunting, to say the least. However, since the manufacturer's rep was offering, and our versatile test magician was eager to try something quite different, I agreed to try to review Blaupunkt's München RDM 169 US Radio/CD unit for automobiles.

Although Chuck Hansen believed the installation instructions were rather slim, I thought they seemed doable—provided you could figure out how to get the old unit out of the dash. Apparently these units are fairly uniform in size and fasten in place with one large captive machine screw in the center of the back plate. I opted to have the unit installed locally by a pro. Prices for this range from $80 to $100.

If you plan on replacing your unit, among other things, I strongly commend what is probably the best mail-order supplier in the autosound business, Crutchfield (1 Crutchfield Park, Charlottesville, VA 22911-9097, www.crutchfield.com, 800-955-3000). This company believes in do-it-yourself and offers very impressive help with any product they sell. Their catalog has vital information on product specifications and automobile data. They have a technical service to guide you to the right decisions about upgrading the sound in what you drive.

TEST DRIVE

I drive a Mazda '97 626 V6 with a stock sound system consisting of four 6 × 8 coax drivers: two in the doors and two on the back deck, with two 1″ tweeters in the dash, pointed at the windscreen. The head unit is an FM/AM radio with cassette tape, so I had no way to compare the recorded sound in the players of the two units directly, but I did know what stations I was able to receive on the old unit.

The RDM 169 is a digital tuner, with an integral CD player, one of the first whose special capability is its sensitivity and its ability to lock onto a weaker station successfully. It also has the capability of separating closely spaced stations for clear reception. In my area, we are 750′ above sea level and are at the end of New Hampshire's White Mountain chain, and 90 miles from Boston's wide range of stations. Reception here is not easy, but this tuner doubled the number of stations I am able to receive.

The 33 pages of instructions are not easy to master quickly. The flexibility of the unit is the reason. It has two parametric equalizers: one covers 32–500Hz, the other 630Hz to 10kHz, with ±8 steps for each. These are useful for adapting the system to your car's acoustics. The system stores 12 FM and AM stations each in presets, and is equipped for RDX, the new system responding to station programming content in 23 categories, as well as automatic clock setting. I cannot imagine anyone finding the München unit deficient in programmability.

The CD player operates smoothly when you open the faceplate and insert a disk. There are facilities for adding a remote ten-disk player that you can program from the head unit. Opening the faceplate, you also see a data card containing the programming preferences of the driver. Remove it on leaving the car and the unit will not function, a deterrent for thieves. A second card is provided for another driver, allowing alternative programming.

The unit also comes with an infrared thumb control mounted on the steering wheel. The small button receiver is installed on the dashboard. This unit allows the driver to change or search for stations and volume level and to mute the system, without looking away from the road. Facilities are available for automatically muting the system while using a telephone. The unit also has preamp outputs for five channels of separate amps and speakers. These are only a few of the really impressive range of features.

LISTENING IMPRESSIONS

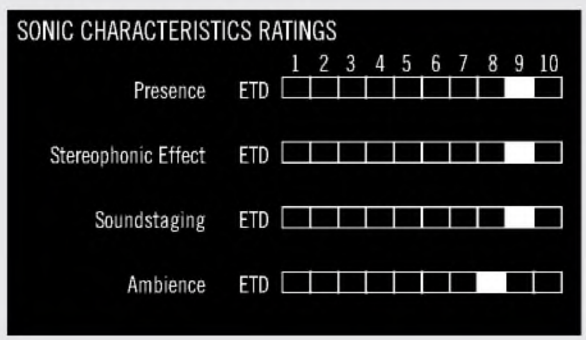
Sound quality must be evaluated using the speaker drivers already in the car, of course. I listened carefully to our standard set of test tracks from the *Hi-Fi News* Test CD III on my house system, which is tri-amplified into Audio Concepts Sapphire 3/Sub 1 bandpass subs, using a much modified Magnavox player with Parts Connection's Assemblage DAC 2.1. My impression is that although the car is a very different environment, and that the stock speaker drivers are probably a significant limitation, the material sounds remarkably similar, despite the confined compartment.

Both the Handel and Parry of the first two tracks recorded out of doors almost convince you the car has become a convertible. The trumpets in the Vivaldi Concerto are just a bit bright with the system at factory settings. You can tame this by adjusting the equalizer down one step. Both of the Peter and the Wolf segments were very crisp and detailed, with excellent articulation of the instruments. The Purcell had a remarkable presence which I rarely hear in broadcast material in my car.

Although the bass response in my auto has always seemed limited, it certainly was not in any one of the Corkhill examples, and in No. 13 the upper registers sounded like water droplets on velvet. The same was true for the Rio Napo selection, where the rich, wide range sound almost feels like it might soak into your skin.

The unit, in addition to all else, looks rather like an elegant Christmas display and even beeps a "goodbye" when the ignition key turns things off. The sensitivity is outstanding and the sound quality bears a very respectable comparison to a good high-end system. Unfortunately, it makes you wonder seriously whether new speakers would make it even better, and...yes there's also the temptation of a changer in the trunk.

If you really want a serious upgrade for your automobile's sound system with elegant and effective sound management, amazing convenience and marvelous sound, the München RDM 169 is a prime choice.



stallation, you should leave this to the experts. Be sure not to lose the right-angle antenna adapter that plugs into the 10mm jack on the radio and adapts to the conventional US automotive antenna plug.

TOPOLOGY

A schematic was not provided. I could see many surface-mount components on the main PC board, visible through the cooling holes. The four internal power amplifiers appear to be contained in a single large metal, heat-sunk, power-integrated circuit.

High-quality auto radios such as the Blaupunkt usually include an RF amplifier stage prior to the tuner section and high-quality automatic volume control (AVC) circuitry. This helps compensate for the continuously varying signal strength caused by constantly changing antenna orientation and RF obstructions.

MEASUREMENTS

I tried to input my distortion test-set oscillator to the preamp via the CD changer line inputs, but I could not fool the controller into believing the changer was present. Thus, I made all my measurements, except for the tuner, using a number of CD test discs in the internal CD player. There was a very low level of hiss with my ear against the speaker, but the München was otherwise quiet (not that this is of primary importance, since the automotive background noise level generally exceeds 65dB).

There was no noise during power-up or shutdown. The unit draws 1.1A DC at 14.4V DC in mute mode, and de-energizes itself below 9.1V DC. Surprisingly, the CD player/amplifier inverts polarity. The CD changer line-input impedance was 10k3 for both inputs. The line-output impedance measured a low 100Ω, and sub out measured 330Ω.

The line-output frequency response was within ±0.7dB from 8Hz to 20kHz, at 0.5V RMS. THD versus frequency increased from a low of 0.031% at 17Hz to 0.82% at 20kHz, using a line-output level of approximately 2V RMS.

Hum and noise (maximum volume, no source selected) measured −78dB. This reading was limited by the internal amp/preamp, which proved sensitive to the very low 120Hz ripple in my test bench 12V DC power supply. It has been my experience that car radio amplifier ICs have fairly low (−50dB) power-supply rejection ratios (PSRR), but that is usually not a problem when using battery power.

The line-output THD+N versus output voltage at 1kHz is shown in *Fig. 1*. Since I was using the internal CD player as the signal source, I engaged my test-set 27kHz low-pass filter to limit the out-of-band noise. Maximum output voltage at 1% THD was approximately 3.8V RMS (resolution was limited by the volume-control steps). The preamp clipped at ±11V p-p with the volume control at "66," while playing the 997Hz 0dBFS CD test track.

The sub frequency response was down −3dB at 4Hz and 61Hz, and −12dB at 127Hz. The sub THD+N versus frequency was higher than the line output, measuring from 0.37% at 17Hz to 2.6% at 127Hz, all at 1V RMS output. Sub THD+N versus output voltage at 31Hz is also shown in *Fig. 1*. The output did not clip at its upper limit. The second harmonic increased above 3.1V RMS starting at the lower half-cycle of the waveform, until it eventually replaced the fundamental.

CD TESTING

Volume-control tracking was excellent, varying less than 0.12dB between four channels, from 0.2W to 10W output. Unity gain occurred at about "40" on the display indicator.

I checked the maximum range of the parametric equalizer at three frequencies, using the default Q level. At 100Hz and 1kHz, the output was ±8dB, exactly as indicated on the display. The high EQ had a wider range of ±9.8dB at 10kHz. Neither the 100Hz nor the 10kHz frequency showed more than 0.1dB variation at 1kHz.

Even so, if you attempt to use the upper end of the LF EQ with the lower end of the HF EQ, there will be some overlap interaction between the ⅓ octave frequencies, even at the tightest Q setting. The LF 500Hz and HF 630Hz responses (set for +8dB) were still up at +5.5dB at 630Hz and 500Hz, respectively.

Assuming the München CD section meets the Red Book spec of 2V RMS at 0dBFS, the overall gain at 2.83V RMS output into 8Ω loads is 17.6dB. The power amplifier uses bridge-connected output stages whose 0.25Ω output impedance at 1kHz was quite respectable for an integrated circuit amplifier. It increased to 0.31Ω at 20kHz.

The amplifier frequency response measured ±1dB from 8Hz to 16kHz at an output of 2.83V RMS into both 4Ω and 8Ω. It dropped to −1.9dB at 20kHz. There was no gain peaking when I connected a load of 8Ω paralleled with a 2µF cap. Likewise, the amplifier was unfazed by the complex impedance of an IHF simulated speaker load.

THD+N versus frequency with 1W into an 8Ω load ran from 0.045% at 17Hz to 0.86% at 20kHz. *Figure 2* shows THD+N versus output power into 4Ω and 8Ω at 1kHz, again using the test-set 27kHz low-pass filter to limit the out of band noise. The München showed absolutely no strain right up to the point of maximum power.

The power amp clipped at ±25V p-p, with the volume control at "52" while playing the 997Hz 0dBFS CD test track. The 1% THD point occurred at 17.5W into 4Ω, and 9.2W into 8Ω. Maximum power, at greater than 20% THD, was 27.5W into 4Ω and 16.5W into 8Ω.

HOT TUNES

The left side of the radio was hot after this testing. In many cases the vehicle dash will provide some additional heatsinking. The München drew 6.9A at 14.4V DC (almost 100W) with the CD operating and all four channels putting out 10W at 1kHz. Clearly the Blaupunkt-specified 4 × 25W at 1% THD and 4 × 40W maximum are optimistic (by about 1.6dBW). It may be the car audio industry doesn't need to adhere to the FTC power rating rules that apply to commercial home hi-fi equipment.

Figure 3 shows the amplifier output spectrum reproducing a combined 11kHz + 12kHz SMPTE intermodulation distortion (IMD) signal at 1W into 8Ω. The 1kHz IMD product is 0.022%. Crosstalk performance wasn't bad for such a compact unit with four output channels. It measured −72dB at 1kHz and −62dB at 16kHz.

The distortion waveform for 1W into 8Ω at 997Hz is shown in *Fig. 4*. The upper waveform is the amplifier output signal, and the lower waveform is the monitor output (after the THD test-set notch filter), not to scale. Even with the test-set 27kHz low-pass filter engaged, the 44.1kHz digital sampling frequency and noise dominate the 0.21% distortion residual signal. I should mention here that this distortion tester does not have the four-pole 22kHz LP filter available in modern DSP-based testers.

The spectrum of a 61Hz CD-generated sine wave at 1W into 8Ω is shown in *Fig. 5*, from 0 to 1.3kHz. The THD+N measured 0.059%, and the harmonics are distributed throughout the spectrum. The second, third, fourth, and fifth measure −72dB, −65dB, −83dB, and −84dB, respectively. However, additional nonharmonic responses occur at 387Hz (−69dB), 591Hz (−65dB), and 713Hz (−65dB).

The 8V p-p square wave into 8Ω at 997Hz (*Fig. 6*) exhibits the Gibbs Phenomenon ringing associated with the steep digital filters used in the DSP section of the receiver. For the same reason, the 10kHz square wave was rounded over into nearly a sine-wave shape.

When I tried to capture reproduction of an undithered 1kHz sine wave at −90.31dBFS, I got nothing but noise on the scope. At this level the signal consists of ±1 bit of data, producing two different voltage levels that are symmetrical about the horizontal axis (time). As a result, *Fig. 7* shows an undithered 1kHz sine wave at −59.94dBFS. Even here, the sine wave is dominated by out-of-band high-frequency noise, which is consistent with the residual distortion in *Fig. 4*.

The München ignored defects on the Pierre Verany Test CD #2 out to track 37, which has a 3mm-long section of blank data. At the last defect test, track 38 (a cavernous 4mm, 3.08ms defect), the unit put out one or two audible clicks over the 500Hz test tone, but for the most part dealt with it well. The Red Book requirement is only 0.2mm maximum. Mobile operation imposes much greater opportunity for mechanically induced skips, so this defect margin is definitely appreciated.

FM TUNER TESTS

My FM stereo signal generator is tuned to 98.1MHz because there is no nearby station on this channel. A few of the more sensitive FM tuners can pull in one distant station (85 miles) in mono at this frequency. The München could do this if it was manually tuned; however, its station seek function passes right over 98.1MHz with a whip antenna.

This is a reasonable default performance for a car radio auto-tune circuit. It would undoubtedly do better in a vehicle in which the sheet-metal body functions as an image reflector for the ¼λ whip. You can also change the sensitivity of the seek run or activate automatic bandwidth switching (SHARX) to accept stations with poorer reception.

Frequency response measured ±3dB from 20Hz to 11kHz. The HF response dropped rapidly above 11kHz and was down −17dB at 16kHz. As with the CD player, the FM tuner inverted polarity (audio signal to the FM modulator compared to the line/speaker output of the radio).

Stereo sensitivity was approximately 21dBf, using a 75Ω RF signal at the antenna input adapter. I can reduce the output of my signal generator to 10dBf using the full range of an external 0–70dB RF attenuator. The mono sensitivity of the München was below this level. Note that Blaupunkt lists this sensitivity as 0.7µV at a rather low 26dB (the IHF level is 50dB) without specifying impedance. A signal of 0.7µV is equal to 8.2dBf at 75Ω.

Stereo separation at 1kHz was 35dB. While you can toggle between mono and stereo in DCS mode, you cannot defeat the mono-blend circuit.

The München FM tuner has reasonable trade-offs in stereo performance, frequency response, selectivity, and sensitivity given the compromised listening environment of an automobile. ❖

REFERENCES

1. RDS is the radio data system that allows text or other information to piggyback on the standard FM radio signal. It uses a 57kHz subcarrier (triple the 19kHz stereo pilot signal) that is divided by 48 in the receiver to recover 1.1875kHz NRZ (non-return to zero) digital data.

Product Review

Monarchy SM-70 Pro Power Amplifier

Reviewed by Gary Galo

Monarchy Audio SM-70 Pro Power Amplifier, DIP 24/96 Jitter Suppressor, and DR-1 Digital Interconnect Cable. Monarchy Audio, 380 Swift Ave., #21, South San Francisco, CA 94080, 650-873-3055, FAX 650-588-0335, monarchy@earthlink.net, www.monarchyaudio.com. Prices: SM-70 Pro—$980, DIP 24/96—$249.99, DR-1 Cable—$239.

In *Audio Electronics* 5/00, I reviewed the first power amplifiers from Monarchy Audio, the SM-70 and SE-100 Delux. Monarchy has been working steadily to improve their power amp designs, and has recently introduced an upgraded version of the SM-70. Also new from Monarchy is an improved version of their high-value Digital Interface Processor clock jitter suppressor.

SM-70 PRO

The SM-70 Pro is the latest version of Monarchy's "zero-feedback" power amplifier design (*Photos 1* and *2*). The new "Pro" version is very similar in design to the original SM-70, with a couple of important changes. The number of output MOSFETs has been doubled—there are now four per channel in a push-pull parallel arrangement. To accommodate the additional heat, Monarchy has increased the chassis and heatsink size. The SM-70 Pro is now the same size as the SE-100 Deluxe.

The power supply has been improved in three ways. First, the SM-70 Pro has a larger toroidal power transformer than its predecessor, raising the raw supply rails to ±32V DC. Second, Monarchy supplies each amp with a pair of low-noise Shindengen D6SB60L rectifier bridges (the original amp had only one rectifier bridge). These are the same rectifiers I described in my *AE* 5/2000 article, "Upgrading Monarchy's Converters and Amplifiers." Finally, power-supply filter capacitance has

PHOTO 1: The SM-70 Pro power amp, front and rear. Like its predecessor, this stereo power amp performs best when used as a balanced-input monoblock.

PHOTO 2: Inside the SM-70 Pro. A pair of low-noise Shindengen D6SB60L rectifier bridges are standard equipment in the new amp.

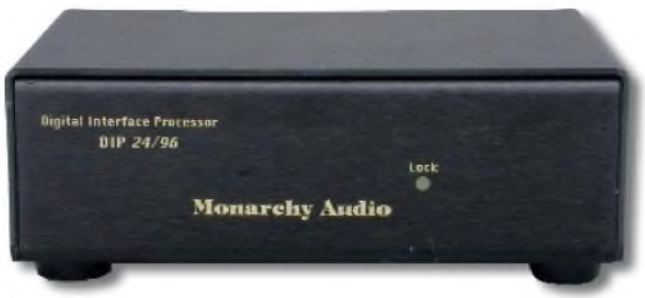


PHOTO 3: Front view of the DIP 24/96. The new DIP chassis includes a digital lock light on the front panel.

PHOTO 4: Rear view of the DIP 24/96. Both coax and Toslink optical inputs are included, along with coax S/PDIF and AES/EBU outputs.

been increased to a total of 60mF (millifarads); the original had 40.

Monarchy originally used one half of a Burr-Brown OPA2604 op amp to supply the voltage gain for each amplifier. The Pro version uses two OPA2604 chips, one per stereo channel. One half of the op amp is used for voltage gain, and the other half is used as a unity-gain buffer/driver for the MOSFET output stage. The DC supply rails for the OPA2406 op amp have been regulated at ±24V DC with an LM317/337 pair.

In my 5/00 review I noted that the SM-70 failed to deliver its rated power of 25W/channel stereo and 75W in the bridged mono mode. The new SM-70 Pro amps are rated at 25W/channel stereo and 80W bridged mono into an 8Ω load; mono output into 4Ω is rated at 120W. I was able to verify the manufacturer's power output ratings for both of the SM-70 Pro review samples. Monarchy's doubling of the output transistors, along with the addition of the unity gain buffer, allows the new amps to make their rated power.

The op amps are configured with conventional feedback techniques. The "zero-feedback" designation refers to the lack of global feedback from MOSFET output stage back to the input stage. The op-amp gain stages and buffers use conventional feedback techniques.

Like the original SM-70, the SM-70 Pro has rather high voltage gain. In the stereo mode, each channel has a voltage gain of 28.25, or 29dB. As bridged monoblocks, the voltage gain is 53.75, or 34.6dB.

The high gain makes these amplifiers suitable for use with passive "preamps." However, the use of a passive preamp will probably preclude operation of the SM-70 Pro amplifiers in the fully-balanced mode, which is where they perform at their best. The new amplifiers are extremely well-constructed, and weigh in at a chunky 24 lbs each.

I did all of my listening to the pair of SM-70 Pro review samples operated as monoblocks using both balanced and unbalanced sources. The SM-70 Pro amplifiers are superb sonic performers. Sonically, they are remarkably similar to the original SM-70 amps, and are among the most musical amplifiers I have heard.

Excellent inner detail has always been a hallmark of Monarchy products, and the SM-70 Pro is no exception. The famous Monarchy "layering" and excellent delineation of instruments within a complex orchestral fabric is ever-present in these new amps. Like the SM-70, the Pro version is somewhat euphonic, with a warm and slightly sweet sonic presentation, and a slightly richer-than-life harmonic palette. The heftier output stage and power supply have resulted in improved dynamic contrasts, and a punchier low end than the original SM-70. Overall, the changes made in the SM-70 design have been extremely worthwhile.

The SM-70 Pro offers the best sonic performance when fed from a fully balanced source, such as Monarchy's Model 22B or Model 33 D/A converters. As before, the amplifiers take on an extra measure of transparency in the fully balanced mono mode. Sonically the SM-70 Pro amplifiers offer extremely high value, and will compete favorably with many of the megabucks amps currently on the market.

The SM-70 Pro comes with a grounded power cord but, like its predecessor, the amp hums unless the ground pin is lifted, even with no input connected. The original SM-70s were silent with the ground pin lifted, but the new amps still have a trace of background hum even with the ground pin lifted, though the hum is lower in level without the ground pin. The hum remains with or

PHOTO 5: Inside the DIP 24/96. The heart of the new DIP is the Cirrus Logic CS8427 digital audio interface transceiver.

PHOTO 6: Monarchy DR-1 75Ω interconnect cable. The DR-1 is an extremely stiff cable with a solid Teflon® dielectric.

without an input connected, and is essentially at the same level with balanced or unbalanced inputs connected.

With my Audio Concepts Sapphire III/Sub-1 loudspeaker system, I needed to have my ear within a few inches of the mid-bass driver in order to hear the hum. At normal listening distances, the hum was not audible. These amps still appear to have a grounding anomaly that hasn't been totally worked out. I don't recommend the SM-70 Pro amps for use with high-sensitivity loudspeakers. This is unfortunate, since these low-power amplifiers would otherwise seem well-suited to high-sensitivity speakers.

DIP 96/24

The DIP 96/24 is the latest product in Monarchy's series of Digital Interface Processors for suppression of clock jitter (*Photos 3, 4* and *5*). The heart of the DIP 96/24 is the Cirrus Logic/Crystal Semiconductor CS8427 Digital Audio Interface Transceiver. Previous input receiver/re-clocking chips from Cirrus, such as the CS8412, did not incorporate a transmitter chip (the CS8402 transmitter was the chip normally used in conjunction with the CS8412). The new chip combines both transmitter and receiver functions in a 24-pin SOIC package.

The CS8427 also operates at 96kHz sampling rates, and at bit rates as high as 24-bit. You can custom-tailor the phase-locked loop (PLL) in the CS8427 to offer optimum performance over a wide range of sampling rates—as low as 8Hz. However, jitter performance is reduced if the PLL is designed to accommodate extremely low sampling rates.

Cirrus Logic's data sheet includes a chart specifying external PLL component values for an 8–96kHz range, and a 32–96kHz range. Monarchy has custom-tailored the PLL component values for sampling rates between the CD standard of 44.1kHz and 96kHz, improving jitter performance even further (Monarchy believes that the audiophile community is not likely to use this product at sampling rates lower than 44.1kHz). The DIP 96/24 is fully compatible with variable-pitch CD transports, and has no problem maintaining lock as the pitch control on my transport is run through the full ±12% range (dropping the pitch by 12% lowers the sampling rate to 38.8kHz).

Monarchy has been extremely careful in designing a proper operating environment for the CS8427. Monarchy has designed their own power-line filter rather than using a stock IEC power cord connector with a built-in filter. They have also used a dual-bobbin power transformer, which further attenuates high-frequency power-line noise.

The CS8427 has three power-supply pins—two digital and one for the PLL. Monarchy has supplied separate 7805 regulators for each of the three 5V supplies. They have also included three ferrite beads in the DC supply lines to provide additional rejection of high-frequency supply noise.

The PLL supply has its own rectifier bridge and raw DC filter cap. The DIP 96/24 has both S/PDIF RCA and Toslink optical inputs. Two outputs are also included—S/PDIF and AES/EBU. Monarchy has coupled the outputs with an extremely high-quality pulse transformer.

The DIP 96/24 is a superb performer. In my review of The Parts Connection's DAC 3.0 in *AE* 6/00, I noted that any outboard jitter boxes I tried actually degraded the performance of this reference-quality DAC. Monarchy's DIP 96/24 is the first outboard device I have used with the DAC 3.0 that actually improves its performance.

This was a shock to me at first, since I really thought I had reached sonic nirvana with the DAC 3.0, and wondered just how much better the sound could really become. Inserting the DIP 96/24 improves soundstage width and depth, both in size and precision of localization. Inner detail and articulation is improved, and the treble region takes on an even more open and airy quality. These improvements are also readily apparent when I used the DIP 96/24 with my DVD player and the 96/24 PCM discs from Classic Records (especially the Rachmaninoff *Symphonic Dances*, DAD 1004).

With permission of Monarchy Audio owner C.C. Poon, I changed the input and output RCA connectors to Canare 75Ω BNC jacks. The DIP 96/24 is refined enough to really allow you to hear the difference between the stock RCA connectors and the BNCs. For those willing to go the extra mile, switching to BNC connectors will allow you to extract the last ounce of performance from your digital audio system.

The DIP 96/24 is a terrific performer. At the factory-direct price of $249, the new DIP continues Monarchy's tradition of offering the best buys in outboard clock jitter suppression. Highly recommended!

DIGITAL CABLES

Monarchy Audio also manufactures interconnect cables, including several for digital audio. Some time ago, Mr. Poon sent me the DR-1 Digital Reference for evaluation (*Photo 6*). It is a ½″ diameter, 1m cable featuring a solid Teflon® dielectric, silver-plated conductors, and a triple shield. The DR-1 is fitted with gold-plated, Teflon®-insulated RCA connectors.

In my listening evaluations the DR-1 provided excellent performance—better, in fact, than any RCA-type digital interconnect I've heard in my system. But, I still prefer the extra transparency and airiness of the DH Labs D-75 fitted with 75Ω Canare BNC connectors. The DR-1 is extremely stiff and rather a beast to use. You may need to physically separate the components you intend to connect, since even moderate bends in the cable are difficult. Connecting a DIP unit to a D/A converter, if they are stacked one on top of the other, may prove to be a challenge.

The DR-1 retails for $239, which certainly isn't cheap. But, if you price the competition in RCA-based digital cables, the DR-1 offers very competitive performance.

Monarchy also makes several other digital and analog interconnects worth investigating, including the thinner and more flexible DR-2 digital interconnect ($119) and the AR-2 Analog Reference ($239/1.5 meter pair). Both feature a solid Teflon® dielectric, silver-plated conductors, and gold-plated RCA connectors with OFC center pins. ❖

The Sovtek 12ax7 LPS Twin Triode

By Don Jenkins

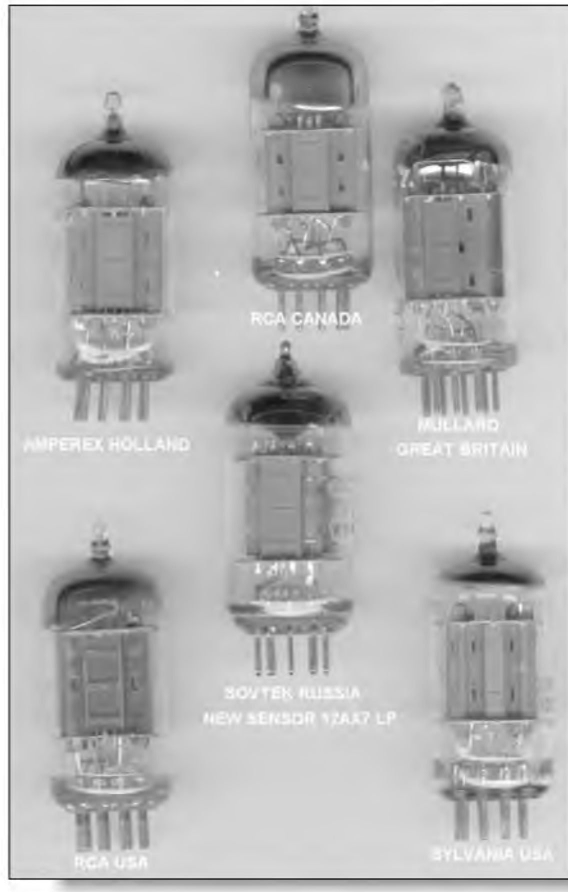


PHOTO 1: 12AX7 tube comparison.

New Sensor Corporation provided eight Sovtek 12ax7 LPS vacuum tubes for evaluation. The New Sensor accompanying data defines LPS as Large Plate.

PACKAGING AND DATA

I received these tubes in single, plain white boxes, without any external identification. I assumed the tubes were from a "selected" lot—i.e., not a random selection from a production run—due to hand-lettered numbers and symbols on the boxes. The data presented should be reviewed with that admonition.

The New Sensor literature specifically identified these as "large plate." The data on the tube itself is without this identification. The tube ID is black stencil on the glass envelope: 12AX7, SOVTEK, Made in Russia, with a date code.

Photo 1 is a photograph of the New Sensor 12ax7 LPS and five other 12AX7s in my inventory, showing the Sovtek model with a glass envelope slightly larger than the other tubes. As to plate size, it is only slightly larger than the USA RCA 12AX7 and the Mullard ECC83/12AX7 from Great Britain. It is definitely larger than the USA Sylvania 12AX7, the smallest of the lot, with the Canada RCA 12AX7 in between.

The tube serial numbers used here are those assigned by the supplier. The last digit refers to the grid pin number and is used to identify the two triode sections.

DIMENSIONS AND ASSEMBLY

The internal structure is very typical and conforms to the more or less standard 12AX7 arrangement relative to the placement of the interconnection buses. As mentioned, the anode area is somewhat larger than that of the USA RCA 12AX7. The glass envelope diameter is a nominal 0.845" while other 12AX7s have a nominal diameter of 0.810". For those users who shield these tubes in a metal jacket and have had difficulty pushing the tube out of this jacket after some hours of use, this increase in diameter will not be beneficial.

PERFORMANCE

I compared individual tube performance and dispersion with the use of the Dynamic Power Integral (DPI). While applicable in a general sense, power delivery from these small-signal triodes is not too useful for performance comparison. Since these tubes are primarily used for voltage gain or impedance matching, the transconductance is a more useful characteristic.

I present two sets of dynamic performance data: *Table 1* is for the classic direct-current performance with the plate voltage at 250V; *Table 2* shows performance when this tube is used as a voltage amplifier with a 20kΩ plate load and an AC signal on the grid. *Figure 1* is the output for the calibration test without any AC component. The grid voltage range is −2 to +2V, the plate voltage is 250V DC.

The transconductance average for the 16 triode units is shown in the last section of *Table 1*. The rigorous transconductance average for the lot was 2621μmhos with a standard deviation, 1 sigma, of 272μmhos. A more useful design parameter is the average linear transconductance, which was 2619 with a standard deviation of 87, or 3.3%. For a 16-unit sample this is a very close match. If these tubes were from a random selection, this could be considered "almost perfect"; if from a selected sample, then the selection process was very good.

In 1997 I ran similar comparisons on a random sample of ten 12AX7s, 20 triode units. *Table 3* is a summary of these tests. *Figure 2* is a plot of the performance average for this ten-tube lot and for the Sovtek lot. This average performance is the same. *Figure 3* is a plot of the plate-current dispersion, one standard deviation, for the ten-tube mixed lot and for the Sovtek eight-tube lot.

Figure 4 is a calibration test using an AC signal on the grid. For this test the grid bias is stepped over the −3.5 to +1.5V DC range, while a 70mV RMS, 1kHz, signal is on the grid. The plate load is

ABOUT THE AUTHOR:
Don Jenkins is a retired aerospace engineer and a consultant in rocket propulsion. He was a frequent contributor to *Speaker Builder*, *Glass Audio*, and was Technical Editor of *Valve and Tube News*. He is a member of the Projection Technology Committee of the Society of Motion Picture Engineers and is President of the Hollywood Motion Picture Projectionists Union.

20kΩ. The plate-voltage AC component is measured and used for gain calculation as well as harmonic distortion, which is shown on the plot as THD%. *Table 2* is a summary of these tests.

The quoted transconductance is really a pseudo transconductance since the plate load is 20kΩ. However, for comparison of the triode sections it is a valid parameter. As noted in *Table 2*, the average linear transconductance, over a −2.0 to +2.0 grid bias, is 1449 with a dispersion of only 2.3%. These 16 triode sections are exceptionally well matched.

Returning to *Fig. 4*, while the transconductance is practically linear over the −2 to +2 grid range, there is a "blip" in the THD as the grid starts to conduct. This is typical of all small-signal triodes, and the distortion depends on the relative magnitude and the nonlinearity of the grid current for positive grid-voltage values. For the absolute best linearity and minimum

s/n	Power	Null	DPI
0152	0.1838	0.0000	0.18682
0157	0.1944	0.0000	0.19718
0042	0.1769	0.0000	0.17946
0047	0.2104	0.0000	0.21373
0027	0.1804	0.0000	0.18291
0022	0.1908	0.0000	0.19392
0057	0.1908	0.0000	0.19387
0052	0.1805	0.0000	0.18309
0147	0.1967	0.0000	0.20024
0142	0.1896	0.0000	0.19260
0117	0.1790	0.0000	0.18127
0112	0.1975	0.0000	0.20081
0067	0.2014	0.0000	0.20558
0062	0.1488	0.0000	0.15047
0097	0.2027	0.0000	0.20629
0092	0.1710	0.0000	0.17319

Average Power Integral 0.1872 1 sigma 0.0148 7.9 %

Composite Plate Current from Average of 16 Triode Sections

Data for Grid Bias Range of 2 to -2.2 step -.2

Grid Bias	1 Sigma	Average ma	%(+/-)	high	low	delta/2
2.0	0.44550	11.66923	3.82	12.1	11.2	0.446
1.8	0.42463	11.11837	3.82	11.5	10.7	0.425
1.6	0.40469	10.55900	3.83	11.0	10.2	0.405
1.4	0.38560	9.99300	3.86	10.4	9.6	0.386
1.2	0.36719	9.42228	3.90	9.8	9.1	0.367
1.0	0.34950	8.84873	3.95	9.2	8.5	0.350
0.8	0.33230	8.27423	4.02	8.6	7.9	0.332
0.6	0.31563	7.70068	4.10	8.0	7.4	0.316
0.4	0.29931	7.12998	4.20	7.4	6.8	0.299
0.2	0.28335	6.56401	4.32	6.8	6.3	0.283
-0.0	0.26766	6.00468	4.46	6.3	5.7	0.268
-0.2	0.25213	5.45387	4.62	5.7	5.2	0.252
-0.4	0.23673	4.91347	4.82	5.2	4.7	0.237
-0.6	0.22136	4.38539	5.05	4.6	4.2	0.221
-0.8	0.20591	3.87151	5.32	4.1	3.7	0.206
-1.0	0.19030	3.37373	5.64	3.6	3.2	0.190
-1.2	0.17446	2.89394	6.03	3.1	2.7	0.174
-1.4	0.15827	2.43403	6.50	2.6	2.3	0.158
-1.6	0.14165	1.99590	7.10	2.1	1.9	0.142
-1.8	0.12454	1.58144	7.88	1.7	1.5	0.125
-2.0	0.10696	1.19254	8.97	1.3	1.1	0.107

Coefficients for Composite Data (3rd Order)
-3.944324E-02 .1065524 2.776942 6.004676
residual variance 2.264541E-12
coefficient of determination 1

Tube data read from file SO12AX7
Sheet 1 of 2 Printed 03-17-1999 12:10:55

Dynamic Power Integral (DPI) 3rd Order Transfer Coefficients CODE PWRSIG04
Sovtek 12AX7 LP 8 tubes 16 triode sections.
Plate voltage = 250 vdc grid bias range -2.0 to 2.0 vdc

Individual Tube Transfer Coefficients (3rd Order)

Plate current = C1 * gb^3 + C2 * gb^2 + C3 * gb + C4 (ma)

s/n	C1	C2	C3	C4
0152	-3.583035E-002	1.127459E-001	2.742698E+000	5.950697E+000
0157	-3.534936E-002	1.116555E-001	2.824611E+000	6.113392E+000
0042	-3.777690E-002	1.138526E-001	2.720832E+000	5.812619E+000
0047	-4.505099E-002	1.157979E-001	2.963595E+000	6.363269E+000
0027	-4.084441E-002	1.029286E-001	2.746312E+000	5.888659E+000
0022	-4.076558E-002	9.918036E-002	2.791126E+000	6.091633E+000
0057	-3.646158E-002	1.011626E-001	2.776408E+000	6.090611E+000
0052	-4.339936E-002	1.003580E-001	2.752152E+000	5.896076E+000
0147	-3.627067E-002	1.179300E-001	2.821054E+000	6.168236E+000
0142	-3.933215E-002	9.486556E-002	2.776347E+000	6.077678E+000
0117	-4.084403E-002	1.120469E-001	2.760896E+000	5.834080E+000
0112	-3.950015E-002	1.042800E-001	2.831977E+000	6.196379E+000
0067	-3.514069E-002	1.143820E-001	2.819917E+000	6.276098E+000
0062	-4.502194E-002	1.061629E-001	2.567250E+000	5.290252E+000
0097	-3.581958E-002	9.750140E-002	2.834639E+000	6.305872E+000
0092	-4.366866E-002	9.998818E-002	2.701218E+000	5.719263E+000

Transconductance Over the Following Grid Voltage Range (micromhos)
Grid voltage 1 -2.00 Grid voltage 2 2.00

s/n	Ip gb1	Ip gb2	Gm
0152	1.2029	11.6004	2599
0157	1.1936	11.9264	2683
0042	1.1286	11.4075	2570
0047	1.2597	12.3932	2783
0027	1.1345	11.4662	2583
0022	1.2322	11.7445	2628
0057	1.2341	11.7564	2631
0052	1.1404	11.4546	2579
0147	1.2880	11.9919	2676
0142	1.2191	11.6952	2619
0117	1.0872	11.4773	2598
0112	1.2655	11.9615	2674
0067	1.3749	12.0923	2679
0062	0.9406	10.4892	2387
0097	1.3132	12.0786	2691
0092	1.0661	11.1723	2527

Average Linear Transconductance 2619 1 Sigma 87.579 Percent 3.3

Gm at GB 1 1877.4 Gm at GB 2 2729.8
Rigorous Transconductance (Composite Data Set) 2621.5
1 Sigma Difference in Transconductance Units (micromhos) 271.9
NonLinear Difference 10.4%
Tube data read from file SO12AX7
Sheet 2 of 2 Printed 03-17-1999 12:10:55

TABLE 1: 12ax7 LPS direct-current performance. Plate voltage = 250V DC.

s/n	Power	Null	DPI
002/2	0.0495	0.0000	0.04951
002/7	0.0477	0.0000	0.04772
004/2	0.0442	0.0000	0.04421
004/7	0.0506	0.0000	0.05059
005/2	0.0474	0.0000	0.04742
005/7	0.0507	0.0000	0.05065
006/2	0.0421	0.0000	0.04206
006/7	0.0531	0.0000	0.05314
009/2	0.0468	0.0000	0.04679
009/7	0.0530	0.0000	0.05296
011/2	0.0512	0.0000	0.05124
011/7	0.0469	0.0000	0.04687
014/2	0.0501	0.0000	0.05006
014/7	0.0495	0.0000	0.04948
015/2	0.0483	0.0000	0.04835
015/7	0.0502	0.0000	0.05023

Average Power Integral 0.0488 1 sigma 0.0030 6.0 %

Composite Plate Current from Average of 16 Triode Sections

Data for Grid Bias Range of 1.5 to -3.5 step -.5

Grid Bias	1 Sigma	Average ma	%(+/-)	high	low	delta/2
1.5	0.14246	5.97129	2.39	6.1	5.8	0.142
1.0	0.13938	5.31301	2.62	5.5	5.2	0.139
0.5	0.13326	4.57053	2.92	4.7	4.4	0.133
0.0	0.12447	3.78061	3.29	3.9	3.7	0.124
-0.5	0.11339	2.98003	3.81	3.1	2.9	0.113
-1.0	0.10035	2.20554	4.55	2.3	2.1	0.100
-1.5	0.08541	1.49394	5.72	1.6	1.4	0.085
-2.0	0.06830	0.88198	7.74	1.0	0.8	0.068
-2.5	0.04832	0.40645	11.89	0.5	0.4	0.048
-3.0	0.02449	0.10410	23.52	0.1	0.1	0.024
-3.5	0.00651	0.01172	55.59	0.0	0.0	0.007

Coefficients for Composite Data (3rd Order)
-4.902903E-02 -2.133379E-02 1.602764 3.780614
residual variance 9.585103E-14
coefficient of determination 1

Tube data read from file 12ax7
Sheet 1 of 2 Printed 04-05-1999 13:53:12

Dynamic Power Integral (DPI) 3rd Order Transfer Coefficients CODE PWRSIG03
SOVTEK 12AX7 (LP) 8 tubes 16 triod sections
B+ = 250 Plate load = 20000 ohms grid bias -3.5 to 1.5

Individual Tube Transfer Coefficients (3rd Order)

Plate current = C1 * gb^3 + C2 * gb^2 + C3 * gb + C4 (ma)

s/n	C1	C2	C3	C4
002/2	-4.932902E-002	-2.695540E-002	1.598345E+000	3.816264E+000
002/7	-5.074540E-002	-2.462627E-002	1.600591E+000	3.740006E+000
004/2	-4.594739E-002	-5.928590E-003	1.561404E+000	3.580419E+000
004/7	-5.097422E-002	-2.064813E-002	1.645304E+000	3.846924E+000
005/2	-4.947930E-002	-2.228153E-002	1.588359E+000	3.727408E+000
005/7	-4.940633E-002	-2.475612E-002	1.618247E+000	3.857355E+000
006/2	-4.773375E-002	-8.431133E-003	1.545388E+000	3.490893E+000
006/7	-4.521080E-002	-2.363798E-002	1.599426E+000	3.956688E+000
009/2	-5.079317E-002	-2.112829E-002	1.599879E+000	3.698299E+000
009/7	-5.105592E-002	-3.415888E-002	1.635425E+000	3.955244E+000
011/2	-4.851773E-002	-2.651872E-002	1.609768E+000	3.883455E+000
011/7	-5.148888E-002	-2.043008E-002	1.610676E+000	3.699206E+000
014/2	-4.869498E-002	-2.303980E-002	1.608969E+000	3.833239E+000
014/7	-4.591347E-002	-1.489763E-002	1.595028E+000	3.804186E+000
015/2	-4.781526E-002	-1.944893E-002	1.590463E+000	3.762833E+000
015/7	-5.135886E-002	-2.445325E-002	1.636959E+000	3.837405E+000

Transconductance Over the Following Grid Voltage Range (micromhos)
Grid voltage 1 -2.00 Grid voltage 2 0.00

s/n	Ip gb1	Ip gb2	Gm
002/2	0.9064	3.8163	1455
002/7	0.8463	3.7400	1447
004/2	0.8015	3.5804	1389
004/7	0.8815	3.8469	1483
005/2	0.8574	3.7274	1435
005/7	0.9171	3.8574	1470
006/2	0.7483	3.4909	1371
006/7	1.0250	3.9567	1466
009/2	0.8204	3.6983	1439
009/7	0.9562	3.9552	1500
011/2	0.9460	3.8835	1469
011/7	0.8080	3.6992	1446
014/2	0.9127	3.8332	1460
014/7	0.9218	3.8042	1441
015/2	0.8866	3.7628	1438
015/7	0.8765	3.8374	1480

Average Linear Transconductance 1449 1 Sigma 32.694 Percent 2.3

Gm at GB 1 1099.8 Gm at GB 2 1602.8
Rigorous Transconductance (Composite Data Set) 1450.7
1 Sigma Difference in Transconductance Units (micromhos) 144.9
NonLinear Difference 10.0%
Tube data read from file 12ax7
Sheet 2 of 2 Printed 04-05-1999 13:53:12

TABLE 2: 12ax7 LPS voltage amplifier with 20kΩ plate load.

10 tubes in lot - 20 triode sections
All tubes either NOS or considered good for use

Grid Bias	1 Sigma ma	Avg ma	1 sigma hi	1 sigma lo	Lot max	ID	Lot min	ID
-3.5	0.11610	0.04	0.16	-0.07	0.33	2	-0.09	19
-3.0	0.09747	0.11	0.21	0.01	0.39	6	-0.04	1
-2.5	0.17011	0.47	0.64	0.30	0.96	6	0.17	1
-2.0	0.24485	1.10	1.35	0.86	1.76	6	0.75	1
-1.5	0.39224	1.97	2.36	1.58	2.79	2	1.37	8
-1.0	0.63575	3.04	3.68	2.40	4.48	2	2.02	8
-0.5	0.96335	4.28	5.24	3.32	6.48	2	2.75	8
0.0	1.36012	5.66	7.02	4.30	8.74	2	3.55	8
0.5	1.81322	7.15	8.97	5.34	11.20	2	4.38	8
1.0	2.31117	8.72	11.03	6.41	13.80	2	5.25	8
1.5	2.84320	10.33	13.17	7.48	16.48	2	6.12	8
2.0	3.39902	11.95	15.34	8.55	19.18	2	6.98	8
2.5	3.96867	13.54	17.51	9.58	21.85	2	7.81	8
3.0	4.54253	15.09	19.63	10.55	24.43	2	8.59	8
3.5	5.11131	16.55	21.67	11.44	26.85	2	9.30	8

Plate voltage 250 vdc. Heater 6.3 vdc
1 sigma value is +/- ma variation from lot average (Avg). This means that 15 of the triode sections will be within this tolerance at the specified grid bias. The columns shown as (Lot) give the lot maximum and minimum plate currents for that grid bias. The ID number identifies the specific tube section.

All tubes in this sample are listed below.

ID #	Tube Type
1	SYLVANIA 12AX7 GRID 2 S/N 001
2	SLYVANIA 12AX7 GRID 7 S/N 001
3	SYLVANIA 12AX7 GRID 2 S/N 002
4	SYLVANIA 12AX7 GRID 7 S/N 002
5	SYLVANIA 12AX7WA GRID 2 S/N 001
6	SYLVANIA 12AX7WA GRID 7 S/N 001
7	AMPEREX ECC83/12AX7 GRID 2 S/N 001
8	AMPEREX ECC83/12AX7 GRID 7 S/N 001
9	MULLARD ECC83/12AX7 GRID 2 S/N 001
10	MULLARD ECC83/12AX7 GRID 7 S/N 001
11	MULLARD ECC83/12AX7 GRID 2 S/N 002
12	MULLARD ECC83/12AX7 GRID 7 S/N 002
13	SYLVANIA kj 12AX7 GRID 2 S/N 001
14	SYLVANIA kj 12AX7 GRID 7 S/N 001
15	SYLVANIA kj 12AX7 GRID 2 S/N 002
16	SYLVANIA kj 12AX7 GRID 7 S/N 002
17	GE 12AX7 GRID 2 S/N 001
18	GE 12AX7 GRID 7 S/N 001
19	RCA 12AX7 GRID 2 S/N 001
20	RCA 12AX7 GRID 7 S/N 001

TABLE 3: Performance dispersion for 12AX7 vacuum tubes.

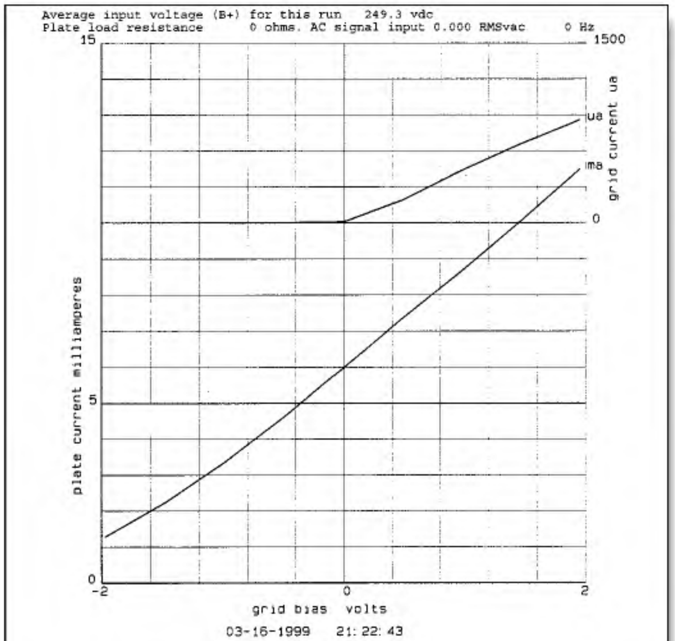


FIGURE 1: Calibration test output without AC component.

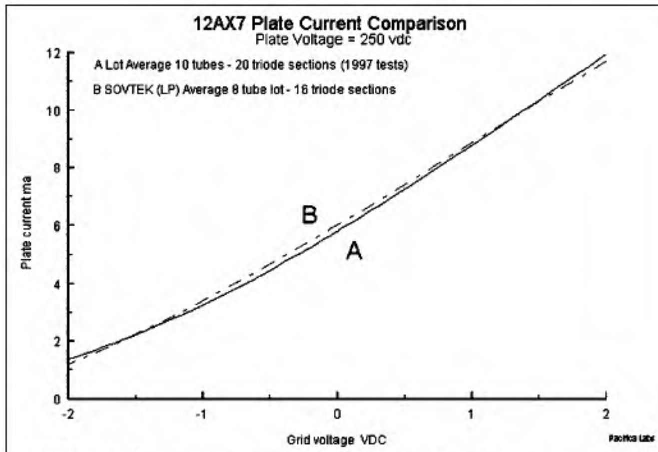


FIGURE 2: Performance average for ten-tube and Sovtek lots.

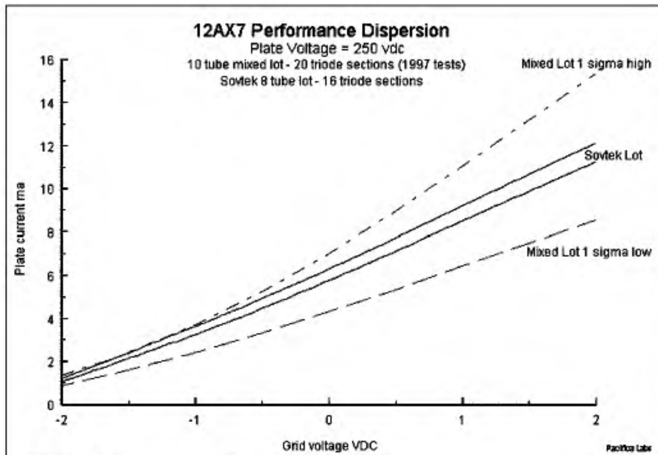


FIGURE 3: Plate current dispersion.

distortion, when this tube is used as a voltage amplifier, you should select the operating point to minimize the transconductance variation and to avoid the THD blip if possible. For high source impedance drivers, it may not, in fact, be possible to drive the grid positive.

Figure 5 is the derived third-order transconductance for a 20kΩ plate load. 100 and 200mV RMS grid signals are shown as peak-to-peak voltages from the maximum transconductance, which occurs at a grid voltage of −0.15V DC. The 100mV signal just reaches the zero grid voltage, which would be the optimum operating point for these conditions. A 200mV signal moves into the positive grid range. For this higher-input amplitude, it is a trade-off between the nonlinear transconductance and the grid-current-induced distortion to select the optimum quiescent grid voltage.

CATHODE BIAS

This performance data is for "hard" reference voltage values. That is, for the grid bias values quoted, a DC component is placed on the grid, relative to ground, and varied with the input signal shown. Few real-world circuits operate in this way. The usual voltage-amplifier circuit for the 12AX7 uses cathode bias.

Figures 6 and *7* show performance for the 12ax7 LPS when using cathode bias. *Figure 6* is the measured plate current for cathode resistance, using a 20kΩ plate load, and a B+ of 250V DC. The grid voltage shown is calculated from the resistor value and the plate current. The cathode resistor is paralleled with a 1.0µF capacitor for all of the data shown.

Figure 7 is the performance, quoted as THD%, for two values of grid signal and for a plate load of 20kΩ. Since there are about as many choices for the plate load and cathode bias resistor as there are circuits, *Fig. 8* covers the range up to 100kΩ for the plate load. I measured the THD values with a Stanford Research SR760 FFT Spectrum Analyzer.

The nominal heater voltage is 6.3V. *Figure 9* shows the plate-current difference for a heater voltage range of 5.8–6.8V.

REFERENCE PERFORMANCE

The first reference I have for 12AX7 characteristics is the *RCA Tube Manual RC 16*, dated 9-1950. The last manual I have, RC-24, is dated 10-1964. Both of these references give the 12AX7 characteristics as being the same as the 6AV6.

For the 6AV6 the transconductance for a plate voltage of 250 and a grid voltage of −2.0 is specifically given as 1600µmhos. The average value calculated for the 12ax7 LPS lot (16 triode sections) and using the third-order transfer function for this specific point,

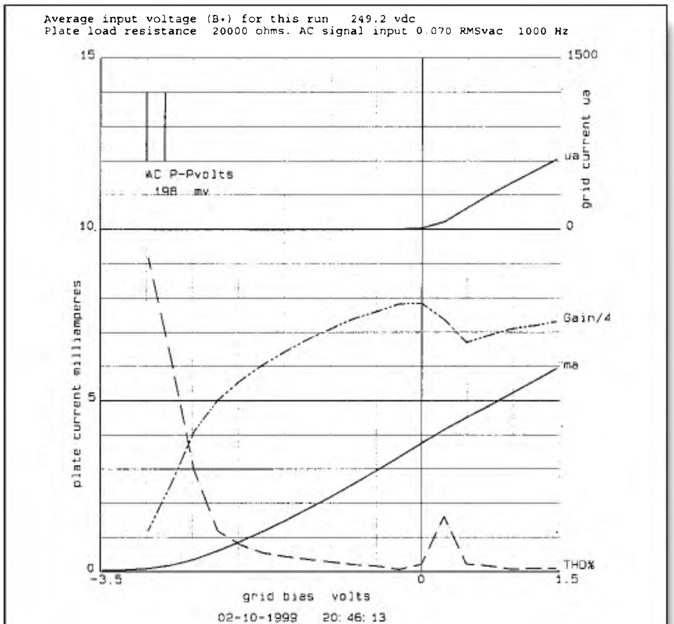


FIGURE 4: Calibration test using AC signal on the grid.

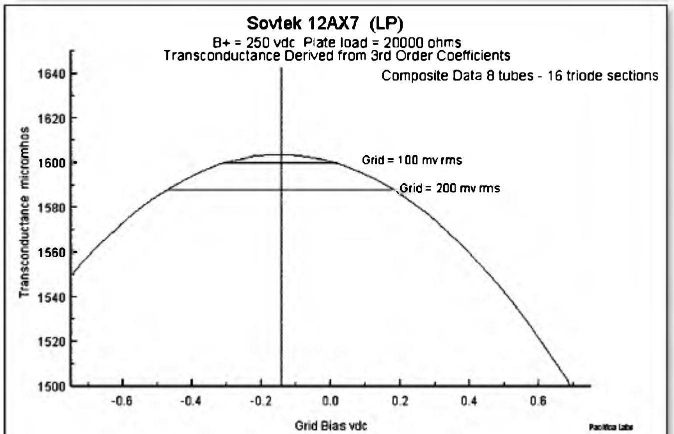


FIGURE 5: Third-order transconductance for 20kΩ plate load.

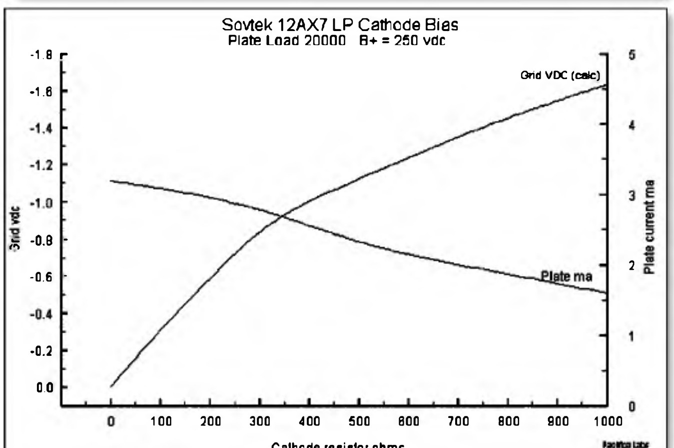


FIGURE 6: Measured plate current for cathode resistance.

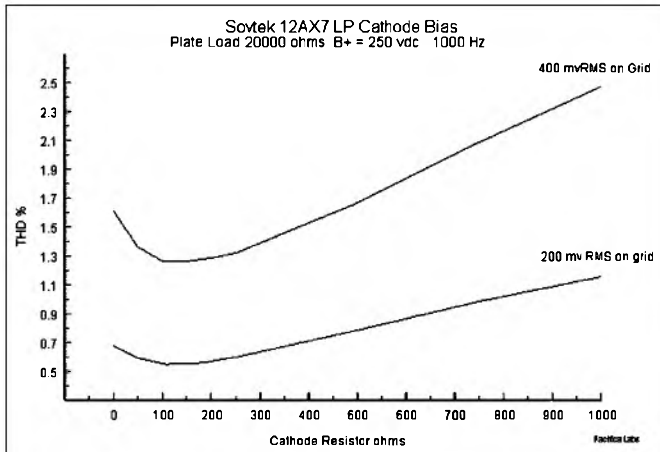


FIGURE 7: Performance of grid signal and 20kΩ plate load.

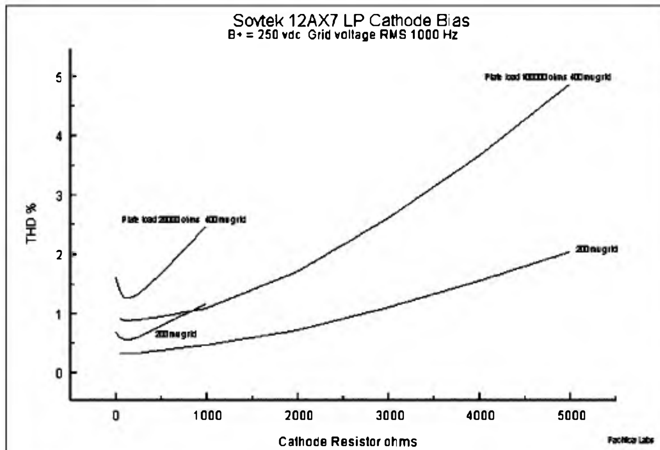


FIGURE 8: Plate load range up to 100kΩ.

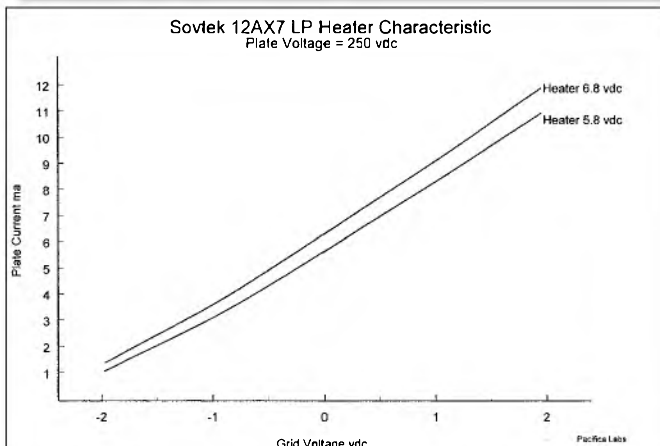


FIGURE 9: Plate current difference for 5.8–6.8V voltage range.

i.e., plate 250V DC and grid −2.0V DC, is 1877μmhos.

SUMMARY

The static performance of the 12ax7 LPS is basically "dead on" the normal 12AX7 curve as determined from an average value of many different manufacturers. The static performance dispersion of this lot is considerably better than what might be expected for a randomly selected lot from a production run. I do not have enough data to comment on the heater dispersion for the 12ax7 LPS; however, the measured values for these tubes fall in line with other evaluated cathode heater tubes.

These tubes should provide excellent results when used in circuits designed for the 12AX7. If all of the Sovtek (New Sensor) lots are as well matched as this lot, you could replace tubes in any circuit without any change in performance.

INTERPRETING TRANSCONDUCTANCE CALCULATIONS

There are several sets of transconductance calculations given in the data sheets. The basic transconductance values are determined from third-order transfer functions derived from the direct measurement of grid voltage and plate current. The values in *Tables 1* and *2*,

(to page 86)

WEBSITE RESOURCES

Adire
http://www.adireaudio.com

All Electronics
www.allelectronics.com

Audio Consulting
http://www.audio-consulting.ch

Linear Integrated
www.linearsystems.com

Marchand Electronics
www.marchandelec.com

McFeelys
www.mcfeelys.com

Solen, Inc.
http://www.solen.ca

Speaker City
www.speakercity.com

The Tube Store
www.thetubestore.com

http://www.partsexpress.com

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PREAMP CIRCUIT UPDATE

My congratulations go to *audioXpress* on the very first issue (1/01) in the new combined format. I hope that future issues will continue to bring audiophiles sage and "sound" advice, as have so many of the past articles within the discrete magazine predecessors.

I have some comments for Dr. Thagard on his article, "A Phono Preamplifier for the CD Era," Part 1, in the above dated issue (also printed in the UK's *Electronics World*, April 2001). First, let me thank you for sharing so many examples of your audio work with us. This applies to not just the present article, but also those of the past on power amps, and so on. They represent truly countless hours of detailed and tedious work—toil that is priceless towards taking real steps in improving our audio system reproduction.

I have a relatively minor contextual but important point on a statement within the first preamp article. Under "Complementary Symmetry," last paragraph (p. 25), you say: "Thus, this design is mostly a JFET-based one."—a point which is obviously valid. You then continue, saying: "It is entirely JFET-based, if you accept the notion that the BJT active-load devices aren't really in the signal path."

Actually, I do question that comment, as follows. Taking the upper-half schematic portion of Figure 1 as an example, clearly the signal-carrying drain current of J1A drives diode-connected Q1A, which in turn drives Q1B. Thus, this bipolar transistor signal-current path also appears at the collector of Q1B, along with the signal current from J1B. In doing so, it flows through the PNP path. A similar reasoning applies for the J2A-Q2A-Q2B NPN-to-JFET signal path at the bottom.

The statement with regard to the bipolar devices not being present in the signal path would, in my estimation, be true only if Q1A and Q2A were AC-bypassed to their respective rails. Or, if bipolar devices Q1B and Q2B were biased with a non signal-dependent voltage (i.e., such as Q5 and Q6, respectively).

Having said that, I'd like to note that these technical points shouldn't be regarded as any overall criticism of the design's utility, but only as a further clarification of how it works. In my opinion at least, there is no inherent harm to an audio signal passing through bipolar transistor current mirrors, which can be operated quite linearly on a current-in, current-out basis. I'm also confident that the circuit works as Dr. Thagard has described, and readers should have few problems in replicating it. Thanks again for all of those interesting articles!

Walt Jung
Fallston, Md.

Norman Thagard responds:

Walt is really too kind. AC does pass through the mirror that serves as an active load for the cascode diff amps. The AC on the left side of the mirror provides an additional 6dB of drive by increasing the amount of signal current sourced by the right side at the same time and by the same amount that the right side of the diff amp is reducing AC signal current. The upshot is that signal current to the common-gate device is doubled.

This is very nicely described and illustrat-

ed on page 690 and figure 13.31 of Microelectronic Circuits *by Muhammed H. Rashid (PWS Publishing Company, 1999). I taught this very same material last year from this text, which means that I can't even remember material after I've taught it. Walt suggests that chapter 4 of Gray and Meyer's* Analysis and Design of Analog Integrated Circuits, *3rd ed., is another good reference.*

There is also increased nonlinearity with this coupling from left-to-right. When I ran a flat gain-of-ten version of one stage of the phono preamp open-loop, bypassing the feedback gates to ground with a large tantalum capacitor, there was 0.23% THD at 20kHz with output set at 1V RMS. I used 20kHz because THD figures are almost always highest at this end of the audio spectrum.

While this is a very respectable open-loop distortion level, when I eliminated the left-to-right coupling across the mirrors with additional large tantalum caps, the THD at a 1V RMS output level fell to 0.068%. Thus, there is room for improvement in the design.

Unfortunately, this improvement may not be feasible for the first stage of the preamp because there is little open-loop gain to spare at 20Hz. First stage closed-loop gain is about 46dB at this frequency, while open-loop gain is a mere 54dB. Reducing open-loop gain by another 6dB means that the first stage will operate nearly open-loop at 20Hz. Since open-loop distortion is so low, THD won't be the problem, but it may induce an RIAA error at low frequency. The builder is certainly free to experiment with this.

I believe that I have given enough explanation for the experimenter to try different approaches to linearity improvements should anyone desire to do so. You could, for example, lower the overall closed-loop gain of the first stage, thus reclaiming enough feedback to keep the RIAA tracking precise even with the loss of 6dB of open-loop gain. You would then increase the second stage gain so that overall gain at 1kHz remained around 38dB or so.

For my part, I like the sound from the preamp design as described in the article. Since I have already built four channels, I'm loath to revise the design despite the demonstrable objective improvements that are possible. My primary interest in this design was the challenge to get good performance from a single-stage discrete op amp. This was accomplished, but the design is probably close to its practical limit. Even so, I am sure that I will take another stab at this at some point in the future.

ARTICLE MIX

I have been a reader of these magazines since the first issue 30+ years ago (*TAA*). I welcome the new combined format, but have a question regarding the choice of content. Do you try to have an equal percentage of the content of each of the previous magazines in every *audioXpress*?

It seems to me that the tube-related content (*Glass Audio*) is a bit higher than the other content. Since I am not a great fan of tubes (I had my fill of them 30 to 40 years ago), I look for more solid-state content than tube content. Occasionally, a tube project catches my eye, but not that often. I do not believe that tube sound is better than solid-state sound (I have worked with solid-state circuits that mimic tube sound and are quite easy to achieve and lose less power).

This is just a personal preference, of course. I thoroughly enjoy each and every magazine, and look forward to retiring soon so that I can spend more time with audio-related projects.

Richard Klassen
Yukon Territory, Canada

Ed Dell responds:

Mr. Klassen raises an interesting question that many audioXpress *readers have been raising, and I am pleased to be able to answer. Our standard yardstick for* Audio Electronics *and* Glass Audio *for article content was four major articles for each of the year's six issues. We used the same standard of four for* Speaker Builder*'s eight issues each year.*

When we started to plan for audioXpress, *we simply added the annual article totals and divided by twelve. That works out to 24 per year for the two bi-monthlies and 32 for* SB. *This is a total of 80 articles, working out to seven* aX *articles per issue: two each for* GA *and* AE *and three for* SB. *Thus, readers would be seeing the same number of articles over the course of a year of* aX *that they would have enjoyed in any subscription to* one of its predecessors, *plus four bonus articles.*

Our issues of aX *have been running 100*

BE PREPARED

pages average since the first issue. The article count does not include reviews, departmental additions, product news, and much else that goes to make up any issue.

I do remind readers, however, that article contributions are not brought to us by the tooth fairy. They come from authors who have a passion for good sound and the capable hands that make the equipment and craft the articles. The tube enthusiasts in our audience are by far the most active these days. For those of you who want more projects, as our surveys show you overwhelmingly favor, the lesson is clear. You speaker builders and solid-state people need to get up out of those comfy chairs and get busy.

You, as readers, are not just numbers to us. You are the flesh and blood of these periodicals. We're here to publish your experience and your achievements—and have been for over 31 years. Our database of authors includes over 700 contributors from all over the world. You can join them, and we welcome you. Our editorial health depends on your insights, experience, and new breakthroughs. Whatever richness of content we all enjcy in these pages depends directly on how well we share—and the quality of your work.—E.T.D.

Great publication. I have been reading *Speaker Builder* for some time and find *audioXpress* excellent. The mixture of articles is great, and I have found new areas that I would not have normally read.

I was disappointed by some of the published negative comments. In my mind the magazine has widened my audio horizons and added to my enjoyment in the area. Well done—keep it up!

Dan Fock
Sydney, Australia

After reading some interesting articles on power transformers, a DSP crossover, and a clever trick for flush-mounting drivers in the June 2001 issue, I felt the urge, as always, to go build something. Then I got to the letters. Evidently some readers are none too happy with the new magazine, to the point of recommending torture as "humor."

The main complaint seems to be insufficient coverage of their particular favorite audio niche. Fortunately, my interests are broad enough to truly enjoy your new magazine. While I might prefer more articles on solid-state, and fewer on tubes, I expect that the balance will even out, over the long run. Nonetheless, the tube articles are fascinating (I'd like to build a tube headphone amp someday), and I always learn something from the speaker articles.

I assume that the article balance is primarily determined by the authors' interests and projects, none of whom are making a living writing for your magazine. Based on the first few issues, it seems that the tube enthusiasts are the ones doing the most work, or, just as good, the ones most willing to write about their projects. Perhaps some of those readers who complain "there aren't enough solid-state articles" would be so kind as to write up some of their projects?

I have two suggestions. The first is for potential authors who might be reading this letter, and who may be willing to accept a little challenge. I think that *audioXpress* should have more "beginner" projects. These would be simple projects, built from readily available parts, that would still result in something that can't be bought off the shelf at BestBuy.

For example, how about a simple tube headphone amp? Or a single-ended Class-A power amp, built only with parts from Radio Shack? These projects might not be state of the art, but they might encourage a potential hobbyist to actually build something. If we want our hobby to remain healthy and thriving, we have a responsibility to encourage and support new enthusiasts.

The second suggestion is to include a short monthly summary of interesting or useful websites. Here's a start: http://www.headwize.com/. This site focuses on headphones and associated electronics, and maintains several discussion groups. I enjoy reading the do-it-yourself discussion group, because the contributors are friendly, enthusiastic, helpful, and extremely encouraging to new builders. The site also has several interesting headphone amplifier projects.

Congratulations on the new magazine. I like it, and I look forward to each issue.

Doug Burkett
Eaton, Ohio

I have been a subscriber to *Speaker Builder* and now *audioXpress* for a couple of years now, and before that an occasional off-the-shelf reader.

I like the new magazine. My interests extend beyond speaker building, but I never felt the need to subscribe to *Glass Audio*. I think it was my loss! However, I now enjoy reading up on these projects as well as the speaker projects and have had an article published in the June issue ("Easy Driver Flush-Mounting").

I also receive *Voice Coil*. I would like to suggest that you add the driver reviews from *Voice Coil* to the *audioXpress* magazine in a fashion similar to the column "New Chips on the Block." Even one new driver review per issue would be great. It is one of the highlights of the *Voice Coil* magazine, and I think the speaker builder readers would love it. Vance Dickason is well-regarded by the DIY community for his excellent book, and I think the column would make for a more speaker-builder-friendly magazine.

Rod Buike
rbuike@home.com

COROIDAL TRANSFORMER

I thought Pete Millett ("Power Transformers for Audio Equipment," June '01, p. 14) might be interested in another transformer type, which is not used very much anymore, if at all. This story was related to me by a power industry engineer I used to work with. The toroidal transformer was invented by GE (late 1800s, after Edison was close to giving up on DC utility power). Transmission efficiency is very important to the utilities, so the gapless toroid helped maximize profits.

In order to work around the GE patent, a company (Line Transformer Co., I think) devised a way to wind a continuous iron strip through the finished winding bobbin using a method similar to GE's winding the coil through the toroid core aperture. I think he referred to it as a "coroid," but I don't know if that's the official name. They pretty much locked up the pole-top distribution transformer business. That left Westinghouse with no way to make a gapless transformer.

Westinghouse was the champion of our modern AC generating/distribution system, so in order to be able to sell distribution transformers, they invented the "C"-core (p. 18 of Mr. Millet's article). The core faces were lapped to almost mirror finishes and carefully matched to mating core halves. By also reducing the flux density (the transformer ended up larger), the result was only 1%–2% less efficient than the toroid and the "coroid." By placing the windings directly over the core gaps (impossible in an E-I core), Westinghouse was able to contain the leakage flux very effectively.

Chuck Hansen
cmhj@concentric.net

SPEAKER SOURCE

Having just finished reading G.R. Koonce's excellent design article in the March issue of *aX* ("A Pair of Computer Speaker Designs"), I noticed that he used the same 6.5″ driver that I used in my Small-Monitor Odyssey (*SB* 8/00). This driver is no longer stocked by Martin Sound, but readers who wish to build either Mr. Koonce's small speakers or my small monitor can obtain the drivers from me as soon as my supply of the custom-made units arrives (early July, we hope). The drivers are identical to the Martin 2736 except that the poly dust cover dome has been replaced by a thick felt one, in order to eliminate a hump in the response at 3kHz. The price will be $59 per speaker.

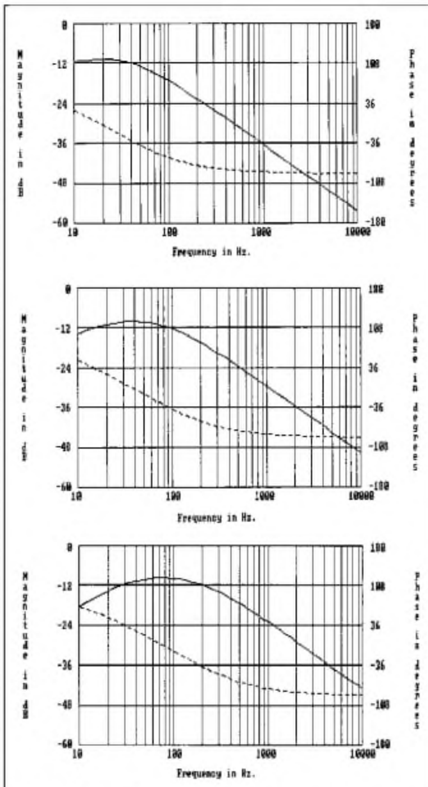
Interested persons can purchase by mail or online at: EDC Sound Services, 404 Olivia Drive, Lexington, NC 27295, www.edcsound.com.

Richard Honeycutt
Lexington, N.C.

PARTS SOURCE

After reading Justus Verhagen's article "Ribbons Made Easy" (May 2001, p. 56), I offer the following information to builders:

Contact All Electronics Corporation, 1-800-826-5432, for a free surplus catalog. They are a good source of inexpensive magnets, and they also have



FIGURES 1–3: Bass tone network.

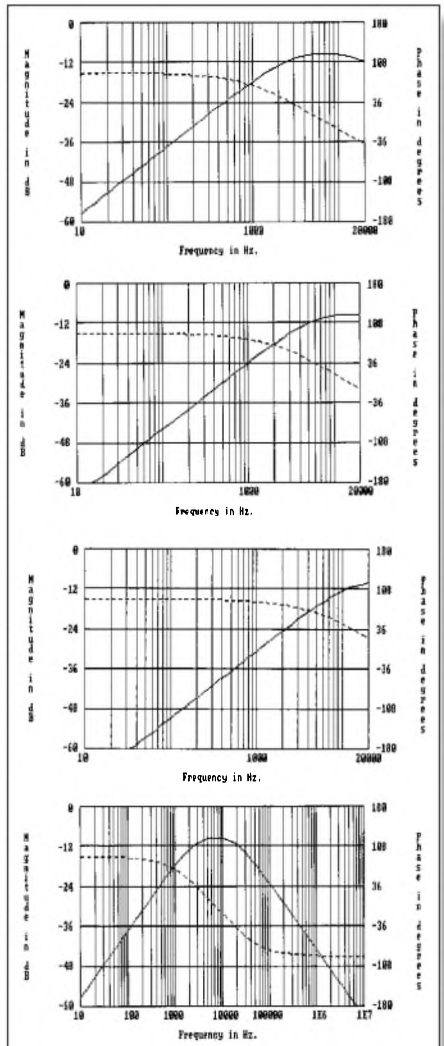
aluminum tape #ALT-1 for $3.50 1.5″ wide, 60 yards long.

Mark McCaslin
Gardiner, Maine

MEASURING MARSHALL

Here are seven printed plots (*Figs. 1–7*) of the curves I got from an analysis program I ran on the passive circuitry of the Marshall "Unusual Tone Control" article (April '01 *aX*). These show three plots of his low-frequency action. (Solid lines equal dB, dashed lines equal phase.) Then, for good measure, I thought you might wonder (as I did) how the same circuit *(Fig. 8)* could be used for both low and high ends, without anything changed except the capacitor values. (Shouldn't there be a symmetry flip-over, or something?) The answer is in plot seven, which covers the whole wide range from 10cps [Hz] to 10 megacycles. This reveals the way the thing works better than any arguments.

Bill Chater
Rancho Palos Verdes, Calif.



FIGURES 4–7: Treble network.

THROW-AWAY CULTURE

I would really rather not welcome your new magazine with loud complaint; however, the cover of the Jan '01 issue says "Build This Preamp." The article describes Dr. Thagard's need to use his personal advantages to obtain the semiconductors he designs around, and the need to buy in large lots. Only one of the semiconductors he lists is available from Mouser.

The cover of the Feb. '01 issue says "Build This Hybrid Tube MOSFET Amp." That article really hits a sore nerve. A couple years ago one of my Dynaco ST-150s began destroying its output devices. In checking my *TAA* back issues I found an article for a 60W MOSFET amp by Erno Borbely. The advantage to me in choosing that design was that the Dynaco power-supply components would work with the MOSFET amp. So I bought the boards and about $200 worth of components to stuff the boards.

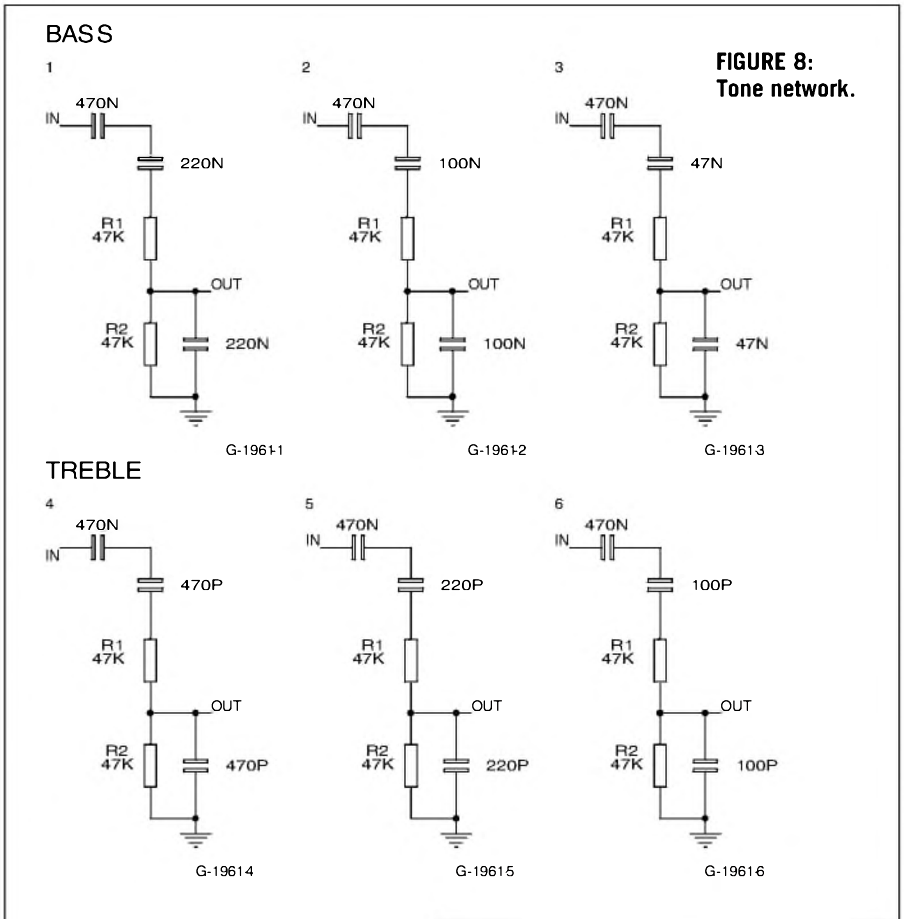
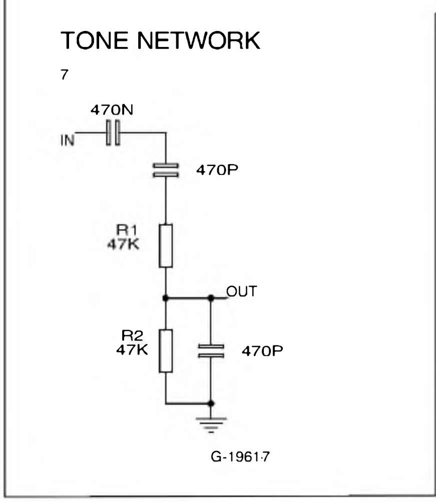


FIGURE 8: Tone network.



But my nearest electronics supply house, in Tacoma, about 60 miles away, didn't stock the Hitachi Power MOSFETs and wouldn't order them. None of my catalogs listed them, and most Seattle supply houses carry only NTE or equivalent. One hole-in-the-wall place took six months to tell me they couldn't get them. By this time, I was trying to get the newer plastic 2SJ162 and 2SK1058. Now I find those same devices specified in Marco Ferretti's design, and no word at all in the article for sources of supply. AARGH. Must we import them from Italy?

Perhaps this is the outcome of becoming a totally throw-away culture with dwindling numbers of repair shops and do-it-yourselfers. Fellow builders be warned: before you start buying parts for a project, make sure they are all available. There is something strange going on when it is easier to buy tubes than transistors.

Please, Mr. Editor, could we have a phono preamp design that uses NTE transistors and non-magical components that can be found in the Mouser catalog?

Art Day
Kingston, Wash.

UNCODED

I offer two comments on the March 2001 issue:

1. Featuring the LM3886, which you note as being on the market since 1993, under "New Chips on the Block" is pretty bogus; and

2. The code for articles is really unnecessary. Anyone who can't distinguish between tubes and solid-state really needs to be reading a far less technical magazine.

Claude Dickson
Lakeville, Mass.

BOGEN RESPONSE

Al Forbes' letter on the Bogen HO-125 (April 2001 issue, p. 82): The schematic published doesn't have complete info, but I believe that C9–10 are used to form a resonant circuit with L1 to increase rejection of the 120Hz output from the rectifier. Without the capacitor(s), L1 will provide $2\pi fL$ of reactance to filter the output of the rectifier. With C9–10 tuning the circuit to 120Hz, the impedance of the L-C combination increases by a factor of Q [which is roughly X_L/R_L (inductance reactance/winding resistance)]. Qs of low-frequency chokes are typically around 20–30, so you can achieve a big improvement in reduction of the principal component of the rectifier voltage with only a moderate bypass cap (C4) (1kV caps are big) at the expense of less attenuation of higher frequencies. I suppose two capacitors are used in series because of the high AC voltage present (nearly 1kV). This circuit also provides a light load to the rectifier tube (it looks like a big resistor), meaning it can be undersized as well. If L1 is 7.5H, then this analysis is valid for this circuit.

Letters about J.N. Still's 20W amp (April '01): In the schematic shown on p. 88, R14 can't do anything but change input resistance and at most short-out the input. To achieve the channel balance effect desired, a resistor should be placed between S1's pole and the connection of R14 and R1.

R14 will now form a variable divider with this input resistor, increasing the signal in one channel as the other increases when the control is moved from center position. The ideal value for R will increase gain in one channel by an amount equal to the loss of gain in the other channel so that the perceived volume of sound stays constant. If R1 (volume pot) was not present, R should be .707·R14 (balance pot) to achieve this. Factoring in R1 (it's in parallel with R) and the value of R14 gives a value of 220k (increasing R1 to 500k reduces R to 82k) for the new resistor.

Unfortunately, this large a resistor in the input will attenuate signal levels by almost 20dB and the amplifier now won't have enough gain. A compromise value for R can be reached if you consider that the balance control will normally be near center and the perfect response at the ends is not essential. Using a resistor of 22k (tube sources) or even 10k (for solid state) will produce a reasonable balance adjust effect with only a moderate gain loss (3–5dB). The output resistance of the sources is in series with this resistor and will need to be considered.

If your listening room requires an off-center balance setting, switching between sources of different output characteristics (say solid-state to tube) will shift the aurally perceived center as well as the volume. Adding resistors in series with the low output resistance sources to match the highest source will preserve the balance from source to source. Moving the balance control to a part of the circuit/system that has a constant source and a high load resistance will avoid this tweak. Another solution would be to leave out R14 altogether and move the furniture around.

This letter is turning into a treatise on the design and philosophy of balance controls. I don't know whether this topic deserves a short article or not; I know that I made the same mistake as Mr. Still on my first audio amp, and I don't think the idea that balance controls should not affect the volume is widely known.

Einar Abell
Three Rivers, Calif.

The two 0.5μF capacitors connected in series across filter choke L1 in Al Forbes' Bogen HO-125 amplifier are there to reduce the 120Hz ripple voltage under conditions of low current draw from the power supply, when there is little or no audio output and hum would be most noticeable. They do this by working with the choke inductance at 120Hz to form a parallel resonant circuit having very high impedance at the ripple frequency. The high impedance acts to greatly reduce the level of 120Hz ripple voltage coming from the full-wave rectifier, but allows relatively pure direct current to pass unimpeded. This improves the filtering action of the power supply at the primary ripple frequency when the amplifier is only drawing idling current; however, harmonics of the 120Hz ripple frequency (240Hz, 360Hz, and so on) will actually be greater with this circuit than without it, although their amplitudes will likely be too small to be of great concern.

The 807s in the Bogen amp operate in Class AB_2 and pull anywhere from 30mA of plate current at idle to as much as 240mA at full output (they also pull a bit of grid current on peaks, which is why the 6SN7 driver tubes are connected as low-impedance cathode followers). When the current load increases with rising audio levels, the choke inductance drops somewhat due to DC polarization of the magnetic core (which reduces the AC permeability), and the parallel circuit will fall away from resonance. The percentage of ripple voltage superimposed on the B+ output can then rise to as much as 10% or more, but the strong audio signal masks the ripple-induced hum in the output.

A little analysis of the Bogen circuit suggests that filter choke L1 is an ordinary smoothing choke of fairly low inductance, and not a swinging choke which, if one had been used, could have avoided the need to parallel resonate choke L1. A swinging choke can provide much more inductance than a smoothing choke at low current levels so there is less ripple at the output of the choke. The 2.0μF filter capacitor is also small for this application; a value of 8.0μF or more would further cut ripple and also improve the dynamic response characteristics of the amplifier. However, Bogen probably used the parts they did for reasons of cost savings (that goes for the misfit power transformer, too, which exceeds the maximum voltage rating for the 5R4GY rectifier tube, already the highest-voltage vacuum rectifier available). Sometimes a manufacturer gets a deal on parts and will sacrifice optimum design in favor of cost savings. After all, this amplifier was intended for use in a public-address system, and hi-fi design criteria were not at the forefront. The Bogen would not make a good hi-fi amp without extensive modifications, in my opinion.

By the way, Bogen probably used two 0.5μF capacitors in series across the filter choke to double the voltage breakdown rating of the caps (resulting capacitance is 0.25μF). The peak ripple voltage across the choke can approach two-thirds of the DC voltage coming from the rectifier tube, which in this case is about 950 to 1000V DC, so the caps must withstand nearly 700V of peak ripple (which is not actually AC—it is pulsating DC, but its effect on capacitors is similar to AC). Any replacement caps used in this way must be rated for high AC current-handling ability or they will soon fail. Four 1.0μF, 275V AC Cornell-Dubilier type 935-C4W1K caps (Mouser part number 5894-400V1) connected in series will give 0.25μF at 1100V AC, and should survive in this application. However, they ain't cheap—expect to spend about $40 for a set of four.

The 0.25μF value discussed here will resonate with a filter choke having 7H of inductance at the amplifier's idling current (30mA to the 807s plus another 21mA of bleeder current through resistor R17; total of 51mA). If you have another value of filter choke, and you can ascertain its inductance value at some particular DC current level, you can calculate the capacitance necessary to resonate it at 120Hz with this equation: $C = 1.76/L$, where L is inductance in henrys, and C is capacitance in microfarads (you can also reverse this equation if C is known and you wish to find L—just swap the positions of C and L; this is how I found the inductance value for the choke in the Bogen amp).

For a single-section choke-input filter such as is used in the Bogen amp, you can calculate the percentage of ripple voltage in the output with this equation: $\%\text{ripple} = 118/(L \times C - 1)$. Again, L is in henrys and C in microfarads. For the Bogen amp ($L = 7$H and $C = 2\mu$F), $\%\text{ripple} + 118/(7 \times 2 - 1) = 9.1\%$. This would be the percentage of ripple at the idling current of 51mA explained earlier, if not for the impedance of the parallel resonant circuit which brings the ripple percentage down to a very low level (not easily calculated). The ripple in this amp is worse at higher supply currents because the choke inductance is lower then, but such excursion should happen only momentarily at peak output.

The values of the filter choke and filter capacitor in the Bogen amp can combine to come dangerously close to a *series* resonant condition at 60Hz during maximum output, when the choke inductance will be at a minimum. Series resonance in a power supply can be extremely destructive due to the abnormally high current and voltage peaks that it causes, endangering every component in the power supply, but especially the filter capacitor and rectifier. It occurs when the product of the choke inductance in henrys times the filter capacitance in microfarads is either 1.77 (resonance at 120Hz) or 7.1 (resonance at 60Hz). With a 2.0μF filter capacitor, series resonance at 60Hz (every other pulse from the rectifier) can occur if the choke inductance drops close to 3.55H, which it may do during periods of maximum current draw. Even if exact series resonance isn't reached, just getting close to it can cause currents and voltages to rise to excessive levels, thereby shortening the useful lives of the rectifier tube and the filter capacitor. The series caps across the choke could also be damaged by abnormal voltage peaks. The simplest preventive measure is to increase the filter capacitance to 4 or 8μF. Maybe this is part of the source of the "tube killer" reputation for this amp (at least for the rectifier tube)?

Lastly, regarding the plate-to-plate load impedance of the output transformer: Bogen may have chosen a slightly lower-than-optimum transformer impedance here, anticipating that the output transformer secondary might not always be fully or properly loaded in PA use, thereby causing the reflected plate-to-plate primary impedance to be higher than the transformer's nominal rating of 4.8kΩ (the optimum transformer impedance is 7.3kΩ for this amp, when the output is perfectly matched). Generally, if the load impedance is too low, tube efficiency will be impaired, and if too high, maximum power output cannot be attained. The negative feedback used in the Bogen amp might make having a perfect impedance match less critical.

Keith A. Kunde
Independence, Ohio

Al Forbes responds:

After my letter titled "Bogus Bogen" was published in the April 2001 issue of audioXpress, *several readers replied to my query concerning the capacitors connected across the power-supply choke.*

I would like to thank readers Michael Kornacker, Fernando Garcia, Keith Kunde, and Einar Abell for taking the time to reply. Their input was very helpful and enabled me to understand an unfamiliar circuit application.

All four readers basically told me the same thing. Apparently, this LC combination forms a resonant circuit, tuned to 120Hz, *which serves as a notch filter to remove ripple from the power supply.*

Although my letter did not specify the inductance value of the choke, all four had done the calculations using the capacitance values and the 120Hz ripple frequency to suggest that the choke was probably around 7.0H. Sure enough, the parts list specified a 7.0H choke for L1. Two capacitors were used in series to obtain a higher voltage rating.

While I had not seen this done in power supplies, I do remember this technique being used back in the '60s to equalize sound systems. This was before graphic equalizers were available. At this time, most line-level audio circuits were 600Ω impedance, and passive LC filters were constructed using capacitors and inductors called "Boner-White coils" to trap out offending frequencies. Altec Lansing called this process "AccoustiVoicing."

Again, I wish to thank the four readers for sharing their knowledge. I also wish to thank the editors for forwarding their replies and for providing this excellent forum through which ideas and knowledge can be exchanged.

HELP WANTED

Can you help with a strange inquiry, please? We are manufacturers of high-end home audio equipment, and occasionally use vacuum-tube circuits.

Does anybody know the history and original application of the Raytheon 5703WB miniature wire-ended tube? It ceased production in the 1980s, and there has been no response from Raytheon themselves. We already have access to the technical specifications, and so forth, but our request is for the historical background.

Any help would be gratefully received—thank you.

Paul Taylor
Musical Fidelity
email@musical-fidelity.co.uk

I'm trying to get the dynamic parameters for a 3.5″ car stereo speaker, to be specific, a Legacy LS328 model. I would like to be able to determine the coefficient of friction (mu) and the spring stiffness characteristic K for the speaker. Also, I would like to learn about the dynamic behavior of voice coils. I am using the LS328 to drive an experimental setup consisting of a mass (aluminum tube attached to the coil itself), and a spring (several configurations used, see http://frahux.netfirms.com). I would welcome any referrals or any information where I can learn this stuff. Thanks in advance for any insight you might be able to offer on this subject!

Francis Huxley
Nonlinear Vibrations Lab
313-577-3806

I'm looking for the schematic of a Gunter Steinfar TV201CD amp or at least a basing diagram for the AEG PL519 power tube used in the amp. This was purchased in South Africa.

Hoby Irwin
xytech@polarcom.com

First, let me begin by saying congratulations on a wonderful publication! I have been a subscriber for over four years now and love the new format.

I am trying to locate a dealer who will sell large heatsinks in the 1–0.1°C/W range (natural convection) in quantities of 1–4 at a time. I have been wanting to build some of the project amplifiers in *audioXpress* but have not had any luck finding heatsinks other than from dealers with a minimum order of $1000.

Steve Skloss
Steven_Skloss@cisgi.com

1. I purchased a couple of DynAudio 30W–100 (12″) woofers 18 months ago for three-way systems, but they were too large for my better-half, so I made smaller boxes. The 8″ units fit and domestic harmony was restored.

Now I am deep-bassless, yet the method-in-my-madness was to make a subwoofer box to get down really low and hide this behind my wife's sewing box. But DynAudio no longer supports the DIY market, and UK suppliers such as Wilmslow Audio are no longer DynAudio.

Dickason's *Loudspeaker Design*

Cookbook leaves gi-normous gaps in parameter usage, impossible to apply (although testing stages covered ok). Book "search" has come up with nought either. So here are loudspeaker parameters in the hope someone out there is kind/mad enough to advise:

Compliance: suspension Cms 1.21 $10^{-3m/N}$, acoustic Cas 1.94 10^{-6} $m^{5/N}$, equivalent volume V_{AS} 2691.
Cone: eff. cone area Sd 400cm^2, moving mass Mms 35.2g, lin. vol. displacement Vd 320cm^3, mech. res. RMS 1.98kg/s, lin. excursion P-P X_{MAX} 8mm, max. excursion P-P 28mm.
Frequency response*: 22–900Hz.
Harmonic distortion: <0.8%.
Intermodulation distortion: <0.6%.
Magnet system: total gap flux 1300µ Wb, flux density 0.51 Tesla, gap energy 445mWs, force factor B × L 6.26 Tm, air gap vol. Vg 4.27cm^3, air gap height 8mm, air gap width 1.68mm.
Net weight: 2.1kg.
Overall dim.: 300 × 104.5mm.
Power handling**: nominal DIN 450W, music 600W, transient 1000W.
Q-factor: mech Q_{MS} 2.70, electrical Q_{ES} 0.803, total Q_{TS} 0.619.
Resonant free-air frequency: f_S 24kHz.
Rise time: 89µs.
Sensitivity: 1W/1m 91dB.
Voice coil: dia. d 100mm, length h 16mm, layers n 2, inductance (1kHz) Le .73mH, nominal impedance Zvc 8Ω, minimal impedance 6.4Ω, DC res Re 5.85Ω.
*Thiele/Small parameters measured dynamically.
**dependent upon box construction.
PS: I would not wish to emulate David Ruethers' 10ft^3 monster from *TAA* 2/1978!

2. Audio-activated mains-on circuit: Jan Didden had a circuit back in *TAA* 3/1978. Is there now a 200W opto-triac to do the job? Has anyone a circuit for the same?

Paul Nelson
2 Cloud View
Congleton, Cheshire CW12 3TP
England
0044 1260 281 085
01260 281 085 (tel/fax)
paulnelson@cwcom.net

I'm looking to buy Scan-Speak 8545 and Focal 120TDX2 drivers but I can't find tests or reviews of them. Can you help? I'm in the process of building two speakers that should look like the Wilson Audio Watt Puppy 5.1 and 6.

Rino L. Long
RinoLo@gov.nb.ca

I have a Hafler model DH-200 stereo power amplifier with serial # 3033176. Its power switch went bad, and I wish to replace it. (It has a lighted red miniature rocker switch.) I would greatly appreciate it if you could send me a schematic drawing showing which wires are attached to the switch.

Frankie "M"
frankiemblues@earthlink.net

I recently purchased a DVD player and connected it to my Sony home theater receiver via the 5.1 channel inputs. When playing DVDs now, I have virtually no bass sound coming from my subwoofer. When I play VHS tapes through my VCR I get *great* bass, as with all the other receiver modes (tuner, TV/SAT, Playstation, and so on).

The indicator LED on the sub (indicating when it's getting an input signal) lights up in the DVD mode, but there is no audible sound whatsoever. What gives?

Terry W. Ruprecht
Twruprec@oandm.uiuc.edu

I am trying to locate information and materials needed to repair the foam that is on the outside edge of the speaker and the frame.

F. Ford
101 Prospect St.
Chula Vista, CA 91911

I love the new combined format. A couple of years ago, my basement flooded. I lost all of my *Speaker Builder* issues. I'm trying to locate an article about a dipole type speaker nicknamed "tombstones" by the authors. Can anyone at the magazine help out?

Steve Frankowiak
BLEHOG96@prodigy.net

Not familiar as named.—Ed.

I have a pair of older KLH speakers model 14 loudspeakers and I am trying to find info on them. Could you tell me if you have had any info in the past about these, also any resources to put a value on them would be great as well. Thank you very much in advance for your help.

Richard Hunt
rwh12@home.com

I want to find replacements for my two Hughes Retriever sound enhancement components that were destroyed by a lightning strike. I believe Hughes sold the rights for their retrievers to a company called Nutronics. They disappeared from their Santa Ana office in the past few years. Can anyone help me find a new retriever?

I would also like to build a triode audio amp. I'm perplexed because most of the designs are expensive. Perhaps someone could submit a reasonable design that would trade off super performance for reasonable cost! I'll build it if it does show up.

Last of all, I would like to see a full-fledged preamp similar to the ones we used to get when vacuum audio components were the rule and not the exception. I would want a unit that would have a tonearm amp as well as the normal high-level CD/tuner capability. In addition, I would like a design that combines all of the switching and input jacks that were built into the better Fisher or Scott (Scott 340) receiver of yesteryear. ❖

John Collins
9320 Notre Dame Dr., Apt. C
Indianapolis, IN 46240

Readers with information on these topics are encouraged to respond directly to the letter writers at the addresses provided. –Eds

AC Power Line

from page 19

spikes but can be up to several milliseconds. Spikes and notches usually come in pairs, or in an oscillating series, due to power-line reactances.

Voltage flicker occurs when you start large motors, or during heavy load changes on a utility distribution system. This flicker is objectionable only if the magnitude of the voltage drop and the frequency of occurrence exceed the threshold of perception documented on the IEC 61000-4-15 voltage-flicker curve.

LIGHTNING

Lightning is one of the leading causes of power outages and transients on utility power lines. The surge currents associated with lightning strikes interact with the distribution-system impedance, producing large voltage transients that are transmitted to remote parts of the grid.

Lightning can cause transients on power lines without even hitting them. The large electric fields (up to 70V per meter) during a cloud-to-cloud discharge can couple into the power system, producing substantial induced-voltage transients.

Normal utility operations, such as load switching, the operation of disconnects on energized lines, adding or removing power-factor-correction capacitor banks, and the automatic operation of line reclosers and transformer tap changers, can all cause transients.

HARMONIC CURRENTS

IEEE Standard 519-1992 sets the US limits for harmonic current distortion, or *total demand distortion* (TDD). European Union (EU) Standard EN61000-3-2 is similar, and must be met before a device can gain the CE Mark required for sale in the European market.

These standards limit the harmonic content of current imposed back into the AC line by any device or its power supply drawing more than 75W. (This was supposed to change to 50W, but the notice has been cancelled.)

The increased use of switching power supplies can generate considerable noise or current harmonics back into the power line, thus contributing greatly to this problem. The maximum TDD varies with the ratio of utility short-circuit current (I_{SC}) available at the particular location in the distribution system to the maximum load current drawn by the user's equipment (I_L), and by the type of device. This ratio, I_{SC}/I_L, is called the short-circuit ratio (SCR)[3].

If the SCR is 20 or less, the device cannot impose more than 5% TDD on the AC line, with a maximum of 4% for any single harmonic below the eleventh. Even-order harmonics are limited to 25% of the adjacent odd-harmonic limits. The harmonic range covered by the regulations extends to the 39th (2340Hz for the 60Hz power line).

Things may not be as stringent in your neighborhood, depending on how many homes the pad or pole-top transformer is designed to serve, and your proximity to industrial loads. If the SCR is 1000 or more, the allowable 519-1992 TDD is 20%, with a maximum of 15% for any single harmonic below the eleventh. Pity the poor audiophile living near that location!

All nonincandescent lighting systems can cause harmonic current distortion. Fluorescent and high intensity discharge (HID) lighting systems can produce from 10% to more than 40% THD, depending on the design. Home-lighting electronic-ballast operating frequencies extend well into the RF range, but they must meet the stringent interference limits of EN55015/CISPR15 (more on these limits later).

Three-phase variable speed drive controllers cause a type of voltage distortion called *line notching*, which produces additional zero crossings on the voltage waveform. In the 1970s I worked for a company that installed four 150HP thyristor-controlled drive motors. The notching was so bad that even the clocks failed to keep proper time, and correcting the power quality problems required some serious power line filtering.

Incidentally, if you hear excessive 60Hz or 120Hz hum in your audio equipment, it is probably not due to injected harmonics, but to equipment power-supply problems. ❖

Part 2 takes a look at electromagnetic interference and how to test for it.

REFERENCES

1. M.G. Say, Wiley & Sons. *Alternating Current Machines*, 5th Ed.

2. IEEE C62.41-1980, *Guide for Surge Voltages in Low-Voltage AC Power Circuits.* January 30, 1981 (ANSI Standard).

3. For those familiar with alternator design, this is not to be confused with generator SCR, which is the ratio of field current at no-load rated voltage, to the field current required to develop rated stator current into a short circuit.

Sovtek Tubes

from page 72

for each triode section, are the "average" values over the grid voltage range, which for these tables is 0 to −2.0V DC.

This average value (identified in the table as the "Rigorous Transconductance Composite Data Set") is the quotient of the integral of the third-order derived transfer function for transconductance and the grid voltage range used for the integration, in this case 2.0V. This value is shown for each triode section, and the dispersion for the lot, based on one standard deviation, is calculated. This dispersion, for the lot, is given as the "Nonlinear Difference."

Third-order functions are difficult to work with for circuit design problems, so a second transconductance value is also shown. This is the "Average Linear Transconductance." This value is derived the same as the third-order value except that the transconductance function is assumed linear between the two grid voltage values and the measured plate current.

The two transconductance values shown in the table for Gm 1 and Gm 2 are the third-order derived values for those specific grid voltage values.

Table 2 has the same set of data as *Table 1*. However, the *Table 2* values are for a plate load resistance of 20kΩ. While identified as transconductance, the values are, in a sense, a pseudo transconductance due to the plate load. For purposes of comparison of individual tube triode sections, these values are satisfactory. The curve of *Fig. 5* is based on the *Table 2* values. ❖

Clubs

ARIZONA
Arizona Audiophile Society
Don Monroe, Secretary
5321 E. Elmwood St., Mesa, AZ 85205
(623) 516-4960
dmonroe@amug.org or
AzAudioS@aol.com

CALIFORNIA
Bay Area Audiophile Society
Dennis Davis
(415) 381-4228
bluedeer@a.crl.com
L.A. Audio File
http://users.aol.com/laaudio
San Diego Audio Artisans
Wendell
(619) 538-6946 eves.
Tube Audio Enthusiasts
John Atwood
65 Washington St. #137
Santa Clara, CA 95050
(408) 985-2006
West Valley Audio Society
Barry Kohan
Woodland Hills
(805) 375-2629

COLORADO
Colorado Audio Society
1941 S. Grant St., Denver, CO 80210
(303) 733-1613

CONNECTICUT
Connecticut Audio Society
Charles King
PO Box 116, East Berlin, CT 06023
(860) 665-2881
James Addison
171 Hartford Rd. A-7, New Britain, CT 06053

FLORIDA
Gulf Coast Audiophile Society
John R. Chait
5746 S. Lockwood Ridge Rd.
Sarasota, FL 34231
(813) 925-1070
Tampa Bay Listening Society
Mike Vans Evers
1250 E. Hillsborough Ave., Tampa, FL 33604
(813) 239-0700
(813) 239-0805

GEORGIA
Atlanta Audio Society, Inc.
John Morrison/Chuck Bruce
1160 Cumberland Rd. NE
Atlanta, GA 30306
(770) 491-1553/(404) 876-5659
chucksaudio@mindspring.com

ILLINOIS
Chicago Audio Society
Brian Walsh
PO Box 313, Barrington, IL 60011
(708) 382-8433
sysop@nybble.com
Prairie State Audio Construction Society (PSACS)
Michael Sims
2711 Bartlow Rd., Urbana, IL 61802
(217) 384-2073
m-simsl@uiuc.edu

INDIANA
Tube Sound
c/o William Schumacher
5417 Armstrong Ct.
Indianapolis, IN 46237-2318

LOUISIANA
New Orleans High End Audio Society
PO Box 50231,
New Orleans, LA 70150-0231

MASSACHUSETTS
The Boston Audio Society
David Hadaway
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Driver Report

The Jordan JX92S

By Vance Dickason

This loudspeaker guru applies his testing techniques to the latest offering from a classic designer—this is a reprinted portion of "Test Bench," from the August 2001 issue of Voice Coil.

I examined the new Jordan JX92S, a full-range product from Ted Jordan (E.J. Jordan Designs). If you are not familiar with Mr. Jordan's career, he began working with Goodmans in the U.K. and developed the Goodmans Axiette, a single cone full-range speaker. Ted began working with aluminum cone drivers in 1963 (Jordan-Watts) and introduced the Jordan 50mm Module driver (well respected in high-end circles) in 1975.

The JX92S is also being promoted as a full-range driver. This unit is constructed on a cast magnesium frame and uses a curvilinear aluminum cone and conical aluminum dustcap. The JX92S is terminated by a rubber surround and a 2⅛″ diameter cloth spider.

TABLE 1
E. J. JORDAN JX92S

	SAMPLE A	FACTORY
F_S	63.08Hz	45.00Hz
R_{EVC}	5.04	4.50
Q_{MS}	1.84	1.35
Q_{ES}	0.83	0.58
Q_{TS}	0.57	0.40
V_{AS}	8.10 ltr	15.28 ltr
Sens. (2.83V)	87.3dB	87.7dB
X_{MAX}	4.5mm	4.5mm

Intended for use near TV or computer monitors, the JX92S uses a bucking magnet and cup for shielding and has a vented polepiece for cooling. Vifa is currently doing the contract assembly of this product, so the quality control will certainly be up to Vifa's usual high standards.

I began by measuring the JX92S's free-air and delta compliance impedance (see *Fig. 1* for the free-air impedance plot) and transferred this data into the LinearX LEAP software for parameter calculation, as shown in *Table 1*. Unfortunately, the shipper destroyed one of the JX92S samples submitted by Canadian distributor Hifi Audio Labs, so only one sample was available.

There was enough difference in my measured parameters and the factory-listed parameters that I programmed the factory numbers into LEAP and compared box simulations using both data sets. The main difference is damping; my data showed a box Q_{TC} of 0.76, and the factory data produced a box Q_{TC} of 0.62, so the variation is more than just f_s/Q_{TC} ratio changes.

As I normally do when I encounter this much variation, I proceeded with the measured data. The box simulation was in a 0.36ft^3 sealed box with 50% fill material. Jordan recommends either a sealed box or a transmission-line variation, but since LEAP does not simulate transmission-line speakers, I have included only the sealed box example. *Figure 2* shows the results.

This 2.83V simulation resulted in an f_3 of 74.7Hz and a box Q_{TC} of 0.76, just about perfect for a home-theater LCR. Increasing the simulation voltage to a sufficient level (11.5V) to cause the driver to excur to X_{MAX} +15% (5.18mm in this case) produced a maximum linear SPL of 97.2dB, which is about average for a 5.25″ driver. *Figures 3* and *4* show the associated group delay and cone excursion curves.

Next I mounted the JX92S in a small enclosure filled with fiberglass and a baffle that measured about 9″ × 6.5″, and I measured the on- and off-axis frequency response at 2.83V/1m. *Figure 5* shows the on-axis response that indeed goes out to 15kHz, validating the claim of full-range operation. However, the off-axis response curve set in *Fig. 6* shows a power response that is about the same as any 5.25″ driver. The –3dB frequency at 30° with respect to the on-

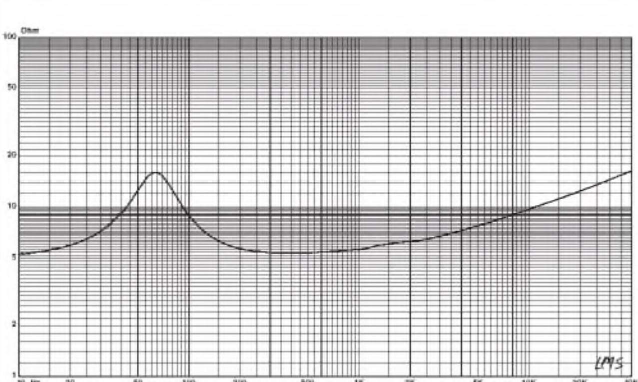


FIGURE 1: Jordan JX92S free-air impedance plot.

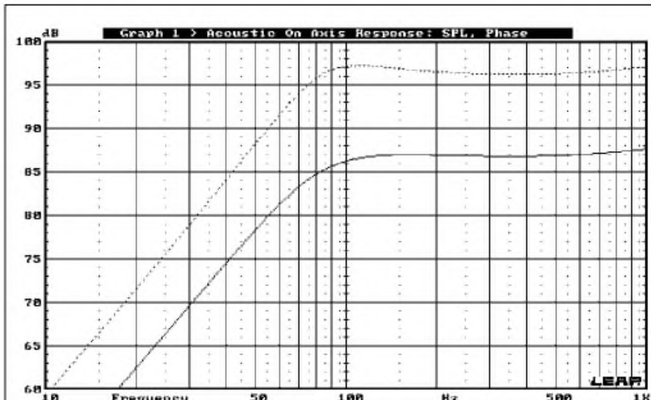


FIGURE 2: Jordan JX92S computer box simulation (solid = sealed box @ 2.83V; dot = sealed box @ 11.5V).

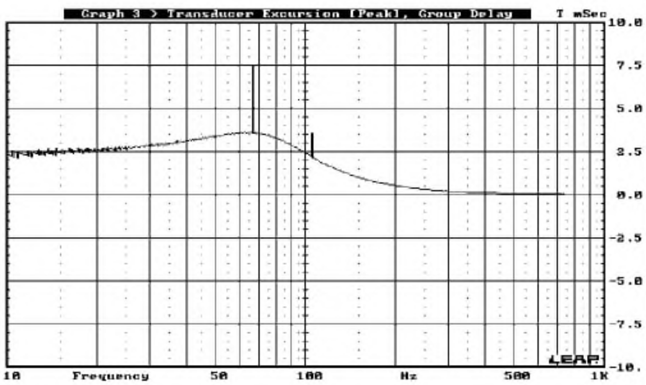


FIGURE 3: Group delay curves for the 2.83V curves in *Fig. 2*.

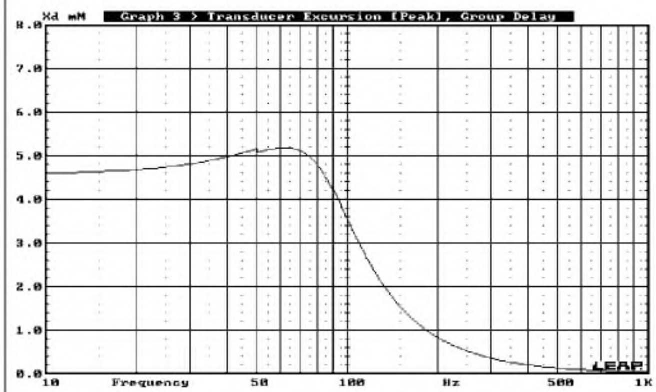


FIGURE 4: Cone excursion curves for the 20.5/18V curves in *Fig. 2*.

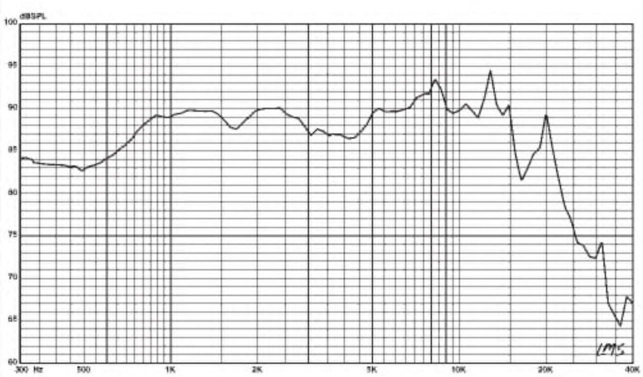


FIGURE 5: Jordan JX92S on-axis frequency response.

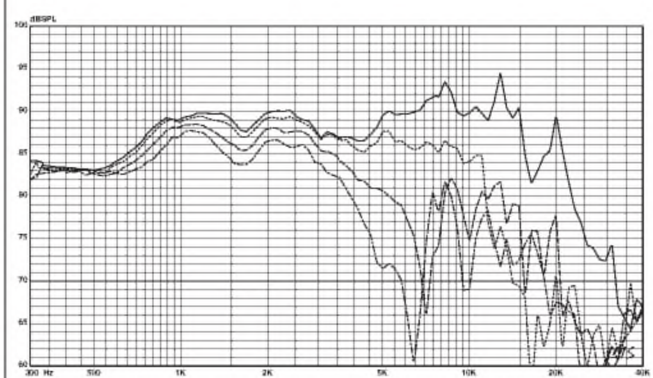


FIGURE 6: Jordan JX92S on- and off-axis frequency response (0° = solid; 15° = dot; 30° = dash; 45° = dash/dot).

axis curve is 3.7kHz, which is somewhat higher than the usual 5.25″ poly mid, but this limitation is due to the circumference of the driver.

Either way, the timbre of this driver is very nice and the response curve is relatively smooth out to the beginning breakup modes at 8kHz. The rise in the response above 500Hz is, of course, dependent upon baffle area and would yield a more flat overall response on a larger baffle. While I would not likely choose to use this product full-range without a tweeter, E.J. Jordan has quite a following of persons who would consider the addition of a tweeter and crossover a bit heretical. Either way, this is an interesting product. For more, contact Hifi Audio Labs (who provided the samples for this report), 31-2088 Pembina Highway, Winnipeg, MB, R3T 2G8, Canada, (204) 261-4975, or visit the Hifi Audio Labs website at www.hifiaudiolabs.com. ❖

New Chips on the Block

Analog Devices AD1852 and AD1854 DACs

By Charles Hansen

Analog Devices, Inc. announced two new audio digital-to-analog (DAC) converters that provide 113–117dB of dynamic range to digital versatile disk (DVD) applications. The AD1852 and AD1854 are the newest additions to Analog Devices' family of audio DACs that are designed to enhance the performance of professional and high-end consumer audio applications such as DVD.

The total harmonic distortion + noise (THD+N) of the AD1852/54 (–97 to –104dB) is claimed to be the best available. THD+N is the ratio of the original signal to harmonic distortion plus unwanted noise (such as when there is no recorded sound in between tracks on a DVD). It is considered a more meaningful performance measurement because it represents how the human ear hears.

AD1852

The AD1852 is a complete 18/20/24-bit stereo audio DAC that is a lower cost, voltage-output, follow-up to the AD1853, which was the industry's first standalone solution to enable OEMs to meet the new 192kHz audio sample rate standard for DVD-Audio players. This new IC uses proprietary data conversion technology, significantly improving audio performance at sample rates from 32kHz to 192kHz.

The AD1852 is a single-chip stereo digital audio playback system. It consists of a multibit sigma-delta modulator, digital interpolation filters, and analog output drive circuitry. Other features include an on-chip stereo attenuator and mute, programmed through an SPI-compatible serial control port. The AD1852 is fully compatible with all known DVD formats, including 192kHz as well as 96kHz sample frequencies and 24-bits. It also is backwards-compatible by supporting 50µs/15µs digital de-emphasis intended for "Redbook" compact discs, as well as de-emphasis at 32kHz and 48kHz sample rates.

The AD1852 operates from a single +5V power supply and is packaged in a 28-pin SSOP. For professional audio systems, the AD1852 is priced at $4.50 in 10,000 quantities.

AD1854

The AD1854 is a high performance, single-chip stereo digital audio playback system capable of sampling data at 16-,18-, 20-, or 24-bits. This monolithic DAC system includes a multibit sigma-delta modulator with dither, continuous time analog filters, analog output drive circuitry, and click-less volume control and mute.

The high performance and low price make the AD1854 well-suited for DVD players, A/V receivers, digital car audio systems, surround-sound home theater systems, high-end CD players, as well as "prosumer" and professional equipment such as digital mixing consoles.

Other features include an on-chip stereo attenuator and mute, programmed through an SPI-compatible serial control port. The AD1854 is fully compatible with current DVD formats, including 96kHz sample frequency and 24 bits. It is also backwards-compatible by supporting 50µs/15µs digital de-emphasis intended for "Redbook" 44.1kHz sample frequency playback from compact discs.

The AD1854 has a very simple but very flexible serial data input port that allows for glueless interconnection to a variety of ADCs, DSP chips, AES/EBU receivers, and sample rate converters.

The AD1854 can be configured in left-justified, I^2S, right-justified, or DSP serial port compatible modes. The AD1854 accepts serial audio data in MSB first, twos-complement format. A power-down mode minimizes power consumption when the device is inactive. The AD1854 operates from a single 5V power supply. It is fabricated on a single monolithic integrated circuit and housed in a 28-lead SSOP package for operation over the temperature range 0°C to 70°C.

The AD1854 is available in two performance grades. The AD1854KRS achieves the same 113dB dynamic range and signal-to-noise ratio (S/N, A-weighted—not muted) at 48kHz (110dB at 96kHz) as the AD1855, while improving THD+N to –100dB. The AD1854JRS achieves 108dB dynamic range and S/N (not muted) at 48kHz (110dB at 96kHz), and –97dB THD+N. The AD1854KRS is available for $3.95 in 10,000-piece quantities, while the AD1854JRS is offered at $2.95 in 10,000-piece quantities.

Analog Devices Inc., 804 Woburn St., Wilmington, MA 01887, (781) 937-1428, Fax (781) 937-1021, www.analog.com. ❖

Cirrus Logic CS43L43

By Charles Hansen

Cirrus Logic recently announced the industry's lowest power digital-to-analog (D/A) converter, the CS43L43, which integrates four chips into one package. This highly integrated chip is also reported to be the industry's smallest solution and offers the highest available audio quality for portable devices, all at one-fourth the power consumption of typical low-voltage D/A converters.

The CS43L43 integrates the functions of four chips into one 16-pin TSSOP package that consumes only 16mW of power. It outperforms other low-power chips on the market by providing higher dynamic range (96dB), better total harmonic distortion plus noise (THD+N), and longer battery life for portable audio applications. The THD+N of the CS43L43 is –85dB, a full 20dB improvement over existing solutions. The D/A converter allows more than 200 hours of music playback.

Cirrus Logic is the only company to provide a single-chip solution with all the capabilities typically found in four separate chips: a D/A converter, an audio effects digital signal processor, an analog volume control, and a headphone amplifier. Cirrus Logic sells its products under the Crystal®, Maverick™, and 3Ci™ brands as well as its own name.

PRICE AND AVAILABILITY

The CS43L43 is available today and is priced at $2.20 in OEM quantities. Contact Molly Muir, Cirrus Logic, mollym@crystal.cirrus.com. ❖

Burr-Brown OPA344 and OPA345 Family

By Charles Hansen

The OPA344 and OPA345 series rail-to-rail CMOS operational amplifiers are designed for precision, low-power, miniature applications. The OPA344 is unity gain stable, while the OPA345 is optimized for gains ≥5, and has a gain-bandwidth product of 3MHz.

The OPA344 and OPA345 are optimized to operate on a single supply from 2.5V and up to 5.5V with an input common-mode voltage range that extends 300mV beyond the supplies. Quiescent current is only 250μA (maximum).

Rail-to-rail input and output make them ideal for driving audio analog-to-digital converters (ADCs). They are also well suited for other audio applications and providing I/V conversion at the output of D/A converters. Single, dual, and quad versions have identical specs for design flexibility:

Single	Dual	Quad
OPA344	OPA2344	OPA4344
OPA345	OPA2345	OPA4345

All packages are specified for operation from –40°C to 85°C. A SPICE macromodel is available for design analysis.

FEATURES

- Rail-to-rail input
- Rail-to-rail output (within 1mV)
- Low quiescent current: 150μA (typ), 250μA (maximum)
- Microsize packages: SOT23-5, MSOP-8, TSSOP-14
- Gain-bandwidth
  - OPA344: 1MHz (typ), G ≥ 1
  - OPA345: 3MHz (typ), G ≥ 5
- Slew rate
  - OPA344: 0.8V/μs (typ)
  - OPA345: 2V/μs (typ)
- THD + Noise: 0.006% (typ)
- Input voltage noise: 30nV/√Hz (typ)
- Input current noise: 0.5fA/√Hz (typ)

BUDGETARY PRICING

Pricing will vary according to grade and package; 1,000 pieces; recommended resale in US$; FOB USA.

OPA344: $.67
OPA345: $.67
OPA2344: $1.10
OPA2345: $1.10
OPA4344: $1.85
OPA4345: $1.85

Texas Instruments, Burr-Brown Products, PO Box 11400, Tucson, AZ 85734, 1-800-548-6132, www.burr-brown.com. ❖

Book Review

GEC Audio Tube Data, 2nd Edition

Reviewed by Larry Lisle

GEC Audio Tube Data: Data and Amplifier Designs KT66, KT77, & KT88, Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, (603) 924-9464, FAX (603) 924-9467, e-mail custserv@audioXpress.com, $16.95. BKAA46.

The good just got better! There's a new edition of *GEC Audio Tube Data* that includes two new sections. The larger one—MOV Application Report Number 20 from 1979—is 36 pages on the theory, design, and construction of preamplifiers. It discusses three functions of preamps: amplifying small signals from a variety of sources, compensating for the deficiencies of the recording process (equalization) so that what you hear is what was originally recorded, and providing tone control if desired.

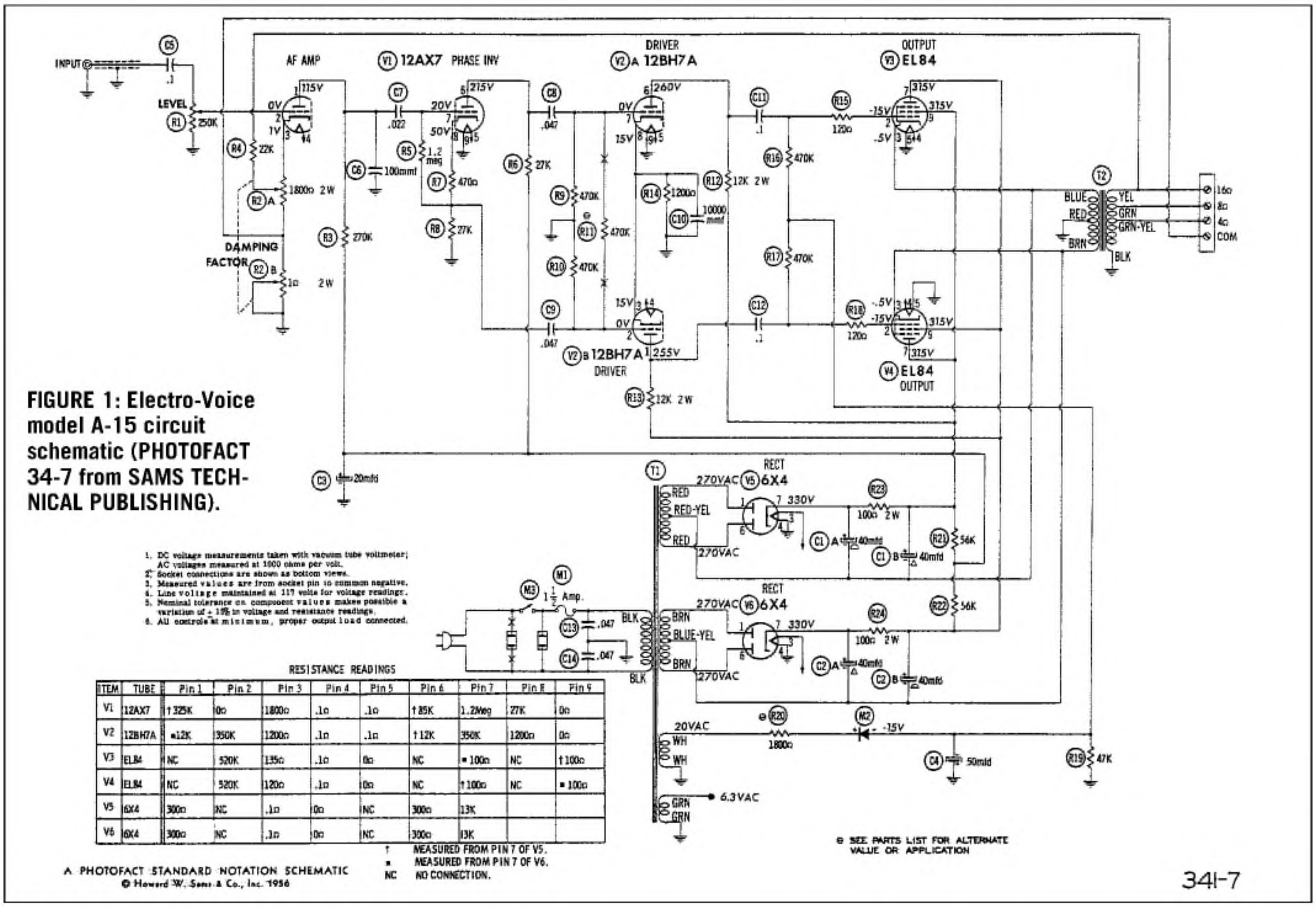
This section offers a very thorough look at microphones, the equalization process for a number of disc and tape formats—both new and old in 1979—and a variety of active and passive tone controls. You can learn much about the theory of equalization, or simply build from one of the many circuit diagrams. The diagrams include "building block" units that you can use with other circuitry and also complete preamplifiers. Several different tubes are used, including some old favorites and some I wouldn't have considered.

The second new section provides amplifier circuit data for the KT77. While the KT77 is not common, the information is certainly interesting, and you can use it in designing amps with other tubes.

The Preamble comparing tubes and semiconductors, and the data and amplifier designs using the KT66 and KT88, as well as the information on the PX4 and PX25, are of course included. ❖

Glass Shard

E-V Model A15



RESISTANCE READINGS

ITEM	TUBE	Pin 1	Pin 2	Pin 3	Pin 4	Pin 5	Pin 6	Pin 7	Pin 8	Pin 9
V1	12AX7	† 325K	0Ω	1800Ω	.1Ω	.1Ω	† 85K	1.2Meg	27K	0Ω
V2	12BH7A	▪ 12K	350K	1200Ω	.1Ω	.1Ω	† 12K	350K	1200Ω	0Ω
V3	EL84	NC	520K	135Ω	.1Ω	0Ω	NC	▪ 100Ω	NC	† 100Ω
V4	EL84	NC	520K	120Ω	.1Ω	0Ω	NC	† 100Ω	NC	▪ 100Ω
V5	6X4	300Ω	NC	.1Ω	0Ω	NC	300Ω	13K		
V6	6X4	300Ω	NC	.1Ω	0Ω	NC	300Ω	13K		

† MEASURED FROM PIN 7 OF V5.
▪ MEASURED FROM PIN 7 OF V6.
NC NO CONNECTION.

FIGURE 1: Electro-Voice model A-15 circuit schematic (PHOTOFACT 34-7 from SAMS TECHNICAL PUBLISHING).

I noticed the interest some readers expressed in the 6V6 3W cathode follower amp (Glass Shard, *GA* 3/98, p. 66), and thought I'd pass along this schematic for the Electro-Voice model A15 push-pull cathode-follower amplifier. (Figure copyright Howard W. Sams & Co.)

Here's some information for those who may be motivated to construct the amp. The output transformer is 2kΩ CT: 4-8-16Ω (E/V part #1590) or similar. It's important to note that the plate in each of the two EL84s connects to the screen grid of the opposing EL84, and that each of the two power supplies has its low side connected to the cathode of the appropriate EL84 output tube. Also, some components connect to chassis ground. Pay close attention to the schematic and stick to it entirely.

The dual-ganged potentiometer (R2) in the damping circuit may be difficult, if not impossible, to locate, but perhaps one of *aX*'s regular contributors may know how to successfully modify the circuit, using currently available parts. Until such time, some may prefer to do the following:

- Substitute the pot with an 1800Ω–2W resistor connected between PIN-3 of V-1 and ground (chassis ground, that is).
- Connect the 22k feedback resistor to PIN-3 of V-1.
- Connect the "com" speaker terminal of the speaker terminal strip to chassis ground, or simply ground the (–) speaker lead to chassis.

As for the power transformer, I know of no manufacturer who has one available; therefore, separate units are probably the best way to go. The –15 bias source supply can be of your own design, but you would do well to use a regulated power supply. Because of the number of transformers required, monoblocks are the best way to go for stereo.

Specifications? I haven't seen any yet and I don't know how it sounds, so therefore I can't guarantee satisfaction. I, for one, intend to build the amp, just for the fun of it. If someone out there builds it, perhaps he or she can report the amp's up side and down side. ❖

Neal A. Haight
Castro Valley, Calif.

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Yard Sale

FOR SALE

23 pieces E130L/Tungsram, unused. Minimum order 4 pieces. Price: 50DM per piece. Walter_burkhardt@web.de.

FOUR BOZAK B-199A 12″ woofers, vintage 1960s—available separately, or if you live in the New England area, in two unfinished fiberglass filled plywood enclosures, 33.5″W × 17″D × 23.5″H. Drivers are flush mounted in sand-filled (½″ of sand) 3-ply front panels. $150 for four drivers, plus shipping. Boxes are free if you transport. AAP, Box 876, Peterborough, NH 03458.

TWO JANSZEN 130-U black utility, four-panel electrostatic tweeters. 7¼″ × 23″ × 12″. Power supplies probably need re-furbishing. Diagram available. $200, plus shipping, as is, where is. Four spare replacement panels included. AAP, Box 876, Peterborough, NH 03458.

WANTED

Three amplifiers, any condition: two late-1940s theater amps—International Projector Corp. (also manufactured as Simplex), Model AM-1026 (60W) and AM-1027 (20W). Also need EICO HF-32 Integrated mono amp. Have one of each—would like pairs for stereo. John Agugliaro, 845-947-2748, JAGUGL4546@aol.com.

Schematic and specifications of RCA tube amp, type MI-4288-J Y, using four 6L6 metal tubes. Will pay expenses and postage. Ken Domminney, 7 Chestnut Close, Eastbourne, BN22 0SZ, England, Phone/FAX +44 (0) 1323 500174.

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WANTED

TUBE HIFI, HORN SPEAKERS, and HIGH END. Altec, Marantz, JBL, Mcintosh, Electrovoice, Fisher, Scott, Dynaco, Leak, Eico, Audio Research, CJ, Quad, Bedini, etc. Sonny, Phone: 850-314-0321 Fax: 850-314-0284 **sonnysound@aol.com**.

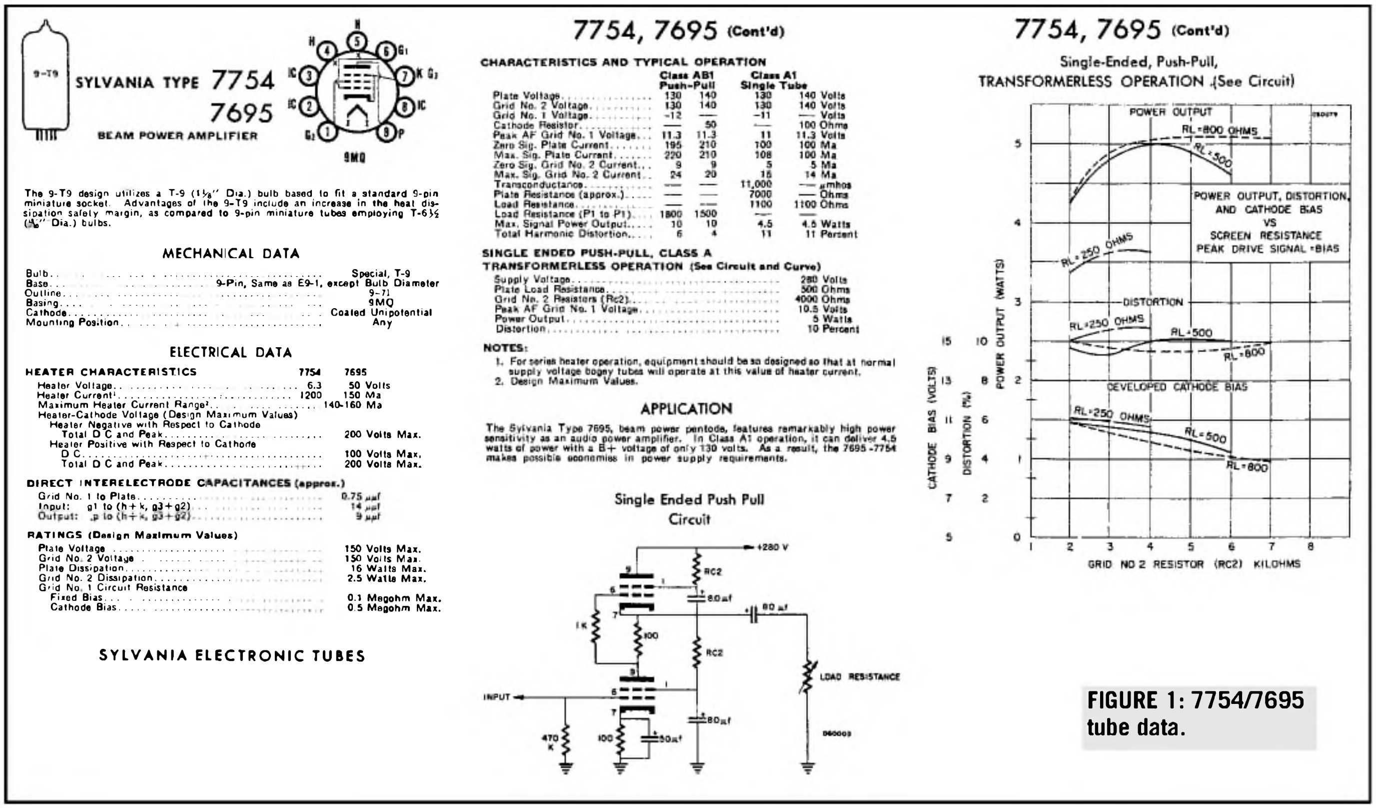
Vintage Glass

Sylvania Audio Amp

I found this diagram in my 1962 Sylvania tube manual. I thought *audioXpress* readers might be interested in it. Since I am new to audio amplifiers, I never heard of a single-ended push-pull amp, which this is. As stated in the data, it can deliver 4.5W at 130 plate voltage in a Class A1 single tube. The only problem I see is that the 7695 tube takes 50V on the heater. The 7754 is a 6V heater. Antique Electronic Supply in Arizona does not list a 7754. ❖

Thomas Sheckels
Rockdale, Tex.

The 14th edition of the Sylvania Technical Manual *is available from Old Colony Sound Lab (PO Box 876, Peterborough, NH 03458-0876, 603-924-6371, Fax 603-924-9467, e-mail custserv@audioXpress.com, BKAA56, $19.95).*



SYLVANIA TYPE 7754 7695

BEAM POWER AMPLIFIER

The 9-T9 design utilizes a T-9 (1⅛″ Dia.) bulb based to fit a standard 9-pin miniature socket. Advantages of the 9-T9 include an increase in the heat dissipation safety margin, as compared to 9-pin miniature tubes employing T-6½ (⅝″ Dia.) bulbs.

MECHANICAL DATA

Bulb	Special, T-9
Base	9-Pin, Same as E9-1, except Bulb Diameter
Outline	9-71
Basing	9MQ
Cathode	Coated Unipotential
Mounting Position	Any

ELECTRICAL DATA

HEATER CHARACTERISTICS	7754	7695
Heater Voltage	6.3	50 Volts
Heater Current[1]	1200	150 Ma
Maximum Heater Current Range[2]		140-160 Ma
Heater-Cathode Voltage (Design Maximum Values)		
Heater Negative with Respect to Cathode		
Total D C and Peak		200 Volts Max.
Heater Positive with Respect to Cathode		
D C		100 Volts Max.
Total D C and Peak		200 Volts Max.

DIRECT INTERELECTRODE CAPACITANCES (approx.)

Grid No. 1 to Plate	0.75 μμf
Input: g1 to (h+k, g3+g2)	14 μμf
Output: p to (h+k, g3+g2)	9 μμf

RATINGS (Design Maximum Values)

Plate Voltage	150 Volts Max.
Grid No. 2 Voltage	150 Volts Max.
Plate Dissipation	16 Watts Max.
Grid No. 2 Dissipation	2.5 Watts Max.
Grid No. 1 Circuit Resistance	
Fixed Bias	0.1 Megohm Max.
Cathode Bias	0.5 Megohm Max.

SYLVANIA ELECTRONIC TUBES

7754, 7695 (Cont'd)

CHARACTERISTICS AND TYPICAL OPERATION

	Class AB1 Push-Pull		Class A1 Single Tube	
Plate Voltage	130	140	130	140 Volts
Grid No. 2 Voltage	130	140	130	140 Volts
Grid No. 1 Voltage	-12	—	-11	— Volts
Cathode Resistor	—	50	—	100 Ohms
Peak AF Grid No. 1 Voltage	11.3	11.3	11	11.3 Volts
Zero Sig. Plate Current	195	210	100	100 Ma
Max. Sig. Plate Current	220	210	108	100 Ma
Zero Sig. Grid No. 2 Current	9	9	5	5 Ma
Max. Sig. Grid No. 2 Current	24	20	15	14 Ma
Transconductance	—	—	11,000	— μmhos
Plate Resistance (approx.)	—	—	7000	— Ohms
Load Resistance	—	—	1100	1100 Ohms
Load Resistance (P1 to P1)	1800	1500	—	—
Max. Signal Power Output	10	10	4.5	4.5 Watts
Total Harmonic Distortion	6	4	11	11 Percent

SINGLE ENDED PUSH-PULL, CLASS A TRANSFORMERLESS OPERATION (See Circuit and Curve)

Supply Voltage	280 Volts
Plate Load Resistance	500 Ohms
Grid No. 2 Resistors (Rc2)	4000 Ohms
Peak AF Grid No. 1 Voltage	10.5 Volts
Power Output	5 Watts
Distortion	10 Percent

NOTES:

1. For series heater operation, equipment should be so designed so that at normal supply voltage bogey tubes will operate at this value of heater current.
2. Design Maximum Values.

APPLICATION

The Sylvania Type 7695, beam power pentode, features remarkably high power sensitivity as an audio power amplifier. In Class A1 operation, it can deliver 4.5 watts of power with a B+ voltage of only 130 volts. As a result, the 7695-7754 makes possible economies in power supply requirements.

Single Ended Push Pull Circuit

7754, 7695 (Cont'd)

Single-Ended, Push-Pull, TRANSFORMERLESS OPERATION (See Circuit)

FIGURE 1: 7754/7695 tube data.

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If you are a tube enthusiast, then you probably have information that would be valuable to *audioXpress* readers. *audioXpress* encourages your feedback, and we'll pay you a modest stipend* for your efforts. Contribute to the wealth of information in this industry by sending your "Glass Shard", and any accompanying photographs or artwork to:

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