AN INTERVIEW WITH THE MULTI-TALENTED AUDIO ENTHUSIASTS BEHIND "TOYMAKER TELEVISION," p. 27

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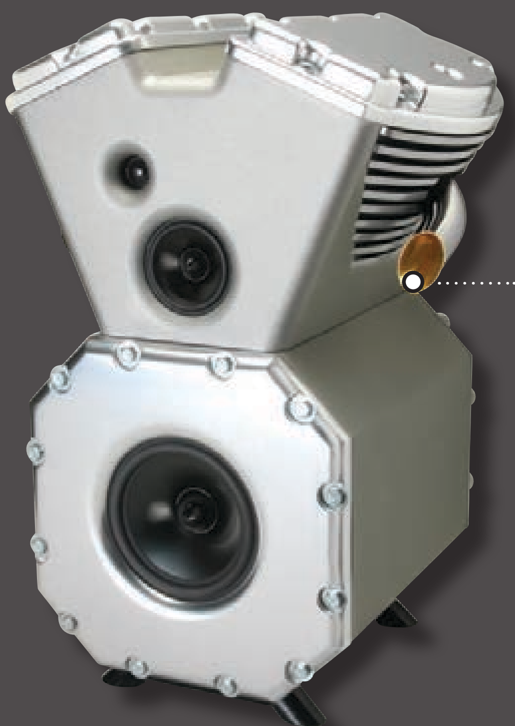

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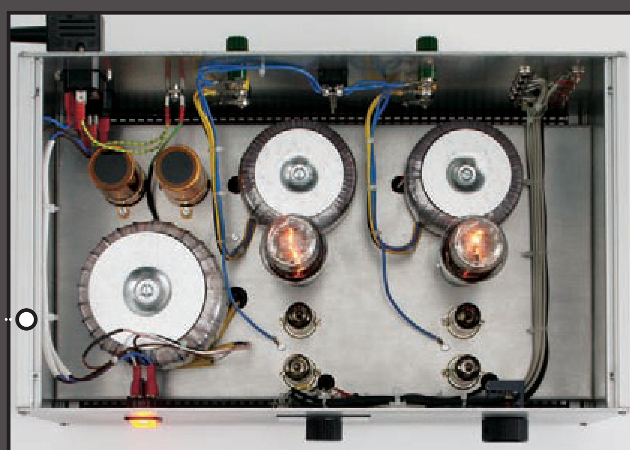

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Innovative Audio Desgin

A few months ago, while reading electronics-related tweets on Twitter, I discovered Toymaker Television (@tymkrs), whose profile quickly caught my attention: "Electronics, programming, avs, building musical instruments, and whatever else we get interested in." Obviously, my interest was piqued and I started following their tweets. Soon thereafter, I checked out their videos and found myself excited about many of their projects: a DIY cigar box guitar, a DIY mic pre/mixer, a SID organ, a laser audio transmitter, and more. This month we feature an interview with the duo behind Toymaker Television (p. 27). I hope their stories and enthusiasm for electronics and music inspire you to start new projects of your own! If so, be sure to contact our editorial department and let us know what you're working on. E-mail us or contact us via @audioXP_editor.

As usual, this issue also features intriguing articles about a variety of other subjects: speaker design, driver testing, an amp project, and glass audio theory.

On page 8, Mike Klasco examines the voice coil. He covers production techniques, bobbin strength, driver temperature limits, wire conductor materials, and more.

On page 14, David Pellegrene writes about his innovative Panheads speaker building project. The design looks cool and sounds awesome.

Turn to page 18 to read about the drivers Vance Dickason recently put to the test: a BMS 15" 15N850V^2 neodymium extended low-frequency woofer and a BMS coaxial 2" 4592ND compression driver. As you'll read, he used a DDS CFD 2-60A Pro 2" 60 × 40 horn for the 4592ND.

Most audio enthusiasts have grappled with the following questions: Tubes or transistors? Single-ended or push-pull? Wim de Jager surely has. Take a look at his a Class-A triode push-pull amplifier project on page 30.

Columnist Richard Honeycutt wraps up the issue with a focus on small-signal amplifiers, (p. 37). He covers the topics of amplifier configurations, vacuum tube biasing, and the voltage gain of amplifier stages.

With all of this content to read (and some Toymaker videos to watch), you're sure to be quite busy until our next issue hits the newsstands. Enjoy!

Regards,

C. J. Abate
editor@audioxpress.com

MEMBER PROFILE

Member Name: Joseph D'Appolito

Location: Wolfeboro, NH, USA

Education: Joseph holds a BEE (Rensselaer Polytechnic Institute), an SMEE and EE (Massachusetts Institute of Technology), and a PhD in electrical engineering (University of Massachusetts)

Occupation: Joseph is president of D'Appolito Laboratories, Inc., which provides test and design services in audio and acoustics, specializing in loudspeaker system design for two-channel and home theater applications.

Member Status: Joseph has been a regular contributor to *audioXpress* and its predecessors *Audio Amateur* and *Speaker Builder* since 1972.

Audio Interests: Joseph is interested in loudspeaker and crossover design, both active and passive. He does home theater design, design and construction of preamps, amplifiers, and other active circuits. He also enjoys classical music.

Most Recent Purchase: An Oppo BDP-95 Universal 3-D Blu-ray disc player

Current Audio Projects: Joseph is currently preparing an article on a closed-box subwoofer for home theater application.

Dream System: Joseph said: "Being in the business, my system is constantly changing. Presently my system comprises an Anthem D1 processor, an Usher P-303 two-channel preamp, and Bryston 4B and 9BSST amplifiers. My program sources include the EAD Ultra Disc CD deck, Denon 5900 and the Oppo. Loudspeakers, which are all my own design, include Snell's Phantom B7, D7 and K7, Usher BE-718 Diamond, and S520 with subwoofers. My home theater subwoofer will appear in a forthcoming article."

 By Mike Klasco and Steve Tatarunis (United States)

The Voice Coil (Part 1)
Parts and Production

A look at the bobbin, the voice coil wire, and the collar

In this month's installment, we're going to discuss the voice coil. The voice coil comprises three major components: the bobbin (also called the former), the voice coil wire, and the collar. The bobbin provides a rigid structure on which the voice coil wire can be wound and the collar can serve several purposes. It secures the coil lead-out wires, reinforces the bobbin, and provides a convenient material for diaphragm attachment (see **Photo 1**).

In some cases—headphone speakers, for example—a monolithic (self supporting no bobbin or collar) voice coil may be used. But this article will focus on the more commonly used bobbin, coil, and collar designs.

Loudspeaker voice coils are seldom considered critical elements that contribute to sound quality, and few technical papers have addressed this issue. But when designing a voice coil, the selection and application of materials can have profound effects upon sound quantity, quality, and power handling. The mechanical energy from the winding stack moves by transconduction through the bobbin and collar before reaching the diaphragm. Any non-linearities in this path are superimposed upon the response of the speaker. Intrinsic characteristics of materials such as internal damping and Young's modulus create specific sonic signatures and contribute to the residual distortion spectrum of the transducer (see **Photo 2**).

In selecting a particular material, a coil winder makes important trade-offs on the winding process. Knowledge of these variables can ensure better, more cost-effective coils, avoid conflicts, and improve production yields. Torsional resonances, internal losses, and electrical conductivity of the bobbin materials are some of the factors effecting the distortion, sensitivity, and sound quality of the finished loudspeaker. If the bobbin's wall strength is inadequate, the materials may flex, creating non-linearities, or they may crease or collapse. Virtually all coil components will soften significantly when heated, creating errors, and if the temperature exceeds their limits, they degrade to the next level of distress, which may be outgassing, blistering, embrittlement, deformation, or burning.

PRODUCTION TECHNIQUES

Most bobbin and collar materials are coated with a thin layer of adhesive on one side, and then dried at low temperature without curing the adhesive. These materials are cut to the required dimensions from master rolls or large sheets of material. Although some manufacturers cut materials in-house, the majority of them receive it cut into "blanks" by a converter. These voice coil blanks are wrapped around tubular steel or aluminum winding-mandrels, at which point their name changes from blank to bobbin. The former is wetted with solvent by brush, by spray, or by in-line wetting of the insulated, adhesive-coated wire. After the wire is wound onto the bobbin, the solvent then activates the adhesives, creating a secure bond between all the components (see **Photos 3** and **4**).

BOBBIN STRENGTH

During sound reproduction, the cone resists movement by the actions of inertia, the air mass load on the cone surface, and tightness of the back chamber, including mechanical losses of the suspension elements. If these restrictive forces acting on the cone exceed the mechanical strength of the former, or of the adhesive bonds, then the speaker will fail. Recent subwoofer designs, employing very stiff suspensions, have created unprecedented stress on the former.

To prevent bobbin crushing, design engineers specify appropriate bobbin and

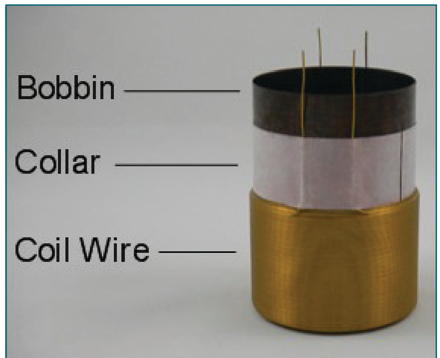


Photo 1: The construction of a voice coil (Source: Precision Econowind)


Photo 2: High-performance, flat wire coils from Precision Econowind (Source: Precision Econowind)


Photo 3: A voice coil goes through the winding process (Source: Po Yun Enterprise)

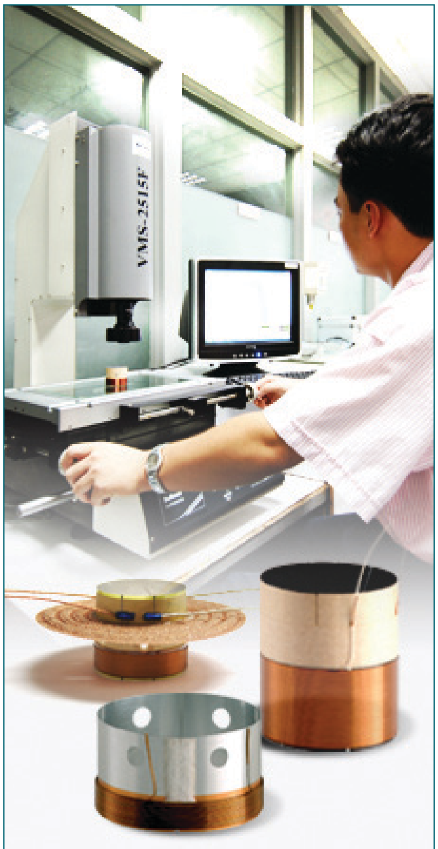


Photo 4: A worker inspects the voice coils (Source: Po Yun Enterprise)

Photo 5: A black bobbin voice coil (Source: MadeInChina.com)

Photo 6: A Kapton bobbin voice coil (Source: Precision Econowind)

collar materials in a thickness calculated to provide adequate wall strength that will endure some abuse. When drivers are operated at high-power levels, the temperature of the voice coil can be greater than 200°C. Clearly the mechanical stiffness of the former and collar materials must be considered at both room temperature and at elevated temperatures. At elevated temperatures, all materials soften to a degree. This is especially noticeable with Kapton (polyimide) which softens significantly at extreme elevated temperatures in the mid-200°C range, resulting in deformation or collapse of the former immediately above the winding. Lower-grade polyimides become soft at even lower temperatures.

Mechanical property considerations for formers include: dimensional stability under varying conditions of temperature and humidity, material mass, (specific gravity/weight), stiffness (torsional and flexural modulus), strength (tensile and compressive strength, plastic deformation limit and ultimate yield limit), fabrication and handling criteria, material fatigue, brittleness, and intrinsic damping characteristics.

THERMAL CONSIDERATIONS

The real temperature limit of a driver may be determined by the failure or softening of any of the following: bobbin material, bobbin adhesive, wire insulation, wire adhesive, adhesive to cone, spider, or dome, or the materials of cone, spider, or dome.

Wire insulation, bobbin or collar materials, or adhesives with upper temperature limits of less than 180°C are not suitable for high-power drivers, but they are appropriate when the input power to the driver is a known and controlled factor, such as a self-amplified subwoofer. Where high-input power is anticipated and not controlled, materials rated for greater than 200°C are required. Because of their graceful over-temperature behavior, some types of 180°C wire are still used. Although voice coil materials and adhesives may tolerate exposures to temperatures well beyond their nominal rating, such exposures tend to have a cumulative effect.

Former thermal conductivity has both positive and negative aspects. On the positive side, it sinks heat from the coil, spreading it over a wider area, cooling the winding, which is always good. The most effective heat path in thermally conductive materials is through the thin film rather than along it, allowing it to transfer heat to the steel pole piece more efficiently, reducing the coil temperature without endangering the cone-coil-spider bond or plastic cone materials. Thermal conductivity also spreads the heat from hot regions over a larger area. Various black materials such as black anodized aluminum, black Kapton (MTB and variants), and black composites exploit the "black body effect," shedding heat more efficiently than pale colored materials. The effectiveness of this emissive cooling is relatively small at approximately 5% (see **Photo 5**).

THE ADHESIVE BONDS

The adhesive coating on the wire and the adhesive coating on the bobbin must be solvent compatible. Compatibility between former material and coating adhesive is more commonly an issue. The low surface energy of polyimide films such as Apical and Kapton, together with their smooth, pore-free surfaces makes them a challenge for any adhesive. Although a driver manufacturer may specify a specific former material, the adhesive coating rarely receives the attention it deserves. Yet, it has dramatic effects on power handling and reliability. At the other end of the scale, Kraft, press, and bond papers have highly porous surfaces, with intrinsically high adhesion. But, their propensity to embrittle and adhesive wicking offset this benefit (see **Photo 6**).

There is also a thermoplastic flow failure, a classic catastrophic failure mode, producing rapid destruction of the coil structure, as windings slide around, or become loose like "slinky" toys. This failure mode is often seen in wire-to-wire bonds within the winding. One contributing factor is butyl rubber-modified phenolic adhesives, which are strong at room temperature, but their strength drops in a predictable, almost linear manner with increasing temperature. Their thermoplastic flow points run from 160°C for elementary types, approximately 190°C for better wire and competitive types, approximately 220°C for highly cross-linked types.

MAGNET WIRE INSULATIONS

The insulation and adhesive coatings must be removed (stripped) to allow electrical termination of the voice coil to

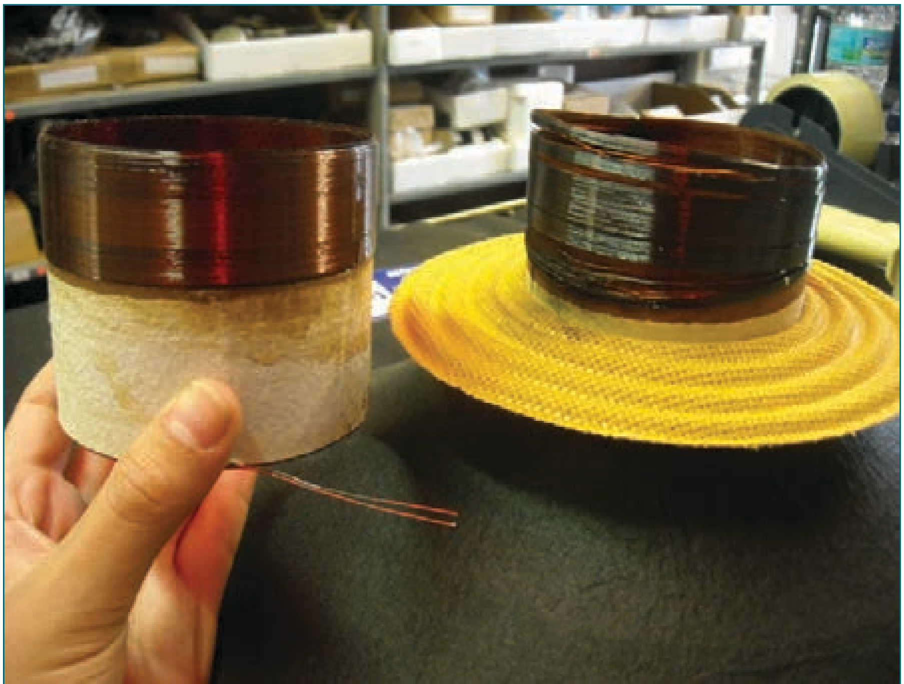


Photo 7: A close-up view of both a good voice coil and a burned voice coil (Source: The Speaker Exchange)

the tinsel lead-wires. Stripping of these coatings can be performed with chemical agents, a hot solder pot, boiling caustic salts, or a mechanical abrasive. Another option may be ultrasonically welding the wire through the coating, or connecting to the tinsel wire with an insulation piercing crimp.

Dipping the wire ends into a pot of molten solder, which also simultaneously tins them, can strip polyurethane and polyester-based insulations. Higher temperature wire requires stronger treatments, such as dipping in boiling fused salts, which are available from the Eraser Co., New York (315-454-3237). Corrosive chemicals can also be used to strip high-performance wire insulations.

There are arguments, occasionally battles, between designers, who need to use 220°C wire to get close to the power specifications that the marketing department writes, and the manufacturing folks, who want to use 155°C wire because "it strips so nicely." Sadly, wire that strips nicely also fails nicely (see **Photo 7**).

Photo 8: A close-up view of an aluminum flat wire (Source: HM Wire International)

DRY WINDING WITH BOND COATED INSULATED WIRE

The majority of voice coil winders find "dry" winding to be the simplest and most economical technique for producing consistently good voice coils. Dry winding employs insulated wire pre-coated with a thermoplastic adhesive, which is reactivated with solvent during winding. It is then baked in an oven, or heated by applying current to the coil, which produces the good general purpose voice coils. Thermoplastic bond-coats allow fast, and fairly clean, winding and processing. But since they re-soften at high temperatures, they cannot tolerate high power.

WET WINDING WITH INSULATED WIRE

As an alternative to the "dry" technique, some coil manufacturers have had success using a technique involving simple insulated wire, which is passed through a bath of wet adhesive during winding. An almost perfect film thickness is achieved by drawing the wet wire through a precision "bullet" extrusion die, although pulling it through a simple felt pad saturated with adhesive is almost as effective. Wet winding allows the choice of adhesives that have been specifically engineered for loudspeaker voice coils, instead of transformer and electric motor industry bond coats. Wet wound coils are acknowledged to be superior to "dry" types, and wet winding is practiced extensively in Europe. But since it requires some operator skill, it is not as popular as it should be. Wet winding provides an excellent wire-to-wire bond, and the highest structural strength when used with adhesive-coated formers.

WIRE DIMENSION

In the United States, wire follows the American Wire Gauge (AWG) scales, which provides convenient increments between sizes, while outside the United States, the metric (IEC) scale dominates. The dimensions of some AWG sizes closely parallel those of IEC wire, and over the range that interests us in the loudspeaker industry, there are near perfect equivalents.

WIRE CONDUCTOR MATERIALS

Copper is the dominant material for voice coils wire due to its high conductivity and widespread availability. The vast majority of copper wire is used in electric motors and transformers, so it is produced in many gauges. Although copper is heavy, it is inexpensive (although it has gone through huge fluctuations during the last decade). Dead-soft annealed copper is the reference standard for conductivity, against which all other metals are measured, so it naturally rates 100% on the International Annealed Copper Standard (IACS) scale. High purity, oxygen-free, and long-crystal copper can rate up to 102% IACS. Copper wire is available insulated with hundreds of different resins, and some are available with a bondable over-coat. Some insulation coats are strong and flexible enough to allow the wire to be flattened (milled) into ribbon profiles. Several manufacturers produce voice coil wire in high-aspect ratios, by starting with a bare conductor, which is highly malleable. They mill it to the desired profile, then apply the insulation and bond coats, but this approach requires a substantial investment in equipment.

Aluminum and copper-clad aluminum are options in voice coils where mass is an issue. Like copper, aluminum wire can be milled into rectangular profiles, although it requires subsequent heat treatment to relieve the work hardening induced by the milling process. For extreme power applications, where wire insulation enamels may be a limiting factor, aluminum conductors can be anodized with a 0.05-mm layer of anodized insulation on all its surfaces (see **Photo 8**).

Copper-clad aluminum (CCA) wire is a combination of an aluminum conductor, clad with a thin outer layer of copper for easier termination. CCA wire is produced with 10% (USA) and 15% (Japan) copper by volume, equating to 25 to 37% by weight, making it appreciably heavier than pure aluminum. CCA wire can be milled into flat profiles, or "ribbon" wire, and is then termed CCAR. All of the normal insulation enamels and adhesives are compatible with aluminum conductors, but few varieties are offered as standard products.

VOICE COILS CONTINUED

Next month, we continue our exploration of voice coils, including secrets of the impact of the materials on sound quality. *aX*

 By David Pellegrene (United States)

The Panheads

An innovative speaker design resembles a Harley-Davidson engine

About two months before Christmas, I got a phone call from my sister-in-law. She wanted me to build a set of speakers for her to give to her husband for Christmas. She wanted them to fit behind his bar. Knowing there wasn't much time, I agreed to do it anyway. He is a big Harley enthusiast, and I wanted to do something that would fit the Harley theme. I wasn't sure what to do, so I researched Harley motorcycles. It didn't take long to figure out that the engine could be built to resemble a speaker. Long story short, I built the "Knuckleheads" and had them ready for her by Christmas. I then decided I wanted to build another set of speakers and spend a little more time designing them. I chose to build the "Panheads" (see **Photo 1**), which I wanted to resemble a motorcycle engine a little bit more. I also wanted to use available drivers not buy outs.


Photo 1: The Panheads speaker system resembles a motorcycle engine.

THE DESIGN

I ended up choosing all Dayton drivers and decided on a three-way design. For base duties, I chose the RS180-8. I went with a vented cabinet of 16 liters tuned to 46 Hz—basically, the bottom of the motor. For mid duties, I chose the RS100-8 and put it in a 5-liter sealed enclosure. This would be the upper part of the motor that would be the cylinders or jugs. The tweeter I used was the ND20FA. I'd used it on a few other designs and, for the money, really liked it. I chose to mount all the drivers on the rear side of the baffle for aesthetic reasons. I also wanted to physically time align the drivers, which mounting them on the rear of the baffle made it easier to do. The woofer was mounted on the backside of the 0.75" baffle, the mid cabinet "jugs" set back 0.75" from the woofer cabinet and was recessed 0.375" into its 0.75" baffle to align with the woofer. The tweeter was mounted to the rear of the same baffle as the mid to align it. I then used a 0.75" round over on the tweeter opening, which made it a small waveguide.

I used the exhaust pipes on each side of the sealed mid cabinet for the ports. I used a 4" tee, (see **Photo 2)** in order to get the vent from the woofer cabinet up into the sealed mid cabinet and split into two vents to exit. I cut a 4" hole in the top of the woofer cabinet (see **Photo 3**), and mounted the tee to the hole which put the tee inside the mid cabinet. On the ends of the tee, I added a piece of 4" pipe that fit tight to the sides of the mid cabinet. There, I cut a 1.5" hole in the side of the mid cabinet to keep the chrome pipe that fit to the outside of the cabinet in line with the 4" pipe. Using my WT3, I cut the chrome pipe until I got the tuning I was after. I wanted the 4" pipe large enough to add volume to the woofer cabinet but not to act as a port. From what I modeled previously, it appears the 4" pipe did play a part in the tuning.

THE CROSSOVER DESIGN

I took measurements of all the drivers in the cabinet to start the crossover design. I used PCD for my initial modeling. Crossover points are 800 Hz and 2.8 kHz. I then sent my results and files to a much more capable crossover designer, Curt C., who is a regular on the Parts Express forum. He tweaked the design for me (see **Figure 1**). After measuring, quite a few listening sessions, and adding some padding on the mid and tweeter, I ended up using the tweaked design.

THE PAINT SCHEME

The only thing left to decide was the paint scheme. The cabinets are built with MDF. The most simple and least expensive way I found to prep MDF for paint is to use several coats of Bin 123 alcohol-based primer. This is basically white-pigmented shellac. I sanded them with 180-grit paper. I followed the primer up with a coat of sprayed on black rustoleum oil-based paint. Again, it is an inexpensive sealcoat. This coat will show any imperfections I might have missed so I sanded it with 400-grit paper. For the final finish, I was looking for something a little different than the black I used on the Knuckleheads. Another Parts Express forum member told me about Aluma Blast paint, which gives the appearance of a freshly sandblasted aluminum finish, similar to a car alternator (see **Photo 4**). I really liked the look so I

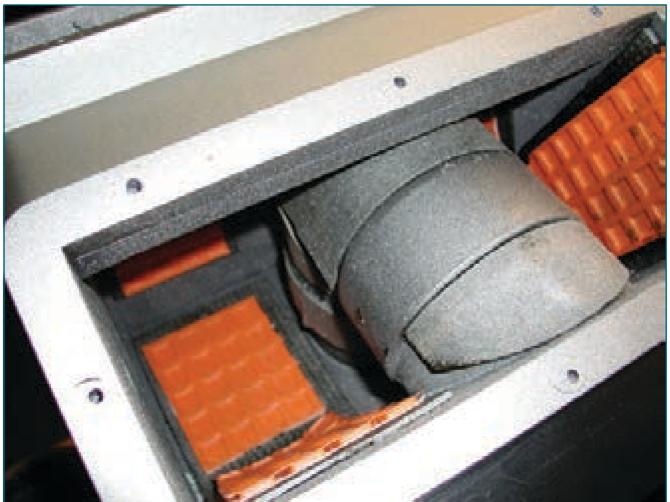


Photo 2: This is a top view of the sealed section of the mid cabinet, showing the 4" tee, attaching the woofer cabinet to the 1.5" chrome exhaust pipes.

Photo 3: This is the woofer cabinet with 4" pipe leading to the 1.5" chrome vent tubes.

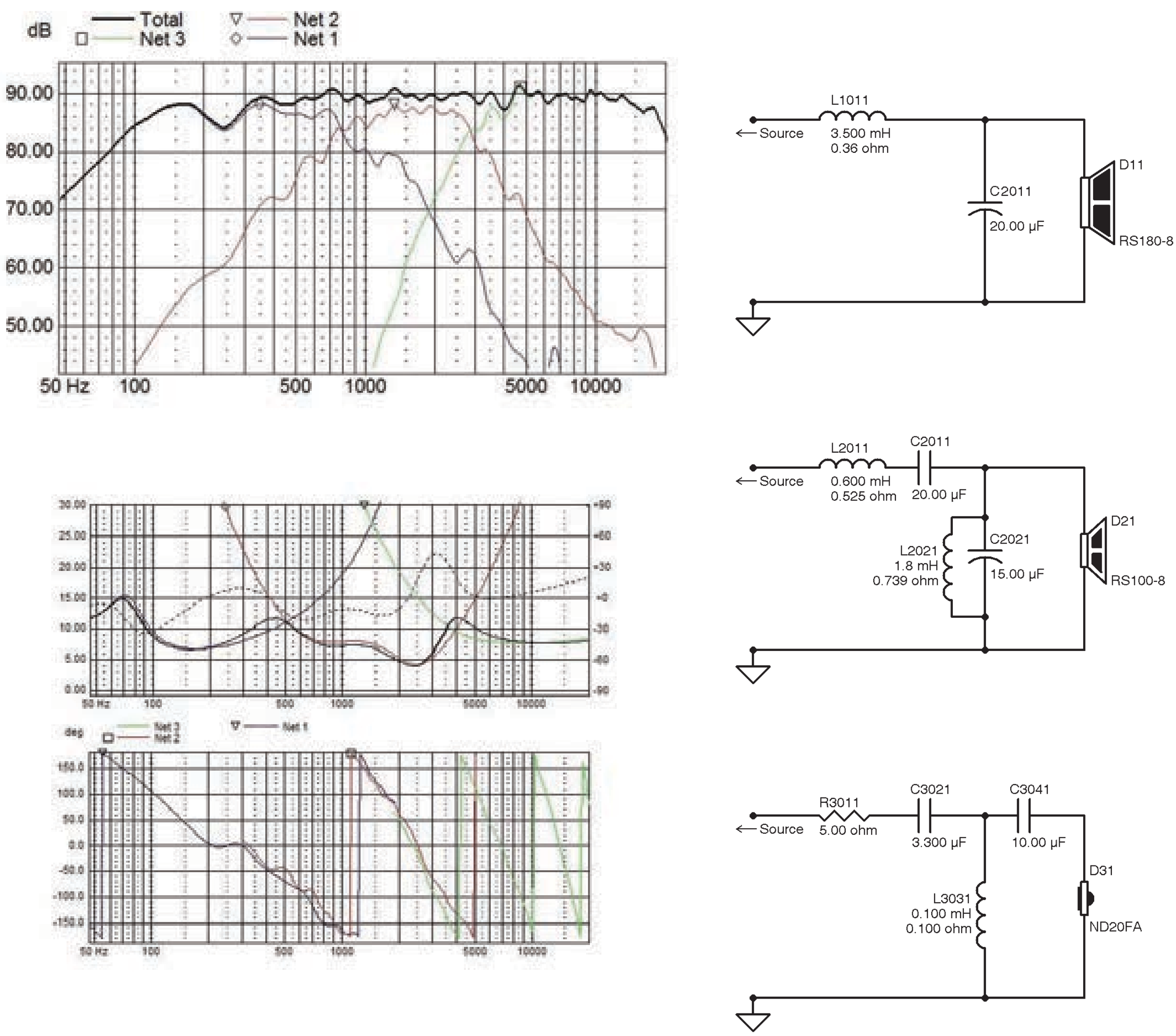


Figure 1: Crossover graph, impedance graph, phase graph, and crossover layout schematic

Photo 4: A close-up picture of the Aluma Blast paint

Photo 5: Sections of the speaker coated with Dupli Color chrome spray paint

Photo 6: Pictured are dowel rods, glued into holes in which to thread lag screws

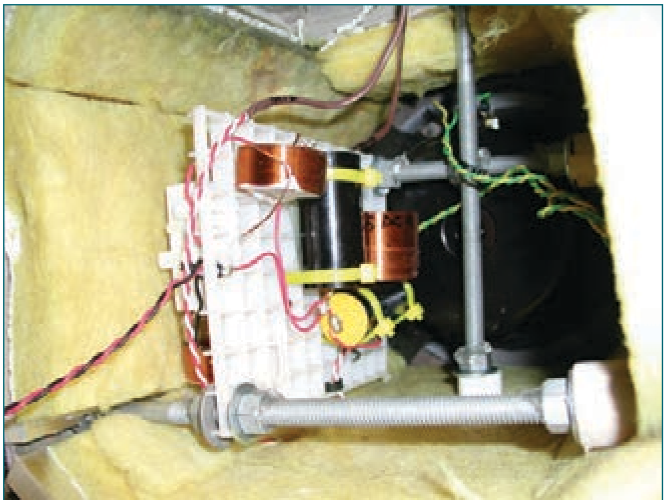


Photo 7: The crossover is mounted on threaded rod bracing

used it on the main part of the motor. The baffles and valve covers were painted with Dupli color chrome paint. It gives a polished aluminum look (see **Photo 5**). The downside to the chrome paint is that you cannot wipe it or rub the finish; otherwise, it will leave a gray smudge that cannot be removed. Any clear coat will change the finish to a gray color so I wouldn't recommend it, unless you want to repaint on regular basis.

THE CABINETS

The cabinets are built of 0.75" MDF. The bass box is mitered with all 22.5° angles glued together using ratchet straps. The baffles and valve covers are installed with lag screws. In order to keep the lag screws from splitting the MDF I drilled 0.75" holes in the MDF and glued 1.25"-long dowel rods into the holes. I pre-drilled holes in the dowels to thread in the screws (see **Photo 6**). I used a 0.125" rubberized material for gaskets. I also used the rubberized material, Ditra by Schluter (the orange material in **Photos 2** and **3),** for cabinet damping. I then covered the Ditra with a layer of Thinset concrete mortar. This made for very solid cabinet walls. For extra stiffening, I used one vertical and two horizontal pieces of All Thread, with spacers on the ends, to apply pressure to the side walls of the woofer section. For the crossover board, I used a piece of suspended ceiling light panel, mounted to the threaded rod. This allowed me to float the crossover board away from all the walls for good air flow. The bracing and crossover can be seen in **Photo 7**.

POSITIVE RESULTS

This was a challenging and very fun build. I am very happy with the results, not only visually but also sonically. The Panheads were well received at a recent audio event. A couple of the comments made included that the speakers image well and that they are also very well balanced.

I think next up is going to be the dragsters. I believe these will be a small book shelf/desk-top design. Thanks for reading. *aX*

Editor's Note: With this design, David Pellegrene won third place in the Midwest Audio Fest 2011 Speaker Design Competition hosted by Parts Express.

By Vance Dickason (United States)

Two BMS Drivers

A test of two high-end home audio drivers

This month's Test Bench drivers were both submitted by well-known German Pro Sound transducer manufacturer, BMS. The first driver I examined was the BMS 15" 15N850V^2 neodymium extended low-frequency woofer, followed by a unique product, the BMS coaxial 2" compression driver, the 4592ND. The horn I used this month for the BMS 4592ND was provided by DDS, the CFD 2-60A Pro 2" 60 × 40 horn.

BMS was founded in 1994 in Hanover, Germany. In a relatively short time (short time when compared to JBL that was founded in 1946), it has developed a world-wide reputation for sound quality and reliability. Their customer list is impressive and includes JBL, Genelac, Yorkville, EAW, Tannoy, and L Acoustics to name a few. The company's R&D, sales and marketing, and warehousing are located in Hanover with its primary factory located in Sofia, Bulgaria. ISO 9001-2000 certified since 2006, BMS is a highly vertically integrated company and manufactures the majority of its own driver parts; including voice coils, CNC machining, plastic parts forming, diaphragm forming, painting, and other manufacturing functions. This month, I received the 15" extended low-frequency neodymium motor woofer, the 15N850V2 that features a rather high-rated nominal power handling of 1200 W (AES).

The BMS 15N850V^2 (see **Photo 1a** and **1b**) is the second version of the 15" neo 15N850 released five years ago. Like its predecessor, the 15N850V^2 is built on a proprietary eight-spoke cast aluminum frame that is further CNC machined at the factory to guarantee better contact between the frame and the motor system for an enhanced thermal contact, supplemented by a special silicone compound for better heat transfer. The frame is also additionally machined on the spider mounting shelve to ensure a ±0.05-mm vertical dimension to avoid DC offset by getting an exact magnetic center location. For cooling, the frame has 48 5 mm × 13 mm vents located between the spider mounting shelf and the back of the frame. The cone assembly comprises a 15"-diameter, double-coated (for moisture resistance) curved profile, carbon fiber-filled paper cone with a 6"-diameter coated paper dust cap. Compliance is provided by a two-roll coated cloth "M" surround and a 6.5"-diameter cloth spider mounted on an aluminum ring. Driving this assembly is a 100-mm (4") diameter voice coil wound both inside and outside with round copper wire on a proprietary 0.22-mm thick glass fiber former. BMS refers to this as a "sandwich" voice coil. Heavy insulated tinsel lead wire are terminated to a pair of chrome color coded push terminals.

a)

b)

Photo 1a: The BMS 15N850V^2 top view. b: The 15N850V^2 bottom view

Testing for the BMS 15N850V^2 commenced using the LinearX LMS and VIBox to produce both voltage and admittance (current) curves with the driver clamped to a rigid test fixture in free-air at 1 V, 3 V, 6 V, 10 V, 15 V, 20 V, and 30 V. However, the device was still measuring very linear at 30 V and would likely have been good at 50 to 60 V, but with 94 to 95-dB sensitivity, it gets extremely loud, even with ear protectors. Going with the established protocol for Test Bench, the fixed Mmd method was used rather than the added mass or the test box method for obtaining the Vas of the driver. This data was provided by BMS transducer engineer Svetly Alexandrov.

Next, the 14 550-point stepped sine wave sweeps for each sample were post-processed and the voltage curves divided by the current curves to derive impedance curves, phase calculated and along with the accompanying voltage curves, imported to the LEAP 5 Enclosure Shop software. Obviously, this is a much more time consuming process that the usual single impedance low-voltage curve method used for deriving Thiele-Small parameters. (See **Figure 1** for the family of impedance curves created by this process.) The obvious thermal change is due to playing a 200-Hz sine tone at the sweep voltage level from 10 to 50 seconds between sweeps in order to approximate the third thermal time constant. The reason for using this method, if you haven't been following this column for a number of years, is that the LEAP 5 LTD model results in a much more accurate prediction of excursion at high voltage levels, which is one of the real fortes of the LEAP 5 software and why I use it for Test Bench testing.

Because most TS data provided by OEM manufacturers is being produced using either a standard method or the LEAP 4 TSL model, I additionally created a LEAP 4 TSL model using the 1-V free-air curves. The complete data set, the multiple voltage impedance curves for the LTD model (see **Figure 2** for the woofer 1-V free-air impedance curve) and the 1-V impedance curves for the TSL model were selected in the Transducer Derivation menu in LEAP 5 and the parameters created for the computer enclosure simulations. Table 1 compares the LEAP 5 LTD and TSL data and factory parameters for both BMS 15N850V^2 samples.

Parameter measurement results for the BMS 15N850V^2 showed fairly close agreement with the factory published data, with the exception of the sensitivity, the difference being that my data showed sensitivity at 2.83 V, while BMS uses the 1 W/1 m criteria with a 5.6-Ω Re that yields a somewhat larger number. Two other variances occurred. Given this, I followed my usual protocol and programmed computer enclosure simulations using the LEAP 5's LTD parameters for Sample 1. This included two vented alignments, both recommended in the BMS brochure for this driver, a 1.77 ft^3 QB3 box with 15% fiberglass fill material tuned to 43 Hz, and a 2.64 ft^3 Extended Bass Shelf (EBS) vented alignment enclosure with 15% fiberglass fill material and tuned to 35 Hz.

Figure 3 displays the results for the BMS 15N850V^2 in the QB3 and EBS vented boxes at 2.83 V and at a voltage level sufficiently high enough to increase cone excursion to Xmax + 15% (13.8 mm for the BMS 15"). This produced a –3-dB frequency of 56 Hz (–6 dB = 46 Hz) for the QB3 enclosure and F^3 = 48 Hz (F6 = 36 Hz) for the EBS-vented simulation. Increasing the voltage input to the simulations until the maximum linear cone excursion was reached resulted in 124 dB at 100 V for the QB3 enclosure simulation and 123.5 dB with the same 100-V input level for the larger vented box. (See **Figures 4** and **5** for the 2.83-V group delay curves and the 100-V excursion curves.) Note that the criterion for these high excursion SPL numbers was the maximum excursion at 25 Hz. If a 20-to-25-Hz high-pass filter were used, the SPL would higher.

Klippel analysis for BMS 15N850V^2 (my analyzer is provided courtesy of Klippel GmbH)—which as usual was performed by Patrick Turnmire, Red Rock Acoustics (author of the SpeaD and RevSpeaD transducer simulation software)—produced the Bl(X), Kms(X) and Bl and Kms Symmetry Range plots given in **Figures 6** to **9**. Like many of the pro sound companies that contribute samples to *Voice Coil* for this column, BMS has its own Klippel analyzer.

The Bl(X) curve for the 15N850V^2 (see **Figure 6**) is nicely broad and symmetrical typical of a high Xmax driver (12 mm is substantial for a pro sound 15" woofer) with a very small amount of coil out (forward) offset. Looking at the Bl symmetry plot (see **Figure 7**), this curve shows negligible 0.79 mm of forward (coil-out) offset in the 5-mm position (a more reliable number than at rest in this case) decreasing to about 0.1 mm, at the driver's physical Xmax of 12 mm, all of which is of no real consequence. **Figures 8** and **9** depict the Kms(X) and Kms symmetry range curves for the BMS 15N850V^2. The Kms(X) curve is also symmetrical with small amount of forward coil out offset. The Kms symmetry range curve shows a minor 0.27-mm forward offset at the 5-mm excursion point (where the data is more reliable), again decreasing to practically no offset at the physical Xmax for this driver. Displacement-limiting numbers calculated by the

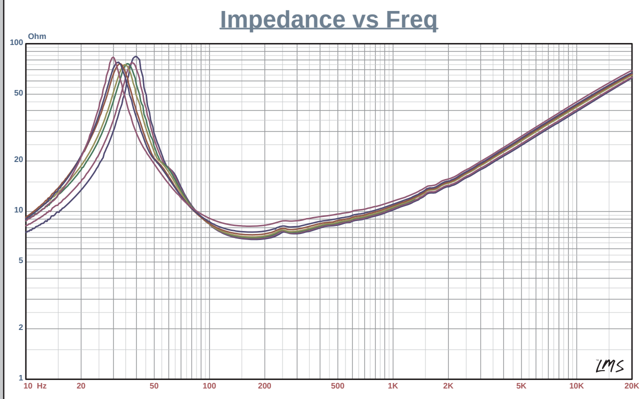


Figure 1: A family of BMS 15N850V2 impedance curves created from the LEAP 5 LTD parameter calculation process

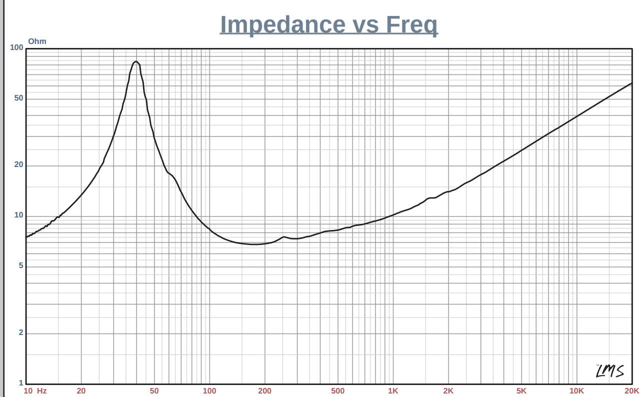


Figure 2: A BMS 15N850V^2 woofer free-air impedance plot

	TSL model		LTD model		Factory
	sample 1	sample 2	sample 1	sample 2	
Fs	39.0 Hz	42.0 Hz	37.7 Hz	40.3 Hz	37.0 Hz
Revc	4.84	4.73	4.84	4.73	5.6
Sd	0.085	0.085	0.085	0.085	0.0845
Qms	5.42	5.20	5.46	5.76	7.38
Qes	0.33	0.33	0.33	0.36	0.34
Qts	0.31	0.31	0.31	0.34	0.33
Vas	95.9 ltr	82.6 ltr	103.3 ltr	90.4 ltr	111.0 ltr
SPL 2.83 V	94.3 dB	94.5 dB	94.1 dB	94.0 dB	95 dB 1 W/1 m
Xmax	12 mm	12 mm	12 mm	12 mm	12 mm

Table 1: BMS 15N850V^2

Klippel analyzer for the BMS 15N850V2 was XBl @ 70% (Bl decreasing to 70% of its maximum value) was 14.5 mm (2.5 mm beyond Xmax!) and for XC @ 50% (compliance decreasing to 50% of its maximum value) was 13.8 mm, which means that for the BMS 15N850V2, the compliance is the limiting factor for the prescribed distortion level of 20% for using this driver as a subwoofer, and again, this number is beyond the physical Xmax of the driver. Data this good is about as good as it gets.

Figure 10 gives the inductance curve L(X) for the BMS woofer. Inductance will typically increase in the rear direction from the zero rest position as the voice coil covers more pole area unless the driver incorporates a shorting ring, and the BMS 15" incorporates a triple aluminum shorting ring format. This is readily apparent in the L(X) curve. As you can see, the inductance variation is extremely small, only 0.04 to 0.07 mH from rest to Xmax in either direction. Minimal inductance variation is critical to a low distortion and accurate sounding driver.

Even though the BMS 15" is intended as a low-frequency

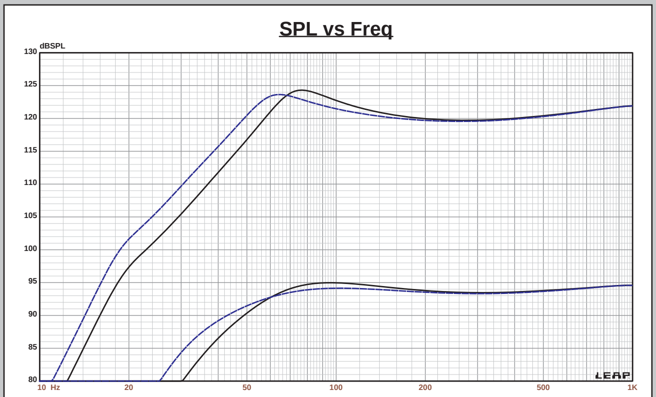


Figure 3: BMS 15N850V2 computer box simulations (black solid = vented 1 @ 2.83 V; blue dash = vented 2 @ 2.83 V; black solid = vented 1 @ 36 V; blue dash = vented 2 @ 36 V)

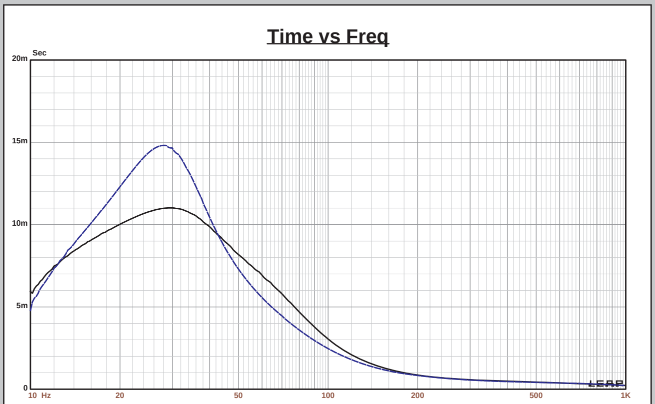


Figure 4: Group delay curves for the 2.83-V curves in Figure 3

extended woofer, a pro sound subwoofer category, this driver could of course be used in a multi-way speaker equally successfully. **Figure 11** gives the on-axis response for this driver, which is ±1.1 dB up to the about 1200 Hz with the high-pass roll off beginning about 2.5 kHz. The on- and off-axis response from 0 to 45° is displayed in **Figure 12**. The –3-dB frequency at the 30° off-axis response with respect to the on-axis response occurs at about 1.2 kHz, probably a good upper crossover point for this

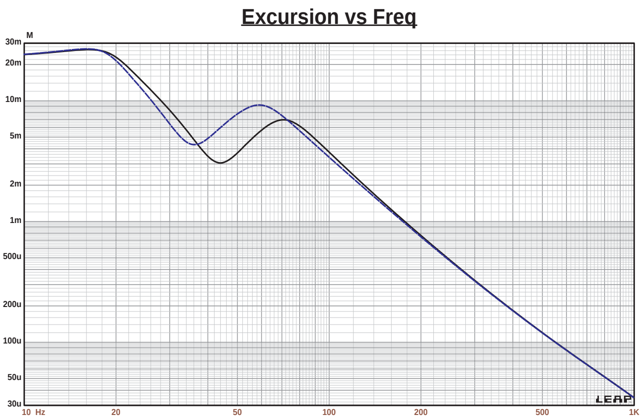


Figure 5: Cone excursion curves for the 100-V curves in Figure 3

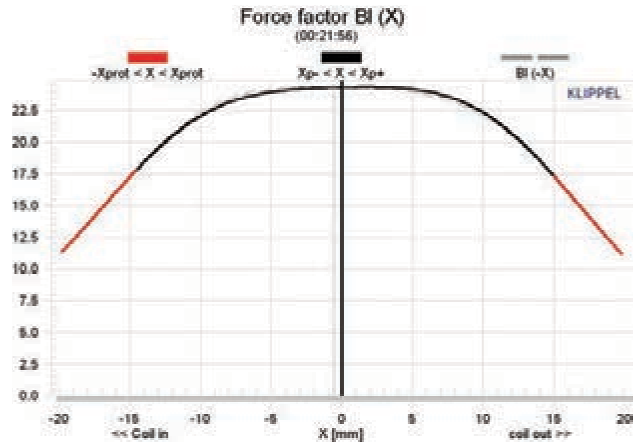


Figure 6: Klippel Analyzer Bl (X) curve for the BMS 15N850V2

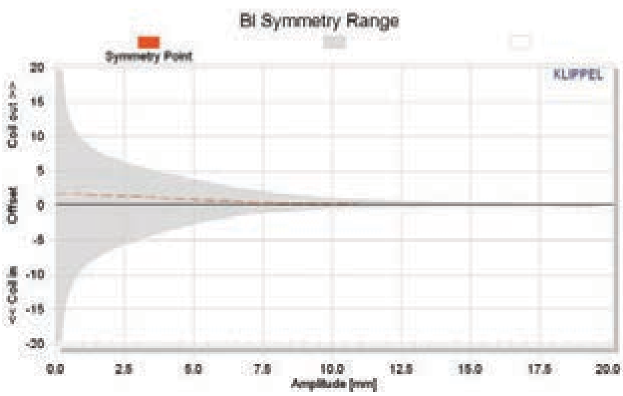


Figure 7: Klippel Analyzer Bl symmetry range curve for the BMS 15N850V2

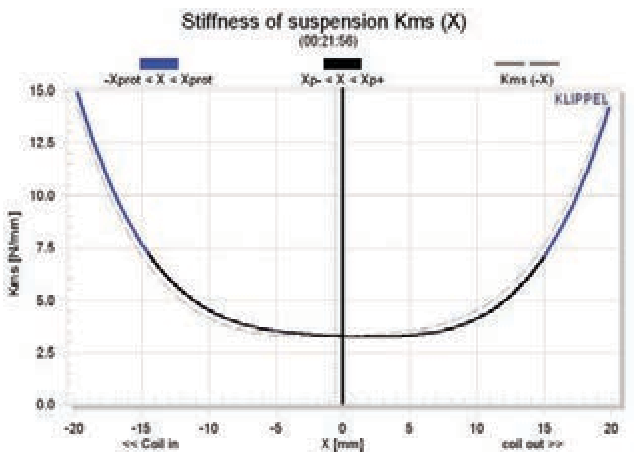


Figure 8: Klippel Analyzer mechanical stiffness of suspension Kms (X) curve for the BMS 15N850V2

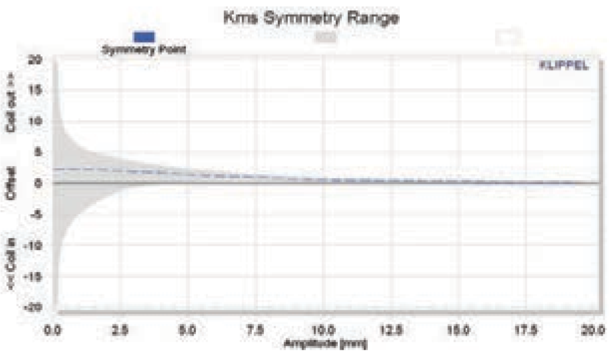


Figure 9: Klippel Analyzer Kms symmetry range curve for the BMS 15N850V2

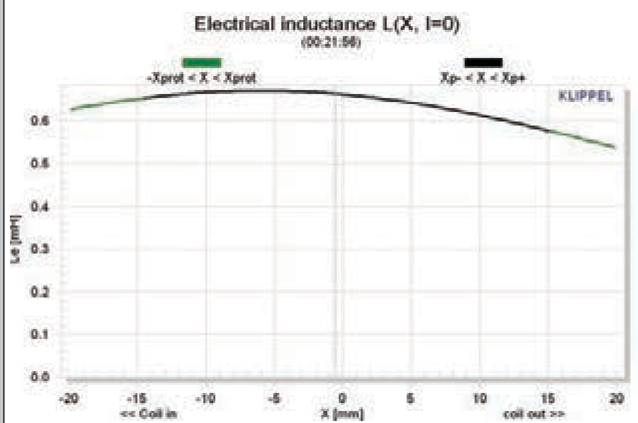


Figure 10: Klippel Analyzer L(X) curve for the BMS 15N850V2

driver.

For the last body of testing on the BMS 15" woofer, I used the Listen, Inc. SoundCheck analyzer and SCM microphone and power supply (courtesy of Listen, Inc.) to measure distortion. Since this driver is really meant for subwoofer applications, I dispensed with the time frequency plots. Setting up for the distortion measurement consisted of mounting the woofer rigidly in free-air, and the SPL set to 104 dB at 1 m (10.6 V) using a noise stimulus (SoundCheck has a software generator and SPL meter as two of its utilities), and then the distortion measured with the SCM microphone placed 10 cm from the dust cap. This produced the distortion curves shown in **Figure 13.**

The BMS15N850V² is the first woofer I have been able to evaluate from BMS, and as can be seen from the above analysis, plus the overall excellent fit and finish that I can observe, this is a very well-crafted product. I was especially impressed with the

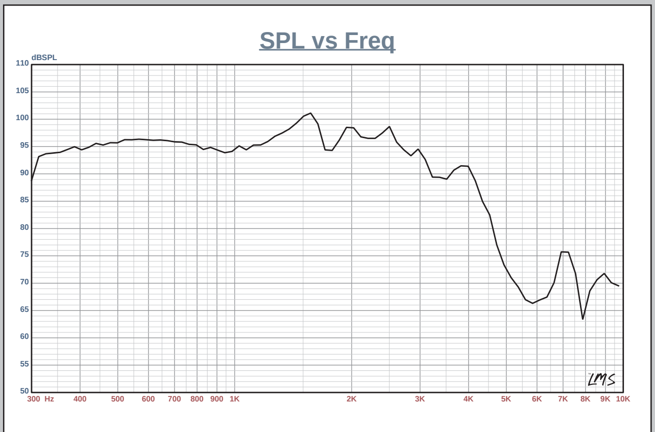


Figure 11: BMS 15N850V² on-axis frequency response

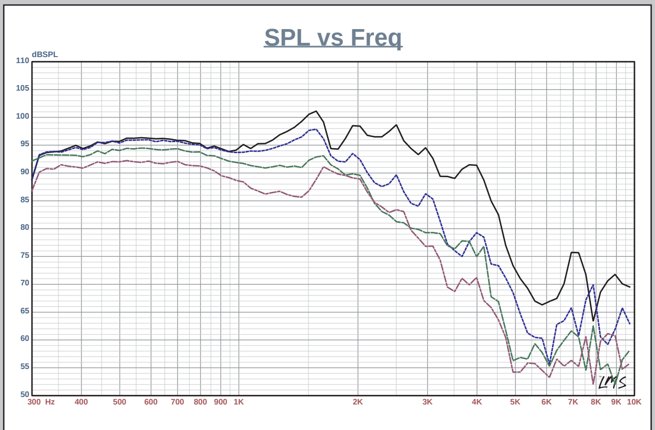


Figure 12: BMS 15N850V2 on-axis and off-axis frequency response

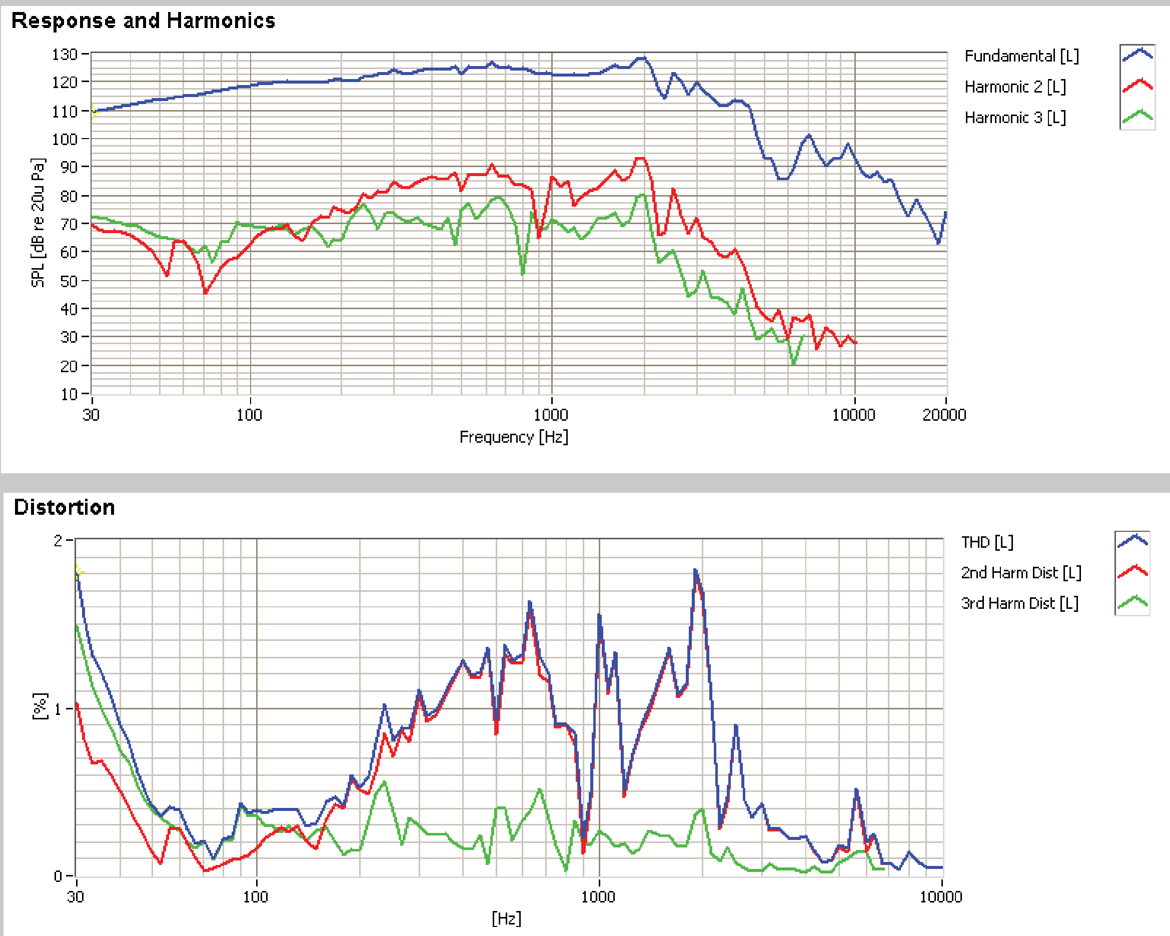


Figure 13: BMS 15N850V² SoundCheck distortion plots

extent (extra CNC processing on the bottom of the frame) the company went to ensure the cone assemble was positioned nearly perfectly at the magnetic center of the motor.

THE BMS 4592ND

The second BMS product I evaluated this month was a very unique compression driver, the BMS 4592ND neodymium coax compression driver. BMS patented this technology in 1997 and now has a four-model line of coax (mid/tweeter) compression drivers, the 1.4" 4593ND and 4597ND, the 1.5" 4595ND, and the subject of this month's report, the 2" 4592ND. The construction is very interesting and merits some explanation. **Figure 14**

Photo 2: The BMS 4592ND neodymium driver and CFD 2-60A Pro 2" horn

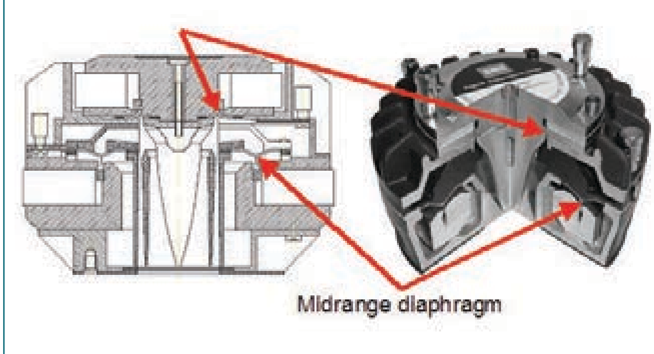


Figure 14: Cutaway drawing of the BMS 4592ND coaxial compression driver

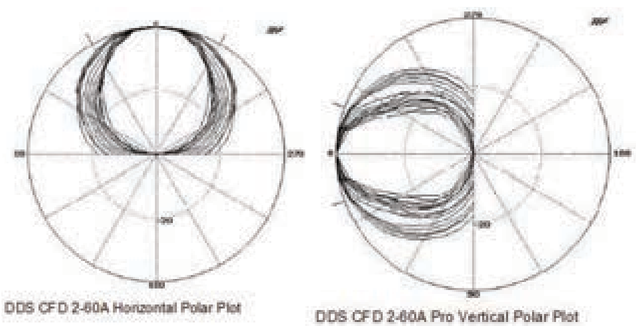


Figure 15: Horizontal and vertical polar plots for the DDS CFD 2-60a 2" horn

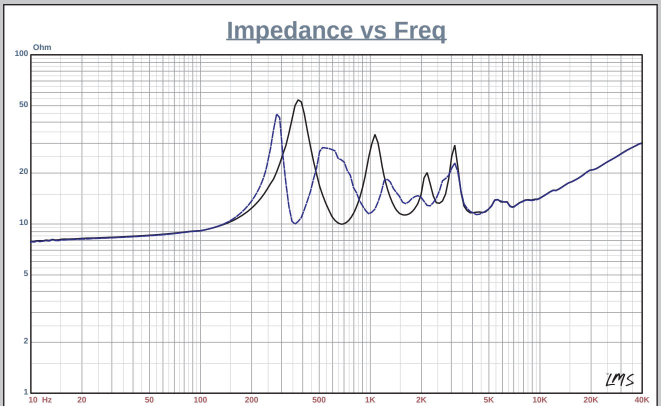


Figure 16: BMS 4592ND midrange free-air impedance plot

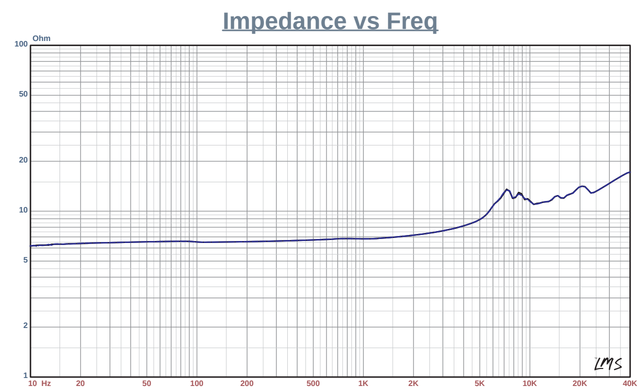


Figure 17: BMS 4592ND tweeter free-air impedance plot

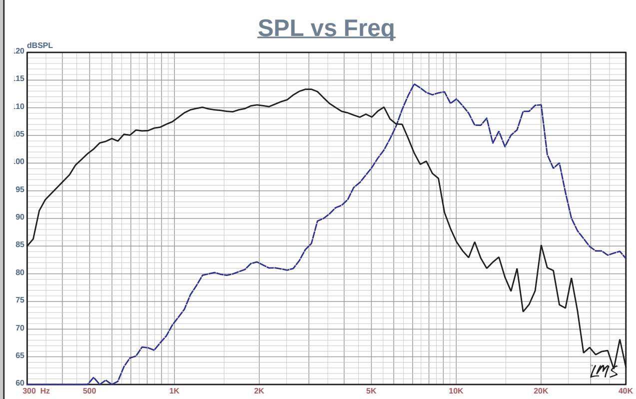


Figure 18: BMS 4592ND/CFD 2-60A on-axis response for both the midrange and the tweeter

shows a cutaway view of the BMS coaxial compression driver. As can be seen, the body of the driver contains two separate subsystems, each driven by a separate motor structure, magnet and voice coil. The midrange diaphragm feeds in from around the peripheral of the phase plug and the tweeter diaphragm is located at the base of the phase plug.

Specific features for the 4592ND include a 300 Hz to 22 kHz bandwidth, Kapton voice coil formers wound with copper-clad aluminum wire (two layers on the inside of the former and two layers on the outside), 1.75" voice coil diameter for the tweeter, and 3.5" voice coil diameter for the midrange, two neo ring magnets, polyester diaphragms, 1,000-W peak midrange power handling above 500 Hz and 320-W peak for the high-frequency driver, and color-coded chrome push terminals for each coaxial

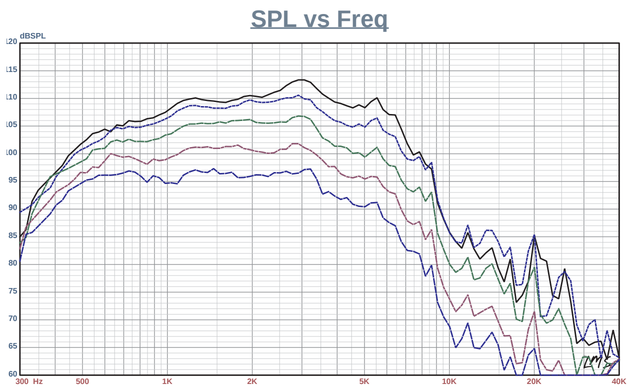


Figure 19: BMS 4592ND/CFD 2-60A midrange horizontal on- and off-axis frequency response (0° = solid; 15° = dot; 30° = dash; 45° = dash/dot; 60° = dash)

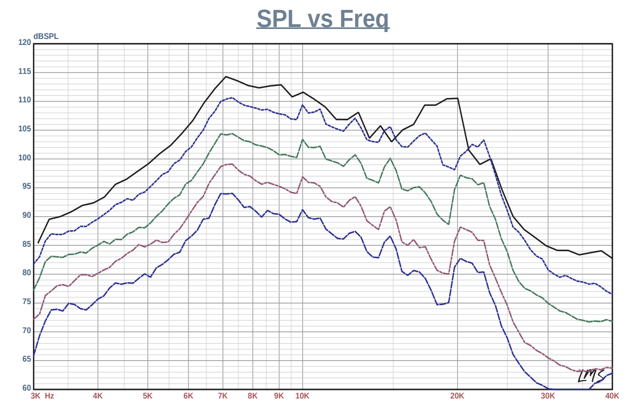


Figure 20: BMS 4592ND/CFD 2-60A tweeter horizontal on- and off-axis frequency response (0° = solid; 15° = dot; 30° = dash; 45° = dash/dot; 60° = dash)

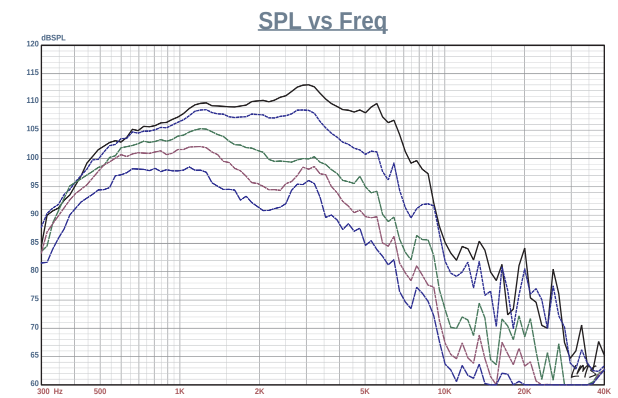


Figure 21: BMS 4592ND/CFD 2-60A midrange vertical on- and off-axis frequency response (0° = solid; 15° = dot; 30° = dash; 45° = dash/dot; 60° = dash)

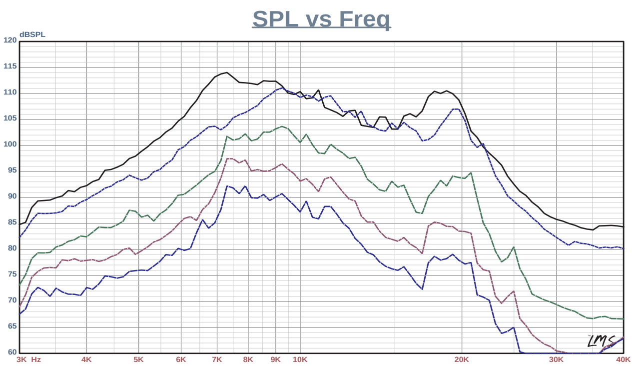


Figure 22: BMS 4592ND/CFD 2-60A tweeter vertical on- and off-axis frequency response (0° = solid; 15° = dot; 30° = dash; 45° = dash/dot; 60° = dash)

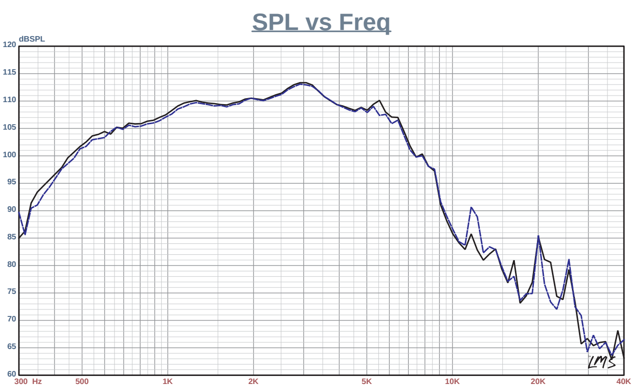


Figure 23: BMS 4592ND/CFD 2-60A midrange two-sample SPL comparison

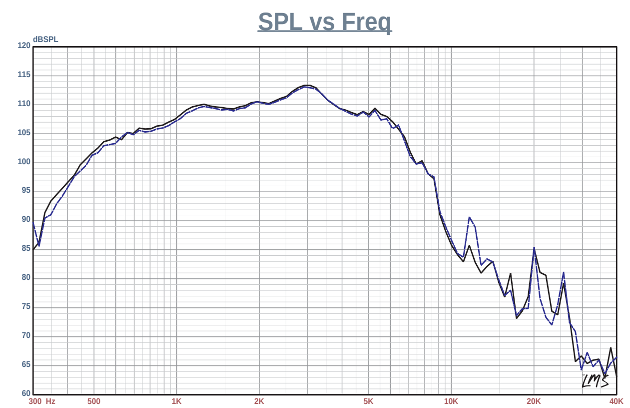


Figure 24: BMS 4592ND/CFD 2-60A tweeter two-sample SPL comparison

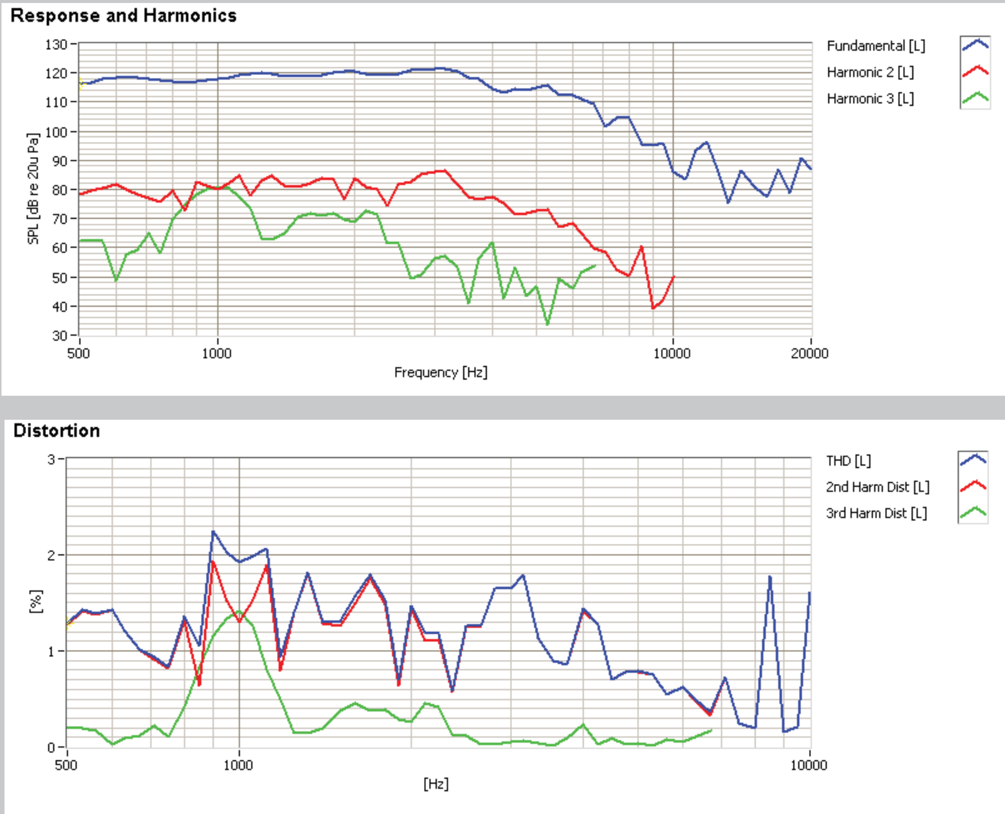


Figure 25: BMS 4592ND/CFD 2-60A midrange SoundCheck distortion plots

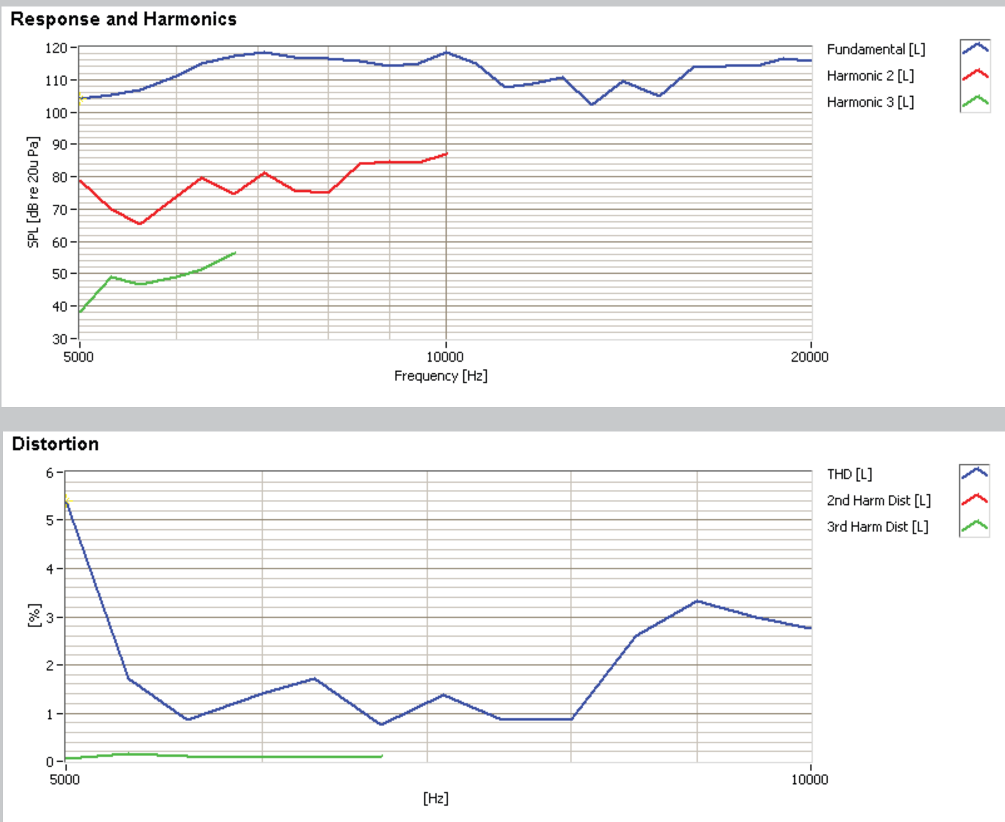


Figure 26: BMS 4592ND/CFD 2-60A tweeter SoundCheck distortion plots

section.

The horn that was provided this month for the BMS coaxial compression driver came from the folks at Design Direct Sound (DDS), who make a complete line of fiberglass horns. For the BMS 4592ND, DDS provided me with their CFD 2-60A Pro 2" horn. The CFD 2-60A Pro is a smaller format 60° × 40° horn designed for applications where large size 60° horns will not fit. (See **Figure 15** for the horizontal and vertical 1/3 octave polar plots for the DDS CFD 2-60A horn.) Specifications include a 1 kHz to 18 kHz frequency range, 113.9-dB sensitivity, nominal gain of +5.9 dB, and a directivity factor (Q) of 15.7 (+2.4–3.7).

I decided to present the test data for BMS midrange/tweeter simultaneously rather each separately. As usual I began with 200-point LMS impedance sweeps for both midrange and tweeter, with and without the horn attached. **Figure 16** gives the midrange impedance with the primary resonance without the horn of 375 Hz, and 281 Hz with the DDS CFD 2-60A horn attached. This also had a minimum impedance of 10 Ω and an 8.43-Ω DCR. For the tweeter, as you might expect (see **Figure 17**), the horn really didn't impact the impedance with the primary resonance at 7.2 kHz. Minimum impedance (above resonance) for the tweeter is 11 Ω with a 6.64-Ω DCR.

Next, I mounted the horn/compression driver combination in free-air without a baffle and produced on- and off-axis frequency response curves out to 60° off from 300 Hz to 40 kHz. **Figure 18** gives the tweeter and midrange on-axis response showing the overlapping range for each section of the 4592ND coax compression driver. Next, I proceeded to measure each section of the 4592ND from 0 to 60° in both the horizontal and vertical planes, 300 Hz to 40 kHz for the midrange, and 3 kHz to 40 kHz for the tweeter. **Figures 19** to **20** illustrate the midrange and tweeter horizontal response respectively, while **Figures 21** to **22** present the vertical response. The final SPL measurements were to compare the two coaxial sections of Sample A with Sample B, illustrated for the midrange and tweeter in **Figures 23** and **24**, respectively. Both sections had a close match within 0.5 dB within the drivers operating range.

For the final set of measurements, I again fired up the Listen, Inc. SoundCheck analyzer and SCM 0.25" microphone to produce distortion and time delay plots. Starting with the distortion plots, I set the SPL of both sections of the 4592ND/DDS CFD horn combination to 104 dB SPL at 1 m, which took 1.47 V for the tweeter, and 1.7 V for the midrange. I next located the microphone 10 cm from the mouth of the horn, and produced the two distortion plots seen in **Figure 25** for the midrange (measured from 500 Hz) and **Figure 26** for the tweeter (measured from 5 kHz).

Following this, I created impulse response curves for the midrange and tweeter at 2.83 V/1 m and loaded this data into Listen, Inc. SoundMap software to produce Cumulative Spectral Decay (Waterfall) plots and Short Time Fourier Transform (STFT) plots. **Figures 27** and **28** give the Waterfall plots for the horn/driver combination of the midrange and tweeter respectively. **Figures 29** and **30** display the STFT plots.

The performance and build quality of the BMS 4592ND is quite good, and the DDS CFD 2-60a horn produced a nice compact combination for this test. For more information, visit www.bmspro.com or www.ddshorns.com. *aX*

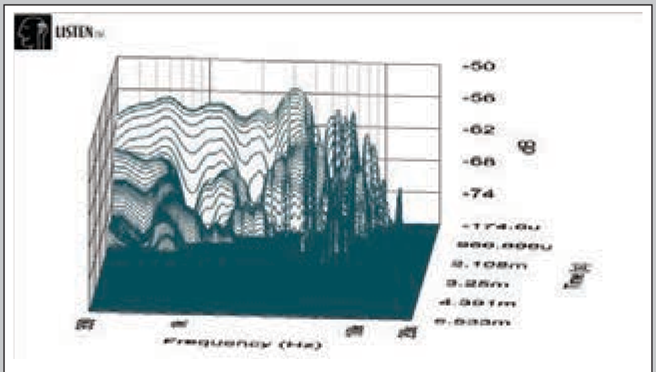


Figure 27: BMS 4592ND/CFD 2-60A midrange SoundCheck CSD waterfall plot

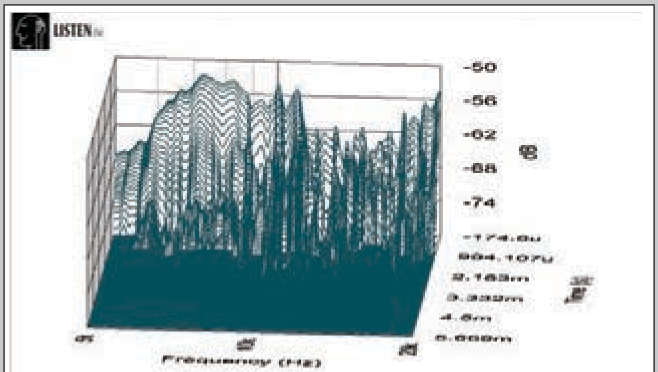


Figure 28: BMS 4592ND/CFD 2-60A tweeter SoundCheck waterfall plot

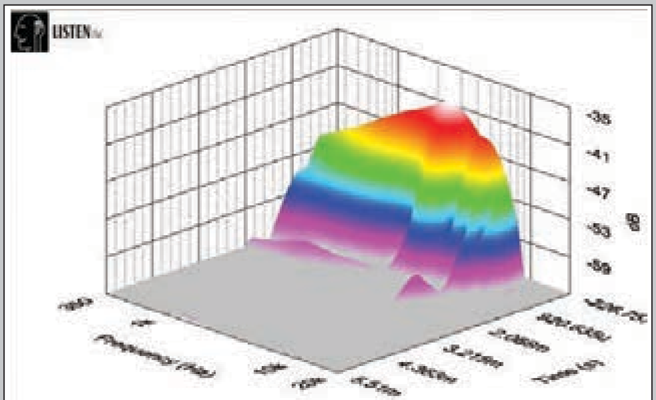


Figure 29: BMS 4592ND/CFD 2-60A midrange SoundCheck CSD STFT plot

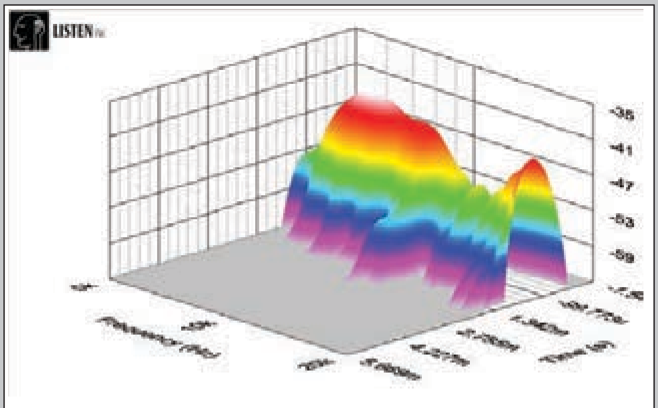


Figure 30: BMS 4592ND/CFD 2-60A tweeter SoundCheck CSD STFT plot

Eclectic Audio Design and DIY Tech

Shannon Becker discusses audio projects and DIY tech with the Tymkrs, Atdiy and Whisker

The duo behind Tymkrs.com distribute free weekly video, audio, written, and music content to thousands of makers, tinkerers, and lifelong learners in the DIY community. With their videos, podcasts, and blogs, they document and explain everything they learn as they build their own production company from the ground up. From analog circuit design, learning microcontrollers, studying how different types of microphones work, and building guitars, they share their knowledge, successes, and failures. On their show ZombieTech.tv, they interview clever people from all walks of life to find out what it would take to rebuild the world in the event of a Zombie Apocalypse!

SHANNON: Where do you produce your shows?

ATDIY AND WHISKER: We reside full time in Rochester, MN. A unique little city overflowing with healthcare professionals and engineers. Rochester is home to IBM's Blue Gene super computer project. It is also home to the Mayo Clinic, a world-renowned research and healthcare institution. As for the shows themselves, we've dedicated the top story of our home to audio and video production. You can make the commute in your pajamas, which is kind of awesome.

SHANNON: What are your current occupations?

ATDIY: I am a cardiovascular surgery nurse at the Mayo Clinic. I use my cheery demeanor to get patients back up on their feet and dancing again.

WHISKER: I am a full-time content producer. From websites to music, podcasts to educational videos, 3-D modeling to graphic design—I do it all.

SHANNON: How did you two meet? And when did you get started doing Tymkrs?


The Tymkrs studio

ATDIY AND WHISKER: We met one another while working as registered solutions providers in Second Life (a virtual reality platform). Together, we worked on many projects: building interactive campuses for universities, educational training systems, international job fairs, and virtual law offices just to name a few. When the global economy was about to take a nosedive, we read the signs, reinvesting everything into professional-grade audio and video production equipment. It took more than a year to fully transition, and in late 2010 "Toymaker Television" began releasing regular episodes.

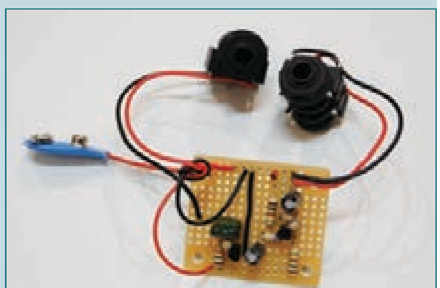
SHANNON: Atdiy, what first sparked your interest in electronics?

ATDIY: I was watching a show about some sort of apocalyptic occurrence, and the only


The Tymkrs cigar box guitar

The Tymkrs, Whisker and Atdiy

person who could find out any information from the surrounding communities was the guy who knew how to fix and operate the ham radio. I told Whisker that it'd be a smart idea to get into ham radio because they're the ones who know how to fix electrical things (especially when the zombie apocalypse comes) and get information. So he suggested I get the licensing manual to read. Not quite leisure reading, I know, but as I went through page by page, I realized that there were *many* concepts and ideas I had no knowledge of. So I researched what I didn't understand as I read, and that included things such as resistors, capacitors, etc. I find that as I learn more, I become more interested, and I enjoy it more when things make sense.


DIY LM2N3904 High-Impedance Preamplifier: "We had a number of high-impedance microphones/cigar box guitars around the studio that we needed amplified, and plenty of LM2N3904s. So we made a high-impedance preamplifier which also sounds great!"

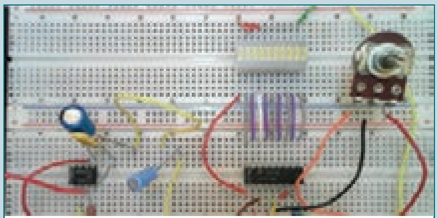
DIY MPF102 High-Impedance Preamplifier (also known as the piezo preamplifier): "This is another high-impedance preamplifier we designed with a JFET, and the sound is loud and clean—can't ask for much more than that!"

SHANNON: Whisker, when did you become interested in electronics?

WHISKER: I cannot remember being uninterested in electronics! Then again, I am interested in everything, so that is not surprising. One early memory that comes to mind is watching my father repair a tube-type black-and-white television set. He explained that the circuit was a tiny city full of millions of tiny people, and that I could watch what they were doing on the screen. In my search for those tiny electric people, I inadvertently became one of them.

SHANNON: How many audio-related projects have you worked on? What is your favorite?

ATDIY AND WHISKER: We do two or more projects a week, many of them audio related and usually filling some role in the recording studio. We really enjoy the DIY aspect of audio production, almost as much as sharing our inventions with the world. One favorite would be the time we built a microcontroller-based square wave synthesizer/sequencer. Instead of a speaker, it output flashes of light via a laser. By soldering up a 0.25" instrument cable through some decoupling capacitors to a small solar panel, we



DIY LM386 Audio Amplifier w/VU Meter: "The LM386 chip is often used in audio amplification projects—so this was one of Atdiy's first audio projects."

could receive the signal on any of our guitar amps or on the mixer. Because the voltage of a solar cell changes somewhat slowly, it acts as a low-pass filter. Filtered square waves are an awesome tool for making super fat bass lines.

SHANNON: *audioXpress* magazine readers are audiophiles in one form or another. Some are musicians, some are speaker builders, and some work on tube amp projects. Tell us about your audio interests.

ATDIY AND WHISKER: We are really fascinated with the entire process. In an inspired moment, one can capture a feeling and fire it down the barrel of history and straight into the ears of our descendants. We like being involved in the whole production chain directly. We build instruments, write music, design preamplifiers, modify amplifiers, build speaker cabinets and audio capture devices, write audio creation and editing software, arrange mix and master, then publish and distribute our own material. On the listening side: nothing beats an Ella Fitzgerald record through a Class A tube amp on a pair of Magnepan loudspeakers. Simple, and delightfully imperfect in the most perfect ways.

SHANNON: When do you find the time to do all of this?

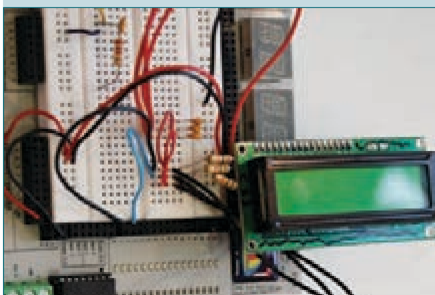
ATDIY AND WHISKER: It is a lifestyle. You have to make the time to do the things

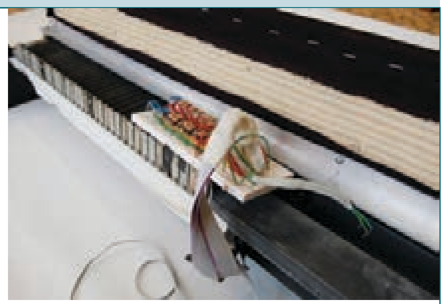
DIY Laser Audio Transmitter: "We wanted to see if it was feasible to transmit audio through use of a laser transmitter and a solar panel receiver. The laser transmission setup was connected to our development board and the solar panel was connected to our guitar amplifier. We managed to transmit pulse-width modulated signals from one floor of the house to another!"

you love. Modern life is full of distractions. If you want to be prolific there are a few things that make it easier: 1. Get rid of the TV. 2. Cell phone goes off at the door. 3. Include the people you are close with. 4. Share your results with other people!

SHANNON: Who came up with the idea of ZombieTech.tv? How many people have you interviewed for this series? And what do you think is the absolute must-have to survive a zombie apocalypse?

ATDIY AND WHISKER: Atdiy loves to yell at people in zombie movies who make poor choices. Because we are deeply involved in the online engineering and DIY community, we get to spend much of our day surrounded by brilliant people (stop by tymkrs.com/chat and join in the fun). ZombieTech.tv grew out of the realization that, with the right collection of people, not only could you survive a zombie outbreak, you could rebuild the world! Over the last six months, we have, so far, interviewed 25 guests: from teenagers building killer marshmallow cannons to famous authors, television personalities,





DIY SidCog Organ: "Whisker loves the SID Chip, the programmable sound chip of the Commodore 64, and all of the tunes it can make, and he also loves organs. So naturally he'd use the sound emulation properties of the chip with the form factor of an old Hammond organ! He's since added the ability to add release, sustain, echo, and decay before or during playing."

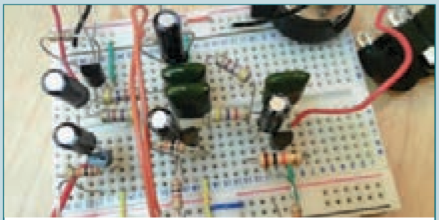
DIY Cabinet Speaker: "We have tons of instruments around and wanted a cheap and gritty way to listen to our jam sessions. So, we built our own cabinet speaker with cheap but functional pine and speaker. It sounds down-to-earth, the way we like it!"

engineers, and scientists. The absolute most important thing any one can bring into a disaster or survival situation is knowledge.

SHANNON: Where did you find your "zombie apocalypse survival team" and what exactly do they do?

ATDIY AND WHISKER: Our zombie apocalypse survival team comes from our favorite group of folks who can be found on tymkrs.com/chat and a few from the Parallax Propeller crowd. The original crew is:

- Whisker—Woodworker/musician, content producer, jack of all trades.
- Atdiy—Cardiovascular surgery nurse and behavioral neuroscientist background.
- Doggiedoc—Veterinarian by day, electronics and Parallax enthusiast by night.
- Aonomus[sic]—Chemist by day, and he wanted to be the weapons specialist for the team.
- JohnS_AZ—Farmer of citrus trees in the desert, but also "retired" manufacturing engineer and EOS/ESD guru. If we're going to make high-voltage defense systems, he makes sure we don't mess things up!
- RoyEltham—He's senior technical architect at Sony Online Entertainment but is also into Parallax. He is in charge of bringing his zombie cat, Buddy, which Doggiedoc will use to cure humanity with!



DIY Mic Pre/Mixer: "Whisker wanted to make a two-channel preamplifier and wanted to be able to mix them into a single output through three stages of amplification. The audio from this sounds amazing."

- Electromanj—He is also a Parallax enthusiast and normally is an electric systems guy who sets up irrigation systems for farms in the Midwest.
- MakerDino—He's an auto-mechanic by day and electronics hacker by night and has agreed to being the team's mechanic.
- Brainwagon—He's my ham radio Elmer and is one of the guys who worked on Renderman at Pixar. He is therefore the ham radio expert and has done *tons* of neat projects that would help fend off zombies.
- JohnR—He is another Parallax enthusiast and IT manager who helps with some of Parallax's event video networking/broad casting. He is endearingly labeled "zombie bait."

This was the original crew, and most have original songs dedicated to them. Since starting ZombieTech.tv, we have inadvertently invited nearly all of our guests. Most notably might be Dr. Cranquis (urgent care clinic doctor), John Cohn (mad scientist/ engineer who participated in *The Colony* on Discovery), Joe Grand (member of l0pht, and cast member of *Prototype This!* also on Discovery), Bradley Voytek (neuroscientist specializing in Zombie Neurophysiology), Mira Grant (Hugo Awards nominee who has considered life after zombies/amateur virologist), and Jeremy Blum (green technology specialist and 3-D printing engineer). Many of our other guests would also be welcome with their amazing skills.

SHANNON: You create original music. How would you categorize your music? What instruments do you play and which instruments did you make yourselves?

ATDIY AND WHISKER: Most of our music centers around either chip tune elements or early blues music. Anything becomes a musical instrument with us around, but a few instruments laying around the studio right now include: a Wurlitzer 206a electric piano, 1916 upright piano, two Hammond M3 organs, several acoustic guitars, a Danelectro U1 electric guitar, violins, a viola, a ukulele, a kalimba, and a resonator guitar! To get the sounds we are looking for, we often build our own instruments. A few of those would include: SidCogOrgan (a synthesizer based around the sound chip from an antique Commodore 64 computer and built into a recycled Hammond Chord Organ), Copper Dragon (a violin constructed from copper and carved maple), and a large collection of electric cigar box guitars.

DIY Foxhole Radio AM Receiver/Germanium Diode Crystal Radio: "Soldiers in WWII used to make these radios with items they found in their foxholes. They can receive AM transmissions with surprising clarity and the radios don't require battery power!"

SHANNON: Atdiy, you posted on FirstSpin.tv—a weekly podcast that you release on Tuesdays—a session about programming a Parallax Propeller microcontroller. Why did you choose that project? How difficult was it?

ATDIY: So FirstSpin.tv is our podcast where Whisker and our friend Roy Eltham teach me how to program a Parallax Propeller microcontroller (which uses the "Spin" programming language). And unlike most programming hobbyists, I've never had any prior programming experience, much less an interest. But, for as long as I've known Whisker, he has always liked the Parallax Propeller microcontroller and has used it in a number of his projects. As he works through his projects, he usually tries to bounce ideas off of me, though I will say that at the time I really had no idea what he was talking about. So once I had become more comfortable with electronics, I realized that I wanted to try to program some additional functionality but that

I didn't know how to. Since Whisker had talked so much about the Propeller MCU, it felt natural to be the first microcontroller I worked with. It helps that I find the architecture of the Propeller quite elegant—a healthy respect for the platform goes a long way. So for a complete newb like myself, it's been a lot like electronics. The more you learn, the more you grasp, and the more interesting it gets!

SHANNON: Whisker, can you tell us more about the SID Cog Organ project?

WHISKER: As a child, I was more interested in my computer than in piano lessons. My family is very musical, so naturally I used my computer to play the music for me. In the early 1980s, eons before computers were being used in recording studios, the Commodore 64 included a sound synthesis chip known as the SID. By manipulating numbers in the computer's memory, you could get this thing to make music. Unlike the NES or Atari of the same era, this chip was designed by a musician, and was capable of some pretty amazing things. As an adult, I decided to learn piano and organ. While I love pianos and that B3 sound, I was missing the sound of my own childhood. This inspired me to start building my own organ. It plays like a Hammond, and sounds like a classic video game! I have learned so much about music and electronics while working on it. It has opened many doors and I would encourage anyone else to start their own impossible project.

SHANNON: Atdiy, can you tell us about your LED Heart project?

ATDIY: Sure! I'm a cardiovascular surgery nurse by day, and it was Halloween, so I decided to make a little project with the Propeller demo board. I had just learned how to turn LEDs on and off, and thought, well, maybe I can make them look/sound like a heartbeat's lub-dub. That worked out fine, and I then decided I wanted to make it have a creepy factor where if you got closer to the heart (in the form of a photoresistor), the heart would start beating faster. If you got farther away from the heart, the heartbeat would slow down. After adding some sound, I showed it to our IRC friends and they were slightly nonplussed. So I decided the project needed more LEDs (because what project doesn't need more), and added the cardiac conduction pathways. That project made it onto Hackaday.com and a few of the medical professionals were inspired but a few were annoyed from my lack of exact conduction timing. The project has now evolved to 14 different rhythms with completely accurate timing of both conduction and valve sounds, and I am currently working on its final form factor—a wearable heart pendant with rhythms that can be changed just for fun!

SHANNON: Do you have any advice for *audioXpress* readers who are considering starting DIY electronics audio projects of their own?

ATDIY AND WHISKER: There is a huge online community of people who spend their time helping each other design and build electronics projects. Come find us! Electronics is much more fun and less frustrating when you have mentors around to help you avoid the pitfalls or to help make your great ideas even greater. For example, we couldn't do half of what we do with out all of the wonderful support from our viewers.

SHANNON: Any projects currently in the works?

ATDIY AND WHISKER: We are always working on something new, check Tymkrs.com for links to all of our latest crazy inventions. One of the biggest projects going on right now is the design and planning of our brand new video production studio. It includes a DIY robotic camera jib built into the ceiling. So much awesome!

SHANNON: Atdiy, do you have a dream project? Something you would build if you had unlimited time and monetary resources?

ATDIY: As silly as it sounds, a zombie apocalypse bunker would be an epic project to get right. When we ask ZombieTech.tv guests what tools they'd bring, we keep getting ideas for new rooms to add into the bunker. So far, the additions have been two MRI rooms, one clean room, zombie power-generating treadmills/wheels, tesla coil defense system, and even a piezoelectric power-generating floor. But if you want realistic, I'll settle for a self-cleaning house.

SHANNON: Whisker, what about your dream project?

WHISKER: Actually, Tymkrs.com is very much my dream project! I give it every ounce of time I have and invested every cent I had into getting it off the ground. It is very important to me to do something which helps, inspires, or at least entertains other people. Fred Rogers is my hero. As such, a fun educational show about music and making stuff is exactly what I want to build. The unlimited monetary resources would help move things along though! Actually, viewer donations are a big part of our budget, so if anyone out there would like to help out with equipment donations or operating expenses, check Tymkrs.com for details. Impossible dreams are made possible with help! There is a whole generation out there growing up without a Fred Rogers, let's see if we can do something about that! *aX*

By Wim de Jager (The Netherlands)

Tube Sound

A triode amplifier design offers optimal quality at a reasonable cost

About 100 years ago, Lee de Forest invented the triode. It is remarkable that triode output stages still occupy an important place in high-end audio amplifiers. Triode amplifiers are often very expensive. The design presented here is based on a combination of optimal quality and reasonable construction costs (see **Photo 1**).

The subject of "Tubes vs. Transistors" has already been frequently discussed in literature. Instead of repeating it all, we are offering a reference instead. A comprehensive comparison is presented in the article, "Vacuum Tubes and Transistors Compared" (refer to www.milbert.com/TVTFC.bdc). In addition to triodes, pentodes are also used for audio amplifiers. The choice between these two is made on the basis of a number of considerations.

Photo 1: A Class-A triode push-pull amplifier

TRIODE OR PENTODE?

With triodes, the influence of the anode voltage on the anode current (negative feedback) is quite large (see **Figure 1a**). The result of this is a lower output impedance and a smaller amplification factor "µ." (This is the maximum achievable voltage amplification when there is a DC current source in the anode path.)

With pentodes (see **Figure 1b**), the screen grid voltage in particular determines the anode current. As a result, the influence of the anode voltage on the anode current is smaller, which results in a higher output impedance and therefore a higher µ.

There are a few disadvantages associated with pentode output stages. Loudspeaker damping is practically zero and negative feedback is required to reduce the output impedance of the amplifier to an acceptable value. The damping of transformer resonances is so small that in the absence of a load (open-circuit output) very strong oscillations can be generated with voltages that can become so high arc-over occurs in the valve sockets or output transformer. With triodes, the loudspeaker damping and transformer resonance damping are much more favorable because of a lower µ.

There are also disadvantages associated with triode output stages. The consequence of a lower µ means that the driving stage has to be able to supply a signal with relatively large amplitude. Because of the lower output impedance the influence of power supply ripple is larger with triodes than it is with pentodes. This means that the power supply smoothing filter has to meet higher requirements.

SINGLE-ENDED OR PUSH-PULL?

The distortions of a triode and a pentode are compared in **Figure 2**. This shows that with triodes d2 is dominant and d3 with pentodes. Even-numbered harmonics are largely cancelled in push-pull circuits, odd-numbered harmonics are not. The result of this is that a triode push-pull amplifier has a lower open-loop distortion than a pentode push-pull amplifier, mostly due to the third harmonic (d3).

The second advantage of a push-pull amplifier is that with correct DC biasing there is no DC bias in the core of the output transformer. The core can therefore be smaller, and as a result the transformer is cheaper to make. Also, the effect of the power supply ripple is mostly cancelled in the output stage.

A disadvantage of a push-pull amplifier is the requirement of a phase shifter to drive the output stage with two signals that have equal amplitude and opposite phase.

OUTPUT POWER AND EFFICIENCY

It is incorrect to think that sound appears twice as loud to the ear when the power rating of the amplifier is two times larger. The sensitivity of our ears is logarithmic and has a range of 0 dB (threshold of audibility) to 120 dB (pain threshold). Doubling the output power of an amplifier gives a power gain of 3 dB.

The amplifier published in this article delivers an output power of 2 × 9 W at a total harmonic distortion (THD) of 2% (open-loop). This is a reasonable amount of power considering the logarithmic characteristics of our ears, and also because of the soft-clipping behavior (excessive signal peaks do not lead to "aggressive" distortion). If desired, high-efficiency loudspeakers could also be used.

The valve handbook from Philipsstates that a pentode push-pull output stage with 2 × EL34 has an efficiency of 53% [1]. This does not, however, take into account the dissipation of the screen grid. This should be included for a fair comparison between triode and pentode output stages. Including the screen grid dissipation, the efficiency is 42%. The efficiency of the triode push-pull amplifier presented here amount to 35%, so that is still competitive.

DISTORTION

The audibility of distortion is strongly dependent on the

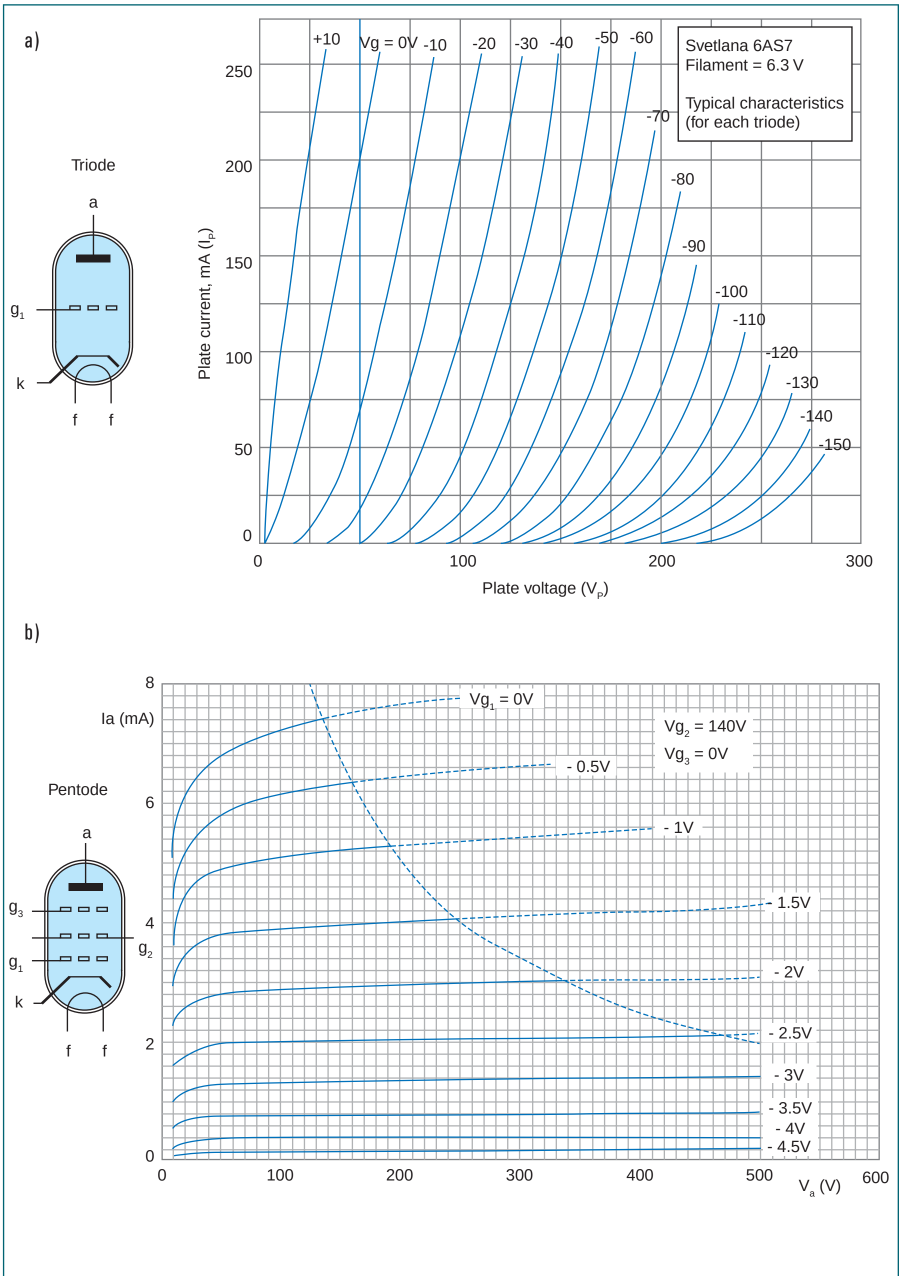


Figure 1a: The characteristics of a triode 1b: The characteristics of a pentode

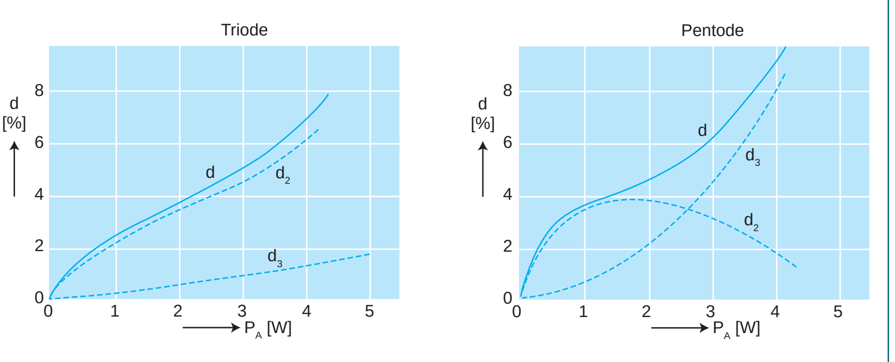


Figure 2: Distortion with a triode and a pentode

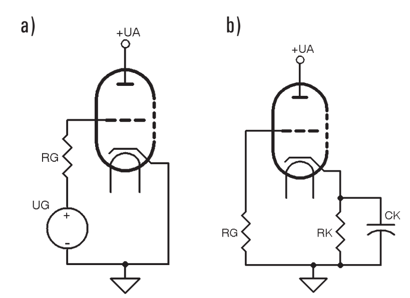


Figure 3a: The fixed value of grid voltage. 3b: With the aid of a cathode resistor, a simple automatic negative grid voltage adjustment can be made.

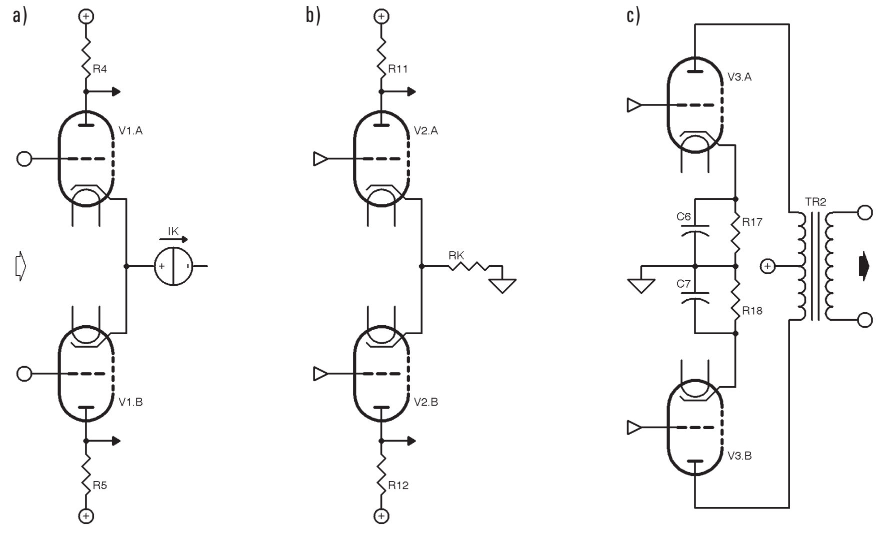


Figure 4: The principle schematic for the triode push-pull amplifier

spectrum. Crossover distortion in transistor amplifiers is very annoying because it has a very broad spectrum. In addition, this distortion increases at lower output powers. With valve amplifiers, the concerns are with lower-order harmonics and according to Douglas Self the threshold of audibility in this case is around 1%[2]. There is therefore very little motivation to design valve amplifiers with lower distortion.

DC BIAS SETTING

A fixed negative grid voltage (see **Figure 3a**) has the disadvantage that the DC bias setting is dependent on device tolerances and ageing of the valve. It is often necessary to make manual corrections for the DC bias setting to obtain the desired result. With an automatic negative grid voltage (see **Figure 3b**) the grid cathode voltage is determined by the voltage drop across cathode resistor R_k, which is caused by the cathode current.

In this method of negative feedback, R_k determines the DC bias setting in a significant way and the effects of valve parameters are strongly reduced. If AC negative feedback is not desired, then R_k typically needs to be decoupled with an electrolytic capacitor.

SIMPLIFIED SCHEMATIC

In **Figure 4**, the simplified schematic for the amplifier, the RC coupling components between the different stages have been omitted for simplicity's sake. The low μ of the triodes used makes a three-stage approach necessary. For the DC bias setting of the output stage, an automatic negative grid adjustment was chosen, separate for each triode. As a result, the DC offset is much smaller compared to a common cathode resistor and this is advantageous with respect to the potential DC bias in the output transformer. Decoupling of the cathode resistors is necessary to prevent the preamplifier from having to deliver an even larger drive voltage.

The output stage of a push-pull amplifier is often driven by a phase shifter. This is not recommended with triode output stages because the wave shapes of the output signals of the phase shifter are not equal at very large signal swings, which would result in more distortion in the output stage than is necessary. The second stage in this amplifier employs a push-pull amplifier with a common cathode resistor. This resistor is not decoupled because the AC cathode currents are in opposite phase and therefore do not cause a voltage drop across R_k. The first stage is a differential amplifier, used as a phase shifter. From all the circuits that are suitable for this purpose, this is the only type where both input terminals remain freely available. By doing so, the design becomes similar to that of an op-amp, which means that the usual methods for negative feedback around opamps can be used here as well. The implementation of the current source I_k can be a resistor, the ohmic value of which must be much greater than the 1/S of the valves (where S is the mutual conductance). This requires a negative power supply voltage of greater than or equal to 50 VDC.

An important advantage of this design is the complete push-pull structure so that the distortion and sensitivity to hum and interference are minimal.

THE COMPLETE AMPLIFIER

Figure 5 shows the complete schematic or the triode push-pull amplifier. The power supply is shown separately in **Figure 6**.

The cost of the output valves and output transformers often make the most significant contribution towards the final price tag of a valve power amplifier. For this design, we chose the 6AS7 valve in combination with the 3A524 output transformer. The 6AS7 is a high-quality valve for a relatively low amount of money. (For details and pricing refer to www.amplimo.nl.) The 6AS7 is a dual-power triode (maximum dissipation 2 × 13 W) that was originally designed to be the series-pass element in regulated high-voltage power supplies. This can be seen from the maximum allowable value for the voltage between the cathode and the filament (±300 V). This indicates high quality isolation. The remaining properties of the 6AS7 are comparable to those of other power triodes.

The 3A524 output transformer (maximum 40 W) is generously sized for a 9-W amplifier. The primary impedance is 3545 Ω. For optimal output drive, the DC settings for the output valves are: U_a = 270 V, I_a = 67 mA, U_k = 80 V, and R_k = 1k2.

Triode push-pull output stages have, by their very nature, a Class-A characteristic and operates as discussed below.

Suppose that, because of the drive signal, the anode current of V3a increases significantly, U_a of V3a will then reduce significantly. The influence of V3a on the output transformer is now dominant and this causes an increase in U_a of V3b. Since the U_a of triodes has a large effect on anode current, V3b does not stop conducting despite the increase of U_g (more negative), which results in a Class-A operation of the output stage.

The "quiescent current" of about 10 mA has very little effect on the efficiency. Because of this Class-A behavior "crossover distortion" is precluded and the current consumption from the power supply is almost independent of the output drive.

The preamplifier (V2) has to be able to deliver two-drive voltages of up to 70 VAC to the output stage and that places high demands on the dynamic range. An ECC82 (12AU7) is very appropriate for this. To obtain the highest possible drive voltage, R7, R8, and R9 are connected to −50 V instead of ground. This increases the power supply voltage to V2 by 50 V to 320 V.

A "hybrid DC bias setting" was selected here. R9 is a common cathode resistor (automatic negative grid adjustment). The voltage across R9 is partially determined by the current through R10.

The consequence of this is that the DC bias setting is a hybrid of automatic and fixed-negative, which results in better drive capabilities. Because V2 is a push-pull circuit of triodes as well, the distortion is also small because of the cancellation of the even order harmonics.

The DC bias setting combined with the values of R11 and R12 ensures that sufficient capacitive current can be delivered to the output stage at higher frequencies. The input capacitance of V3a and V3b amounts to about 40 pF.

For the phase shifter, an ECC83 (12AX7) is the most suitable, in particular because of its relatively high μ, the gain of the first stage is as high as possible. The influence of R6 on the offset (the nonideal current source functionality is of concern here) is about 4%. Further improvement by increasing the negative power supply voltage and a larger value for R6 is not worthwhile.

The input signal for the amplifier is connected to V1b and the negative feedback signal, via a switch and R3||C1 and R2, to V1a. Swapping the input terminals results in positive feedback, which causes the amplifier to oscillate badly. It is, therefore, very important to carefully check the numbering on the valves and the color codes of the output transformer.

Because of the good open-loop characteristics, a moderate negative feedback factor of 3 (9.5 dB) was selected. The negative feedback can be switched on and off to be able to determine the difference in listening tests. When the negative feedback is switched on, the input signal needs to be larger by a factor of 3 (600 mV instead of 200 mV for an output power of 9 W).

It is important to consider that with valve amplifiers, large negative feedback can cause RF instability in addition to AF instability. Because of the AF phase shift resulting from the RC coupling between stages and the output transformer, a very low-frequency oscillation can occur. In the past, this has sometimes been called "motor-boating." Resistors R15 and R16, the so-called "grid stoppers" reduce the risk of RF instability in the output stage.

THE POWER SUPPLY

The power supply section can be seen in **Figure 6**. The power supply transformer, with the part number 5N1609, has been specifically designed by Amplimo Netherlands for this project, and is available from them (www.amplimo.nl). The electrostatic shield has to be connected to the chassis and its task is to divert mains borne disturbances to earth that could otherwise be capacitively coupled through the transformer to the power supply.

Triodes make high demands on the power supply ripple filter. The design of these filters is based on a so-called π-network, such as C10||C11, R20 and C9. The series resistor has a comparable effect to that of the forward resistance of a valve rectifier. This forms a low-

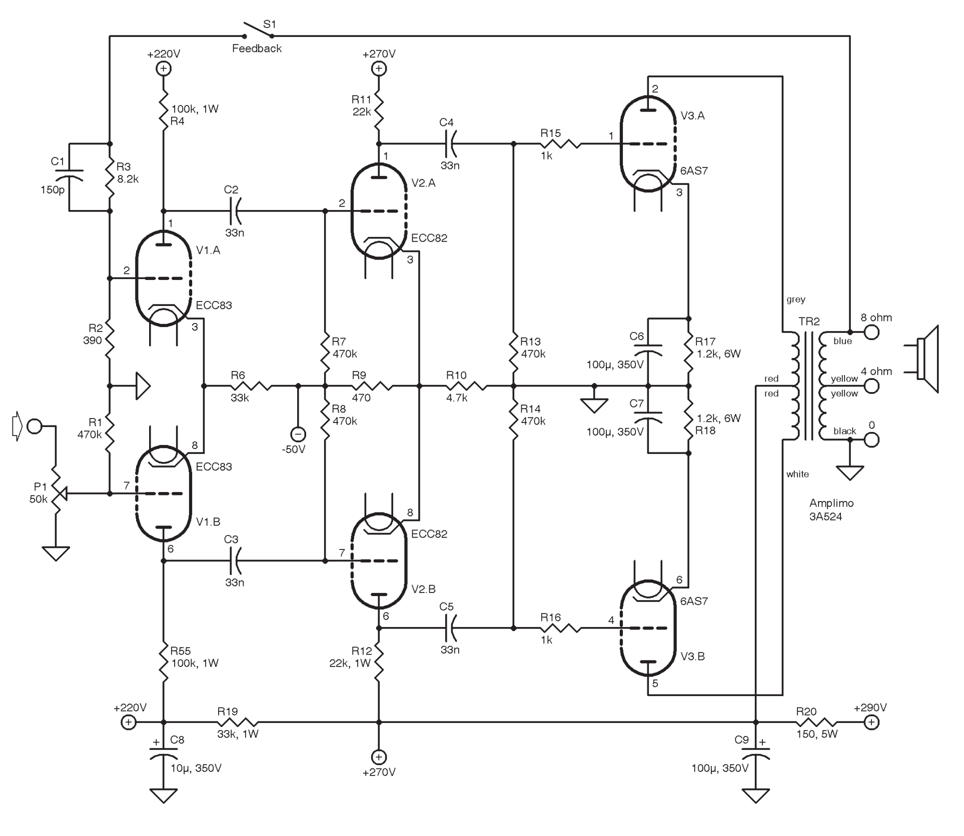


Figure 5: The complete schematic of the amplifier (one channel shown). Note the switchable negative feedback.

pass filter, which effectively suppresses the ripple voltage (the higher harmonics in particular), mains disturbances that are inductively coupled through the transformer and switching pulses from the rectifier diodes. Since the amplifier (including the output stages) operates in Class A, the current consumption from the power supply is practically constant and the use of series resistors is not a problem.

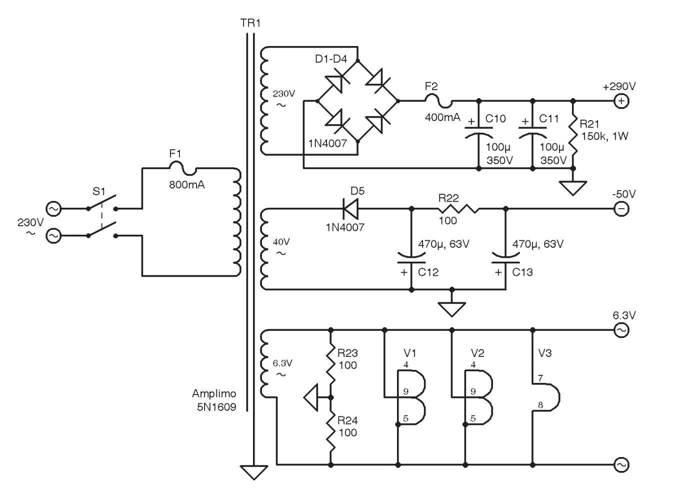


Figure 6: The power supply uses a special mains transformer that is available from Amplimo

In order to limit the hum from the filament supply, the "electrical center" of the filaments is connected to ground via R23 and R24.

Under certain circumstances, when, for example, the amplifier is switched off shortly after it has been switched on, it is possible that the high-voltage electrolytic capacitors in the power supply filter can take a very long time to discharge completely. This is undesirable, particularly from a safety perspective. Connecting a "bleeder" resistor (R21) in parallel with C10 and C11 discharges the electrolytic capacitors to less than 50 VDC in about two minutes.

REALIZATION

The starting point for the construction of the enclosure (see **Photos 2** and **3**) is an aluminium corner profile (e.g., type 04.90.2208, available from Tessatronics, Display, and others). The profile is normally supplied in a length of 1 m and can be used for four corner profiles of 25 cm each. The front and rear panels are made from 3-mm thick hard aluminium sheet and are attached to the ends of the corner profiles with M5 bolts. The side panels (1.5-mm thick aluminium sheet) fit in the slots of the corner profiles. These side panels support the chassis plate (2-mm thick aluminium plate with four-cornered edges) with M5 nuts and bolts. Between the chassis and the front and rear panels is a gap of 15 mm to ensure sufficient cooling. The top and bottom plates (perforated steel plate, 1-mm thick, with cornered front and rear edges) is attached with M3 bolts and square nuts that fit in the corner profile. The connecting and the mounting of the components is done using the "hard-wired" method. This takes a little bit more time compared to assembling a printed circuit board, but that should not be an objection considering the simplicity of the circuit. An added benefit is that the results are better, among other things because twisted wiring for the filaments has a lower risk of hum and is better suited to handle large currents. The assembly (see **Photo 3**) uses discs cut from circuit board for the star-point ground, strips of circuit board for ground and cable supports. The discs, strips, and cable supports are attached to the chassis with self-adhesive standoffs. There are two options regarding the coupling capacitors C2 to C5: cheaper polyester foil (e.g., from ERO) or high-quality paper-in-oil (e.g., from Jensen).

It is recommended to use 1 W/1% metal film resistors for anode resistors R4, R5, R11, and R12. That is because the larger dimensions of these resistors are better suited for dealing with high voltages. Since this amplifier is implemented as a "complete amplifier," the front panel has a power switch, input selector switch, and a volume control. On the back of the enclosure, you can add, in addition to the usual inputs and outputs, the switch for turning the negative feedback on and off, so that it becomes very easy to experiment with this.

PARTS & COMPONENTS

Capacitors
- C1 = 150 pF ceramic
- C2–C5 = 33 nF 400 V
- C6, C7 = 47 µF, 350 V electrolytic
- C8 = 10 µF, 350 V electrolytic
- C9–C11 = 100 µF, 350 V electrolytic
- C12, C13 = 470 µF, 63 V electrolytic

Miscellaneous
- Tr1 = mains transformer type 5N1609, primary 240 V, secondary 230 V/40 V/6.3 V (www.amplimo.nl)
- Tr2 = output transformer type 3A524 (www.amplimo.nl)
- F1 = fuse, 600 mA, slow
- F2 = fuse, 300 mA, slow
- Aluminum corner profile (e.g. #04.90.2208) (www.tessatronic.nl; www.display.nl)
- Self-adhesive standoff (e.g., Conrad Electronics #532027-89)
- Self-Adheseive mounting for cable strap (e.g., Conrad Electronics #543234-89)

Resistors (all metal film 1%, 0.25 W or 0.6 W unless otherwise specified)
- R1 = 470 kΩ
- R2 = 390 Ω
- R3 = 8 kΩ2
- R4, 5 = 100 kΩ, 1 W
- R6 = 33 kΩ
- R7, R8 = 470 kΩ
- R9 = 470 kΩ
- R10 = 4 kΩ7
- R11, R12 = 22 kΩ, 1W
- R13, R14 = 470 kΩ
- R15, R16 = 1 kΩ
- R17, R18 = 1 kΩ2, 6W (wire wound)
- R19 = 33 kΩ, 1 W
- R20 = 150Ω, 5 W (wire wound)
- R21 = 150 kΩ, 1 W
- R22–R24 = 100 kΩ
- P1 = 50 kΩ logarithmic law pentiometer

Semiconductors
- D1–D4 = 1N4007 or bridge rectifier 400 V piv @ 1 A
- D5 = 1N4007

Valves
- V1 = 12AX7 or ECC83
- V2 = 12AU7 or ECC82
- V3 = 6AS7

TEST RESULTS

With valve amplifiers it is customary to give the maximum output power at THD = 5%. Because this amplifier can also be used open-loop, we assumed a value of 2%. This value also corresponds to the limit before clipping occurs.

Photo 4 shows the soft-clipping behavior at an output power of 11 W (5 V/div). **Photo 5** shows the open-loop square-wave response at 2 kHz and 5 V/div. It is of interest to note that this amplifier—because of the characteristics of the triode output stage—is open-loop stable. aX

Editor's note: This article originally appeared in the June 2007 edition of Elektor.

RESOURCES

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[2] D. Self, *Audio Power Design Handbook*, Newnes, Maryland Heights, MO, 2000.

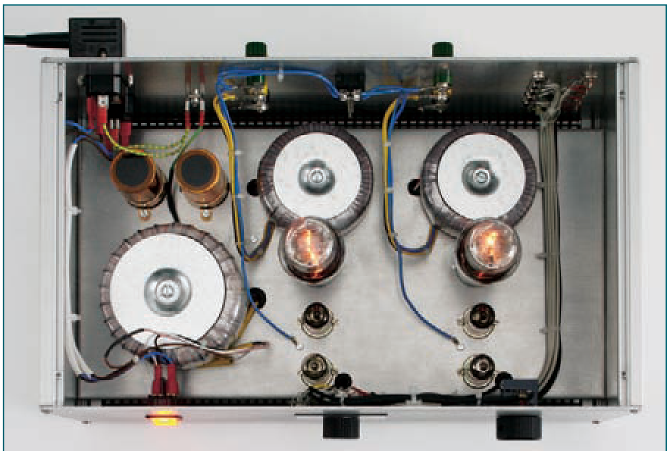


Photo 2: The amplifier viewed from above

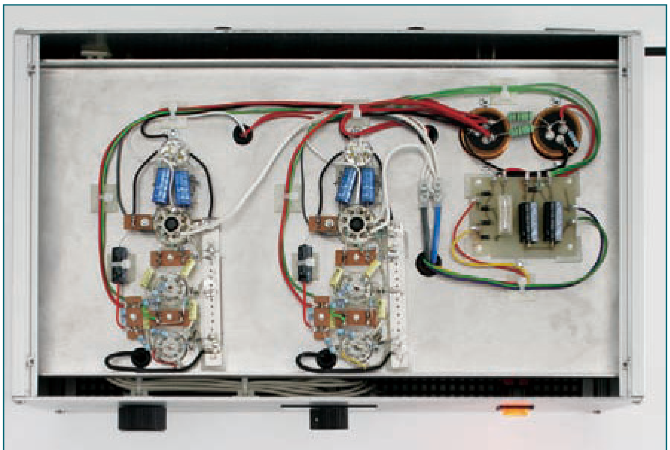


Photo 3: The components mounted on the bottom side

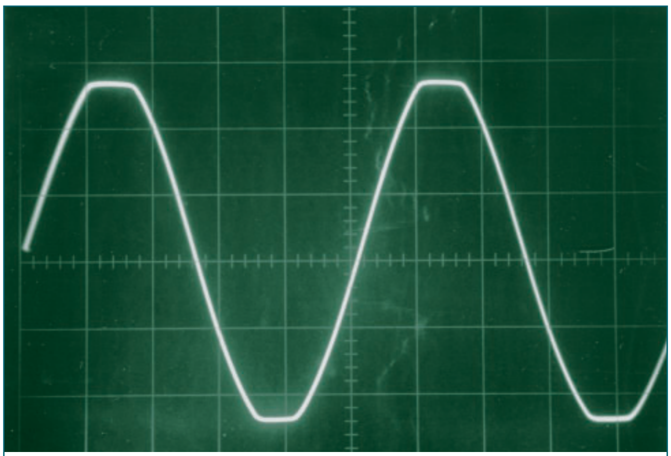


Photo 4: The clipping behavior at 11 W/1 kHz (5 V/div)

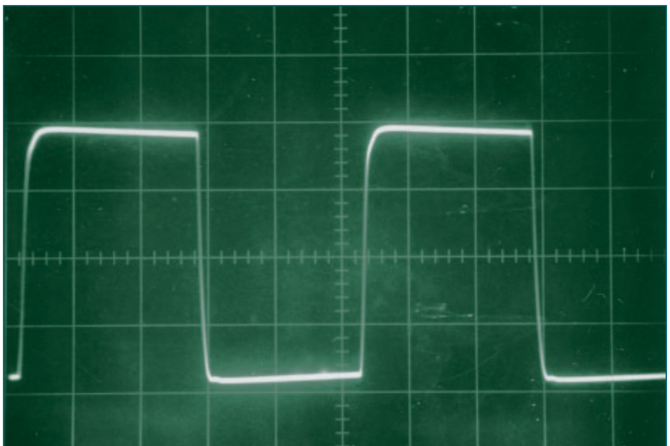


Photo 5: The square-wave response without feedback (2 kHz, 5 V/div)

SPECIFICATIONS

Test measurements carried out with a load of 8 Ω

Output power (1 kHz, 2%):	9.5 W into 4 Ω and 8 Ω	
Input sensitivity (9 W, 1 kHz):	200 mV (without negative feedback)	600 mV (with negative feedback)
Output impedance (1 kHz):	2.3 Ω (without negative feedback)	0.8 Ω (with negative feedback)
Bandwidth (1 W):	20 Hz to 27 kHz (without negative feedback)	less than 20 Hz to 64 kHz (with negative feedback)
THD + noise (1 W, 1 kHz):	0.15% (with negative feedback)	0.3% (without negative feedback)
Hum/noise:	–72 dB (without negative feedback)	–82 dB (with negative feedback)

By Richard Honeycutt (United States)

Hollow-State Amplifiers

A look at small-signal amplifiers

Last month, we took a look at the development of vacuum tubes. In this column, we'll see how triodes and pentodes are used in small-signal amplifiers. We'll discuss power amplifiers in a later column. Let's begin by looking at amplifiers in general, before examining any particular type. **Figure 1** shows an equivalent circuit of an amplifier.

The triangle symbolizes an amplifier. An AC voltage is applied at the input, and an amplified (enlarged) version of that AC voltage appears at the output. The factor by which the amplifier increases the input voltage is called the voltage gain. A certain amount of current flows in the input part of the amplifier when the input voltage is applied. The amount of current is controlled by the input resistance. In general, there can also be capacitive and/or inductive components controlling the input current, so we often use the more general term, input impedance.

The output part of the amplifier circuit is symbolized as an AC voltage generator. This generator is considered a pure voltage source. Its output voltage is not affected by whatever is connected to the output terminals. No real amplifier behaves exactly this way, so our equivalent circuit includes an output resistance. When current is drawn from the amplifier output, it causes a voltage across the output resistance. This voltage is subtracted from the output voltage; thus, the more current a real amplifier puts out, the lower its output voltage. For example, if we apply a 1-V input to an amplifier with a voltage gain of 10, the voltage produced by the amplifier when no output current flows is 10 V. If the output current is 0.01 A, and the output resistance of the amplifier is 100 Ω, a voltage of 0.01 × 100 = 1 V appears across the output resistor, decreasing the output voltage to 9 V.

It is not necessary that one side of the input and one side of the output be grounded, although this is often done. When neither side of the input or output is grounded, the circuit is referred to as balanced. Balanced circuits are used in pro audio and broadcasting. Balanced circuits using tubes are generally built by using balancing transformers at the input and/or output.

The equivalent circuit in **Figure 1** applies to any amplifier: triode, pentode, transistor, op-amp, or whatever. Mainly, it helps us to picture the meaning of R_{IN}, R_{OUT}, and A_V.

AMPLIFIER CONFIGURATIONS

The input voltage to a vacuum tube amplifier must be applied between the control grid (often just called the grid) and the cathode. If the circuit is unbalanced, as most are, either the cathode, grid, or plate can be grounded. Thus, there are three basic tube amplifier circuits: grounded cathode, grounded grid, and grounded plate. These are shown in **Figure 2**.

Two things should be noted about this drawing. The first is that the only DC voltage shown is the B+ supply. As we will see later, a real amplifier must include provisions for proper grid biasing. The second thing is that, at first glance, the plate does not seem to be grounded in the grounded-plate amplifier stage. Actually, however, the B+ supply is provided by a circuit that keeps AC voltage from appearing on the B+, so the B+ supply is really at AC ground, though certainly not at DC ground. By the way, other names are sometimes used for these stages: common-cathode, common-grid, and common-plate. The latter is also often

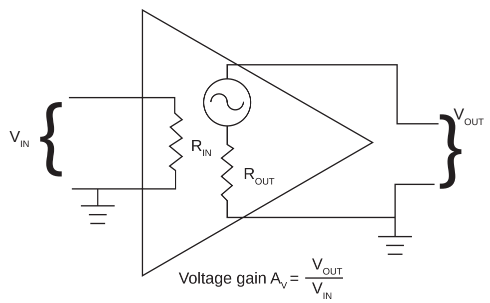


Figure 1: Generalized amplifier

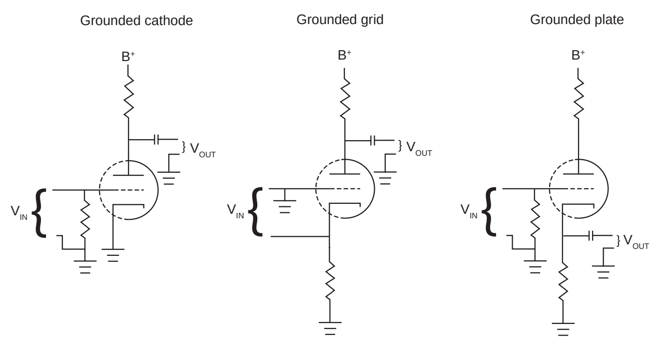


Figure 2: Basic triode amplifier stages

called a cathode follower stage.

Typically, both the grounded-grid and the grounded-cathode stages provide a lot of voltage gain, up to about 100 or so for a triode stage, and more for a pentode stage. The input resistance of the grounded-cathode stage is quite high. It would be almost infinite except for the resistor that must be connected from grid to ground in order to provide the correct DC voltage on the grid. The input resistance of the grounded-grid stage is fairly low—a few kilohms or so. This fact, plus the excellent high-frequency performance of the grounded-grid stage, makes it useful at radio frequencies. The output impedances of the grounded-cathode and grounded-grid stages are somewhat lower than the value of the plate resistor—tens to hundreds of kilohm. The cathode-follower has a voltage gain slightly less than 1. However, it does provide a high-current gain. This makes it useful as an impedance-matching stage for driving load resistances that are uncomfortably low for being driven from the plate of a tube, since low impedance loads require higher current for a given voltage.

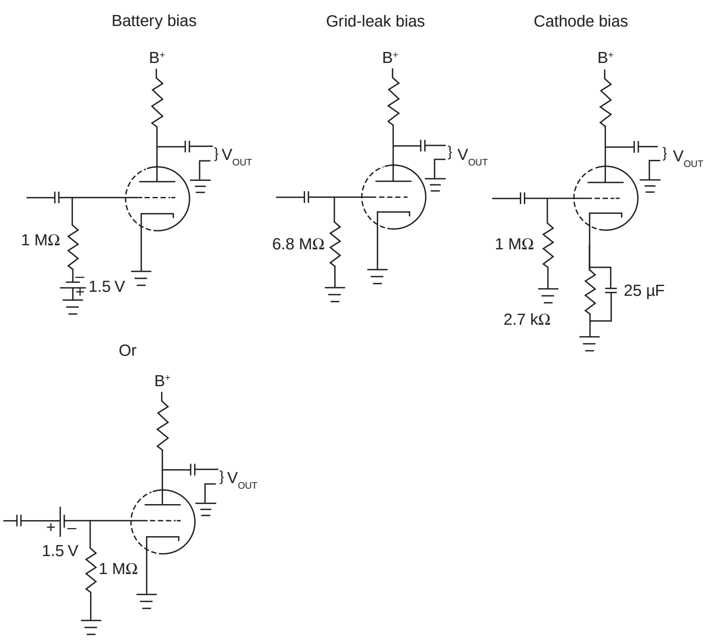


Figure 3: Biasing triode amplifier stages

VACUUM TUBE BIASING

In order to work properly in an amplifier circuit, the grid must have a DC "bias" voltage that is negative with respect to the cathode. Although we will not go into detail concerning setting the operating point (DC conditions) for tube amplifiers, we will point out that if the grid is driven positive with respect to the cathode, the output signal will be distorted. So normally, the grid's "quiescent" (no-signal) bias voltage is negative enough that the highest positive swing of the signal will not drive the grid positive. Note that the present discussion refers only to the DC conditions of the tube circuit. AC signals are ignored for the time being.

There are three ways to provide grid bias. The earliest used was a low-voltage dry cell (C) battery. (We have already met the B battery, as the source of the name B+ for the plate-supply voltage. The A battery powered the filaments.) Since the grid draws almost no current, the C battery could be connected through a large resistor (a megohm or so) without the resistor causing much voltage drop. An alternative way to implement battery bias is to connect the C battery in series with the input source, negative terminal toward the grid. This method was used in some battery-operated measuring equipment, such as the 1960s-vintage H. H. Scott 412 sound-level meter.

Note, though, that the grid draws almost no current, not absolutely no current. In fact, a few electrons do bump into the grid on their way to the plate, through what one professor calls the "pitchfork-in-the-rain effect." Thus if we connect a 5 MΩ or higher "grid-leak resistor" between the grid and ground, this small flow of electrons will result in a negative bias of a volt or so on the grid. (Remember that electrons exit the negative end of a current source but enter the negative end of a load. The tube, in this case, acts as a source; the grid-leak resistor, as a load.)

The most common method of biasing the grid is called cathode bias (sometimes also called self bias). A resistor is connected between the cathode and ground. If this resistor is not bypassed by a capacitor, the gain of the stage will be reduced. This effect will be discussed in greater depth later. The current in the tube must flow through that resistor, resulting in a few volts DC appearing at the cathode. The grid is then connected to ground through a grid resistor of perhaps a few hundred kilohm, keeping the DC grid voltage very near zero. Although the grid voltage is not negative with respect to ground, it is at DC ground voltage and the cathode is positive with respect to ground, making the grid negative with respect to the cathode. **Figure 3** shows these methods of biasing the grid of a triode. Although these biasing circuits are shown in the grounded-cathode circuit, the same approaches can be used for grounded-grid or cathode-follower stages.

Grid-leak bias is used only for circuits whose input signals are expected to be very small. For example, early guitar amplifiers often used grid-leak bias, but with the introduction of hotter humbucking pickups in the 1950s, cathode bias became more common because of its lesser tendency to be overdriven by the higher input voltages.

The considerations for biasing the control grid of a pentode are the same as for a triode, and the same circuits are used. A pentode, however, requires a bias voltage for the screen grid as well. (There have been a few circuits that used the screen grid in a different way than simply to accelerate electrons toward the plate. One such circuit was used as an audio compressor. We will limit our discussion to ordinary amplifiers, in which the screen-grid bias is a constant DC voltage.)

Most often, the screen grid is fed from the B+ supply using a screen-dropping resistor, and a bypass capacitor is connected from the screen grid to ground, keeping a signal voltage from building up on the screen grid. **Figure 4** shows a typical pentode amplifier circuit, which illustrates this screen-grid biasing arrangement.

So far, we have only considered amplifiers in which we wanted the output waveform to be as close as possible to an exact replica of the input waveform. Such amplifiers are designated as Class A. If instead of biasing the control grid so that the entire input signal wave is amplified, we bias the control grid at cutoff, then only the positive half-cycle of the wave will be amplified. This is Class-B operation, and is often used in power amplifiers, where two tubes share the job of providing a high power output. If the control grid is biased more negatively than cutoff, less than the entire positive half-cycle is amplified, it is a Class-C operation. Class-C amplifiers are mainly used in RF circuits, where resonant circuits are used to reconstruct the missing portions of the signal wave. In general, Class-A circuits are the least efficient (least number of watts out per watt input from the power supply). Class-B circuits are more efficient than Class A, and Class-C circuits are the most efficient. In a later column, we will discuss the use of Class-B circuits in audio, and also hybrid Class-AB circuits. For now, though, we'll stick to Class A.

VOLTAGE GAIN OF AMP STAGES

The voltage gain of a tube amplifier stage depends upon the characteristics of the tube itself: the operating voltages, including control-grid and screen-grid bias voltages; the load resistance or impedance of the load that the stage drives; and the presence or absence of negative feedback. We'll discuss feedback later, but for now, let's look at the gain of a simple triode stage and a simple pentode stage. The gain of a grounded cathode triode stage using no negative feedback is given by:

$$A_V = \frac{\mu R_L}{(r_P + R_L)}$$

where μ is the amplification factor of the tube, R_L is the load resistance driven by the stage, and r_P is the plate resistance of the tube.

The amplification factor typically has a value between 30 and 100. R_L is the parallel combination of the plate resistor (resistor between B+ and plate) and the external load (maybe about 220 kΩ if the amplifier drives another stage of amplification). Plate resistance r_P is the dynamic resistance of the tube (the ratio of the change in plate-to-cathode voltage to the corresponding change in plate current.) Amplification factors and plate resistance values for typical circuits can be found in tube manuals.

For example, **Figure 5** shows half of a 12AX7 dual triode in a common guitar amplifier input stage. The voltage at the plate is about 260 V.

From a 1973 *RCA Receiving Tube Manual*, one sees that $\mu = 100$ and $r_P = 62.5$ kΩ for a 12AX7 in this configuration. RL is the parallel combination of the 100-kΩ plate resistor and the 220-kΩ input resistance of the next stage, which comes out to 68.75 kΩ. Thus the gain is:

$$\begin{aligned} A_V &= \frac{\mu R_L}{(r_P + R_L)} \\ &= \frac{100 \times 68.75\ \text{k}\Omega}{(62.5\ \text{k}\Omega + 68.75\ \text{k}\Omega)} \\ &= 52.4 \end{aligned}$$

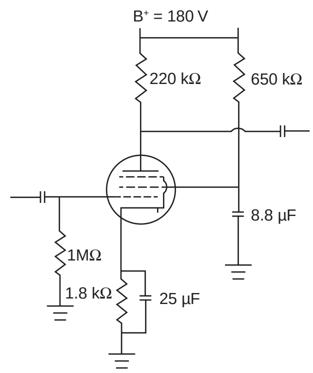


Figure 4: Pentode amplifier showing screen-grid biasing

The gain of a grounded-cathode pentode stage is calculated in the same way; however, tube manuals often do not list the m for pentodes. Instead, they list the transconductance.

Conductance G is the inverse of resistance: R = V/I, so G = I/V. In this case, V is the grid-to-cathode voltage, and I is the plate current. Since we are interested in the AC behavior of the tube, we use a lowercase g to represent (change in plate current)/(corresponding change in grid-to-cathode voltage):

$$g_M = \frac{\Delta i_P}{\Delta v_{G-K}}$$

The trans is used because, whereas ordinary conductance would deal with voltage and current at the same terminals—for example, conductance of a resistor is (current in the resistor)/(voltage across the resistor)—here we are dealing with electron current from cathode to plate and voltage between grid and cathode (symbolized by the G-K subscript). The subscript M indicates transconductance, rather than ordinary conductance G. (In years past, the term was "mutual conductance" rather than "transconductance;" hence, the "M" rather than a "T"). Thus, the idea is that a pentode is analyzed as a device whose current is controlled by the input voltage.

We can derive an amplification factor for a pentode by calculating: $\mu = g_M r_P$

If a 7199 pentode is used in the circuit shown in **Figure 4**, we find from the tube manual that gm is about 3000 microsiemens (μS). (These were formerly called "micromhos," because the inverse of resistance in ohms was conductance in mhos. Note the reversal of spelling.) Dynamic plate resistance r_P is 1 MΩ. If we assume the input resistance of the next stage to be 220 kΩ, then R_L is 220 kΩ paralleled with the 220-kΩ plate resistor = 110 kΩ. Thus, $\mu = g_M r_P$ = 3000 μS × 1 MΩ = 3000, and the voltage gain of the stage is:

$$A_V = \frac{\mu R_L}{(r_P + R_L)} = \frac{3{,}000 \times 110\text{ k}\Omega}{(1\text{ M}\Omega + 110\text{ k}\Omega)} = 297.3$$

The other amplifier configuration that is widely used in audio is the grounded plate configuration, more commonly called the cathode follower. This latter name comes about because the voltage at the cathode (from which the output signal is taken) follows the grid voltage. If the grid voltage becomes more positive, more plate current flows, and since the plate current flows in the

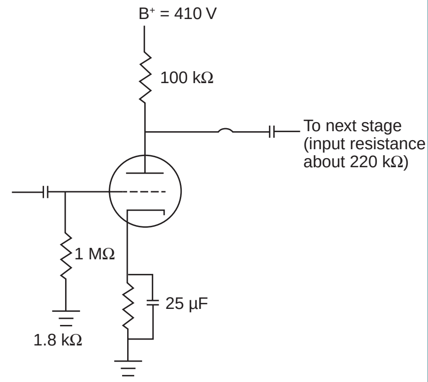


Figure 5: Triode stage for gain calculations

cathode resistor, it produces a more positive voltage at the cathode. Thus, the voltage gain of a cathode follower is near unity.

Any amplifier circuit produces some distortion, since the output/input characteristic is never completely linear. One way of reducing this distortion is to omit the capacitor bypassing the cathode resistor. The purpose of this capacitor is to keep the cathode at AC ground: to prevent the AC component of the plate current from producing an AC variation in cathode voltage that would cancel out part of the grid-to-cathode signal voltage and thus reduce the gain. If we omit this capacitor, the stage gain is indeed reduced but distortion is reduced by a greater factor. This is considered to be a form of negative feedback, in which a portion of the output voltage is fed back out of phase with the input voltage to make the amplifier behave more linearly.

The voltage gain of a grounded-cathode triode amplifier stage having an unbypassed cathode resistor R_K is given by:

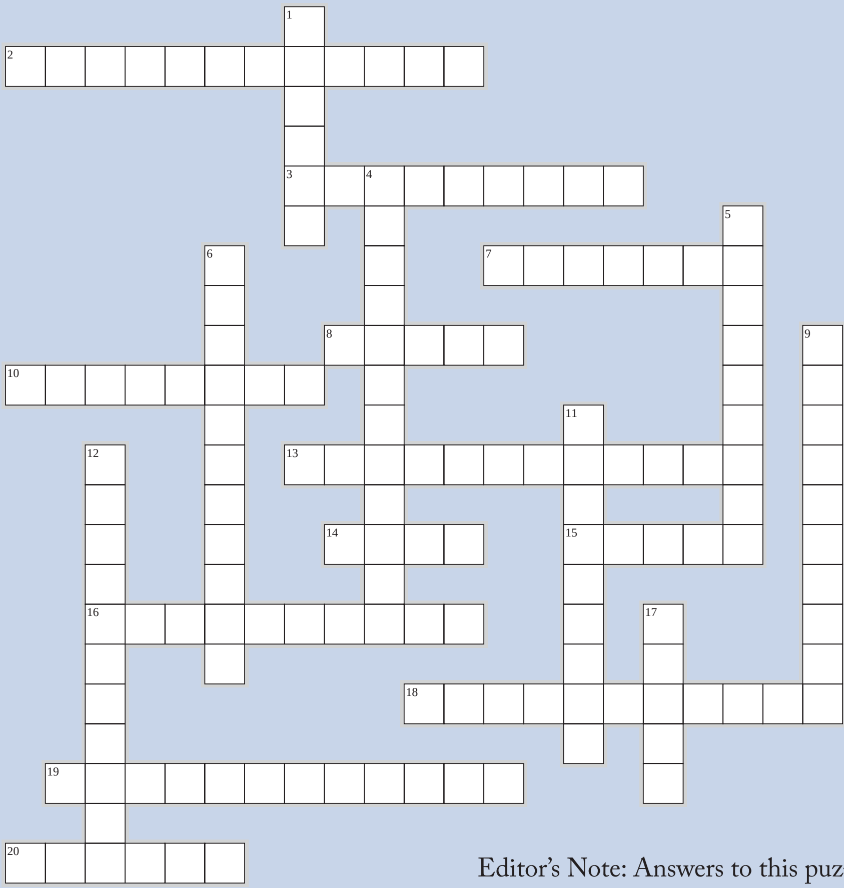
$$A_V = \frac{\mu R_L}{\left[r_P + R_K(\mu + 1) + R_L\right]}$$

Applying this formula to the circuit shown in **Figure 5**, we find that omitting the bypass capacitor drops the stage gain to 22.

In our next column, we'll look at the frequency response of amplifiers. *aX*

RESOURCE

RCA Receiving Tube Manual, Antique Electronic Supply, Tempe, AZ, 1973. It is out of print, but used copies are available online.



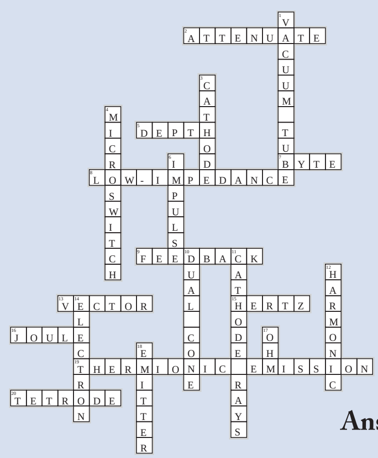
Editor's Note: Answers to this puzzle will appear in the next issue.

ACROSS

2. A delay effect created by feeding the output of a delay back into itself to cause a delay of the delay of the delay.
3. The tendency for a rapidly increasing or decreasing electrical impulse to momentarily exceed its actual peak level
7. A measure of the relative liveness of a room (two words)
8. The biasing and current-flow parameters of the active devices in a push-pull amplifying circuit
10. The ringing in the ears that often results from prolonged exposure to very loud sound levels
13. The digital representation of an analog signal, which involves sampling the amplitude of the signal at a fast rate
14. Acronym for musical instrument digital interface
15. Any unusual or unscientific improvisation that makes a device work better, or work at all
16. Opposition to the flow of direct current through an electrical circuit
18. The ratio of the peak value to the RMS value (two words)
19. Another term for micromho, the unit of conductivity
20. The material which attaches magnetic recording oxide to its plastic-tape base

DOWN

1. A specialized channel-balance control for recording, which allows a monophonic signal to be positioned between loudspeakers (two words)
4. Purposefully changing the frequency response of a circuit
5. The natural vibrating frequency of an object or a tuned circuit
6. The mean ratio of the signal output of a system to the signal output of the same system (two words)
9. Any device which changes energy from one form into another, as from sound waves into electrical impulses or vice versa
11. The fixed plate of a condenser microphone (two words)
12. The bending of sound waves around an obstacle (Huygens Principle)
17. The time relationship between two signals, expressed in degrees around a circle



Answers to the February 2012 Puzzle

Multichannel passive VU meter

PASSIVE METER BRIDGE FOR STUDIO & STAGE

SM Pro Audio's VU800 Passive Meter Bridge is the next-generation version of their popular VU8. Perfect for use in broadcast, studio, live, or just about any other audio production situation, this multichannel passive VU meter with built-in headphone monitoring is designed for use as an all-purpose visual audio level indicator.

The VU800's metering channels are completely passive and allow for "peak" or "average" metering configuration. This ensures not only that the signals remain clean and unaltered by any possible interference, but also that signal levels can be visually represented precisely as intended.

One of the more notable enhancements in the VU800 is the ability to independently configure each of the eight channels for –10 dBV or +4-dBu operation via rear-panel selector switches. The on-board headphone system provides convenient, user-selectable audio monitoring in channel pairs of the available eight channels.

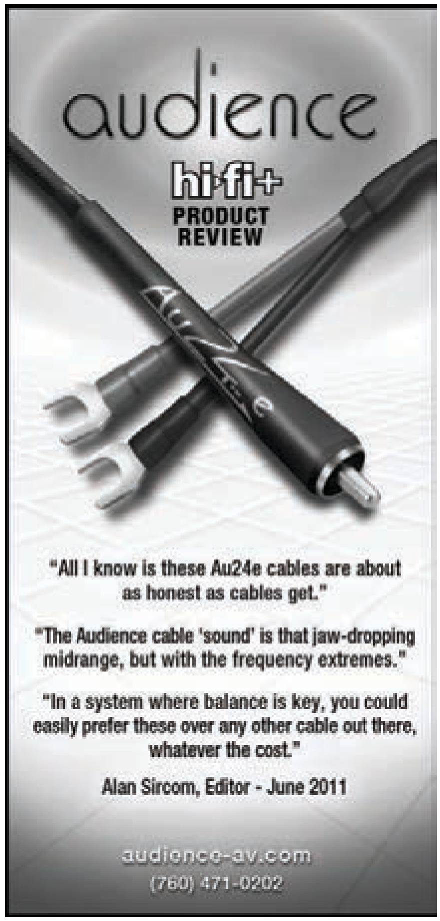
The VU800 has an estimated retail price of $349.99. Find out more about the passive meter bridge and other products at www.smproaudio.com. For a list of North American dealers, go to MV Pro Audio (www.mvproaudio.com/dealers.html).

WIRELESS SUBWOOFERS OPEN UP NEW POSSIBILITIES

Now, powered subwoofers can be placed anywhere in a room with no new wires!

Sunfire's Universal Wireless Subwoofer Kit comprises a receiver and transmitter. The kit is compatible with any third-party manufacturer's powered subwoofer and eliminates new wires, making the placement of a subwoofer anywhere in a room easier than ever before.

Available for a suggested retail price of $80 each, the SDSWITX Wireless Transmitter and SDSWIRX Wireless Receiver allow consumers to place a subwoofer near any electrical outlet, up to 25' from the audio source, enabling them to find the optimal placement in most rooms. For even more bass punch and fidelity, one transmitter can support two receivers for two separate subwoofers. And with digital signal transmission, the audio signal is reproduced with 100% accuracy. For more information, go to www.sunfire.com *aX*



WOMEN IN AUDIO

A few months ago, I received my first issue of *audioXpress*, published under the leadership of Hugo van haecke and the Elektor Group, respectively.

Like you and other readers, I too wish the best for former aX leader Edward T. Dell, Jr., and want to thank him for 41 years of the greatest published audio information anyone could ever want to read.

But now, Ed (Dell) has turned the reins of control over to the Elektor group and you, and I have my sights set pretty high on the belief that *audioXpress* will continue to be a viable publication well into the future.

As for changes at aX, the only change that I think needs to take place is getting girls and women to take an interest in audio technology (ie., design and construction). We have women doing electrical work and designing computers and building computers (and even cars and trucks!), so why haven't they lined up to get their hands dirty in our fun hobby? It could very well begin in your own home—perhaps you have a daughter or niece you could teach theory and construction skills, which in turn might lead to her building a great amplifier, thus challenging the daughters and nieces of aX readers to build their own audio gear. In the end, women would begin to subscribe to aX, drastically increasing the number of subscribers and contributors.

Thanks for reading and for a great magazine.

Neal A. Haight
Tracy, CA

READER SEEKS HELP WITH GUITAR AMP PROJECT

I enjoyed reading Patrick Brunner's article, "Upgrading the Phase Linear 200 Amplifier," in the October 2011 issue of *audioXpress*. I especially liked the part about updating the transistors with more modern units. Perhaps someone could help me out with a project I am working on.

I am modernizing a Fender Squier 15 practice guitar amp. This means replacing the electrolytic power caps with Panasonic FCs, replacing the signal and feedback path electrolytics (seven of them!) with polypropylene, replacing the ceramic caps with silver mica, replacing the 5% resistors with 1% metal film, doubling the power supply caps to 4400 µF per rail and bypassing them with polypropylene, replacing the toy power transformer with one that weights 2.5 times as much, using a bigger full-wave ridge rectifier, and replacing the IC with an OPA2134 unit.

The only thing left is replacing the transistors. I was able to find a site that gave more modern units for the small transistors. The 2SC-1815s can be replaced by NTE85 NPNs. The 2SA-1015s can be replaced by the NTE290A PNPs. But I couldn't find anything for replacing the power transistors, the 2SD-313s.

Could someone recommend a replacement for the big transistors? And would someone recommend the NTEs for the small transistors? I would like to order from either DigiKey or Mouser, and I'll consider ordering 10 of them to match them. Thanks for your help.

Steve Ball
Austin, TX

Patrick Brunner responds:

I appreciate your interest in my article. Please remember that the following advice is not guaranteed, but should be okay. Use it at your own risk and work carefully.

The 2SD313 is an older 60 V, 3 A, 30 W (at 25°C case temperature), 5 MHz Ft, NPN, TO220 cased part with a DC gain (Hfe) of 40 to 100 or so at 1 A, and drops like a rock after that.

A good choice, given money is no object, I would suggest that you try a sturdier DigiKey P/N KSE44H-11TU-ND, at $0.73 in lots of 1. It is an 80 V, 10 A, 50 W (at 25°C case temperature) NPN TO220 cased part with a DC gain (Hfe) of 100+ at 1 A and is still close to 70 at 10 A—really a nice flat gain curve. It has a much wider bandwidth too, with an Ft of 50 MHz. With the better gain and bandwidth, it is remotely possible that your amplifier may be prone to oscillate (hot transistors, flaming tweeters, etc). If so, try adding 100 pF to 470 pF from the base to the collector of the output transistors to restrict their high frequency gain. Or even easier, slightly increase the value of the capacitor in parallel with the gain setting feed back resistor (C19) from the output to the input stage. But you may not need this at all. Don't forget to properly insulate the cases from the heat sink.

I am assuming that the small transistors are TO-92 cased. For the small signal NPN parts, you could probably use a MPSA06, and for the PNP parts a MPSA56. These usually cost about $0.35 each in lots of 10 pieces. Buy several of each so you can select (using a cheap DMM with transistor testing) some reasonably matched pairs for Q2 & Q3. The suggested transistors are complementary parts with decent gain up to about 300 mA.

A small caution: if you change Q1 (or not), you may need to slightly tweak the value of R18. The voltage at Q1's collector should be about 0.5 of the voltage "A" (7-8 VDC). If it is low, increase R18's value. If it is high, decrease it. Also, this will vary with Q1's temperature, so let it cool if you just soldered it, and recheck it after the amp has run a little while and stabilized.

With all of the upgrades you have described, it is almost as if you were trying to make a guitar amplifier into a Hi-Fi one. If so, you may want to add a switch to disconnect D1 and D2, which are basically 0.65-V clamps.

These back-to-back diodes are typically used to create the "overdrive" sound effect without actually killing the amplifier. This generates a lot of higher harmonics by clamping the signal to ±0.65 V. Good luck. *aX*

CONTRIBUTORS

Vance Dickason ("Two BMS Drivers," p. 18) has been working as a professional in the loudspeaker industry since 1974. He is the author of the *Loudspeaker Design Cookbook*—which is now in its seventh edition and published in English, French, German, Dutch, Italian, Spanish, and Portuguese—and *The Loudspeaker Recipes*. Vance is currently the editor of *Voice Coil: The Periodical for the Loudspeaker Industry*, which is published on a monthly basis. For several years, Vance has been working as an engineering design consultant for various speaker manufacturers and is responsible for numerous award-winning designs, including 20 THX home theater certifications. Speakers that Vance designed that have won magazine and industry design awards include the Artison Portrait, Atlantic 170, 270, 370, 450, and 8200, and the coNEXTion Sonoma 10 and z600c.

Richard Honeycutt ("Hollow-State Amplifiers," p. 37) has been an audio enthusiast, musician, recording engineer, broadcast radio engineer (FCC First Class Commercial licensee since 1969), radio announcer, electronics repairman, tube guitar amplifier designer, speaker system designer, pro-sound/video designer, and college electronics teacher. He is now a consultant in acoustics and electroacoustics. He possesses a PhD in Electroacoustics from the Union Institute, and lives in Lexington, NC, with his wife Betty Jane, near his daughter Alyson, her husband, and two of Richard's three grandchildren.

Wim de Jager ("Tube Sound," p. 31) first contributed his article to *Elektor*. It appeared in the June 2007 issue of *Elektor* and turned out to be very popular at the time. Wim is one of countless people who have contributed articles about their projects to *Elektor*.

Mike Klasco ("The Voice Coil (Part 1)," p. 8) is the president of Menlo Scientific Ltd., a consulting firm for the loudspeaker industry, located in Richmond, CA, . He is the organizer of the Loudspeaker University seminars for speaker engineers. Mike contributes monthly to *audioXpress*. He specializes in materials and fabrication techniques to enhance speaker performance.

David Pellegrene ("The Panheads," p. 14) is a remodeling contractor and has been in the business for over 30 years. David's favorite part of speaker building is the cabinet design. One of the reasons he got into the hobby was to learn more about electronics.

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