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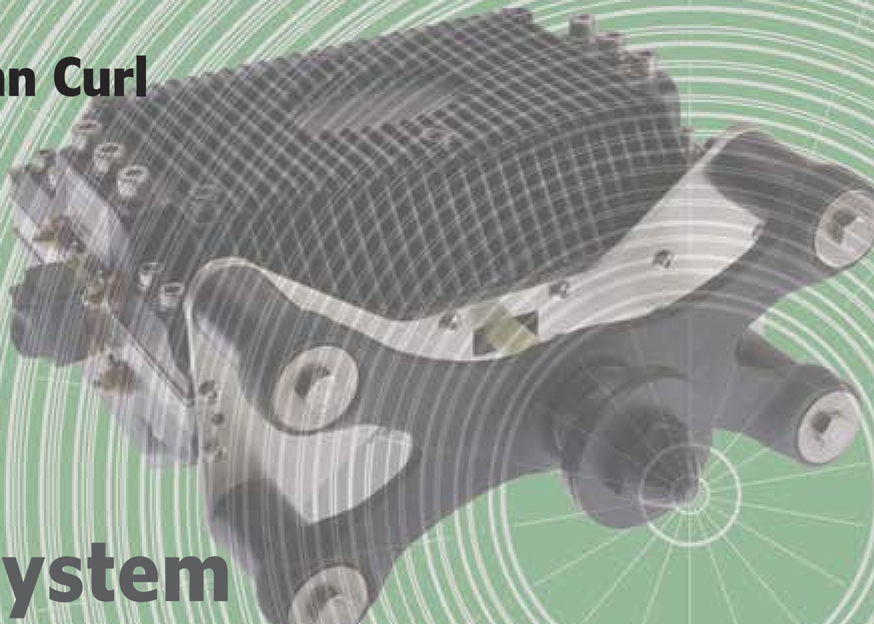

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SEPTEMBER 2015

September 2015 **ISSN 1548-0628**

www.audioxpress.com

audioxpress (US ISSN 1548-0628) is published monthly, at $50 per year for the US, at $65 per year for Canada, and at $75 per year Foreign/ROW, by Segment LLC, Hugo Van haecke, publisher, at 111 Founders Plaza, Suite 904, East Hartford, CT 06108, US Periodical Postage paid at East Hartford, CT, and additional offices.

Head Office:
Segment LLC
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Speaker Design—The Cooks and the Wizards

audioXpress—and our sister magazine *Voice Coil,* the only industry publication focused exclusively on loudspeakers—targets the audio industry as a whole, not just the finished products but also the product development side.

Among the areas we address, speaker design is one of the most fascinating areas for me. I have had the privilege to meet and exchange views with some of the legends in electro-acoustic science and also meet with driver and speaker part manufacturers—even manufacturing equipment suppliers—and gain privileged access to industry associations, of which we naturally endorse the role of the Association of Loudspeaker Manufacturing & Acoustics (ALMA) International and its associates.

Through the eyes of many of those industry experts, it's possible to gain a broader vision on how to approach the art and craft of speaker design. I've met brilliant minds that approached their creations based purely on self-acquired knowledge and research, deliberately trying to distance themselves from existing trends and debates. Other similarly brilliant designers base their work on team efforts, industry partnerships, and sometimes market-oriented options to achieve the best economies of scale and sensible solutions for their target consumers. These are the people I've started to categorize as the Wizards and the Cooks.

The categorization is valid for designers targeting the absolute high-end market—where price is not a primary consideration—as well as for mass-market portable PA solutions. I was even more amazed to discover the same "personalities" in speaker designers targeting in-wall and ceiling speakers and studio reference monitors.

Of course, as anyone can discover by reading some of the industry "bibles" on speaker design—of which I would recommend Vance Dickason's *Loudspeaker Design Cookbook*—understanding the basics is essential to achieve the best results. As the vast majority of authors who submit DIY projects to *audioXpress* will agree, following a recipe is a great way to learn and perfect this craft. It's no coincidence that Vance's books are titled "cookbook" or "loudspeaker recipes." But even Vance Dickason would agree that, once in awhile in the history of loudspeaker design, there are great examples of achievements that seem to defy logic and science. Also, great technology advancements result from "Eureka moments" and those tend to gain more visibility than achievements resulting from pure engineering method and experimentation.

In speaker technology, those advancements have proven crucial in areas such as audio electronics and signal processing, material research, and driver design. Still, most loudspeakers combine a solid knowledge of established designs, which can be perfected with the right amount of attention to detail, incremental improvements, and always where possible, specifying new technologies and materials.

audioXpress tries to make that engineering process visible when we publish reviews or R&D stories. We want to address how R&D efforts are combined with sourcing strategic partnerships with suppliers, providing expert advice on innovative solutions and how to customize critical components.

Loudspeaker driver companies continuously renew their catalogs of ready-to-use solutions, benefiting from their own research, manufacturing improvements, precision engineering, and sometimes, pure innovation. On the other hand, every time we see a new innovative speaker, we find the usual reference to "custom-made drivers." And yet, few manufacturers even have the ability to design speaker drivers in-house, let alone have the manufacturing capabilities to turn something simulated in software into a successful working sample.

That's why it is so important for all speaker designers to understand how suppliers such as speaker driver/transducer manufacturers operate. When approached by a speaker designer with a new project—and a solid business plan—they can act as expert-consultants, working as an extension of the original R&D department.

In speaker design, those partnerships are essential to convert great projects into reality. It's always remarkable to see apparently simple designs, "cooked" in-house, result in outstanding products. Sometimes it actually seems like wizardry.

João Martins
Editor-in-Chief

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COLUMNISTS

Vance Dickason has been working as a professional in the loudspeaker industry since 1974. He is the author of *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 the editor of *Voice Coil: The Periodical for the Loudspeaker Industry*, a monthly publication. Although he has been involved with publishing throughout his career, he still works as an engineering consultant for a number of loudspeaker manufacturers.

Dr. Richard Honeycutt fell in love with acoustics when his father brought home a copy of Leo Beranek's landmark text on the subject while Richard was in the ninth grade. Richard is a member of the North Carolina chapter of the Acoustical Society of America. Richard has his own business involving musical instruments and sound systems. He has been an active acoustics consultant since he received his PhD in electroacoustics from the Union Institute in 2004. Richard's work includes architectural acoustics, sound system design, and community noise analysis.

Mike Klasco is the president of Menlo Scientific, a consulting firm for the loudspeaker industry, located in Richmond, CA. He is the organizer of the Loudspeaker University seminars for speaker engineers. Mike specializes in materials and fabrication techniques to enhance speaker performance.

Steve Tatarunis has been active in the loudspeaker industry since the late 1970s. His areas of interest include product development and test engineering. He is currently a support engineer at Listen, in Boston, MA, where he provides front-line technical support to the SoundCheck test system's global user base.

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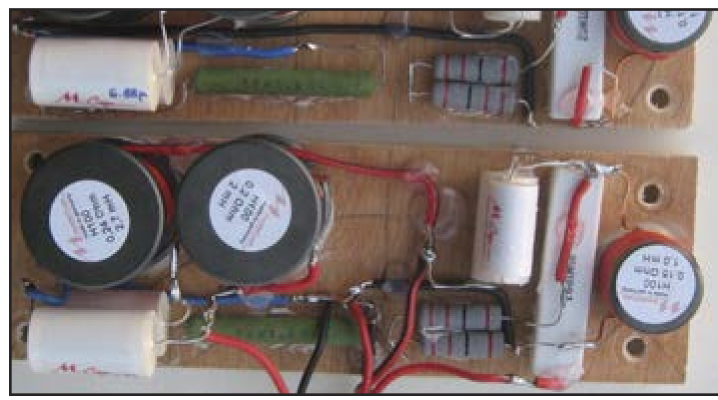

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Speaker Focus Edition!

Volume 46 – No. 9 September 2015

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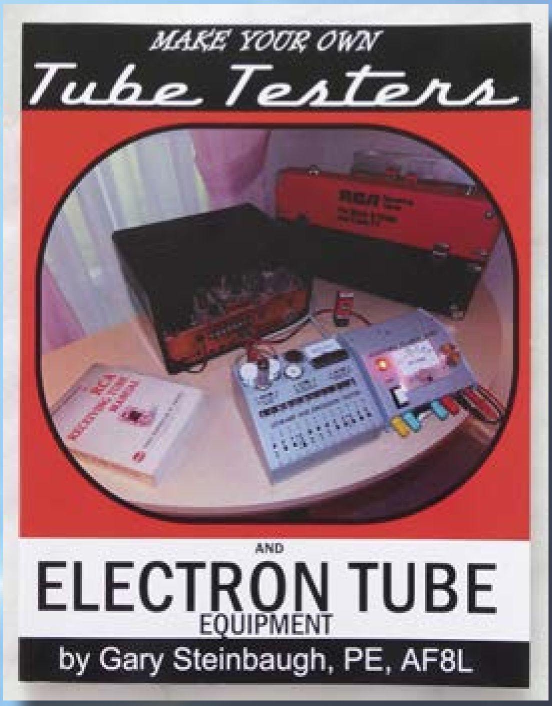
Exploring *Make Your Own Tube Testers*

With the renaissance of tube technology, today's experimenter is faced with three choices: find and restore an old tube tester, buy a modern computer driven curve tracer, or build one. Gary Steinbaugh's book, *Make Your Own Tube Testers and Electron Tube Equipment* offers DIYers help with the third option.



Review By
Stuart Yaniger
(United States)

Those of us with more grey in our hair than we'd care to admit can remember the days when vacuum tubes were ubiquitous in consumer electronics. Because of their perishable nature, they were a constant source of product failure.

Recognizing a consumer need, tube companies put do-it-yourself tube testers in almost every drug store in America, with an array of different sockets, switches to set the test parameters and pinouts, and a roll chart of tube types with the switch settings for each one.

There was, of course, a caddy of replacement tubes helpfully stashed in a cabinet below the array of sockets and knobs, which was really the whole point of providing this service. Just about anyone could operate these simple machines, and the big meter with "Good" and "Bad" provided reassurance to the non-specialist. These machines were relatively crude, and measured for shorts, gas, and emission. More subtle defects were overlooked, but at least these covered the "80" of the 80:20 rule ("80% of problems come from 20% of issues"). And indeed, most problems with radios and TVs could be fixed by Joe Everyman (and in those days, it was nearly always a man) by pulling all the tubes and carting them over to the tester to determine which ones were bad.

Make Your Own Tube Tester and Electron Tube Equipment
By Gary Steinbaugh
Publication Date: 2013
Format: Paperback
Pages: 240
ISBN: 978-1-57074-089-3
Greyden Press

Repair shops generally had more sophisticated (and portable) versions of these testers, with added features such as transconductance checks, which were typically done at a single point. More advanced experimenters and designers needed to be able to measure all tube parameters (e.g., mu, transconductance, and plate resistance) as a function of operating point. Doing this manually, point by point, took very little equipment (meters and variable power supplies) but was a tedious and time-consuming process. Nonetheless, those of us who couldn't afford to buy a Tektronix curve tracer bit the bullet and did it.

Steinbaugh is a licensed Professional Engineer and a ham radio operator who (like me) grew up in the tube era. This book (which I'll call *MYOTT* for brevity) is in a sense a brain-dump of things that Steinbaugh wants to pass along to younger folks, keeping the old knowledge alive.

The first section covers basic tube equations and characteristic curves. The basic parameters (mu, transconductance, and plate resistance) are defined and explained. There's not much new here that isn't covered in more depth elsewhere, but one nice touch is some photos of 3-D tube curve models the author constructed. This reinforces the idea that the curves we generally visualize are actually slices of a 3-D surface (or even more for pentodes and other multigrid tubes), and that when searching for operating points, we need to simultaneously consider all the variables—plate voltage, plate current, and grid voltage—involved.

MYOTT then describes common tube faults and the basic service testers used in repair shops to detect them. There are several quotes and paraphrases from Robert Tomer's classic *Getting the Most Out of Vacuum Tubes*, (Audio Amateur, 1960) and many photos of tube testers from the 1950s and 1960s. Some simplified schematics explain their basic operation. The section ends with a short survey of some "old time" parts and a brief elementary explanation of the basic terms RMS, peak, accuracy, and precision.

At this point, *MYOTT* veers back to its raison d'etre, the construction of a service-type tube tester, capable of measuring emission and leakage. The construction diagrams are quite detailed, but it does rely on some rather hard-to-obtain switches. However, most experimenters will find it relatively easy to substitute more commonly available rotary switches for the multi-position slide switches used in the prototype.

Next up is an adjustable filament (more properly, heater) supply, based on a standard LM338 circuit. It is capable of 0 to 15 VDC at 4 A. This is followed by a series of photographs of laboratory tube testers and curve tracers. Then, the book provides a review of high voltage power supplies.

Perhaps the most useful part is an examination of a simple but effective pentode-based regulator that incorporates both feedback and feedforward. This is a circuit not commonly seen today, but it deserves further examination. Construction details are provided, but there are no measurements or calculations of basic power supply performance (e.g., ripple, noise, load regulation, line regulation, or source impedance). Steinbaugh uses these supplies to perform point-by-point measurements of tube characteristics and gives detailed instructions on how to do so.

The book ends with a short section on construction, offering some of the basic projects that DIYers used to find in Frederick Collins' *The Radio Amateur's Handbook* (Harper & Row, 15th revised edition, 1983) and photos of some example projects.

Review Findings

First, the book's organization leaves something to be desired. It lurches a bit from topic to topic, with no real guiding theme or logical arrangement.

Second, it is somewhat unfocused. In his desire to provide a wide variety of information, Steinbaugh can't seem to decide what the book is really supposed to be—a DIY guide to test equipment, a nostalgia collection, a survey of tube operation, or a design guide to power supplies.

Third, the use of tube high voltage supplies for measurement purposes strikes me as a bit off. Yes, it's fun to build them. Yes, they look cool. Yes, they have some cred in audiophile circles. But for metrology, the important characteristics are stability, efficiency, noise, and repeatability, and as much as I love tubes and tube design, there is no way that a tube power supply regulator can come close to matching a good solid-state design.

I was reminded of the line from the TV show *Breaking Bad*: "You don't want a criminal LAWYER, you want a CRIMINAL lawyer." In this case, you're not building a tube TESTER, you're building a TUBE tester. That said, these supplies would work well as components in an audio amplifier. If you want to get accurate and efficient measurement of tube characteristics, you might do better with one of the modern PC-based curve tracers.

My main issue with *Make Your Own Tube Testers and Electron Tube Equipment* is the ratio of "filler" to "meat." There's some really good information in here (especially the sections on tube regulator design), but far too much of the book comprises cut-and-paste photos from public domain sources, Internet-sourced quotes, scans of old advertisements, public domain pictures of old scientists each of which fill an entire page, and restatements of topics that have been well-covered by others without clarifying or adding anything new. The useful contents of the book could be condensed down to two or three *audioXpress* articles without losing very much.

While being appreciative of what Steinbaugh is trying to do here and being fully convinced of his sincere passion for the subject, I just don't see $35 of value. If this were available in an electronic format at a much lower price, I'd have an easier time ignoring the unimportant and focusing on the valuable.

About the Reviewer

Stuart Yaniger has been designing and building audio equipment for nearly half a century, and currently works as a technical director for a large industrial company. His professional research interests have spanned theoretical physics, electronics, chemistry, spectroscopy, aerospace, biology, and sensory science. One day, he will figure out what he would like to be when he grows up.

Resources

F. Collins, *The Radio Amateur's Handbook,* 15th revised edition, Harper & Row, 1983.

R. Tomer, *Getting the Most Out of Vacuum Tubes*, Audio Amateur, Inc., 1960.

The Armature Driver

When it comes to in-ear monitor earphones, most designers opt for the armature driver. This article details the appeal of a balanced armature transducer and provides some insight on the device's history.

By
Mike Klasco and Steve Tatarunis
(United States)

For the In-Ear Monitor (IEM) earphone connoisseur, the balanced armature transducer is the driver of choice and the more of these devices in each earpiece, the better. An armature driver is about a quarter the size of a 13.5-mm dynamic driver, yet has equivalent sensitivity. The compact size enables two-way (woofer/tweeter) and three-way configurations, with the number continuing to trend upward.

The armature electro-dynamic configuration was originally used in early telephone/ radio receiver communication headsets dating back to the 1920s. These early designs were incredibly efficient by today's standards. For the last half century, it has been mostly used for hearing aids due to its ideal form factor and minuscule size. Larger variants of armature devices have been placed in back-up warning beepers on commercial vehicles and in other audio for applications such as VU meter movements and even in phono cartridges.

A balanced armature driver employs a voice coil just like a conventional moving coil dynamic driver. However, unlike the coil in the dynamic driver, the coil in a balanced armature transducer is in a fixed position, just like the magnet. The moving part is not a diaphragm but rather a long arm (the armature) that is placed through the coil's center. The armature does not come in contact with the magnet. Instead, the arm is held in place in its center, so it can still move (pivot) and is the source of the driver's sound radiation (see **Figure 1**).

Armature Driver Designs

Not easy to design or fabricate, typically armature drivers use manual or semi-automatic production techniques. Depending on the factory, armatures have a reputation for being variable in unit-to-unit consistency, hard to use with many of the popular simulation modeling tools, and expensive.

Designers who implement armature drivers in IEMs seek greater refinement from their initial application as hearing aids. Armature drivers generally do not reflect the same recent advances in materials made by most of the dynamic driver vendors. Sophisticated magnet wire, ferrofluids, high-tech armature materials, and so forth have only recently been introduced by armature driver vendors.

Balanced armature design is a combination of wisdom and witchcraft. There are several design considerations that influence a driver's final performance. The maximum output is limited by the enclosure size (larger = more output) and more often than not, the enclosure size is dictated by the size of the IEM. Next, the designer must decide if single or dual (or more) armatures will be used. A single armature offers the highest efficiency but dual designs provide better sonic performance. The volume behind the membrane/diaphragm also influences output and sometimes designers use vent holes to increase the apparent back volume.

Armature Motors

Once these mechanical aspects are in place, the motor design can be addressed. The armature is typically U-shaped but its thickness, stiffness, and width can be adjusted to optimize parameters (e.g., low frequency extension, overall bandwidth, sensitivity, frequency of resonance, etc.). The membrane divides the driver's back and front chambers. Its location defines the volume of these chambers. Membrane size determines the low-frequency output while stiffness and mass combine to limit the upper response.

Fine-tuning the driver response can be achieved via damping screens on the driver's output port and/or introducing tiny "compensation holes" into the membrane itself. Last, armature shock protection can be applied either by the use of mechanical bumpers or ferrofluid. Shock protection can prevent fatal damage to the armature with the benefit of lowering distortion at high output levels, but it may reduce the driver's maximum output potential.

System design, not just the drivers—especially in the case of two-way or multi-way IEMs—is an art form (see **Figure 2**). The location of the drivers, lining up of the acoustic centers, the artful optimization of the crossover network, the casing material, and the eartips all contribute to their good, bad, and ugly attributes.

Armature Driver Manufacturers

While there are endless earphones assembled from armature drivers, there are only a half dozen companies that build the armature driver itself (see **Figure 3**). Knowles is the most established manufacturer of armature drivers for IEMs, having introduced its first wideband armature drivers for this market in 2007. Knowles has factories in China (Suzhou, Weifang, and the Beijing vicinity) and Malaysia (Penang), although some of these locations are also used for microelectrical-mechanical system (MEMS) microphones and microspeaker production. A few years back, Knowles bought the microspeaker business from nxp, and there has been some downsizing as the two groups converged.

Several European hearing aid companies have manufactured armature drivers, mostly for in-house requirements of their own assisted hearing devices. One of these is Sonion, which offers its balanced armature products on an ODM basis. Originally known as Microtronic, the company bought a Dutch armature device company in the mid-1990s and later moved production to Poland. The company changed its name to Sonion in 2000. Its armature driver production eventually shifted to Malaysia, with a recent opening of an armature factory in the Philippines.

The legacy of armature production lines shifting locations and downsizing appears to have created the right environment for startups. Today, a half dozen new armature manufacturers have popped up in China. Yet, not all armature drivers are copies of Knowles and Sonion, and we still have not discussed the Japanese contingent of Sony, Yashima, Final Audio Design, and Star Micronics.

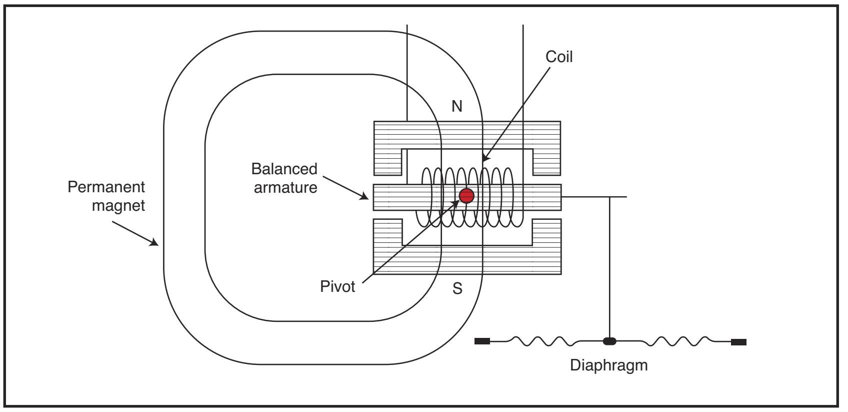


Figure 1: This is a simple schematic diagram of a balanced armature driver. (Image courtesy of JA Davidson, English Wikipedia)

Star Micronics makes electro-mechanical and mechatronic gizmos. If you owned a dot matrix printer (or its print-head guts), they might have come from Star Micronics. Its armature drivers typically find homes in hearing aids but not in IEM monitors. The other Japanese manufacturers are more involved in IEMs. There are also new topologies of armature drivers such as the Sony Quad-Balanced armature and the newly developed Yashima single magnetic pole used in Ortophon and Grado IEMs.

Armature Driver Variations

In the legacy balanced armature (BA) transducer, the "arm" where the miniature coil rests is placed between two magnets (this is the "balanced armature" part). When electrical signal is applied to the coil, the arm flexes back and forth according to the change of magnetic field and pushes a rod at the end of the arm. The rod transmits the motion to a membrane and creates the sound. In contrast to BA, the dynamic (moving) transducer coil is placed inside a single larger magnetic field (a rounded magnet). The coil's motion is directly transmitted to the diaphragm since they are attached to each other. The difference in construction reflects the differences

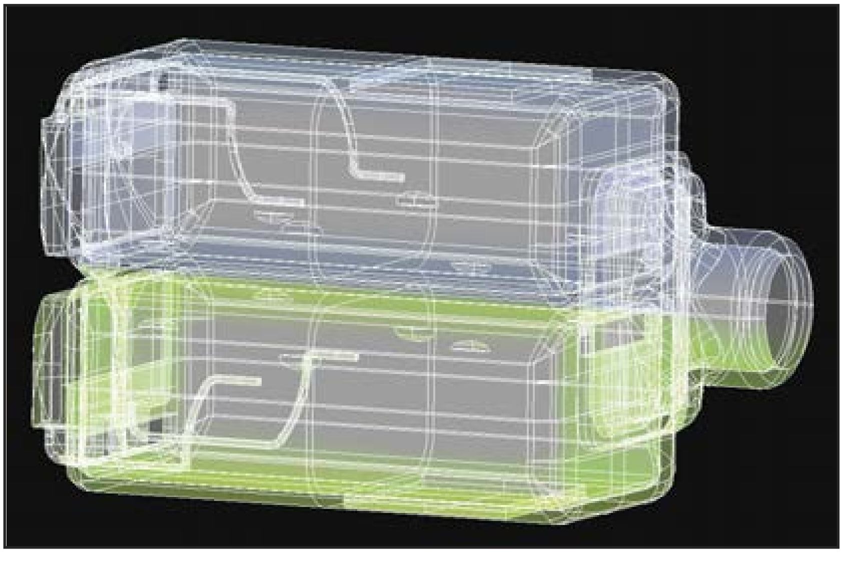


Figure 2: This is a 3-D drawing of a dual-armature package. (Image courtesy of Knowles Electronics)

Figure 3: Here is an exploded view of a armature drive in-ear monitor (IEM). (Image courtesy of Knowles Electronics)

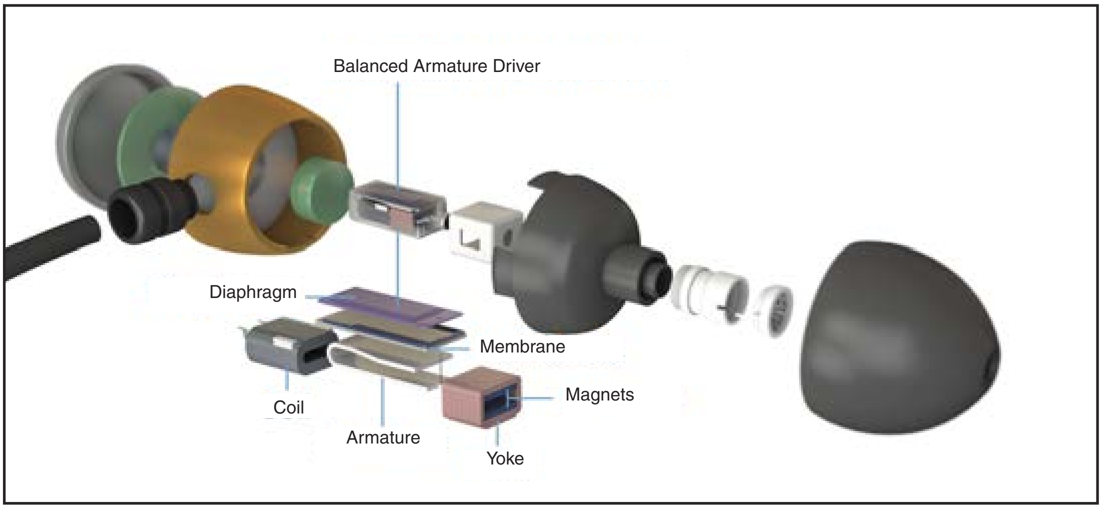


Photo 1: The Grado GR8 has a proprietary wide bandwidth moving armature design that provides the listener with great performance. (Photo courtesy of Grado)

Photo 2: The The Sony XBA-4 is the flagship model in the XBA line and uses four balanced armature drivers (full range, tweeter, woofer, and super woofer drivers) in each earpiece. (Photo courtesy of Sony Corp.)

in sonic characteristics. BA is often better at detail, speed, and resolution while dynamic is often better at "bass pumping power" due to its mass and air moving excursion.

This variant of the original type armature is called the "moving armature" and is similar to a dynamic transducer without the moving coil. Instead, the diaphragm is connected to a large armature placed in a single magnetic field. This interesting alternative is used by the Japanese Yashima single magnetic pole configuration, which can be found in the Ortophon e-Q7, the Grado GR8, and a few other premium earphones (see **Photo 1**).

Sony's XBA sub-brand, which also stands for Xperience Balanced Armature, is comprised of four models, having single through quadruple armatures. The armature drivers were designed for music (hearing aid design criteria have not compromised the design).

On the inside, the armature is damped stainless steel laminated to polyethylene terephthalate (PET) film. There is also a high-grade resin casing along with an outer cosmetic casing, which provides high-resolution and high-sensitivity audio along with high sound isolation. The new design also provides size and weight benefits that come along with miniaturization.

There is one other Japanese innovative armature driver manufacturer, Final Audio Design. Long known for audiophile products (e.g., phono cartridges, tone arms, turntables, and speakers), the company has also developed an armature driver and a wide range of armature earphones. While in Munich for the High End Show in May, Mike Klasco had the chance to listen to Final Audio Design's excellent sounding armature IEMs. The company has also coined a new term for its unique design, Balance Air Movement (BAM), which provides a bit more oomph to the bass.

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Sound Cards for Data Acquisition in Audio Measurements (Part 4)

The Missing Link

In this article series, we are examining ways to create a low-cost system for lab-grade audio electronics measurements. Thus far, I hope the first three articles in this series have whetted your interest in inexpensively extending your measurement capabilities. Now, we must select the right interface.

By
Stuart Yaniger
(United States)

In the first few installments of this series, I looked at the basic hardware and software aspects of using computer sound cards for audio measurements. The goal is to create an inexpensive (less than $200 or so) system that can do a variety of audio measurements, which was previously the exclusive dominion of expensive test gear (e.g., distortion meters, spectrum analyzers, and impedance bridges).

Thus far, I have provided an overview starting with the computer, moving on to the sound card, and some of the software options. Now, we're only a step away from the lab bench, so let's consider the crucial part where the metaphorical rubber meets the road—the interface to the device under test (DUT) that you want to characterize (e.g., an amplifier, a phono cartridge, a CD player, or a loudspeaker).

How do we get the sound card and the DUT to "talk" to one another? The sound card will likely have an input impedance of 10 kΩ or so. It will also have an input and output voltage capability of 2 V or so. The former means that measurement of any circuit with a moderate or high impedance will be loaded down by the sound card (and the cable connecting it to the sound card) and give inaccurate values. The latter means that low-voltage measurements will be compromised by noise, high voltage measurements will overload the inputs and potentially damage the sound card, and low impedance loads or loads that require significant voltage drive will cause the sound card to prematurely stop working.

Audio Interfaces

These limitations aren't as onerous as they might seem at first glance. We can certainly measure preamps, DACs, CD players, and similar line level components without much fuss. Since these are levels comparable to a microphone preamp output, we can also do loudspeaker frequency response measurements without difficulty. But, what about power amps? High impedance circuits? Phono cartridges?

Traditional instruments (i.e., oscilloscopes, spectrum analyzers, distortion analyzers, wave analyzers, and AC voltmeters) typically feature high impedance (1M or higher) to avoid loading down higher impedance circuits. For different voltage levels, there is generally a stepped attenuator to specify the voltage range, so the instrument has the ability to measure signals in the microvolt range, and then with the twist of a knob, hundreds of volts.

We can add some of these features directly to the sound card measurement system by setting up external jigs. **Figure 1** shows the schematic for one that I have already built. It is a unity gain single-ended circuit with a 1-M input impedance. The input is capacitively coupled so that DC will not affect the

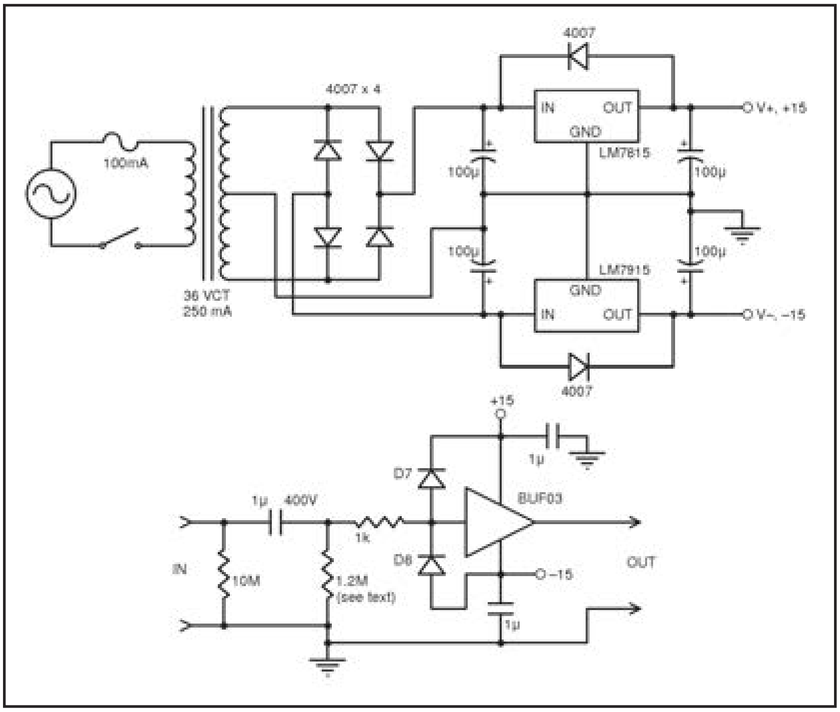

Figure 1: This is a simple single-ended buffer to use between the device under test (DUT) and the sound card to give a high input impedance and input protection.

performance of the jig or—in the case of voltages commonly found in tube circuits—burn out the jig and/or the sound card. In my usual low-rent fashion, I built it using point-to-point wiring on a perfboard inside an inexpensive RadioShack project box. I built this as a two-channel device. Typically, I use one channel between the DUT and the sound card's input, since the jig's 1-M input impedance significantly reduces the loading on the DUT compared to the 10 to 20 kΩ of the sound card's direct input. I use the other channel when I need to drive a low-impedance load or one with significant capacitance where the sound card output might be compromised.

The buffers I used (BUF03) are no longer available, but there are several newer parts that will give equivalent or even better performance. You'll want a buffer with low-bias current to successfully keep the input impedance high. An excellent example is the EL5x23 series from Intersil, which has lower noise than the old BUF03s in my interface box.

The Intersil part does not have the BUF03's current capability, but with a 30-mA drive available, it should be sufficient for most applications. If you need more drive (e.g., for applying signal to a highly capacitive load), you can use the EL5x23 to drive the input of a Linear Technology LT1010 buffer. The LT1010 buffer has 150 mA of drive capability, but high input bias current, so its input needs buffering to maintain a high input impedance. If you're using a low-impedance source and don't care about having a high input impedance (e.g., for being driven directly from a sound card output), you can substitute the LT1010 into my circuit by changing the input resistor from 1M to something between 2 kΩ and 4.7 kΩ.

In either case, the output impedance of the buffer will be quite low—the BUF03 is 2R, the LT1010 is 5 to 10R, and the EL5x23 is not specified. However, my measurements suggest that it's under 5R. If you require a higher source Z for a measurement, like 600R for studio equipment, a resistor can be tacked in series with the output. That's the one advantage of the DIY construction that I used compared to a nice PCB—it's easy to add temporary parts for measurement.

For oscilloscopes (and often the other instruments), the capacitive loading of a circuit by the test cable to the instrument, which can run in the hundreds of picofarads, can be reduced by using a 10x scope probe. This decreases capacitance to a few picofarads but drops the measured voltage by a factor of 10. Those probes aren't just limited to oscilloscopes—they also can be connected to this interface. The probe will expect the loading of the interface box to be 1M, so this is why that value was chosen. The 10x reduction in measured voltage is acceptable for larger signals and can, in fact, extend the range of measurement for the sound card to 20 V or so. Likewise, you can also buy x100 probes, which will enable you to measure to 200 V. Again, the load on the instrument end of the probe needs to be 1M, so an interface box like mine is a must. And BNC connectors are standard for this application.

Millett's Interface Box

If you prefer a more convenient, more versatile, neater, and lower noise solution (the BUF03 is not very quiet), Pete Millett has designed a nice interface box with switchable voltage range from 200 mV to 200 V, a built-in AC voltmeter, and a choice between single-ended and quasi-balanced inputs and outputs. The interface either amplifies or attenuates the signal so that the sound card is presented with a 2-V signal at the maximum of the selected range of the interface. It uses 0.25" stereo phone plugs for balanced connection to the sound card input and output, and BNC plugs for the DUT's signal output and input. Power is taken from the computer's USB port.

One issue I have with the Millett interface is that its input impedance is 100 kΩ rather than the 1M I would prefer. Millett's logic is that higher input impedance could potentially lead to more noise. I am unconvinced since his chosen input amplifier is

About the Author

Stuart Yaniger has been designing and building audio equipment for nearly half a century, and currently works as a technical director for a large industrial company. His professional research interests have spanned theoretical physics, electronics, chemistry, spectroscopy, aerospace, biology, and sensory science. One day, he will figure out what he would like to be when he grows up.

the OPA2604, which has extremely low input current and current noise. Likewise, the input impedance is shunted by the DUT's source impedance, so that would be expected to reduce the effect of the input impedance on the interface's noise. If you do what I suggest (and Millett has updated his webpage to mention it, along with his reasons for not recommending it!), increase the values of R4 through R7 by a factor of 10.

As a practical matter, when I did this, I couldn't see any significant increase in noise. And the 1 M input impedance enables us to use the 10x probes with a greatly reduced capacitive loading on the DUT. If you think all your measurements will be of low impedance sources, use Millett's specified values. Remember that using the 10x probe will change the voltage ranges shown on the selector switch and AC voltmeter by a factor of ten.

The balanced outputs are provided by DRV134 differential line drivers, so the source impedance is about 50R and current drive is up to 50 mA.

As with any sort of measurement, grounding of the measurement interfaces, sound card, and DUT

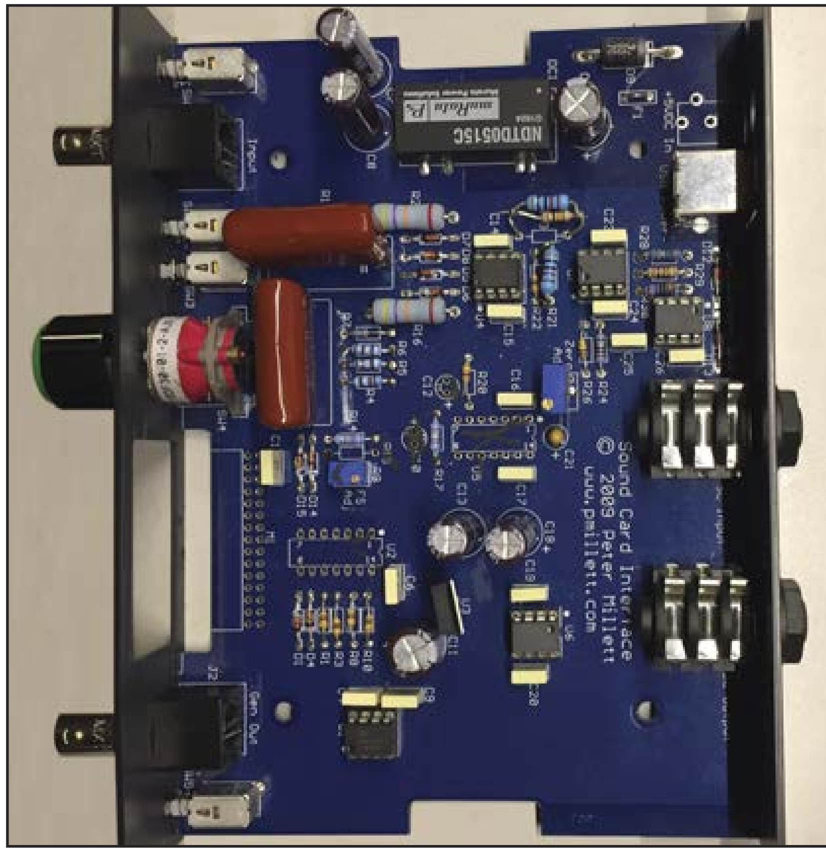

Photo 1: One option is to use Pete Millett's interface box design. The "missing" components are for the optional digital voltmeter.

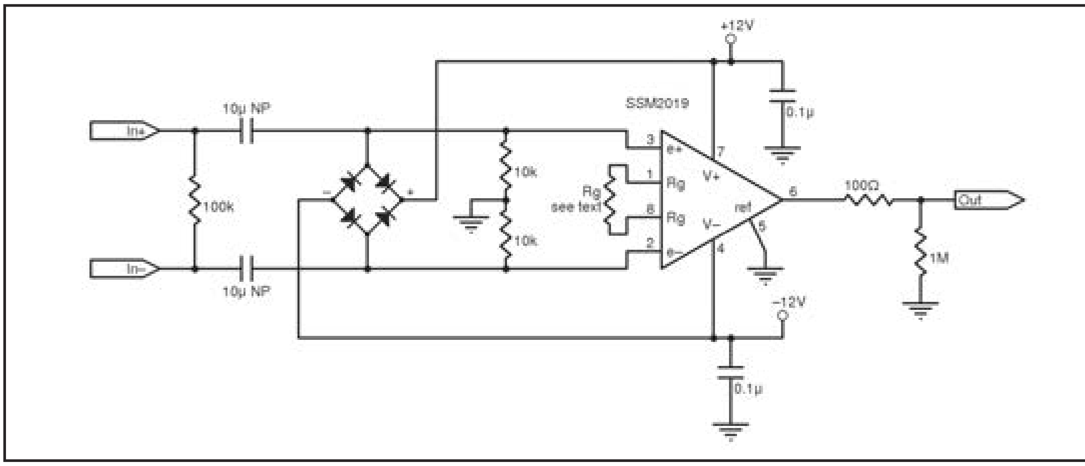

Figure 2: A balanced differential input amplifier with high common-mode rejection will measure small signals. The Rg's value determines gain.

can be a bit of trial and error. I found that if I powered the interface from a battery pack, I had an easier time avoiding ground loops and noise. A stack of four D batteries gave me 6 V at the power input, and my test setups became much quicker to optimize.

All in all, I would strongly recommend starting out with Millett's interface for speed, flexibility, and expediency. If it had been around 10 years ago, I probably wouldn't have bothered building my own. If you're ambitious, you can play around with upgrading the op-amps and line drivers, altering some of the grounding, and building a battery pack, but that's somewhat gilding the lily. My own build of this box omitted the internal voltmeter, which made the construction a very inexpensive proposition (see **Photo 1**).

Resources

Analog Devices, Inc., "Self-Contained Audio Preamplifier, datasheet, 20111, www.analog.com/media/en/technical-documentation/data-sheets/SSM2019.pdf.

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P. Millett, "Sound Card Interface," www.pmillett.com/ATEST.htm

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SSM2019 Audio preamplifier
Analog Devices, Inc. | www.analog.com

EL5x23 series
Intersil | www.intersil.com

LT1010 buffer
Linear Technology Corp. | www.linear.com

OPA2604 Op-amp and DRV134 differential line driver
Texas Instruments, Inc. | www.ti.com

A Question of Balance

Measuring small signals is even more of a challenge. Noise pickup and gain are the key issues. Analog Devices (ADI) has come up with a terrific solution aimed toward the microphone market—the SSM2019 balanced input audio preamplifier. This chip can run balanced or unbalanced inputs (which is useful if your sound card only has single-ended inputs). The chip also has impressive common-mode rejection and can be varied in gain from 0 dB (unity gain) to 60 dB (gain of 1,000) by selecting a single resistor. It has a single-ended output and boasts outstandingly low input voltage noise (about 1 nV/rt Hz).

This is perfect for looking at the output of a microphone or phono cartridge—a 10R gain setting resistor will give a gain of 1,000, which is suitable for most applications like this. For other gains, use the resistor values recommended on the SSM2019 datasheet (see Resources).

Figure 2 shows the circuit I used. The diode bridge at the input is for overvoltage protection. The type of diode used is noncritical, but try to find one with relatively low leakage and reverse capacitance. I used UF4007 because I had a drawer full of them. The power supply is also noncritical. As before, I used batteries to prevent any possible ground loops. I also built my unit on perfboard and housed it in a small project box. The input connector choice is yours. For convenience, I used XLR connectors. I plan to use this interface in some demonstration measurements discussed in a future article in this series.

Things to Come

Next month, we'll delve into a bit of the basic math to understand Fourier Transforms at a level that will enable us to choose appropriate software settings for good distortion and frequency response measurements.

Enhanced Stereo (Part 3)

Adding Realism to Sound Files

Continuing our series on realistic sound reproduction, this month we discuss impulse responses. Learn how to add realism to sound files with reverberation produced from the impulse response of real spaces (e.g., various size rooms, concert halls, cathedrals).

Photo 1: Altiverb is the industry standard convolution reverb plug-in for Mac OS X and Windows. It uses top quality samples of real spaces—ranging from Sydney Opera House to Amsterdam's Concertgebouw—to create reverb. (Photo courtesy of Audio Ease, www.audioease.com)

By
Ron Tipton
(United States)

The impulse response (IR) of a space is a recording of all the reflected sound that is picked up and recorded after a very short loud sound (the impulse) is produced in the space. To make this recording, a microphone is set up somewhere in the space. The microphone is typically placed in the center if it's a small room or in several different locations in a large room or concert hall to obtain different IRs (see **Photo 1**). The impulse, which can be an electric spark, gunshot, or balloon burst, is produced and the IR recorded and analyzed. The IR length can vary from less than a second to as much as 10 seconds for a very reverberant space. The IR file is usually edited to remove the impulse, leaving just the reflected sound.

Photo 2: Select a preset at the screen top to get started with FreeVerb. The Quality button is set at 10 so leave it there unless you are using an older computer with a slow CPU. The Gate section should be cautiously used, perhaps to add some sparkle to a cymbal clash or high-hat strike.

Reverberation is added to an input sound file by convolving it with the IR. Leaving all the math aside, convolution can be thought of as a filtering operation in which the IR is the filter. The original sound file is passed through the IR filter and comes out with the added reverberation. The amount added is set by the VST plug-in, the Mix control. I have included an article, "Creative Convolution," in the Supplementary Material if you want to delve deeper into this subject.

There are several IR file collections or libraries to download, many of them are free. The file name tells you something about the space, for example, small-room.wav, concert-hall-1.wav, and so forth. I have included some of the download sites in Resources.

There are also quite a few convolution-reverb plug-ins from which to choose. I have included four of the free ones here because they each have a somewhat different user "face" and you may find one that you like better than the others. The first two are "algorithmic" plug-ins, which means they use built-in simulated IRs, but they can produce realistic reverberation.

FreeVerb

This is an easy to install and simple to use VST plug-in (see **Photo 2**). Unpack the downloaded zip file and copy the single dll file to your VST plug-in folder. The four-page datasheet explains the control functions. This simple plug-in has multiple built-in presets that simulate IRs and can be altered with the control sliders.

I tried several presets with the "lady" file, in particular the stereo file attenuated by 4 dB, to avoid clipping in the output. I found the added reverb to sound best with a fairly small room and I made a clip, lady-S-FreeVerb.wav, using the Medium Room 2 preset which has a room size of 11, a relative number that ranges from 3 to about 200. There is a graphic on the screen that shows the relative room size with the sound source (a loudspeaker icon) and listener locations.

Then, I tried a Melody Road track with a reduced loudness percentage using the MultiDynamics plug-in (which I discussed in my May 2015 *audioXpress* article). I discovered a larger room was fine here so I made the clip, MDS-MD5-FreeVerb.wav, using preset Club 2, which has a 29 room size.

Ambience

Install by unzipping the download file and copying ambience.dll to your VST plug-in folder. This is another plug-in with many presets that simulate IRs (see **Photo 3**). The settings can be changed with the control knobs and it appears to save your changes when the plug-in is closed. I don't necessarily think this is good so you may want to keep an unaltered copy of ambience.dll in another folder to recopy it to the VST plug-in folder to restore the original presets. The two-page "Mini Tutorial" explains the function of the knobs and sliders. There is a .txt copy in the Supplementary Material.

I also used the Melody Road track with a reduced loudness percentage and attenuated 3 dB as the input to several presets. The most pleasing to me was Pady-VocalMellow with the Wet slider set to −4 dB from its original −7 dB. I saved this clip as MRS-MD5-Ambience.wav in the Supplementary Material. Changing the ratio between the Dry and Wet (processed) signals in the mix is the easiest way to alter the audible reverb. Adding some reverb has the effect of widening the stereo image without any other processing.

Both FreeVerb and Ambience have a Gate section, which is better left off for a natural sounding output. However, it could be fun to experiment with. Also, both plug-ins have an EQ section. The settings can be changed using the preset and it seems generally better to leave them that way because usually some bass cut is needed to keep the output from sounding muddy. But this depends on the amount of bass in the input file, so listen and make EQ changes as needed.

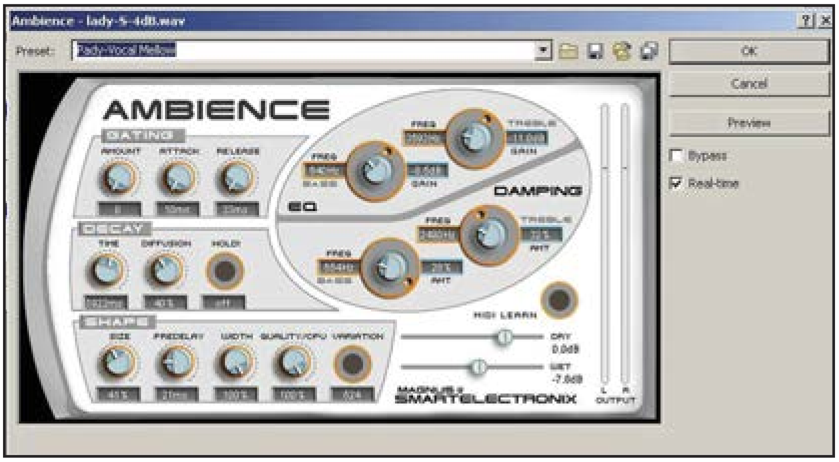

Photo 3: Loading a preset will set the Ambience knobs to get you started. If you change the settings, it seems they are saved when you close the plug-in, which could be inconvenient. There are lots of knobs but their functions are explained in the manual.

Reverberate LE

The "LE" version is free, but you are encouraged to make a donation if you find it useful. The "CORE" version is not free (about $50) but it will run in demo mode for 30 days. Both are installed by running their setup .exe files and can be added to your computer at the same time. The preset files are placed in a default folder but they can be moved because clicking the small up-arrow icon on the right side of the screen just under the row of knobs asks for the preset folder path. FLAC presets are supplied with the LE version and .wav with the CORE version but either preset type will load in either plug-in version. They

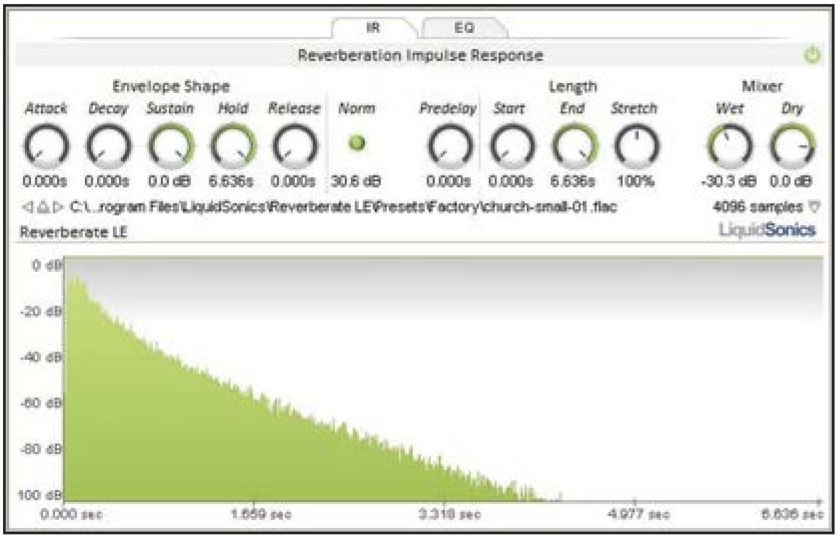

Photo 4: The loaded Impulse Response (IR) is displayed in the screen's lower half and its shape can be altered with the control knobs. The changes can't be saved as a new IR file in the LE version so make notes. This plug-in can be set to zero time delay for real-time processing.

Photo 5: The screen shows SIR because SIR-2 had not been created. Just load an IR file to get started. The graphic equalizer is nice if you need to use it. This plug-in has a fixed delay of 8960 samples so it may not be suitable for real-time processing. (The delay may be compensated by the VST host so consult its documentation.)

can also load presets from the various free and not-free preset libraries.

The LE version can produce a good output: listen to lady-Reverberate.wav in the Supplementary Material. After trying several presets, I used the Small Church preset with the Wet side of the mix increased about 4 dB for a bit more reverb (see **Photo 4**).

The CORE version can load a pair of preset files, left and right, for true stereo reverb. Most of its furnished presets are left-right pairs but whether you can hear an improvement over a mono file preset probably depends how much left, right difference there is in the input stereo file. I have included User Guides in the Supplementary Material.

The Dizzy Gillespie "ShhhPeacefull" clip has a large variation between the left and right sides and, I think, some original reverb. Anyway, I processed it with the demo CORE version using the stereo Cavernous Hall TS preset which has a decay time to −60 dB of about 3.5 seconds—fairly reverberant. The resulting clip is Shhh-Cavernous-Hall-TS.wav—it's rather lively.

SIR-1

SIR-1, shown in **Photo 5**, also uses loadable mono IR files so both stereo channels are processed the same. As I've mentioned, this usually produces very good enhancement. You can download a small IR library from the SIR website and you can also load any of the mono IR files from the libraries listed in Resources. Clicking on an IR file name will attempt to load the whole library with the file names appearing in the column on the screen's right side. The rectangle in the lower left corner is a graphic equalizer, which is normally a flat response. Clicking in the rectangle creates control points which can be moved around to set up an EQ curve. There is very little documentation, but the controls are similar to those in the Reverberate plug-in so just refer to its user guide.

I opened lady-stereo.wav and loaded the IR file L960large_ambient.wav. I wasn't quite satisfied with the resulting music so I changed the Mix to Dry at 0 dB and Wet at −1 dB. The enhanced clip is lady-sir.wav in the Supplementary Material, which also has the IR file.

SIR-2 has additional features including true stereo output using stereo IR files, but it costs $189. The download file will run as a full-featured demo but it puts a chime sound into the output every few seconds. I've included a copy of its user guide in the Supplementary Material.

About the Author

Ron Tipton has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at White Sands Missile Range. In 1957, he started Testronic Development Laboratory (now TDL Technology) to develop audio electronics. He is still the TDL president and principal designer.

Project Files

To download additional material and files, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html.

Resources

Audioease Altiverb Impulse Responses and Altiverb Sampled Acoustics Processor, www.audioease.com/IR and www.audioease.com/Pages/Altiverb/sampling.php.

Bricasti M7 Impulse Response Library v1.1, Simplicity, www.samplicity.com/bricasti-m7-impulse-responses.

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R. Tipton, "Signal Dynamic and Loudness," *audioxpress*, May 2015.

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Ambience
Magnus | magnus.smartelectronix.com/#Ambience

SIR-1 and SIR-2
SIR Audio Tools | siraudiotools.com

FreeVerb
Sinus | www.sinusweb.de/freetoo.html

Powersoft M-System

From Switched-Mode Amps to a Novel Transducer

Powersoft's M-System combines the M-Force motor transducer and the M-Drive switching-mode amp module, taking advantage of Differential Pressure Control (DPC) technology among several other company innovations. The project is the result of the company's continued search for new technologies and ideas from innovative concepts. In this article, we follow the effort—from a Eureka moment to a well-devised solution, which is now available to the OEM market.

By
João Martins
(Editor-in-Chief)

After 20 years convincing the market that Class-D could replace existing linear amplifiers designs, Powersoft has reached a milestone in loudspeaker engineering. But just how did an amplifier company decide to develop a transducer?

Founded in 1995 and headquartered in Florence, Italy, Powersoft designs and manufactures technologies and solutions for the professional audio industry. Mainly recognized for its lightweight, high power, energy efficient, switched-mode technology Class-D amplifiers, Powersoft is also recognized as a "rich" company in terms of developments, and not just in the field of pure amplification technologies.

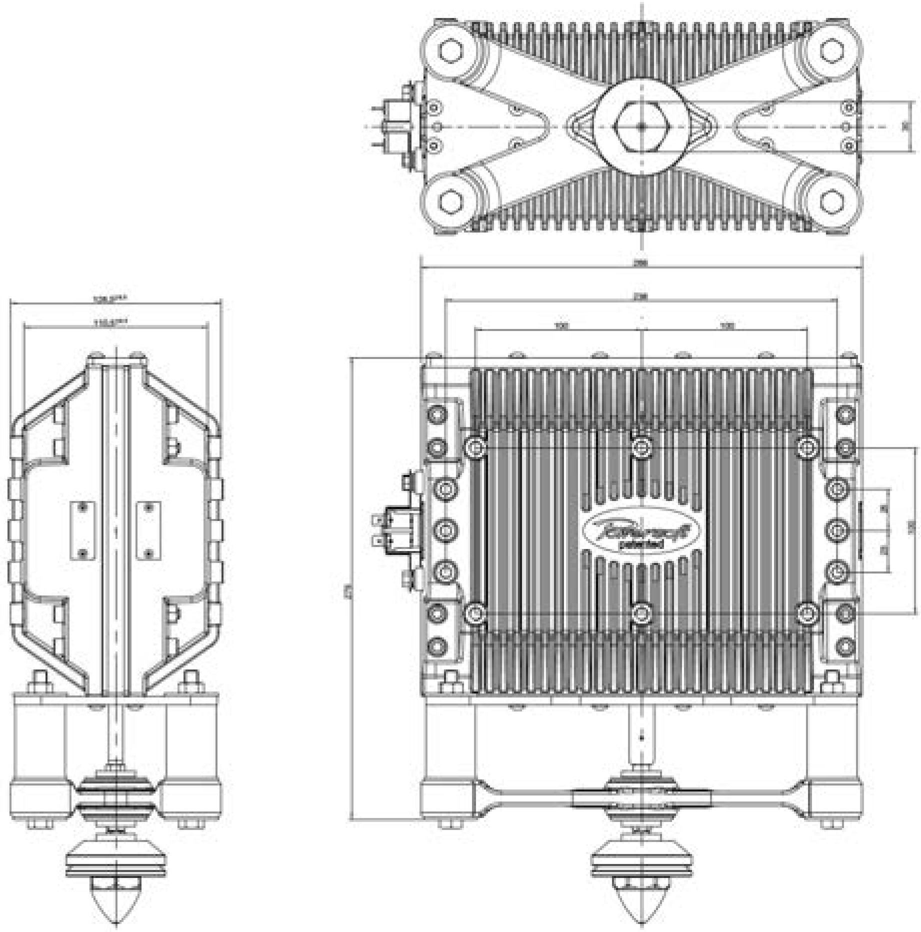
As Matteo Bianchini, Powersoft's OEM Account Manager states, "We're well aware that amplification is just a link in the audio chain, and any technical progress cannot bring a real major benefit if it doesn't involve, in some way, other elements of such a complex system."

That statement, might help us understand how Powersoft got involved in such innovative projects as the DEVA multifunctional device, which is basically a speaker equipped with several sensors (i.e., a microphone, a presence detector, a twilight switch, and a accelerometer), a video camera, LED lights, and everything needed to combine public address with audio messaging and video capturing. This device would be, in itself, a remarkable concept, combining an original design with a complete rethinking of the applications, which the DEVA addresses. However, Powersoft also refined the electronic design to minimize power consumption, enabling uninterrupted use, powered by an internal rechargeable battery that can be quickly recharged using a latest-generation solar panel.

We highly recommend a look at the DEVA. It has a multiple enclosure finish that emphasizes its exquisite raindrop-shaped design that is specifically intended to protect the internal circuitry from exposure to the atmospheric elements, making it an ideal solution for outdoor applications (IP 65). The DEVA system even features a GSM/UMTS communications module, GPS, dual band Wi-Fi, Bluetooth, and wired Ethernet connectivity.

Inside, we find Powersoft's renowned Class-D amplifier modules, making the DEVA a unique solution for background music and paging applications, combined with video and/or audio surveillance.

This project is a fine indicator of Powersoft's engineering department's capabilities and willingness to bring to market. It also serves as the ideal introduction to the topic of this R&D article. But first, we need to examine some of Powersoft's fundamental technologies.



The M-Force transducer features "push-pull" technology, an extremely powerful motor, and unique displacement capabilities.

Rethinking the Audio Chain

Powersoft was one of the first companies to successfully introduce pulse width modulation (PWM) technology in a way that enabled its products to achieve a previously unthinkable level of efficiency and power, with power factor correction (PFC) technology integrated into switching mode power supplies.

Synonymous with Powersoft's beginnings in 1995, this technology was gradually adopted in professional audio applications worldwide due to its extremely high efficiency, which enables it to transform all the energy drawn from the mains into usable power and recycle the reactive energy coming back from the loudspeakers (back EMF energy recovery, which typically can be damaging for traditional linear amplifiers). The combination of those technologies produces equipment with useful features for touring and fixed installation equipment, allowing for a drastic reduction of the required energy for the same output power, less energy wasted on cooling systems, and certainly fewer amplifiers in less space.

Powersoft has previously used PFC to improve performance and contain mains current draw and consumption, enabling 40% energy savings and important efficiencies including the size of the electric generators used in shows and fewer cable sections due to lower RMS currents, which in turn allows for less hum and induced distortion. PFC-based audio amplifiers also seem to be immune to mains voltage fluctuations and load impedance since the power supply automatically regulates itself, allowing amplifiers to be used worldwide, whatever the mains voltage.

Powersoft also leveraged the benefits of Smart Rails Management (SRM) technology. In any amplifier output stage, the efficiency is a function of the difference in the output voltage delivered to the load with the rails voltage delivered from the power supply. This is true for any amplifier topology but Powersoft's SRM implements real-time voltage tracking in the power supply, minimizing the differences between the output voltage rails voltage to improve overall efficiency. The SRM system feeds back the output signal to the power supply and modulates the rails voltage to reduce heat dissipation and improve efficiency, which includes reducing switching losses and EMC problems. It also provides a noise floor reduction on the output stage.

But the PWM technology, the PFC, and the SRM were only the beginning for Powersoft. The Italian company gradually gained recognition because of its lightweight, high power, Class-D amplifiers with reliable "green" credentials, and the company's ability to continuously innovate new products.

Year after year, Powersoft introduced smaller, lighter amplifier modules and racks, gradually introducing advanced DSP modules, network connectivity, and management software. Signal

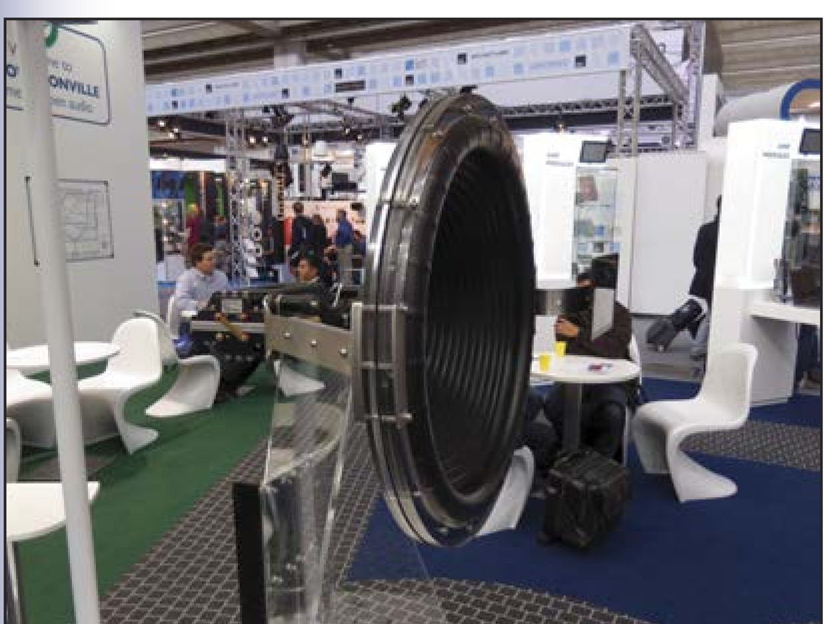
Powersoft's DEVA automated audio messaging and video device is a remarkable engineering feat, demonstrating the Italian company has more to offer than amplification.

processing in the digital domain enabled Powersoft to modify the audio response to optimize its own products' audio quality and efficiency. Powersoft also designed more complex systems and increased its products' market appeal by providing integrated processing tools in response to the needs of speaker manufacturers and audio engineers.

Powersoft was also one of the first amplifier manufacturers to experiment with networked control and software tools. First, it introduced KAESOP, an optional board for its popular K Series and Installation amplifiers for remote control and audio networking. The company also experimented with combining Ethernet protocol for remote control functions with a recognized digital audio protocol (e.g., AES3) to use the same CAT5 cable remote control and two channels of AES3 audio, which can be converted to four analog channels. Even before other audio networking protocols became popular (e.g., Audinate's Dante, which Powersoft currently implements), Powersoft was already pioneering software for remote control and monitoring for all its products.

Powersoft's Armonía is a fine example of total system integration, offering full online or offline system setup and tuning, real-time management and monitoring of all vital functions from a remote computer (and now mobile devices) via a single intuitive graphical user interface (GUI). Above all, Powersoft's software offered another way to significantly improve sonic performance and system reliability, allowing for innovations such as Active Damping Control to compensate for speaker wire losses and improve cone control with a virtually negative output impedance, and efficiently limit the amplifier's output power at safe levels depending on the varying load impedance over frequency.

So, it's not surprising that Powersoft invests more than 10% of its turnover in research and development, encouraging experimentation. The company believed in active speakers and it pioneered its technologies in amplifier modules that could be integrated into the speakers. All those things were the necessary prelude to the company's M-System project, which started in late 2008.



Powersoft displays the M-Force concept as an innovative and unique transducer based on a patented moving magnet linear motor structure at Prolight+Sound 2013 in Frankfurt, Germany.

Revolutionary New Concept

Next, Powersoft started working on a new amplifier module, the M-Drive (originally called the SUB-Drive) and the patented differential pressure control (DPC) concept, allowing for close integration between the amplifier and the speaker. The concept was originally named IPAL, which stands for Integrated Powered Adaptive Loudspeaker, and consists of "a combination of a high-power, high performance Switching Mode Amplifier in conjunction with an embedded DSP that performs the double operation of both managing the loudspeaker system processing and taking care of the Differential Pressure Feedback Loop Control implemented on it."

Powersoft introduced this concept in 2011, at the 131st Audio Engineering Society (AES) convention in New York, presenting a paper titled "Practical

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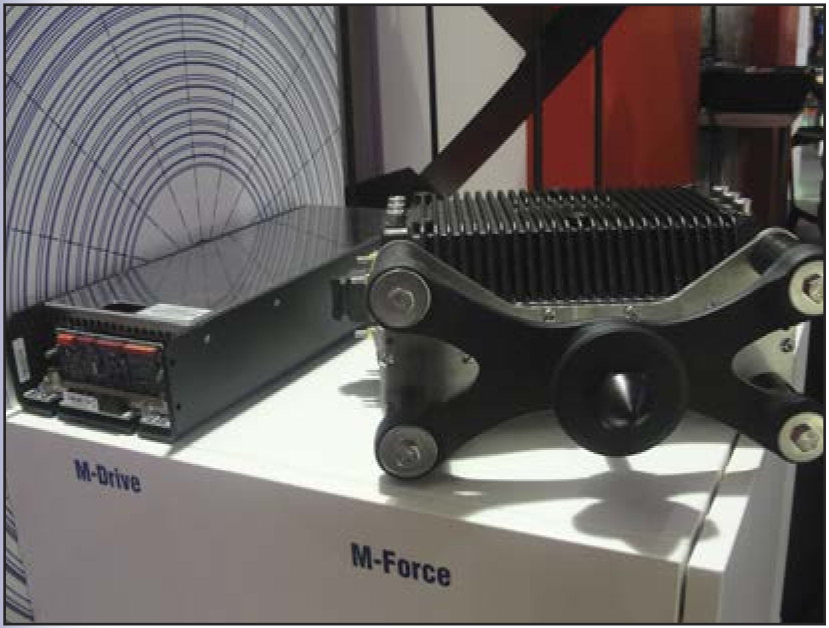
As co-founder and R&D Development Director of Powersoft, Claudio Lastrucci's fields of expertise include electric power, audio amplification, signal processing, acoustic design, and other areas. (Photo courtesy of Powersoft)



Powersoft's M-System combines the M-Force motor transducer and M-Drive switching-mode amp module, taking advantage of Differential Pressure Control (DPC) technology to establish a global feedback between the electrical and the acoustic domains. The Zero Latency DSP performs real-time processing of the differential pressure control signal to adaptively correct the diaphragm displacement.

Applications of a Closed Feedback Loop Transducer System Equipped with Differential Pressure Control." The paper was co-authored by Fabio Blasizzo (then a freelance transducer engineer who conducted practical measurements in the early stages of the project), Mario Di Cola, Paolo Desii, and Claudio Lastrucci. Desii is one of Powersoft's R&D engineers, responsible for embedded systems development and software. Lastrucci is Powersoft's co-founder, President, and Research & Development Director—he is also the driving force of the company's technologies. According to Lastrucci, Di Cola was involved in the evaluation of the first prototypes "as soon as there was something to test," which means about three years after the project started.

Basically, the paper described a closed feedback loop transducer system dedicated to very low frequency reproduction. As stated, "The use of a feedback control loop can be very helpful to overcome some of the well known transducer limitations and to improve some of the acoustical performances of most subwoofer systems. The feedback control of this system is based on a differential pressure control sensor. The entire system control is performed by a 'Zero Latency DSP' application, specifically designed for this purpose in order to be able to process the system with real-time performances."

The M-Drive, used in conjunction with DPC was the technology at the core of IPAL systems, with a specifically designed transducer to implement the novel approach of a closed feedback loop for direct control over the system's acoustical output.

The first amp module—equipped with DPC and a PFC-equipped power supply—delivered up to 8,500 W and was tested by several speaker manufacturers. Italian company B&C Speakers was among the first companies to introduce two IPAL high-efficiency transducers. B&C Speakers developed two different IPAL compatible speakers, the 21IPAL (21") and the 18IPAL (18"). The 21IPAL was intended for use with single speaker applications and the 18IPAL was recommended for 2x18" configurations. Eighteen Sound, another Italian speaker manufacturer, also developed its own design with IPAL-compatible products.

The original AES paper introduced the concept of Virtual Speaker Modeling, again leveraging the company's expertise in software and DSP control to take advantage of the DPC concept and create a so called "Virtual transducer." This powerful software tool makes the real transducer behave as a "user defined" transducer, synthesized by the speaker designer with a dashboard to manage the desired driver's Thiele–Small (T-S) or electromechanical parameters, based on a mathematical model.

Previous attempts at motor-driven speakers had some elements of Powersoft's concept, but lacked much of the refinement. Some readers may remember Tom Danley's motor driven speakers and their distance cousin, the Phoenix Gold Cyclone. Moving magnet shakers include the Bodysonic, the Clark, the Earthquake, and others.

From Concept to M-Force

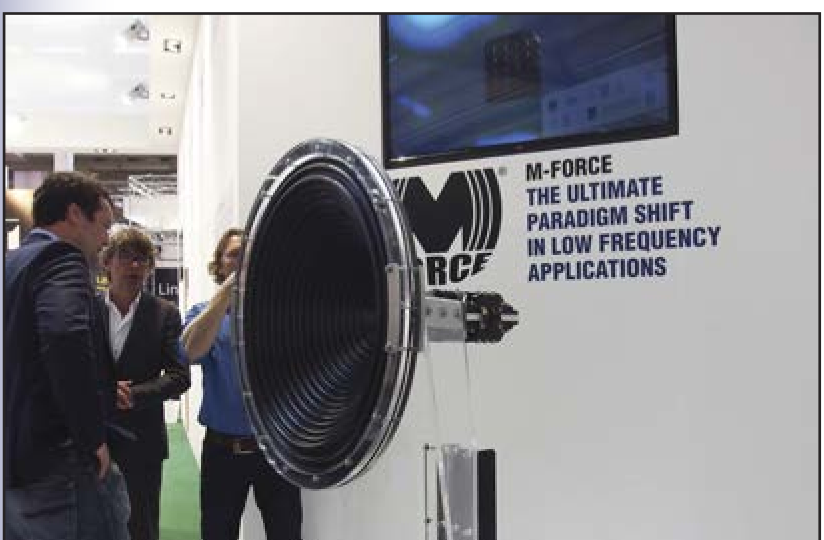
We specifically asked Lastrucci about the "genesis" for the whole M-Force project. And, Lastrucci admitted this was a "eureka moment." "The moment we imagined something not axisymmetric but 'planar' in the magnetic design. However, the move for a different transducer technology came from the knowledge of switch-mode amplifiers, their native properties and the desire to fully exploit the real benefits of the native energy recycling capabilities. No switch-mode amplifier, no M-Force..."

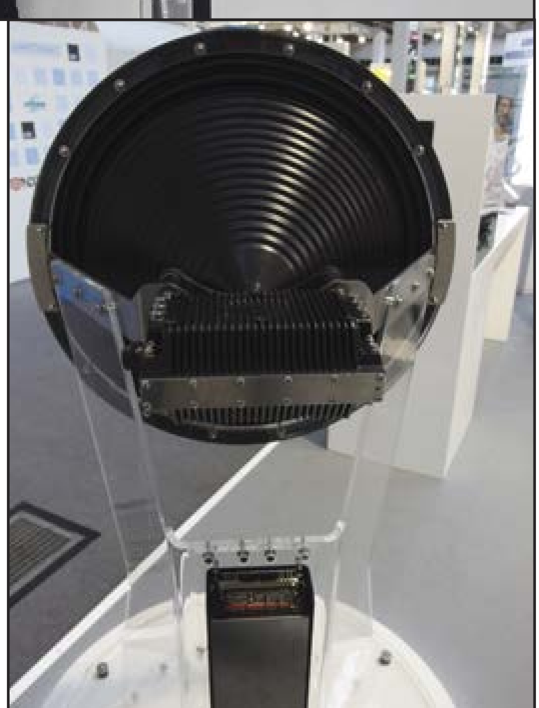
As previously described, the M-Force evolved from the earlier efforts to optimize a transducer for subwoofers. As Matteo Bianchini's stated to

"We don't like to use too big words, but M-System really 'risks' to be one of the few big things happened in audio transduction in the past decades," says Matteo Bianchini, Powersoft's OEM Account Manager.

audioXpress, "In our R&D efforts, we always tried to take into account the interaction of our technology with "adjacent domains," especially the world of transducers and the actual effects of this interaction in the acoustical domain. As a consequence, some of the technologies are more complex than the simple amp, and involve many perspectives, from processing to transduction, as well as acoustic design and, of course, high power electronics."

He adds, "The first things you might notice are the very high output figures (310 V_{PEAK}, 200 A_{PEAK}), making it able to drive lower-than-1-Ω loads, but there's much more than that: The M-Drive, despite compatible with applications using traditional transducers, was designed in order to fully drive the M-Force, an innovative low frequency transducer (a linear motor based on a moving magnet approach) that no other amplifier would be able to push to the limits."





The complete M-System was presented at the 2014 Prolight+Sound show, when several manufacturers were already working on practical implementations.

After showing it at multiple shows and providing samples to some exclusive partners, Powersoft decided to make it available for OEM projects.

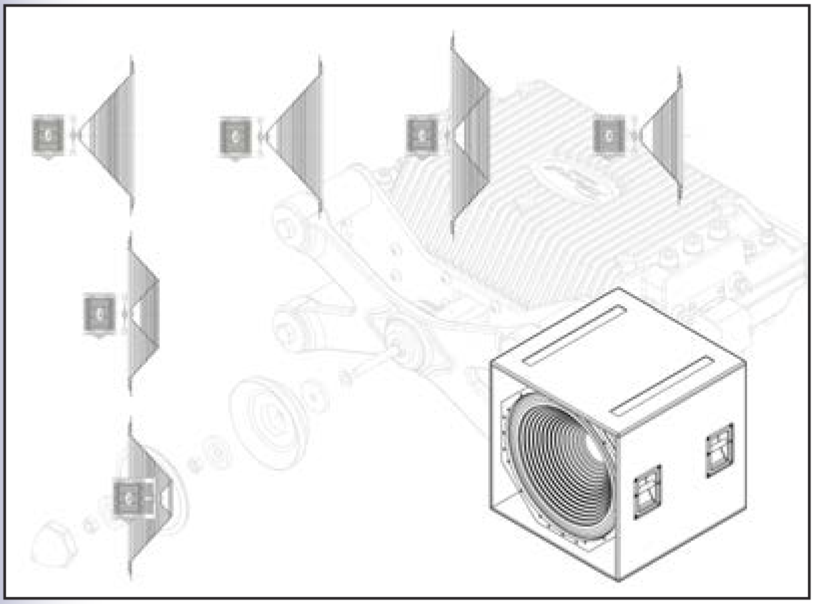
The M-Force was the second stage of Powersoft's R&D efforts, following the M-Drive amp module and the IPAL concept. In 2014, Powersoft presented another paper at the 136th AES convention in Berlin, Germany, explaining how the company designed "A Novel Moving Magnet Linear Motor." The paper, presented by Lastrucci describes how the company explored a new "electrical to acoustic conversion approach" combining "new technologies in the electronic amplification domain and latest magnetic materials." The new transducer is presented as a "new electrodynamic device that considerably improves electrical to acoustical conversion efficiency, sound quality, robustness and power handling," based on a "fully balanced and symmetrical moving magnet motor design, along with anisotropic magnetic compound integration," to deliver substantial performance improvements in terms of "acceleration, linearity and efficiency, providing additional degrees of freedom in high quality professional speaker design," different from the "ubiquitous moving coil approach."

The topology is closer to a bass shaker than a conventional subwoofer. In crude terms, the M-Force is a bass shaker coupled to a spider, a cone, and a surround. However, the response is far more linear than a shaker with some benefit from the servo-control feedback circuitry.

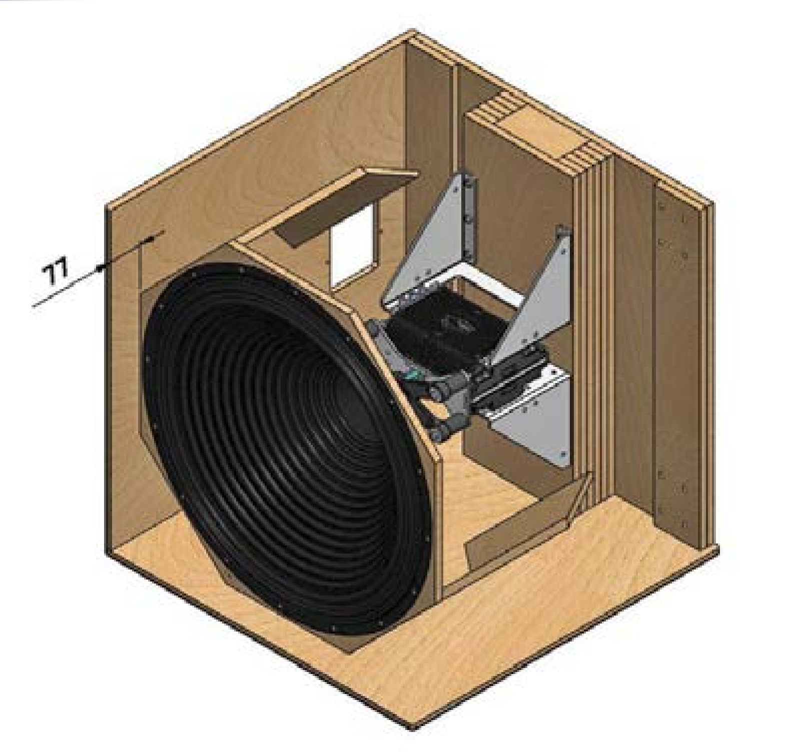
The new transducer leverages a specific radiating element developed to cope with the high force and displacement provided by the motor "using large-size vacuum-formed conical polymeric diaphragms that embeds in a single material piece, the piston, the connecting elements, and the outer surround as the load for the motor and the coupling device to the acoustical domain."

This paper describes the M-Force system introduced at the 2013 Prolight+Sound and InfoComm shows. It features an impressive 40" diaphragm, as well as 22" and 30" variations. The 40" unit reportedly is able to generate a maximum radiating surface up to 6,500 cm^2, or the equivalent radiating surface of four 21" drivers.

Powered by the powerful new "pistonic" motor, the system effectively opened the way for powerful low-frequency cabinets with "a much smaller form factor for a given SPL performance and low-end extension."



Powersoft provides a range of two motors and six M-System diaphragms models for very low-end subwoofer (40″) and low-frequency applications (30″ and 22″).



This Powersoft subwoofer reference design features M-Force.

Powersoft also noted other possible fields of application in materials stress testing, vibrational active damping control, mechanical to electrical conversion, and even mechanical energy harvesting.

The M-Force Drive

According to the paper presented at AES Berlin, the "motor structure" uses forces generated by current and magnetic field interaction, the same as on a conventional moving coil actuator. "However the stationary reference has been moved from the magnet and yoke assembly to the exciting coils, which become the "heavy" part of the system," Lastrucci details.

Powersoft has written three complementary papers, which can be accessed from the AES website (www.aes.org). As for the new system, it is "based on two parallel bars of NdFeB magnets, facing to a common plane but with opposed magnetic field orientation. Two coils are placed facing the bars of magnet in a way to create a sandwich structure that holds the magnet within the coils."

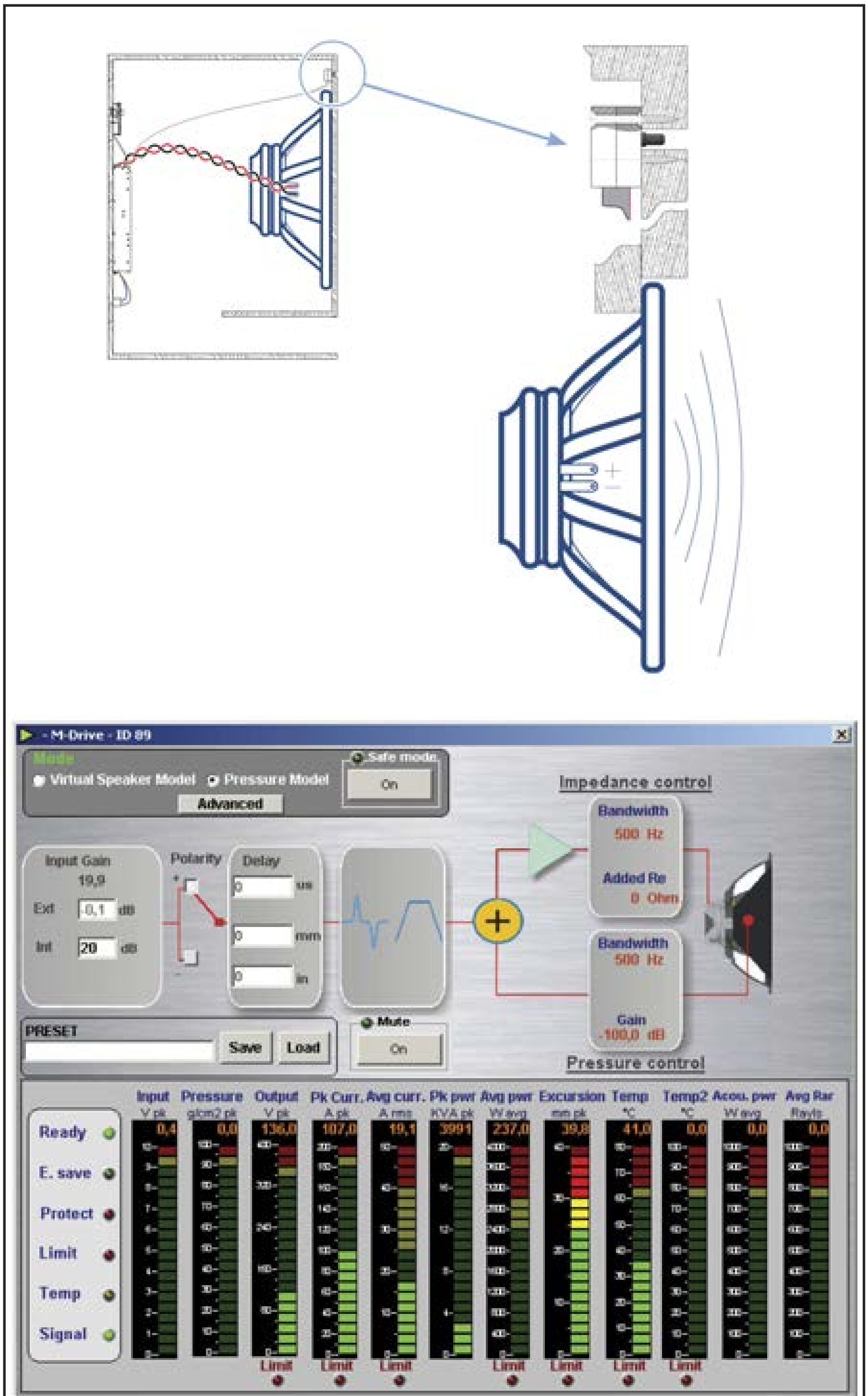
"The coils are wound using a ribbon of solid conductor, forming a winding of rectangular shape. The magnetic field generated by the magnets is forced to cross the conductor of the coils, producing, once the coils are subjected to current, a relative force between the coils and the magnet bars. This force linearly depends on the intensity of the field generated by the magnets and the current flowing into the coils."

The paper also describes how the symmetrical design is surrounded by an outer ferromagnetic shell "to allow easy circulation of the steady field generated by the magnet bars and to create a defined path for the variable flux produced by the current flowing in the coils," and "a frame of composite material" to "provide a mechanical connection to the radiating part of the complete acoustical transducer."

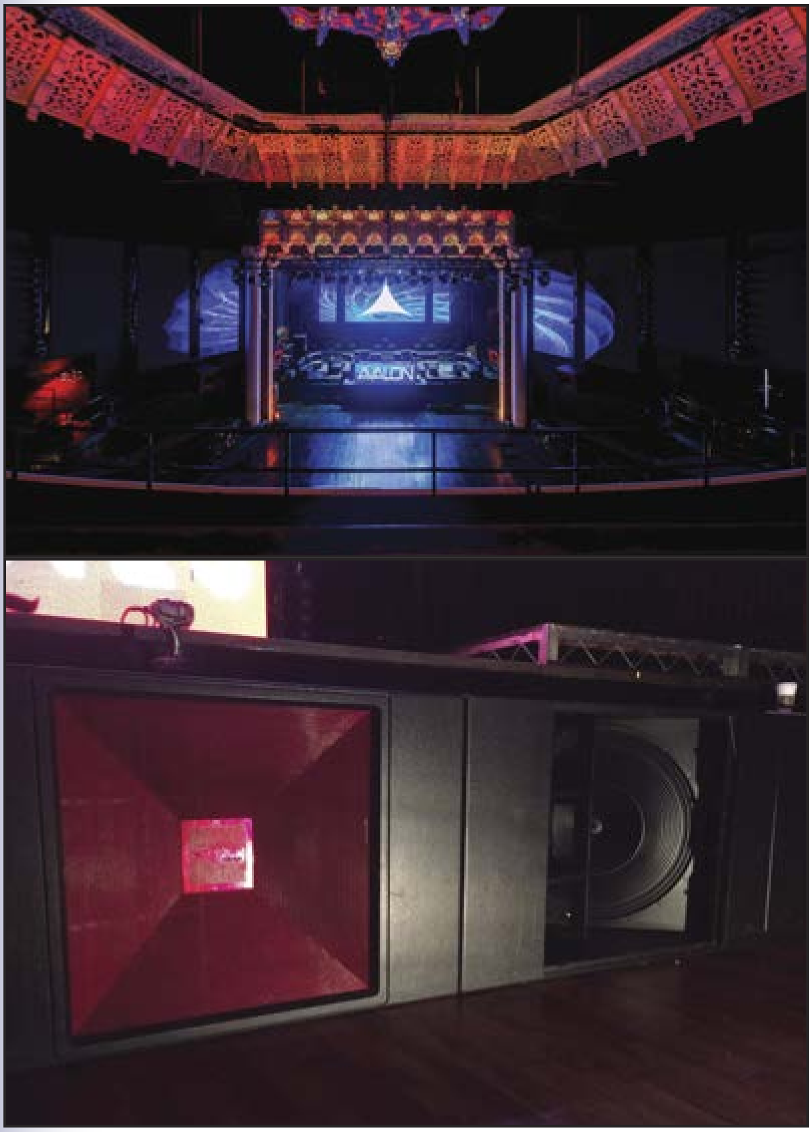
Lastrucci says "this can be considered a real push-pull device, where both magnet bars and coils work completely symmetrical in respect to the axial displacement and each portion of the conductors provide either push or pull action on each magnet bar in a complementary fashion to the other specular portion of the motor," while the most evident feature is the absence of conductors in the moving portion of the motor.

"The forces are provided by the interaction of the field that is generated by the steady coils and the field generated by the magnets that are not energized by any connection," allowing for a "free and very reliable operation even with extreme acceleration and displacement" and thermal stability.

To optimize such a system, and specifically its peak force to mass ratio, over a linear displacement of 30-mm peak to peak, Powersoft applied all the technology from its M-Drive amplifier module with on-board DSP to achieve actively synthesized values from –10 Ω to +10 Ω resistance and –2 mH to +2 mH induction. To create global



The differential pressure control (DPC) measures the difference in pressure between the front and the rear sides of the radiating diaphragm and uses this information to alter the transducer's behavior, according to the actual boundary conditions.



The Avalon Hollywood club now has a new improved audio system, featuring "the world's largest subwoofer club installation ever" with new subwoofers powered by Powersoft's new M-Force technology and 40" cones.

feedback in the system, Powersoft used a differential pressure sensor to detect the overall pressure acting on the radiating surface of the radiating diaphragm. The DPC measures the difference in pressure between the front and the rear sides of the radiating diaphragm and uses this information to alter the transducer's behavior, according to the actual boundary conditions. Via software, it's even possible to weigh and customize the feedback between the electrical and the acoustic domains.

"This method allows the definition of a predictable behavior in the electrical–mechanical–acoustical signal chain, and allows reduction in the sensitivity of the system performance against aging and boundary condition," says Lastrucci.

All the main parameters of the transducers are supervised. The voltage, the current, the power, the pressure, the displacement, and the forces are maintained within the safety conditions and limited within a global amplifier and transducer combination. Power management uses a combination of energy recycling output stage and PFC integrated into the power supply.

The M-System in Practice

A new paper, presented in 2014 at the 137th AES Convention in Los Angeles, CA, titled "Subwoofer Design with Moving Magnet Linear Motor" and co-authored by Di Cola (Audio Labs Systems), Lastrucci, and Lorenzo Lombardi (Powersoft), takes the new electro-dynamic transducer described in Lastrucci's previous paper and details specific proposals in subwoofer design using this technology. Among the designs, we find a high output, high Q vented box design tuned at 30 Hz, a small-size compact vented design tuned at 34 Hz and a hybrid short transmission line. The paper details actual measurements from those units and provides advice on system optimization using the available tools.

Powersoft also released a series of documents that includes "Loudspeaker Project Examples" for subwoofer designs based on M-Force. The company has also been expanding the information it provides to manufacturers, including information for two M-Force models, one for extremely low frequency applications with $(Bl)^2/Re$ = 2215 $(T{\cdot}m)^2/\Omega$ and maximum acceleration of 3800 m/s^2 ; and another with $(Bl)^2/Re$ = 3000 $(T{\cdot}m)^2/\Omega$ and maximum acceleration of 4800 m/s^2. Details of different diaphragm designs are also provided in Powersoft's literature.

Since its launch, AV specialists and integrators including ATK Audiotek, Maryland Sound International (MSI), and others have used Powersoft's M-Force and M-Drive solutions in a variety of high-profile applications.

Meanwhile, the first commercial implementation examples have also become public. One of those involves the world's largest subwoofer club installation, in a collaborative project with Eastern Acoustic Works (EAW) at the Avalon club, a Hollywood, CA, historic venue.

John Lyons, Avalon Hollywood's owner and audio expert is known for the design and the installation of audio systems at some of the world's leading nightclubs, restaurants, and lounges across the globe, including the Avalon clubs in Hollywood, Singapore, San Diego, CA, and Boston, MA. Lyons and EAW's co-founder Ken Berger worked together on the original Avalon loudspeaker system's design launched in 1997.

The system was redesigned from the ground up as the "Avalon by EAW" Sound System, and re-launched in 2011. Avalon Hollywood's Avalon-series system has now

Spanish manufacturer D.A.S. Audio introduced its new Sound Force Series line-up of dance systems at the 2015 Prolight+Sound show in Frankfurt, Germany. The series includes the SF-30A, a M-Force powered ultra-low system, using 30" diaphragms, that provide 7,500 W of continuous power.

been improved with the help of Powersoft's latest technology, implemented in a new range of Avalon subwoofers.

The innovative setup was unveiled at Avalon's Grand Reveal 2014, replacing the original Avalon EAW system, which was recognized as a world reference and can still be heard in hundreds of nightclubs around the globe. Each speaker cabinet now includes a 40″ Powersoft diaphragm and M-Force, resulting in a cleaner, more powerful sound than was previously possible to produce.

Each subwoofer cabinet (there are six in total in the reference installation) is powered by two Powersoft M-Drive modules that are used to supply power to the M-Force motor, the cone, and each of the traditional speakers, generating 15,000 W at 2 Ω.

Another example about to reach the market is the Sound Force Series of club systems by Spanish manufacturer D.A.S. Audio.

The new Sound Force Series of dance systems is aimed at top-level clubs and discos. The system is comprised or four speaker models for mid-high, mid bass, and two subwoofer options, including the impressive SF-30A, a M-Force powered ultra-low system, using 30″ diaphragms, offering 7,500 W of continuous power.

Other M-Force powered systems are currently in development from other manufacturers but no details have become public at the time of writing.

Open Applications

Powersoft has been extensively promoting the M-Force concept in different forums. Lastrucci introduced the technology also at the Association of Loudspeaker Manufacturing & Acoustics (ALMA) International's Winter Symposium 2015, in Las Vegas, NV, convincingly demonstrating the efficiency of the new moving magnet linear motor technology in subwoofers and detailing working methodologies, cabinet design, and loudspeaker application scenarios.

According to Lastrucci, the M-System is currently an exciting new platform for OEM applications. "As soon as the results on the initial prototypes provided some effective performances, we started to pursue a process of optimization and engineering that leaded to a finished product. Since our goal, as Powersoft, is not to be a speaker manufacturer but a supplier of components for their products, this was naturally in line with the company's vision."

Another surprising aspect was how the concept was quickly converted into actual products. According to Lastrucci, "We do like to have and maintain a tight relationship with our existing partners and users, always listening to their needs and looking for their feedback on our "crazy" ideas: speaker manufacturers, rentals and FOH all have their own unique perspective on a system and it is important to verify that a new technology brings some real benefits to all of them. With M-Force it was exciting to see the reaction when we started to talk about this new technology, and above all, after getting them listening to it and playing with the first samples. We know well that new technologies take a very long time to be accepted in our industry (it happened so many times to us...) but it sounds like this time the word is spreading faster... Maybe the industry is beginning to trust we can make crazy ideas real."

Lastrucci adds, "The technology is intended also for other applications, not necessarily to produce sound... Likely, otherwise, to remove it... This is a field of application [active noise canceling] we are investigating for a considerable period of time and foreseeing interesting possibilities."

Apart from high power and energy efficient subwoofers, Powersoft says its M-System is currently being tested for diverse applications such as steerable low-frequency arrays, low-frequency noise, and standing waves active removal systems and dipole low-frequency acoustic generation. "We are already working on other fields of application for the technology, there is a kind of market that is demanding large forces and medium stroke, typical of the M-Force technology," concludes Lastrucci.

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M. Klasco and S. Tatarunis, "Bass Shakers (Part 1): Enhancing the Deep Bass Experience with Tactile Energy," *audioXpress*, April 2014.

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———, "DEVA: The revolutionary unit for automated audio messaging and video," www.powersoft-audio.com/en/multimedia/deva.

———, "Loudspeaker Project Example n° 1: A Simple Compact Subwoofer Concept for M-Force®, powersoft_MForce_kit_en_v1.0 PDF document.

"Powersoft's New M-Force Technology Powers World's Largest Subwoofer Club Installation," audioXpress, February 2015, http://audioxpress.com/article/Powersoft-s-New-M-Force-Technology-Powers-World-s-Largest-Subwoofer-Club-Installation.html.

Fresh fom the Bench

Benchmark SMS1 Loudspeaker

Setting the Benchmark Reference

Photo 1: An SMS1 loudspeaker is set up for testing in Benchmark's R&D room. Benchmark measures frequency response in both the near-field and at a distance of 6′ to produce an anechoic combined response. Its calibration microphone was set up at the 6′ distance, and an additional boom stand can be seen for the near-field measurement. (The large JBL box served only to support the SMS1. It was not connected.)

Benchmark Media Systems has been known for its excellent digital conversion products, microphone preamps, and the remarkable AHB2 stereo power amplifier. Last year, the company expanded its product line to include a loudspeaker, which it has designated the SMS1. The new loudspeaker has been in short supply, selling faster than Benchmark can make them, so getting a loaner pair for review was not easy. The solution was to visit Benchmark.

By
Gary Galo
(United States)

Benchmark's headquarters is less than three hours by car from Potsdam, NY, so I made a day trip to check out the new loudspeakers. Benchmark occupies the entire second floor of a commercial building on East Hampton Place in Syracuse, NY, and I was impressed by the bright, spacious facility. Vice President and Director of Engineering John Siau was a gracious host and gave me a tour before we got down to the business of auditioning Benchmark's new speaker.

Benchmark designs and manufactures all its products at the Syracuse headquarters, and a high level of technical sophistication was evident from the start of my tour. We began in the R&D room, where an SMS1 loudspeaker was set up for testing (see **Photo 1**). Benchmark measures frequency response in both the near-field and at a distance of 6′ to produce an anechoic combined response.

Next, we next moved to the assembly room (see **Photo 2**), where one of the technicians was building a DAC2 D/A converter (highlighted recently on the audioXpress website). The DAC2 converters are built around six-layer circuit boards designed in house, but manufactured overseas (not in China, however). Benchmark contracts with two facilities based in New York state for the actual stuffing of parts on

SMS1 Loudspeaker
Benchmark Media Systems, Inc.
203 East Hampton Place, Suite 2
Syracuse, NY 13206
800-262-4675
sales@benchmarkmedia.com
www.benchmarkmedia.com
List Price: $2,450/pair with black side panels; or $2,650/pair with Honduran Mahogany or African Padauk side panels
Optional Stands: $240/pair

Photo 2: Benchmark Media Systems' assembly room houses several workbenches and an array of sophisticated test equipment, including two Audio Precision SYS-2722 audio analyzers. Most products are assembled and tested to order.

the PC board which, in itself, is a sophisticated operation and can be fraught with pitfalls if the equipment manufacturer doesn't exercise careful control over the process.

In recent years, the trend in audio manufacturing is the use of surface-mount, "chip"-type passive components, and Benchmark is no exception. With through-hole parts, it's easy to identify the types and the OEM manufacturers of the resistors, capacitors, and other passive components found in electronic equipment. With surface-mount parts, it becomes nearly impossible. Metal- and carbon-film resistors look essentially the same, and forget about trying to identify the manufacturer.

Benchmark's solution to ensuring the exact parts it specifies are actually used in the finished products is to purchase all the parts themselves. Many equipment manufacturers leave this to the plant that stuffs the PC boards. When the finished boards arrive for final product assembly, they can only hope that the parts they specified were the ones actually used. However, all parts used in Benchmark's PC boards are directly shipped to them and inspected. Benchmark then ships the parts to the PC board assembly plant. This level of quality control ensures excellent unit-to-unit consistency.

Most products are assembled and tested as orders are placed, and orders made by 3:00 PM are generally shipped the next day.

Photo 3: The SMS1, shown in a front view, is an acoustic suspension design using a proprietary 170-mm long excursion woofer and a 25-mm soft-dome tweeter. (Photo courtesy of Benchmark Media Systems, Inc.)

SMS1 Details

The array of small studio monitor loudspeakers available today shows a decided preference for vented designs, which many designers and users favor because of their high sensitivity. Powered loudspeakers are also popular, eliminating the need for a separate power amp. Some powered loudspeakers include a digital input and built-in D/A converter, and some incorporate active crossovers with multiple power amps. I admit a preference for acoustic suspension designs over vented types because they roll off at 12 dB/octave below cutoff rather than 24, and because vented systems are unloaded below cutoff.

The latter was a more serious problem when playing LP records, but if the user's primary sources are digital it's not as much of a concern. Although there are strong arguments to be made for tight integration of a power amplifier and a loudspeaker, I prefer the flexibility of being able to use an amplifier of my own choice, one not compromised by the necessity of having to make it fit in the loudspeaker enclosure. I also don't like the idea of having your entire loudspeaker out of commission if a power amp fails.

Benchmark has gone decidedly against the industry grain with the SMS1 (see **Photo 3** and **Photo 4**). First, it's a passive loudspeaker, requiring an external power amp. Second, it has an acoustic suspension design, requiring more amplifier power than a vented loudspeaker. Benchmark's own AHB2 power amp is an ideal match for its new loudspeaker, and will deliver 160 W into SMS1's 6 Ω load. The AHB2 has plenty of power to drive this speaker to high levels with very low distortion. Although the AHB2 is extremely compact and efficient, it wouldn't be possible to achieve this level of performance if even one channel of this amp had to fit inside the SMS1 enclosure. Benchmark does not recommend low-power tube amplifiers with the SMS1. If a tube amp is used, it recommends 30 W RMS or more.

Regarding vented loudspeakers, Benchmark notes the following on its website: "We know that small ported loudspeakers have become fashionable.

We don’t do fashion. Ported speakers create the illusion of bass extension by emphasizing certain frequencies (hence the name “tuned port”). In smaller speaker enclosures, ports often produce a nonlinear bass response—sometimes described as “one note bass.” Ports can make a small speaker seem large, but accuracy can suffer in the name of big bass.”

Some loudspeaker designers will probably take issue with this. Since the work of A. Neville Thiele and Richard H. Small some 45 years ago, the boom-box or “one-note bass” character of vented loudspeakers has largely been a thing of the past. I prefer to use the term “vented” to describe modern loudspeakers designed in the post Thiele/Small era, in contrast to “ported” or “bass-reflex” terminology used to describe older, less accurate designs. Vented designs typically have a Q of 1.0, and although an acoustic suspension loudspeaker can be designed with a Q of anywhere from 0.7 to 1.5, the most accurate ones are usually on the low end of that range (for a thorough discussion, see Vance Dickason’s *Loudspeaker Design Cookbook).*

A vented design can deliver bass response that is superficially more impressive than that of a lower-Q acoustic suspension loudspeaker. I normally find the bass performance of acoustic suspension systems to be more natural than vented designs, particularly on classical music, which is my main area of interest.

Photo 4: The rear view of the SMS1 loudspeaker shows that each unit has a four-pole Neutrik NL4 SpeakOn connector and a pair of high-quality, gold-plated binding posts. Passive bi-amping is possible via the SpeakOn connector. (Photo courtesy of Benchmark Media Systems, Inc.)

High-Performance Components

The drivers in Benchmark’s two-way system include a proprietary 6.7” (170 mm) woofer and a

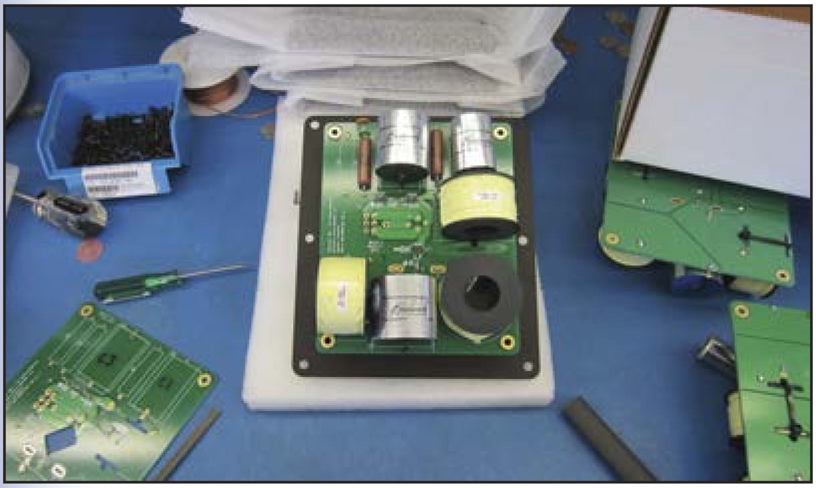

Photo 5: Each crossover board for the SMS1 loudspeaker is hand assembled, using premium-quality parts. ClarityCap custom manufactures capacitors to the exact values Benchmark needs. Custom-wound Renco air-core inductors are oriented to avoid magnetic interactions. Close tolerances are maintained in the crossover components to ensure unit-to-unit consistency.

1″ (25 mm) soft-dome tweeter. The woofer is a long-excursion design with a co-polymer cone, specifically designed for use in Benchmark's acoustic suspension enclosure. The crossover boards for the SMS1 loudspeakers are hand assembled from premium quality components (see **Photo 5**). Benchmark uses ClarityCap ESA-series polypropylene film capacitors and, to avoid paralleling capacitors to achieve desired values, Benchmark has exact values custom made for them.

There are no iron core inductors in the SMS1. Air-core inductors are custom wound for Benchmark by Renco Electronics and are physically oriented to avoid magnetic interactions with each other and the driver voice coils. Power resistors are non-inductive, wire-wound types with low temperature coefficients. The crossover frequency is set at 3 kHz.

Like the AHB2 power amp, Benchmark includes four-pole Neutrik NL4 SpeakOn connectors, along with high-quality, gold-plated binding posts. The rear panel also includes a toggle switch for selecting Normal and Bi-Amp operation. SpeakOn connectors are recommended for the best performance. Benchmark notes that speaker cables fitted with Neutrik NL2 connectors can be used to connect one AHB2 power amp to a pair of SMS1 loudspeakers operating in the Normal mode. The Bi-Amped mode, using a pair of AHB2 power amps, requires a pair of four-conductor cables fitted with NL4 connectors (using the AHB2's "1 & 2 Out" NL4 connector). The SMS1 can also be bi-wired, using a four-conductor cable with an NL2 connector at the amplifier and an NL4 at the loudspeaker, with the switch in the Bi-Amp position.

Benchmark notes that Bi-Amp operation can reduce InterModulation (IM) distortion in the power amp. This is true only if an active crossover is used ahead of the woofer and tweeter power amps. Passive bi-amping involves operating both amplifiers full-range, using the passive crossover built into the loudspeaker. This is how the SMS1 operates and under no circumstances should an active crossover be used with this loudspeaker. Doing so will seriously degrade its performance.

Figure 1, supplied by Benchmark, shows the SMS1's frequency response measurements. The yellow curve shows the woofer's near-field response, while the blue curve shows its far-field response, measured at 6′. This measurement method combines the near- and far-field measurements to calculate the anechoic combined response shown by the green curve.

The SMS1's basic price includes black side panels. For $200 more per pair, they can be ordered with Honduran Mahagony or African Padauk panels. The side panels are not purely cosmetic—they further stiffen and dampen the side walls of the enclosure.

Benchmark uses a stainless steel mesh for the grills, a type normally used for microphone screens. The screen is more acoustically transparent and more durable than conventional grill cloth. Small monitor loudspeakers should normally be placed on loudspeaker stands, which can be purchased from Benchmark (see **Photo 6**). Benchmark recommends that you have 36″ from the floor to the base of the loudspeaker.

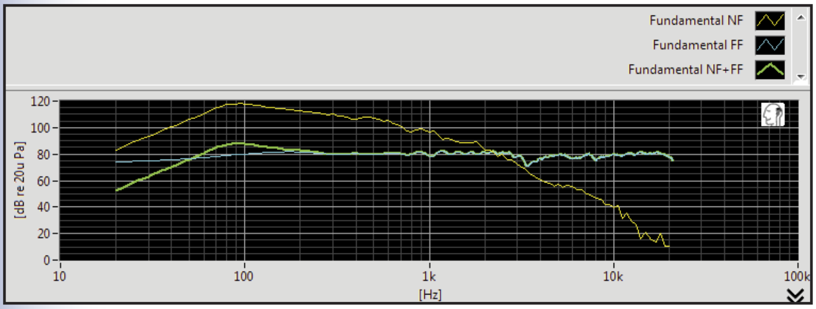

Figure 1: These are the SMS1's frequency response measurements. The yellow curve shows the woofer's near-field response, while the blue curve shows the loudspeaker's far-field response, measured at a distance of 6′. This measurement method combines the near- and far-field measurements to calculate the anechoic combined response shown by the green curve. (Image courtesy of Benchmark Media Systems, Inc.)

The Listening Room

When Benchmark moved to its current facility seven years ago, it built a dedicated, properly dimensioned listening room with acoustical treatment (see **Photo 7**). The concrete floor in this commercial building is certainly an asset. The listening room showcases the SMS1 loudspeaker, fed by a DAC2 HGC DAC/Preamp and the AHB2 power amplifier, with an OPPO Digital BDP-80 digital player and a computer server providing high-resolution digital sources. Benchmark has three other pairs of loudspeakers set up in the listening room for comparison to the SMS1. The room is arranged in such a way that left and right speakers are the same distance apart for each pair. All the listeners need to do is rotate their heads when moving from one pair to another.

The other speakers in the room included the Adam Audio A7X, the Focal Solo 6Be, and the Klein + Hummel O300D. The Adam A7X is a vented design with a 7″ (175 mm) woofer and a 2″ (56 mm) eXtended Accelerating Ribbon Technology (X-ART) tweeter. The X-ART tweeter is a new adaptation of the Air-Motion Transformer tweeter designed by Oskar Heil. The A7X is a powered, bi-amped system, with a 100-W PWM (i.e., Class D) amp on the woofer and a 50 W Class AB amp on the tweeter.

The Focal Solo 6Be is another vented design with a 6.5″ (165 mm) composite sandwich-cone woofer and a 1″ (25 mm) inverted dome tweeter (Focal makes its own drivers). The Solo 6Be is also a powered, bi-amped system, with a 150-W amp using BASH technology driving the woofer, and a 100-W Class AB amp driving the tweeter. BASH amplifiers claim to combine the efficiency of Class D with the performance of a conventional Class AB amplifier.

The Klein + Hummel O300D has a sealed, acoustic-suspension design with a tri-amplified,

Photo 6: A pair of optional stands is available for the SMS1. Some type of stand is highly-recommended for any small monitor loudspeaker. (Photo courtesy of Benchmark Media Systems, Inc.)

SPEAKER FOCUS

powered, three-way system with an 8″ (210 mm) polypropylene cone woofer, a 3″ (76 mm) fabric dome midrange, and a 1″ (25 mm) titanium/fabric dome tweeter. The woofer is powered by a 150-W amplifier, and the midrange and the tweeter by 65-W each. The manufacturer's product brochure notes that the amplifiers are high-efficiency designs, each tailored to the specific driver that they power. The Klein + Hummel loudspeaker is the only one of the four in which the drivers are not vertically aligned—the woofers are mounted inside of the midrange-tweeter array.

Photo 7: Benchmark's dedicated listening room was designed with proper dimensions and acoustical treatment. The four loudspeakers are arranged so that the distance between each left-right pair is equal. From left to right are the Adam A7X, the Benchmark SMS1, the Focal Solo 6Be, and the Klein + Hummel O300D. A third Benchmark SMS1 can be seen in the center, but it was not used for these two-channel stereo comparisons.

Switching between loudspeakers can be fraught with problems, but Benchmark has a sensible solution. In the Benchmark listening room, each of the three powered loudspeakers is fed from its own DAC2 HGC DAC/preamp. Switching is done on the S/PDIF digital lines feeding the various DAC2 HGCs.

During my afternoon in the listening room, I played excerpts from several classical CDs and high-resolution PCM discs that I brought from my own collection. Siau then played a variety of high-resolution popular music recordings, all from files on the digital server. Several of the selections Siau chose featured solo vocals that the recording engineers placed in the center of the stereo image.

I easily preferred the Benchmark SMS1 to the other three loudspeaker systems, for both the classical and the popular recordings. It provided the largest and most spacious soundstage, the most precise imaging, with solo vocals accurately placed in the center of the stereo image.

The SMS1 also had the most neutral tonal balance, with excellent inner detail and resolution. The Adam A7X came in second, acquitting itself extremely well. But, imaging and depth perspective were not as precise as on the Benchmark SMS1. The Focal Solo 6Be had a curious upper midrange depression in its tonal balance, which I found most noticeable on the classical orchestral recordings and the solo vocals on the more popular selections. The Klein + Hummel O300D produced the least precise stereo image, with the location of solo vocals rather vague.

All four systems produced impressive bass for their sizes, but I preferred the smoother, more natural character of Benchmark's acoustic suspension design to the Adam and Focal vented designs. The Benchmark also had the best detail in the bass region.

Results can certainly vary when loudspeakers are heard under differing conditions, and different associated equipment. I admit to being biased toward my own reference system and listening room, and I never feel that I've really heard another piece of equipment unless I hear it with my own system. But, under these conditions, and fed by the AHB2 power amp, Benchmark's SMS1 loudspeaker delivered a very impressive performance. If you're in the market for a small, accurate monitoring system, the SMS1 deserves serious consideration.

About the Author

Gary Galo retired in June 2014 after 38 years as an Audio Engineer at the Crane School of Music, SUNY, in Potsdam, NY. He now works as a volunteer, transferring vintage analog recordings in the Crane archive to digital format. He is the author of more than 280 articles and reviews on both musical and technical subjects. He is an active member of the Association for Recorded Sound Collections (ARSC) and has been a regular presenter at ARSC conferences. Several of his conference papers have been published in the *ARSC Journal*.

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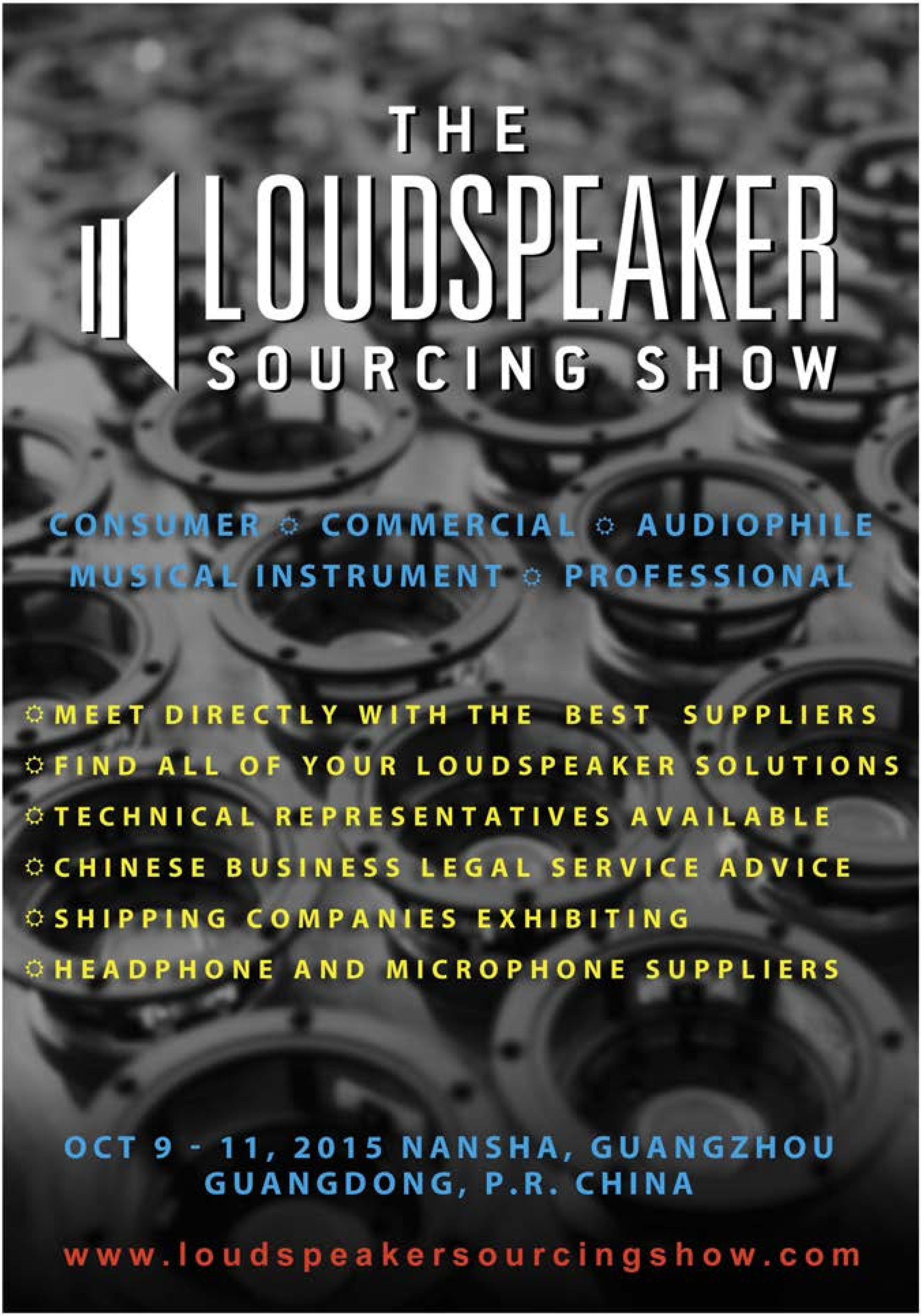
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SPEAKER FOCUS

Construct Your Own Atmo Speaker

For a non-cubistic approach, try this fun loudspeaker design that will provide unique and pleasing results.

By
Ward Maas
(Netherlands)

The idea for the Atmo loudspeaker design originated at an audio fair in Krefeld, Germany. The fair's organizer was the Analogue Audio Association, which is a group of enthusiasts and manufacturers who strive to preserve analog audio sound. The result was a two-day event in which analog aficionados showed up with all kinds of old equipment (think turntables, tape recorders, etc.). Several manufacturers also attended the event with their equipment, which matched the products shown by the analog fans.

One of those manufacturers was Wolf von Langa. He is mainly known for his fantastic field coil loudspeaker systems. These systems range from large to very large. During the show, he demonstrated his latest product, a very affordable small loudspeaker called "The Sphere." Knowing his typically huge systems, many people's first reaction was "What is he up to?"

Photo 1: During an audio fair in Krefeld, Germany, Wof von Langa presented his "The Sphere" loudspeaker.

Well, if you look at these loudspeakers, you immediately know that a very deep bass is something that cannot be expected from such a speaker (see **Photo 1**).

Listening to the "Sphere" speakers revealed that they sounded nice, with a peculiar room-filling soundstage. The loudspeakers were mounted on photo tripods that you could pick up and move. But, the sound stage did not change that much at all. Only when they were put in a corner could you hear extra bass (not surprisingly).

The particularly interesting aspect was that they use two full-range loudspeakers. One speaker faced toward you and the other faced away from you. The intention was to provide a point source, giving the sphere an almost ideal loudspeaker "cabinet."

Replicating the Sphere's Design

After returning home, the thought of trying to make a similar speaker was intriguing. So, I started with a little research. Sphere-shaped loudspeaker enclosures are not new. An Audio Engineering Society (AES) paper presented by Harry F. Olson in 1950, discussed a comprehensive analysis of enclosure configuration and the effects on sound distribution pattern and frequency response characteristics. In other words: What shape of loudspeaker system sounds best? The sphere-shaped enclosure was deemed the best. Unfortunately, making such a sphere-shaped enclosure is not easy.

Yes, there are some successful attempts by Elipson or Cabasse (see **Photo 2**). They made /make very nice products. However, besides the positive effect of the sphere-shape on audio quality, the rest of its qualities can be characterized as "potentially problematic." They are not easy to make, not easy to ship, not easy to sell, not easy to.... you get the drift.

So sphere-shaped loudspeakers, certainly for a DIYer remained an obscure cult product until two unlikely product developments met.

First, due to the increasing demand for soundbars, loudspeaker manufacturers developed small good quality full-range speakers, which meant that reasonable quality loudspeaker systems can be built with them.

Second, there was IKEA. Yes, the Swedish furniture giant. Apart from its range of stylish affordable furniture, the company offers a wide range of matching accessories. It has become a popular sport to take an IKEA product and change it into something unintended. You can find lots of "IKEA hacks" on the Internet. One of the hacks is the "Blanda matt" loudspeaker. Blanda matt is basically a half-sphere serving bowl made from bamboo. Two bowls make a sphere with two flattened surfaces. Typically, one of the bowls is used for the loudspeaker; the other for the loudspeaker terminals.

Wolf's sphere is a high-quality milled, polished, and lacquered sphere with full-range speakers specially made for this product. Yes, they are highly recommended. This, however, is about something you can make that will bring a smile to your face while enjoying the music.

Speaker Search

Normally, you would choose a loudspeaker and design an enclosure for it. Instead, we start with two Blanda bowls and find loudspeakers to fit.

Ikea's Blanda bowls come in three sizes—12 cm (4.72"), 20 cm (7.87"), and 28 cm (11.02"). I chose the 20-cm bowls for two reasons: they more or less matched the original's size and the price was reasonable. The smaller one is suitable for a tweeter and the larger can be used for a midrange or small bass speaker. But those are projects for another day. The two bowls have an overall diameter of 20 cm and

Photo 2: The Elipson AS 50 features a futuristic design with iconic spherical speakers.

an inner diameter of 18.5 cm. That makes roughly a volume of 1.7 ltr per bowl or 3.4 ltr per sphere. That is a number you want to keep in mind for your calculations.

The flat surface has a 90-mm diameter. That should be roughly the size of the loudspeaker we are going to use. Ideally, the loudspeaker will be flush to the bowl. As a rule of thumb when looking for loudspeakers, you only take into account loudspeakers that come with a datasheet. That way you avoid low-quality loudspeakers. Of course, if you have a loudspeaker that looks fantastic and is really cheap, it is worth the experiment.

After a long search I found the Monacor SPX-30M, which looked most pleasing with regard to specs, appearance, and price, which was important as it was intended as an experiment. If the results are very good you can always find more expensive loudspeakers.

This loudspeaker has an F_S of 100 Hz, a Q_{TS} of .71 and a V_{AS} of 1.78 ltr. With a double bass, as in this situation, V_{AS} drops to half of the value to 0.89 ltr. In reality, it does not matter that much. Play around with the figures in a speaker calculator and you will find

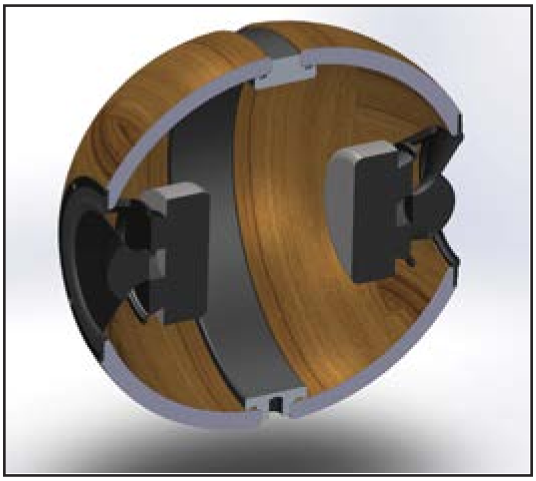

Photo 3: This drawing shows the Atmo speaker's interior.

Photo 4: These are the parts I used to construct the Atmo speaker.

the obvious. The deep bass is not there, but it will be able to go surprisingly deep. A volume mismatch will give a higher system Q.

Resource

H. F. Olson, "Direct Radiator Loudspeaker Enclosures," Audio Engineering Society (AES) paper, 1950.

Sources

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Construction

First, make the hole for the loudspeaker. Determine the exact center of the bowl's bottom and drill a little hole that serves as the center for the hole saw. If you have something like a milling machine you know what to do. The hole should be large enough for the magnet and the top part of the basket to nicely fit. As I said, ideally, the rim of the loudspeaker should be flush with the bowl. Therefore, we need to sand down the bottom of the bowl a bit more. Rough sanding paper on a hand sanding tool works nicely and accurate. The bamboo material handles the sanding well, so that makes life easier. Then, drill the holes for the loudspeakers.

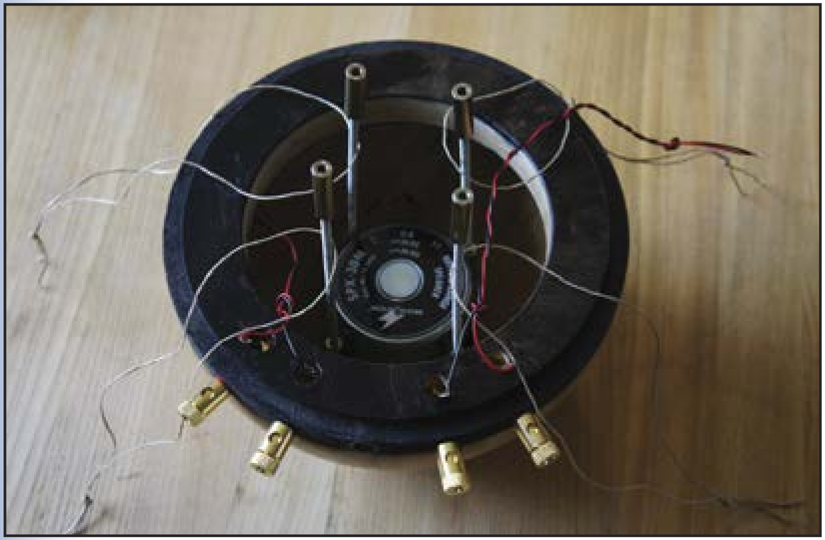

Photo 5: Attaching strings around the threaded rod makes it easier to line up the holes in the bowl.

Next, put the bowls together. That sounds easier than it is. Yes, you simply can sand the top of the bowls flat (stick a sheet of sanding paper on your work bench and move the bowl around in circles), glue them together, and drill holes for the speaker connections. That is a bit crude and leaves you with the problem of how to position the loudspeaker. After all it is a bowl/sphere.

An elegant detachable solution is to use a ring between the two halves. That provides a nice place for loudspeaker connections and a spot to fit a 1/4-20 UNC thread. That thread is not readily available in Europe. But, I found a thread-making set on the Internet. I had a lathe available to me, so construction was rather easy. If you do not have access to a lathe, a bit of creativity is required. Of course, you are also free to alter the construction.

The ring has to fit both bowl halves and have a ridge in the middle flush to the outside diameter of the bowl. On that ridge, drill holes for the loudspeaker connections and the "photo" thread. You can use all kinds of material for the ring. Plastic, wood, MDF, or metal (isolate the loudspeaker connections). This depends on the material and tools at hand. If you have a lathe or milling machine that can handle that diameter, it is easily done. But there is nothing wrong with doing it the hard, old-fashioned way using a jig saw and gluing the thin multiplex rings together. **Photo 3** shows the use of a nicely designed ring, which has two grooves for rubber sealing rings. Of course, this is a bit overdone for a fun project. I made mine from MDF. **Photo 4** provides an overview of the parts.

You can use some stuffing material, if you want. Remember that tightly packing the cabinet reduces the housing volume and loose filling increases that volume.

You also want to ensure the loudspeakers are now perfectly aligned. This also provides the method used to put both bowl halves together. The (nice looking) screws I used to fit the loudspeakers are not fastened by bolts, but by threaded rod connectors. Between those threaded rods rigidly connect both speakers together. Of course, it is a bit challenging to exactly line up the threaded rod connector with the holes in the bowl and the loudspeaker. A piece of string around the threaded rod helps and makes it rather easy (see **Photo 5**). Cut and remove the strings before tightening the last screws and you are ready!

To Finish the Loudspeaker Sphere

I put the loudspeaker sphere on a table microphone standard that fits beautifully (see **Photo 6**). The loudspeaker connections are on the left and right side. They are completely separate. That leaves some intriguing possibilities. You can use one in a single-source way. Turn the speaker 90° with the left channel to one speaker and the right channel to the other. You can build a loudspeaker pair and use them in stereo with the speakers in series or parallel. Or, you can use them in stereo with the back speaker connection reversed. You can also use the front speakers for the left and the right and cross-connect the back speakers for right and left (!) (reversed or not). Another option is to use them in a multi-speaker system with a subwoofer. There are plenty of possibilities to explore.

The Results

Listening to music, these little spheres perform nicely and probably very differently from what you have experienced. At the audio show, one man who stepped into Wolf von Langa's demo room, expecting his usual large systems, said to him "Du schertzt wohl!" (which translates to "you've got to be joking"). Well, that man became silent after hearing Wolf's "Sphere" loudspeakers.

I hope you do the same with your DIY ones. You might find yourself in a situation where you are listening to these experimental speakers that you built a bit longer than you would like to admit. But hey, that's what DIY audio is all about. Surprise, be surprised, and smile!

Photo 6: The Atmo speaker is my DIY attempt to replicate Wolf von Langa's "The Sphere."

My thanks go to Bertus Devente , Gert de Beer, and Dylan Goulooze for their support.

About the Author

Ward Maas is the owner of Pilgham Audio. He studied electronics, marketing, and amplifier design. During his career in consumer electronics, Ward worked in areas ranging from CD standardization to radio and television to personal GPS navigation. Ward has worked on an extreme low-noise magnetic cartridge preamplifier and several special amplifier products. As the CTO of "Witchworld," a theme park near Amsterdam, he is also works with animatronics. He lives in Almere, Netherlands, with his wife and son.

You Can DIY!

Build a Bookshelf Speaker

Many people live in compact spaces and DIY enthusiasts often seek smaller speaker options. In this article, I will demonstrate how to build the LIB-2, a compact loudspeaker designed to fit on a bookshelf.

Photo 1: I designed the LIB-2 loudspeaker so it would fit on a bookshelf.

By
George Ntanavaras
(Greece)

"What we need are loudspeakers designed with knowledge of how they are to be mounted, where they are to be placed. The idea of a universally applicable, one-type-does-all loudspeaker is a 'steam-era' concept," wrote Floyd Toole in his book *Sound Reproduction.* And, his words should be on the mind of every loudspeaker designer who wants his loudspeaker to offer maximum reproduction capabilities. Unfortunately, in most cases loudspeaker placement is beyond the control of its designer, but this is the advantage of a DIY design. So keeping Toole's words in mind, I will share this loudspeaker designed to be placed on a bookshelf (see **Photo 1**).

Loudspeaker Design Considerations

The first decision I had to make was the loudspeaker's dimensions. So, I checked the measurements of the shelves of several bookcases that I found in the houses of relatives and friends. Once done, I decided my loudspeaker height should be 33 cm (approximately 12.9") and the total depth should be less than 35 cm (approximately 13.78") including the input connectors. I will finalize the loudspeaker's width later because it will define the total volume of the box.

My initial thought was to design a three-way system because I wanted a wide dispersion from the loudspeaker. But in the end, I decided to stay with a two-way system because a three-way would considerably increase the cost and the size. Additionally, it would have required a more complicated and more expensive crossover.

For the woofer alignment, I prefer a closed box. However, for this loudspeaker, I decided to use the bass reflex alignment for several reasons. I wanted to further evaluate this alignment by using a rather high volume box with a lower than usual tuning box frequency. Also, the bass reflex box will increase the loudspeaker's efficiency and greatly improve the low-frequency extension. And finally, a bass reflex box could always be easily transformed into a closed box simply by plugging its bass reflex port. As we will see later, this proved to be very convenient.

Choosing the Drivers

The next thing I did was select the drivers. For the woofer, I selected the SEAS Prestige H1456-08-ER18RNX unit. I have used this driver for other projects with excellent results. It is a 6.5" reed/paper pulp cone driver with a 12-mm peak-to-peak linear excursion and up to 22 mm maximum excursion. According to the manufacturer, it has the following Thiele-Small (T-S) parameters: F_S = 37 Hz, Q_{TS} = 0.32, and V_{AS} = 32 ltr. These parameters are suitable for a bass reflex loudspeaker.

For the tweeter, I chose the SEAS Prestige H1212-06 27TBFC/G. This is a dome tweeter with a wide, soft polymer surround and a rear chamber. Its diaphragm consists of an aluminum/magnesium alloy. It has a grid that protects the diaphragm, and supports a phase plate, which compensates for a slight axial roll off toward 20 kHz. The voice coil windings are immersed in a magnetic fluid that increase the short-term power handling capacity and reduce the compression at high power levels. Another interesting feature of this loudspeaker design is that it has a rear chamber with acoustic damping that enables the tweeter to have a low-resonance frequency and can be used with moderately low crossover frequencies.

To begin, I measured the T-S parameters of the two H1456-ER18RNX drive units that I had available. **Table 1** shows the results. **Table 1**'s third column shows the average of the two previous columns. As shown, the measured F_S and Q_{TC} parameters are very close to the manufacturer data but the V_{AS} that I measured was larger by about 16%.

Parameter	ER-18 No. 1	ER-18 No. 2	Average
F_S	36.46 Hz	37.2 Hz	36.83 Hz
R_E	6.81 Ω	6.66 Ω	6.74 Ω
Q_T	0.32	0.33	0.33
Q_{ES}	0.42	0.44	0.43
Q_{MS}	1.37	1.35	1.36
M_{MS}	13.34 grams	12.73 grams	13.04 grams
R_{MS}	2.23 kg/s	2.20 kg/s	2.21 kg/s
C_{MS}	0.001428 m/N	0.001438 m/N	0.001433 m/N
V_{AS}	37.11 ltr	37.36 ltr	37.24 ltr
Sd	136 cm^2	136 cm^2	136 cm^2
B_L	7.08 Tm	6.75 Tm	6.91 Tm

Table 1: These are the results from the Thiele-Small measurements I took for the two H1456-ER18RNX drive units.

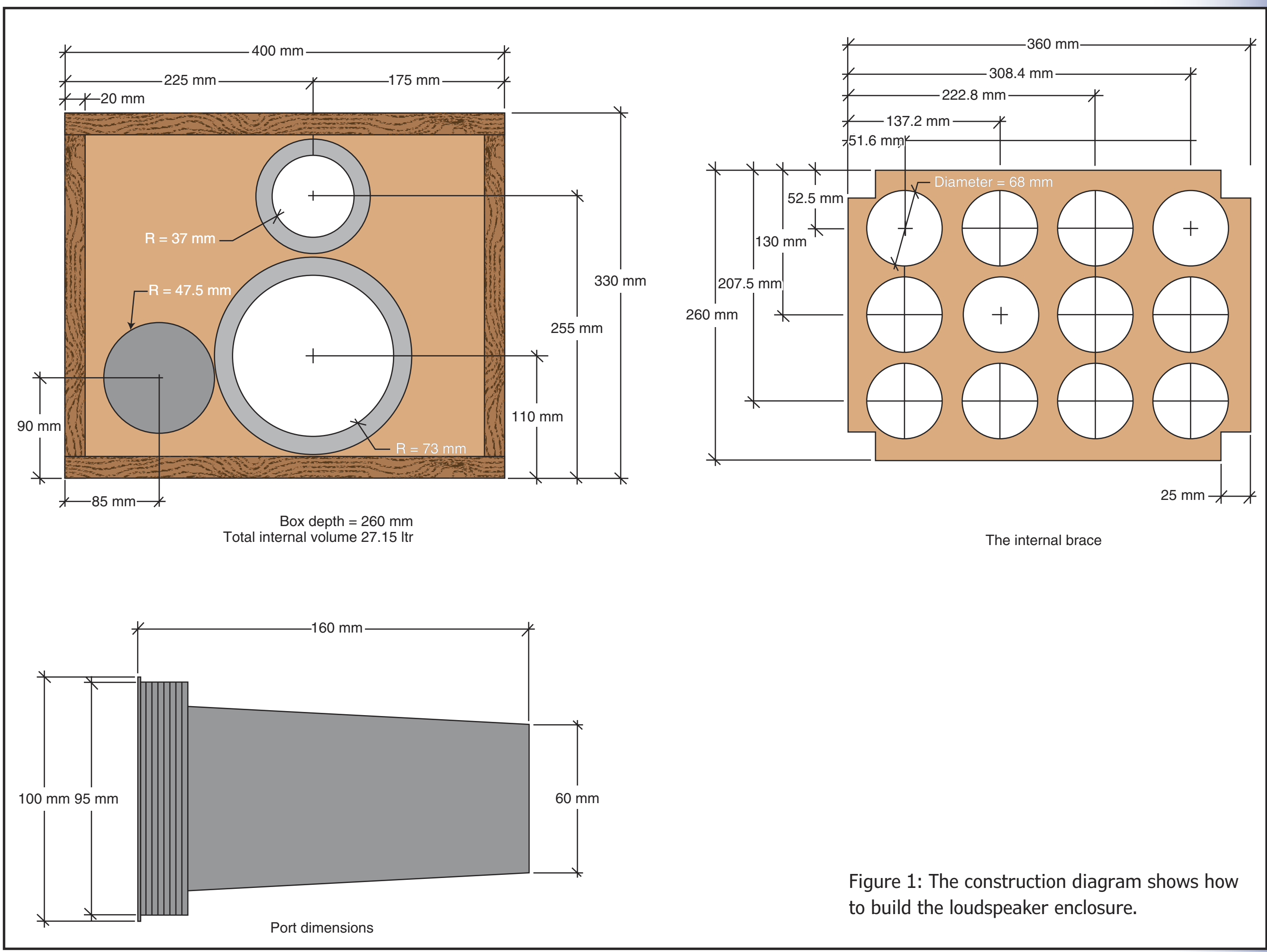

Figure 1: The construction diagram shows how to build the loudspeaker enclosure.

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Photo 2: Here is an internal view of the loudspeaker box's without the back side attached.

The Box Design

D. B. Weems in his book *Great Sound Stereo Speaker Manual* describes a simple method for a vented-box design based on the pocket calculator method of Don B. Keele, Jr. According to this method, only three parameters of the driver are needed to design the box: the F_S, the Q_{TS}, and the V_{AS}.

From these parameters, the volume of the loudspeaker box Vb is calculated: Vb = 15 × V_{AS} × $Q_{TS}^{2.87}$.

Then, the optimum alignment with a frequency response that has no hump or dip will have a box resonance frequency (f_B) given by the formula: f_B = 0.42 × F_S × $Q_{TS}^{-0.9}$.

Inserting the values of F_S, the Q_{TS}, and the V_{AS} shown in **Table 1**'s third column into the formulas, gave the following results: Vb = 23.2 ltr and f_B = 42 Hz.

Since the room gain at low frequencies will increase the loudspeaker's bass level, I decided to build a box with the same volume as calculated from the above formula but to lower the box resonance frequency (f_B) at 36 Hz. This alignment will be close to third-order rolloff offering better transient response and smoother bass response when the loudspeaker is placed in a room. This becomes even more important because the loudspeaker will be placed on a bookshelf, close to a wall.

Figure 1 shows the box's dimensions. The total external volume of the box is 34.3 ltr and the internal volume is 27.1 ltr. I estimate the box's net volume, after subtracting the volume of the drivers, the tube, the crossover, and the internal braces to be close to the desired volume of 23.5 ltr.

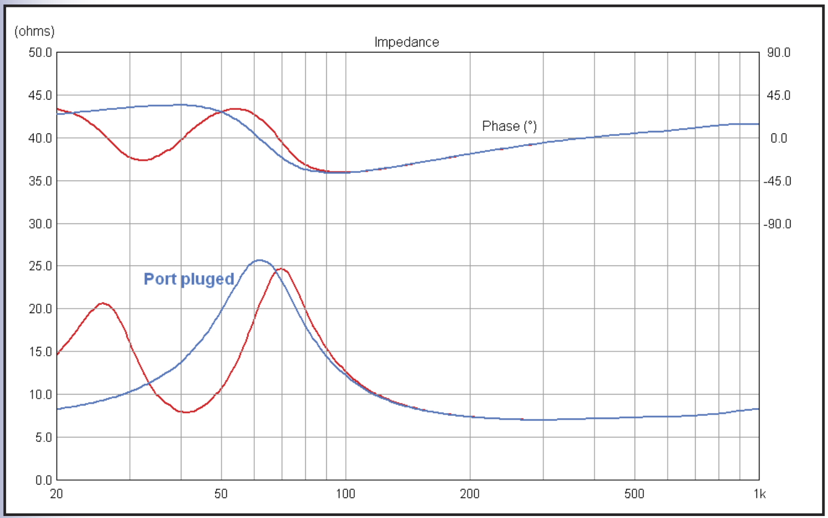

Figure 2: The woofer's impedance in the bass reflex box is measured.

Box Construction

I used 20-mm plywood to construct the box. I glued all the sides together with a good-quality wood glue. Based on my experiences from previous projects, I designed an adequate bracing to eliminate the box vibrations as much as possible. I used a main horizontal internal brace as shown on the right side of **Figure 1**. I placed the internal brace between the two drivers to separate the box into two parts. Several openings in the brace ensure that all the volume of the box is seen by the woofer. Also, I used two other vertical braces. **Photo 2** shows a view of the bracing.

For the bass reflex port, I used a black plastic tube-type EP-68.220 from AB Sound. The front opening has an 68-mm internal diameter while the end of the tube has a 60-mm internal diameter. The final length of the port tube is 160 mm as shown in the bottom of **Figure 1**.

To increase the damping, I placed self-adhesive bituminous panels with 2-mm thickness on each side of the box. I also used additional glue to affix these panels to the wood.

For the box stuffing, I used two different materials. I glued 20-mm thick acoustic foam on each side and loosely filled the inside of the box with lamp's wool. I kept the path between the back of the woofer and the opening of the port clear.

Photo 2 shows an internal view of the enclosure without the back and all the details for the construction of the box.

After, sanding the outside of the box, I applied several layers of a wood-imitation varnish.

	Impedance Measurement No. 1 Loudspeaker	Impedance Measurement No. 2 Loudspeaker	Near-Field Measurement No. 1 Loudspeaker	Near-Field Measurement No. 2 Loudspeaker
f_L	24.87 Hz / 20.3 Ω	25.6 Hz / 18.6 Ω	-	-
f_B	40.68 Hz / 8 Ω	39.5 Hz / 7.9 Ω	-	-
f_H	68.36 Hz / 24.5 Ω	66.4 Hz / 22.1 Ω	-	-
f_C	60.33 Hz / 25.4 Ω	60.8 Hz / 23.9 Ω	-	-
f_B	40.6 Hz (calculated)	37 Hz (calculated)	36.5 Hz	36 Hz
Q_L	13.3	14.8	15.6	15.7

Table 2: Here are the low-frequency measurements for both loudspeaker boxes.

Measurements and Evaluation

Once the box was ready, I placed the drivers and took some measurements to evaluate the box's design. **Figure 2** shows the impedance with the woofer direct driven and the woofer's impedance with the port completely plugged. **Figure 3** shows the woofer's near-field responses and the port. The woofer's effective diameter is 13.16 cm, while my estimation of the effective diameter of the port is about 6.5 cm. For this reason, I scaled down the port output by 6.5/13.16 or −6.1 dB.

From those measurements, I recorded the following values that are needed for the evaluation of the box following the method that is described in Vance Dickason's *Loudspeaker Design Cookbook:*

- f_L = the frequency where the impedance has the first peak
- f_H = the frequency where the impedance has the second peak
- f_B = the frequency between the two above frequencies where the impedance has its

lower value. This is the box tuning frequency.

- f_C = the frequency where the impedance of the box, with the port plugged, has its maximum value

Usually the impedance curve in the region around f_B is very shallow so that it is not accurate to find the minimum value. In this case, the f_B can be measured more accurately from the woofer's near-field frequency response. It is the frequency at which the woofer has the minimum output.

Another way to find accurately the f_B is to calculate it mathematically from f_L, f_H, and f_C using the formula $F^2_B = (f^2_L + f^2_H - f^2_C)$.

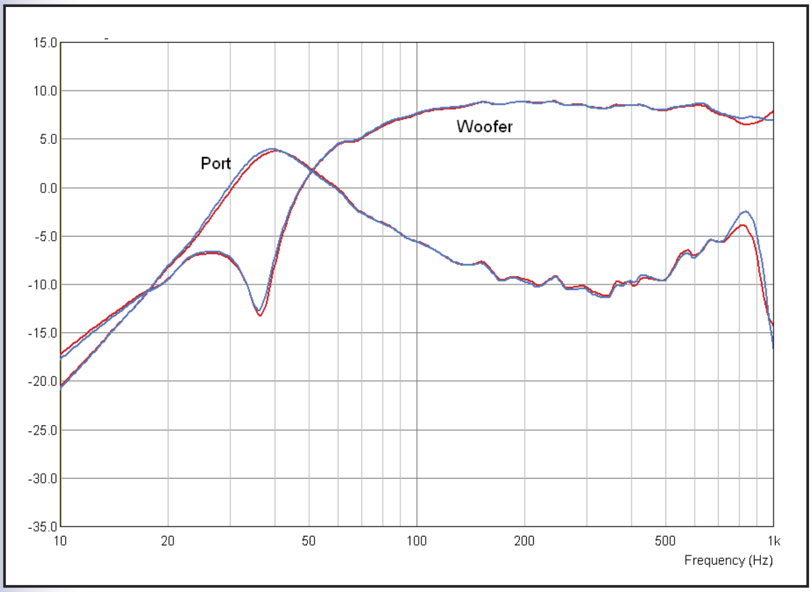

Figure 3: These are the near-field responses of the woofer and the port.

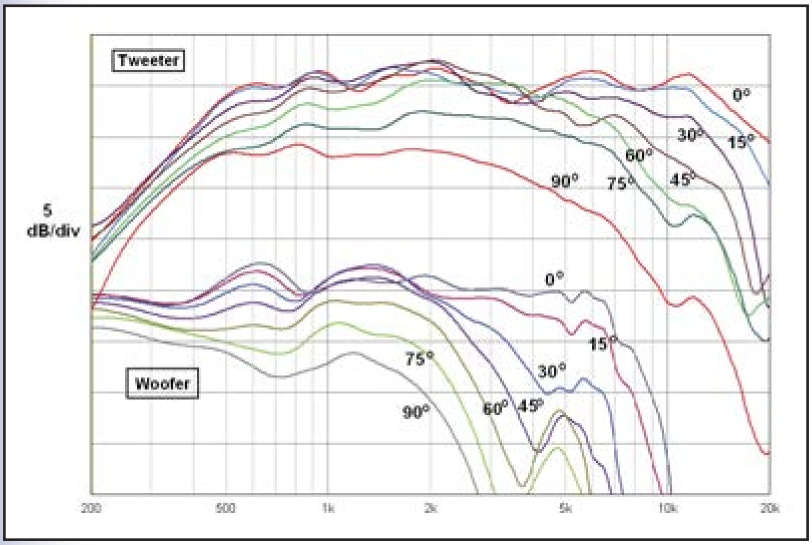

Figure 4: These measurements show the horizontal polar response after equalization for the woofer and the tweeter.

All three values should be very close but differences may exist due to box losses and filling, the voice coil's inductance, or in series crossover components.

Table 2 summarizes the results of the above measurements. The leakage losses of the box Q_L was calculated with two different methods to check the accuracy. First using the calculated value of f_B and then using the value for f_B that was taken from the near-field measurement. Both methods gave a value of about 15 for the loss of the box due to leakage (Q_L).

The near-field responses of the two woofers, as shown in **Figure 3**, are similar indicating that the chosen alignment is not very sensitive to changes in loudspeaker and box parameters.

Crossover Design

The crossover frequency selection is an important decision with regard to this loudspeaker design. Several parameters of each driver should be measured before making a decision. In my opinion, one of the most important of them is the off-axis response of the drivers.

I mounted the woofer and the tweeter in the enclosure and and measured the polar response of the two drivers after they were equalized using my Behringer DCX2496 device to have a nearly flat on-axis response. The woofer was equalized up to the frequency of 5 kHz, while the tweeter was equalized down to 500 Hz. This helps provide a clearer view of the capabilities of the drivers without introducing any irregularities in the frequency response.

Regarding the measurements, I took them all in my listening room and this did introduce some errors. To eliminate these errors as much as possible, I placed the loudspeaker on a stand in the middle of the room so that the driver was at half the height of the room (about 1.35 m from the floor). Then, I placed the microphone 0.75 m from the speaker. This gave a gate window of about 6.5 ms for the measurement.

To take the measurements, I used an EMM-8 microphone with a MP-1r preamplifier from iSEMcon. The microphone according to the manufacturer was calibrated against a Brüel & Kjær 4133 reference microphone. I loaded the calibration file that I received to the ARTA software that I use for the measurements and the analysis. I did this to compensate for any possible errors from the microphone response.

Figure 4 shows the results. The woofer has a good polar response up to about 2 kHz. Above this frequency, it rolls off smoothly as expected. Also a wider dispersion in the response is shown between

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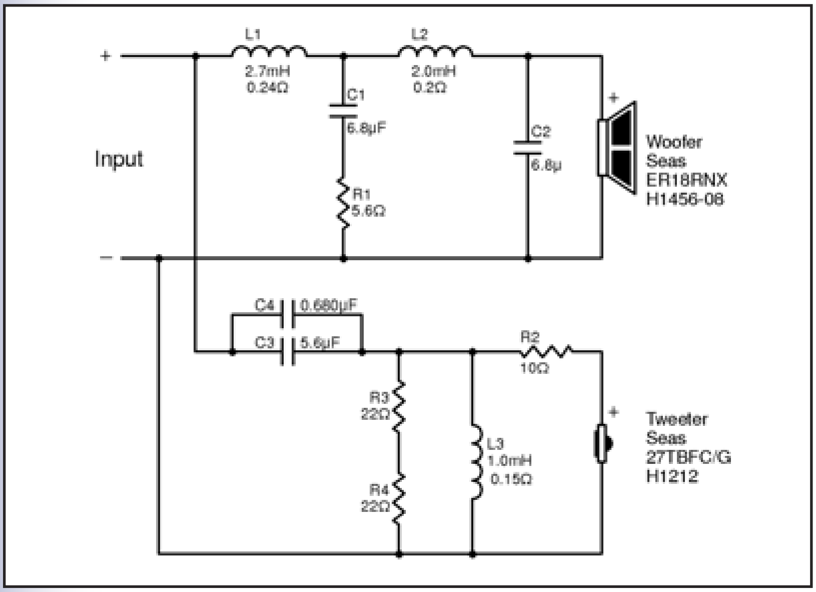


Figure 5: Use this crossover electrical diagram to build a replica of my LIB-2 bookshelf loudspeaker design.

800 Hz and 2 kHz. The tweeter has a good polar response up to 5 kHz and a good response up to 8 to 10 kHz. A wider dispersion in the response at the region of 3 kHz is also apparent. Above 10 kHz, it rolls off as expected from its 1″ diaphragm.

I chose a third-order Butterworth filter with the crossover frequency set at 1,700 Hz. The 18 dB/octave slope will give sufficient attenuation to the woofer resonance, which exists in the region of 5 to 6 kHz and will protect the tweeter from high excursions at the lower frequencies.

I computed the initial values of the components using relevant theory for the design of the filters. The box's front baffle is 33 cm (13″) × 40 cm (15.75″). According to the Edge diffraction simulator program, it should have a baffle step of 6 dB between 140 and 300 Hz. The crossover should compensate for this baffle step, and the Edge simulator gave some initial values for the components that should be useful.

Next, I used the LspCAD 6 demo version for the passive crossover simulation. The demo version contains all the functionality of the unlimited version with a few exceptions. It is not possible to save projects. It is also not possible to export data, and the crossovers are limited to two drivers.

First, import each driver's frequency response and impedance to the program. Then, the loudspeaker's total response can be easily computed by the program. It is instantly displayed. When the value of any component is modified, the result is immediately shown in the frequency response, in the impedance curve, and many other useful displays.

When I was satisfied with the simulation results, I used a piece of plywood to assembly the prototype crossover. Then, I remeasured the loudspeaker's frequency response to confirm that everything was as the simulation predicted. Next, I proceeded to the listening tests with the loudspeakers placed in the bookshelves.

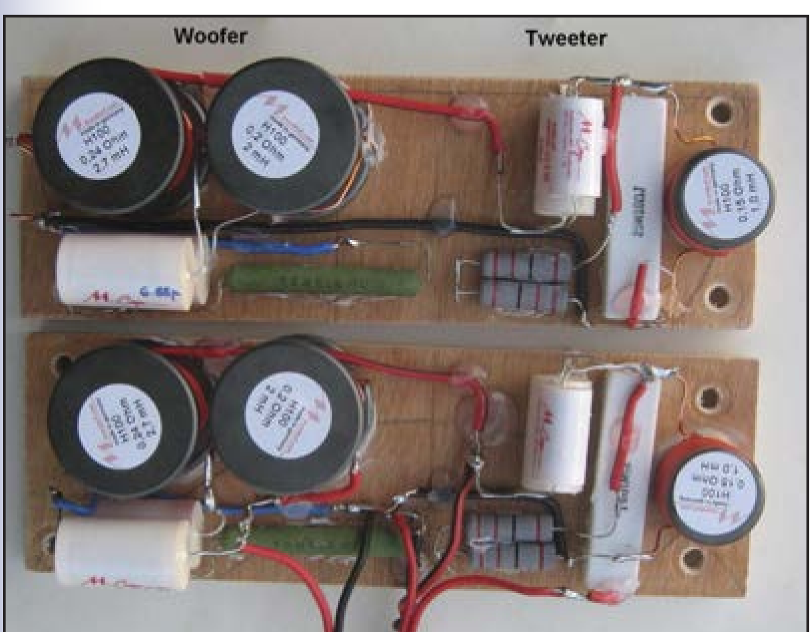


Photo 3: These are the crossovers I used for the loudspeakers.

Listening to the Loudspeakers

From the first notes of familiar music, it was obvious that the low-frequency levels were excessive. I modified the baffle step at the level and at the transition frequencies, but this did not solve the problem. When I took a frequency response measurement, I noticed a strong peak of approximately 20 dB high at around 36 Hz.

My listening room has a length of about 9.5 m, which sets the main axial room mode at about 18 Hz and the second-order axial room mode at 36 Hz. This mode was strongly excited because the loudspeaker was placed close to the end of the room. The peak was so strong that with the loudspeaker placed on the bookshelf, it was present even in the woofer's near-field response.

However, I found a simple solution to this problem when I remembered that this frequency was close to the tuning frequency of the bass reflex port. I used a piece of cloth to plug the bass reflex port, and the peak was almost eliminated. Now, the reproduction of the bass frequencies was evenly balanced. The loudspeaker's placement in the bookshelf, near to the listening room's wall, produced an increase in the bass frequency that made the bass reflex port unnecessary.

I continued the listening tests with the port plugged, and I finalized the crossover design after making some other small crossover changes.

Although my procedure is time consuming procedure and requires great attention to details, it proved to be absolutely necessary. Besides, this is the advantage of designing and building a DIY loudspeaker.

As a final check, I placed the loudspeaker in a free-standing position. As expected, the loudspeaker sounded lean at the bass frequencies. To my surprise, when I unplugged the port everything was just fine. The loudspeaker's balance was smooth and no other change to the crossover was necessary.

Crossover Construction

Figure 5 shows the final circuit diagram for the crossover. Both drivers are connected in phase. Below the value of the each inductor, I have indicated the internal DC resistance. I used inductors with ferrite cores and 1-mm oxygen-free copper (OFC) round-wire that gave a low internal resistance. I used metalized polypropylene capacitors with a low loss factor and wire-wound power resistors. I glued all of them on a piece of 21 cm (8.3") × 7 cm (2.75") plywood.

Photo 3 shows the completed crossovers. The woofer part is shown on the left side and the tweeter part is shown on the right side.

I used four screws to secure the crossover to the back side of the cabinet behind the port. Then, I used 1-mm² cable for the woofer connection and 0.75 mm² cable for the tweeter connection.

Loudspeaker Measurements

I took a lot of measurements after the loudspeaker's completion.

Figure 6 shows the loudspeaker's total on-axis response, the response of the woofer and the tweeter, and the total on-axis response with the tweeter inverted. The crossover frequency at 1.6 kHz is close to the initial value of 1.7 kHz. Also, the clear notch at the crossover frequency, when the tweeter is inverted, indicates a good integration between the woofer and the tweeter at the crossover frequency.

Figure 7 shows the horizontal dispersion of the loudspeaker. The measurements were taken on the loudspeaker's axis and at 15°, 30°, 45°, 60°, 75° and 90°.

In **Figure 8**, the loudspeaker's vertical dispersion is shown with measurements taken on the loudspeaker's axis and at 15° above and below its axis.

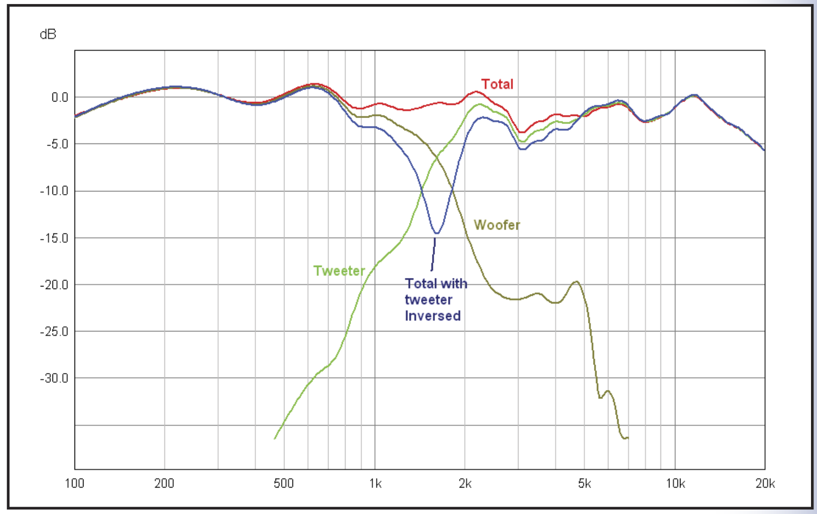


Figure 6: I measured the loudspeaker's on-axis responses.

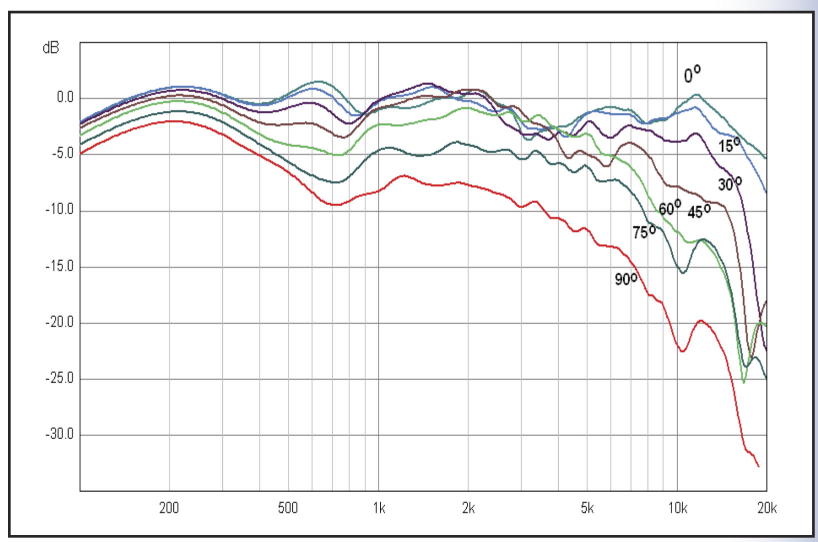


Figure 7: This is the loudspeaker's horizontal polar response.

About the Author

George Ntanavaras graduated from National Technical University of Athens (NTUA), Greece in 1986 with a degree in Electrical Engineering. For more than 20 years, he has worked in the Research and Development Department of a large Greek telecommunications company. He began his career as a design engineer and then worked in several managerial positions. He currently works as a freelance consultant. His audio interests include the design of preamplifiers, active crossovers, power amplifiers, and loudspeakers. He also enjoys listening to classical music.

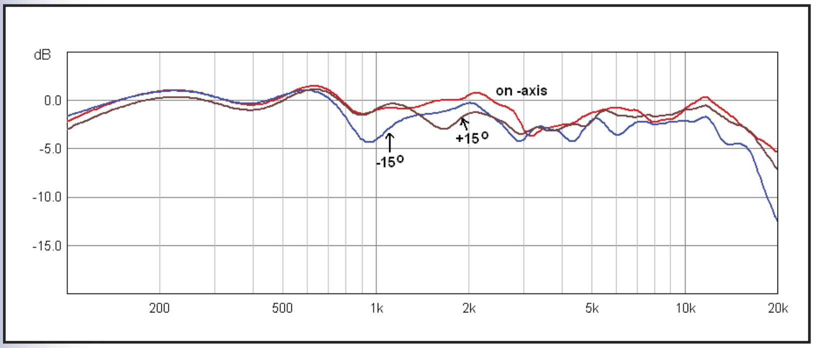


Figure 8: I also measured the loudspeaker's vertical polar response.

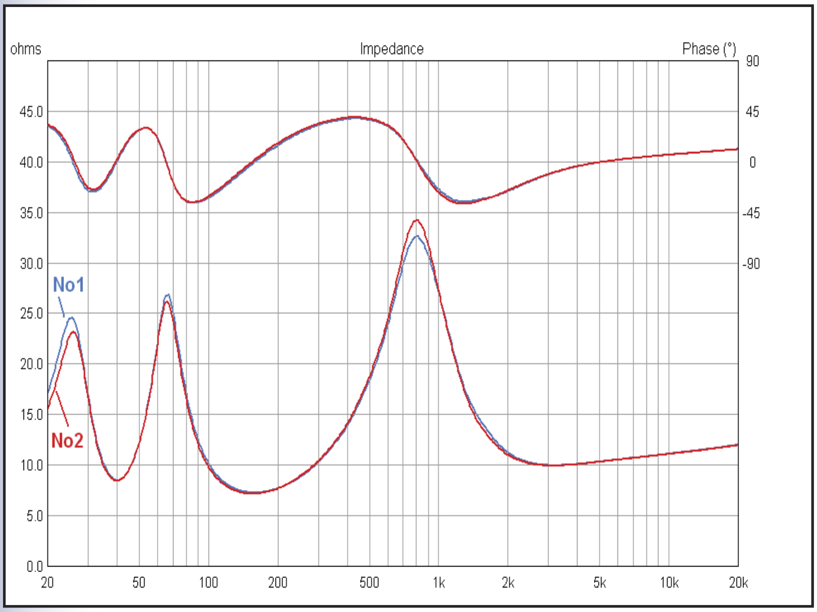


Figure 9: These measurements show the loudspeaker's electrical impedance and phase.

Photo 4: This is my revised accelerometer system, which I used to take vibration measurements.

Figure 9 shows the measured impedance of the two loudspeakers. They are similar with some minor differences in the values of the peaks. The impedance's minimum value is 7 Ω at the region of 150 Hz. However, in this frequency, the impedance's electrical phase is close to 0°, resulting in a very easy load for any amplifier.

This was one of the design targets because I also wanted to eliminate the frequency response irregularities due to the cable length between the loudspeaker and the amplifier. The maximum impedance value is 34 Ω at about 800 Hz. The phase has a minimum value of −40° at 1,200 Hz but the magnitude is 17 Ω and a maximum value of +40° at 440 Hz but the magnitude here is 15 Ω.

Vibration Measurements

I used my accelerometer system during the loudspeaker's development to test the enclosure's vibration behavior. This is a reliable test system that I designed a few years ago. It can easily identify any vibration problem and help evaluate the effects of bracing, damping, material selection, and so forth. It consists of the ACH-01-03 accelerometer, an amplifier/processor that computes the acceleration input signal's velocity and displacement, and an external 12 VAC power supply adapter (see **Photo 4**). The system I used here is an improved version of the design that I previously presented in *audioXpress* (see Resources).

I investigated the enclosure's panel vibration behavior by attaching the accelerometer to the panel using a very thin double-sided tape. I had to shield the accelerometer for these low-voltage measurements because strong interferences from the

Crossover Components List

Capacitors

(All capacitors are 250 V, MKP)

C1	6.8 µF
C2	6.8 µF
C3	5.6 µF
C4	0.68 µF

Inductors

(All inductors are cored with non ferrite material, except as noted)

L1	2.7 mH, 0.24 Ω
L2	2 mH, 0.2 Ω
L3	1 mH, 0.15 Ω

Resistors

R1	5.6 Ω/10 W
R2	10 Ω/25 W
R3	22 Ω/5 W
R4	22 Ω/5 W

50-Hz line voltage were present at all measurements. This was easily done with a 5 cm (2”) × 5 cm (2”) piece of aluminum foil around the accelerometer, which was electrically connected to the amplifier ground.

For the analysis, I again used the ARTA software, except this time the input signal was the output of the accelerometer amplifier. I used pink noise produced from the ARTA software for the excitation signal, which I input to a power amplifier that drove the loudspeaker at a high sound level.

It was necessary to use an additional 20 dB gain on the accelerometer preamplifier because the signals measured by the accelerometer were very low in level. I used ARTA software’s “Scale” function to compensate for the additional gain by –20 dB down.

Figure 10 shows one of the measurements that I took. The top curve is used as reference, and it was taken with the accelerometer attached on the woofer’s cone with the crossover connected. The other three curves are measurements of the vibration from the loudspeaker box’s sides. The top curve is the acceleration of the front panel, the middle of the back side, and the bottom curve of the left side. Comparing them with the woofer’s cone curve shows that they are 50 to 60 dB down, confirming the box was adequately braced and damped and that no serious problem could be expected due to panel vibrations.

Overall Design

The LIB-2 is a good sounding loudspeaker with a well-balanced frequency response. Its bass extension is adequate and, in most cases, no subwoofer will be needed. It is also a very easy load for any amplifier.

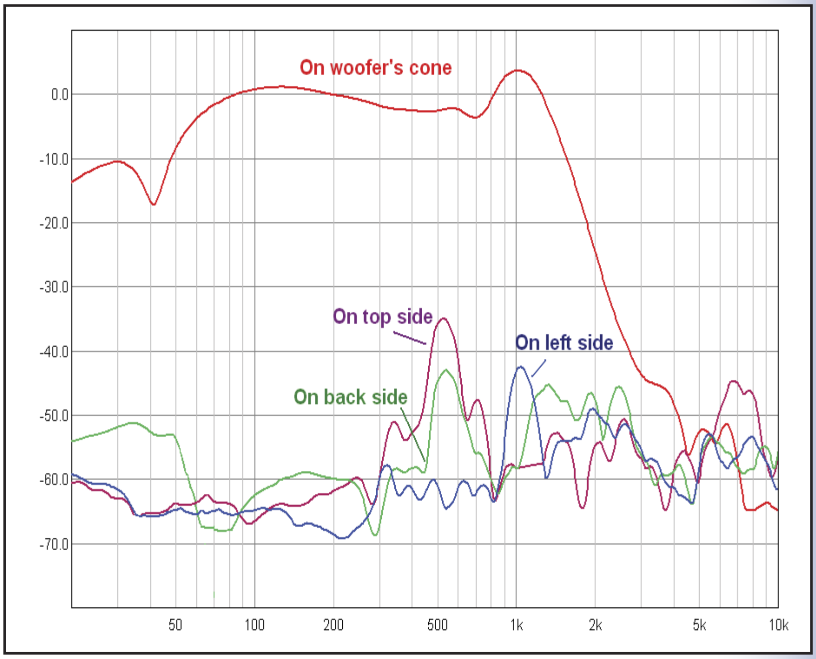


Figure 10: I used an accelerometer system to measure the vibration of the enclosure panels.

If I were to build another bookshelf loudspeaker, I would choose a closed-box alignment since I found the increase in the level of the bass frequencies due to placement and wall proximity made it unnecessary to have a bass reflex box.

Author’s Note: I have a small quantity of high-quality PCBs (with solder mask and silk screen) and engraved front panels (as shown in Photo 4), which can be used to construct the accelerometer amplifier. I can be reached at gntanavaras@gmail.com.

Resources

ARTA software, www.artalabs.hr.

V. Dickason, *The Loudspeaker Design Cookbook*, 7th edition, Audio Amateur Press, 2005, www.cc-webshop.com.

EDGE program, www.tolvan.com/index.php?page=/edge/edge.php.

LspCAD program, www.ijdata.com/LspCAD_demo.html.

G. Ntanavaras, “Accelerometer Testing of Loudspeaker Drivers,” *audioXpress*, June 2011, http://audioxpress.com/article/Test-your-speakers-performance-with-this-do-it-yourself-measurement-system.html.

F. Toole, *Sound Reproduction, Loudspeakers and Rooms*, Focal Press, 2008.

D. B. Weems, *Great Sound Stereo Speaker Manual with Projects*, TAB Books 1990.

Source

H1456-08-ER18RNX Prestige Series woofer and the H1212-06 27TBFC/G tweeter
SEAS | www.seas.no

Variation on a Theme

Photo 1: The completed Variation transmission line speaker system enclosures are shown in 2.1 mode.

Build this Variation design, which is a classic transmission line loudspeaker system based on a proven commercial type. This design provides an alternative to a much more costly original system.

By
Ken Bird
(United States)

Composers like to take a musical theme from another composer and produce a musical variation of the original. Many great classical pieces have resulted from this semi-plagiarism. In the world of loudspeaker design, there are few designs developed by manufacturers and amateurs that do not reflect those that preceded the "new" system. The basic laws of physics and acoustics usually produce some similarities in all speaker system designs.

Such is the case with the Variation transmission line speaker system outlined below. Many will recognize this design's similarities to one that was priced at $10,000/pair a few years ago. And while it may have some physical attributes to that design, the materials and drivers are very different.

The Variation incorporates three major characteristics of a transmission line system. It has a lined, rather than fully stuffed tunnel, a tapered sound path, offset speakers, and damped chambers at each end (see **Photo 1**). These combined features eliminate the odd harmonics of the rear wave and produce a broad warm bass. Does the Variation produce $10,000 worth of sound? Placing a dollar value on reproduced sound is beyond my expertise, but I think the results are at least worth many times over the time and the money that will be spent on its construction. **Figure 1** shows the overall 3-D design concept drawing.

The Drivers

For drivers, I used 4" MCM 3 55-1856 woofers. They feature a 17 oz magnet with an 8 Ω, 1" aluminum voice coil, which has 86 dB sensitivity, 100 W RMS power handling, and a flat wide response from 65 Hz to 20 kHz. When used in parallel, it closely matches the 90-dB sensitivity of the Dayton tweeter. I also used Dayton Audio ND25FA-4 1" soft dome neodymium tweeters with 4-Ω voice coils. The woofers' 0.3 Q_{TS} met the low Q criteria for use in a transmission-type enclosure . After some listening tests, I also chose to use a 4,000 Hz, first-order crossover, which I devised using the WinSpeakerz crossover design program.

Construction

I used a Jasper router jig to cut the holes for the drivers, but they can also be cut using a jig saw. The hole dimension for the woofer is 3.66″ and for the tweeter, 1.77″. The hole for the speaker terminal cup is 3″.

Begin by consulting the drawing shown in **Figure 2** and cutting out the enclosure frame from the 1″ × 6″ dimensional lumber. Then, cut out the sub baffle front and enclosure sub top from a piece of 0.5″ one-side finished plywood. Next, round the sub top front and side edges with a router. As for the frame material, you can use either pine or hardwood. Using a compound miter saw will enable accurate cuts in the frame material. After cutting the holes for the terminal cup in the rear panel and the speakers holes, in both the front panel and the sub panel, nail the pieces together using 1.25″ finishing nails and liberal amounts of wood glue. Use corner clamps to hold the assembly while nailing. Pre-assemble the internal tunnel then attach it to the front panel (see **Photo 2**).

Once the frame is assembled, lay it on one of the 0.25″ side panels and carefully line it up with

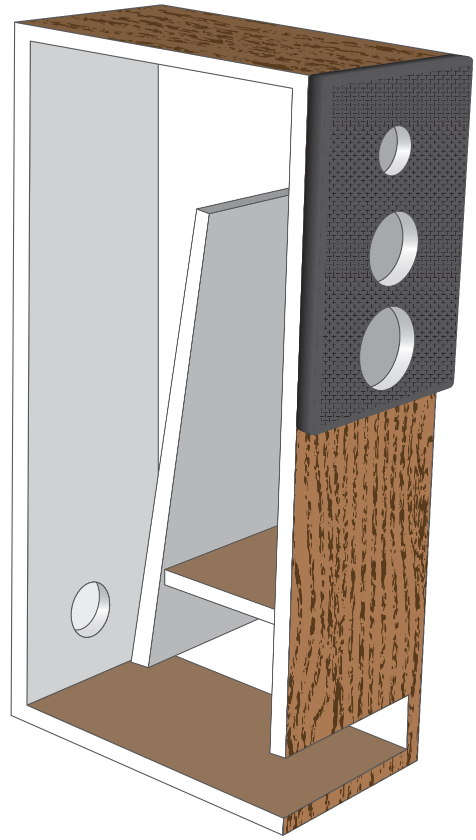


Figure 1: This is a 3-D concept of overall speaker design.

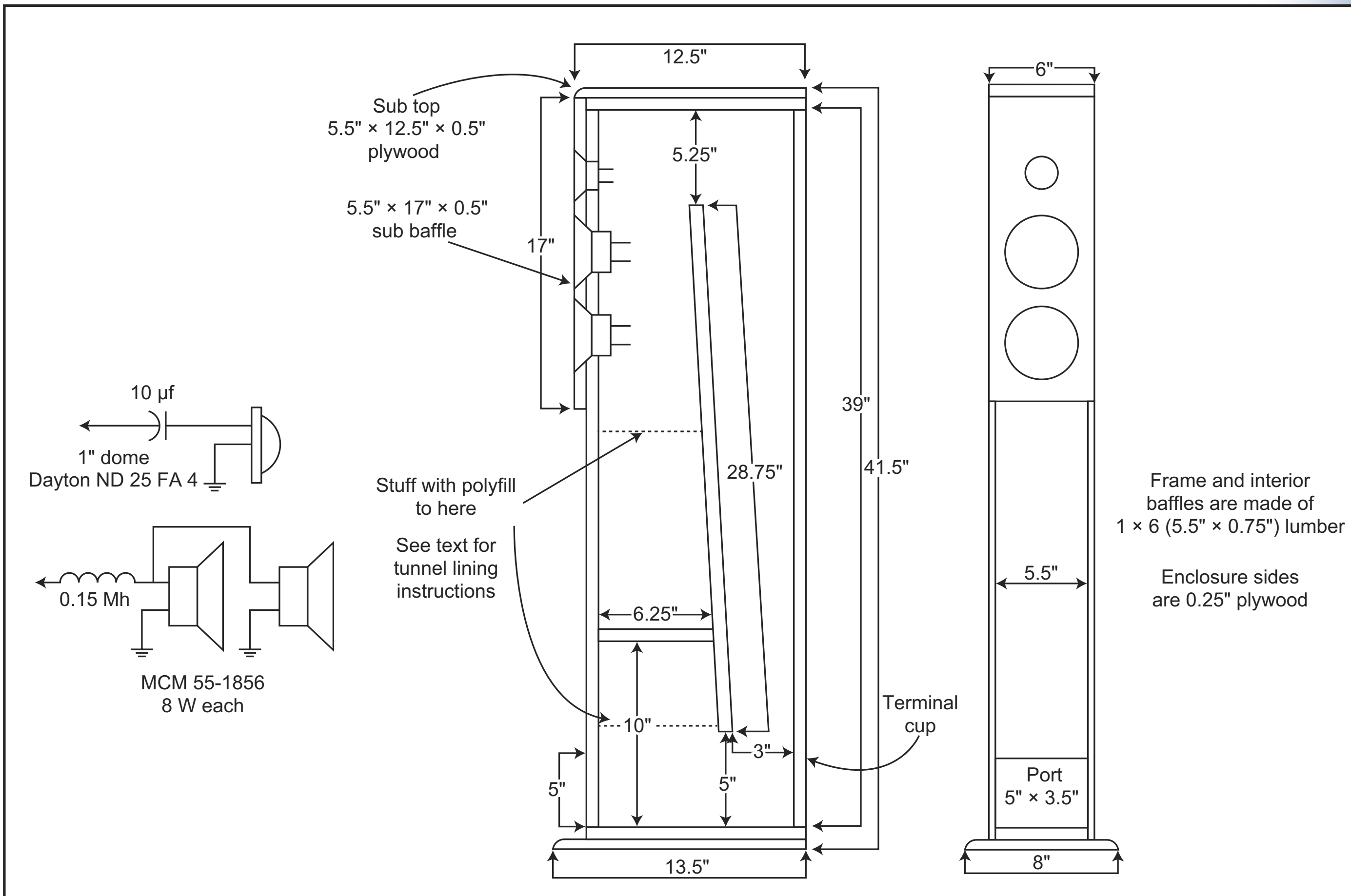


Figure 2: Follow the detailed enclosure drawing and schematic to build this classic transmission line system.

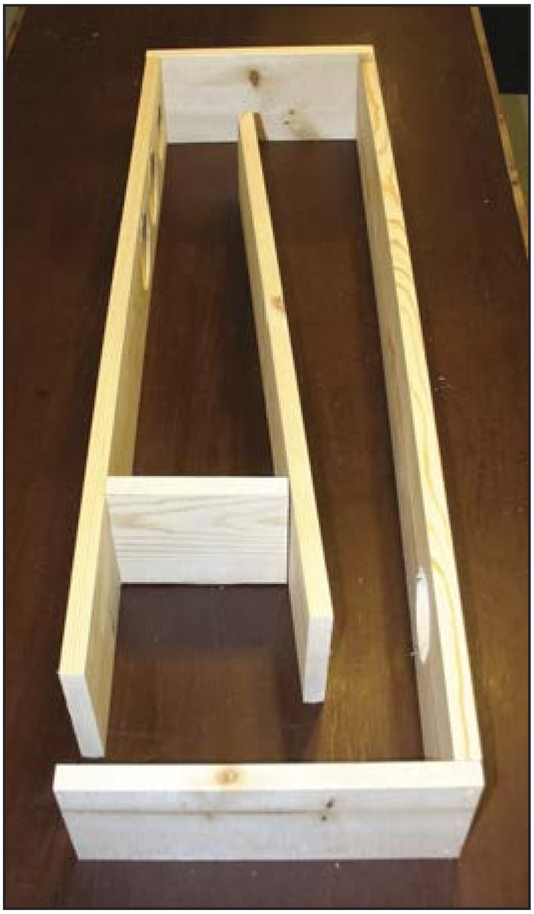


Photo 2: Construction of the enclosure frame.

the edges. Use a pencil to trace out the tunnel pattern on the inside of the panel. Remove the side and drill some 0.062″ (1/16) guide holes spaced 3″ apart in the center of the frame tracings. This will enable you to accurately nail the tunnel frame to the sides. Use wood glue and 1″ finishing nails to attach the side. Drive each nail in 0.062″ using a nail set. Fill the holes with wood filler when all the sides are assembled.

With one side in place, run three different color wires through the tunnel area from the terminal cup location to the space behind the speaker locations. One wire is for the woofer, one wire is for the tweeter, and there is one common wire. Staple the wire to the tunnel frame to prevent any rattle.

Caulking, Stuffing, and Lining the Tunnel

Caulk all the joints of the outer wall and the tunnel frame. Then line and stuff the tunnel. For stuffing, use ordinary polyfill material available from the local craft store. In this design, I used 6 oz of

Photo 4: The raw enclosures are ready for stain.

Photo 3: Stuffing and lining the enclosure.

polyfill—3 oz in the area below the speaker and 3 oz in the chamber near the port. Next, staple a piece

Photo 5: The drivers used in the system are shown with the terminal cup. Note the crossover components mounted on the cup.

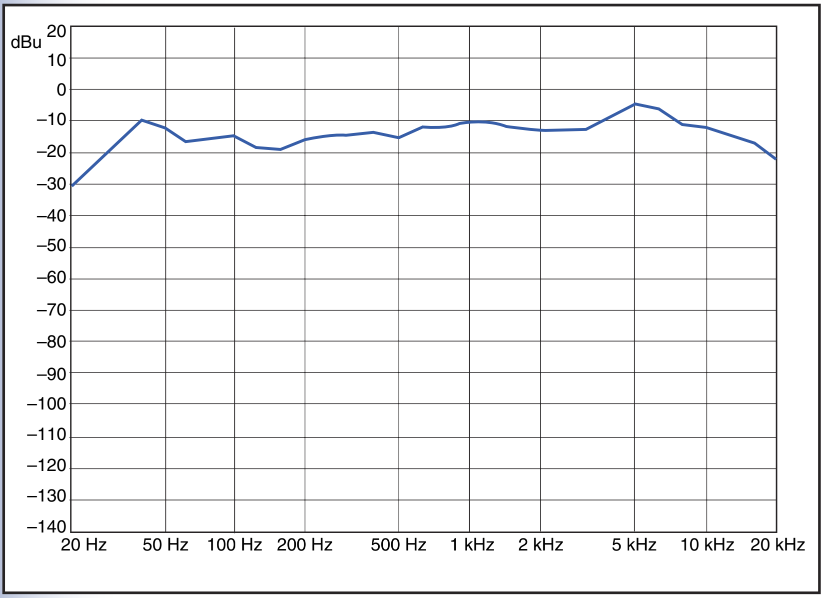


Figure 3: The True Audio RTA records the new system at a 1/3 octave in-room response with a flat EQ.

About the Author

Ken Bird has had a 40-year engineering and product management career focusing on consumer, industrial, and telecommunications electronics. Ken retired to his native Missouri and now pursues his interest in building speakers and tube amplifiers, along with model railroading, amateur radio (WOKLB), and digital photography.

of fiberglass screen wire below the lower chamber to hold the polyfill in place.

The tunnel lining can be low-cost indoor-outdoor carpet, which can be found at many home improvement stores. The 0.25″ material can be easily cut with a carton knife and glued to the inside of the tunnel walls with craft glue.

Once the tunnel walls have been lined with the carpet, line the side wall before you attach it. The trick to locating the right lining position is to lay masking tape face up on the edge of the tunnel frame. Then, carefully lay the side wall down on the frame. When you lift the frame, the tape will stick to the plywood and outline the area where you need to glue the carpet lining. It can also serve as a template for where to drill holes to attach the side final panel (see **Photo 3**). Before attaching the side panel, run a bead of caulking along each surface of the frame and tunnel. This will provide a good seal. Then, lay the panel on the frame and nail it in place. Stuff wadded newspapers into the cabinet holes and then sand the entire exterior (see **Photo 4**).

I trimmed the lower front of enclosure with 0.75″ screen door lath. The enclosure's finish is the builder's choice. For my design, I used a traditional dark walnut stain and covered it with two coats of gloss polyurethane. I painted the speaker sub baffles gloss black, but you could also give them a wood finish. I chose not to use speaker grilles as they complicate the design and can be problematic with the system's high-frequency response. As a safety measure, I made base plates from 0.5″ plywood. The base plates should be 13.5″ × 8″ and attached to the speaker bottoms by nails or screws.

Installing the Crossover and Speakers

Use a hot glue gun to install the crossover coil and the capacitor on the rear of the speaker terminal cup. A "Christmas tree" style terminal served as the wire terminations (see **Photo 5**). Next, attach the common and the two separate wires to the tree terminal. An alternate crossover mounting is possible directly behind the speakers on the tunnel board. Then, wire the common lead to the negative leads of the speakers and the separate leads to the tweeter. Wire the woofers in parallel. The woofer gaskets can be made from 0.25″ foam insulation. Apply self-stick strips to the rear edge of the woofer frame. This is necessary since the speaker frame is of the "pin cushion" type and the foam provides a good seal at the edges of the speaker holes. Use 0.75″ number 2 sheet metal screws to attach the speakers. Then, insert 0.5″ fiber washers under the frame edge at each screw location. This prevents the pin-cushion-style frame from being bent down by the screws.

Parts List

Electronics
4 MCM 55-1856 4″ woofers
2 Dayton 275 059 tweeters
2 10 capacitors, non-polarized. . . . . .Part-Express # 027-340
2 Inductors 0.15 mH . . . . .Parts Express #255-022
1 Speaker terminal cup . . . . .Part Express 260-293
16′. . . . . .18-gauge wire, (black for common wire, red for woofer, and green for tweeter)

Lumber
2 Speaker cabinet backs . . . . .1″ × 6″ × 39″
2 Speaker cabinet fronts . . . . .1″ × 6″ × 34″
4 Top and bottom pieces . . . . .1″ × 6″ × 12″
2 Plywood sub tops (0.5″). . . . . .6″ × 12.5″
2 Plywood bottom bases (0.5″) . . . . .8″ × 13.5
2 Front sub baffles (0.5″) . . . . .6″ × 17″
4 Plywood for sides (0.25″). . . . . .12″ × 40.5″

Miscellaneous
Finishing nails . . . . .1″ and 1.5″
Wood glue. .1 bottle
Silicone window caulk . . . . .2 tubes

Stuffing and Tunnel Lining
Indoor-outdoor carpet . . . . .1 yd
Polyfill pillow stuffing . . . . .12 oz

Testing

Outside of subjective listening, it is difficult to get a speaker system's scientific measurement without an anechoic chamber. However, I used the True Audio RTA Program and recorded the room response (see **Figure 3**). Positioned 15″ from the wall, the system has an excellent bass output and superb midrange imaging and does much justice to my preferred pipe organ, orchestral, and choral music. It is also quite nice with other forms of music, including heavy metal, and it does not cost $10,000.

Resource

V. Dickason, *Loudspeaker Design Cookbook*, 7th Edition, Audio Amateur Publications, 2005, www.cc-webshop.com/Loudspeaker-Design-Cookbook-AA-BKAA068.htm.

Sources

MCM 55-1856 cone woofer
MCM Electronics | www.mcmelectronics.com

Dayton Audio ND25FA-4 tweeter
Parts Express | www.parts-express.com

WinSpeakerz and True RTA
True Audio | www.trueaudio.com

Questions & Answers

Insight into a Legendary Industry Designer

An Interview with John Curl

John Curl is one of the most respected circuit designers of all time and the creative genius behind Parasound's high end audio and home theater amplifiers. He is responsible for the classic Mark Levinson JC-2, the SOTA head amplifier, and his own Vendetta preamplifier. (Photo courtesy of Anatoliy Lisovskiy)

By
Shannon Becker
(United States)

SHANNON BECKER: How did you become interested in audio electronics?

JOHN CURL: Before I discovered audio electronics, I studied engineering-physics and I also studied guitar. I found that I could discern small but significant differences in musical instruments, such as acoustic guitars, better than most people. Then, I realized that audio was a way to do engineering, without building weapons or some such thing. Instead, I could help people by providing audio reproduction.

SHANNON: Tell us about your early years.

JOHN: My first real experience in audio was getting a part-time job at a high-end audio store in 1965. I was fortunate to work for the audio engineer who owned the store, and his idealism matched mine. We sold, installed, and repaired products made by Klipsch, McIntosh, Marantz, Fisher, Ortofon, and others —the best of that time. I wasn't a real designer yet, but I was finishing my degree in engineering-physics, and this job introduced me to people in the business—people like Paul Klipsch and Saul Marantz—and exposed me to the best commercial products of that time and even "better sounding" amps, such as triodes, which had fallen out of favor in 1965.

When I finished college in 1966, I was immediately employed in aerospace (not bombs, thank goodness), and my professional career began. I also joined the Institute of Electrical and Electronics Engineers (IEEE) and the Audio Engineering Society (AES). It took me several years to work up to even being an "engineer." Mostly, I helped senior engineers with their projects. It was frustrating at first, but I finally got to the point where I could design my own amplifier circuits.

SHANNON: How did you get involved with the American rock band the Grateful Dead?

JOHN: While working at Ampex in the late 1960s, doing both audio and video work, I befriended a guy who was equally interested in audio problems. He was the person who eventually hired me to work with the Grateful Dead, starting in 1970. It was quite a change in lifestyle, but I had several years of experience in formal engineering design so I was able to contribute something useful.

Our biggest problem was the sound quality of solid-state designs—commercial and custom. We came to realize that we needed to better understand what it took to make a good sound amplifier. I decided that Matti Otala [a Finnish college professor and engineer known for his work on a distortion mechanism that he called transient intermodulation] had some good ideas on the subject. I tried using some of his ideas in my custom designs and was finally successful after a couple years. This led to other opportunities, including designing high-end electrics with Mark Levinson, starting in 1973.

SHANNON: Can you tell us about your current work?

JOHN: I have been almost exclusively working as a design consultant with various audio companies for the last 25 years. I find that it best fits my temperament. Oh, I tried being president of my own company, vice president of a larger one, and even a senior design engineer for an even larger company, but my best fit is design consultant, without having to run the company administration.

I was introduced to Richard Schram [founder of Parasound] in 1989 and have designed all of Parasound's high-power amplifiers and consulted on the design of the low-level circuits of many of its other components. My fit with Parasound over this period has been one of the easiest because so much of the effort in the layout, manufacture, and even incidental circuitry (such as blinking lights, etc.) are done by others. Having these circuits built in Taiwan saves a great deal of money. This means we can offer great bang for the buck in comparison to my more intimate designs, in which I have control of almost everything and don't worry about the cost. For example, the Vendetta Research products that I made in the 1980s and early 1990s.

SHANNON: What design are you most proud of?

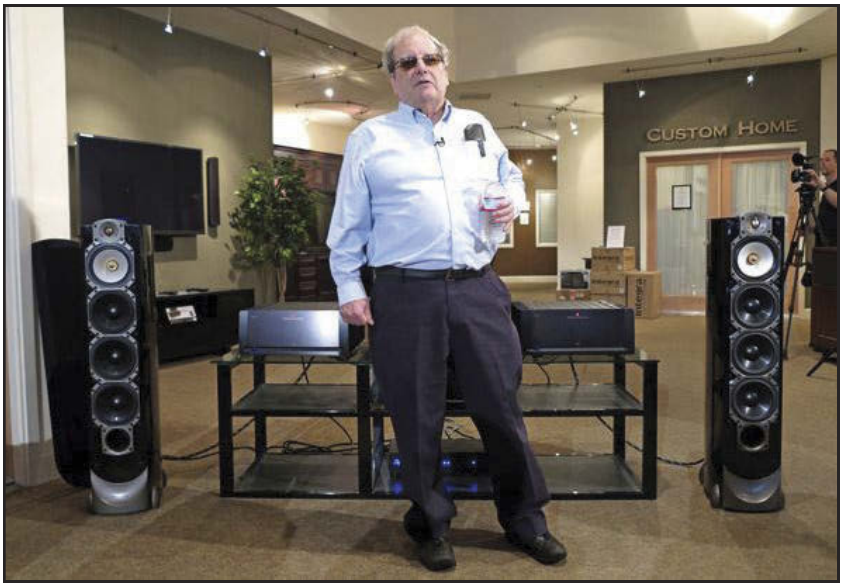
JOHN: The design that I am most proud of is the Vendetta Research SCP-2 phono preamp. I still use one today in conjunction with the CTC Blowtorch preamp that is my next best design.

When I am in total charge of a project, very little gets by me. It is almost a compulsion to get it virtually perfect. It is *always* more expensive than what I can do for Parasound, but there is usually a sonic difference.

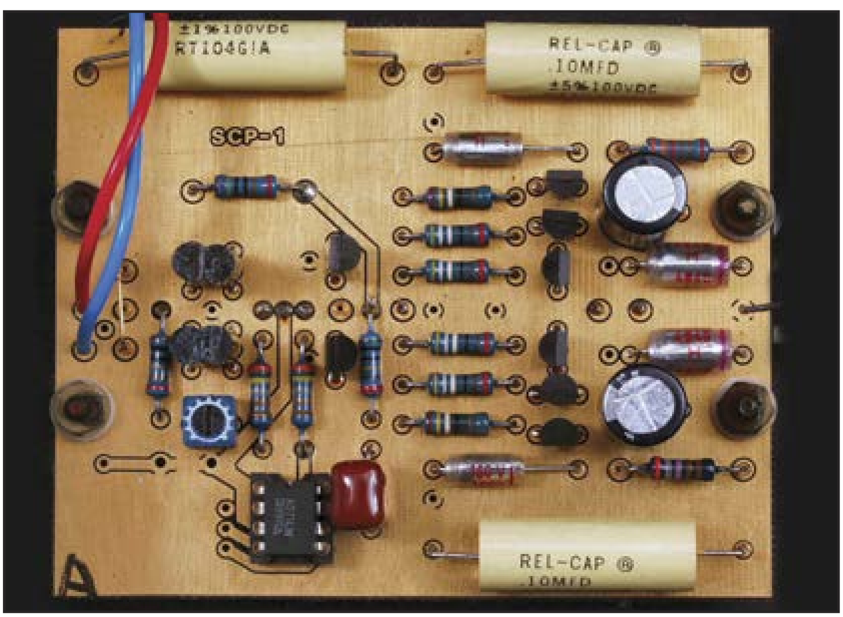
SHANNON: Can you share some thoughts about your former company, Vendetta Research?

JOHN: Vendetta Research started as an inside joke based on complaints I had with certain former employers, who berated me and refused to pay me the royalties that they promised. When we decided to make it a real company, we chose the name because

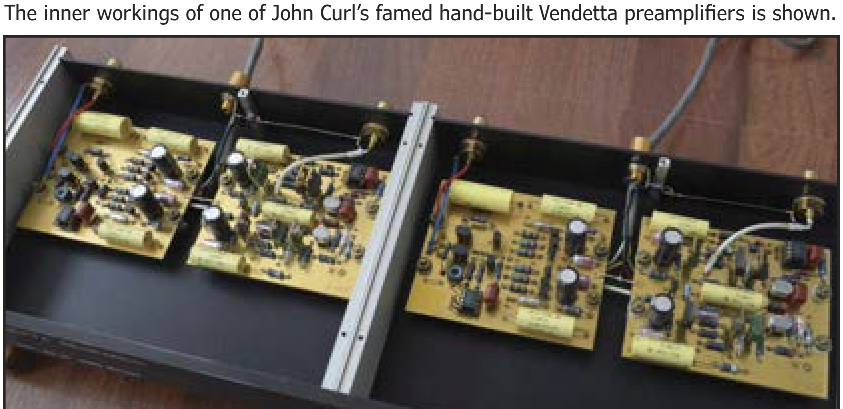
John Curl shows Nelson Pass a very old design from 1968.



In July 2014, John Curl discussed amplifiers during a presentation he attended at Century Stereo in San Jose, CA, with Parasound founder Richard Schram.



The inner workings of one of John Curl's famed hand-built Vendetta preamplifiers is shown.



This is one of John Curl's complete Vendetta preamps.

we knew we would not have any competition for the name when it came to registering it.

I spent four years making Vendetta Research phono stages. While the product was successful, it did not make me much of an income or give me much peace of mind because I also had to worry about taxes, employees, and other administrative tasks almost all of the time, so real audio design took a back seat.

During that time, we made hundreds of SCP-2s, but I don't know exactly how many. When people moved away from phono reproduction and a firestorm destroyed my inventory, I quietly closed shop. By then, I was already working for two other companies as a design consultant and found it much more pleasant.

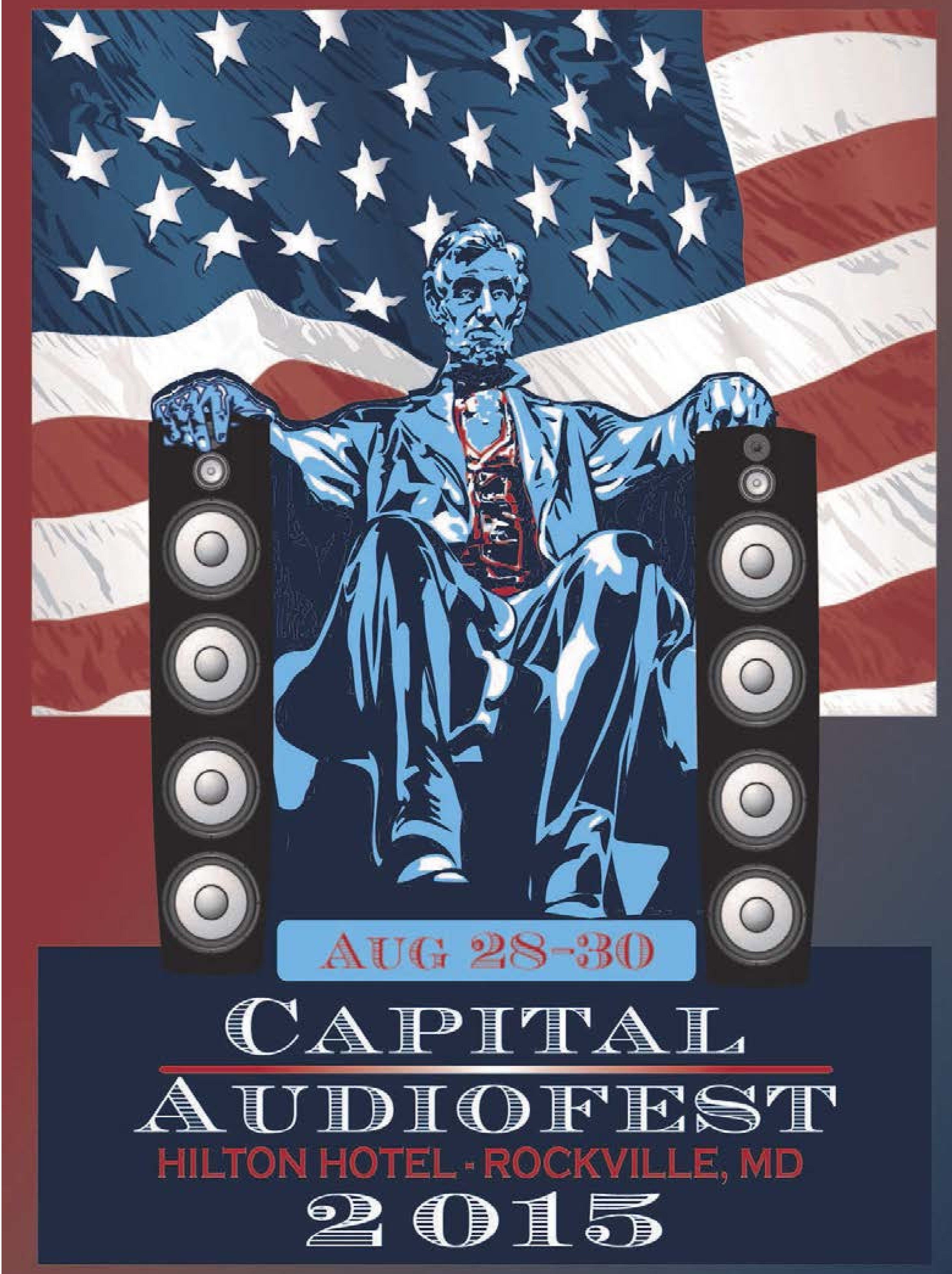
I have been making amps and preamps for more than 40 years. Most of my designs are based on what I have found to "work well" and what does not work as well. I do use several "specified" components that I have found over the decades including polystyrene capacitors, certain brands of 1% resistors (not always the most expensive), and if possible complementary JFETs. I examine everything and use only the best possible component materials, especially if I am in control of the entire fabrication. I get the best results when I don't compromise.

SHANNON: What is it about your amplifier designs that give them such longevity in the audio market?

JOHN: They are usually very similar in topology. Each is adapted for the specific function that it has to do, but they are like members of a family and have many obvious similarities. I try to make them balanced, complementary, and fast. I use JFETs where possible, and Class A when I can. However, I have found that other circuit details, such as solder type, circuit board material, capacitors, and even wires and connectors contribute to the overall sound. This is what makes my best designs excel over my cost-effective designs. I don't have a specific preference for any new audio technology. I keep hoping for improvements in digital reproduction, and they are slowly (slower than I had hoped) coming.

SHANNON: What do you see as the most challenging aspect that lay ahead?

JOHN: I think it is getting digital playback really better than the best analog playback, including vinyl and analog master tapes. I don't think we really understand how we actually hear audio differences and I can only suggest to people that they "use their ears" as the final evaluator, even if that is scoffed at by some academics. One's perception of sound quality is, like the taste of fine wine, the final arbitrator.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Design Hollow-State Amplifiers Using Pentodes

Learn how to use pentodes in your hollow-state amplifier designs and achieve a pleasing audio performance.

By
Richard Honeycutt
(United States)

In the recent resurgence of interest in hollow-state audio, triodes have captured the limelight. This is primarily true because of triodes' distortion signature, which consists mainly of second harmonics with a bit of third. The pentode's significant percentage of higher harmonics creates a more noticeably distorted sound, which can also be more annoying. All this is understandable if you remember that the difference between the sound of a trumpet playing C_5 and a clarinet playing C_5 is the harmonic content.

Altering the harmonic content thus changes the timbre of music, but does not insert new notes. Added high harmonics are characteristic of "buzzy" instruments such as lower bagpipe notes and some styles of saxophone playing. These timbres are more likely to be annoying than the purer sounds of the flute, the clarinet, and the classical trumpet.

However, pentodes have certain characteristics that make them more advantageous for certain purposes, and if pentode amplifiers are carefully designed and care is taken not to overdrive them, very acceptable audio performance can be achieved.

The Development of Pentodes

The most common use of pentodes in audio electronics is in push-pull output stages. In fact, most hollow-state power amplifiers use either power pentodes or beam-power tetrodes (sometimes denoted as beam power pentodes) for the output stage. However, for the present, we will concentrate on pentode-based preamplifiers.

The limiting factor in the high-frequency response of a triode is the grid-to-plate capacitance, which provides increasing amounts of negative feedback as the frequency is increased. One way to reduce the grid-to-plate capacitance is to insert an additional grid between the control grid and the plate. This screen grid is grounded through a fairly large capacitor, from an AC point of view, thereby shielding the grid from the plate, and breaking up the grid-to-plate capacitance.

A tetrode may have a control-grid-to-plate capacitance of about 0.01 pF, compared to a few picofarads for a triode. DC-wise, the screen grid is biased at a high positive voltage, near or equal to the plate voltage. Thus, the screen grid attracts

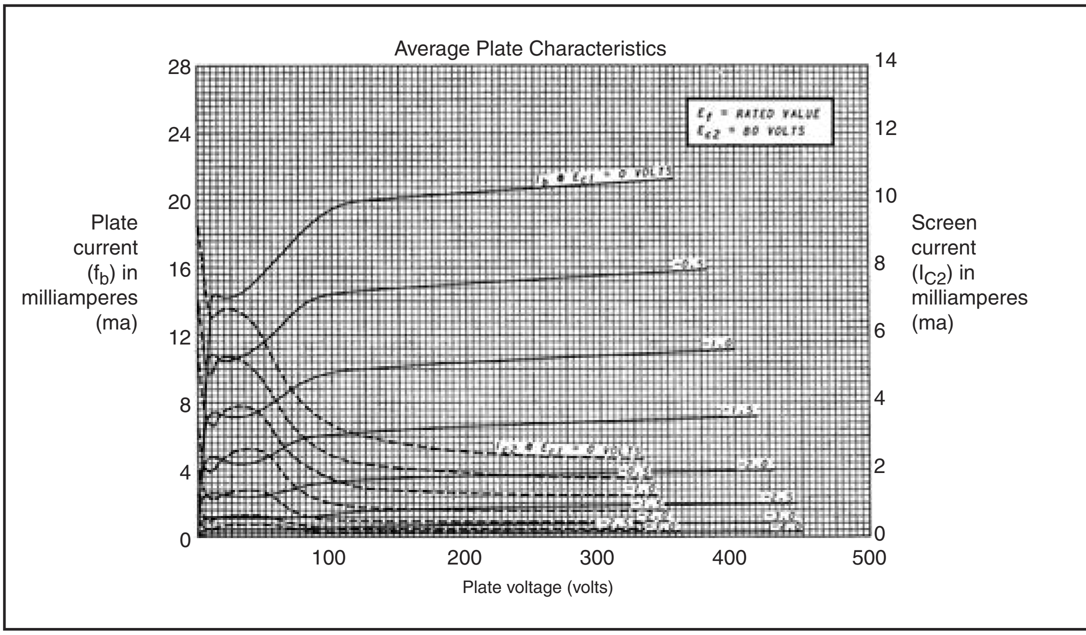

Figure 1: This plate characteristic for a 6CY5 tetrode shows the kink (look at plate voltages below about 40 V).

electrons that have passed through the control grid, accelerating them on their way to the plate. Since the wires of the screen grid are widely spaced, most of the electrons pass through it to the plate, after being accelerated by the screen grid's high positive voltage. Under these conditions, the plate current depends primarily upon the screen-grid voltage, and only slightly upon the plate voltage. A tube built this way has four active internal elements, and is called a tetrode. Tetrodes can have much higher voltage amplification than can triodes.

A problem unique to tetrodes is a "kink" in the plate characteristic curve, in which for a small range of operating conditions, increasing plate voltage is accompanied by decreasing plate current. This negative resistance or "kink" (see **Figure 1**) makes a tetrode subject to undesired oscillations. The kink is produced by secondary emission: electrons that strike the plate with enough energy to rebound, producing a secondary space-charge (cloud of electrons) between the screen grid and the plate.

Eliminating the kink is accomplished by adding a "suppressor grid" between the plate and the screen grid. The suppressor grid is usually connected to the cathode. Thus, it repels the secondary-emission electrons back to the plate.

Types of Pentodes

Pentodes are classed as either "sharp-cutoff" or "remote-cutoff." Sharp-cutoff pentodes have a plate characteristic that is as straight as possible, resulting from the use of uniformly-spaced control-grid wires. Remote-cutoff pentodes require a strongly negative control-grid voltage to cut off the plate current. **Figure 2** shows the plate-current curves for both kinds of pentodes. Thus sharp-cutoff pentodes are preferred for audio amplifiers, since the straight plate characteristic correlates with lower distortion; whereas, remote cutoff pentodes serve as automatic gain control (AGC) amplifiers.

The effect of screen-grid bias voltage on the plate characteristic curves is to move each entire curve upward (more plate current for a given control-grid bias and plate voltage) as screen-grid bias is increased. This effect allows the screen grid to be used as a gain control terminal in dynamics processors (e.g., compressors, expanders, etc.).

Although the shape of each characteristic curve is not changed by varying the screen-grid bias, screen voltage has an effect upon distortion, gain, and maximum output voltage. A higher screen voltage gives a higher gain and increases the maximum output voltage. For example, increasing a 6AU6A pentode's screen voltage from 100 to 150 V, while keeping the plate voltage at 250 V and the control-grid voltage at –1 V, increases the transconductance from 2,800 to 5,600 µS.

The suppressor grid bias voltage does have an effect on plate current, but it is small and usually ignored. In fact, most "pentode models" for the simulation software PSpice are actually tetrode models. Indeed, in some pentodes, the suppressor

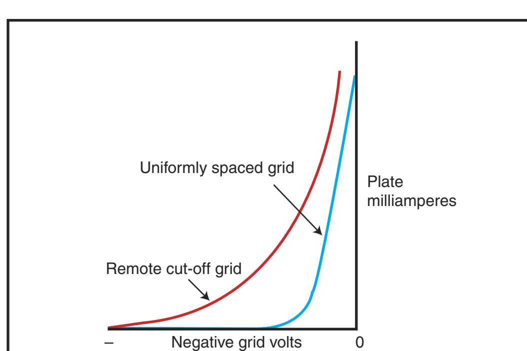

Figure 2: The difference in shape of the plate characteristics makes sharp-cutoff pentodes more suitable for low-distortion applications, and remote-cutoff pentodes more suitable for AGC, compressors, expanders, and so forth.

grid is internally connected to the cathode, ensuring that the suppressor-grid bias is always zero.

Readers interested in more mathematical depth may want to consult *Fundamental Amplifier Techniques with Electron Tubes*, by Rudolph Moers, or read Norman L. Koren's article "Improved Vacuum Tube Models for SPICE Simulations: Part 1: Models and Examples" (see Resources).

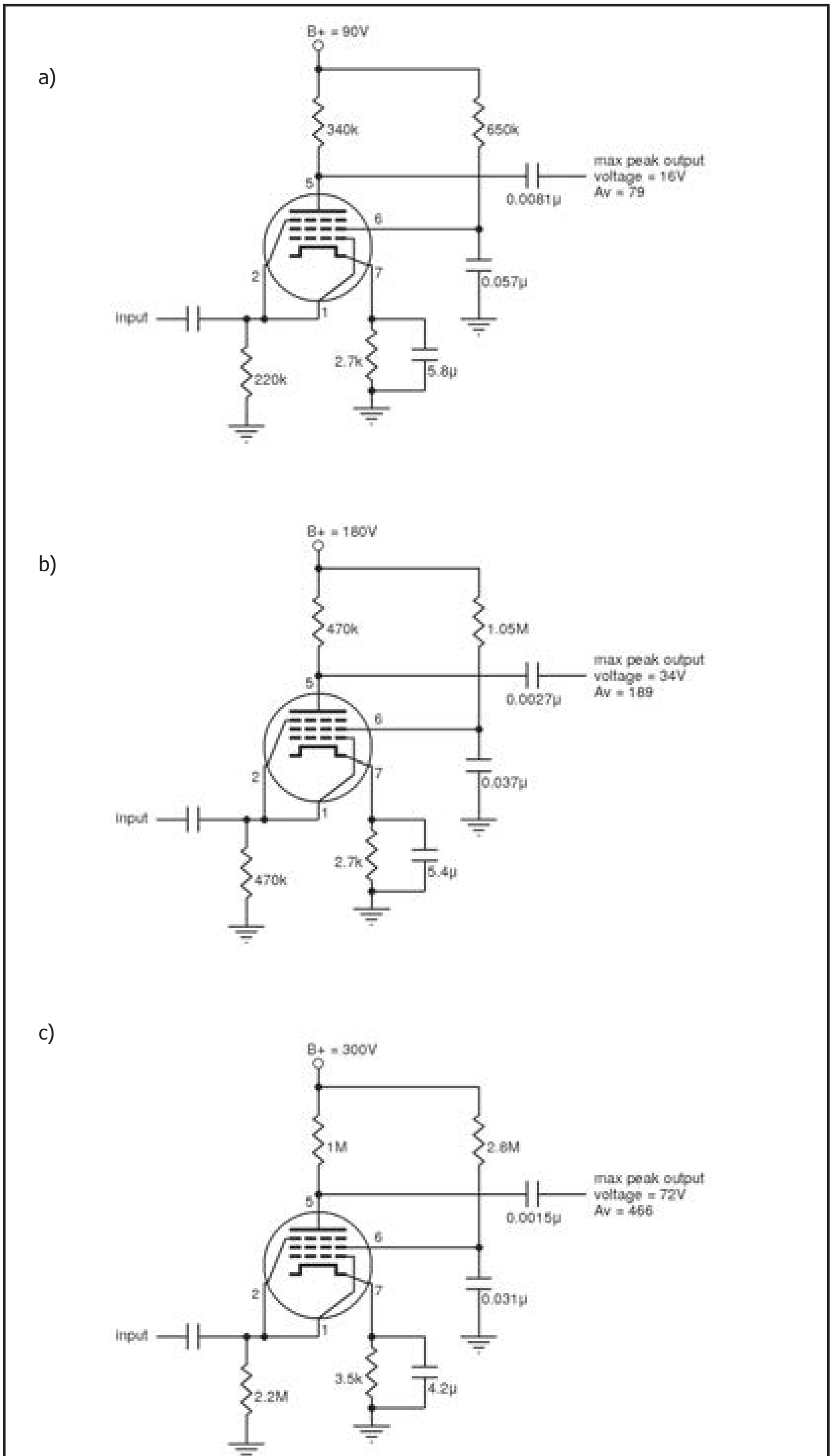

Figure 3: These three circuits illustrate the wide range of gain and maximum output voltage available from a single pentode amplifier stage.

Figure 3 shows three small-signal amplifiers using 6AU6A pentodes, and their corresponding performance characteristics. Note that the voltage gain of the stage can be much higher than is available from a single triode stage. I should mention that the values of the output capacitors, cathode-bypass capacitors, and screen-grid bypass capacitors are selected to give a −3 dB low cut-off frequency of 100 Hz. For a different low cutoff frequency ($f_{desired}$), each capacitor value should be multiplied by the factor ($100/f_{desired}$).

The gain and maximum output voltage values are predicated upon the output feeding an identical following stage, or some other load whose resistance is equal to the control-grid resistor shown in the circuit. These values are based upon the design tables in the *RCA Receiving Tube Manual*, and may be somewhat optimistic, based on my past experience, and the experience of other designers. However, they do provide a basis for preliminary understanding of the performance of pentode amplifier stages.

Commercial Pentode Preamp Stages

It is interesting to look at some commercial pentode preamp stages. For example, the circuit shown in **Figure 4** is from a small Silvertone guitar amplifier. The 12AU6 was grid-leak biased, and the hot-chassis circuit only provided 120 V for the 50C5 beam power tube. The B+ was dropped to 75 V for the pentode preamp. A screen-dropping resistor further lowered the DC to 26 V for the screen grid. The low plate and screen-grid voltages provided a nice distortion sound that was generally hated because it did not match the musical style of the time. When these amps were rediscovered in the 1970s, they were prized as practice amps, precisely because of this distortion.

Although the B+ supply fed the 35W4 half-wave rectifier directly from the power line, the circuit did use a filament transformer to supply 12.6 VAC for the 12AU6. While it would have been possible to connect all three tube filaments in series instead of only the 50C5 and 35W4, since all three tubes require the same filament current (0.15 A), better hum and noise performance was obtained by using the filament transformer. When this amplifier was built, AC power-line voltage in the US was specified as 115 ±10 VAC, although sometimes, particularly during peak-load situations, the voltage would drop lower. Therefore, it was not uncommon for electronic

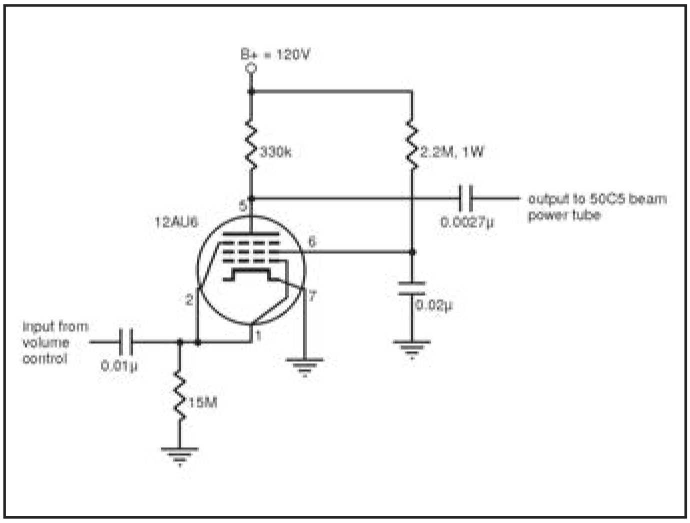

Figure 4: This pentode preamp stage was used in the 1950s-vintage Silvertone Model 1430 guitar amplifier.

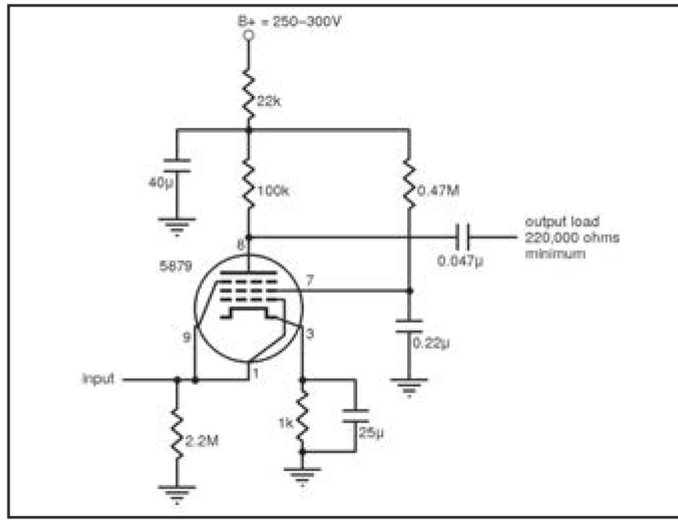

Figure 5: This pentode-based microphone preamp was designed by RCA.

designers to design equipment to operate with the power mains as low as 95 VAC. Indeed, when you add together 50 V for the 50C5, 35 V for the 35W4, and 0.15A × 100 Ω = 15 V for the dropping resistor, you get 95 V.

Figure 5 shows a pentode-based microphone preamp circuit from the 1975 *RCA Receiving Tube Manual*. The extremely high input impedance of 2.2 MΩ would have worked well with crystal or ceramic microphones, which were ubiquitous at the time, due to their low cost. The requirement of a very high load resistance meant that this circuit would have probably been followed by another tube amplifier stage, perhaps a cathode follower.

The circuit used a 5879 pentode, which was designed for this sort of application. Design-center input voltage was 3 mV RMS, at which level the output voltage was 220 mV RMS: a voltage gain of about 73. The extremely high input impedance required a shielded input lead to avoid hum, noise, and AM radio pickup.

Figure 6 shows a pentode-based microphone preamp designed by Mullard, using its EF86 pentode tube. This tube was designed for low noise and microphonics. Pins 2 and 7 connect to an internal shield. Tubes typically used in sensitive preamp circuits are fitted with an external shield that slides over the tube envelope. In either case, the shield is grounded, effectively enveloping the electrodes in a Faraday cage. The Amperex 6267 or the Mazda 6F22 are essentially equivalent tubes.

The tube is grid-leak biased to provide the high input resistance needed by crystal microphones. Either this circuit or the RCA design shown in **Figure 5** can be adapted to lower-impedance microphones by the use of an input transformer, which also provides some essentially noise-free gain.

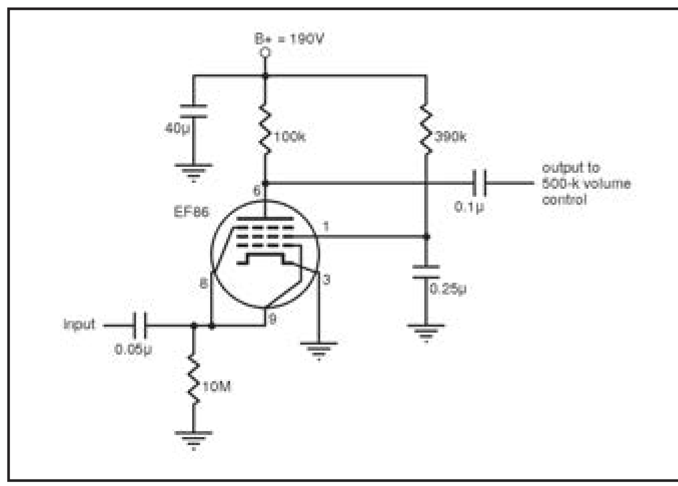

Figure 6: Here is a pentode-based microphone preamp designed by Mullard.

Pentode Use

In summary, pentodes are a viable choice for preamplifier tubes when a high voltage gain and/or high peak output voltage is needed, and when the input voltage is restricted so that overdriving is not a problem, or the harsher "pentode-type" distortion is desired.

Resources

N.L. Koren, "Improved Vacuum Tube Models for SPICE Simulations Part 1: Models and Examples," www.normankoren.com/Audio/Tubemodspice_article.html.

R, Moers, *Fundamental Amplifier Techniques with Electron Tubes,* Elektor, 2011.

RCA Receiving Tube Manual, RCA Corp., 1975, available at www.cc-webshop.com.

Industry Calendar

Here are a few places where you might find a copy of *audioXpress* or possibly meet one of our authors and staff members:

September 10–13, 2015 — Prolight+Sound NAMM Russia 2015

Sokolniki Exhibition and Convention Centre, Moscow, Russia
www.namm-musikmesse.ru
www.prolight-namm.ru

A joint-venture between the National Association of Music Merchants (NAMM) and the Frankfurt Messe, NAMM Musikmesse Russia and Prolight+Sound NAMM Russia is becoming a significant effort to reach the eastern countries and gain a foothold in the massive Russian market.

Russia's only international music-trade exhibition and its biggest international trade fair for professional audio and event production technology will return to Moscow's Sokolniki Exhibition and Convention Centre, following a reasonably successful event in 2014. The fourth edition of NAMM Musikmesse Russia and Prolight+Sound NAMM Russia will be visited by Russian and CIS music product retailers, distributors, educators, theater and production professionals, and musicians. Globally known brands along with public and private supporters are already on board for the September event, indicating that NAMM Musikmesse Russia and Prolight+Sound NAMM Russia 2015 are on track to be the largest gathering yet.

Since the Moscow shows' inception in 2012, nearly 30,000 Muscovites and international visitors have come to see, hear, and demonstrate products from more than 500 companies and 25 countries. In 2014, the shows' exhibitor square footage increased 40% and the number of exhibitors and visitors expanded by 42% over 2013. The 2015 shows will be presented in conjunction with the Moscow Ministry of Culture and of the Russian State Duma have the endorsement of The Moscow Ministry of Culture, the Moscow Conservatory, the Russian Guild of Theatre Managers, the Russian Union of Theatre Workers, the Russian state musical TV center "Orpheus," the Association of Musical Theatres, and many other public and private organizations.

NAMM Musikmesse Russia will fill two pavilions in 2015. One for electronic music products and instruments, including synthesizers and electronic keyboards and the Pavilion Classique is devoted entirely to classical and acoustic music products and instruments. Held September 10-12, Prolight + Sound NAMM Russia will spread out across three pavilions at Sokolniki Exhibition Centre. Building on the enthusiasm of last year's show for performance technologies, one entire pavilion will be devoted to theater production. Additional Prolight+Sound NAMM Russia pavilions will hold forums on rental and event management, radio-industry summits, educational seminars in lighting, media and audio technologies, and demonstrations of concert sound installations.

September 10–15, 2015 — IBC 2015

IBC 2015 Conference—September 10-14, 2015
IBC 2015 Exhibition—September 11-15, 2015
www.ibc.org

The world's leading Electronic Media and Entertainment Event continues to get bigger and better every year and IBC 2015 is expected to once again attract large numbers of visitors and exhibitors. Recognized as the most international event for the electronic media and entertainment industry—in parallel with the National Association of Broadcasters (NAB) convention held in Vegas every April—IBC takes place in Amsterdam in September every year. Multiple initiatives are turning the IBC convention increasingly attractive to the industry, focusing on converging media technologies and business models in Broadcast, IT and Telecoms—increasingly in the management and multiplatform disciplines, of all forms of electronic media content.

Audio exhibits, most of them concentrated in Hall 8, include a unique combination of radio technology suppliers, microphone manufacturers, recording and production solutions, including ENG systems and many unique companies in the areas of communication and wireless transmission, apart from important studio technology manufacturers. For those interested in learning about the latest trends in audio streaming and digital radio, including multichannel and binaural distribution, IBC 2015 is a valuable opportunity.

Over 1,700 key international suppliers provide an unrivaled environment for over 55,000 attendees from more than 170 countries to connect across countries, technologies and industry sectors. For 2015, the newly constructed Amtrium will be a stunning addition to the 14 Halls of exhibits. IBC will also be introducing the IBC Launch Pad, for first-time IBC exhibitors who will be shining a spotlight on the mega-trend toward IP based infrastructure and software tools which are transforming the industry. IBC 2015 will also be deploying Touch & Connect technology: a valuable networking tool which will be free to all visitors, enabling them to instantly acquire and exchange contact information and exclusive content. Complementing the Conference and Exhibition, there will be Industry Insights conference sessions; IBC Content Everywhere Europe conferences; the Future Zone exhibits; and the IBC Awards Ceremony. Focusing on the most recent developments in digital cinema, the IBC Big Screen Experience will take place in the RAI's auditorium, which will be equipped with the latest cutting-edge cinema technology including Christie 6P 2-D and 3-D laser projection and Dolby Atmos immersive audio.

October 2–4, 2015 — Rocky Mountain Audio Fest (RMAF)

Marriott Denver, Denver Colorado
www.audiofest.net
www.audiofest.info

This will be the 12th edition of the annual Rocky Mountain Audio Fest (RMAF), the largest high-end consumer audio show in the US, featuring approximately 200 exhibitors. From affordable audio systems to ultra-expensive high-end gear and CanJam at RMAF—a show within a show organized by Head-Fi.org—featuring the latest in headphone technology, this is certainly a great show for product introductions away from the noise of other CE-oriented tradeshows. The RMAF also features live entertainment and informative seminars, equipment show specials, and prize drawings throughout the weekend.


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