An Interview with Audio Icon Walt Jung, p. 32

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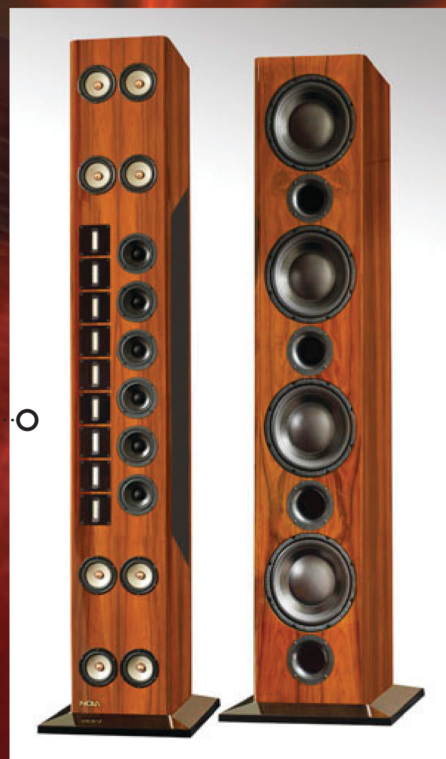

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From the Editor's Desk

Expert Reports & Projects

The redesigned AudioAmateur.com website is now up and running. As you'll see, it is a more modern, user-friendly website than its predecessor. You can access your digital subscription to *audioXpress* as well as read feature articles, interviews, member profiles, and news. I encourage everyone to bookmark the site and visit each day. Stay informed!

Now let's focus on this issue.

Did you attend the 2012 Munich High End Audio Fair? If not, no worries. Ward Maas walked the floor and took copious notes on interesting audio products and systems (p. 12).

In the first part of his series on electrostatic speakers (*audioXpress* 9/12), Richard Mains provided tips for selecting the proper materials for an electrostatic loudspeaker (ESL) project. This month he covers the fabrication process (p. 22).

Turn to page 26 for a thoughtful product review. Gary Galo provides an analysis of Monarchy Audio's SE-100 MK2 and SM-70 amplifiers.

This month's interview is fascinating (p. 32). Walt Jung's interesting life and audio-related work will inspire you to learn, design, and write.

On page 8, Mike Klasco and Steve Tatarunis describe what's behind a speaker cone. They detail the functionality of speaker stuffing.

Are you engaged in the "sound system debate"? On page 18, Richard Honeycutt investigates the differences between hollow-state and solid-state amps.

The last article in this issue is a thorough review of B&C's new 18" woofer (p. 38). According to Vance Dickason, "the 18TBW100 uses a ferrite motor design, which is probably going to be the trend in pro woofer design until neo's prices drop to a more competitive level."

Regards,

C. J. Abate
editor@audioxpress.com

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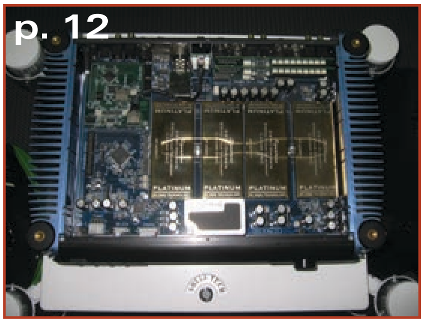

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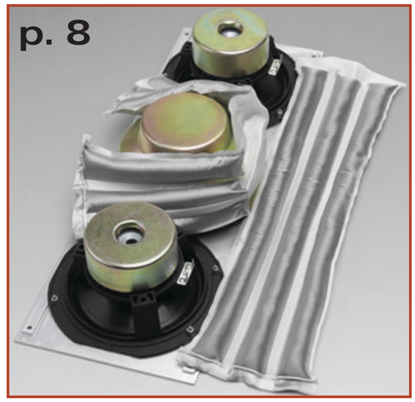

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By Mike Klasco and Steve Tatarunis (United States)

Behind the Speaker Cone

Box Stuffing and Beyond

Selecting the right cabinet prevents unwanted resonance

With the exception of speaker cables, perhaps no other audio component carries more baggage than speaker enclosure stuffing. Part of the confusion stems from a misunderstanding of the different application requirements, which are diverse. Some box stuffing materials work well in most cases, while others provide more psychological comfort than acoustical benefit. Likewise, costs, manufacturability, and stability must be considered. Over the last 25 years, “magical” box stuffing has been introduced from time to time, and some of the options do make a difference. Ever since loudspeaker designers realized the acoustic benefits of partially or fully enclosing a loudspeaker in a cabinet, they have sought to extract the maximum bass performance from a minimum cabinet size. We can begin separating wisdom from witchcraft by understanding box stuffing’s functions.

BOX STUFFING

Since half the energy coming from a speaker’s cone emanates from its back side, the “backwave” has to go somewhere. Certain enclosure types (e.g., the bass reflex) constructively utilize the component’s low-frequency portion by tuning and phase inverting the energy to contribute to useful bass output.

Other enclosures (e.g., acoustic suspension) depend on the stuffing to enhance the air spring compliance to provide a bit more low-frequency response. In this case, the sound energy inside the speaker enclosure is slightly turned into heat energy by the box stuffing fibers’ frictional movement (shearing action). This frictional shearing is modeled in electrical circuit analogies (e.g., Thiele-Small parameters or SPICE circuit simulation programs) as an increase in the size (value) of a capacitor in series with the woofer. In a mechanical description of the mechanism, the fiberglass stuffing enables the air spring of the trapped air in the enclosure to become less stiff and the speaker’s resonant frequency in the enclosure drops somewhat—perhaps 20% to 40%. Since stuffing costs can be significant with large acoustic suspension designs, it’s up to the engineer to justify the cost. As with bass reflex-tuned port designs, the stuffing’s absorption, at higher frequencies, suppresses box modes, smoothing upper frequency and transient response. Acoustic suspension designs benefit from increased stuffing to a point, while too much stuffing in a vented design will reduce the useful output from the port.

Omitting box stuffing in most of the less-expensive, compact audio products (e.g., sound bars and docking stations) is a disturbing trend, indicating a clear misunderstanding of this component’s importance. I just finished reviewing a $1,200 pair of mobile, self-amplified DJ speakers for a DJ trade magazine and was surprised to see absolutely no box stuffing even though the woofer had a cast frame, heat fins, and a neodymium magnet (all premium components) and the high-end section was a ribbon planar speaker. The crossover was at 2 kHz to enable the ribbon to reach high sound levels without distortion. You could hear this backwave midrange “noise” coming out of the ports next to the woofer.

While enclosure stuffing is generally understood to influence the low-end response, its effects on midrange clarity are less widely appreciated, especially in the case of small surround-sound satellite enclosures. These all have their woofers crossed over high in the midrange. With these 3” or 4” woofers, there’s a lot of midrange energy being dumped off the cone’s back. This sound energy bounces around and comes back through the cone—or the bass reflex ports, if one is used.

STUFFING CHOICES

There are two main categories of stuffing: foams and fibers. Each type has advantages and disadvantages.

Open-cell polyether and polyester foams have similar properties, where foam density and cell structure are related to acoustical performance. The foam’s primary advantage is ease of handling. Foam can be cut to order in fixed, non-friable geometries, making for ease of warehousing, inventory, handling, and assembly. Foam’s poor, low-frequency absorption is of limited benefit in sealed enclosures at low frequencies. Foam can deteriorate when exposed to UV light, fungus, ozone, and so forth, but these don’t tend to be issues in sealed or passive radiator systems, where the foam cone surrounds may go first!

The major fiber groups used in acoustics include acrylics, wools, and glass fibers. Fiber length, texture, diameter, and packing density tend to be the major acoustical variables for a given fiber type. Optimal fiber geometries provide the best results, though attention to detail tends to increase costs.

Wool is an expensive fiber and impractical for mass-produced speakers. On the other hand, when properly applied, wool does have excellent acoustical characteristics. Since the fibers tend to bunch up, it’s difficult to get consistent packing density and volume due to its dimensional instability.

Fiberglass is a low-to-medium priced fiber with good-to-excellent performance. Fiberglass is generally used pre-cut in sheet form, with specified thickness and density. As anyone who’s

Photo 1: Cerwin-Vega produced an advertisement to promote its Thermo-Vapor suspension bags.

ever worked with fiberglass can attest, fiberglass's primary disadvantage is adverse worker reactions (e.g., itching and allergic reaction).

Acrylics are medium-priced fibers with good-to-very-good acoustical performance, depending largely on fiber geometry and packing density. Unlike fiberglass, acrylic fibers are stable and non-allergenic, with no special handling issues. Acrylics can be specified in sheet for lining or in bulk for filling. Small buttons of acrylic fiber are occasionally produced for back-of-dome applications by thermally fusing the fibers around the button's periphery.

HOFFMAN'S IRON LAWS

Hoffman's Iron Law was first formulated back in the early 1960s by Anthony Hoffman (the H in KLH). Hoffman's Iron Law is a mathematical formula that was later refined by Thiele and Small, whose work now forms the basis of all modern loudspeaker design.

Hoffman's Iron Law states a woofer system's efficiency is directly proportional to its cabinet volume and its cut-off frequency's cube (the lowest frequency it can usefully reproduce). To reproduce even lower frequencies at the same output level, you need a larger box. Larger woofers are better woofers. Larger cabinets are more efficient and produce deeper bass with less power compression. If size efficiency or range is compromised, a smaller design can be made. The old audio maxim: a good large loudspeaker will always beat a good small one, reflects a core aspect of the physics of loudspeaker design: the interdependency of box internal volume, bass extension, and sensitivity. Because a fundamental relationship interlocks these three factors, changing one tends to alter one or both of the others. There are a few ways to get more bang for your buck and bend the Iron Laws. Some tricks include gain/bandwidth tradeoffs and magical enclosure stuffing.

WEIRD SCIENCE

In the 1950s, Edgar Villchur invented a small loudspeaker capable of producing deep, rich bass tones. This opened the high-fidelity music market to millions of everyday listeners. Villchur used fiberglass in the enclosure because it was more compliant (compressible) than air alone. There is a means of making a speaker cabinet's internal volume appear much larger than it is by using esoteric internal "stuffing," although practical application has been elusive. A sophisticated and more complex solution is a "module" or bag containing gases and liquid phases that is more compliant and compressible than fiberglass. Eugene Czerwinski received a patent around 1978 for packaged two-phase (gas/liquid) porous cocoon and trademarked the technology as Thermo-Vapor. Then, Cerwin-Vega S-1 and S-2 went into production using Thermo-Vapor bags inside their enclosures (see **Photo 1**).The effectiveness and long-term stability of this approach would be an interesting study. At the 1984 AES convention, Ralph Marrs presented a paper on supercompliance for speaker enclosures. He provided an update on the topic at the 1990 AES convention. The original design required auxiliary heat to operate at its boiling point of 117.6°F. In the later version, the system boiling point passively tracked environmental conditions from 65°F to 80°F, without requiring heat or support systems. It promised the virtual enclosure acoustical volume would be doubled with superior dampening.

Photo 2: The KEF KHT9000 uses ACE active carbon internal stuffing.

This seemed too good to be true. Around 1991, Marrs and I briefly explored a pro sound implementation. In previous years, I had consulted with Yamaha Proaudio on its YST servo control technology application using negative output impedance for tighter control of bass drivers in arrays. Woofers talk to each other, especially in close-coupled arrays. The Yamaha YST 1510, along with the rest of the YST pro audio series, was the first to use digital signal processing (DSP) loudspeaker management processors. Adding Marrs's supercompliance to the enclosure along with the YST amplifier output stage topology had the potential to reduce the enclosure size for touring sound applications while improving bass response.

Unfortunately, the test results, at least as far as the collaboration was taken, indicated the woofer's enhanced damping absorbed more bass than the supercompliance provided. At the time, Marrs concluded that supercompliance was best suited to acoustic suspension designs, as the contributed bass output from the port was reduced.

Aside from Cerwin-Vega and Marrs Developments, Polk, KEF, and AR all experimented with solutions less complex than supercompliance, but they still promised potentially higher compliance than conventional glass fiber stuffing. These approaches typically used activated carbon granule variations. The allure to achieve "big bass from small boxes" is hard to ignore.

Around 2004, KEF introduced its Acoustic Compliance Enhancement

Photo 3: The N'Bass is a custom-developed (white) material shown here with typical microspeakers.

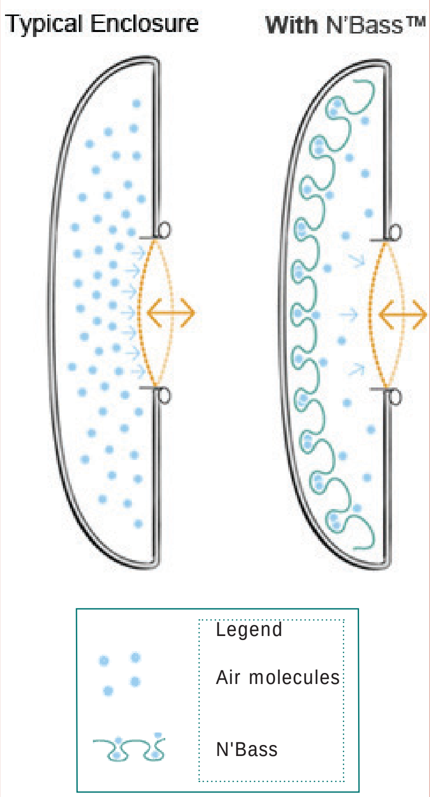


Figure 1: A typical speaker enclosure (a) and a speaker enclosure with N'Bass (b) show the difference in available back volume.

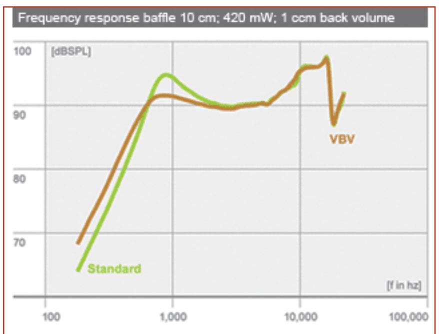


Figure 2: The frequency response of an 11 mm × 15 mm × 3.5 mm speaker with 1 cm³ back volume.

(ACE), using the surface action of activated carbon. Activated carbon grains have a complex surface structure comprising numerous pores of different sizes. These pores supply a high ratio of surface area to volume and provide a large number of sites where molecules can attach to the material surface or "adsorb." (Adsorption is strictly a surface phenomenon, in contrast to absorption where a substance is carried into the material's body.) Certain activated carbon forms have a high affinity for air, and it's this property that is exploited in ACE. A loose-weave bag of activated carbon granules is placed within the cabinet and the speaker system then it is assembled. As the speaker diaphragm compresses the air in the cabinet, the activated carbon absorbs more air molecules. The diaphragm consequently experiences less resistance to its motion, as if the air's volume in the cabinet were larger. As the diaphragm moves in the opposite direction, reducing air pressure within the cabinet, the activated carbon releases air molecules (a process called desorption), so it appears to the drive unit as if it is operating within a larger enclosure.

Although this increase in cabinet volume is apparent rather than real, it elicits the same benefits as a physically larger cabinet. This means bass extension and/or sensitivity can be improved while maintaining the same cabinet volume. Or, the cabinet volume can be reduced while maintaining bass extension and sensitivity. KEF's research showed the apparent increase in volume achievable in practice can be as large as three times the original (actual) volume. Still greater enhancements are feasible but rendered impractical because the activated carbon then adds too much internal damping (as with the Marrs supercompliance technique). It has been eight years since KEF first introduced ACE, and it is still used in its higher-end speaker systems, including the KHT9000 and KHT 6000 (see **Photo 2**). With the need for laptop and cellphone microspeakers and related mobile audio products, engineers searching for better bass from "shrink-wrapped" sub enclosures stumbled upon magic box stuffing.

More recently, Knowles, a major manufacturer of hearing aids and microspeakers, introduced its N'Bass Virtual Back Volume Technology for mobile devices.

N'Bass is a sound-enhancing technology intended to increase consumer devices's acoustic performance with small form factors. It enables the smallest possible acoustic designs in smartphones, tablets, and other portable media devices. N'Bass is a custom-developed material, which Knowles claims increases the loudspeaker's back volume up to 100% (see **Photo 3**). It provides either better acoustic performance—specifically more bass—or up to 50% smaller loudspeaker box designs with the same performance. It also facilitates bigger loudspeaker usage, which provides superior sound performance within the same gross application volume. Similar to a sponge, the N'Bass material adsorbs the air in the loudspeaker box. In this way, it virtually increases the loudspeaker's available back volume (see **Figure 1**).

Figure 2 shows the frequency response of an 11 mm × 15 mm × 3.5 mm speaker with 1 cm³ back volume. The difference in the resonance frequency is about 150 Hz. This means a sound pressure level-boost below resonance frequency of approximately 3 dB using N'Bass virtual back volume.

BOX STUFFING AS AN AFTERTHOUGHT

Proper box stuffing applications can greatly enhance a loudspeaker system's performance. Today's engineers have a variety of box stuffing materials. The push for smaller speakers and smaller speaker enclosure volumes continues to be a factor that speaker designers must accommodate. The enclosure stuffing's benefits, particularly the sophisticated higher compliance enclosure stuffing technologies, are a real consideration for sound quality, and they offer a unique "out of the box" approach to Hoffman's Iron Laws.

Unfortunately, it has become commonplace for some manufacturers to view box stuffing as an afterthought, leading to poor application or even total elimination of box stuffing. While this practice might be understandable in low-cost systems, the fact that we're beginning to see it in more premium products is a disturbing trend. *aX*

 By Ward Maas (The Netherlands)

The Munich High End Audio Fair 2012

A Look at Innovation in High-End Audio

A serious event with an excellent turnout

The 2012 Munich High End Audio Fair took place earlier than usual, running from May 3 to May 6, 2012 at the Munich Order Center (MOC). A total of 366 exhibitors from 33 countries represented more than 900 brands, and 4,427 visitors—a 4% increase over last year—came from more than 70 countries (see **Photo 1**). All the ingredients were set for a fantastic happening, and it was. But, it differed somehow compared to prior years.

Some companies with well-known brands chose not to participate and new alliances were formed. There was also a great deal of competition among suppliers. For instance, the fair's catalog listed seven pages of just connection cable and plug suppliers. There were also manufacturers, OEM suppliers, distributors, and sales houses competing for visitors by promoting everything from best products to best value for money to color.

There was also a change in the manner new products were offered. Some manufacturers chose to be more modest, just showing last year's products. Luckily, quite a few other manufacturers also showed their new products.

NEW SPEAKER PRODUCTS

Backes & Mueller (www.backesmueller.de) is a company that has been on the scene for decades, but never seems to get the breakthrough it deserves. Not only is it very competent when it comes to technology, but it also has the know-how to produce extremely good-sounding loudspeakers. It is an "ear-opener" to hear a voice in a familiar recording as "background mumbling" become clearly understandable. At the fair, Backes & Müller showcased its BMline100 speaker on the Atrium floor next to its listening room. I got

Photo 1: The Munich High End Audio Fair drew 366 exhibitors.

the impression many visitors regarded the speaker as a piece of art, nicely matched with the MOC (see **Photo 2**).

While the Backes & Mueller system was large, the new Nola Grand Reference Series VI was very large (see **Photo 3**). Accent Speaker Technology (www.nolaspeakers.com) presented this massive four-tower system with 23 drivers per side is as a major upgrade to its previous system, which the company described as "breathtaking." Of course, the external passive crossovers and the ball-bearing crossover isolation platforms are also needed.

Silbatone (www.silbatoneacoustics.com) was apparently afraid someone was going to top last year's gigantic system, so it brought an even larger system this year: a WE-15A horn set with field coil drivers, field coil tweeters, and a subwoofer that I mistook for a shielded crew area. I only discovered it to be a subwoofer when I left the room and saw an EV30 (one of two) in a window reflection. Yes, it sounded very pleasant, but I do not think a single system component

Photo 2: The Backes & Müller BM100 speaker resembles a piece of art.

Photo 3: Nola Grand Reference is a massive four-tower system with 23 drivers per side.

Photo 4: An ADN aluminum segment is one piece of a unique speaker cabinet.

will fit in my living room.

What will fit is the KEF LS50 mini monitor speaker, which is KEF's 50th year anniversary product. Inside it is a coaxial driver similar to its "Blade" system. It has special cabinet damping, an optimized baffle shape, a nice price tag, and it sounds great. I've seldom heard a precise bass that low and loud from such a small system. This is definitely going to be a hit (www.kef.com).

It is easy to get overwhelmed attending a fair like this. So, sometimes a product or a company can get overlooked. Luckily, I did not overlook the products of ADN Acoustics (www.adnacoustics.com). A casted, aluminum thing on the floor of the booth, which turned out to be a loudspeaker system segment, was on the floor of its booth (see **Photo 4**). A number of the segments are bolted together with a top and bottom plate to form a stack. A front plate is welded onto the stack, then sanded and polished to form a loudspeaker cabinet (see **Photo 5**). The walls are then filled with a special damping material. Using Scan-Speak drivers and Mundorf crossover parts, it has everything needed to form an interesting speaker. Unfortunately, ADN Acoustics did not demonstrate this system at the show.

Photo 5: The ADN segments were stacked to form a speaker cabinet.

But, even ADN Acoustics's smallest variant "The Secret" is a backbreaker, weighing 46 kg (101.2 lb) and measuring 57 cm (22.4") a piece. Its larger brother "The Column" weighs 90 kg (198 lb) and is 110 cm (43.3") high. It was interesting to see this Spanish high-end initiative.

HORN SYSTEMS

Of course, the horn systems always attract attention. This year, a few companies showed milled plywood horn systems. Among them were Cessaro (www.cessaro-horn acoustics.com) and Tuneaudio (www.tuneaudio.com), a Greek company that showcased its Anima. It was worth a look and a listen.

In the horn section, Autotech's products (www.horns.pl) could not be overlooked. It makes a wide range of multilayer composite horns and waveguides. The standard version comes in white, but all RAL (a color-matching system used in Europe) colors are available on request. Also, for the DIYer, it offers products in eye-catching colors (see **Photo 6**).

MSB Technology (www.msbtech.com) impressed me with a series of high-tech products. They were really showing off the Platinum Signature DAC IV with its Space Shuttle ceramic tile-style casing for the clock oscillator, modular approach to inputs (just plug in the kind of input you need), and its ability to

Photo 6: Autotech horns are available in many colors, including this bright red.

be updated in many aspects. I have to admit I loved it. Besides having impressive specifications, its appearance is impressive as well (see **Photo 7**).

Just before the show ended, MSB Technology debuted its new "affordable" DAC, the Analog, which is "just" a black/natural-colored 22-mm aluminum slab, with a rather minimalist user interface (one button, one knob),

Photo 7: MSB Technology debuted its new Platinum Signature DAC IV.

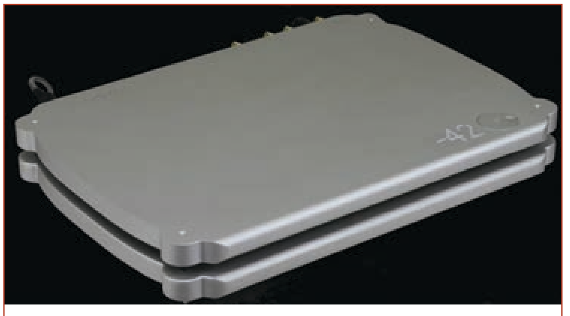


Photo 8: The MSB "Analog" offers a rather minimalist user interface (one button, one knob).

but, what an impression. For me, it confirms this is one of the most prestigious new products on the block (see **Photo 8**).

SOCIETY'S NEW INITIATIVE

The High End Society (the organizers of the Fair) offered young, new companies the opportunity to participate free of charge. The invitation was open and to all high-quality entertainment electronics companies that opened in the past 15 months. The High End Society selected six start-up companies from the huge number of applicants, enabling them to present their newly developed technologies at the fair.

Bohne Audio (www.bohne-audio.de) was a pleasant surprise, as it offered a complete range of loudspeakers and amplifier/controllers. I hope this initiative will continue!

TECHNOLOGY LECTURES

The technology lectures on all aspects of audio have already gained a solid place on the Fair. Unfortunately, this year's program was ready/available very late, and interesting lectures were held at the same time as other events. A bit more coordination in the future might enable more people to attend the lectures.

MEDIA PLAYERS

While high-end media players were new to last year's fair, this year I got the impression that those products have matured very quickly. All the technical hurdles had been met, with all possible user interfacing now available. Take Aurender's S10 (www.aurender.com). It has good specs, and it is good looking. Check one out for a listening session.

THE FUTURE

"What next?" seems to be the logical question. Next, the price will come down, meaning that, in the coming years, high quality can be taken for granted. I guess the shape of things to come will also be determined by products like the Reelbox Avantgarde 3 (www.reel-multimedia.com), (www.reelbox.co.uk). This new product plays a list of audio formats, with FLAC-HD (96 kHz/24 bit).

Photo 9: The Reelbox Avantgarde 3 could be a sign of the high-quality audio products to come.

If you read its functionality overview, you might get the impression that the audio capabilities are just a by-product. But, this is, indeed, a multi-media home system that does almost everything from TV reception to distributing everything throughout your home. The set up is modular and the users are active on an Internet user forum. Just don't look at the price, you might want one (see **Photo 9**).

The German company Accuton (www.accuton.com) also made a solid impression with a range of high-quality loudspeaker chassises. It does not shy away from difficult, exotic material (e.g., ceramics) and it has a large online product overview, which includes pricing. I once saw a midrange ceramic cone break like a porcelain teacup, so I was a bit reluctant regarding the use of such material, but Accuton is confident that incorrect handling is the main reason for failure. In the case I witnessed, that was clearly true. So, there should be no reason not to test them in your next project—although I am afraid they are not exactly intended for low-budget projects.

Of course, a technological leap forward shows commitment, and it is a good way to position yourself as a brand. But, it doesn't necessarily mean commercial success. Mainstream products represent a large market segment value. Certainly in Europe a few very large outlets dominate that market. I admire the courage of a company like TMA Audio (www.tmaaudio.com) that decided to enter the market with a complete range of good-quality loudspeakers.

Certainly, it is a huge effort to get products on the market that satisfy customers, dealers, and the companies themselves.

THE HIFIDELUXE SHOW

At the same time as The Munich High End Audio Fair, the smaller Hifideluxe show took place in the Munich Marriott hotel. There, you could see a group of small brands that apparently feared they would drown in the large MOC where the larger High End Fair was held. A shuttle bus service was offerred to the Hifideluxe show so quite a few fair attendees went to listen to some good music in a cozier environment. Unfortunately, this affair was a bit too small to stay for long time. But, the idea was well received.

THE FAIR DIFFERED FROM PAST YEARS

One evening, I was sitting in a restaurant musing why this year's fair made a different impression on me. Was it the different timing? The fair didn't take place during a public holiday and the number of visitors was more evenly distributed. The result gave a quieter impression, even with the increased number of visitors. Was the euro forcing companies to change partners and alliances? Apparently not.

Just as I was about to leave my musings and pay attention to my beer, I heard from someone who has plans for a new audio publication called *Katzenjammer* (German for cat wailing, and also used in reference to a hangover). The magazine would be devoted to the "dark side of audio," so it would include stories about malevolent glowing tube amplifiers and audio cables from hell. Then, I understood what was missing. This year's fair was a serious event. But fortunately that conversation provided the perfect counterbalance. Cheers! *aX*

By Richard Honeycutt (United States)

Differences in Amp Sound

How Do We Find the Truth?

Sound reproduction requires the use of art and science

In the last couple of articles, I've explored distortion in a quest to find the origin of "tube sound." Now it's time to take a more general view of possible amplifier differences, solid- or hollow-state. A sound system's purpose is to provide a functional, pleasant, and accurate rendition of the original sound field. For this article, I will limit the discussion to systems designed for the high-fidelity reproduction of music—either live or prerecorded. (Some of my colleagues would substitute other descriptors. A broadcast engineer, from whom I learned much, used the term "high-fiddle-ity" in reference to those who continually fiddle with their equipment, and a well-known Midwestern consultant denotes the area as "high-futility," based upon much the same set of observations!)

SOUND SYSTEM DEBATE

Whether the sound system's goal is "accurate" reproduction or simply "pleasing" reproduction is often discussed. Often, accuracy of reproduction is assumed to be the goal. In saying this, one must recognize the impossibility (futility?) of recreating, in one acoustic space, a sound field generated in another acoustic space. Yet, much of this difficulty stems from differences in directional cues and reverberation. Directional cues are primarily a matter of loudspeaker positioning, room geometry, and acoustics. Reverberation in a listening room seems to be largely factored out of perception by the human ear/brain mechanism. We become accustomed to the sound of the room we're in. Thus, we can theoretically come pretty close to recreating the same perceptual experience in our listening room as was experienced in the original venue. The focus in this series of articles is the presence, or absence, of different amplifiers's differences in sounds.

Given the goal defined in the previous paragraph, I have dived headlong into the "measurement versus perception" debate.

Numerous sound-system attribution studies have been made in an effort to correlate physical measurements with listener perceptions in the evaluation of sound systems. Albert Einstein said everything should be made as simple as possible—but no simpler. If we hear differences in the sound of different amplifiers, we try to explain why the differences exist. At any point in our understanding, we may fall prey to the assumption that all perceived elements correlate with the presently popular set of measurable characteristics. The result of this unfortunate tendency has been to alienate a certain portion of audiophiles from the scientific side of the field, to the point that they may suggest perception is everything and measurements are meaningless. At the other extreme are those claiming our present level of understanding is sufficient to completely characterize perceptual response to a sound system in terms of measurements alone. While this is certainly the goal, it has not yet been realized. So, neither of the extreme views helps us progress in the art and science of sound reinforcement and reproduction.

OUTSIDE INFLUENCES

Recognizing that listeners' sound equipment opinions do not always correlate with measured performance ratings, many researchers have studied the subjective evaluation process of components and systems. Their intent has been to maximize the amount of equipment-related information that can be extracted from such evaluations while minimizing extraneous information input. Examples of extraneous input are numerous. Listeners evaluating speaker systems respond more favorably to equipment with known brand names, even if the equipment they are praising was manufactured by company A, and the nameplate from the well-known manufacturer company B was substituted for the original. Listeners have been shown to prefer speaker A over speaker B when the sound level from speaker A was only a few tenths of a decibel higher than that of speaker B (due to differences in electrical drive levels). They switched their preference to speaker B when its level was raised above that of speaker A. The cabinet appearance also affects listeners' sound quality perception. A speaker's room location, both in relation to the listener and in relation to the room boundaries, affects preferences. The list goes on, and some of it applies to amplifiers as well as speakers.

The multitude of extraneous information that can intrude into a subjective evaluation is such that some professionals have characterized subjective testing as useless. And yet, we live in subjectivity. A perception that is true for the perceiver may not be true for other persons, meaning a subjectively true perception is not necessarily objectively true. The problem is not with subjectivity, but with deriving relevant and repeatable information from subjective tests while rejecting the irrelevant and non-repeatable. When this is accomplished, the subjective test results yield useful buying recommendations for listeners and can provide valuable input for product design improvements.

Two excellent examples come from the history of power amplifier design. In the early days of solid-state amplification, transistor amplifiers suffered a bad reputation due to what was described as a "harsh sound." After design theory advanced to include techniques for applying the correct bias current throughout the full operating temperature range, crossover distortion of solid-state amplifiers was reduced to low levels, and most of these complaints subsided. But, since distortion was usually measured at full output and crossover distortion is negligible except at low output levels, the objectionable levels of distortion were not noticed until subjective complaints alerted us to them. A similar situation obtained in the 1970s, beginning with Matti Otala's work on transient intermodulation distortion (TIM). This distortion is difficult to measure, as it only occurs during transients, and then only if the amplifier's slew rate is so low the negative feedback signal is unacceptably delayed in its action. When this occurs, the negative feedback's distortion-reducing action does not occur, and momentary high levels of distortion occur in the signal. TIM's effects are clearly audible to careful listeners, but until the cause was identified, that audible perception could not lead to the design improvements that eliminated TIM.

AES GUIDELINES

In 1996, the Audio Engineering Society (AES) published "AES20-1996: AES Recommended Practice for Professional Audio—Subjective Evaluation of Loudspeakers." It was written after years of collaborative work by some of the finest minds in the field. This standard (revised in 2007) provides valuable guidelines that should be observed in any effort to obtain useful subjective information on any piece of sound-system equipment. It is based upon careful, thorough science, and while it makes no claim to being the last word on the topic, it well summarizes a helpful approach to this emotion-laden area. AES20 is focused on speaker evaluation, but many of its insights apply to any sound equipment's subjective evaluation. Guidelines included in the standard can be grouped into four areas: room requirements, listening arrangement, program material, and test procedures. I will examine the last two areas, as they are most appropriate to tube amp sound.

PROGRAM MATERIAL

Program material should be chosen with several specific goals in mind. First, it should be material for which almost anyone can recognize whatever aural defects the test system may introduce. Foremost among such program material is the human speaking and singing voice. Acoustical instruments are also good. Electronic instruments have no native sound: their sound varies depending on the instrument's setup and amplification. Therefore, no matter how much a particular listener may personally like any given recorded performance or group, electronic instruments should be used only for specific purposes during listening tests, (e.g., width of frequency response and ability to handle power). They are not generally suitable for evaluating frequency-response flatness ("accuracy") or subtle distortion levels.

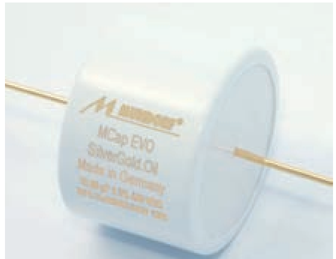
Second, the material should provide extremes of both frequency and dynamic content. Since music sounds most natural when heard at concert levels normal for that style of music, electronic music can be profitably used to test a system's ability to produce high levels. It is also important to remember sound systems can easily produce damaging sound levels, so high playback levels should be limited to safe time durations. Many references are available to help determine these levels. Occupational Safety and Health Administration (OSHA) guidelines set a maximum limit of SPL and duration, in that a certain percentage of the population will experience permanent hearing damage even when adhering to OSHA guidelines. **Table 1** shows the OSHA noise exposure guidelines. It is a good idea to limit exposure to 5 dBA below the levels.

Third, percussive instruments (e.g., piano and steel drums) can provide useful tests of a component's transient performance. And finally, imaging tests require program material that contains well-defined sonic images. This is not always true of otherwise good recordings. (Imaging concerns apply mainly to speakers. It is difficult to imagine any way in which an amplifier can affect imaging.)

AES20 contains the following recommendations concerning program material, plus others specifically applicable

Duration per day, hours	Sound level, dBA, slow response
8	90
6	92
4	95
3	97
2	100
1.5	102
1	105
0.5	110
0.25 or less	115

Table 1: OSHA's noise exposure guidelines are used to determine safe listening levels.



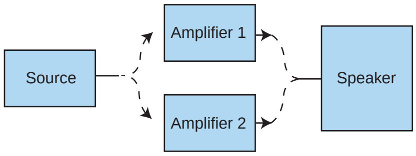


Figure 1: A test setup for comparing amplifiers can be simple.

to testing of equipment other than amplifiers:

- The material should include one anechoic recording of the male speaking voice, made using an omnidirectional microphone.
- The material should contain solo male and female singers.
- The above material by itself should contain sufficient breadth of frequency, transients, and dynamic range to provide a good system test.
- Test signals (e.g., pink noise and sine sweeps) should be used first to check for defects (e.g., speaker buzzes, out-of-phase connections, etc.).
- Music featuring electronic instruments should be used last to test the system's dynamic capabilities. (Loud levels may induce fatigue or even a temporary hearing threshold shift in listeners, rendering them less suitable for further testing until the ears have rested.)

TEST PROCEDURES

The test setup used for comparing amplifiers can be as simple as the one shown in **Figure 1**. Since the AES20 recommendations for test procedures are designed to accomplish specific goals, I added comments (in parenthesis) as some of these recommendations are listed. I have not included recommendations that do not apply to amplifier tests.

First, sound levels should be realistic for the type of program material. Second, a previously evaluated "anchor" system should be used for comparison with the unit under test. (This recommendation is not absolutely necessary if the test's goal is only to compare two or more units, but it is essential for making any sort of absolute judgment of a piece of equipment since even the same listener will experience variations in perception from day to day. Thus, a given listener may rate the same piece of equipment differently on different days unless there is a standard of comparison. Some listeners believe they have time-invariant golden ears. There is no evidence such belief makes them more immune to perceptual variations than other human listeners. Recognizing human limitations is not a sign of weakness, but of scientific integrity.)

Third, in A/B tests, the levels of the "A" and "B" systems must be matched (using pink noise and a sound level meter on slow setting) within 0.5 dB. Fourth, listeners should stand or sit in a variety of locations, moving from time to time during the tests. (This recommendation helps prevent contamination of test results by room-position-dependent acoustical factors.)

Fifth, each program sample should last from 30 s to 60 s. (Shorter clips do not provide sufficient time for a judgment, and longer times risk prolonging the total test to the point of inducing listener fatigue.)

Sixth, trained, experienced listeners should be used. (Studies have shown that untrained listeners tend to show poor consistency, and their responses may

Spectral Uniformity

1. Uncolored
2. Octave balance, clarity
3. Balance, bass to treble
4. High-frequency extension
5. Low-frequency extension

Dynamics and Distortion

1. Maximum loudness
2. Transient impact or punch
3. Not modulated, compressed
4. Pianissimo clarity
5. Loud, no strain or glare
6. Loud, low frequencies, no timbre change

Listening Fatigue

1. Long time without fatigue
2. Fatigue only on some selections

Transparency

1. Ability to pick out individual instruments
2. Free from harshness or stridency
3. Lack of graininess

Quietness

1. Free from electronic hiss
2. Free from electronic hum
3. Free from electronic buzzes
4. Free from mechanical hum, buzz
5. Low operating noise

Figure 2: This checklist of descriptors will help ensure consistent results.

tend to cancel statistically, reducing the confidence rating of the test program. Also, tests using just one listener cannot yield objective results; they can only indicate that listener's preference.)

Seventh, a checklist for characteristics to be described should be used. The use of consistent descriptors that are well-understood by all listeners is vital in order to ensure valid results. (Such descriptors are illustrated in **Figure 2**.)

Breaks should be taken as needed to avoid listener fatigue. And last, testing should be blind, and, if possible, double-blind. This means that at least the listeners, and preferably those conducting the test, should not know which unit they are evaluating at any particular time. Any form of blind testing adds to the complexity of the process. However, experimental science has established the reality of a placebo effect (i.e., attributing a perceived experience to an incorrect cause). In tests using expert listeners at an AES meeting some years ago, many listeners heard several solid-state and tube amplifiers, using a wide variety of program material and a double-blind protocol. Listeners were not able to reliably distinguish among the amplifiers, even though many of these professional audio engineers were convinced they heard consistent differences. Refusal to believe in the placebo effect provides no exemption from experiencing it!

TRAINING PROGRAM

The training program should involve three primary phases. First, each listener should become familiar with the aural meaning of each descriptor used on the checklist so everyone uses the same vocabulary. Second, each listener should become familiar with the program material, using excellent reproducing equipment. Third, each listener should be qualified for self-consistency using blind listening tests. Any listener's hearing defects should be noted on the test report.

Blind evaluation of amplifiers requires locating the equipment in another room so listeners never know which amplifier they are hearing at a given time. The most important element in a meaningful subjective test is blind testing. Those who disagree often call upon the proponents of blind testing to prove their cases, or they object to blind testing for reasons that have been more than adequately met in literature from many scientific fields. I submit the absolute requirement of blind testing has been proved, and thus, the onus of proof falls upon those who oppose its necessity.

Blind testing requires at least two participants in the test: one who periodically switches the equipment under test into or out of the audio chain, and the other who listens. The switching protocol should allow for a "switch" operation to either swap or not swap the connections, so when the listener hears artifacts from the switching, (s)he will not know whether a swap has occurred. The person doing the switching must carefully record which piece of equipment was used in each trial, so the equipment's identity can be correlated with the corresponding listener reports. After one series of trials, the participants can trade positions, so there will be two listeners. Switching can be performed by simple cable swapping. No complex relay arrangements are needed.

Second in importance is correct equipment setup. As an absolute minimum, the test room should not have an unusual acoustical character, in terms of either resonant modes or reverberant characteristics. Speakers should be located to avoid extraneous reflections that can color the sound. Level matching within 0.5 dB is essential. Failure to observe this requirement will invalidate the results, since research has clearly shown listeners relate higher sound level to higher quality, even if they do not consciously recognize the sound level as being higher.

Many articles have been published based on testing that does not adhere to these standards. Thus, much dubious material has been promulgated, contributing more heat than light to the "great debate."

ART & SCIENCE TOGETHER

Earlier in this article, I used the phrase "the art and science of sound reproduction and reinforcement." Too often progress in this field is hindered by our failure to integrate both aspects. But if we scientists will respect the art, and we artists will respect the science, progress will be achieved. In the next article, I will examine results obtained by some excellent researchers. *aX*

By Richard K. Mains (United States)

Experimenting with Electrostatic Speakers (Part 2)

Learn to fabricate electrostatic cells

In the first article of this three-part series, I discussed the basic materials required to make electrostatic cells, along with information on how to obtain them. In this installment, I discuss the procedures I followed for electrostatic cell fabrication. I will also try to clarify wherever the steps differ from the instructions provided by Roger Sanders in his article "A Compact Integrated Electrostatic/Transmission Line Part II," *Speaker Builder*, February 1990.

GLUE THE STATORS TO THE SPACERS

The first major step in the fabrication process is to glue the perforated metal stators to the spacers. Put the spacers in their proper positions and tack them together with plastic cement (e.g., Weld-On). I put a layer of plastic wrap under the spacers for these gluing steps. The plastic wrap serves two purposes. It prevents spills, and it also tends to keep the plastic spacers from shifting. Put a bead of polyurethane glue around the spacers where the metal stators will go. Take care to leave an unglued space at the corner. It will be bent back later to make the stator contact. At this point, a problem arises that I encounter in all the gluing steps. How can I lower the stator onto the spacers so that it registers in the proper position? And, how can I do this without getting glue on my hands? The stator is big enough that, unless you are more coordinated than I am, it will not be lowered into exactly the right position. This means it must be moved after it has touched the glue, which means the glue will be smeared.

I gave this problem some thought and came up with a simple and effective solution. **Photo 1** illustrates my solution, as well as some other features about the fabrication process.

In **Photo 1**, the materials are placed on top of one of the two 48" × 24" × 0.25" sheets of glass I used to fabricate the cells. I had these sheets made by a local glass supply company, and—this is very important—I had them sand the edges. In the photo, you can see the template of masking tape I put underneath the glass. I used it to line up the spacers for the gluing steps. Of particular importance is the 0.5" spacer which must be located exactly in the middle for the cell's two halves to register properly. The layer of plastic wrap I put underneath the spacers is also visible. But what **Photo 1** is mostly intended to illustrate is the technique I used in all gluing steps to line up whatever I am trying to glue. I temporarily placed the stator in the position I wanted it to end up without any glue. Then I placed layers of masking tape around the bottom corners of the stator. You have to use at least four or five layers for the corners to effectively locate the stator. Hold each piece of masking tape taut and locate it so that it gently touches the edge of the stator, without enough force to disturb the stator position. When the masking tape corners are in place, remove the stator and go through one or two practice runs to ensure the stator will end up in the correct position. The method I used to locate the stator in place is shown in **Photo 2**.

Hold the stator nearly vertically and put the bottom corners in the masking tape locators. Then lower the stator as shown in **Photo 2** so it makes contact starting at the bottom and ending at the top of the spacers. When you are convinced the tape locators are doing their job, remove the stator and put down a bead of glue underneath what will be the stator's permanent location (except not at the contact corner!). Don't forget to put glue on the center plastic spacer as well. Just before the stator is about to touch the top spacer,

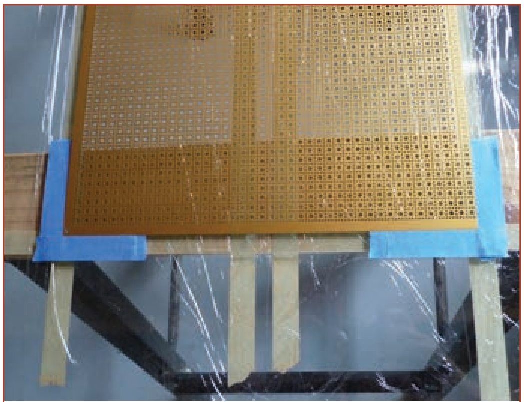


Photo 1: The tape corners are used to locate the stator in the gluing step.

Photo 2: The tape corners are used to to line up the stators.

Photo 3: The glass is weighed down for curing after every gluing step.

release it so your hands will not contact the glue. If the stator does not register perfectly, it should be very close so only a slight adjustment will be necessary.

After every gluing step, I put a layer of plastic wrap on top, then I place the second sheet of glass on top of the assembly. Next, I weigh down the assembly with books and place wood clamps in the middle to keep the glass from sagging. A gentle tightening of the wood clamps is sufficient (see **Photo 3**).

Two stator/spacer assemblies are prepared for each electrostatic cell, so carry out the gluing and curing process twice (see **Photo 2** and **Photo 3**). Orient the top spacers with the diaphragm contact holes so the two holes line up properly when the two stator assemblies are put together.

DIAPHRAGM PREPARATION

Next, I want to discuss the diaphragm's preparation, which is quite different from the Sanders procedure. In his article, the first step is to apply a graphite coating to the diaphragm. In my procedure, I wait until the very last step to apply a Licron coating. The main reason is that I can't imagine it is good for the Licron's adhesion to heat shrink after it is applied.

First, cut the film to size, leaving enough excess around the edges to attach pieces of masking tape. Stretch the diaphragm by placing the tape on alternate sides, just enough to remove all the wrinkles since it will be stretched more during the heat-shrinking process. By placing the diaphragm over the template, it will be clear where to put the polyurethane glue. Now use the same technique as for locating the stators when they are glued to the plastic spacers. Temporarily put a stator/spacer assembly down on the film, in the position where you want it to end up. (I start with the assembly that has the smaller diaphragm contact hole in the top spacer, which will be on the cell's back side.) Place strips of masking tape around the lower corners of the spacers. **Photo 4** shows the diaphragm after it is stretched out and the corner locators are in place. The template I made on the bottom of the glass is also visible.

After the locators are in place and you have made one or two dry runs to ensure the stator will be properly aligned, place a bead of polyurethane glue on the diaphragm where the spacers will be. Use the template as a guide for the glue, and don't forget the area under the center spacer. Remember to leave some clearance near the diaphragm's contact hole in the top spacer. You don't have to interrupt the glue there. Direct the bead to the edge of the spacer opposite the hole rather than putting it in the middle. Once the glue is in place, use the same method shown in **Photo 2** to lower the stator assembly onto the diaphragm film. Once the stator is properly located on top of the film, weigh it down with the second sheet of glass (see **Photo 3**). Wait about a day for the glue to cure if you use the polyurethane glue from ER Audio.

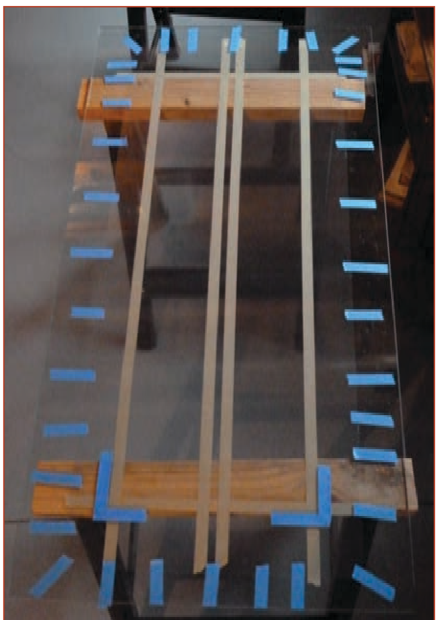


Photo 4: Remove the wrinkles and place the locator corners on the diaphragm.

After the glue is cured, remove the clamps and top sheet of glass. Then, remove the pieces of masking tape used as corner locators and to eliminate the diaphragm wrinkles. When removing the tape, pull in a horizontal direction away from the assembly. If you pull upward, there is a tendency for the diaphragm film to tear.

HEAT SHRINKING

Once the assembly is free, I do the heat shrinking. At first, I tried turning the assembly over and taping it to the glass with the diaphragm side up. But, I found I achieve better heat shrinking if I hold the assembly vertically with the bottom resting on the glass. It is helpful to wear white fabric gloves (the kind you wear when working with photographs) to prevent getting fingerprints on the film. I set my programmable heat gun to 390°F and I usually make three slow passes over the entire surface to achieve maximum heat shrinking. There is some bowing of the spacers, but they will flatten out in the final gluing step. I should mention one of the disadvantages of heat shrinking is that it is difficult to precisely control the amount of shrinking. The amount of shrinking affects the resonant frequency of the cell and its output. As to how tight you can stretch the diaphragm by heat shrinking, I am typically able to place a stack of about 20 quarters on the diaphragm before it touches the stator.

After heat shrinking, I tape the assembly down on the glass with strips of masking tape around the edges. The

Photo 5: Stretch the assembly out after the heat-shrinking process is performed.

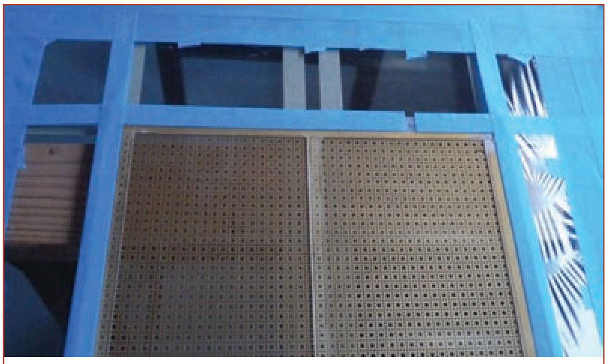


Photo 6: Masking the assembly is crucial at this stage of the process.

goal is to make the assembly as flat as possible in preparation for gluing the second stator on top (see **Photo 5**).

At this point, I apply the Licron coating. But before I do, it is necessary to mask off any adjacent areas where I don't want the Licron, since it is difficult to precisely control the spray. One of these areas is on the outside of the plastic spacers, where the polyurethane glue will be in the final fabrication step. I do this because I imagine the glue will adhere better without the Licron coating. It is important not to completely mask this area, since there will be a copper tape contact ring on the opposite stator that must make electrical contact with the diaphragm. You could also mask the area of the diaphragm along the center spacer, but I don't because I've noticed when you remove the masking tape, the Licron can partially come off in the direction you pull the tape. You can avoid this problem on the perimeter spacers by pulling away from the middle of the diaphragm, but you can't do this in the middle.

Photo 6 shows a closeup of the assembly's upper portion after it is masked. Only the outside of the area around the peripheral spacers has been masked. I do not mask in the vicinity of the smaller diaphragm contact hole, because this is where the diaphragm contact screw will have to make contact with the film. (Keep in mind that in this photo the diaphragm is on top of the spacers).

As I indicated earlier, it is difficult to apply the Licron in a single, even, and cosmetically pleasing layer. I often respray it one or two more times, letting each coat cure before the next is applied. I'm not sure exactly how much Licron needs to be on the diaphragm. I like to see it everywhere on the film. Also, I sometimes place two pennies close together in different spots on the diaphragm after it has cured and ensure I get at least some reading on my VOM's 200-MΩ scale.

After the diaphragm is sufficiently coated with Licron and the coating has cured, remove the masking tape in the spacers' vicinity. Don't remove the tape on the glass's perimeter yet. Be sure when you remove the tape to pull away from the assembly to avoid either tearing the film or removing a portion of the Licron. At this point, I am almost ready for the final gluing step, which attaches both stator assemblies together. Again, you can use the masking tape corner locators to ensure the front stator aligns with the assembly that is now taped to the glass. Temporarily lay down the front stator/spacer assembly so it aligns with the rear assembly. In particular, ensure the diaphragm contact holes align. Without moving the front stator assembly, place tape locators around the bottom corners of the front spacers. These locators will lie on the film surface. Then remove the stator assembly and test it a couple of times to ensure the locators work properly (see **Photo 7**).

FINAL GLUING PROCESS

Two additional things must be done before gluing the front stator. First, you need a copper tape contact ring on the front stator spacers. The tape is placed along the inside of the spacers, since the polyurethane glue will be on the outside. **Photo 8** shows the front stator after the contact ring is in place.

It takes a little practice to evenly apply the copper tape. First, I cut the length of tape I need for one side and separate the backing from the tape just at one end. I press the tape's end at the starting position, then I use both hands to simultaneously place the tape and separate the backing from it as I move along an edge. You can overlap the tape's end with the next piece that starts along a different edge. I then use an ohmmeter to verify the two pieces of stacked copper tape make electrical contact. In **Photo 8**, the assembly's top is on the bottom, where you can see the larger diaphragm contact hole. This hole is where the contact washer will fit. I make a contact out of copper tape on the opposite (rear) stator assembly for this washer to electrically contact (see **Photo 9**). This contact is on the film currently attached to the rear stator assembly, around the smaller diaphragm contact hole. It extends into the area where the copper ring on the opposite

Photo 7: The masking tape has been removed and the tape locators are in place in preparation for the final gluing step.

Photo 8: A copper tape contact ring is placed on the front stator assembly.

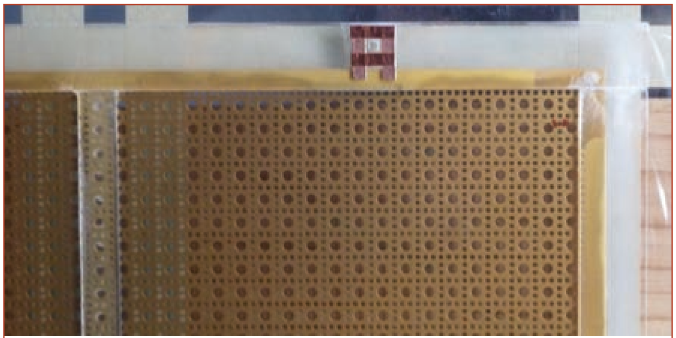


Photo 9: Copper tape contact around smaller contact hole on diaphragm extends into the area where the copper ring on the opposite stator will be located.

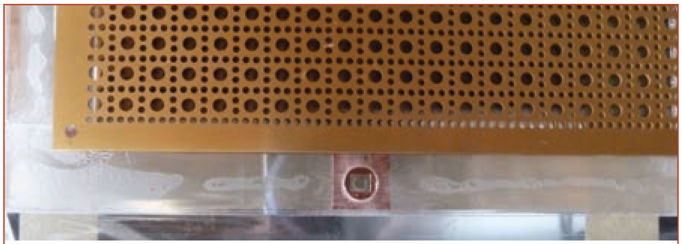


Photo 10: Alignment of the diaphragm contact holes is the final stage of the gluing process.

stator will be located.

We are now ready for the final gluing step. The corner locators are already in place to ensure the two stator assemblies register. Place a bead of glue on the diaphragm film along the spacers's outer edges and also along the middle spacer. Avoid the areas where the copper tape contact ring on the front stator assembly will lie. Also avoid the area around the diaphragm contact hole, which is easy because copper tape is now placed there. Lower the front stator assembly so it contacts the glue, using the locator corners as guides. I put a layer of plastic wrap on top before placing the second sheet of glass over the cell.

After the top sheet of glass is in place, inspect the cell to ensure the edges line up and, in particular, the diaphragm contact holes line up. If not, you can easily make adjustments by shifting the top sheet of glass. It moves the top stator assembly along with it. **Photo 10** shows the diaphragm contact holes as they should appear in this final step.

Weigh down the glass and put wood clamps in the middle. Let the glue cure overnight.

Remove the top glass and any masking tape, including the tape used for the corner locators. Again, be careful to pull the tape horizontally away from the cell. It is not necessary to completely remove the tape because you can use a small Exacto knife to trim away the excess film sticking out from the edges. Carefully puncture the diaphragm with a tool or drill bit the size of the smaller diaphragm contact hole to accommodate the #4 contact screw. I insert a small flat punch on one side of the hole and use a drill bit to carefully cut the film on the other side. Finally, carefully bend back the stators' corners where the #4 contact screws will be placed. I put something flat near the corner to support the stator's glued part while I use a putty knife to bend back the corner. If you use the Lincaine with albras finish, I would recommend using internal tooth lock washers at the stator contacts, to ensure the contacts pierce the coating on the stators.

TROUBLESHOOTING

In the second article of this series, I illustrated the major fabrication steps I follow to make electrostatic cells. In the third and final article of this series, I will discuss some issues that might arise after completing the cells. In particular, I will discuss the electronics used for my hybrid electrostatic speaker system. I will also offer some important considerations when using step-up transformers to drive electrostatics. *aX*

RESOURCES

R. Sanders, "An Electrostatic Speaker System Part I," *Speaker Builder*, February 1980.

———"A Compact Integrated Electrostatic/Transmission Line Part II," *Speaker Builder*, February 1990.

Monarchy SE-100 MK2 & SM-70 Pro Power Amplifiers

A look at Monarchy's new pro power amps

Monarchy SE-100 MK2 and SM-70 Pro Power Amplifiers
Monarchy Audio
380 Swift Avenue, #21 South
San Francisco, CA 94080
(650) 873-3055
www.monarchyaudio.com
monarchy@earthlink.net
Price: $1,179 each (either model)

Monarchy Audio has been manufacturing affordable, high-performance power amplifiers since the mid-1990s, when it introduced the original version of the SE-100 Delux monaural power amp. The SE-100 Delux featured a complementary MOSFET output stage and discreet, single-ended bipolar circuitry for the remainder of the feedback-based topology. A few years later, Monarchy introduced the SM-70 Pro, a slightly higher-powered version of its original SM-70 amplifier. The SM-70 amps were also based on complementary MOSFET output stages, along with op-amp-based input/voltage-gain circuitry. They were billed as "zero feedback," since they did not employ global feedback from the output to the input stages. The SM-70 could be used as a stereo or a monaural power amp. These high-value amplifiers held their own in audiophile circles for 15 years and were favorably reviewed in many high-end audio publications. I reviewed the SE-100 Delux amplifier in the September 2000 issue of *Audio Electronics* (a predecessor to *audioXpress*), and the SM-70 Pro in *audioXpress* (September 2001). Both amplifiers have been recently updated. The SE-100 MK2 design has been significantly improved, and both amps have undergone changes that result in higher reliability.

SE-100 MK2

Like its predecessor, the SE-100 MK2 is a monaural power amplifier, requiring two amplifiers for stereo. I have long believed in monaural power amps, which are often called "monoblocks" in high-end audio circles. Complete isolation of the two amplifiers yields superior soundstage presentation, and often better dynamics when compared to stereo power amps. With monaural amplifiers, the amps can be placed close to the loudspeakers, so loudspeaker cables can be kept as short as possible. Longer interconnects and short speaker cables are usually preferable to short interconnects and long speaker cables. Although the SE-100 MK2's case and heatsinks are identical in size to the original SE-100 Delux, the amplifier's appearance has been noticeably improved (see **Photo 1**). The new, black-anodized handles are oval. The front and rear panels are laser engraved, and the front panel has a cleaner, less cluttered look than its predecessor.

Although the SE-100 MK2 resembles its predecessor, the amp has a substantially upgraded design. Like its predecessor, the SE-100 MK2's input and voltage gain/driver circuitry is based on discrete, bipolar technology (see **Photo 2**). The MK2 amps employ a current-mirror input, making the input stage more stable throughout its entire dynamic range. The "SE" in the model designation refers to the single-ended differential input and voltage gain/driver stages, in contrast to the full-complementary differential inputs used in many power amp designs. By definition, the single-ended circuitry operates Class A. The current sources for both the input and cascode voltage gain/driver stages have been improved, lowering distortion and increasing speed.

All circuitry ahead of the output stage is powered by a pair of LM317/337 three-terminal IC regulators, fed from the main ±50-V unregulated DC rails. This is exactly as it should be, since the input and voltage-gain stages have the most to gain from clean, regulated DC supply rails. The output stage provides current gain, so absolute stability of the DC supply rails is not nearly as critical. The SE-100 MK2 is supplied with a 450-VA toroidal power transformer. Shindengen D6SB60L low-noise, soft-recovery rectifier bridges are standard, and the amp contains separate bridges for the positive and negative supplies (see **Photo 3**). The raw filter capacitor bank consists of four 15,000-µF capacitors, for a total of 60,000 µF.

The old power amps used Hitachi 2SK1058 and 2SJ162 MOSFET output transistors in a full-complementary, push-pull arrangement, four of each per amplifier. These transistors have a 160-V drain-to-source rating, a 7-A drain current, and 100-W channel dissipation. The original

Photo 1: A pair of Monarchy SE-100 MK2 monaural power amplifiers. The rear panel (a) shows the unbalanced RCA and balanced XLR inputs, binding posts for loudspeaker connections, and an IEC AC power connector. The front panel (b) sports new oval handles and a cleaner appearance than the original SE-100 Delux amplifiers.

output stage had virtually no protection circuitry for the output devices. The 2SK2221/2SJ352 complementary pair have replaced these transistors for the MK2 amplifiers. Channel dissipation is still 100 W, but drain-to-source voltage is rated at 200 V, and the drain current is 8 A. Monarchy notes the higher current capability enables the SE-100 MK2 to drive 2-Ω loads (the original amp was limited to 4 Ω). The new amps also feature a sophisticated output protection scheme. This translates into higher reliability. Monarchy also notes the amps have positive overdrive protection and very clean clipping.

Although the SE-100 MK2 became available in 2009, the latest version reflects significant recent changes. First, some samples of the SE-100 MK2 and the original SE-100 exhibited a low-level oscillation. MOSFET output stages can be finicky when it comes to layout and compensation. In the amplifiers's current production, gate-to-drain compensation capacitors have been added to the N-channel output transistors. This appears to have completely killed the oscillation. My latest samples are problem free. The main PC board has also been redesigned with a new layout and wiring scheme, resulting in an even lower noise floor.

The MK2 amplifiers have an extremely low output DC offset. My amps were both well under 5 mV and, unlike the old SE-100, there's no need to short the inverting input of the XLR connector (Pin 3) to ground when feeding the amp from unbalanced sources. The SE-100 MK2 seems relatively immune to impedance mismatches on the noninverting and inverting inputs, at least as far as DC offset is concerned. (Common-mode rejection is another matter.) The old SE-100 amplifier produced a hefty turn-on thump, which has been eliminated in the MK2 version with a time-delay relay in series with the output. The output relay is a Bestar CS-115CA, rated at 12 A/120 VAC, and the relay activation circuit senses the output offset and closes the relay after the offset has dropped to a safe level. Turn-on and turn-off on the new amps is completely silent.

For owners of the original SE-100 Delux amplifiers, Monarchy offers a reasonable upgrade plan. Replacing the main PC board or both MOSFET output boards costs $100 per amplifier. A pair of the Shindengen rectifier bridges costs $50. Monarchy has a flat labor charge of $50, so one amplifier can be completely updated for $300. Although you can update the main PC board, the output boards, or the rectifiers, I highly recommend a complete upgrade, combining an extremely worthwhile sonic improvement along with increased reliability.

SM-70 PRO

The latest version of the SM-70 Pro is nearly identical to the original

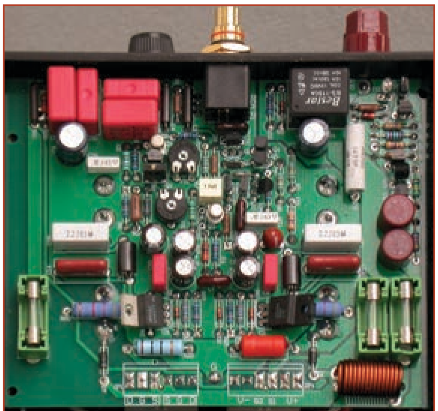


Photo 2: The SE-100 MK2's main PC board houses the input and voltage-gain/driver circuitry, which is based on discrete, single-ended, bipolar technology. This circuitry is regulated with a pair of LM317/337 three-terminal regulators shown near the bottom. The output muting relay in the upper right is activated with a delay circuit that ensures silent turn on and shutdown.

Photo 3: This is an inside view of the SE-100 MK2 amplifier. The new MOSFET output PC boards are mounted on the heatsinks, under the main PC board. The hefty, 450-VA toroidal power transformer and Shindengen low-noise, soft-recovery rectifier bridges are in the front portion of the amplifier.

in appearance (see **Photo 4**). For the newest version of the SM-70 Pro, Monarchy has given the amp the same MOSFET upgrade as the SE-100 MK2, replacing the old 2SK1058/2SJ162 transistors with the heftier 2SK2221/2SJ352 devices (see **Photo 5**). With the new output devices, the amps can drive 3-Ω loads continuously, and 2-Ω loads on an intermittent basis. The SM-70 Pro is a stereo amplifier. In each channel, two of each device are used in a full-complementary, push-pull arrangement. The SM-70 Pro's output stage operates as an open-loop buffer, with zero global feedback. The circuit topology for the output stage includes trim pots for DC offset, which I have found extremely stable over time. The trim arrangement is used in a pair of original SM-70 amplifier that I have in my office at The Crane School of Music, SUNY Potsdam. After nearly 15 years of use, the offset trim has never required adjustment.

a)

b)

Photo 4: A pair of Monarchy SM-70 Pro power amplifiers are pictured with the rear panel (a) showing the unbalanced RCA connectors for stereo connections and a single XLR connector for fully-balanced monaural operation.

The input/voltage-gain circuitry is based on a TI/Burr-Brown OPA2604 FET-input dual op-amp (see **Photo 6**). The SM-70 Pro's maximum voltage output is limited not by the output stage, but by the supply rails powering the op-amp. The OPA2604 is unique in that it can run on up to ±24-V rails, making it ideal for this application. In the amp's original version, Monarchy used trim pots in the dividers that set the LM317/337 regulators's output voltages. I was always uncomfortable with these since it was too easy to exceed the op-amp's absolute maximum supply rating. For the SM-70 Pro's latest version, Monarchy uses fixed resistors and set the supply rails at ±23.5 V. In each stereo channel, Monarchy uses the second half of the op-amp as a unity-gain follower in a current doubling arrangement similar to the one described in Morgan Monks's application bulletin, "Double the Output Current to a Load with the Dual OPA2604 Audio Op Amp," (Burr-Brown AB-051, 1993) on Texas Instruments's website (www.ti.com). The added output current is necessary for the op-amp to drive the output stage to full rated power. The SM-70 Pro's raw power supply is similar to that of the SE-100 MK2, with a 350-VA toroidal power transformer, a pair of Shindengen low-noise, soft-recovery rectifier bridges, and four 15,000-µF capacitors in the raw filter bank.

One welcome change in the SM-70 Pro is the lower voltage gain. In the amp's original version, Monarchy set the op-amp for a voltage gain of 34 (30.6 dB), using feedback/divider resistor values of 33 kΩ and 1 kΩ in the op-amp's feedback circuit. I found this too high, forcing me to operate my pre-amp volume control at a very low setting. In the amp's latest version, the 33-kΩ resistor has been lowered to 22 kΩ, for a voltage gain of 23 (27 dB). The SM-70 Pro can be operated as a bridged mono amplifier, fed from an unbalanced source, or as a fully balanced mono amplifier when fed from a balanced source. Most users will operate this amplifier in a "monoblock" arrangement, since a single stereo SM-70 Pro's power output is fairly modest. But, there are certainly applications for a low-power stereo amplifier with the high-level performance offered by the SM-70 Pro. It would be ideal with high-sensitivity loudspeakers, and it would make an excellent computer audio amplifier, particularly when fed by a USB-connected DAC/pre-amp, such as the NuForce Icon HDP (reviewed in *audioXpress* April 2011) or Benchmark DAC-1 USB (reviewed in *audioXpress* January 2009). It would also be suitable for powering a home theater system's rear channels.

CLASS A DETAILS

Both the SE-100 MK2 and SM-70 Pro power amps have output stages biased for Class A operation up to 15 W of output. They run quite warm in normal operation, but they are a lot cooler than an amp operating pure Class A up to full-rated power. Above 15 W, the output stage slides into Class AB operation. Monarchy President C.C. Poon offers the following comments on Class A operation as it applies to MOSFET output stages: "The Class distinction is a moot point with the MOSFETS we use, which operate into the gigahertz region. Class distinction is much more critical with bipolar transistors, which operate in the megahertz region at best. In other words, the MOSFETs operate thousands of times faster than bipolar transistors, rendering crossover distortion almost immeasurable. Class A operation at full power output becomes unnecessary and actually quite inefficient. The MOSFETS in both the SE-100 and SM-70 Pro also operate purely in current gain mode only. The voltage gain stages in both amps operate in

Photo 5: A look inside the SM-70 Pro power amplifier shows two pair of complementary MOSFET output transistors mounted on each heatsink. The output transistors are soldered directly to the main PC board. The hefty, 350-VA toroidal power transformer and Shindengen low-noise, soft-recovery rectifier bridges are in the front portion of the amplifier.

pure Class A, all the way to full power output."

I would add that operating the output stages Class A up to 15 W also ensures the output devices operate with excellent linearity at low-signal levels.

THE SOUND: SE-100 MK2

The latest Monarchy SE-100 MK2 amplifier version is clearly a more refined amp than its predecessor, improving on an already excellent design. The amplifier's most immediately striking characteristic is its detail and transparency. On well-recorded material, it offers a clean window on the original performance, without ever sounding sterile or overly analytical. The detail level on well-recorded, high-resolution sources is remarkable. Listed in parenthesis are the websites and product numbers where these recordings can be found. Especially noteworthy in this regard is Reference Recordings's amazing 176.4-kHz/24-bit HRx release of Stravinsky's *The Rite of Spring* with Eiji Oue and the Minnesota Orchestra (www.referencerecordings.com, product number HR-70), and the Esoteric SACD remastering of the "Interlude and Dance" from Manuel de Falla's *La vida breve* with Ernest Ansermet and L'Orchestre de la Suisse Romande, a superb Decca/London recording made in Geneva's Victoria Hall in 1961 (www.esoteric.teac.com, product number EESD 90016). The Reference Recordings HRx releases are .wav files supplied on DVD-R discs, intended for playback on a computer music server. But, if you own an Oppo BDP-93 or BDP-95 Universal Player, you can play the .wav files on the Reference Recordings discs. There's no need to use a computer server. (Amplifiers as refined as the SE-100 MK2s deserve a suitable digital source. The Oppo BDP-95 is an excellent complement to these amps.)

The SE-100MK2 excels in revealing subtle harmonic details. I found this particularly striking at the beginning of Dukas's *The Sorcerer's Apprentice* with Charles Munch and the Boston Symphony Orchestra, where the violins play soft, delicate harmonics (www.classical.net, RCA Victor Living Stereo CD 68978-2). Tonally, the amplifier is very well-balanced, and the upper midrange and treble region are silky smooth and utterly natural. The midrange has excellent detail and liquidity, and articulation across the entire spectrum is most impressive. The beginning of the 3rd movement of the famous Vox/Turnabout recording of Rachmaninoff's *Symphonic Dances*, with the Dallas Symphony—conducted by Donald Johanos and superbly engineered by the late David B. Hancock—contains figurations in the high violins that are rendered with uncanny clarity and realism by the SE-100 MK2 amplifiers (www.analogueproductions.com, Analogue Productions SACD CAPC34145 SA).

MOSFET amplifiers are often unfavorably compared to bipolar designs when it comes to low-frequency performance. The SE-100 MK2 should put those biases to rest. This bass region is extended, clean, and well defined. On the Reference Recordings *Rite of Spring*, the bass drums's power, impact, and extension are amazing. The SE-100 MK2 achieves this along with remarkable clarity and control. This amplifier enables you to hear the instruments's characteristics. In the 1980 Telarc recording of the same work with Lorin Maazel and the Cleveland Orchestra—using the Soundstream digital recorder—the bass drums were tuned fairly tightly. (www.classical.net, Telarc SACD-60563).

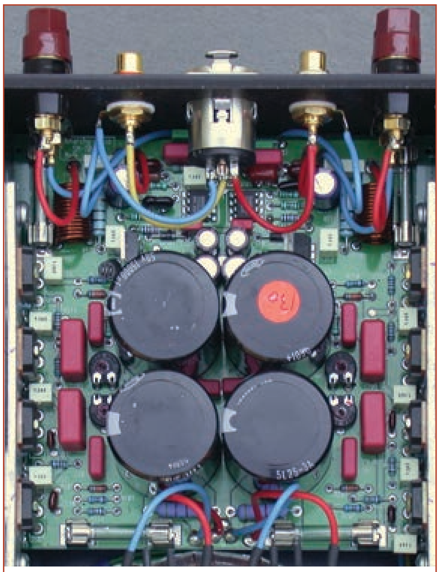


Photo 6: A close-up of the SM-70 Pro's main PC board shows the front-end circuitry, which is based on OPA-2604 dual op-amps running on ±23.5 V, regulated by an LM317/337 pair. Voltage gain is accomplished with conventional feedback, but the amp doesn't employ any global feedback from output to input.

The Minnesota Orchestra's bass drums, in the Reference Recordings version, are tuned more loosely, so they ring longer after the initial impact. The SE-100 MK2 amp enables you to hear the striking differences between these instruments and the recording venues' acoustics.

I've been a believer in mono power amps for many years, primarily due to the excellent sound-staging complete power supply isolation provides. The SE-100 MK2 amps reproduce a large lateral soundstage, with precise localization and a lot of depth. These amps reveal the soundstage differences between conventional Red Book CDs and carefully mastered, high-resolution digital sources. These gutsy amplifiers also deliver dynamics that belie their relatively small footprints, and they never become congested, even on heavily scored orchestral recordings. The detail level I noted on the Reference Recordings *Rite of Spring* is maintained even in the score's very dense passages. Monarchy has also lowered the noise floor on the MK2 amplifiers. The new amps are very quiet.

Having owned a pair of the original SE-100 Delux amplifiers, I highly recommend Monarchy's total upgrade package for existing amplifiers. Monarchy continues its high-value tradition with the SE-100 MK20—a refined audiophile amplifier with high-end performance at a reasonable price.

THE SOUND: SM-70 PRO

I evaluated the SM-70 Pros using a pair of the amps as mono amplifiers, both in bridged mono mode from my unbalanced pre-amp, and in true balanced configuration driven directly by the balanced outputs on my Oppo BDP-95 universal digital player. The SM-70 Pro amplifiers offer many of the virtues of the SE-100 MK2s, namely detail, resolution, excellent soundstage reproduction, and impressive dynamics. Monarchy's goal has been to combine the virtues of solid-state and tube designs, and it's been remarkably successful. I don't use any tube equipment in my system. I prefer the well-designed solid-state equipment's accuracy to the tube's sometimes overly euphonic colorations. Monarchy has succeeded admirably in balancing those two characteristics. The SM-70 Pros have a bit of tube-like warmth, without being excessively euphonic, along with the transparency, detail, dynamics, and bass I expect from a good solid-state amplifier. The tube-like warmth can be captivating on low strings. In the "Gnomus" section of the Fritz Reiner/Chicago Symphony recording of the Mussorgsky/Ravel *Pictures at an Exhibition* the violas, cellos, and contrabasses have a gutsy, authoritative sound that is palpably real (www.classical.net, RCA Victor Living Stereo Gold CD 68571-2).

As with the SE-100 MK2 amps, the SM-70 Pros belie the negative views often held on the bass performance of MOSFETs. The bass is clean, well defined, and powerful. Although the SM-70 Pro amps are rated at less power output than the SE-100 MK2, you'd never know it from the sound. Advocates of zero-global feedback designs often cite the sense of unrestrained dynamics as one of their virtues. The SM-70 Pros certainly leave

SPECIFICATIONS

SE-100 MK2

Damping Factor
- Better than 600 from 10 Hz to 400 Hz

Dimensions
- 9" × 5" × 12"
- Net weight: 22 lb
- AC voltage: 117/234 V, 50/60 Hz

Frequency Response
- –0.1 dB, 20 Hz to 20 kHz at 1 W
- Signal-to-noise ratio: Greater than 120 dB below rated FTC full bandwidth power
- Slew rate: 50 V µs
- Input impedance: 40 kΩ

Input Sensitivity
- 1.5 V_{PP} for full output

Intermodulation Distortion (IMD)
- Less than 0.05% from 250 mW to full-rated FTC power

Power Consumption
- 400 W

Power Output
- 8-Ω load: 100 W RMS
- 4-Ω load: 200 W RMS
- Power bandwidth: –3 dB from 5 Hz to 100 kHz

Total Harmonic Distortion (THD)
- Less than 0.01% at full rated FTC power from 20 Hz to 20 kHz

Voltage Gain
- 26 dB

SM-70 Pro

Dimensions
- 9" × 5" × 12"
- Net weight: 20 lb
- AC voltage: 117/234 V, 50/60 Hz

Frequency Response
- –0.25 dB, 20 Hz to 30 kHz
- THD: less than 0.05%
- Signal-to-noise ratio: Better than 90 dB in stereo mode; better than 100 dB in balanced mono mode

Input Impedance
- 75 kΩ

Input Sensitivity
- 0.7 V RMS for full output

Minimum load impedance:
- 3 Ω continuous
- 2 Ω intermittent (stereo or mono)

Power Consumption
- 200 W

Power Output
- Stereo mode:
  8-Ω load: 25 W RMS × 2
  4-Ω load: 40 W RMS × 2
- Mono mode:
  8-Ω load: 80 W RMS
  4-Ω load: 140 W RMS

me with that impression. Like the SE-100 MK2s, orchestra strings are smooth and sweet without a trace of harshness. The soundstaging is superb. Once you've experienced separate mono amplifiers, you'll never go back to a stereo-power amp.

The SM-70 Pros perform at their best when operated as mono amps driven from balanced sources. When I reviewed the Oppo BDP-95 Universal Disc Player (*audioXpress* January 2012), I mentioned trying the SM-70 Pros directly driven directly from the Oppo's balanced output. The ESS Sabre-32 DAC used in the Oppo has an internal 32-bit volume control, which is adjusted using the BDP-95's remote control. At the time I reviewed the Oppo, the volume control operated in rather coarse 5-dB increments, but the most recent firmware upgrade to the player changes this to 1-dB increments. This makes the player ideal for driving a power amp directly if the digital player is your only source. The BDP-95/SM-70 Pro combination performs superbly, offering exceptional detail level, transparency, and soundstage reproduction.

EVALUATION EQUIPMENT

- Custom-built, belt-driven turntable with Grado signature tone arm and Grado Signature XTZII moving iron cartridge: *Audio Amateur*, March 1985 and March 1988; *audioXpress* July 2008
- Electronic power supply/speed control for the custom-built, belt-driven turntable (with recent upgrades): *Audio Amateur*, January 1986
- Audio Concepts Sapphire III loudspeakers and Sub-1 subwoofers with custom, all-polyprpylene subwoofer-to-satellite crossovers: *Speaker Builder*, March 1991 and March 1995
- D.H. Labs Air Matrix interconnect cables (for unbalanced and balanced cables); D.H. Labs Q-10 loudspeaker cables, including bi-wiring for the Sapphire IIIs, and separate runs to the Sub-1 subwoofers: *audioXpress*, October 2002
- Custom-built pre-amplifier on Adcom GFP-565 chassis: *audioXpress* November 2003, December 2003, January 2004, February 2004, and December 2004
- D.H. Labs Power Plus AC cables with Marinco/Wattage connectors: *audioXpress*, February 2005
- Three PS Audio Power Plant Premier AC regenerators: *audioXpress*, April 2010
- Three dedicated AC power lines—one for each power amplifier and one for low-level equipment: *audioXpress*, January 2011; *Multi-Media Manufacturer*, July–August 2010
- Oppo BDP-95 Universal Blu-ray digital player: *audioXpress*, January 2012

PERSONAL PREFERENCE

Making a decision between these amplifiers is difficult and will come down to personal taste and your specific listening requirements. If you have a true balanced source to feed your power amps and you prefer a tasteful amount of tube-like warmth in your sound, you should strongly consider the SM-70 Pro. But, if your source is unbalanced and your tastes tend to favor ultimate accuracy over what is admittedly a bit of euphonic coloration, the SE-100 MK2 will be your choice. Overall I'd rate the SE-100 MK2 as slightly more transparent, and the SM-70 Pro as slightly warmer. Either amp will provide hours of musically satisfying, nonfatiguing, high-end sound, at prices that continue Monarchy's high-value tradition. If you'd like to help the trade deficit, Monarchy is an American company, with its power amplifiers hand-assembled in its San Francisco, CA facility. These amps are highly recommended. *aX*

MEMBER PROFILE


Mark Driedger

Member Name: Mark Driedger

Location: McKinney, TX, north of Dallas

Education: Mark studied Electrical Engineering at University of Waterloo, Canada, graduating with a BSc and MSc. His graduate studies focused on communications and digital signal processing applied to speech.

Occupation: Mark worked in the telecom industry for 24 years. He spent the first 11 years in engineering and management roles building wireless base stations for the early digital cellphone systems. He is currently a VP for a division of Tektronix, responsible for a large engineering, service, and manufacturing organization.

Member Status: Mark has subscribed to *audioXpress* since 2002. In 1977, he discovered *Elektor* at a local Canadian newsstand. Each month for the next five years, the newsstand put aside a copy for him. He said he spent many hours studying and building projects from *Elektor*.

Affiliations: Mark presently has no affiliations, although he is a past member of the Institute of Electrical and Electronics Engineers (IEEE).

Audio Interests: He is primarily interested in tube power amps, both for Hi-Fi and guitar.

Most Recent Purchase: Mark's most recent purchase was a USB DAC kit from Hifidiy.com. He also bought a couple NAD solid-state power amps that he restored to power some outdoor speakers.

Current Audio Projects: Mark said he is working on a KT-88-based, tube power amp with an integrated USB DAC. He completed the power supply and output stages and he is working on the driver stages.

Dream System: Mark said he receives a lot of satisfaction listening to what he builds. So, his dream system would be the tube amp on which he is currently working, if/when he gets it finished! He uses a computer as a source, with all his CDs stored in lossless format. Mark said his dream system would also include a speaker upgrade from his PSBs—perhaps a pair of Focals. If he went with a commercial amp, he said he would choose a classic Marantz or Luxman tube amp. *aX*

 By Vance Dickason (United States)

B&C's New 18" Woofer

A high power-handling speaker with a ferrite motor is put to the test

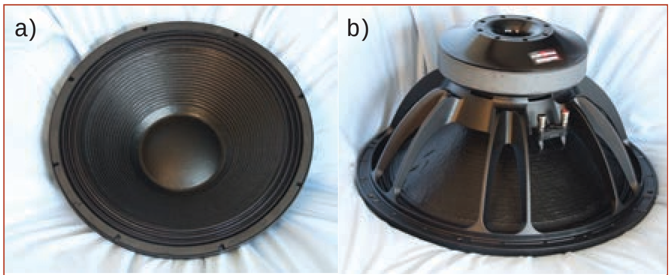


Photo 1: The B & C Pro Sound woofer's top (a) and bottom (b) view

B&C's new 18" woofer, the 18TBW100-8 is another in a series of high, continuous, power-handling rated pro sound woofer/subwoofers (anything that goes much lower than 35 to 40 Hz is pretty much a subwoofer in the pro sound world) B&C has been producing over the last several years (see **Photo 1**). Like the B&C 18SW115-4 rated at 3.4-kW continuous program material (*Voice Coil*, May 2010), the 18TBW100 has a high continuous-power handling rating of 3 kW. However, unlike the neodymium motor 18SW115, the 18TBW100 uses a ferrite motor design, which is probably going to be the trend in pro woofer design until neo's prices drop to a more competitive level. In terms of weight, the neo 18SW115 weighs 22.6 lb, while the ferrite motor 18TBW100 weighs 33.3 lb (a difference of 10.7 lb), which is certainly more weight, but not a deal breaker by any means.

As expected, the feature set for this kind of performance is substantial, starting with a proprietary cast frame. Like many of B&C's pro sound drivers and pro sound and high-powered car audio woofers, the frame is very much part of the cooling system. This frame has six double spokes, or 12 spokes, thermally coupled to the large ceramic ring magnet motor structure. Venting is fairly complex and includes six 8 mm × 38 mm vents around the peripheral of the base of the frame that couples air to the area between the spider-mounting shelf and the front plate. Additional cooling is provided by a flared 45-mm diameter pole vent, plus eight peripheral 8-mm diameter vent holes on the back of the motor cup, which, when taken together, provide substantial air flow throughout the motor structure.

The cone assembly consists of a ribbed 18" cone that is coated on both sides, plus a large 6"-diameter convex coated dust cap. Note that this coating is a special TWP waterproof coating. Compliance is provided by a triple-roll coated cloth surround and a double-silicone coated cloth spider assembly. Coupling the cone assembly to the motor is a 4"- (100-mm) diameter glass fiber split winding voice coil wound with round copper wire.

A large 30-mm thick, 21.8-cm diameter ceramic ferrite ring magnet provides the horsepower. Other motor features

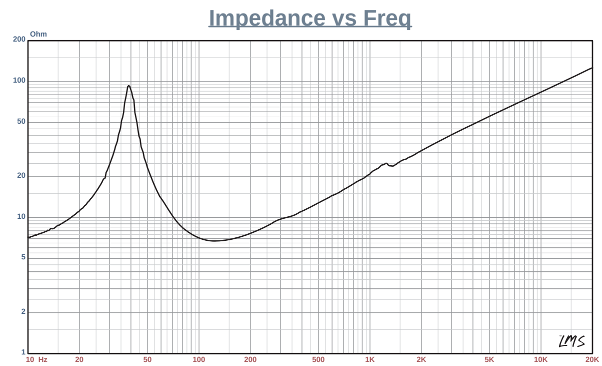


Figure 1: The B&C 18TBW100-8 woofer free-air impedance plot

	TSL model		LTD model		Factory
	sample 1	sample 2	sample 1	sample 2	
F_S	39.1 Hz	39.1 Hz	37.5 Hz	37.6 Hz	35 Hz
R_{EVC}	5.38	5.37	5.38	5.37	5.3
Sd	0.119	0.119	0.119	0.119	0.121
Q_{MS}	8.25	7.72	7.47	7.03	8.00
Q_{ES}	0.47	0.48	0.44	0.44	0.41
Q_{TS}	0.45	0.46	0.41	0.42	0.39
V_{AS}	136.0 ltr	136.0 ltr	149.5 ltr	148.1 ltr	175 ltr
SPL 2.83 V	94.3 dB	94.1 dB	94.4 dB	94.4 dB	96.0 dB
X_{MAX}	11 mm	11 mm	11 mm	11 mm	11 mm

Table 1: The LEAP 5 LTD, TSL data, and factory parameters are compared for both B&C 18TBW100-8 samples.

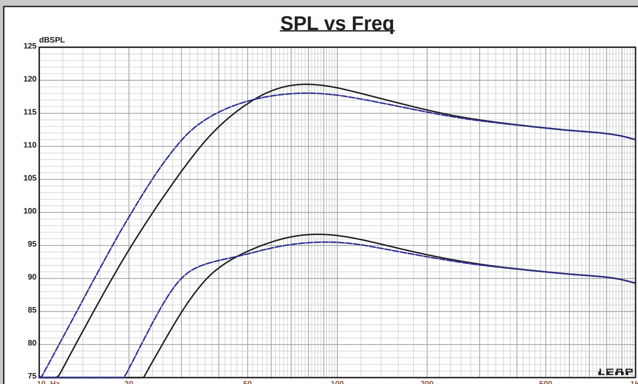


Figure 2: The B&C 18TBW100-8 computer box simulations (black solid = sealed 1 @ 2.83 V; blue dash = sealed 2 @ 2.83 V; black solid = sealed 1 @ 7 V; blue dash = sealed 2 @ 5 V)

include a T-shaped pole and an aluminum demodulation ring (i.e., shorting ring or faraday shield). Last, the voice coil tinsel wires are connected to a set of chrome, color-coded push terminals.

I began testing the 18TBW100-8 using the LinearX LMS and VIBox to produce both voltage and admittance (current) curves with the driver clamped to a rigid test fixture in free-air at 1 V, 3 V, 6 V, 10 V, 15 V, 20 V, 30 V, and 40 V. Note the driver remained linear in free air up to the 40-V sweep and probably would have remained linear up to 50 V or 60 V, but with 96-dB sensitivity, 40 V is usually my limit wearing ear protectors. Also, please note I use a procedure that attempts to achieve the third time constant on each sweep, the LMS oscillator is turned on for a progressively increasing time period between sweeps. Also, following the established Test Bench testing protocol, I no longer use a single added-mass measurement and instead used actual measured cone assembly weight provided by B&C.

Next, 16 550-point stepped sine wave sweeps were post-processed for each sample. The voltage curves were

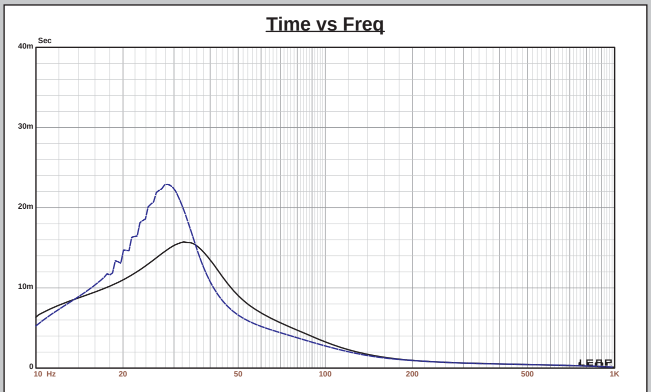


Figure 3: Group delay curves for the 2.83-V curves in Figure 2

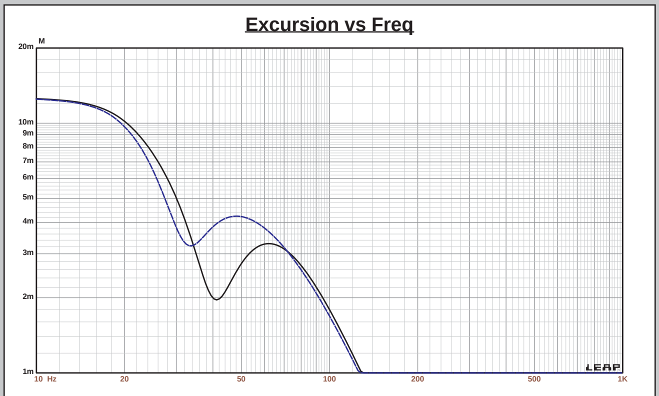


Figure 4: Cone excursion curves for the 40-V curves in Figure 2

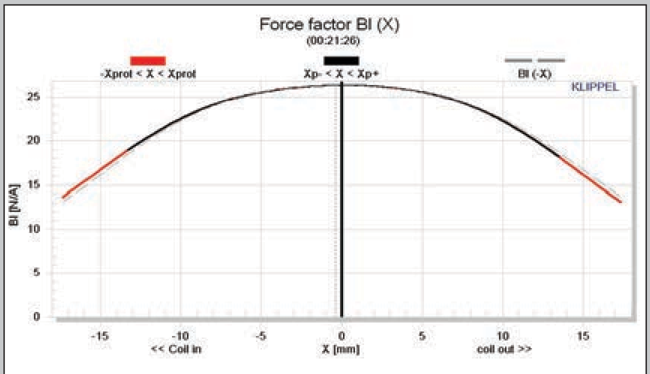


Figure 5: Klippel analyzer Bl(X) curve for the 18TBW100

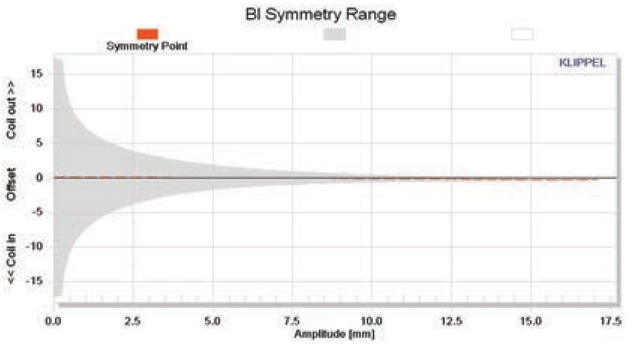


Figure 6: Klippel analyzer Bl symmetry range curve for the B&C 18TBW100-8

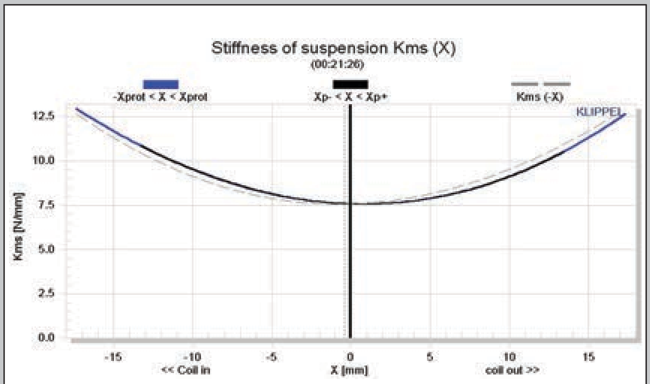


Figure 7: Klippel analyzer mechanical stiffness of suspension K_{MS} (X) curve for the B&C 18TBW100-8

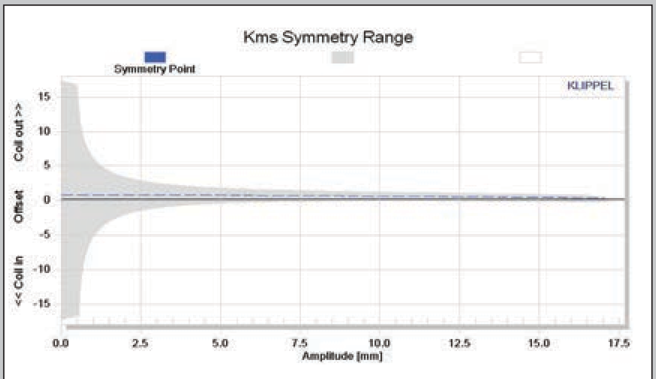


Figure 8: Klippel analyzer K_{MS} symmetry range curve for the B&C 18TBW100-8

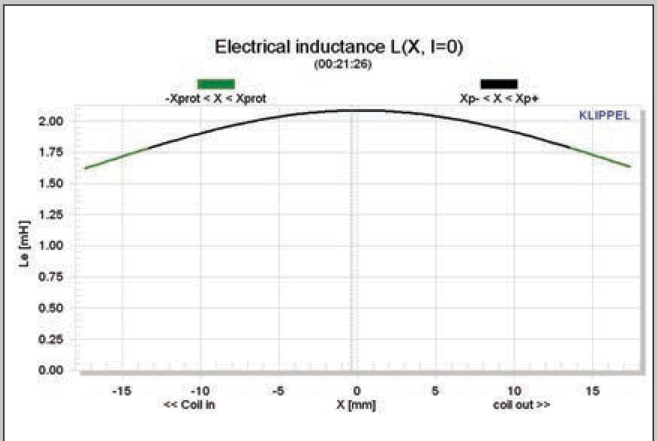


Figure 9: Klippel analyzer Le(X) curve for the B&C 18TBW100-8

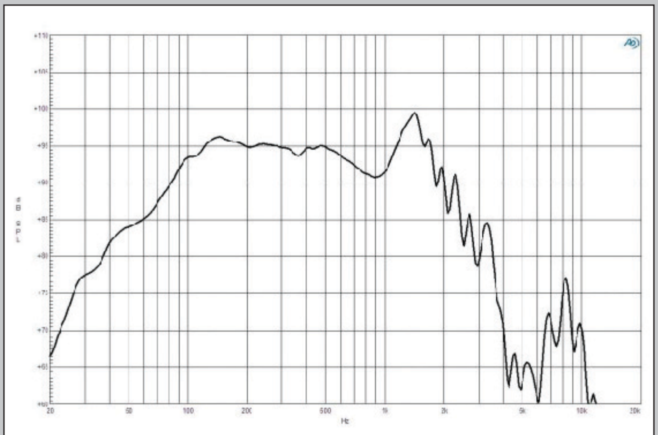


Figure 10: B&C 18TBW100-8 on-axis frequency response

divided by the current curves to derive impedance curves, phase calculated along with the accompanying voltage curves, and imported to LEAP 5 Enclosure Shop software. Obviously, this is a more time consuming process than the usual low-voltage impedance curve used for deriving Thiele-Small parameters. The reason for this, if you haven't been following this column for a number of years, is that the LEAP 5 LTD transducer model methodology results in a more accurate prediction of excursion at high-voltage levels, one of the real fortes of the LEAP 5 software.

Because most TS data provided by OEM manufacturers is being produced using either a standard method or the LEAP 4 TSL model, I additionally created a LEAP 4 TSL model using the 1-V free-air curves. The complete data set shows the multiple voltage impedance curves for the LTD model. See **Figure 1** for the woofer 1-V free-air impedance curve. The 1-V impedance curves for the TSL model were selected in the Transducer Derivation menu in LEAP 5 and the parameters were created for the computer enclosure simulations. **Table 1** compares the LEAP 5 LTD, TSL data, and factory parameters for both B&C 18TBW100-8 samples.

Parameter measurement results for the 18TBW100 Q_{TS} numbers for my measurements were somewhat higher and the V_{AS} was somewhat lower, plus the sensitivity data I derived was 1.6 to 1.8 dB lower than the factory data. However, my sensitivity rating is at 2.83 V, and the factory is 1 W/1 m based on a 5.3 Ω Re. While not absolutely identical, my parameter measurements were obviously close to the factory data. This is almost always the case when I measure B&C products, which not only speaks to the excellent group of transducer engineers at the company, but also to the fact that it makes extensive use of its Klippel analyzer during the product development stage. Given this, I proceeded to set up two computer enclosure simulations using the LEAP LTD parameters for Sample 1. This included two vented alignments, a 4.05 ft^3 Chebychev/Butterworth type box with 15% fiberglass fill material tuned to 42 Hz, and the factory recommended 7.06 ft^3 vented alignment enclosure with 15% fiberglass fill material tuned to 32 Hz. This was actually an extended bass shelf (EBS) type of alignment.

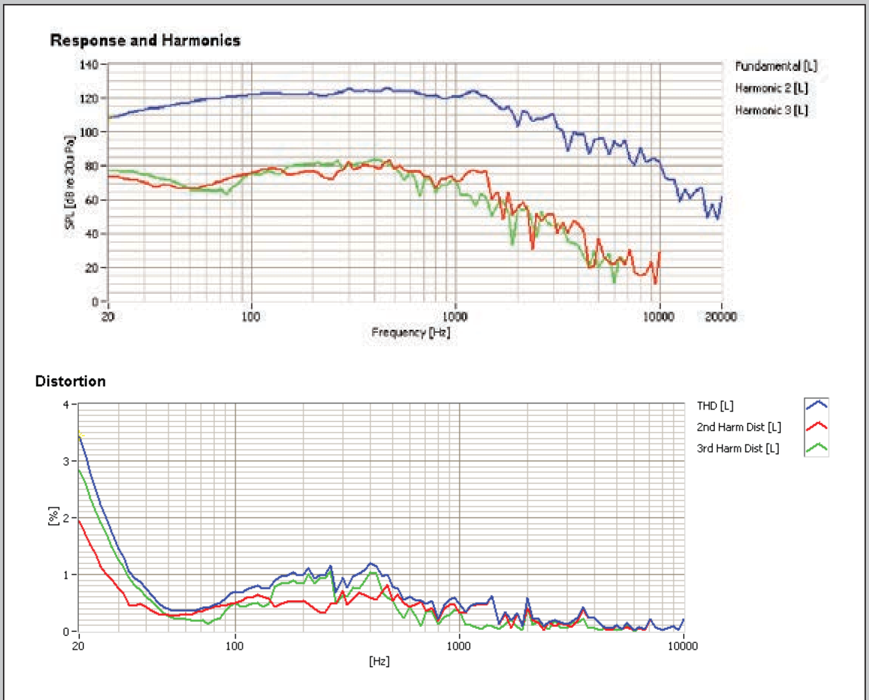


Figure 11: The B&C 18TBW100-8 SoundCheck distortion plot

Figure 2 displays the results for the 18TBW100-8 in the Chebychev/Butterworth and EBS vented boxes at 2.83 V and at a voltage level sufficiently high enough to increase cone excursion to X_{MAX} + 15% (9.2 mm for the 18TBW100). This produced a –3-dB frequency of 47.0 Hz (i.e., –6 dB = 38.0 Hz) for the 4.05 ft^3

C/B enclosure and F3 = 38.0 Hz (i.e., F6 = 29.0 Hz) for the 7.06-ft^3 factory-recommended EBS-vented simulation. Increasing the voltage input to the simulations until the maximum linear cone excursion was reached resulted in 119.5 dB at 40 V for the C/B enclosure simulation and 118 dB the same 40 V input level for the larger vented box. See **Figure 3** and **Figure 4** for the 2.83-V group delay curves and the 40-V excursion curves. If you look at the excursion curves, you will see I cut the simulation off at 40 V mostly because the excursion curve reached 9.2 mm at close to 20 Hz.

Klippel analysis for the B&C 18TBW100-8 produced the Bl(X), K_{MS}(X) and Bl and K_{MS} symmetry range curves shown in **Figures 5–8**. The Bl(X) curve for the 18TBW100 is very broad and extremely symmetrical (see **Figure 5**). It is what a loudspeaker engineer would call "picture perfect" since both the Bl curve and its accompanying offset curve are almost exactly overlaid. Looking at the Bl symmetry plot, the offset curve is mostly invisible since it tracks at the 0 offset level out to 10 mm and only slightly below (coil-in rearward offset) beyond that (see **Figure 6**). **Figure 7** and **Figure 8** show the K_{MS} (X) and K_{MS} symmetry range curves for the 18TBW100. The K_{MS} (X) curve is as symmetrical as possible in both directions, with a trivial 0.75-mm offset, which is further confirmed by the K_{MS} symmetry curve, almost a straight line at the zero rest position tracking again at a nonsignificant 0.75-mm coil-out (forward) offset. Great job, guys!

Displacement-limiting numbers calculated by the Klippel analyzer for the 18TBW100 were XBl @ 82% (Bl decreasing to 82% of its maximum value) 10.7 mm, and for XC @ 70% (compliance decreasing to 70% of its maximum value) was 11.5 mm, which means that for this B&C 18" woofer, the Bl is the most limiting factor for a prescribed distortion level of 10%, but for either element, the contribution is several millimeters beyond the physical X_{MAX}. If I use the subwoofer criteria for 20% distortion, the numbers are XBl greater than 13.3 mm and XC greater than 13.3 mm.

Figure 9 shows the inductance curve L(X) for the B&C pro sound subwoofer. Inductance will typically increase in the rear direction from the zero rest position as the voice coil covers more pole area unless the driver incorporates a shorting ring. Since the 18TBW100 does incorporate an aluminum demodulation ring (shorting ring) in the motor assembly, there is only an extremely minor change in inductance, about 0.15 mH, from X_{MAX} in to X_{MAX} out.

Since the B&C 18TBW100 will likely be used in a subwoofer configuration for use mostly below 100 Hz to 150 Hz, I dispensed with the SPL measurements, but also because I don't keep 18"-or-21"-size cabinets in my inventory of test fixtures. However, **Figure 10** shows the factory SPL curve, and obviously you could cross the product over certainly higher than 100 Hz. Without the off-axis response it's not possible to guess exactly what that frequency would be, but probably 500 to 800 Hz.

Since I decided not to perform SPL measurements, I moved on to the last group of tests, which were performed using a SoundCheck analyzer, an SC-1 microphone, and SoundConnect power supply (courtesy of Listen) to measure distortion. Again, because we are not dealing with frequencies much above 100 Hz as a subwoofer, I also did not use SoundMap software for time frequency presentations. Setting up the distortion measurement consisted of mounting the woofer rigidly in free-air, and setting the SPL setting to 104 dB at 1 m, using a noise stimulus. (SoundCheck has a software generator and SPL meter has two of its utilities.) Then I measured the distortion with the microphone placed 10 cm from the dust cap. This produced the distortion curves for this woofer (see **Figure 11**).

B&C offers a well-designed, robust 18" woofer for pro-sound applications. For more information, contact B&C Speakers N.A., B&C Speakers NA LLC, 220 West Parkway, Pompton Plains, NJ, 07444 USA, (973) 248-0955, e-mail info.usa@bcspeakers.com, or visit the B&C Speakers website at www.bcspeakers.com. *aX*