SPECIAL FOCUS ON SPEAKERS , p. 17–33

SEPTEMBER 2013

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From the Editor's Desk

Teacher and Innovator

We'd like to note the recent passing of world-renowned pro-sound innovator and longtime MIT professor Amar Bose. The sound and electrical engineer founded Bose audio company in 1964.

News of his death in July, at age 83, came as this issue was in production. Since that time, much has been written about Bose's contributions. Jan Didden, editor/publisher of *Linear Audio,* will explore Bose's legacy more deeply in an article in next month's issue of *audioXpress.*

In the meantime, here is a small tribute to Bose. His many accomplishments include inventing the 901 Direct/Reflecting Speaker System in 1968. The speakers were among the first to utilize the space around them, producing a mix of direct and reflected sound that recreated the dynamics of concert hall acoustics. Other popular innovations followed, including the Bose noise-canceling headphones and the Bose Wave radio.

But Bose was also a committed and beloved teacher. In 2011, he gave MIT the majority of stock in the Bose company, as nonvoting shares. The dividends must be used to support MIT's educational and research missions.

To hear Bose tell his own story, watch his 1996 talk titled "Experiences of an Academician in Industry," available on the MIT website at http://video.mit.edu.

Regards,
Mary Wilson
editor@audioxpress.com

Editor's Note: Figures 2 and 3 in Vance Dickason's Test Bench column ("Beston's Horn-Loaded Transducer," audioXpress, July 2013) were incorrect. The correct figures for the Beston Technology RT004A ribbon tweeter are available online at audioamateur.com/errata-corrections-updates.

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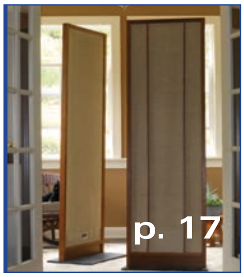

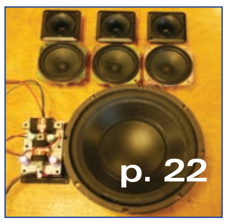

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By Mike Klasco and Steve Tatarunis (United States)

Loudspeaker Failures and Protections (Part 2)

Passive Speaker Protection

Protection techniques to prevent overpower damage

In last month's article about speaker failures, we introduced the topic by describing what occurs in your speaker when it is pushed to its mechanical and thermal limits. We also mentioned that there is a new generation of effective and powerful speaker projection technology.

Sophisticated loudspeaker protection signal processing for concert sound was pioneered by Meyer Sound and Apogee Sound in the 1980s. Now these complex techniques have found their way into smartphones, laptops, tablets, Bluetooth speakers, and docking stations. This month we will discuss passive protection techniques used between the amplifier and the speaker.

For speaker designers, loudspeaker overpower protection has always been a problem, particularly in delicate high-frequency drivers. As voice coil adhesive temperature capabilities have increased, so has amplifier power output. While ferrofluids have helped move heat off the voice coil, higher power amplifier clipping and wide dynamic range program sources (e.g., CDs and lossless digital recording) have raised the bar on what speakers must tolerate.

SPEAKER PROTECTION

In the 1970s, Mike Klasco owned an audio company that pioneered disco speakers. The speakers needed to have high sensitivity, wide bandwidth, and high power handling at a cost-effective price to compete with the established pro audio speaker brands. Mike will take over the narrative from here.

At the time, the big brands were primarily selling their products to professionals. But my firm sold products to audio retailers, which sold them to DJs and club owners. Although the DJs knew their music, they were not as wise about getting high sound pressure levels (SPLs) out of their gear. Remember, 40 years ago power amplifiers did not overload gracefully and they created even more serious issues (e.g., turntable acoustic feedback and record warps, which provided subsonic speaker problems.)

Photo 1: A typical E-T-A thermal circuit breaker is shown. (Photo courtesy of E-T-A Engineering Technology)

With that in mind, I built in "global protection" of the entire speaker system, including the woofers and crossover network, and fast "local protection" of the midrange and tweeters.

For the overall protection, I used a thermal auto resetting circuit breaker on the entire speaker system (see **Photo 1**). E-T-A Engineering Technology was the vendor at the time, and in fact, it still makes this type of product today.

Inside the circuit was an internal coil with an impedance that would shift upward as it heated. I empirically selected the breaker's value to protect the product, enabling it to shut off before the speaker was permanently damaged. The speaker would turn back on in a minute or two, after the breaker coil and the speaker's voice coils cooled down.

While this method is not for audiophile or studio monitor use, the device prevented field failures that would be devastating to a DJ. The brief interruption was sort of a slap on the DJ's hand rather than a death blow to the loudspeakers and the party.

Aside from a breaker that would shut down the speaker if it was driven too hard over an extended time, the compression drivers were still getting damaged because the breaker time constants were intentionally slow acting. In other words, the breakers were able to protect the woofers but they did not adequately protect the mid- and high-frequency drivers from the peak signal levels.

The array of piezo tweeters could survive if we put a resistor in series to work as an ultrasonic stabilizing network. The piezo tweeters have a falling impedance at ultrasonic frequencies and some amplifier output stage protection circuits "chattered" when overdriven into this lossy capacitive low-impedance load without a series resistor, which tended to burn out just about everything.

We used General Electric's new (in the mid-1970s!) metal oxide varistor (MOV) to protect the midrange compression driver (see **Photo 2**). At the time, we had a unique protection scheme for the MOV, which worked better than back-to-back Zener diodes. Zener diodes are hard clippers that

Photo 2: A metal oxide varistor can protect the midrange compression driver. (Photo courtesy of EPCOS AG)

sound terrible when they cut in. We soon realized the MOV's Joule power dissipation ratings had to be the largest possible or they would fail. We eventually put a copper heat tab on one of the leads and used large copper traces on the circuit board. This not only enabled the devices to survive, but somewhat maintained their operating characteristics while they were in use.

Electro-Voice, Community Professional Loudspeakers, and Cerwin-Vega adopted light bulb protection, which is still popular today in high-output loudspeaker products (see **Photo 3**). In the 1980s, Raychem (now Tyco Electronics) introduced its PolySwitch polymeric positive temperature coefficient (PPTC) protection device (see **Photo 4**). The simplest configuration consists of a PolySwitch PPTC device in series with a high-frequency driver. Some designers would like to reduce the power by a smaller fixed amount in case of overdriving, in which case the PPTC is used with a parallel shunt resistor (ideally, in the form of light bulbs as a sort of a passive limiter). When the threshold is exceeded, the output drops. With the time constants, the result is imperceptible. You don't hear its operation, but you can avoid damaging the driver and experiencing any subsequent distortion.

This sounds good, plus you now have a light bulb in your bill of materials! The light bulb is typically used only for its positive temperature coefficient (PTC) effect, but it can also be used as an overload indicator if it is mounted to a panel where it can be seen. An LED in series with a resistor can also be used as an overload indicator, but it does not have any PTC effect. The choice between a shunt resistor and a light bulb resistor, or the choice to use nothing at all in parallel with the PPTC device, depends on the speaker designer's protection philosophy.

For a compression driver's biamplified speakers, you could also use a series capacitor set to roll off the signal an octave below the crossover point. This technique provides the compression driver with some protection from hum, DC offsets from the amplifier, and other unintended energy.

Photo 3: Loudspeaker protection light bulbs are sometimes used in high-output speaker products. (Photo courtesy of JBL)

Photo 4: The PolySwitch helps protect sensitive electronics from damage caused by overcurrent and overtemperature events. (Photo courtesy of Tyco Electronics)

In the ensuing years, there have been several ambitious semiconductor designs intended to protect the speakers, and the protection circuit was driven by the amplifier's signal power. The most recent and perhaps most ambitious and effective commercialized product is Eminence Speaker's D-fend. The D-fend is a programmable standalone unit designed to protect passive loudspeakers from excessive power conditions. The user sets the thresholds and D-fend monitors and limits the amount of input power it passes through to the loudspeaker. It's USB compatible and can be programmed from a desktop or laptop. Operating from a standard speaker-level signal, the D-fend requires no auxiliary power unless it is being used in low-power applications. This protection device works by reducing the power delivered to the loudspeaker, while also reducing the net electrical current required from your amplifier. In other words, the D-fend technology does not dissipate the undesired power into heat or light output; rather, it reduces the amount of power that is requested from the amplifier through the use of high-efficiency digital factoring (see **Photo 5**).

Photo 5: Eminence's D-fend SA300 enables you to set the thresholds. D-fend monitors and limits the amount of input power it passes through to the loudspeaker. (Photo courtesy of Eminence Speaker)

PROTECTION LIMITATIONS

Aside from sophisticated and expensive solutions (e.g., the D-fend), passive solutions are somewhat clumsy and have some negative impacts on sound quality. Because they must face the full brunt of the amplifier, robust components should be used, but they are expensive.

Next month, in Part 3 of this article series, we will discuss the advanced signal processing speaker protection and explore active protection and the psychoacoustic techniques used to increase apparent loudness. *aX*

RESOURCE

D-fend Loudspeaker Protection Technology, "About D-fend," www.d-fend.net.

THE PPTC AND SPEAKER PROTECTION

PolySwitch is the trade name for a polymeric positive temperature coefficient (PPTC) overcurrent protection device, which acts as a self-repairing fuse by nonlinearly going from a low-resistance to a high-resistance state in response to an overcurrent. This is called "tripping" the device. In the tripped state, the device resistance is proportional to the square of the applied voltage—until the device resistance reaches its maximum (see **Figure 1**). This is where some of the more inexpensive PPTCs fail to measure up. While generically referred to as "resettable fuses," technically they are not fuses. They are nonlinear positive-temperature coefficient thermistors.

At normal operating temperatures, these polyswitch devices have resistances from 30 to 800 mΩ. They essentially have no capacitive or inductive reactance nor will they cause any measurable distortion over the audio range of frequencies. The PPTC rating should be selected so its time-to-trip at any particular current is less than the time required to damage the driver at that current.

When the current flowing through the PPTC device exceeds the current limit of 1 A, for example, the PPTC device warms up above a threshold temperature. The crystalline structure starts to degrade and the PPTC device's resistance suddenly "trips" (i.e., its resistance increases from a few ohms to a few kilohms). The time-to-trip varies based on how much voltage is present. "Low-fault" currents of two to three times the "hold" current take some time to heat up the PPTC and depend on local heat transfer (e.g., copper tabs or "wings"). High-fault currents, say 10 times the hold current, trip much faster, since there is no time for the heat to be conducted away from the device.

If too much current passes through the device, the device will begin to heat due to I^2R heating. As the device heats, it will expand. As it expands, the carbon particles will separate and change phase and the device's resistance will increase. This will cause the device to heat faster and expand more, further raising the resistance. This increase in resistance is sufficient to substantially reduce the circuit's current. A small amount of current will still flow through the device and is sufficient to maintain the device's temperature and keep it at the high resistance level.

When the power and fault are removed, the PolySwitch device will cool. As the device cools, it contracts to its original shape and returns to a low resistance level where it can hold the current specified for the device. The actual time to reset will vary based on the rate at which heat can transfer out of the device, but it typically will take 1 to 2 min.

Raychem's (now Tyco Electronics's) PolySwitch is a bit expensive for most consumer speaker products. There are also some inexpensive Chinese-made PPTC devices, some of which are marginal in stability and distortion, but they cost a fraction of the Raychem device's price. There have been patent suits in the past, but at this point, the original patents have expired and the general design is in the public domain.

Raychem itself was sold in 1999 and Raychem PPTC devices are now manufactured and sold by Tyco Electronics, a TE connectivity company. Today there are several vendors for this component (e.g., Sea & Land from Sea Galleon, Semifuse from ATC Semitec, Fuzetec from Fuzetec Technology, Polyfuse from Littelfuse, and Multifuse from Bourns), but product characteristics vary depending on the vendor's specific implementation of the technology.

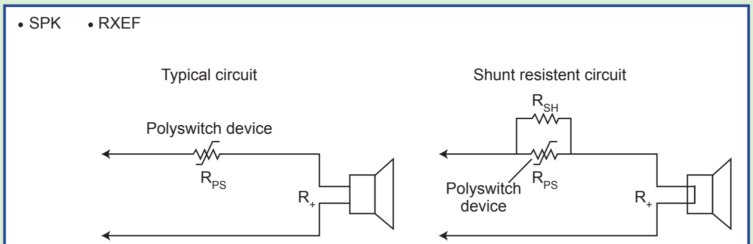

Figure 1: Tyco Electronics offers two different schematic designs for polymeric positive temperature coefficient (PPTC) speaker protection.


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Morel Founder Finds Inspiration in Innovation

Meir Mordechai attributes the company's success to its revolutionary designs

Morel founder Meir Mordechai (left) with his son Oren Mordechai. Oren leads Morel's R&D and many of Morel's latest innovations. (Photo courtesy of Morel)

SHANNON BECKER: What sparked your initial interest in audio?

MEIR MORDECHAI: Music has always been a big part of my life. The love of music and my curiosity as a child to know how musical instruments play and produce different sounds led me to explore the audio field and experiment with electronics and speakers.

I'm still as passionate today about music as I was then. I regularly attend concerts and live performances to make sure that my ears are "tuned" to what musical instruments sound like, unedited, in their natural state, if you will.

SHANNON: Describe your first personal loudspeaker project. Why did you build it? Is it still in use?

MEIR: I was relatively young when I began my research and building projects. It was at the age of 11 when I tried to improve a speaker by changing its cone for another that I put together from wood veneer.

In my generation, information or electronic products were not readily available as they are today. The lack of access to valuable resources was possibly a key reason that motivated me and many others to build and innovate.

SHANNON: Your company, Morel (www.morelhifi.com), headquartered in Israel, has introduced major technological and design innovations that redefined the state-of-the-art loudspeaker technology. Can you discuss some of them?

MEIR: The desire to innovate has always driven me forward. I realized a long time ago that replicating a product would not necessarily generate business and definitely wouldn't differentiate me from the competition. The principles of investing significant resources in new product development and in innovating and designing new ways to build speakers has been the core value behind Morel to this day and contributes greatly to our success.

There are several key technologies that were developed over the years that became the platform for our products. Our large external voice coils (EVCs) are one of the signature elements in our speakers. It turned traditional speaker design inside out. The idea was to place the magnetic drive system within the voice coil, eliminating stray magnetic flux by effectively directing all the magnetic energy to the voice coil. The result is an ultra efficient and powerful design that is highly compact with efficient heat dissipation and reduced cone breakup for lower distortion.

Because our magnet systems are placed within the large-diameter voice coils, we always had to invent new ways to achieve more magnetic energy from small magnets. Developing the double and hybrid magnet systems has enabled us to continue to enjoy the great acoustic benefits of the large-diameter voice coils.

There are many more technologies that were developed and employed in our products over the years. We continue to refine these technologies with newly developed materials that become available to us. At the same time, we keep improving the production process to increase consistency and efficacy. We never cease to research and find new ways to improve the sound reproduction; it's a never-ending journey.

SHANNON: Tell us about Morel's very first product. How did it come about and is it still being sold today?

A glimpse into a soon-to-be released driver. Meir Mordechai meticulously inspects his latest development—a titanium-based subwoofer (which is a new addition to the Ti driver series) during its cone-assembly process. (Photo courtesy of Morel)

MEIR: The first Morel-branded product I made was a 9″ mid-bass driver, which employed a 3″ coil with a paper composite cone and a single ferrite magnet motor system. Later on the development of the double-magnet system replaced the single-magnet system, immensely improving the product performance. The earlier version of the 9″ driver became the foundation for several products that employ some of the original attributes of the design to this day.

SHANNON: Morel has been in business for 38 years. To what do you attribute your success?

MEIR: It can be attributed to many decisions made over the course of 38 years, but most importantly the core values I mentioned earlier, that I adopted early on, and the people who make up this company, are at the heart of our success.

I have always insisted on total design and manufacturing control. It enabled us to have the flexibility in speaker design and the ability to react quickly when needed. Time and time again it has proven to be the best decision I have ever made.

Our focus on innovation and exploring new ways to design speakers has always driven us to new levels. This approach has put us at the forefront with the best speaker manufacturers and also differentiated us from the competition.

A lot of the credit for that has to go to the many people at Morel. Without everyone here we would not have achieved such a high degree of success today. I could not have done all of this by myself. I have been blessed to meet and work with some of the most amazing, hard-working individuals throughout my life. Our production workers, designers, engineers, salespeople, managers, and our global business partners abroad are passionately working to promote our company.

SHANNON: As Morel's founder and principal designer for many years, what has been your best experience?

MEIR: The joy of designing a product, manufacturing it, and following its acceptance in the marketplace is something very special to me. The thrill and excitement of seeing other people enjoy our creation and design is a feeling that revitalizes me with the energy and drive to work on the next project.

SHANNON: Your son, Oren Mordechai, is now responsible for Morel's unique design. How did that come about? Did you encourage his interest in audio?

MEIR: Oren grew up practically in the factory, because that is where I spent most of my time. From a very early age, he loved to be on the production floor and try everything. I would not say I encouraged his interest—he just loves it as much as I do. I am lucky to say that today he is doing excellent work developing new products, venturing into new materials and innovative designs that have received industry recognition and many prestigious awards worldwide. There is no bigger satisfaction for a father than to see his son following one's lead and continuing the tradition.

SHANNON: Are you working on or planning any new audio innovations?


In-house voice coil winding has been a fundamental element for Morel. Here, a Morel employee winds aluminum wire on an aluminum bobbin (former) to manufacture a tweeter voice coil. (Photo courtesy of Morel)

MEIR: We are always working on various projects. There is never a dull moment in our Research and Development department. Since Morel caters to the home audio, car audio, and raw drivers market segments, it seems we always have new products to release.

Our most recent innovation for the DIY/OEM segment is the Ti series, which employs titanium as a bobbin (former) for our oversized coils. The new Titanium series presents very special sound characteristics and parameters that enable speaker designers to achieve better resolution and dynamics in smaller enclosures. We are also developing new magnetic systems and chassis structures that will be implemented in future products to be released later this year.

A tray of soft dome tweeters works its way through the manufacturing line at Morel's factory in Israel. (Photo courtesy of Morel)

SHANNON: Your dream has been to create the perfect loudspeaker. What achievements have you made along the way and how close are you to fulfill your dream?

MEIR: The dream of creating the perfect loudspeaker can only be a dream. Over the years I have learned that in the loudspeaker field, which is partly "science" and partly "art" there is no such thing as "perfect." My ambition is to create a speaker that appeals to the largest audience possible, while reproducing music in its most natural and authentic form, free of any distortion.

Only those who have experienced speaker building can understand the obstacles and compromises that have to be made in order to achieve this objective. Even with 38 years of know-how and a fully capable manufacturing facility, we still struggle to overcome the physical boundaries and acoustic challenges to achieve the sound we want.

SHANNON: What do you see as some of the greatest audio innovations of your time?

MEIR: There were many, but the one that transformed the industry is the transition from analog to digital. I never liked the format of a music record, as it is bulky and highly vulnerable to being damaged. At the same time, this format produces high-quality sound. As I child, I always thought this format had to change. The shift to CDs to complement, you might say, the vinyl record suits me. But it seemed to me a radical change at the time.

Whether you like it or not, nobody can refute the fact that the digital age has made music more accessible to people. Because of the new digital formats, the interface that people use today can be very common, ranging from a high-end CD player, digital streamers, or smartphones, and so forth. That's remarkable.

SHANNON: Do you have any advice for *audioXpress* readers who want to build their own sound systems?

MEIR: You do not need to have a speaker company or an engineering degree to design and build good speakers. I began as a hobbyist like many in your audience probably. Reading, researching, and experimenting with different components and ideas were the only ways I was able to learn what it takes to build a good speaker. In today's digital culture the amount of information available is almost infinite; use it to your advantage.

Keep your designs as simple as possible. Some will make you believe that a complicated crossover is a must for a good speaker to perform well. I have always spent most of the time developing quality drive units. Using quality components will minimize the need for corrective measures when you build your crossovers and cabinet.

As with anything, have fun doing it. Put your soul and heart into it. I was lucky enough to be able to transform my passion for music and building speakers into a successful business. It may happen to you. *aX*


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Maggies Reclaimed

Rebuild a Magneplanar MG-IIA (Maggie) Loudspeaker

By Tom Yeago

I've learned a lot about stereo gear from books (kudos to Vance Dickason's opus, *Loudspeaker Design Cookbook*, and *The ARRL Handbook*, especially). But mostly I'm a disciple of Yogi Berra, who sagely noted that "You can observe a lot by looking."

I am an inveterate "let's look under the hood" kind of guy. I've dissected loudspeakers from Acoustic Research (formerly AR, now part of VOXX International), KLH Audio Systems, KEF, Bowers & Wilkins, and most of the rest of the alphabet soup that is or was the stereo industry. Berra was right. Much of what I've learned came from taking something apart and wondering why it was constructed this way or that way.

I digress. The Magneplanar MG-IIAs (i.e., the Maggies, serial No. 110751) dropped into my lap from out of the blue (see **Photo 1**). The Magneplanar brand name was the speaker design of Magnepan, a private, high-end audio loudspeaker manufacturer in White Bear Lake, MN. Jim Winey, Magnepan's engineer, conceived and implemented the company's loudspeaker technology in 1969.

I only recently received my Maggies from an acquaintance. The man was a Methodist minister whose wife had tired of schlepping them from parish to parish. The treble sections were broken, they were heavy and they took up space. I thought I might have some use for them, some day, so they became mine.

I had other irons in the fire, but eventually I got around to the Maggies and was pleased, on the whole, with what I found. The original packaging had been retained. Other than a slit in the grille cloth on the backside of one, they were in fine shape, cosmetically.

Photo 1: The reclaimed Magneplanar MG-IIAs (Maggies) contain 1970s standard-issue cosmetics. The Maggies' innards are something altogether different. Note their resemblance to diving boards.

I removed the woodwork and the cloth covers. I placed the panels on top of the sawhorses and, under the lights, the problems became obvious. Magnetic planar loudspeakers are essentially conventional magnetic drivers whose motor is unwound and spread across the surface of a film diaphragm in hopes of driving the surface evenly. Sometimes they're said to be "full-range." Sometimes they submit to reality and have separate bass and treble sections.

That's what Winey did with these Maggies, considered his down-market loudspeakers. A single sheet of tensioned polyester (I assume), about 10″ × 56″, is mounted on a 24″ × 70″ baffle. The magnets are arrayed on a perforated steel plate on the frontside (to let the sound out). The backside is open, no box (see **Photo 2** and **Figure 1**).

The woofer section is 8″ wide, with 23 loops of No. 24 copper wire glued down with what looks like contact cement. It covers approximately 450 in^2, uses about 120 g of copper, and has a DC resistance just shy of 6 Ω. Other than a few spots where the glue looked questionable, it was just fine.

That was not true of the treble sections. Each diaphragm's edge had three loops of wire about 45″ long, which took up approximately 1.5″ at the edge of the 10″ × 58″ panel. This wire was not copper, but aluminum. It had turned to oxide, a result of years at Virginia Beach seaside. It wasn't wire, it was dust. And Magnepan would fix it for me for approximately $600 to $800, return shipping included. (They also sell a $50 kit). They're really nice folks at Magnepan. The company has a toll free number (800-474-1646), and it still support all its 1970s-era gear. But my curiosity had been piqued, so I opted to fix it myself.

REPAIRING THE MAGGIES

I knew I would have to lay down some wire loops to establish a treble section, but what thickness wire, specifically? Aluminum was out of the question so I began reviewing my copper options.

A look at the crossover showed the treble section, like the bass, was a 6-Ω load. This indicated the stock wire (approximately 270″ total) was probably about 7 mils in diameter with a 6-Ω DC resistance. The stock wire weighed about 0.6 g. Associated masses include the polyester film and the estimated air load, which brought the estimated moving mass to 6.5 g.

After reviewing the problem, I settled on No. 36 AWG enameled copper wire, which is, in the spool I have, about 0.44 Ω per foot. Those of you

Photo 2: The Maggie's backside is shown without the obscuring grille. A stretched 10″ × 56″ polyester diaphragm on a 24″ × 70″ baffle is visible. The woofer section is 8″ × 56″ (note the heavy wire). The treble section is 1.5″ × 45″. The magnet structure is on the frontside, and not visible here. Maggies are inherently "single-ended."

doing the math will (rightly) calculate 270″ of this wire at almost 10 Ω.

To solve my problem, I ran twin runs of No. 36 AWG wire, at 0.22 Ω per foot, and I glued down four loops (not three) by doubling up on the center loop. That totals slightly more than 360″ and about 6.7 Ω. Perfect.

Well, almost perfect. Copper wire is heavier than aluminum, and the difference in mass is about 1.5 g (i.e., 2 g vs. 0.6 g). Adding in the other moving bits, the new tweeters come to 8 g vs. 6.5 g. That translates to a 2-dB deficit in output, all things being equal.

But they're not equal. Now there are four loops instead of three, which increases the force by a third, giving the copper-wired unit more output and bringing the rewired unit within spitting distance of the stock speakers. And by spitting distance, I mean within a fraction of a decibel, which I think is insignificant given normal manufacturing variance.

So, I had a plan. But, I needed to decide how to execute it.

LAYING DOWN TRACKS

Getting the wire was no problem. I have spools of thin wire from small transformers, stepper motors, and so forth. Much of it is conveniently color-dyed. So I had the wire. I simply had to determine how to lay it down.

First, I gave the polyester film a good scrubbing. I used cotton swabs to wash it with alcohol, then acetone. When the oxide and the glue were mostly gone, I used the remaining faint trace as a guide for the new wire. This is significant, because the wire should be positioned exactly between the magnet faces on the other side of the diaphragm for maximum efficiency (see **Figure 1**).

Then, I laid down the loops, 45″ per loop, in sequence. Using a lot of light and a pair of inexpensive reading glasses for magnification, I glued down the copper, about 6″ at a time.

I put the wire in place and held it with a little piece of painters' tape every 0.5″ or so, keeping the twin wires parallel and flat. I used cyanoacrylic (i.e., "super" glue) because it's cheap, cures quickly, and is thin enough to spread between wire and film with capillary action. However, I didn't apply it directly from the tube. I put a drop on a piece of masking tape that I had placed off to the side and used a sharpened bamboo chopstick to transfer the glue to the work piece. After the glue set, I lifted off the small pieces of painters' tape one piece at a time and glued each section.

To recap: First, I laid down the wire and held it in place with tiny bits of tape. Then I glued the sections between the tape, carefully applying small amounts of adhesive to prevent it from flowing under the tape. After it hardened, I removed the tape, applied glue where the tape had been, and moved on to the next 6″ section.

Tedious? Sure. Time-consuming? You betcha. Maddening? Yes, if you're not careful. But it worked.

TECHNICAL BITS

I want to review the rewired unit's output vs. the stock speaker's output. The moving mass is 8 g (my modified version) vs. 6.5 g (stock version). The DC resistance is 6.8 Ω (modified) vs. 6-Ω assumed (stock). The force factor (Bl) is four loops (modified) vs. three loops (stock). If you do the multiplication, you find the rewired unit's sound pressure will be about 95% of the stock unit's sound pressure, which is within half a decibel. I can live with that. Other details of the modification are worth discussing as theoretical, but they are probably not significant.

Using twin runs of thinner wire may provide a tiny improvement, because it leads to a field around the two wires that is flatter and squashed down, as opposed to the magnetic field you'd find around a single round wire. I may speculate that this fits into the magnetic field from the magnets slightly better, yielding marginally lower distortion. The twin wires also have more surface area and will glue down more securely than a single round wire.

Then there's that fourth loop, doubled up over the middle of the three stock loops. This could conceivably lead to a slight deformation of the treble section (i.e., a bit of bulge in the center) that could improve lateral

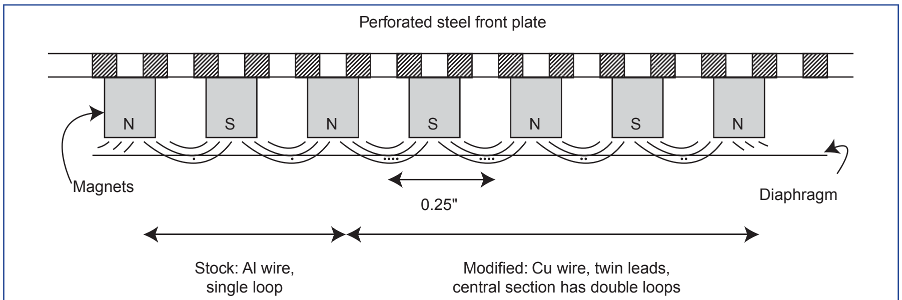

Figure 1: This is a section through the treble regions of the revived vs. stock Maggies. Note the non-symmetric magnetic circuit. A second magnetic structure on the backside would be nice but, under the circumstances, is hugely impractical. Building a structure to keep two magnetic arrays in proximity would be extremely challenging, not to mention heavy and expensive. If you've ever taken apart one of Infinity's EMIT units, you understand.

dispersion. Maybe. It could happen.

Other measurements relate to the Maggies' original design. It is nothing I can take credit for. The crossover (XO) is textbook Butterworth third-order, so its Q is approximately 0.71 and is down 3 dB at resonance. The impedance curve suggests that the low-pass and high-pass sections are slightly overlapped (see **Figure 2**, **Figure 3**, and **Photo 3**).

Because the load is a virtually perfect, nonreactive 6-Ω resistance, you can look at the phase response of the sections and see how they will be at a 90° difference throughout their range. The two sections' output will add power as opposed to simple pressure. It makes perfect sense.

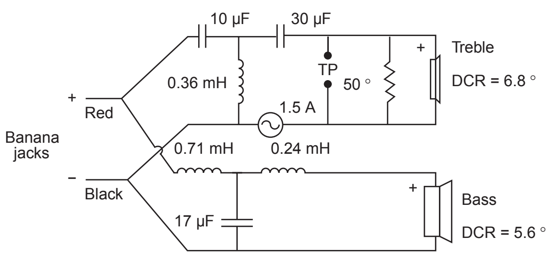

Figure 2: The modified Maggie's schematic shows everything is stock except the 50-Ω shunt across the tweeter section. It brings the rebuilt Maggies' 6.8 Ω back down to the designed 6 Ω.

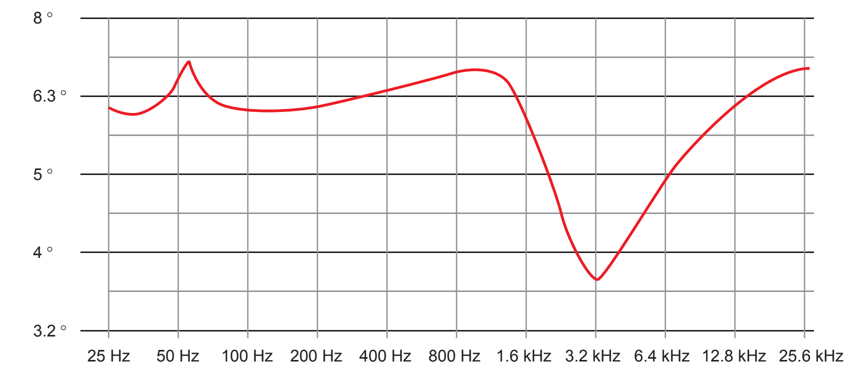

Figure 3: Impedance of the modified Maggies reveals that other than the 0.5-Ω blip at 52 Hz indicating panel resonance and a dip to 3.8 Ω at 3.2 kHz, we're looking at as nonreactive a loudspeaker as we're likely to find. I ran a curve for the tweeter, not shown, and they're a solid 6.1 Ω, rising to 6.4 Ω at 25 kHz.

Photo 3: A close-up view of the crossover is shown. These units were built to a high standard. Note all the film caps and the air core. Other than the aluminum wire in the treble section, not much can go wrong.

A curious thing about these units is that the bass and treble sections share a single, contiguous membrane. They cannot act independently. So while the difference in the crossover region will be 90°, that difference will be resolved on the diaphragm itself, not in the atmosphere in front of the loudspeaker.

Occasionally, I read of listeners who note they have to sit a good distance from a loudspeaker before it sounds natural (i.e., before the various drivers meld or cohere). There may be something to this, if the Maggies' reputation for a "seamless" sound quality arises from this physical oddity. Or maybe not, because subsequent magneplanars, also described as seamless, have sections that don't share a common membrane.

The loudspeaker/amplifier interfaces are wonderful, nonreactive 6-Ω loads. A pair per side wired in series would be a terrific load for a tube amplifier on the 8-Ω tap. I didn't find them noticeably power-hungry. I used an unremarkable but decent 50-W solid-state amplifier and it seldom registered

SPEAKER SPECIAL

beyond the 5-W range in a 20′ × 33′ lively room.

They are dipoles, so you can also perform the parlor trick of walking around the side into the cancellation null. Amaze your friends! But the bass needs rolloff because the wavelengths become large compared to the baffle width. Plus the panel resonance appears to be approximately 52 Hz.

I found you can stretch things a bit by siting the panels about 5.5′ out from the front wall. This means at 50 Hz, the panel's out-of-phase back wave bounces back to emerge in-phase with the wave off the panel's front. It's a gimmick, but it's better than nothing.

Readers will note the 1.5-A fuse for the treble section. Owing to copper's higher mass, higher melting temperature, and additional surface area of the twin wire runs, I figure the modified treble section can safely handle about twice the power of the stock model. So keeping the 1.5-A fuse is more than prudent.

The crossover from the 8″ wide bass section to the treble is 2 kHz, where the wavelength is about 6.8″ and very sensible. The design makes sense dispersion-wise and power-wise.

COSMETICS

The oak framing is stained dark enough to resemble walnut. The open-weave grille cloth is beige or off-white, with more texture than the linen-look that was in vogue. I like its understated look.

I gave the oak a few light coats of rub-on polyurethane to spruce up the finish. I also ensured the holes for the woodscrews were still snug. The bases were painted flat black and a bit damaged, so I spackled and streaked several different paint colors for a fake-granite finish that's fine if you don't look too closely.

I also gave the normally open backs a mote of protection by installing a layer of window screen, held by big "splines" of polystyrene form blocks. It was not perfect, but it provided some protection if they should someday tip over backward.

The project's most frustrating aspect was the narrow oak trim strips down the front of the speakers. They're stapled into hard plastic pads glued to the front of the baffle. The only remaining job was to tediously work the staple prongs through the grille cloth back into their original holes. What a pain.

IN USE

Modern Maggies are low-impedance, current-sucking monoliths. At 6 Ω, these units are quite tractable. I like them a lot, but I miss the low-end frequencies of a good, closed-box unit. I should mention Quad ESLs are my all-time favorites.

I loaned the Maggies to a friend who had no experience with planar loudspeakers. As he's had little experience with anything other than vented boxes, the Maggies were something entirely different. In fact, he is unfamiliar with turntables, not to mention closed-box designs. He really liked the Maggies, but after six months or so it was time for him to return them. I suspect one day soon the Maggies will find their niche as sound reinforcement (!) in a small venue nearby. They can handle the power and the side nulls of a dipole are a natural fit for this job. I'll keep you posted. *aX*


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SPEAKER SPECIAL

The Single-Tower Stereo

One enclosure with two channels

By Ken Bird

I received a number of comments from potential builders following my *audioXpress* article about the twin Castles ("Castle Tower Omni Speakers," September 2011). Most commenters said they found the project interesting but daunting, due to the complicated woodwork. Some DIYers were concerned about the cost of 26 drivers. However, those who heard the system were impressed with its open sound quality. I gave some thought to their comments and decided that a simpler approach was needed for audiophiles who want an omni stereo.

STEREO HISTORY LESSON

When stereos burst forth in the late 1950s, commercial mass market manufacturers struggled to incorporate two sound channels in a package that was inexpensive to make, yet retained the stereo sound people wanted. Most of the designs involved coffin-sized console cabinets with enclosed speakers on each end, separated by housing for the electronics and record changer. Audio quality varied by manufacturer, with most of the low-end models containing only single-ended tube amplifiers and $1.50 speakers. (I was an engineer with a large manufacturer at the time and our 12″ woofers cost us $1.50 in 1,000 or more quantities.)

No matter the system's initial cost, the stereo effect in most of them was poor. Some people complained they heard only one channel or the other. And even if you did get the stereo in a "sweet spot," there seemed to be a "hole" in the center. Complaints of this "hole" were also heard from users of separate stereo speakers. The speakers were usually large boxes poorly placed in the living rooms.

Imaging was a word not used in early-day stereo. Tower designs with modern drivers were still a few decades away. To address the stereo "hole" complaints, console manufacturers (e.g., Magnavox, Admiral, Zenith) added a "third-channel" speaker, typically a 8″ or 6″ × 9″ oval speaker placed in the cabinet's center. The component manufacturers (e.g., Fisher, H. H. Scott, and Dynaco) also added third-channel speaker terminals to their high-end amplifiers.

Photo 1: The completed single-tower stereo is shown here.

Bell Laboratories first used the center-stereo channel in its stereo experiments in 1934. The late Paul Klipsch authored several papers on deriving third-channel signals. Many other professional audiophiles published articles on pre- and post- third-channel solutions. By the end of the 1960s, the audio "hole" issue was largely forgotten as speaker technology and consumer education on speaker placement improved.

Thoughts on devising a simpler omni speaker brought me back to those early days when both channels were incorporated in the same console enclosure with signals from both channels sent to a third channel.

Why not use modern drivers to do the same in a single-tower design? The problem is how to derive the third channel without damaging the amplifier. In the early days, third-channel solutions were designed for tube amplifiers and used multiple taps on the output transformer to attain a matrixed left and right signal.

MODERN SOLUTION

Elias Pekonen, a participant in the web-based DIY Audio website, proposed a similar method of deriving a third channel. His paper is available on the DIY Audio website (www.diyaudio.com).

I like Pekonen's approach and his detailed analysis of incorporating the left and right signals within a third channel. His term for the method is "vector steering," which uses a RC network in parallel with the center-channel speaker to form a psychoacoustic filter. The filter's action "steers" the high frequencies from the third channel and changes apparent sound distribution by reducing the treble response. I decided to build a mock-up of his design to evaluate the system (see **Photo 1**).

MOCK-UP

To build a mock-up, I used poly stuffed coffee cans for enclosures, mated

Photo 2: This mock up shows the three-channel system.

with a 10″ DVC woofer that crosses at 300 Hz (see **Photo 2**). Pekonen recommends a 4.7-µF capacitor in series with a 5-Ω resistor. Increasing the capacitor values reduces the center channel's high-frequency response and can change the system's soundstage. I did not see any change in the sound using the 4.7-µF capacitor, but there was a change in the sound with a 10-µF capacitor (see the optional circuit in **Figure 1**). I suggest builders experiment with the filter based on the system's location in a room. I found the filter is a function of both component values and system location. In my final design I left the filter out and just changed the system's position until I found the ideal location for maximum stereo performance.

The single-tower stereo provides a full-stereo sound system in approximately 1 ft^2 of space, which most rooms can accommodate. The ideal placement for the system is where wall reflections would be less than 4′ to 6′ from the side speakers. Yet, it was also found that corner placement was equally good, if not better, for a wide stereo effect along with bass enhancement.

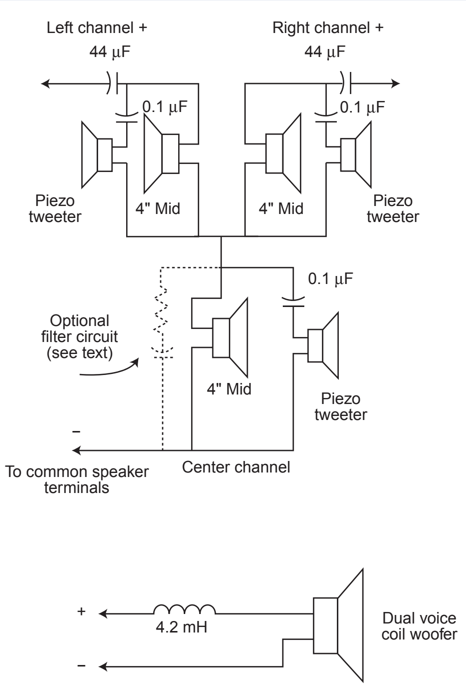

Figure 1: The loudspeaker schematic with the optional filter circuit shows only one woofer coil circuit. You will need to make one for each input.

DRIVERS

Driver choice is not critical for this system. They should all have the same specification and be 4″ to 6″ in size with a 200-Hz to 12-kHz or higher midrange response. Likely candidates are types with whizzers cones or even some automobile types with coaxial tweeters. The total impedance per channel is about 12 Ω when using 8-Ω speakers. If you use 4-Ω speakers, the impedance will be close to 6 Ω. If you use the steering filter, it's best to use only 8-Ω drivers. Since the system stereo effect is mostly dependent on reflected sound, efficient midrange and treble is important in the driver.

I used the drivers I had in my inventory, a 4″ Pioneer with a 100-to-10,000-Hz wide, flat range and a 6-to-20-kHz piezo tweeter in parallel. A Goldwood 10″ DVC woofer in a sealed configuration handles the bass. The two-way crossover is a 6-dB first-order at 300 Hz (see **Photo 3**). The Pioneer speaker is no longer available, but some substitutes are listed in the Single-Stereo Tower Parts List at the end of this article.

CONSTRUCTION

Construction is easy if you use precut MDF shelf panels. You will also need a jig saw, a handsaw, or a skill saw, and a router to build the tower (see **Figure 2**). I used five shelf panels for the entire project. They were only $2.99 each, making this a low-budget project. The enclosure is divided into two chambers, the top is 1 ft^3 and the woofer chamber is 1.8 ft^3. You will need a piece of scrap 2″ × 2″ lumber for the woofer chamber's internal braces.

Start by cutting out the speaker holes. The sizes will vary depending on the speakers you choose. I made a 3.6″ hole for the midrange with a 6.25′ wide inset that was 0.25″ deep. I also made a 2.5″ hole for the tweeters. The woofer required a 9″ hole with a 10.25″ wide inset. Then, I made a 3″ hole in the rear panel for the terminal cup. I opted to router an ogee pattern to the front and back edges of the side panels (see **Photo 4**).

Next, I cut out the blank panel. For the top, cut an 11.25″ × 12.75″ piece. The bottom piece should be 11.25″ × 13.25″ and the internal partition should be 11.25″ × 9.75″. I used a router and a Jasper Model 240 circle jig to cut all the insets and the holes.

I butt nailed the panels with 2″ finishing nails and used a liberal application of Gorilla Glue to secure them. I used corner clamps to hold the sides together and I used larger clamps to ensure the panel centers were even. I used a 1/16 bit to drill pilot holes to a depth of 5/16″ for the nails (see **Photo 4**). I used a nail set to drive the heads 1/8″ deep and I filled each nail hole with Weldwood wood putty. I added the partition before

Photo 3: This system uses two drivers and a crossover.

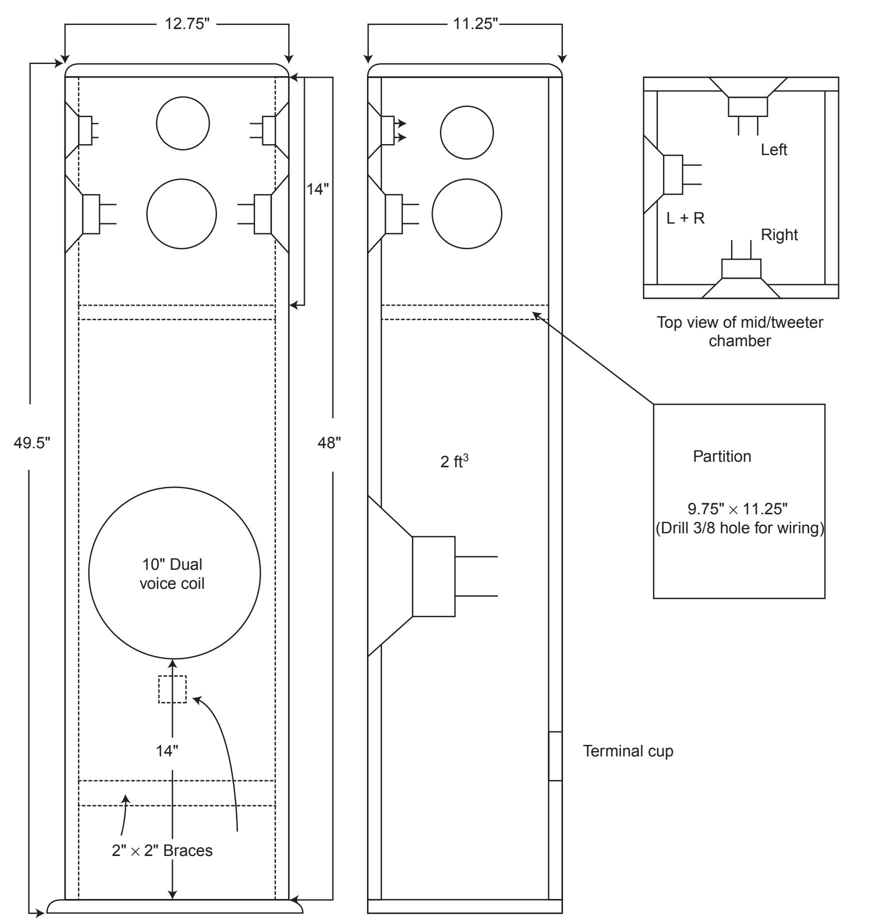

Figure 2: The single-tower stereo design includes the dimensions for the speaker cabinet and placement of the 10″ dual voice coil.

I nailed the back panel in place. Be sure to drill a 3/8″ hole in the partition for the wiring. Before adding the top and bottom panels, caulk all the joints in the upper and lower enclosure chambers. Once nailed in place, caulk the bottom panel joints through the woofer hole.

Instead of gluing the top, I placed a liberal bead of white window caulking on the enclosure's top edges and nailed the top, spacing the nails every 2″. Use a rag to wipe off the excess caulk. This method forms a good seal and a strong joint. I added 2″ × 2″ cross braces to the woofer chamber. Locate the longer 11.25′ brace 8″ below the woofer and the shorted 9.75″ brace 3″ above the woofer. I used 1 5/8″ drywall screws countersunk 0.25″, along with glue to secure the braces.

Once the enclosure is fully assembled, use Weldwood or another filler to seal all the nail and screw holes and let the glue and caulk set overnight before proceeding with sanding and applying a finish (see **Photo 5**).

FINISHING

The enclosure can be painted or veneered. I painted my version after I sanded the panels to remove the excess hole filler and smoothed all the surfaces. The MDF needs to be properly primed before you apply paint. I applied two coats of KILZ primer. I used 120-grit sandpaper to sand each coat, and applied the final paint coat.

The color choice was based on the inventory of my paint locker. The black trim matched the speaker grilles. My "aesthetics critic" said it resembles a

Photo 4: The shelf panels are ready for assembly. The blank panel (shown at left) will be used for the top, bottom, and internal partition.

Photo 5: Simple butt joints were used in the assembly, along with glue and 2″ finishing nails.

Photo 6: The enclosure is ready for finishing.

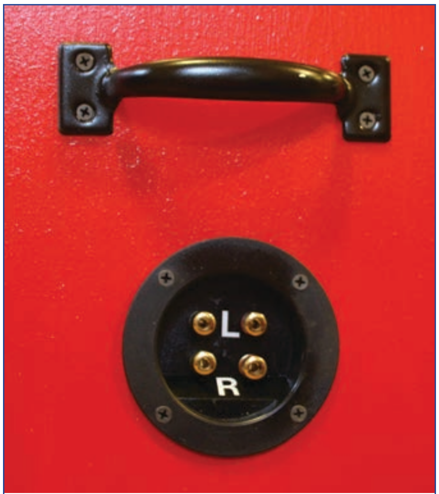

Photo 7: I added banana jacks to a single-speaker cup to accommodate both channels. The optional handle makes moving the tower easier.

British telephone kiosk or a Chinese magician's magic box (see **Photo 6**).

CROSSOVER

I made the crossover from a modified commercial unit and I have listed the components necessary to duplicate it for the system. It is a first-order, 6-dB type crossing at 300 Hz.

INSTALLING THE SPEAKERS

I used 18-gauge wire to install the upper chamber speakers. You only need to run three wires up to the top chamber. I used red wire for the right channel, green wire for the left channel, and black wire for the common ground. When wiring, follow the schematic shown in **Figure 1**. Each piezo tweeter required a 0.1-μF capacitor wired in series to match the midrange units 86-dB SPL. I conventionally wired the dual voice coil woofer for a two-channel system. I modified a 3" terminal cup by adding another set of terminals. You could also order the one suggested in the parts list. One optional item I always add to my tower speakers is a carry handle mounted above the terminal cup on the back. With one hand on the handle and the other on the top front, it is easy to tilt and lift the enclosure (see **Photo 7**).

I placed 10 oz of polyfill stuffing in the upper chamber and I lined the lower chamber with polyfill blanket material to a 1" depth. I used a gasket bead of clear silicone caulking and 1" drywall screws to mount the midrange and tweeter speakers. The woofer gasket was 0.25" foam peel-and-stick insulation. This enables easy removal of the woofer to service the crossover. I made the inset grilles from fiberglass pet screen stapled to circular frames made from Masonite (see **Photo 8**).

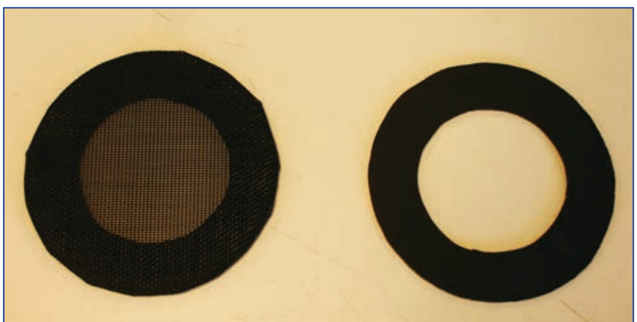

Photo 8: The grilles fashioned from pet screen on a 1/8" Masonite wood frame shown on the right.

PLACEMENT AND SOUND QUALITY

The speaker works well when placed 2' to 4' from the back wall and centered 4' to 6' from the side walls. It is an ideal speaker for a bedroom, a den, or a dorm room. For larger rooms, you can improve the speaker quality by placing it in a corner where the stereo spread will be better reflected and the bass enhanced. Another option is to turn the enclosure backward, facing the wall at a 4' to 5' distance. If you like an airy omnidirectional sound, placement is not critical. No matter how you position it, you will find the single-tower stereo a unique audio experience. *aX*

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___"Simplified Phantom Third Channel," *Audio Craft Magazine*, Audio Amateur, January 1958.

E. Pekonen, "Stereophonic Sound from a Single Loudspeaker," www.diyaudio.com.

H. Tremain, "Method for Deriving a Third Channel," *The Audio Cyclopedia*, Howard W. Sams, 1969.

Single-Stereo Tower Parts List

Crossover Components
- Two each: 4.2-mH inductors (www.parts-express.com, 266-580)
- Two each: 44-μF non-polarized capacitors (www.parts-express.com, 027-358)

Enclosure
- Five each: 75" × 11.25" × 48" MDF shelf boards
- 3' length of 2" × 2" wood for cross braces
- Wood glue
- 2" finishing nails
- 1" drywall screws
- Silicone caulk
- Polyfill material

Speakers
- Three full-range 4" or 5" 4- or 8-Ω speakers

Speaker Options
- Visaton Full Range Part #55-4610 80 to 20 kHz with whizzer—no tweeter needed (www.mcmelectronics.com)
- Paper cone woofer (www.parts-express.com, 292-404), use with piezo tweeter (www.parts-express.com, 270-011)
- One 10" dual voice coil woofer (www.parts-express.com, Goldwood 290-362)

Speaker Grilles
- 1 yd of fiberglass pet screen
- 1 2 × 4 piece of Masonite
- 1 box of 0.25" staples

Speaker Terminals
- Dual terminal (www.parts-express.com, 260-304)

The Horizon Subwoofer

A subwoofer designed to complement the Horizon loudspeakers

By George Ntanavaras

The Horizon loudspeakers are open-baffle speakers that can be used as full-spectrum loudspeakers for music reproduction (see my article "The Horizon Loudspeaker," *audioXpress*, September 2012). But in a home cinema where effects with low frequencies are common, it is better to use a subwoofer for the low-frequency reproduction. The benefit of the Horizon loudspeaker's open-baffle subwoofers is that they don't use the large cone movements necessary to reproduce the low frequencies, which dramatically reduces their distortion.

In this article, I will describe the procedure I followed to design my Horizon subwoofer and how I used DSP to adjust it for a smooth in-room frequency response.

I will also present some interesting measurements that I received using an accelerometer with its accompanying preamplifier/processor while evaluating the subwoofer's performance. These measurements concern the subwoofer's frequency response, the vibration behavior of the subwoofer box, and an estimation of the voice coil's displacement of the subwoofer cone in relation to the voltage drive and the total harmonic distortion (THD) of the reproduction.

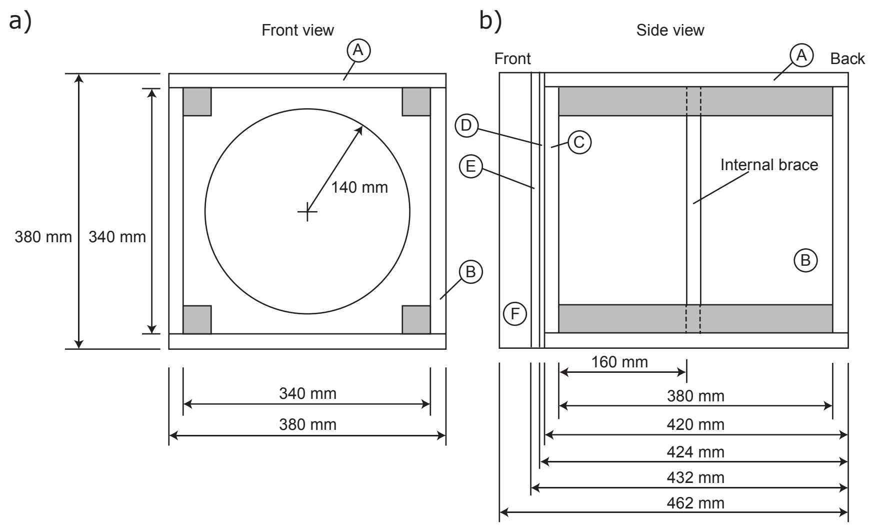

Figure 1: The Horizon subwoofer box is shown in a front view (a) and a side view (b). The design uses 20-mm plywood for the sections labeled A–F. A = 380 mm × 420 mm; B = 340 × 420 mm; C = 340 mm × 340 mm; D, E, and F each = 380 mm × 380 mm. The corner braces measure 40 × 40 mm × 380 mm.

DRIVER SELECTION AND BOX CONSTRUCTION

The subwoofer design is based on the Peerless 830500 subwoofer driver. This is a unit designed for subwoofer applications. Peerless has an excellent reputation for its subwoofer drivers and its 830500 is no exception.

It has an "extra-long stroke" capability for long voice coil excursion and an effective 466-cm² piston area. According to the manufacturer, the driver also has low distortion. The unit's construction is impressive.

The manufacturer lists the driver's Thiele-Small (T-S) parameters as follows:

$Q_{ES} = 0.21$
$Q_{MS} = 3.7$
$Q_{TS} = 0.2$
$F_S = 18.1$ Hz
$RDC = 3.5\ \Omega$
$Sd = 466\ cm^2$
$V_{AS} = 139$ L

I choose a closed box for the subwoofer although it is not very common for subwoofers because it has higher resonance frequency compared to a bass-reflex box. But it has some benefits including the transient response, which is much better than any other box, and the sound pressure, which decreases by only 12 dB/octaves. This can be corrected more easily with electronic equalization.

I choose a Vb = 35-L volume for the subwoofer box. This is a good compromise between the box size and the needed power amplifier drive voltage.

Applying the following formulas from the closed-box theory:

$a = V_{AS}/Vb = 139/35 = 3.97$
$Q_{TB} = Q_{TS} \times SQRT(a + 1) = 0.45$
$F_{SB} = (Q_{TB} \times fs)/Q_{TS} = 40.4$ Hz

Where Q_{TB} is the closed box quality factor and F_{SB} is the resonance frequency.

Figure 1 shows the subwoofer box design. The box's internal dimensions are 340 mm × 340 mm × 380 mm while its external dimensions are 380 mm × 380 mm × 462 mm. These dimensions give an approximate 66-L external volume and a 44-L internal volume. If the

Photo 1: The subwoofer box's interior is shown in the early phases of contruction.

volume occupied by the driver itself and the internal braces is subtracted, the net volume is about 35 L.

I used 20-mm plywood with corner braces to construct the subwoofer box (see **Photo 1**). In the middle of the box, I glued an internal brace with large openings. I did this to increase the box's rigidity and minimize the panel vibration. I did not use any internal stuffing in the subwoofer box since it is unnecessary when using electronic equalization.

Photo 2 shows the completed subwoofer with the front baffle after I painted it with an imitation-wood varnish.

I decided to construct two subwoofers for my listening room. This will provide an extra 6-dB output, lower distortion, and more importantly, a smoother in-room frequency response due to the excitation of the room from two different positions.

Photo 2: I designed the Horizon subwoofer to work with my Horizon open-baffle loudspeakers.

SUBWOOFER'S PARAMETER MEASUREMENTS

After the box's construction, I used the ARTA software (www.artalabs.hr) to measure the box's T-S parameters. The program's demonstration mode is fully functional, however, it does not enable you to load or save files.

The software package consists of three separate parts. One of them, the Limp, measures the impedance of a loudspeaker driver or cabinet then automatically extracts the T-S parameters.

The Horizon subwoofer's impedance is shown in **Figure 2**. The box's resonance occurs at 40.5 Hz with a 57.6-Ω maximum impedance. The minimum impedance is 4.8 Ω at 120 Hz, but fortunately the impedance phase at this frequency is close to zero.

A small "glitch" in the impedance curve occurs at 400 Hz. This is caused by a standing wave corresponding to the subwoofer box's large dimension, which is equal to 0.38 m. The standing wave is created when $\lambda/2 = 0.38$ m so $f = c/\lambda = 342/0.76 = 450$ Hz. It is not a concern since it occurs well outside the subwoofer's operation range. Another small "glitch" is also shown at about 550 Hz. This is probably due to the loudspeaker basket.

After the impedance measurement, the program extracted the first subwoofer's T-S parameters:

F_{SB} = 40.5 Hz
R_E = 3.73 Ω
Z_{MAX} = 57.6
Q_{TB} = 0.48
Q_{ES} = 0.51
Q_{MS} = 7.42

The second subwoofer's T-S parameters were:

F_{SB} = 39.7 Hz
R_E = 3.73 Ω
Z_{MAX} = 58.2
Q_{TB} = 0.47
Q_{ES} = 0.50
Q_{MS} = 7.35

The F_{SB} and the Q_{TB} of both subwoofer boxes are close to their theoretical values.

ADJUSTMENT OF THE SUBWOOFER RESPONSE

To adjust the subwoofer's frequency response, I followed a step-by-step procedure. I will describe this procedure in the next paragraphs and it consists of the following steps:

- Equalize the subwoofer's near-field frequency response
- Optimize the subwoofer's location in the listening room
- Provide additional electronic

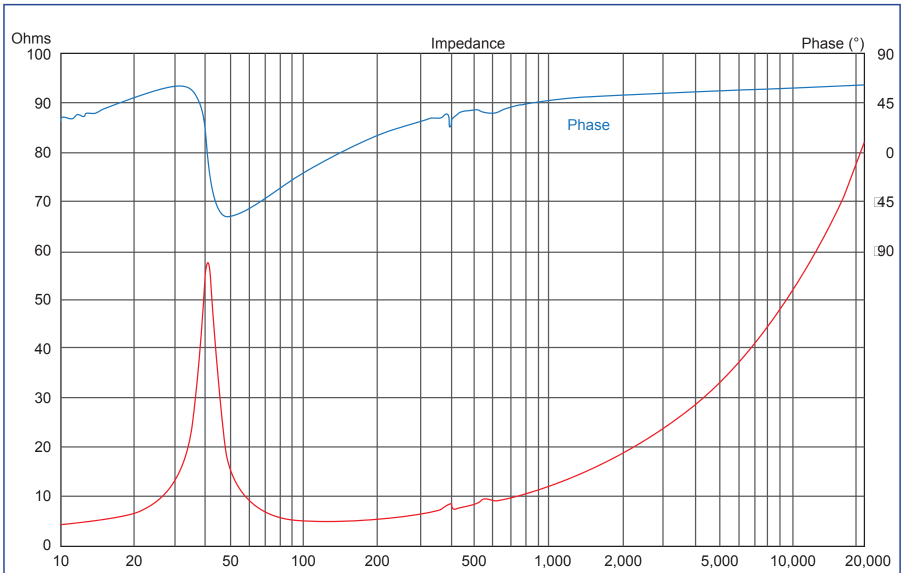

Figure 2: The subwoofer has a 57-Ω maximum impedance.

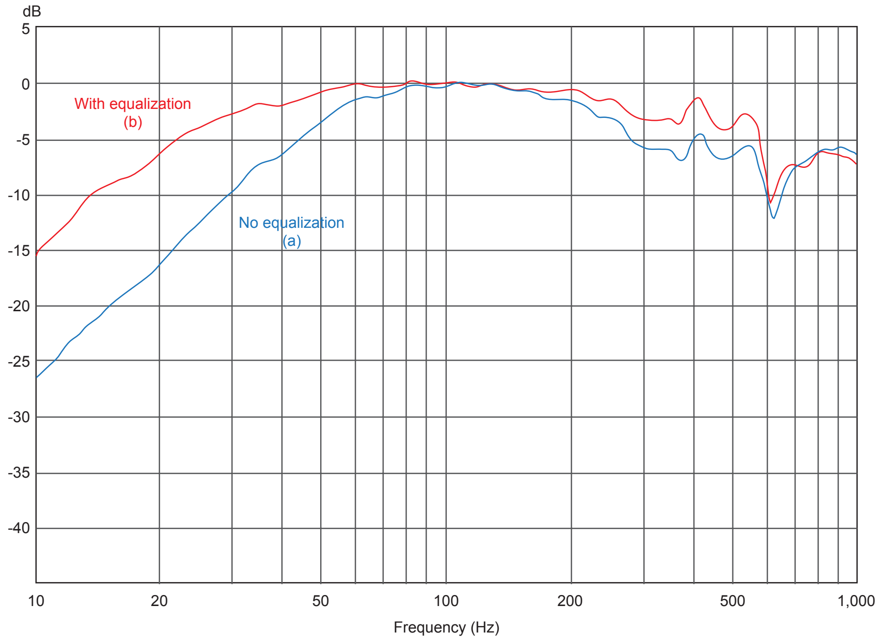

Figure 3: The Horizon subwoofer's near-field response is shown without equalization (a) and with equalization (b).

equalization to the subwoofers for flat response at the listening position

- Perform final subwoofer adjustments for integration with the main loudspeakers

NEAR-FIELD FREQUENCY RESPONSE

The subwoofer's near-field frequency response is shown in **Figure 3a**. It has −3-db frequency at 47 Hz. It drops to −9.5 db at 30 Hz and −16 dB at 20 Hz. This frequency response is not acceptable for a subwoofer and should be corrected with equalization.

The equalization could be applied using different methods including: an analog circuit (e.g., a Linkwitz transform circuit) or digitally (e.g., a DSP equalizer). Carefully apply your chosen

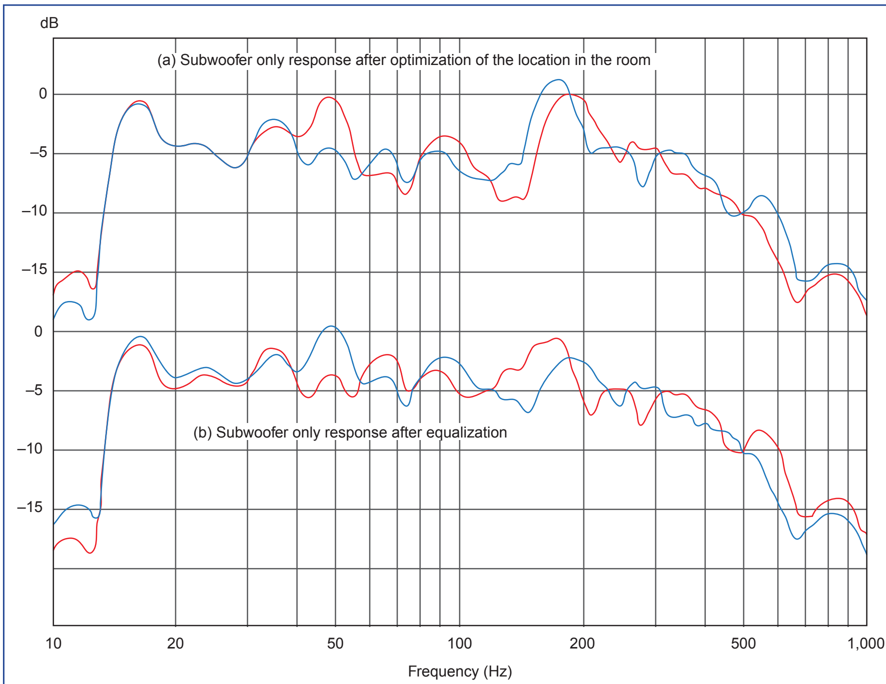

Figure 4: The subwoofer's responses correspond to the two different microphone positions—after finding the optimized location (a) and after equalization (b).

Photo 3: I used the ACH-01-03 accelerometer system to measure the subwoofer.

method to prevent the equalizer from voltage clipping at its output due to excess gain.

I also wanted to equalize the in-room subwoofer response so I used a Behringer DCX2496 loudspeaker management system for the equalization. This is a versatile device with three analog inputs and six analog outputs. It uses 24-bit/96-kHz ADCs and DACs.

I used one of the three analog inputs as the subwoofer input and one of the outputs as the subwoofer output. The DCX2496 includes several different types of filters for crossovers and several filters for equalization for each input and output, including high-pass shelving filters, low-pass shelving filters, band-pass filters with variable Q, delays, phase change circuits, and so forth.

Knowing the room gain will boost the low frequencies, I selected a target response that has a −3-dB point at about 30 Hz and a −6 dB at 20 Hz.

To begin, I adjusted the Behringer filters' parameters then measured the near-field subwoofer response. I did this several times until I achieved the desired result. Although it sounds complicated, this is a simple procedure to perform and will not take much time.

I used the following equalization parameters to receive the response shown in **Figure 3b**:

- LP-shelving filter at 32 Hz with a 9-dB gain to boost the low frequencies
- HP-shelving filter at 323 Hz with a 7-dB gain to flatten the response at the higher frequencies
- 6-dB/octave low-pass filter at 254 Hz to gently cut off the higher frequencies that are out of the subwoofer's range

LOCATION

Next, I optimized the location of the two subwoofers in my listening room. By this, I mean that there are some locations in every listening room where a subwoofer can be placed so the resulting low-frequency response is much smoother. This optimization is important and useful since the most flat in-room frequency response is naturally succeeded without any additional electronic equalization applied.

It is a difficult and time-consuming procedure. First for practical reasons, it is not easy to move two big heavy subwoofers around the room. Second and most important, it is difficult

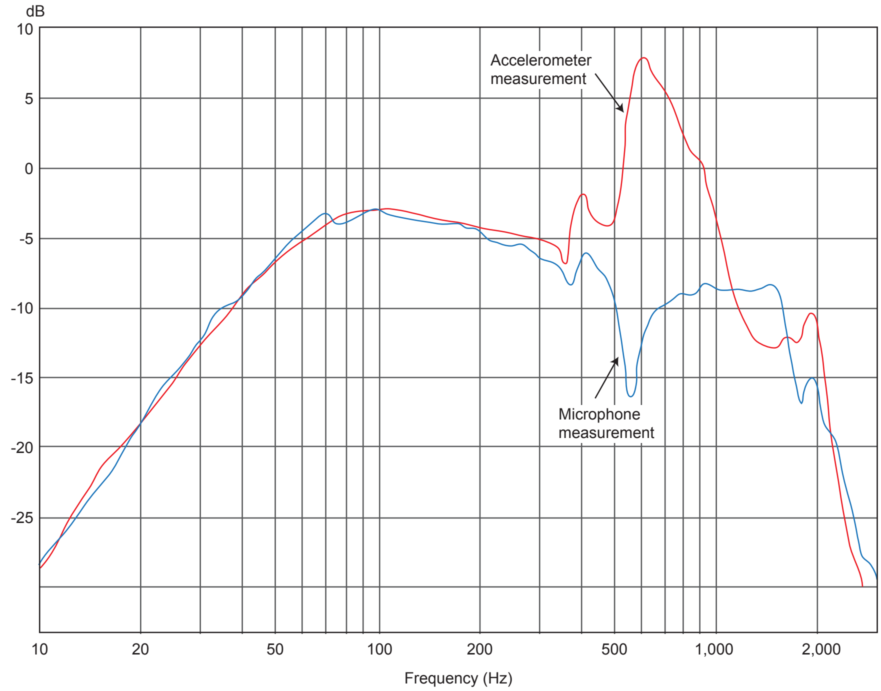

Figure 5: I used an accelerometer and a microphone to compare measurements.

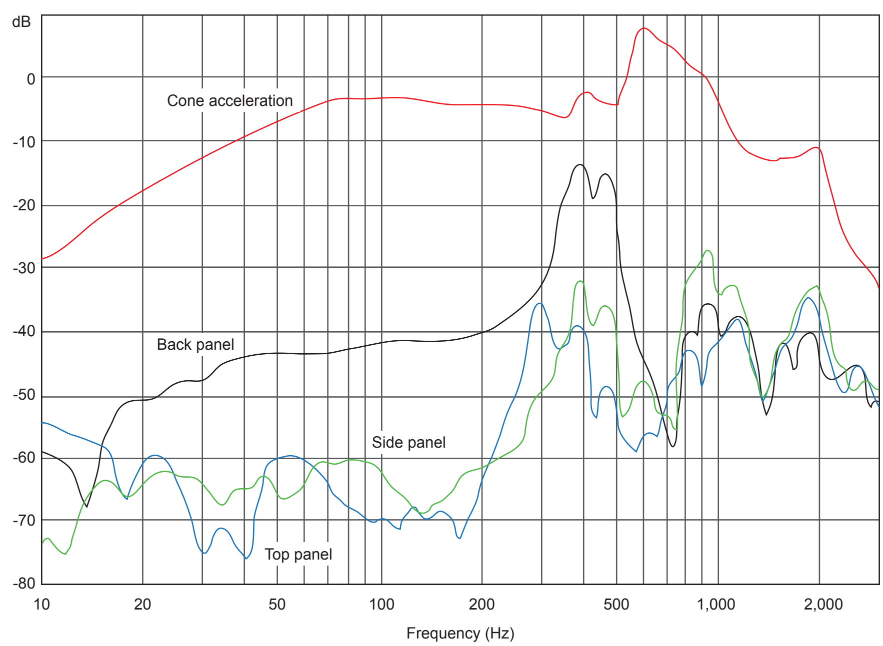

Figure 6: I used an accelerometer to take cabinet vibration measurements.

to interpret the measurement results. Usually the in-room frequency response measurements at low frequencies have a 10- or even 15-dB variation. Also, large measurement variations could occur by moving the microphone just a few centimeters.

For this reason, I decided to optimize the response in only a small area of my listening position. My listening room's dimensions are approximately 4-to-5-m width × 10-m length. After several attempts, I placed the subwoofers near the walls but not exactly at the corners. I also placed them to the rear of the listening area, but since the subwoofer will only reproduce low frequencies this is not a problem. I later confirmed this by listening.

The distance between the subwoofers and the listening position is similar to the distance between the main loudspeakers and the listening position.

This setup's response is shown in **Figure 4a**. The two curves correspond to the two microphone positions around the main listening position. The response errors are within ±5 dB from 16 to 200 Hz, which I think is a good result for in-room response without any electronic correction.

EQUALIZATION AT THE LISTENING POSITION

After I determined the best location for the subwoofers in my listening room, I carefully applied equalization, using the Behringer filters to further smooth the response. I placed the microphone in two positions very close to the main listening position. The final result is shown in **Figure 4b**. Now the response errors are much lower. To achieve this response I used the following filters:

- band-pass filter at 184 Hz with a –3.5-dB gain and Q = 3.5
- band-pass filter at 69 Hz with 3-dB gain and Q = 3.5
- band-pass filter at 135 Hz with 4-dB gain and Q = 4.5
- band-pass filter at 28 Hz with 2-dB gain and Q = 5

Now the subwoofers are ready for the final step, integration with the main loudspeakers.

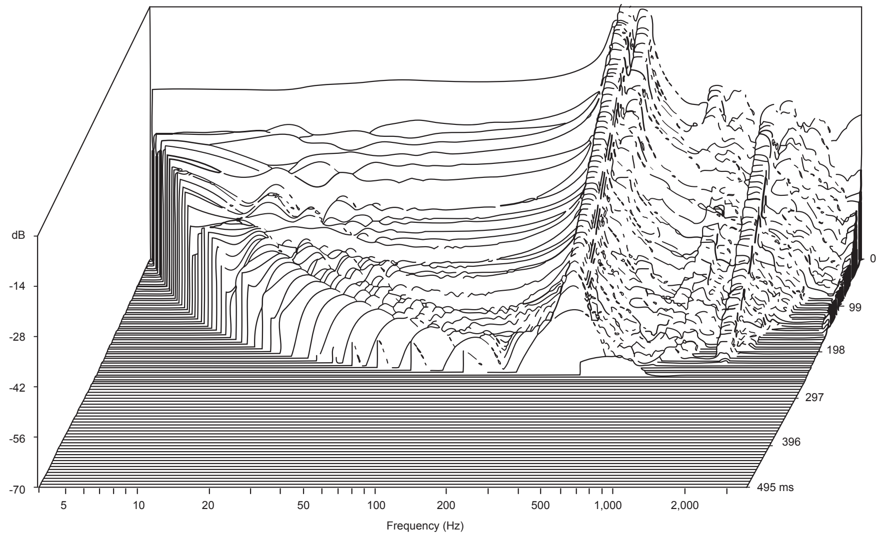

Figure 7: The cumulative spectral decay (CSD) plot (smoothed to 1/6 octave) shows the back panel vibration. The results indicate strong resonances in the 500-Hz region, which is well above the subwoofer's operating range.

INTEGRATE THE SUBWOOFER

I used an analog active crossover for the separation of the spectrum between the subwoofers and the main speakers. This sums the left and the right channels to produce a mono bass signal and uses a fourth-order Linkwitz-Riley low-pass filter at 62 Hz, which then drives the Behringer DCX2496's analog input of the subwoofer.

I connected the two subwoofers in series to the power amplifier because I only had one available. In the future, I plan to build two high-power amplifiers so each subwoofer can be driven by its own amplifier. It is important to adjust the subwoofer's levels in relation to the main loudspeakers to properly integrate it into the sound system. I achieved this by taking several measurements with the microphone at the listening positions.

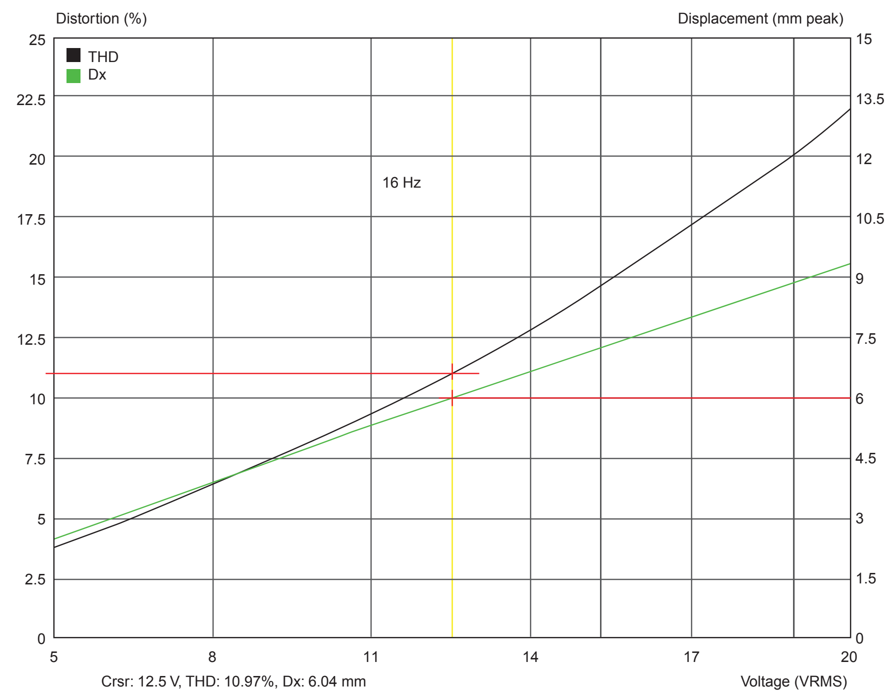

Figure 8: This shows the cone displacement measurements at 16 Hz.


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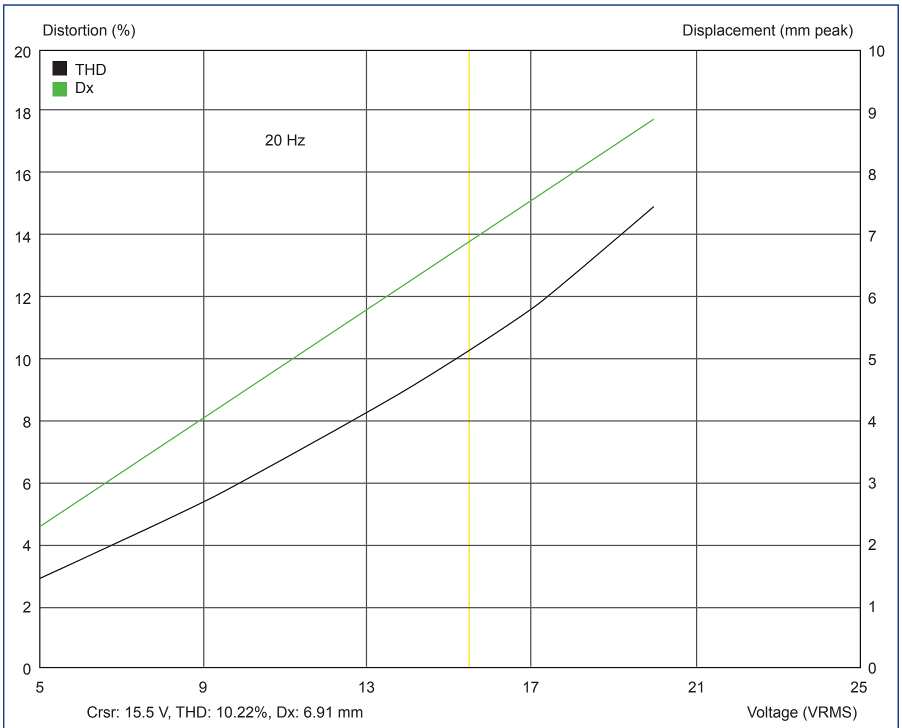

Figure 9: I also took the cone displacement measurements at 20 Hz.

After I was satisfied with the measurement results, I completed the final tuning by listening to well-known music pieces to verify the correct level adjustments. I believe this is necessary since one or two decibels up or down with the subwoofer could make an important difference.

ACCELEROMETER MEASUREMENTS

Next, I used my accelerometer system to measure the subwoofers. I designed this system a few years ago. It consists of an ACH-01-03 accelerometer, the accelerometer amplifier, which is an amplifier/processor that also computes the acceleration input signal's velocity and displacement, and an external 12-VAC power supply. The complete system is shown in **Photo 3**. It is a slightly improved version of the design that I presented in my article, "Accelerometer Testing of Loudspeaker Drivers" (*audioXpress*, June 2011).

I performed all the measurements without any electronic equalization applied on the subwoofers. To begin, I attached the accelerometer to the center of the subwoofer cone using thin double-sided tape and measured the cone acceleration. The cone's acceleration and the far-field sound pressure are proportional to each other.

Figure 5 shows the two measurements. I used the near-field method with the microphone a few millimeters away from the cone's center to take one measurement and I used the ACH accelerometer to take the second measurement.

Both measurements are almost identical up to about 200 Hz, where the deviation begins. Above 400 Hz, the two measurements are very different. This happens because the subwoofer cone ceases to behave like a rigid cone above this frequency so the accelerometer measures the local acceleration of the part of the cone where it is located and not the whole cone's acceleration.[1]

Another interesting observation is that some small peaks in the near-field measurements do not exist in the accelerometer measurement. These peaks are probably due to my listening room's standing waves.

With the accelerometer system, I then investigated the subwoofer box's panel vibration behavior. To do this, I used thin double-sided tape to attach the accelerometer to the center of subwoofer cabinet's top panel. To take these measurements, I had to shield the ACH accelerometer because a 50-Hz line voltage interference was present. This was easily done with a 5-cm × 5-cm piece of aluminum foil around the accelerometer, which was electrically connected to the amplifier ground.

As an excitation signal, I used pink noise produced by the ARTA software. A power amplifier drove the subwoofer at a high sound level. The subwoofer

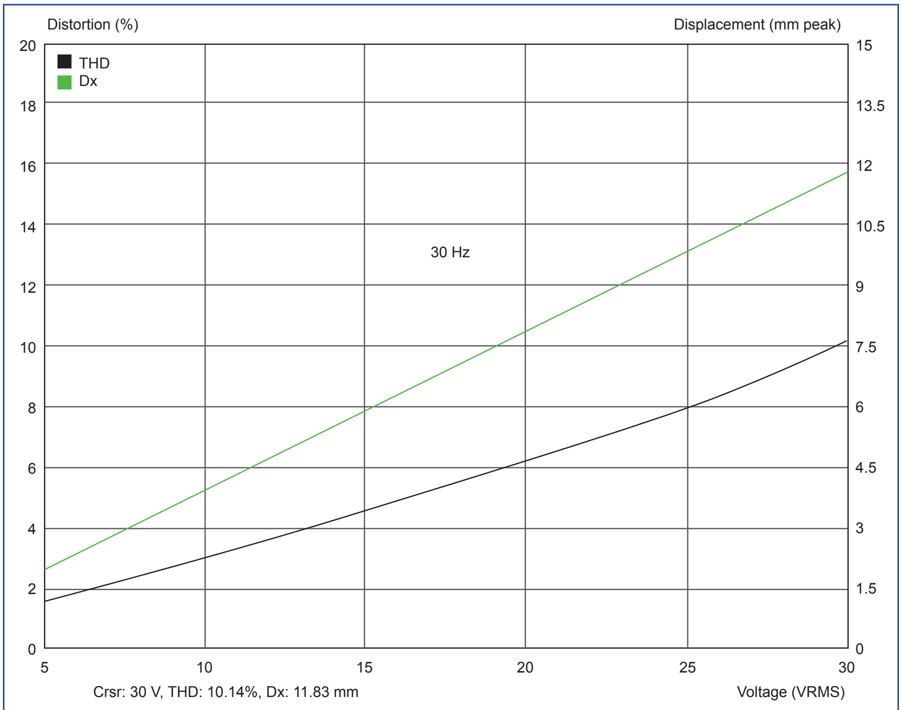

Figure 10: At 30 Hz, the cone displacement measurements show a 12-mm peak displacement.

was isolated from my listening room's wooden floor using three feet made from Sorbothane (Sorbo Shoes). These feet are hemispherical with a 50-mm diameter and a 25-mm height.

On the accelerometer preamplifier, I used an additional 20-dB gain for the panel vibration measurements because the signals measured by the accelerometers are very low in level. The ARTA software's "Scale" function compensated for the additional gain.

The measurement results are shown in **Figure 6**. The top curve is the acceleration of the subwoofer cone, which is used as a reference. Then the back panel's acceleration is shown, followed by the side panel's acceleration, finally that of the front panel. The difference between the back panel's vibration and the front and side panels' vibrations is very large. Obviously the brace I used in the box's center dampened the top and side panel vibrations (see **Photo 1**). If I built the subwoofer box again, I would use a similar brace between the back panel and the middle brace.

Although the back panel's vibration behavior is not as good as the cabinet's other panels, **Figure 7** shows there is no serious acoustical drawbacks. The cumulative spectral decay (CSD) plot of the back panel's vibration indicates strong resonances only in the 500-Hz region, which is well above the subwoofer's operation range.

I used the accelerometer system to perform some other interesting measurements including the voice coil's displacement (X_{MAX}) of the subwoofer cone in relation to the applied voltage drive and the THD of the reproduction.

These measurements were made possible with the ARTA program's STEPS capability. For more information, see the Resources section at the end of this article.

The program increases the drive voltage to the subwoofer in steps at the selected frequency and uses the microphone to measure the THD of the subwoofer's sound pressure output. The accelerometer measures the displacement of the subwoofer's voice coil. After processing the accelerometer signal, the accelerometer amplifier provides the voice coil's displacement output.

I measured the cone displacement at several frequencies to test the subwoofer capabilities. **Figure 8** shows the measurement for the 16-Hz frequency. The subwoofer needs 12.5 V_{RMS} for a peak 6-mm displacement with an 11% distortion. When the drive voltage rises to 20 V_{RMS} the peak displacement reaches 9.3 mm but now the distortion is almost 22%.

Figure 9 shows the 20-Hz measurement. The subwoofer needs 15.5 V_{RMS} for a peak 7-mm displacement with a distortion of about 10%. When the drive voltage rises to 20 V_{RMS}, the cone displacement reaches the 9-mm peak but the distortion is now 15%.

Figure 10 shows the point where things start to improve. The test frequency is now 30 Hz. The subwoofer is driven with 30 V_{RMS} and produces a peak displacement of almost 12 mm with a distortion of 10%.

Although not shown, things improve even more at 40 Hz. At this frequency, the subwoofer is driven with 30 V_{RMS}, which produces a 10.3-mm peak displacement while the distortion is only 6%. Obviously more voltage drive is needed, but unfortunately the power amplifier I used for the tests was not able to provide more than 30 V_{RMS}. Even if I had a more powerful amplifier, it would have been difficult to continue the test due to the high sound levels that it would produce.

However, the measurements provide interesting insight on the subwoofer's abilities to produce low frequencies at large peak displacements. *aX*

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ACH-01-03 Accelerometer
Measurement Specialties | www.meas-spec.com

830500 Subwoofer driver
Tymphany Peerless | www.tymphany.com/peerless

Author's Note: I have a small quantity of high-quality PCBs (with solder mask and silk screen) and engraved front panels (manufactured as shown in ***Photo 3****) for construction of the accelerometer amplifier. For more information, e-mail me at gntanavaras@gmail.com.*

By Richard Honeycutt (United States)

Tone Controls (Part 2)

Usage and Frequency Responses

"Tone" preferences vary among listeners

The Baxandall bass and treble tone controls introduced in last month's article have more or less emerged as the standard configuration for home-stereo amplifiers, although there was some experimentation in the late 1960s with the addition of a midrange control. This added control did not meet with market acceptance, probably for two reasons.

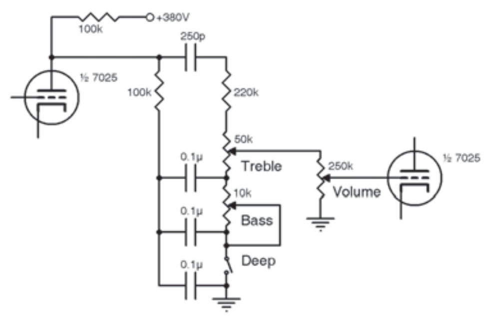


Figure 1: This bass-treble stack was used in the Fender AA-864 guitar amplifier.

MIDRANGE CONTROL DRAWBACKS

First, the definitions of "midrange" are not consistent from one listener to another. Some people think of "midrange" as the frequencies that sound to the ear as if they are in the middle of the range of hearing—about 640 to 1,280 Hz. Some think of "midrange" as the frequencies that sound as if they are in the middle of the fundamental voice range—about 250 to 300 Hz. Some manufacturers seem to think that "midrange" is from 1,000 to 5,000 Hz. These differences, plus differences in audible frequency range from listener to listener, plus the fact that our ear/brain system perceives pitch in a very nonlinear fashion, make selecting the operating range for a midrange control extremely challenging for a circuit designer.

The second reason is that changing the response in any of these ranges can have drastic effects on musical quality, so it is unlikely that a casual listener will easily find a "good" position for a midrange control.

INSTRUMENT AMPLIFIER CONTROL STACKS

However, hollow-state guitar amplifiers often have bass-mid-treble "tone stacks" with a BRIGHT or DEEP switch added as well. Since guitar players are not trying to accurately reproduce recorded music, but are instead voicing their own instruments, experienced players have little difficulty adjusting tone controls to their own liking.

Figure 1 shows a bass-treble tone stack used in the Fender AA-864 guitar amplifier chassis. This stack has a DEEP switch (shown in the "off" position). **Figure 2** shows the effects of the bass and treble controls when the DEEP switch is ON so that it places the lower 0.1-µF capacitor in parallel with the middle one. The green curve shows the frequency response with bass and treble controls set at the midpoint. The red curve shows what happens when the treble control is set at minimum. The blue curve is maximum treble. The yellow curve is maximum bass. The magenta curve is minimum bass. The red and blue curves were made with the bass control at the midpoint. The yellow and magenta curves were made with the treble control at midpoint. Actually, the bass control has very little effect in this circuit when the DEEP switch is off, since the bass frequencies are blocked by the two 0.1-µF capacitors in series, giving an equivalent 0.05-µF capacitor.

These curves also show an interesting characteristic of some tone controls. The midpoint settings of the tone potentiometers do not produce a flat response. To obtain a flat response, the treble control would have to be set below the midpoint with the bass control at the midpoint and the DEEP switch on.

Figure 3 shows a three-potentiometer (bass-mid-treble) tone stack used in many Fender lead guitar amplifiers (e.g., the Twin Reverb, etc.) This particular version is from the AB-763 chassis.

Figure 4 shows the effects of the treble and middle controls, with the BRIGHT switch off. The green curve is

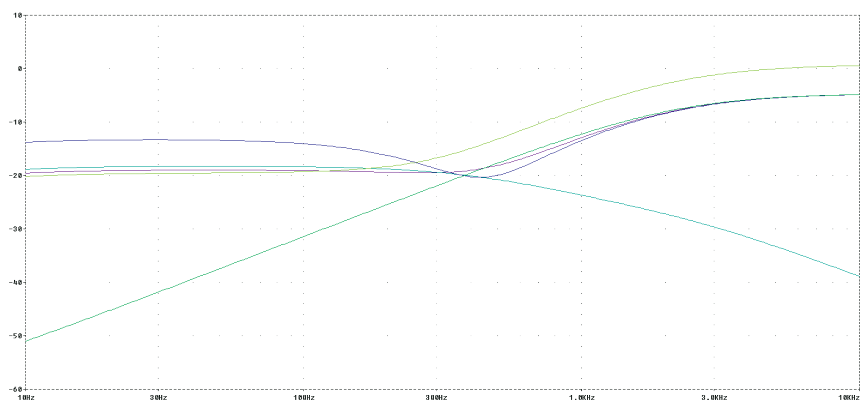


Figure 2: These curves show the effect of Figure 1's tone control circuit.

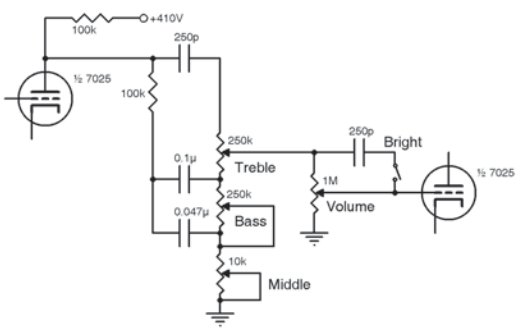


Figure 3: This bass-mid-treble tone stack was used in the Fender AB-763 chassis.

the response with all controls at their midpoints. The red curve is the maximum treble. The blue curve is the minimum treble. For both curves, bass and middle were left at midpoint. The yellow curve is minimum middle and the magenta curve is maximum middle. In these last two cases, bass and treble were at midpoint.

Note that with this circuit also, midpoint control settings do not yield a flat response. In fact, it is nearly impossible to set the controls in this circuit to obtain a flat response. Also notice that there is always a dip in the midrange, even at the middle control's highest setting. The middle control affects frequencies between 300 and 500 Hz.

Figure 5 shows what the bass control and the BRIGHT switch do when applied to the same circuit used in **Figure 3**. The green curve shows the all-controls-at-midpoint condition for reference. The red curve is minimum bass with treble and middle at their midpoints. The blue curve is maximum bass, again with treble and middle at midpoints. The yellow curve shows the effect of switching the BRIGHT switch on with all potentiometers and the volume control at midpoints. The BRIGHT switch's effect depends on the volume control's setting. At the minimum volume setting, the BRIGHT switch will have a maximum effect. At the maximum volume, it will have no effect at all.

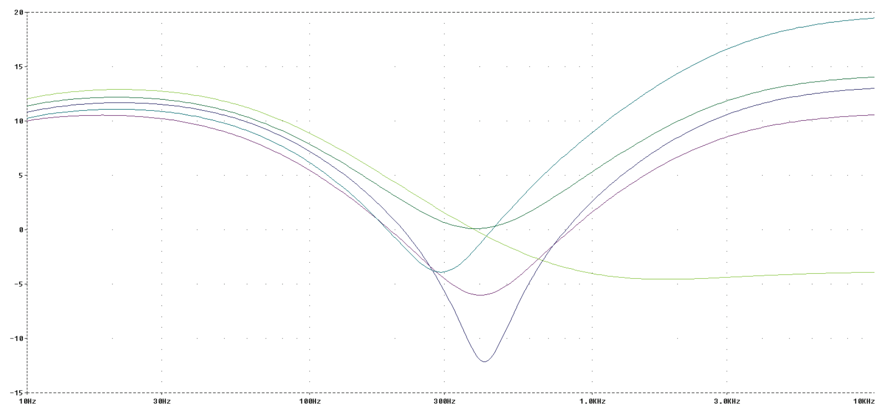


Figure 4: These curves show the effects of the treble and middle tone controls of Figure 3's circuit.

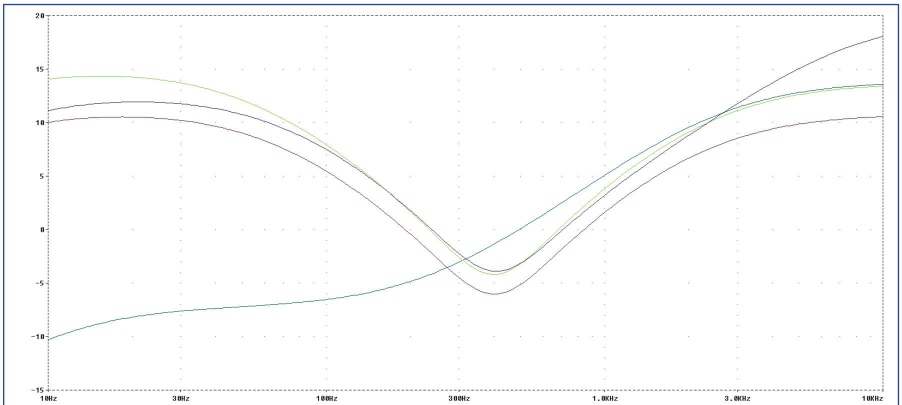


Figure 5: The effects of the bass control and the BRIGHT switch of Figure 3's circuit are shown.

LOUDNESS CONTROLS

In last month's article, I discussed loudness controls. Let's look at those in a bit more depth. First, I should mention that loudness is not the same as sound pressure level (SPL), although they are related. This is true because loudness is a psychoacoustic quantity measured in phons and SPL is a physical quantity measured in decibels (dB) SPL. In other words, loudness is the measure of a certain sensation experienced by a human when listening to a sound source. Since the human hearing apparatus has a varying frequency response depending on the SPL, loudness depends on frequency as well as SPL. This fact is illustrated by the set of curves shown in **Figure 6**.

This figure is based on the work of Harvey Fletcher and Wilden Munson of Bell Laboratories, as updated by D. W. Robinson and R. S. Dadson. Each curve represents the SPL that a pure tone must have to sound exactly as loud as a 1.3-kHz tone with an SPL equal to the number shown on the line. For example, a 20-Hz tone must have a 110-dB SPL to sound as loud as a 1.3-kHz tone at 70 dB SPL. Note that the ear is most sensitive at 4 kHz and that there is also an increase in sensitivity between 10 and 15 kHz.

I should also point out two other things. First, the curves are not parallel, so the lower the SPL, the greater the increase at low frequencies needed to sound exactly as loud as a given midrange tone. Second, these curves were created from averaged measurements of young adults with normal hearing. If these measurements were made on three people, three different sets of curves would result because of variations among the hearing characteristics of different individuals. Younger people, in general, would have higher sensitivity, especially at high frequencies. On average, older people would have decreased high-frequency sensitivity, and also a marked dip in hearing sensitivity near 4 kHz. Some older women, in particular, would have decreased sensitivity at low frequencies. The reasons for these variations are known, but are not within the scope of this article.

The important point is that when we listen at lower levels, the bass and high treble sound weak. Many pieces of audio equipment produced in the past included loudness compensation circuitry intended to boost bass and high treble when the volume control was turned down. Often, a "loudness" switch was also included so that listeners could disable the loudness compensation if they so desired. **Figure 7** shows the frequency response of a loudness compensation control formerly manufactured by IRC. The percentages correspond to the control's setting, which also acted as the volume control. These controls were used in some home-audio amplifiers and radio receivers.

Figure 8 shows the control's schematic diagram. The left-hand potentiometer had a 500-kΩ value. The other two were 100 kΩ each. Typically the LC1 was fed from one tube's plate, and the output went to the following tube's grid.

As a lower-cost alternative, some manufacturers used tapped volume control with capacitors connected to the taps to create a frequency response that boosted bass and high

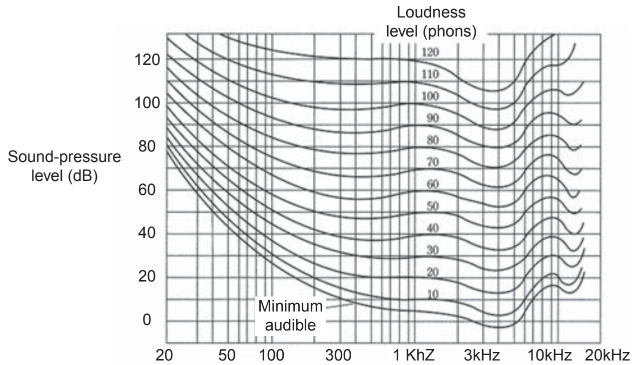


Figure 6: The loudness depends on the frequency as well as the SPL.

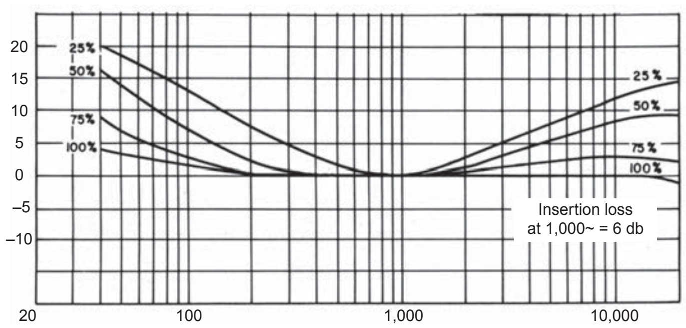


Figure 7: The IRC model LC1 loudness control produced these frequency responses.

treble at low volume settings.

The underlying idea of loudness compensator controls was to enable the "volume control" to really control loudness—not just audio level—by automatically compensating for the ear's level-dependent frequency response. The problem with that idea is that the rest of the audio chain—including the listener's hearing apparatus—is not standardized. Although in a table radio the speaker sensitivity (decibel at 1 W at 1 m) would be known by the circuit designer, the same cannot be said for hi-fi speaker systems. If the sensitivity varies by 12 dB from what the amplifier designer assumed when designing the loudness control, the bass and treble boost could easily be off by 6 dB or more. Thus critical listeners have generally preferred to use bass and treble controls to make frequency-response adjustments so that they are in complete control of the results. In this way, they can compensate for the variability of the speakers and the listeners' hearing.

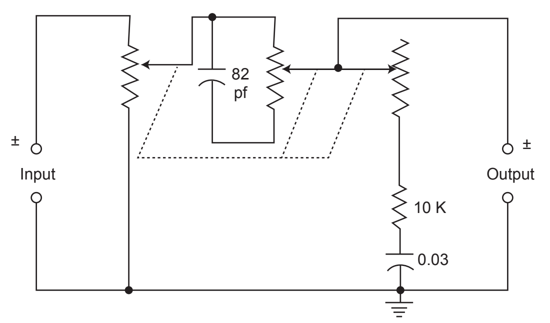


Figure 8: This is the circuit of the IRC LC1 loudness control.

The use of loudness compensation circuitry has often been somewhat like the use of graphic equalizers set to a smiley face: too much bass and too much treble. For that reason, loudness compensation circuits are pretty much a relic of the past.

In my next article, I'll examine hollow-state dynamics processors, mainly compressors. *aX*

MEMBER PROFILE


Costas Sarris

Member Name: Costas Sarris

Location: Thessaloniki, Greece

Education: BSc, Business Computers

Occupation: He currently works as a medical engineer at Biolab Lp Diagnostics

Affiliations: He is a monolith magnetics audio transformers tester for Mythos Audio, a high-end speaker manufacturer, and he belongs to several popular web-based hi-fi forums.

Audio Interests: Costas said he is most enthusiastic about projects that involve tubes. Although he enjoys all things audio, Costas said he prefers to work on power tube amplifiers, tube preamplifiers, and guitar amplifiers.

Most Recent Purchase: Costas recently bought a pair of NOS 12AX7 tubes, which he plans to use in an upcoming project.

Current Audio Projects: He has several projects in the works including a 2A3 single-ended tube amplifier, a 300B SE power tube amplifier, and a single-ended guitar tube amplifer based on a 6L6GC power tube.

Dream System: Costas said his dream system includes an Altec Model 19 Circa 1977 with a Audio Note Ongaku power amplifier and an Audio Research Reference 2 preamplifier.

 By Vance Dickason (United States)

A FaitalPRO Compression Driver and the STH100 Horn

Loudspeaker features a Ketone polymer annular-shaped diaphragm

For this month's article, I put one of FaitalPRO'S new compression drivers, the HF106 along with a FaitalPRO STH100 Elliptical Tractrix horn to the test (see **Photo 1**). The HF106 is a new edition to the company's series of 1" diameter compression drivers that use Ketone polymer diaphragms. The entire series is designed to be used with the STH100 horn.

FaitalPRO HF106

The HF106, like the entire series, is an interesting compression driver that shares several unique features with the other models. It includes a Ketone polymer annular-shaped diaphragm and an annular-shaped phase plug (note the HF100, the HF104, and the HF105 use radial-shaped phase plugs). The driver's throat diameter is 25.4 mm (1") with the diaphragm coupled to a 44 mm (1.73") diameter voice coil wound on a Kapton former with aluminum wire. Other features include a neodymium ring magnet, a cast aluminum body, 60-W AES-rated power handling (120-W

Photo 1: Faital's HF106 compression driver is shown coupled with the STH100 horn.

Photo 2: Faital's STH100 horn is shown without a driver.

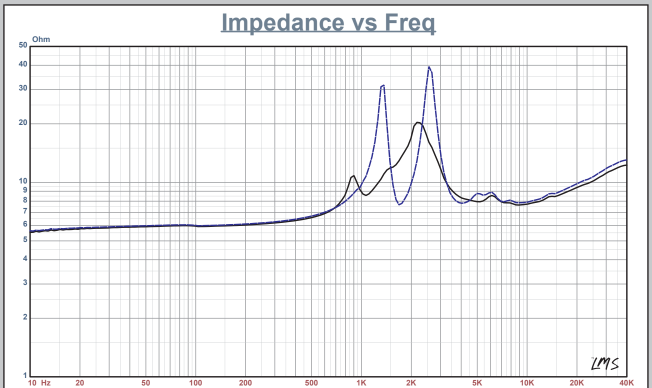


Figure 1: Faital HF106/STH100 free-air impedance plot

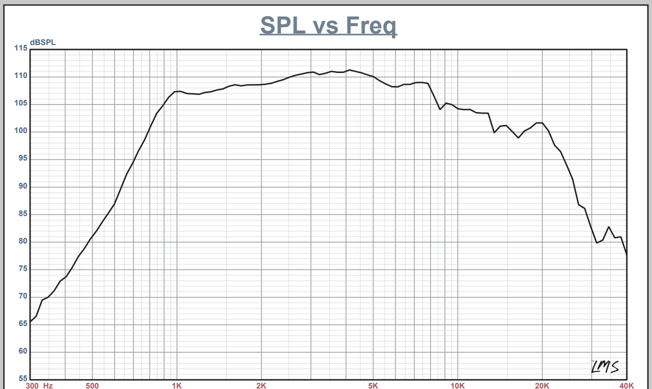


Figure 2: Faital HF106/STH100 on-axis frequency response

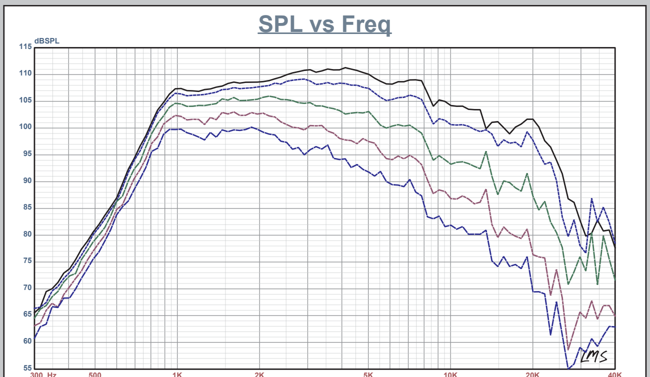


Figure 3: Faital HF106/STH100 horizontal on- and off-axis frequency response (0 = solid; 15° = dot; 30° = dash; 45° = dash/dot; 60° = dash)

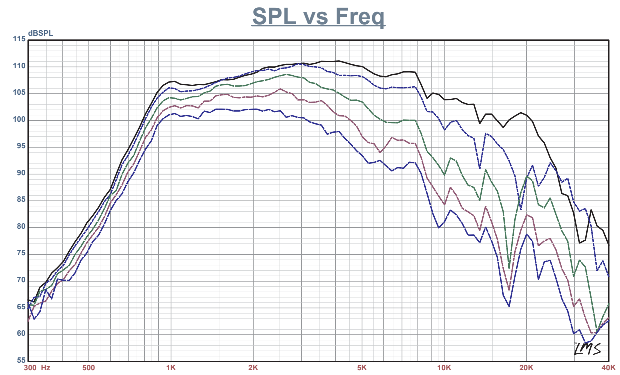


Figure 4: Faital HF106/STH100 vertical on- and off-axis frequency response (0° = solid; 15° = dot; 30° = dash; 45° = dash/dot; 60° = dash)

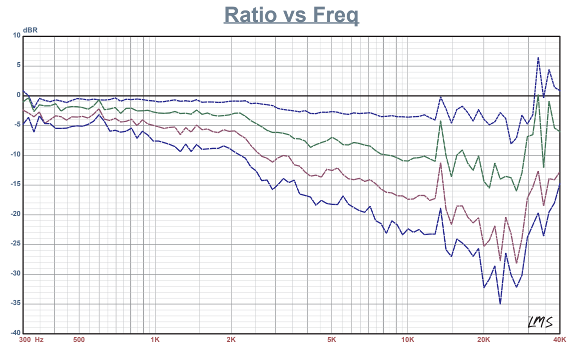


Figure 5: Faital HF106/STH100 normalized horizontal on- and off-axis frequency response (0° = solid; 15° = dot; 30° = dash; 45° = dash/dot; 60° = dash)

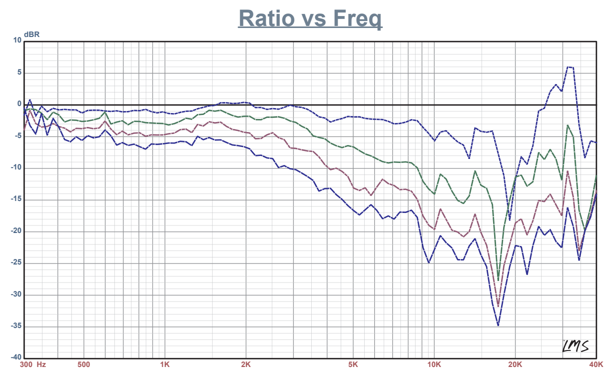


Figure 6: Faital HF106/STH100 normalized vertical on- and off-axis frequency response (0° = solid; 15° = dot; 30° = dash; 45° = dash/dot; 60° = dash)

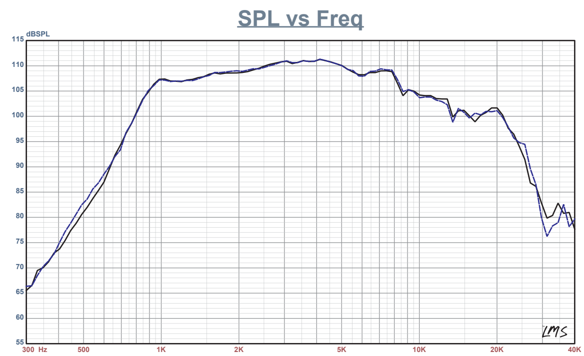


Figure 7: Faital HF106/STH100 two-sample SPL comparison

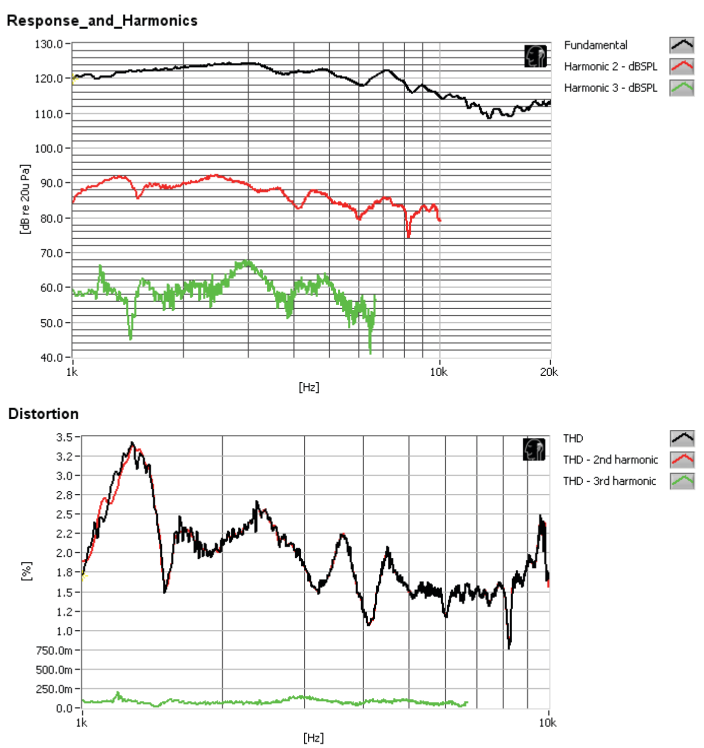


Figure 8: Faital HF106/STH100 SoundCheck distortion plots

maximum), and solderable aircraft terminals. The STH100 horn supplied with the HF106 driver has a 1″ diameter throat with 80° horizontal × 70° vertical short elliptical tractrix flare (see **Photo 2**).

I used the LinearX LMS analyzer to produce the 200-point stepped sine wave impedance plot shown in **Figure 1**. The solid black curve shows the HF106 mounted on the STH100 horn and the dashed blue curve represents the compression driver without the horn. With a measured 5.4-Ω Re, the HF106/STH100's minimum impedance was 7.65 Ω at 8.9 kHz.

For the next test sequence, I mounted the HF106/STH100 combination in an enclosure with a 10″ × 15″ baffle and used a 100-point gated sine wave sweep to measure the horizontal and vertical on- and off-axis at 2.83 V/1 m. **Figure 2** displays the compression driver/horn combination's on-axis frequency response. With a 110-dB, 1-W/1-m rated sensitivity, it has a 111.3-dB, 2.83-V/1-m peak output at 4.1 kHz. The sound pressure level (SPL) profile measures ±4.5 dB from 1 to 10 kHz. (The HF106's recommended crossover frequency is a 1.3-kHz minimum with a second-order network.) Since this horn's coverage is 80° horizontal × 70° vertical, you wouldn't expect much difference in

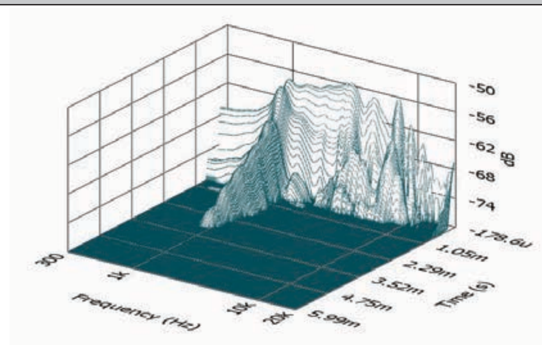


Figure 9: Faital HF106/STH100 SoundCheck CSD waterfall plot

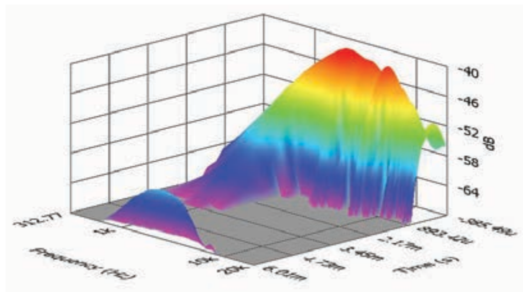


Figure 10: Faital HF106/STH100 STFT plot

the horizontal and vertical off-axis plots. **Figure 3** shows the horizontal orientation. **Figure 4** shows the vertical orientation. **Figure 3**'s plot with the off-axis normalized to the on-axis response is shown in **Figure 5**. **Figure 4**'s plot with the off-axis normalized to the on-axis response is shown in **Figure 6**. **Figure 7** shows the two-sample SPL comparison. Both samples are closely matched.

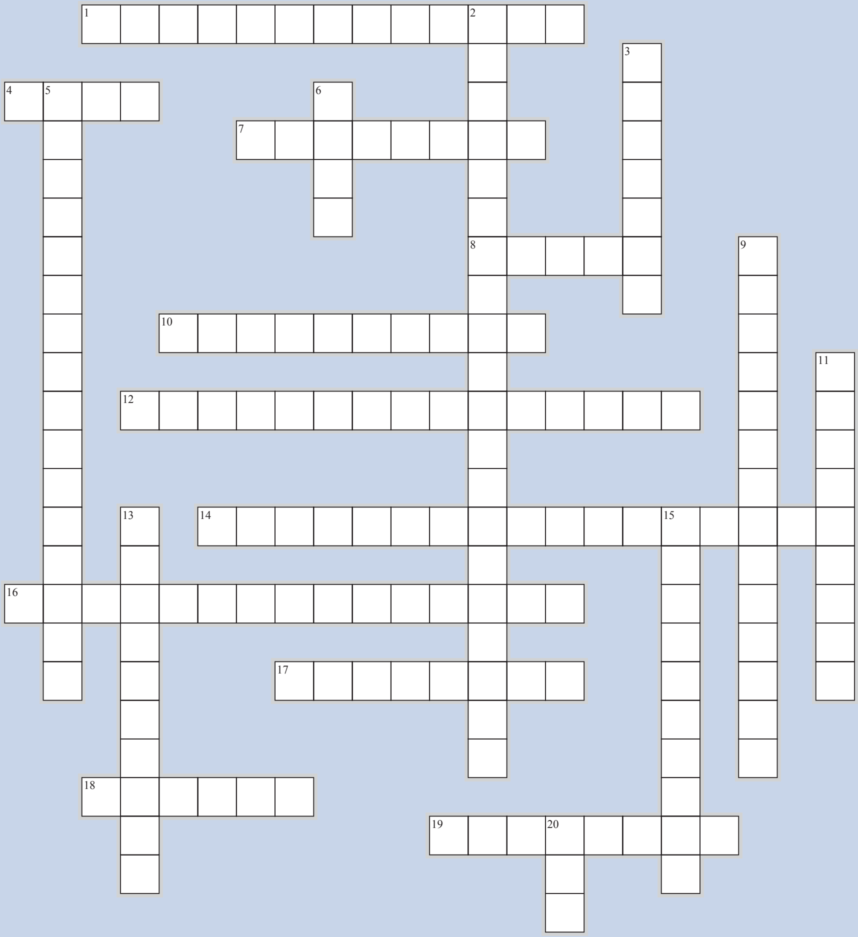
For the remaining tests, I used the Listen SoundCheck software, the AmpConnect ISC analyzer, a 0.25" SCM microphone, and a power supply to measure the distortion and generate time-frequency plots. For the distortion measurement, I mounted the HF106/STH100 combination with the same baffle I used for the frequency response measurements. I used a pink noise stimulus to set the SPL to 104 dB at 1 m (1.73 V) and placed the Listen microphone 10 cm from the horn's mouth to measure the distortion. This produced the distortion curves shown in **Figure 8**.

I used SoundCheck to get a 2.83-V/1-m impulse response for this driver and imported the data into Listen's SoundMap time-frequency software. **Figure 9** shows the resulting cumulative spectral decay (CSD) waterfall plot. **Figure 10** shows the short-time Fourier transform (STFT) plot. The SoundMap software, supplied with SoundCheck 12.0, no longer supports the classic "MLSSA" waterfall curve orientation. I used the default orientation for both SoundMap plots, and I will continue to do so in the future.

I have been told that the Ketone polymer diaphragm yields a smooth subjective performance. However, from the previous data, the HF106/STH100 looks like a nice addition to FaitalPRO's lineup of pro sound compression drivers.

For more information, visit www.faitalpro.com, or in the US, contact Faital USA, Keith Gronsbell, 220 West Parkway, Unit 13, Pompton Plains, NJ 07444, Phone (516) 779-0649, Fax (973) 835-5055, and e-mail kgronsbell@faital.com. *aX*

Crossword



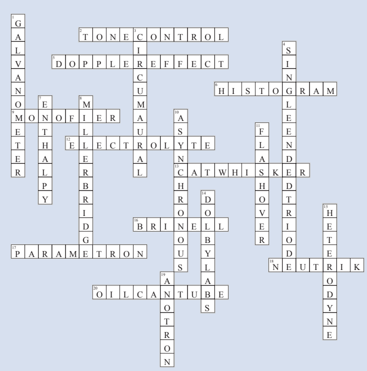
Editor's Note: Answers to this puzzle are posted at audioxpress.com/crossword and will be available in the next issue.

ACROSS

1. Optimized for the highest audible frequencies
4. Provides single-tile mulitcore microcontrollers
7. Loudspeaker with two or more identical woofers operating simultaneously in a single enclosure
8. Credited with dynamic drivers' modern design
10. 19th-century horn loudspeaker
12. Relates to the natural extension f* of a real function f
14. Measures a rotating shaft's speed and direction [two words]
16. One cause of ringing artifacts in signal processing [two words]
17. An audio engineering visionary
18. Enables an object's rotation around a single axis
19. Annular shaped

DOWN

2. Microphone and preamplifier specification [three words]
3. Located between the voice coil and the primary cone
5. These speaker types don't require suspensions or voice coils
6. Limits use of dangerous materials
9. Environmental factors affecting transducers [two words]
11. I.e., cumulative spectral decay
13. Uses multichannel mixing technology
15. T is the ANSI symbol for copper and this material
20. Alt + 234 provides this unit's symbol



Here are the answers to the August 2013 puzzle.

APHEX RELEASES AUDIO XCITER APP UPDATE

Aphex, a leader in audio enhancement technology for more than 35 years, has updated its popular Audio Xciter app for the Android smartphone and tablet platform. The updated version provides a more responsive user interface and greater stability, fixing previous problems with in-app upgrades.

Audio Xciter has been popular with iPhone and iPad owners. This update brings Android users a new listening experience with Aphex's patented audio processor technology—the same technology pros have been using for years in the studio and on stage. The Audio Xciter app's familiar interface provides instant access to the user's library, playlists, and Bluetooth playback support, with studio-quality enhanced playback.

An update for the Audio Xciter app on the iOS platform (iPhones, iPods, and iPads) is currently in the works. Try Audio Xciter for Android platforms for free on GooglePlay (http://bit.ly/17Rt6Z7).

GLOBAL SPECIALTIES RELEASES NEW DIGITAL AC/DC POWER SUPPLY

Global Specialties has introduced the 1315D, a new rugged, general-purpose AC/DC power supply that features variable outputs of 0 to 25 V at 5 A with digital LED (green) displays for motoring voltage and current outputs.

This unique power supply is a low-cost, space-saving solution for applications requiring AC/DC voltages. The 1315D is specifically designed for educational use and prototyping, covering a range of circuit testing applications. This simple AC/DC power source can demonstrate the concept of alternating and non-alternating voltages. It can also be used for simple experiments powering lamps, resistors, and so forth. The unit includes a voltmeter and an ammeter for AC/DC voltage and current readings. The 1315D costs $590.

For more information, visit www.globalspecialties.com.

Global Specialties's 1315D is a durable general-purpose AC/DC power supply.

ADLINK INTRODUCES A SIGNAL-ACQUISITION MODULE

ADLINK Technology now offers the USB-2405, a four-channel USB 2.0 dynamic signal-acquisition module. The module's built-in IEPE excitation current source provides 2 mA on each AI channel. BNC connectors enable the USB-2405 to provide high accuracy and excellent dynamic performance for microphone and accelerometer measurement in vibration and acoustic applications.

The USB-2405 offers a portable solution for time-frequency analysis and research that includes accuracy with low temperature drift, built-in anti-aliasing filters, support for flexible trigger mode, and a USB power.

ADLINK's USB-2405 supports four analog input channels.

The USB-2405 supports four analog input channels simultaneously sampling up to 128 kS/s. It delivers 100-dB dynamic range and –94-dB THD of dynamic performance. Built-in anti-aliasing filters enable the filter cutoff frequency to be automatically adjusted to the sampling rate, suppressing out-of-band noise and avoiding measurement distortion.

ADLINK's USB-2405 also supports auto-calibration for ensured accuracy and minimizes temperature drift in the field. The USB-2405's low DC measurement drift, along with temperature deviation, optimizes accuracy in spite of the environment.

The USB-2405 provides a lockable USB to enhance connectivity. The included multi-functional stand fully supports desktop, rail, or wall mounting. ADLINK's easy-to-use U-Test software is included at no extra charge. With no programming required, the USB-2405 delivers fast, easy instrument setup and quality data acquisition. The USB-2405 supports Windows 8, Windows 7, and Windows XP OSes and is fully compatible with third-party software (e.g., LabVIEW, MATLAB, and Visual Studio.NET).

For more information, visit www.adlinktech.com.

XL2-TA RECEIVES TYPE APPROVAL

The XL2-TA sound level meter has received Class 1 type-approval from the Physikalisch-Technische Bundesanstalt (PTB) in Germany. The XL2-TA in combination with the M2230 measurement microphone, is officially listed as a type-approved Class 1 sound level meter. The XL2-TA is suitable for environmental noise, occupational health, and sound insulation applications where the measurements require certification.

Receipt of the type-approval certificate confirms the XL2-TA (in combination with the M2230 measurement

The XL2-TA Sound Level Meter has received Class 1 approval.

microphone and the ASD cable) offers full compliance with the Class 1 sound level meter requirements in accordance with the International Electrotechnical Commission (IEC) standards 61672 and 61260 and meets the American National Standards Institute (ANSI) S1.4 Type 1 performance. The XL2-TA offers certified spectral measurements in octave and third-octave resolution, which further comply with Class 0 filter specifications.

The M2230 measurement microphone connected to the XL2-TA via the ASD cable configuration enables the user to monitor the level readings on the instrument display while Class 1 measurements are taken. The NTi Audio Precision Calibrator completes the portable measurement kit offered in the compact system case.

Customers may upgrade their XL2 analyzers to a type-approved XL2-TA by adding a retrofit kit. Contact your local NTi Audio partner for the upgrade. For more information about the XL2 Analyzer and obtaining the type-approval Option, visit www.nti-audio.com/XL2.

TASCAM SUPPLIES PROFESSIONAL-GRADE HEADPHONES

TASCAM, the Pro Audio Division of TEAC America, has been supplying and revolutionizing the market with professional-grade products for decades. Its most recent contribution is the TH-2000 headphones.

TASCAM's TH-2000 professional-grade headphones are designed for serious audiophiles.

The TH-2000 headphones offer a powerful bass response, midranges that round out the mix, crystalline high-frequency ranges for a strong presence and clarity, comfort for hours of listening and use, durability, and TASCAM quality.

TASCAM, creator of the original TH-02 headphones is taking its professional-grade headphones a step further with the TH-2000. The TH-2000's main features include: a foldable design for easy compact transport, circumaural ear cuffs with an industrial strength flexible headband, and a closed-back isolating design.

The headphone specifications include a 53-mm driver diameter, 60-Ω impedance, 101-dB ±3 sensitivity, 22-kHz-to-18-Hz frequency response, and 1,800-mW maximum power.

For more information, visit http://tascam.com. *aX*

CONTRIBUTORS

Ken Bird ("The Single-Tower Stereo," p. 22) has had a 40-year career in engineering and product magagement in consumer, industrial, and telecommunications electronics. Ken has since retired to his native Missouri and now pursues his interest in building speakers and tube amplifiers, along with model railroading, amateur radio (WOKLB), and digital photography.

Vance Dickason ("A FaitalPRO Compression Driver and the STH100 Horn," p. 38) has been working as a professional in the loudspeaker industry since 1974. He is the author of *Loudspeaker Design Cookbook*—which is now in its seventh edition and published in English, French, German, Dutch, Italian, Spanish, and Portuguese—and *The Loudspeaker Recipes*. Vance is the editor of *Voice Coil: The Periodical for the Loudspeaker Industry*. Although he has been involved with publishing throughout his career, he still works as an engineering consultant to a number of loudspeaker manufacturers. This currently includes Samsung, The AVC Group (Niles and Sunfire), Artison, and Emotiva.

Dr. Richard Honeycutt ("Tone Controls (Part 2): Usage and Famous Frequency Responses," p. 34) has been an audio enthusiast, musician, recording engineer, broadcast radio engineer (First Class Commercial licensee since 1969), radio announcer, electronics repairman, tube guitar amplifier designer, speaker system designer, pro-sound/video designer, and college electronics teacher. He is now a consultant in acoustics and electroacoustics. He earned a PhD in Electroacoustics from the Union Institute, and lives in Lexington, NC, with his wife, Betty Jane, near his daughter Alyson, her husband, and two of his three grandchildren.

Mike Klasco ("Loudspeaker Failures and Protections (Part 2): Passive Speaker Protections," p. 8) is the president of Menlo Scientific, a consulting firm for the loudspeaker industry, located in Richmond, CA. He is the organizer of the Loudspeaker University seminars for speaker engineers. Mike specializes in materials and fabrication techniques to enhance speaker performance.

George Ntanavaras ("The Horizon Subwoofer," p. 26) graduated from the National Technical University, Athens, Greece, in 1986 with a degree in Electronic Engineering. He currently works in the Development Department for a Greek electronics company. He is interested in the design of preamplifiers, active crossovers, power amplifiers, and most loudspeakers. He also enjoys listening to classical music.

Steve Tatarunis ("Loudspeaker Failures and Protections (Part 2): Passive Speaker Protection," p. 8) has been active in the loudspeaker industry since the late 1970s. His areas of interest include product development and test engineering. He is currently a support engineer at Listen, in Boston, MA, where he provides front-line technical support to the SoundCheck test system's global user base.

Tom Yeago ("Maggies Reclaimed," p. 17) has been a stereo buff since the early 1970s. However, his formal training lies elsewhere, which is not necessarily a bad thing. He's been mucking about with loudspeaker innards and other audio hardware since the late 1980s, leading to considerable accretion and occasionally an understanding of the whys and wherefores of the builds. Then he begins the modifications. Tom's last published project was a turntable/tonearm with an alignment piece thrown in as a bonus (*audioXpress*, December 2010).

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