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Speaker FOCUS Edition!

Fresh From the Bench
SB Acoustics Rinjani
and Ara Speaker Kits

By Oliver A. Masciarotte

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A Pro DIY Subwoofer Project
Using a 1975 EV 30" Woofer

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You Can DIY!
A Hybrid Electrostatic Speaker
with Tailored Dispersion

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Questions & Answers
With Zoltán Bay,
Creator of the
Bay Radial Speaker

Sound Control
Acoustical Measurements
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Fresh From the Bench
RME ADI-2 Pro Converter
and Headphone Amplifier
By Bennett Prescott
and Stuart Yaniger



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Speaker State of the Industry

Loudspeakers and speaker building are at the core of *audioXpress*. In this edition, we once again celebrate the spirit of *Speaker Builder*, the publication which, together with other previous titles, is the origin of *audioXpress* and our sister publication *Voice Coil*.

This year, *Voice Coil* celebrates its 30th year since Vance Dickason and Ed Dell decided to create the magazine—initially as a newsletter, which quickly turned into an indispensable information resource for the loudspeaker industry. Both *audioXpress* and *Voice Coil* are proud to serve—each from its own specific perspective—as an information and education resource for loudspeaker professionals. While *Voice Coil* addresses the driver design and manufacturing from the OEM and ODM perspectives, *audioXpress* widens its scope to DIY and general speaker building and design, including electronic and electroacoustic components. *Voice Coil* is available free, with controlled circulation for industry professionals (subscription available here: www.audioxpress.com/page/Voice-Coil-Subscriber-Services)

Considering this month's Speaker Focus, I should also highlight that our 2017 edition of the *Loudspeaker Industry Sourcebook* (LIS) is now available (gotomyllis.com), together with a complete updated online directory (www.loudspeakerindustrysourcebook.com). LIS 2017 is filled with high-quality industry features and reports, which provide an invaluable perspective of the speaker industry. Best of all, the content is freely accessible. Just follow the respective Internet links.

As we do every year, for LIS 2017, we asked audio and speaker industry experts about their expectations for business and the trends they consider most relevant. And from that small but extremely knowledgeable sample, we can anticipate significant trends in perspective for the next 12 months.

No doubt, headphones, earphones, and truly wireless earbuds, will continue to propel the personal audio category and generate even stronger growth. As consumers embrace the wireless revolution, and audio quality and reproduction also continues to improve, those will continue to be growth areas for several years.

Nevertheless, there is no doubt that the fastest growing segment of the audio industry is currently in loudspeakers. Consumers are embracing the new class of portable wireless, multi-room audio, and new categories of smart- and voice-controlled speakers are gaining traction. Battery-operated, low-power electronics, and wireless technologies are now propelling a new class of devices, which will soon be transversal from home audio to personal and wearable categories, and even automotive.

In 2016, we have seen how the Amazon Echo voice-controlled speaker inspired the industry for the possibility of smart speakers, creating a new category. Connected speakers—because that will be the new class, not necessarily just voice-controlled—are already the fastest growing audio product segment for 2017 and will be large-volume products for 2018, with all major global brands getting on board. Adding to those platforms the combination of powerful DSP, microphone arrays, all types of sensors, and the latest wireless standards and network connectivity creates a foundation with incredible potential to revolutionize all loudspeaker applications, including whole-home audio and home theater. Manufacturers need to recognize the advances in these areas and leverage the extraordinary resources that will be available to them.

There's also another notable trend that's important for the loudspeaker industry. Different approaches in speaker design and certainly in transducers, will inspire new designs in applications that are not always obvious. New ideas are entering the market and loudspeaker manufacturers are rethinking the conventional and challenging what's been possible.

João Martins
 Editor-in-Chief



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COLUMNISTS

Vance Dickason has been working as a professional in the loudspeaker industry since 1974. He is the author of *Loudspeaker Design Cookbook*—which is now in its seventh edition and published in English, French, German, Dutch, Italian, Spanish, and Portuguese—and *The Loudspeaker Recipes*. Vance is the editor of *Voice Coil: The Periodical for the Loudspeaker Industry*, a monthly publication. Although he has been involved with publishing throughout his career, he still works as an engineering consultant for a number of loudspeaker manufacturers.

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

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

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

Ron Tipton has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at White Sands Missile Range. In 1957, he started Testronic Development Laboratory, which became TDL Technology, to develop audio electronics. All product sales and services were terminated on December 31, 2015, but the TDL website is still online with a variety of audio information and downloads.

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● Speaker Special Features

20 SB Acoustics Rinjani and Ara Speaker Kits

A Gift for Fun

By Oliver A. Masciarotte

Learn more about SB Acoustics' Rinjani and Ara speaker kits, which feature high-end Satori drivers. Oliver Masciarotte had the opportunity to live for some weeks with the exquisite floorstand Rinjani speakers. He also compares the Rinjani's performance with the smaller Ara two-way bookshelf speakers.

30 True Bass in a Large Space A Pro DIY Subwoofer Project from 1975 Using Electro-Voice's 30W 30" Woofer

By Don Keele, Jr.

Inspired by a DIY *audioXpress* article from February 2017 titled: "True Bass in a Small Space," Don Keele describes an equivalent solution from 1975 that provides about the same response and acoustic output but uses a high-efficiency low-excursion 30" driver mounted in a massive 76 ft³ vented-box, driven by a 60 W amplifier, creating a total system weight that exceeds 800 lbs.!

44 The Thinman Project

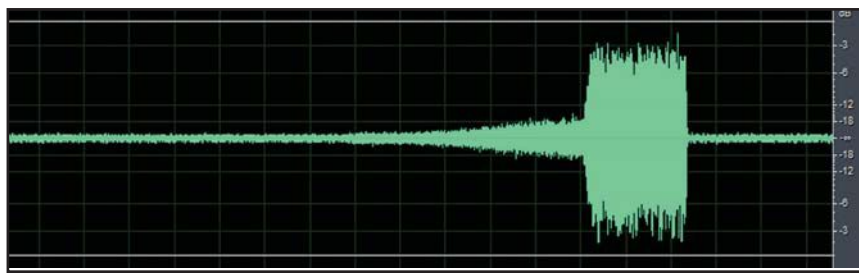
By Ken Bird

Galvanized by the revival of 2.1 analog stereo, this variation of a thin speaker is 5.5" wide and 49.5" high, with a footprint of less than a square foot, built in a three-way configuration, using an 8" woofer side firing in the ported section of the enclosure, with a dual midrange and dome tweeter in a classic D'Appolito MTM arrangement in their own sealed enclosure.

52 Hybrid Electrostatic Loudspeaker with Tailored Dispersion

By Charlie Mimbs

Build your own hybrid electrostatic speaker with tailored dispersion and "a really cool stator design." As Charlie Mimbs explains, "This speaker is three generations evolved from my original ESL described in the April 2012 issue of *audioXpress*. What's new and completely different is the electrostatic panel, which now uses insulated copper wire stators and electrical segmentation to tailor its dispersion."



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8 RME ADI-2 Pro Review

By Bennett Prescott and Stuart Yaniger

The RME ADI-2 Pro is a feature-packed update to the RME ADI-2 converter/headphone amplifier, providing a host of innovative features that will appeal to both mastering engineers and high-end audio aficionados alike. Independent measurements conducted by Stuart Yaniger back up the manufacturer's claims.

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RME ADI-2 Pro Review



Photo 1: The ADI-2 Pro converter/headphone amplifier is shown with a pair of Sennheiser HD 700 headphones.

After spotting the RME ADI-2 Pro at a recent National Association of Music Merchants (NAMM) show, Bennett Prescott requested the loan of a unit so he could examine it more closely. He also enlisted his colleague Stuart Yaniger to put the converter/headphone amplifier combination through a series of measurement tests.

By
Bennett Prescott and Stuart Yaniger
(United States)

I took advantage of the relative peace and quiet during the last day of the National Association of Music Merchants (NAMM) show this year to walk the halls and see if I could find anything new and interesting. This seems to be the golden age of headphones. Even at a decidedly not hi-fi-oriented event such as the NAMM show, there were a dozen manufacturers showing new or updated models. I'm always looking to deliver great audio on a budget, and today \$800 can buy one of the best headphone setups available. Try buying a good pair of monitors, not to mention an acoustically treated room, for that much! Headphones will never replace loudspeakers for me, but for less dedicated listeners, they are an outstanding option right now.

My meandering eventually brought me by the large, well-attended, and very blue RME booth. While there, I noticed RME's new ADI-2 Pro converter/headphone amplifier (see **Photo 1**) plugged into a pair of nice planar headphones. The ADI-2 Pro was introduced in late 2016, but new to me, so I gave it a listen and determined that it's impossible to

critically listen to anything at a trade show. Never one to be discouraged, I emailed Synthax—RME's rep in the Americas—after the show and asked if I could borrow a unit to review, to which they generously agreed.

First Impressions

If you don't know the German brand RME, it has a stellar reputation in pro audio and recording for making no-frills, technically perfect, supremely useful audio tools. Its product line includes optical and copper format converters, high channel count interfaces with and without microphone preamps, and several kinds of DAs and ADs for USB, Thunderbolt, and PCI. RME is also one of the strongest purveyors of MADI (AES10) multichannel audio hardware, used for transport in many high-end concert and broadcast mixing consoles. The short story is that this company knows conversion and digital.

The ADI-2 Pro (not to be confused with the older, less headphone-oriented ADI-2) is a sort

Sound thinking in coaxial design

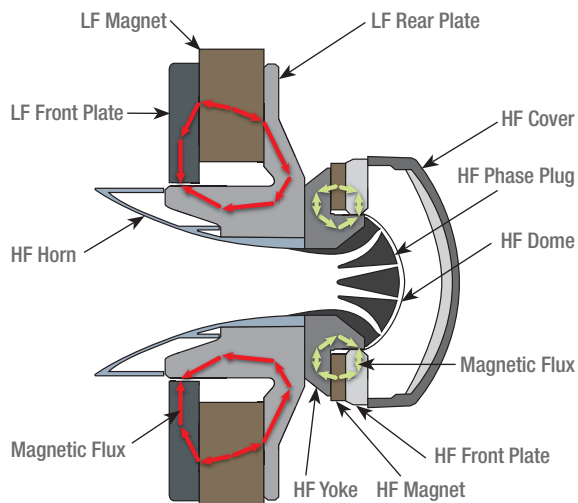
How a common magnet motor delivers a better sounding, lightweight, full-range speaker

Coaxial drivers have long been chosen as an alternative to two-way speaker systems, delivering benefits in size, weight and off-axis response. By concentrically aligning the low and high frequency components, coaxial loudspeakers act as a single source.

Moving the concept forward, Celestion's FTX coaxial range features fully combined LF and HF components that are powered by a Common Magnet Motor Assembly - i.e. using the same magnet for both elements. This enables the voice coils and hence the acoustic centres of the two drivers to be brought much closer together. The end result is a significant improvement in signal coherence and time alignment, and the most natural sounding audio reproduction.

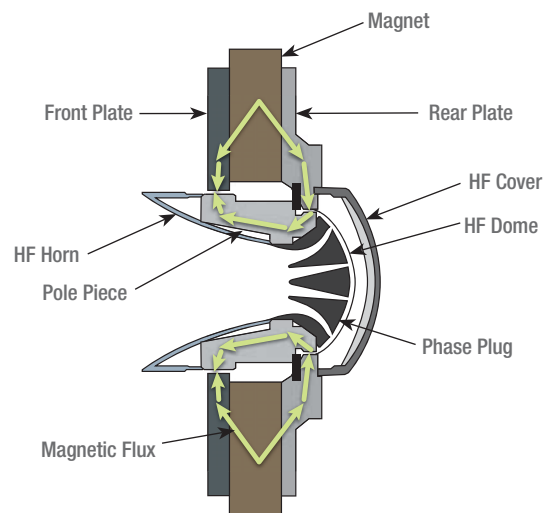
The magnetic flux is optimally distributed between the two elements (calculated using FEA), to achieve the best balance between HF and LF response (see diagrams below). Additionally, the use of a single magnet assembly also means lighter weight and a more compact profile, compared to more conventional dual motor designs.

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Traditional Coaxial Driver

With a magnet for each driver, the traditional coaxial design is heavier and the voice coils are further apart.



Celestion Common Magnet Motor Design

Celestion's single magnet design is a more compact, lighter-weight alternative with significant improvements in signal coherence and time alignment.

Find out more



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CELESTION

Project Files

To download the user guide for the RME ADI-2 Pro, visit <http://audioXpress.com/page/audioXpress-Supplementary-Material.html>

of do-everything bi-directional digital-to-analog (DAC) converter with dual headphone outputs on the front, XLR and optical S/PDIF inputs and outputs on the back, plus copper S/PDIF and AES available on a breakout cable (included), which connects via DB-9 (see **Photo 2**). The converter came in a compact and well-padded retail box with nice graphics, and included an external power supply that solves one of my usual complaints by using a twist lock mechanism to hold its jack in place.

The retail price is just under \$2,000 US, and for that you get two powerful DSP engines and two powerful headphone amplifiers, plus the ability to work with PCM sample rates up to 768 kHz at 32 bits, and true native DSD support up to DSD256. Those sample rates are presumably in case we ever reach a future where we all exist as vibrating, massless strings able to detect information in the megahertz range. The bottom line is this unassuming desktop unit will do instrumentation-grade conversion of audio in any imaginable format—today or probably until we start listening to quantum audio directly on our positronic brains.

Initial setup is simple, although I did have to read the manual (which can be found in the Supplementary Material section of the audioXpress website) to see how to reset the entire unit to factory defaults. Fortunately, for the uninitiated like myself, there is an auto-configuration mode, which assumes it should pass the first digital signal it sees to the XLR and headphone outputs.

On my Mac, operation was plug and play, I could immediately get iTunes and high-resolution audio playback. The internal DSP engines can run separate multi-band parametric EQ on the headphones vs. the XLR outs (or different EQ on the two headphone outputs), provide sample rate conversion, adjust stereo width, provide calibrated loudness control (a personal favorite), run balanced or unbalanced headphones, and even display an RTA. At sample rates of 192 kHz or below, they can do all of this at the same time, at higher sample rates some features become limited as DSP horsepower is assigned to support those extreme data rates.

However, I had some small complaints about the user interface. The screen is bright and clear, but I always seemed to be pressing the wrong button to get to the menu option I wanted. Some options are only available in some modes or for certain inputs, so I found myself re-reading the manual to find the menus I needed to navigate, check or adjust a particular setting.

The master volume control has no feel to it, which makes quickly adjusting volume a pain: Too slow and it takes forever to change half a decibel, too fast and suddenly 10 dB have come and gone. The two smaller knobs take on different personalities depending on the mode, but can often be used to adjust the headphone output volume separately from the XLR output volume—with the same complaint. Overall, the ADI-2 Pro looks nice but lacks the gravitas of other high-end DACs. For something that costs as much as a nice laptop and that I might interact with every day, a machined faceplate and volume knobs I can set with my eyes closed do not seem like much to ask for.

Listening

I got excited about the ADI-2 Pro in the first place due to its two high-power headphone outputs, which is how I did the majority of my listening. I have a few headphones but have settled upon the Sennheiser HD700 as a reliable reference. They have a pretty hyped high end but few other flaws. It's nice to be able to do critical listening with headphones: A lot of audio errors are much easier to hear as room effects are more or less eliminated, the full audio bandwidth is available without crossovers, and stereo separation is complete. I can also listen to headphones at moderate volumes for hours without generating noise complaints.

First, a digression: One of the options available in the setup menus is to select the DA oversampling filter. Choices are Sharp, Slow, and Non-Oversampling—the former two are available in IIR “short delay” versions as well. These different



Photo 2: The ADI-2 Pro has XLR and optical S/PDIF inputs and outputs on the back.

reconstruction filters enable you to trade frequency response flatness for an impulse response that looks better to the untrained eye. Why this choice, especially NOS with its zero reconstruction and stair-step response, would be available at all is beyond me. The need for a reconstruction filter to behave a certain way is well understood by DSP engineers; there are a few defensible choices and anything else just enables end-users to add error to their audio. The manual is quite clear on the trade-offs, at least.

Unfortunately, either by default or by accident, I did all my listening using the "Slow" filter, which has substantial high-frequency rolloff. As a result, in comparison to my reference Benchmark DAC2 DX, I found the ADI-2 Pro to be a little dull. Aside from this dullness I found sound quality to be excellent, with wide imaging and very clear details in instrument tones and room reverberation. Fortunately, my mistake was discovered and invalidated in measurement and should not be held against this product.

Aside from my unfortunate use of the wrong reconstruction filter, I found the ADI-2 Pro to be extremely consistent and high quality in every way. The metering is accurate, quick, and useful. Headphone output sounded the same whether "Extreme Power" was selected or not (and it wasn't necessary with my 150 Ω headphones). Driving two headphones seemed to have no effect. Although I could only provide 192 kHz source material, different sample rates were handled gracefully and changes accomplished quickly.

The software brains are especially commendable, they provide a nice volume ramp-up when connecting headphones, remember the last used gain including different gains and EQ for each headphone output, and generally make using the ADI-2 Pro easy. The five-band parametric EQ on each headphone output is a killer feature, it sounds good and behaves as expected, making it much easier to compare or switch between different pairs of headphones—or translate between headphones and studio monitors.

Measurements

Stuart Yaniger, a regular contributor to *audioXpress*, did extensive measurement and listening with the ADI-2 Pro using his ears as well as an Audio Precision APx515. A subset of his impressions and results can be found in the sidebar article that accompanies this review.

Conclusion

It's very hard to find a piece of gear that appeals to studio purists, audiophiles, pro audio engineers,

and measurement experts. The RME ADI-2 Pro may have enough horsepower to make everyone happy. It can drive any headphone at any conceivable volume, convert analog sources to digital with zero compromise, output any digital source to multiple formats at multiple clock rates, and even equalize headphones to match monitors and monitors to correct for room modes. If you want one box to put on your desk or take with you on tour that does it all, I suggest you take a close look at this great converter from RME. 

About the Authors

Bennett Prescott is Sales & Operations Manager for B&C Speakers North America, and co-founder of the popular discussion and news site SoundForums.net. He lives in southern Connecticut, and can be reached at bennettprescott@gmail.com.

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

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RME ADI-2 Pro Measurements

As usual, all measurements were taken using an Audio Precision APx515, with software running under Windows 7. The ADI-2 Pro has so many input, output, equalization, and sampling options that a complete set of measurements would occupy 100 pages or more, so I'm restricting this set to a few of the highlights.

Distortion vs. frequency with an AES (digital) input and the headphone output at 1.5 V into 32 Ω is shown in **Figure 1**. It remains impressively low across the audible spectrum, and does not exhibit the commonly-seen rise at higher frequencies. Distortion versus level at 1 kHz is shown in **Figure 2**, again with a 32 Ω load at the headphone output. It is shown using both the "normal" headphone output and with the "Hi Power" option engaged. The downward slope of the curve with increasing voltage indicates that the results are noise-limited up until the clipping point; actual distortion is minuscule. The Hi Power option increases clipping voltage to 6 V, at the expense of a small amount of extra noise because of the increased gain.

Note that in extended Hi Power testing, the ADI-2 Pro would have a thermal shutdown after 5 minutes or so of continuous output at 5 V into 32 Ω . The protection circuitry is effective—after shutdown, the unit returned to normal operation after a few minutes, unscathed. For the majority of headphones, this is a moot issue; these levels would destroy any headphones that I have on hand.

Figure 3 shows the residual distortion expanded 100,000 times; this certainly looks noise dominated, as implied by the Figure 2 curves.

One unusual feature is selectable reconstruction (anti-imaging) filters for each of the two DACs. Each DAC has five options (Slow, Fast, Short Delay Slow, Short Delay Fast, and NOS). And each of these options yields a different frequency response, impulse response, image rejection, and delay. **Figure 4** shows the frequency response for a 48 kHz sample rate with each of the filter options. The two Sharp options overlay one another, as do the two Slow

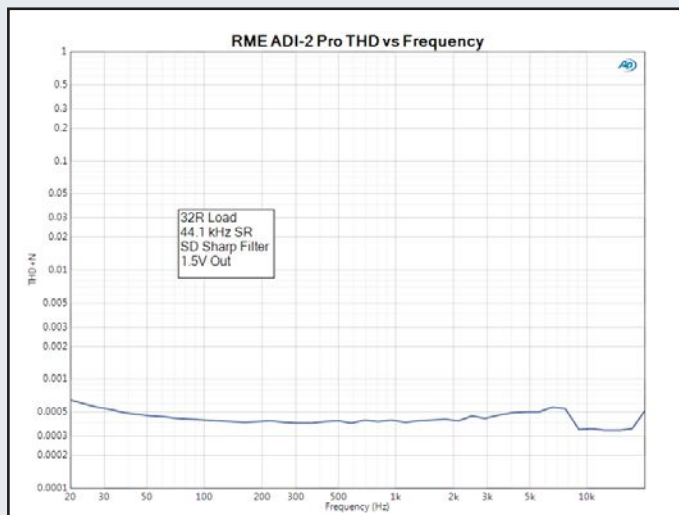


Figure 1: Distortion vs. frequency

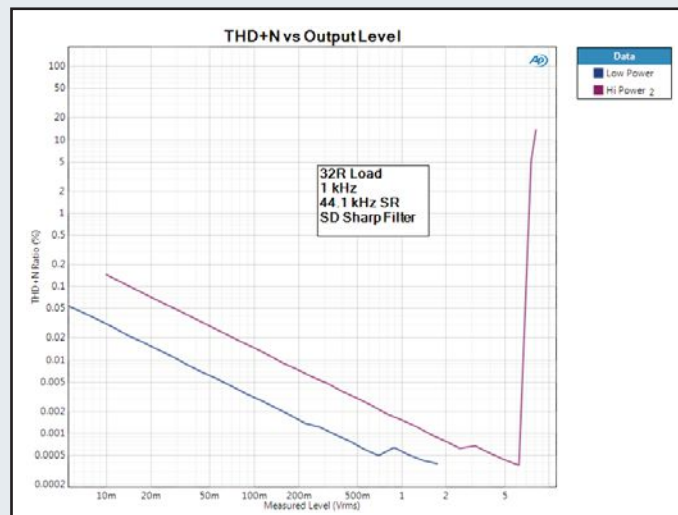


Figure 2: Distortion vs. level at 1 kHz

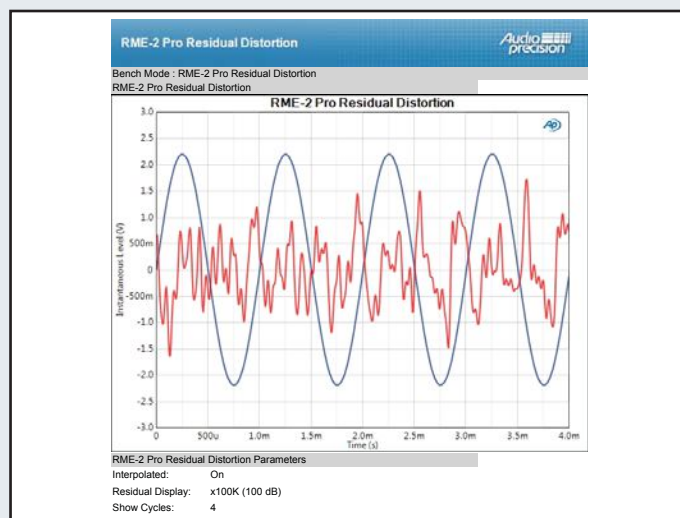


Figure 3: Residual distortion expanded 100,000 times

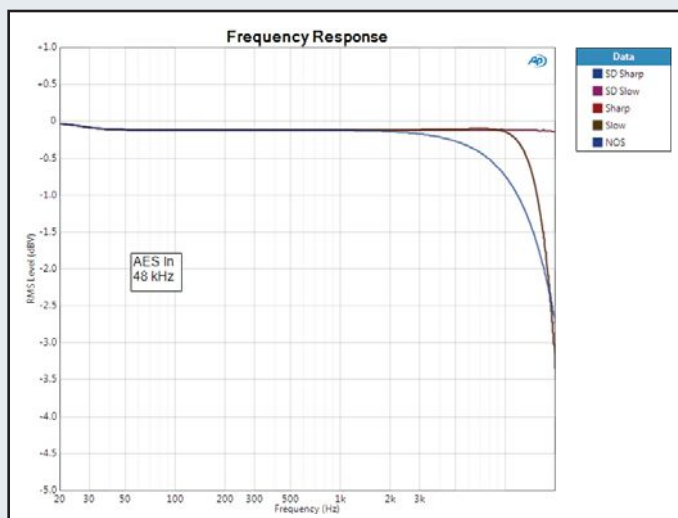


Figure 4: Frequency response for a 48 kHz sample rate

options, with the latter showing significant rolloff in the top octave. The NOS option rolls off even earlier, something that should even be audible to a casual listener.

The amount of treble rolloff with the Slow and NOS is naturally a function of the sampling rate. **Figure 5** compares the frequency response with a Slow filter at 44.1 kHz and 96 kHz sampling rates, the difference being well into the “audible” magnitude.

The choice of filter also affects the suppression of images. The 10 kHz distortion spectra illustrated in **Figure 6** show this clearly. I displaced the fundamental by 500 Hz between the two filter measurements so that the differences would be more visible in the graphs. The image at sample rate minus fundamental (at about 34 kHz) shows 60 dB better for the Sharp filter compared to Slow.

Figure 7 shows impulse response differences for three of the filter options at a 44.1 kHz sample rate. The delay varies from about 0.2 to 0.9 ms, with NOS showing the shortest delay. All of

those numbers are well below audibility, but the Short Delay (SD) option will reduce them even further, with SD Sharp showing a delay of under 350 microsecond.

My favorite signal torture test is a multitone measurement, with equal level tones spaced a half octave apart, covering the audio spectrum. Any misbehavior of the electronics shows as tones or hash between the “teeth” of the “comb.” **Figure 8** shows the multitone spectrum of the ADI-2 Pro through the headphone output loaded with 32 ohms. It’s textbook clean, an outstanding result. I ran this test two ways, using an AES input and using the analog inputs to exercise the AD converters. The results look identical, so the AD half of this unit does not let down the superb performance of the DA and headphone amps.

All in all, the bench test performance of the ADI-2 Pro is flawless. The sound can be tailored using the filter options or equalization if a departure from neutrality is desired. 

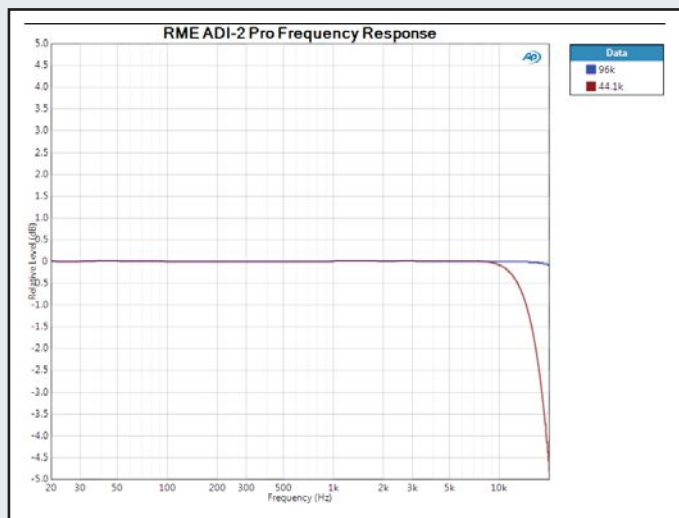


Figure 5: The frequency response with a Slow filter at 44.1 kHz and 96 kHz sampling rates

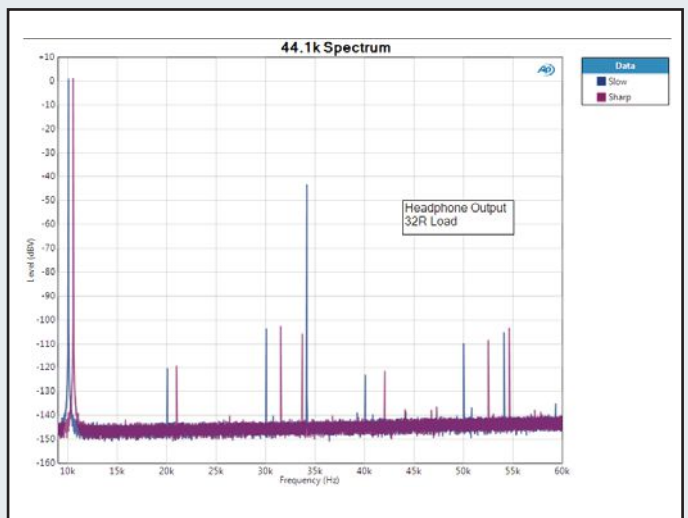


Figure 6: The distortion spectra at 10 kHz show the choice of filter also affects the suppression of images.

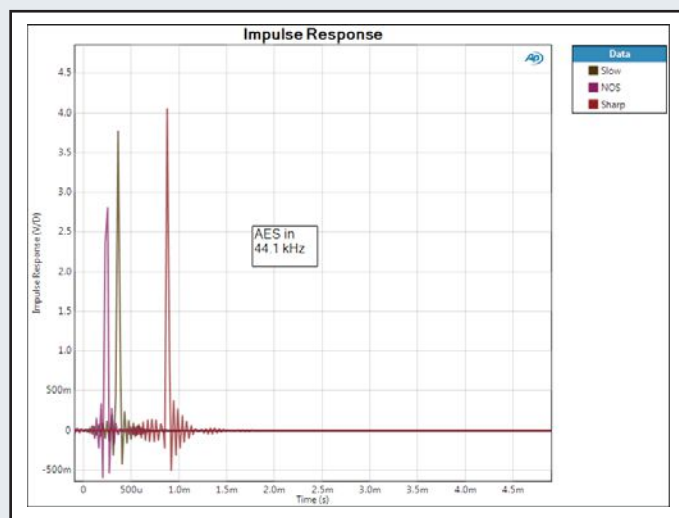


Figure 7: Impulse response differences for three of the filter options at a 44.1 kHz sample rate

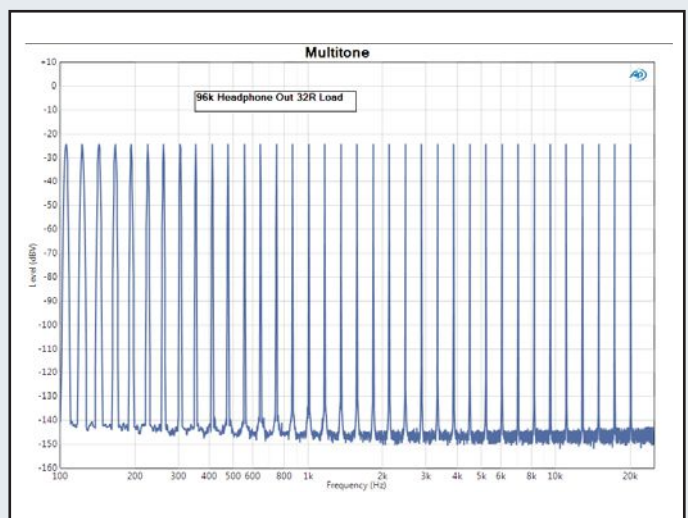


Figure 8: The multitone spectrum of the ADI-2 Pro through the headphone output loaded with 32 Ω



Acoustical and Electroacoustical Measurements on a Tight Budget

Learn how to take useful acoustical measurements without purchasing expensive equipment.

By
Richard Honeycutt

(United States)

Many audio enthusiasts know the importance of acoustics and may wish to measure the acoustical performance of particular spaces. A lot of us made tape recordings in our parents' living rooms; then later, in our own living rooms. We might have been strictly amateurs or we might have taken advantage of the occasional opportunity to make a few bucks recording semi-professionally. During those experiences, some of us became aware of acoustical problems with school auditoria or other venues in which we performed or attended performances.

Taking Acoustical Measurements

Articles or books on acoustics may stimulate amateurs or semi-pros to make acoustical measurements. Depending upon our areas of special interest, these measurements might include room mode effects, reverberation time (RT), background noise level, or speech intelligibility. Established professional acousticians make a place in their budgets to buy the appropriate instruments for making these measurements. Amateurs might find the cost of such equipment prohibitive. There are ways, though, to make useful acoustical measurements without expensive equipment.

The most obvious approach is to find good, low-cost instruments. The Sound Control articles in *audioXpress* issues from September 2014 through August 2015 discussed a range of acoustical

measuring instruments. These were all professional instruments. If all you need is a sound level meter (SLM), several fairly inexpensive ones can be found on the Internet. Occasionally you can also find a good quality used instrument for sale.

If you can't find an affordable source for the test instrument(s) you need, but you have a smartphone, a mind-boggling number of free or inexpensive apps are available. (Accuracy of smartphone-based acoustical measurement systems varies from excellent to wretched, so caveat emptor!) A representative survey of some of these options appeared in *audioXpress*' November and December 2015 Sound Control articles. Because of the difference in operating systems and hardware, Apple mobile products are more suitable for acoustical testing than Android ones.

For some of us, an iPhone is not a trivial investment and might be a cost-prohibitive solution to obtaining acoustical test equipment. Although as part of my consulting practice, I own and use a number of professional acoustical test instruments, I do occasionally need to find a very-low-cost testing solution. For example, I have a client in New Zealand for whom I did a significant project some years ago. At that time, mobile test apps were not an option. The project budget did not include travel expenses for me and my test equipment, so I had to devise ways for the client (who was technically savvy) to take the measurements I

needed. Any technically savvy person can use the same approach.

Signal Sources

First, you need a signal source. It must produce fairly low-distortion sine waves adjustable to the desired audio frequency. While 20 Hz to 20 kHz is the commonly accepted range, RT tests can be done using a more limited range. The ability to sweep the test frequency can be convenient. Pink noise, preferably in 1/3-octave bands, is extremely useful. If speech intelligibility is to be measured in Speech Transmission Index (STI), a special test signal is required.

One convenient way to provide test signals is to use a test CD. Tejera Microsystems used to sell a CD with signals including sine waves at 1/3-octave intervals from 20 Hz to 20 kHz, 1- and 5-kHz square waves, octave-wide sine sweeps from 31.5 Hz to 16 kHz, and broadband pink noise, as well as other tests less useful for acoustical measurements. This product no longer appears on its website, but may be available used.

The Computer Aided Room Acoustics (CARA) test CD (www.cara.de) includes:

- 28 pure sine-wave tones from 16 to 200 Hz (distortion < -80 dB)
- 27 narrowband pink noise signals (bandwidth about 1/18 octave) within the range of 20 to 200 Hz. The center frequencies equal those for the pure sine-wave tones mentioned above (except 16 Hz)
- Four special bandwidth-limited pink noise signals: two "CARA-Noise" (to analyze midrange sound colorations), "Polarity-Noise," and a standard pink noise signal flat between 20 and 20,000 Hz.

The Dayton test CD (www.parts-express.com/dayton-audio-omcd-version-4-test-track-cd-for-omnimic-v2--390-791#lbiProductDetails) contains:

- Monophonic pseudo-noise sequence, short sine sweep, long sine sweep, and bass sweep
- Stereo pseudo-noise sequence
- Left and right channel short sine sweep, short sine sweep with bass removed, short sine sweep with bass and midrange removed, long sine sweep, bass sweep, and pseudo-noise sequence

As discussed in the June 2017 Sound Control article, "Assessing Audio Quality Using the Matt Test," the Musical Articulation Test Tones (MATT) test is available in either CD or downloadable form from www.acousticsscience.com/matt. This test

signal is uniquely useful for room-mode testing.

An Internet search will reveal other test CDs, but those already mentioned comprise the ones with which I have had some experience. But you can see the range of signals available from these inexpensive CDs. Even a small portable CD player can be connected to a sound system to make the sounds desired for testing.

If the room does not have a sound system, you'll need one capable of producing reasonably low-distortion sound throughout the frequency range of interest. For room mode tests, you need the response to be pretty flat down to as low a frequency as possible, which can be a challenge for a system that you can carry around. For RT and speech intelligibility tests, 100 Hz to 5 or 6 kHz is enough. While such systems can be purchased for as little as \$200, the least expensive ones may not be able to produce signals that are at least 30 to 40 dB above the room noise, as required for RT tests in the lower octave bands.

Measurement Microphones

Measuring what the room does to the sound will require a measurement microphone. *audioXpress's* November and December 2014 Sound Control articles discuss choices of measurement microphones. However, RT, speech intelligibility, and background noise level testing does not really require a professional measuring microphone: A good-quality recording or PA mic will suffice. It is better if you know the microphone's frequency response, but even that is not absolutely essential. (As a side note, advertising claims of a frequency response "from X to Y Hz", with no plus-or-minus decibel tolerances, are useless.) Published response curves that look really smooth (see **Figure 1**) are not much better, as they were probably produced by a marketing department rather than being

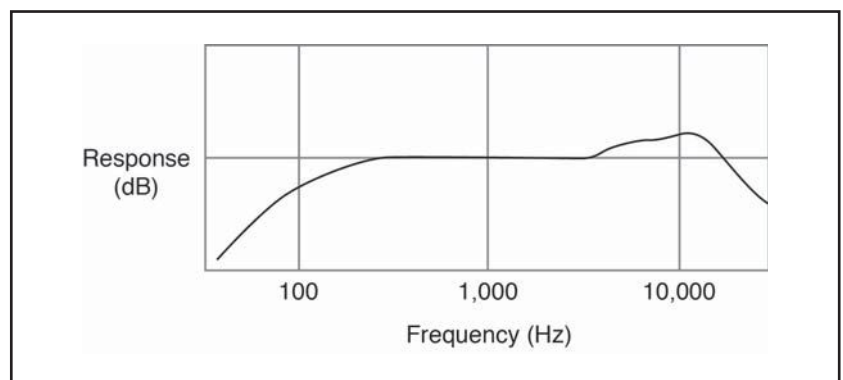
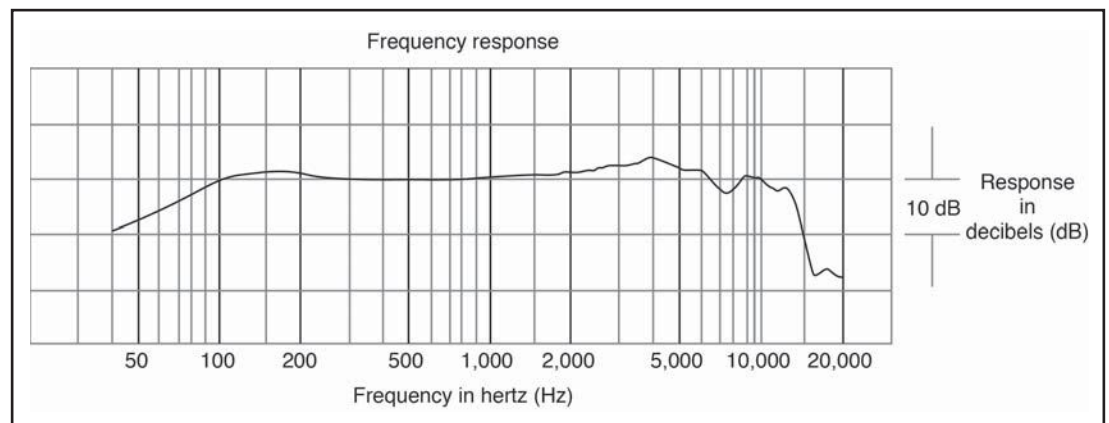


Figure 1: We can't trust microphone frequency response curves that are too smooth!

Figure 2: Though probably not precisely accurate, this curve for a low-cost microphone does contain some useful information.



measured in a lab. An exception is very high-quality recording or test microphones. Most low-cost microphones have a response similar to the one shown in **Figure 2**. Although this response has probably been massaged by the manufacturer, it does warn us to be careful about trusting the accuracy of measurements below 100 Hz, above 10 kHz, or in the 4-kHz octave band.

Finally, we have to display the results of our measurements. If we have access to a laptop, notebook, or tablet computer, we can use recording software, if we have it. We can download free recording software such as Audacity (www.audacityteam.org). You can record the test signal from an appropriate location in the room, and then analyze it using the software.

Room Mode Testing

For room mode analysis, I recommend using the process described for the MATT test mentioned earlier in the article. It enables you to aurally determine the severity of modes from the sound made by the MATT signal being played in your room.

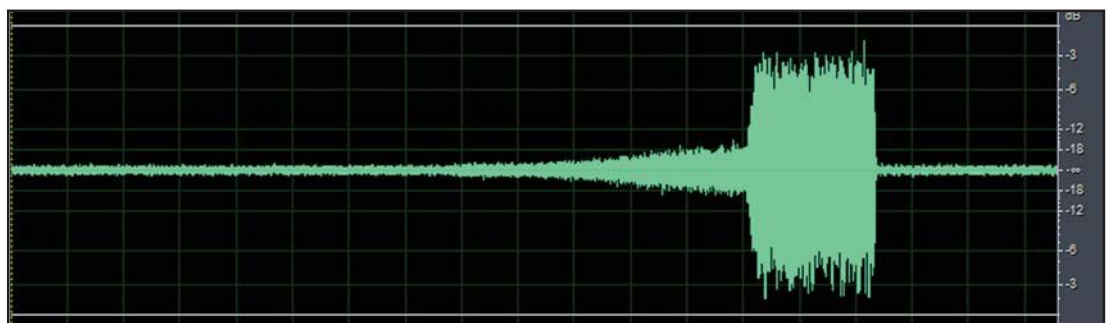
RT Analysis

First, start the recording software and notice the average level of the background noise. Then set the test signal level in the room about 30-40 dB higher

than the background noise. Cut off the pink noise, wait a few seconds, and then stop the recording. You should get a display similar to the one shown in **Figure 3**. After you zoom in to expand the time axis, you can identify the peak value just before the test signal was cut off, and the time at which the amplitude has decreased by 5 dB. This is the start of the portion of the decay we will measure. In this case, we were only able to get the pink noise signal about 30 dB above the background noise, so we'll need to find the time at which the level has dropped by 20 dB from the 5 dB down time. The results of these operations are shown in **Figure 4**.

Now, a reverberantly decaying sound decreases at a constant number of decibels per second. Thus, if the amplitude were logarithmically scaled, the decay slope (represented by a blue line in Figure 4) would be a straight line. But even though Adobe Audition can show the time calibrations logarithmically, the actual amplitude values are plotted linearly. Therefore, the decay slope appears on this plot as a curved line. The time required for the sound to decay by 20 dB is 30.720-30.647 = 0.073 seconds. The 20-dB decay represents 1/3 of the 60-dB decay by which RT is defined, so the measured RT of this room is $3 \times 0.073 = 0.219$ seconds. Technically, what we have done is called a T20 measurement. You could also do a

Figure 3: This is a full trace of the RT measurement recording.



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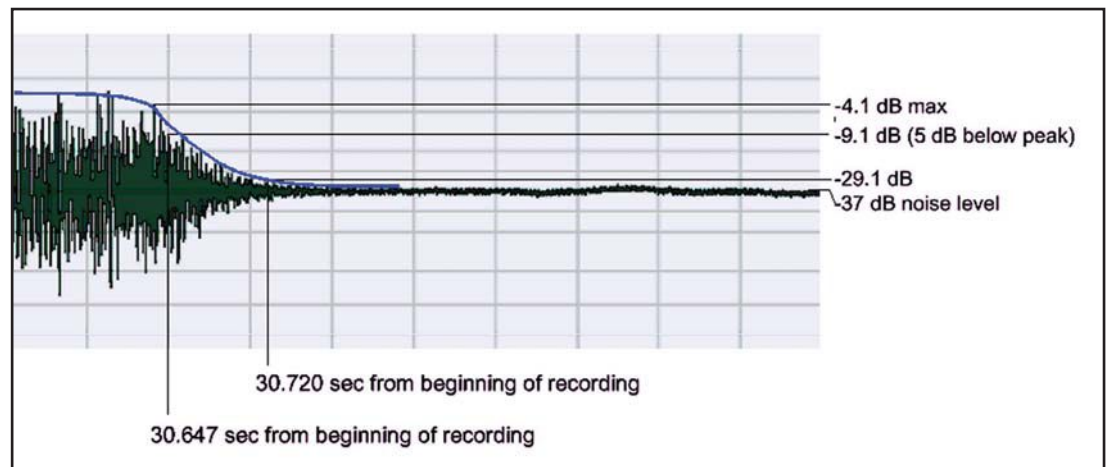
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Figure 4: This expanded view of the end of the trace shows the decay, from which the RT is calculated.



T10 measurement—the decay between -5 dB and -10 dB multiplied by six—or a T30 measurement—time between -5 dB and -35 dB multiplied by two. Actually, this is a rather contrived example, since the RT of a room this dead is not well-defined: The reverberant sound in the room probably never becomes statistically diffuse, and so the room does not meet Sabine’s assumptions. Nevertheless, the process serves well as an example of measuring RT without specialized equipment.

For finding RT within an octave or other

frequency band, Sabine used organ pipes which he sounded by blowing into them. Of course, even if he used pipes pitched 1/3 octave apart, the pipes produced a single tone plus negligible harmonics, rather than a continuous range of frequencies such as bandlimited pink noise would produce. We can do something similar by using sine waves at the frequencies for which we want to know the RT, if we do not have a way to create bandlimited pink noise. If we use sine waves to test below a few hundred hertz, it is a good idea to vary the frequency up and down while listening to the speakers to make sure our test frequency is not on a modal peak (loud tone) or null (weak tone). The RT at peak or null frequencies is not characteristic of the entire 1/3-octave band or octave band.

Resources

Archive of Anechoic Chamber Recordings,
http://ece.nmsu.edu/~pdeleon/ASAP/BSS/Anechoic_Chamber.html.

Audacity, www.audacityteam.org.

Musical Articulation Test Tones (MATT), Acoustic Sciences Corp.,
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R. Honeycutt, “Assessing Audio Quality Using the Matt Test,” *audioXpress*, June 2017.

——, “Audio Tools from Studio Six Digital,” *audioXpress*, December 2015.

——, “Smartphone Apps for Sound Measurement,” *audioXpress*, November 2015

——, “Measurement Microphone Suppliers,” *audioXpress*, December 2014.

——, “Measurement Microphones,” *audioXpress*, November 2014.

Sources

Adobe Audition

Adobe | www.adobe.com

Computer Aided Room Acoustics (CARA)

ETS Services GmbH & Co. KG | www.cara.de

Dayton Audio OMCD Version 4 Test Track CD for OmniMic V2

Parts Express | www.parts-express.com/dayton-audio-omcd-version-4-test-track-cd-for-omnimic-v2--390-791#lProductDetails

Audio Tools for iPhone

Studio Six Digital | <http://studiosixdigital.com>

Background Noise

For approximately measuring background noise without a calibrated SLM or its equivalent, you can adjust a pink noise source so that when you hum at a normal level, the pink noise is barely loud enough so that you can’t hear yourself inside your head. If you are recording the mic output, the recorded level is about 80 dB, provided that you have normal hearing. You can then calculate what SPL is represented by any other level you record with that microphone using the same settings on your computer’s audio adapter “mixer.” Usually, background noise level is highly variable, but you can “eyeball-average” the level on your recording, as illustrated in **Figure 5**. (Remember that the amplitude scale is linear, so dB SPL is logarithmically related to vertical distance on the waveform plot.) The average level is about the center of the range of amplitudes that the peaks most frequently reach.

Background noise is also position-dependent, so you’ll want to be careful how close you are to

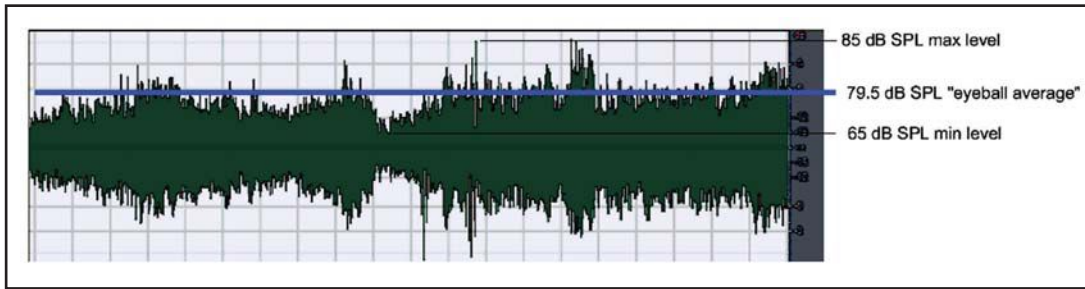


Figure 5: You can "eyeball-average" the instantaneous levels from the recording to get the approximate average SPL.

noise sources such as the bar in a restaurant to avoid measuring in a location having an atypical background noise level.

Speech Intelligibility

Properly measuring speech intelligibility pretty much requires a computerized STI tester. Probably the least expensive good instrument for measuring STI is the system in Audio Tools for iPhone by Studio Six Digital. It costs a little less than \$600, including the Audio Tools app and the in-app STI measurement upgrade, but not including their microphone.

However, there is a way to roughly gauge intelligibility without such tools. You can play a good anechoic speech track (e.g., one that can be downloaded from http://ece.nmsu.edu/~pdeleon/ASAP/BSS/Anechoic_Chamber.html) and listen from what you would expect to be the worst listening position in the room. Remember, factors that hurt intelligibility include distance from

the source (i.e., direct-to-reverberant sound ratio), background noise, and the presence or absence of discrete echoes or comb filtering, so conduct this test with the HVAC system running. If you (or another person with normal hearing) can understand what is being said on the track, the STI is probably at least 0.6, which is the bottom of the "good" range.

While no substitute for excellent acoustical measuring equipment, these inexpensive techniques can be of great help in evaluating the acoustics of a room. The ability to separate measurement from analysis is helpful. In the case of my New Zealand client, he made on-site recordings and e-mailed them to me. I did the analysis in my office. 📧

Editor's Note: All audioXpress articles from 2001 to present can be found on the aX Cache, a USB drive available from www.cc-webshop.com.

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Fresh From the Bench

SB Acoustics Rinjani and Ara Speaker Kits

A Gift for Fun

By
Oliver A. Masciarotte
(United States)

Oliver Masciarotte shares his experiences with the surprising SB Acoustics Rinjani and Ara speaker kits, which feature high-end Satori drivers. Masciarotte compares the floorstand Rinjani speakers' performance with the smaller Ara two-way bookshelf version. Both versions share the same design concept and high quality finish, typical of high-end speakers costing multiple times the price.

Unlike many reviewers, I usually only request samples from a company when I'm already aware of a product and admire the manufacturer's work. This review was an exception. The editor of *audioXpress* and I were discussing kit speakers and he suggested I listen to SB Acoustics' new products. The company had e-mailed him to confirm that both the Rinjani and Ara kits were ready for review, and wanted to send both products, as "...they are a close family." They would even be delivered assembled so, no fuss or muss on my part. My only concern was: What if they were not good?

So, one day in late winter a big FedEx truck threaded its way along the back alley that accesses my garage, and a pallet was unloaded with not quite 200 lbs. of speakers and packing. Since a stand is not available from SB Acoustics, I unpacked and set up the Ara pair on my well-used, 21" Sanus Foundations Steel stands. Since they were already broken in, I didn't perform the usual flip-polarity-and-cook ritual. I simply started to listen to them.

A Bit of Background Information

Unless you've been using your back issues of

audioXpress as shims rather than reading material, you might remember the July 2016 issue with a feature on touring the Sinar Baja Electric factory, which is a high volume source for SB Acoustics' components. One particularly salient feature I found in my review units was, as the article notes, the exceptional quality of the drivers. "...for those accustomed to lower standards from Asian suppliers, it's important to reinforce that Sinar Baja's mission statement includes verified ISO/TS quality standards, supported by RoHS standards." The Satori line of drivers possess a linearity seldom heard by any premium transducer family. It may not have anything to do with what's in Surabaya's water, but the design DNA of the Danish engineers certainly contributes to the calibre.

Initial Impressions

On first exposure, the single aspect of both speakers that was obvious to me was their decidedly neutral voicing. I know this is oft repeated, but I did hear elements of my test recordings that I had not previously noticed. Considering how many times I've listened to these files, I found it a bit disconcerting

and quite fun. If I was forced to find any fault, I'd point you to a subjective correction I dialed in for most of my listening. I should explain that this is not something I'd even bring up unless obliged to.

The other aspect of both models that I found most pleasing and atypical, was their distortion or lack thereof. These are sweet systems, not in the sense that they flatter, round off, or velvetize the music. There simply was no sense of harmonic grit that sometimes passes for detail. They also display none of the peaky or resonant aggressiveness that plagues many far more expensive loudspeakers. If you're looking for a close shave, these SB Acoustics kits are not the place to go. What they do deliver is oodles of plausible detail and sound stage, timbrally and temporally. When there was sparkle in the content, it glinted realistically in my living room. If actual reverb was captured in the recording, I could sense the walls in my sweet spot.

Speaking of which, both the Ara and the Rinjani were a bit more difficult to place than I had expected. Rather than the more common 60° equilateral angle between the speaker and the listening position as per ITU-775, in my room I preferred a quite wide 80° placement relative to my head, with the speaker's top edge 24" and the bottom edge 28.5" from the rear wall. Both models have a fair amount of backward lean, the better to lower group delay via time-coincident arrival from each driver. I tried toe-in but found, as recommended by the designer, that only a few degrees from perpendicular to the wall provided the most satisfying imaging. The speakers are not designed for typical toe-in, where the front

baffle is perpendicular to the listener. A maximum of 15° off square is suggested.

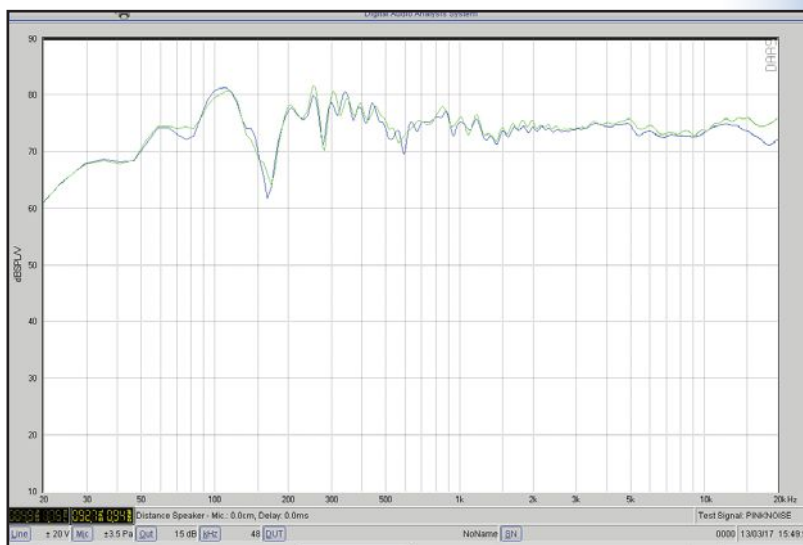
In the golden listening spot, I settled in so my ears were about an inch lower than the tweeter's center. Warkwyn's measurements (see the Measurements text) revealed a rather narrow beaming around 10 kHz, but I didn't notice anything untoward. In a similar vein, I found that the soundstage had a slightly cloudy spot between phantom center and the left speaker, along with a corresponding vagueness between phantom center and the right speaker. When I mentioned this at AXPONA 2017 to Bo Albrechtsen, Sales and Marketing Director at SB Acoustics, he told me he hadn't heard of that deficiency from anyone



Rinjani's sophisticated sound and look makes the entire family happy.



The SB Acoustics Rinjani, Ara, and the new Micro speaker kits were on display at High End Munich 2017.



Rinjani's measured frequency response as supplied by Danesian Audio.

else. Warkwyn's measurements showed wide, even dispersion in the horizontal plane, so I chalked this observation up to room interaction, though I haven't heard this from other speakers in my room.

While the Rinjani can't create a plausible version of the opening notes from Richard Strauss' "Also sprach Zarathustra," it can dig about as deep as I need. Most folks, and I am one of them, either don't have the room, space, budget, spouse, or neighbors

that will sanction speakers that are flat to 20 Hz. I also am quite comfortable with my self-image and don't need to prove anything, so the slim 40" tall Rinjanis fit perfectly into my world view. At Warkwyn, I was surprised to see that Rinjani's low-frequency response was not as good as I'd subjectively expected, with it starting to roll at about 50 Hz. Still, I found the low end to be fine. Note that the EQ I used did not include any low-order low shelving, which I typically use when I feel low end is lacking. With a 3 dB upper limit of 25 kHz, the tweeter is capable of nicely controlled behavior in the audio passband while delivering, thank you Anthony Burgess, that touch of the old ultrasonic that I enjoy.

Supporting Gear

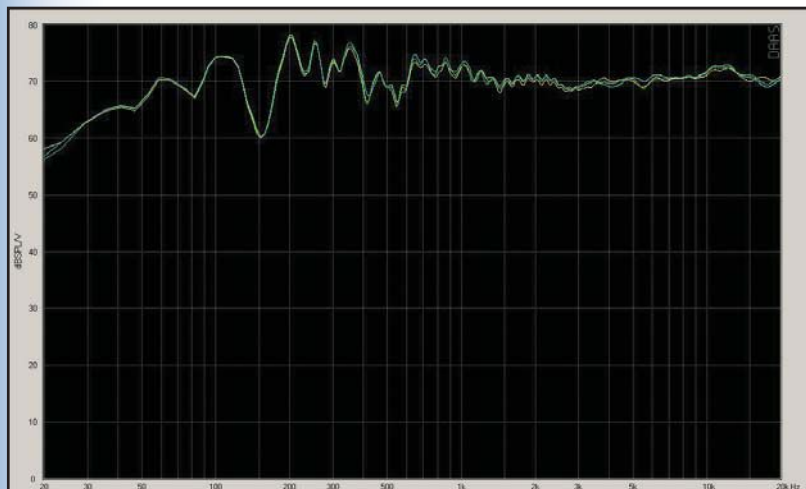
For those who keep track, the supporting gear for this review is modest, in keeping with the cost of the speakers themselves. A Mac mini running Audirvana Plus v3 and Amarra v4 players fed my exaSound e22 Mark II DAC via a \$200 top-of-the-line VUE USB cable. Out from the DAC, balanced GEN II Beta 2-22S cabling from Soundstring fed a AMPX Class A stereo amp from Lindell. Finally, AntiCable Level 3 speaker wires animated the elegant white wonders. During their time at my home, I also powered the Rinjani with a 2 x 50 W into 4 Ω Anaview AMS0100-based custom amplifier, but preferred the top end and midrange of the AMPX.

Madisound Speaker Components is the sole distributor of the SB Acoustics kits in the US. Adam Johnson at Madisound told me that the company is going to stock the following kits:

- Rinjani—\$2,500/pair for the kit, including white, high-gloss cabinets, Satori drivers, and assembled crossovers
- Tweeter crossover capacitor upgrades, from ClarityCap and Mundorf will be available.
- Ara—\$1,370/pair for the kit with white, high-gloss cabinets, Satori drivers, and assembled crossovers
- Tweeter crossover cap upgrades will also be available.

While at AXPONA 2017, I got to hear the Micro, a tiny stand-mounted two-way that also sounded great. In white satin, the Micro runs \$450/pair for a kit with cabinets, SB Acoustics drivers, and assembled crossovers.

The Ara and Rinjani occupy a special place in the audiophile ecosystem, being uncommon quality and affordably priced kits. I can think of a handful of factory-built speakers that would sit comfortably alongside either model—Andrew Jones' Uni-Fi line



Ara's measured frequency response as supplied by Danesian Audio.



This is a bit of high-resolution EQ I dialed in to taste.



The solid quantity terminals are in keeping with the rest of the hardware.

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Top rated and reviewed:

Headphonics 12/2016: "This is a game changer, even for pure music listening"

Studio Magazin 12/2016: "Fantastic work!" **Sound on Sound Magazine 04/2017:** "Cons? None!"

Sonarworks 03/2017: "...the ADI-2 Pro impressed me enough to become the reference unit here at Sonarworks."

Mixdown 03/2017: "RME have pulled out a winner on this unit (...) Hearing is believing."

Dale Pro Audio 03/2017: "The sound – wow. Pristine."



Take just a tweeter and mid-woofer, and you'd have an Ara.



The slim but deep mini-tower is dual ported

Resources

Madisound Speaker Components, www.madisoundspeakerstore.com.

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Satori MW16P-4 Midwoofers (Ara)

SB Acoustics | www.sbacoustics.com/index.php/products/midwoofers/satori/6-satori-mw16p-4

Satori MW16P-8 Midwoofers (Rinjani)

SB Acoustics | www.sbacoustics.com/index.php/products/midwoofers/satori/6-satori-mw16p-8

Rinjani Kit

SB Acoustics | www.sbacoustics.com/index.php/sb-acoustics-kits/sba-satori-rinjani

Ara Kit

SB Acoustics | www.sbacoustics.com/index.php/sb-acoustics-kits/ara-kits

or KEF's LS50. Yes, those are both British designs, and the SB Acoustics offerings can hold their own in terms of tonal neutrality and uniformity across the audio spectrum relative to those competitors. In addition, Rinjani and Ara exhibit much lower distortion than either, while delivering truthful timbre and exceptional good looks.

As with all things audio, your taste is not mine, but I found both speakers, ahem, spoke to me with a compelling and authoritative voice. I haven't had so much fun since a pair of \$5,000 Gauder Akustik ARCONA 40s were in the house. If you need to have reasonable bass without excess clutter, then Rinjani is for you. If you have a difficult room or wish to optimize the low end, I'd highly recommend the Ara, coupled with a mono or, better yet, stereo set of subwoofers. Either way, if you go for truth more than euphonious color, and enjoy a bit of tinkering and DIY, then either kit is highly recommended. 

About the Author

Oliver A. Masciarotte has spent more 30 years immersed in the tech space, working on facilitation, optimization, marketing, and product development for clients worldwide. As an author and speaker, he enjoys informing folks about technological best practices. More information is available at seneschal.net and othermunday.com.

Tips from Danesian

All the Sinar Baja transducers used in both Ara and Rinjani are designed by former Danish Vifa/Scan-Speak experts and current Danesian Audio ApS co-owners Frank Nielsen and Ulrik Schmidt. I asked Schmidt for specifics on the products. Here are excerpts from our conversation.

OLIVER MASICAROTTE: Are these kits time coherent? If so, how? Baffle angle, crossover, or both?

ULRIK SCHMIDT: The baffle on both speakers is inclined about 7° to 8° to fully compensate for the difference in time delay between the (upper) midwoofer and the tweeter—but it is also a design feature. We provide this data for some combinations of drivers on our website (Time Alignment: www.sbacoustics.com/index.php/technical-notes). It is not strictly necessary to do this, but in this case it made it a lot easier to design the crossover network.

MASICAROTTE: Tell me about the individual drivers, and why did you specify those particular units?

SCHMIDT: For these speakers, we chose our top-of-the-line Satori drivers (that we had) at the time. We didn't have the beryllium tweeter back then and though we did have the neodymium version of the Satori tweeter, we didn't use it here, as we didn't need the higher sensitivity. Also, these are speaker kits that are supposed to be fairly affordable. [For a list of features for the drivers, see Sources.] Both the Rinjani and the Ara use the Satori TW29R tweeter. The Ara uses the Satori MW16P-4 (4 Ω) midwoofer and the Rinjani uses two MW16P-8 (8 Ω) midwoofers.



The measurement setup that the designers, Danesian Audio, used for their measurements. Note the asymmetrical shape of the speaker, which are delivered as mirrored pairs.

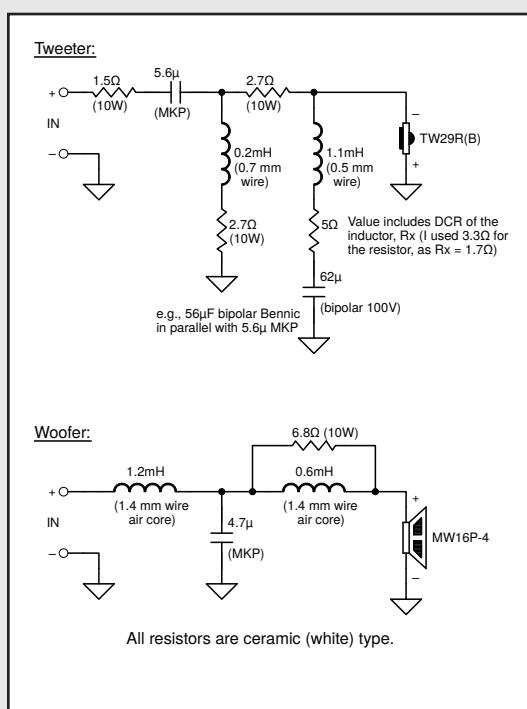
MASICAROTTE: Tell us about the crossovers—topology, materials, and why you chose those?

SCHMIDT: I could write a book about this, but I am not going to. I am not religious when it comes to crossovers. The objective is to make it all work—to make the drivers integrate and work together as a whole. Given the inclined baffle and the very flat impedance, at higher frequencies, of the midwoofers, we are looking at second order filters with some modifications, with the tweeter connected in reverse phase.

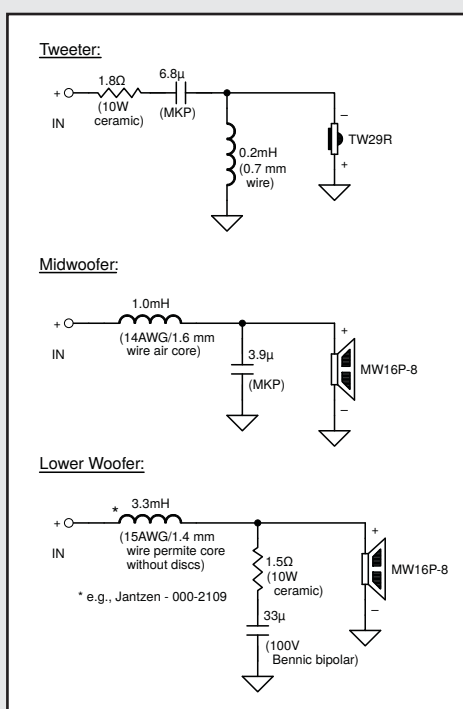
The Ara uses a second-order network with a shelf filter for baffle-step compensation on the midwoofer. This ensures a very neutral tonal balance. The high-pass filter on the tweeter might look a little bit complicated at first, but really it isn't. Basically,

it's a second-order filter with some attenuation. To achieve perfect integration with the midwoofer in the entire crossover region, it was necessary to use a resistor in series with the inductor. This is a really neat trick that often makes the whole difference, (as it) changes the phase and roll-off. This resistor makes it more of a first-order filter at lower frequencies and therefore it is necessary to equalize the impedance peak at the tweeter's resonance frequency, hence the LCR-network in parallel with the tweeter.

The Rinjani is a little bit more straightforward. As a 2.5-way speaker, it doesn't need the shelf filter. The lower woofer adds



Crossover schematic for Ara



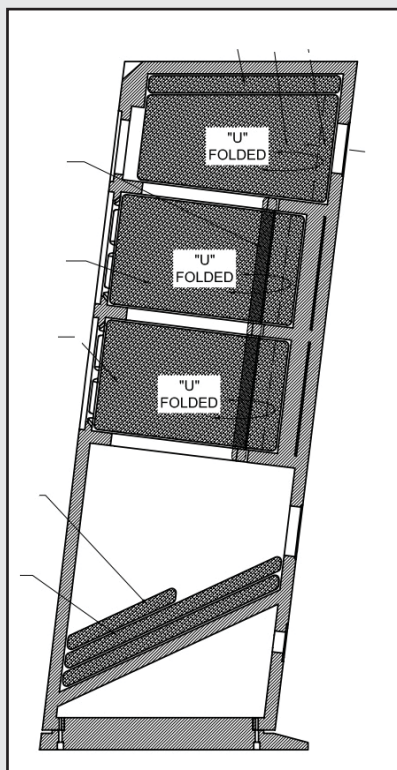
Crossover schematic for Rinjani

the extra/missing energy below the baffle frequency, where it is crossed over. Furthermore, the relatively flat upper response of these midwoofers makes them easy to work with. Second-order filters on all drivers, though a low value resistor is used in series with the capacitor on the lower woofer to make it blend in perfectly with the upper midwoofer near the crossover frequency. So, this is not your typical RC network used to equalize the inductive part of the impedance curve on a midwoofer—inductance is already very low, thanks to the copper sleeve on the pole piece.

There are no exotic parts or components. We use good quality, standard components that we like: Jantzen Audio Cross-Cap capacitors, wire wound inductors with sufficiently low resistance in the right places, and ceramic resistors—my preferred type, actually. One inductor for the Rinjani network uses a core, which is a very good quality Permite core as opposed to a laminated iron core. People who build these speakers don't have to buy our cross-over networks, though fully mounted PCBs are available, of course. (The crossover is not bundled with the cabinet, it's purchased separately, as are the drivers.) Anyone is

welcome to upgrade to more expensive/exotic components.

MASICAROTTE: How about the cabinets? Flexy or stiff, materials, shape, port details, internal details, etc., and why?



Damping material for Rinjani



The cavity in which Rinjani's crossover resides



Looking into the lower portion of the Rinjani prototype cabinet. This photo also reveals internal bracing and the inner wedge-shaped rear wall, behind the midwoofers only, that significantly reduces the reflected sound wave.

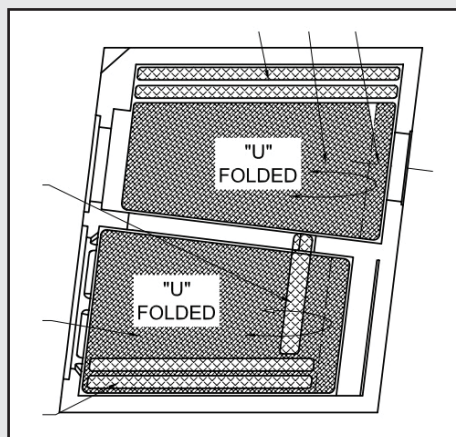
Do the mids and/or woofers have their own enclosures/spaces? If there is damping, what do you use and why?

SCHMIDT: The cabinets are very much alike, as the Ara cabinet is quite simply the top of a Rinjani with a bottom plate. The way I made the Ara prototype cabinet was to cut off the top section of a Rinjani prototype cabinet with a plunge saw and mount a new bottom plate.

The cabinets are made with 19 mm (0.75") MDF. The facets in the top area reduce edge diffraction at higher frequencies, though this is also a design feature to give the speakers a unique appearance. The Ara uses one 50 mm diameter port, (with) both ends flared, and the Rinjani uses two of these ports (with) only the outer end flared. In the Rinjani, the two midwoofers share a common enclosure. I have always preferred to do it this way with 2.5-way speakers, but you could certainly use separate cavities as well. Potentially, a common enclosure can cause problems with vertical standing waves, depending on how the drivers and port(s) are positioned in this internal "transmission line." The internal slanted bottom plate, in combination with the damping material, effectively deals with that. This is absolutely nothing to worry about in the Rinjani.

We use a synthetic porous damping material, Dacron, with a sheet thickness of 25 mm (1") and an area density of 300 g/m² (1 oz./ft²). Notice, there is an additional layer on the back wall in the midwoofer sections.

We use Dacron because it works well and it is available everywhere. Where to put the damping and how much is determined by experiments and listening tests, (plus) a lot comes with experience. We spend a lot of time listening, fine tuning the crossover, ports and damping.



Damping material for Ara

The Measurements

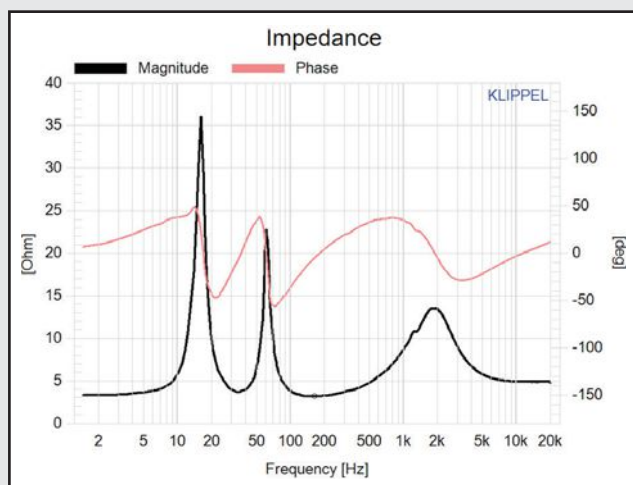
The SB Acoustics V1r2 “Rinjani” was measured at Warkwyn’s facility using the Klippel Near Field Scanning (NFS) system. The NFS is an autonomous speaker measurement system that utilizes robotics to move the microphone around the device under test (DUT), thus leaving the DUT in a stationary and stable condition while measurement occurs. In this way, the loudspeaker has consistent interaction with the room. In addition, stationary loudspeakers are easier to support with minimal gear—a big advantage for anyone measuring large heavier loudspeakers—the Rinjani a notable inclusion.

Since the speaker is stationary its interactions with the room are consistent and valid measurements are obtainable in non-anechoic conditions with the NFS. By scanning around the DUT in two cylindrical layers of measurement points the system is able to remove the effects of the room using a combination of acoustical holography and field separation. An acoustic model is then created and SPL measurements can be generated from any point in 3D space. For more information, visit www.klippel.de/our-products/rd-system/modules/nfs-near-field-scanner.html.

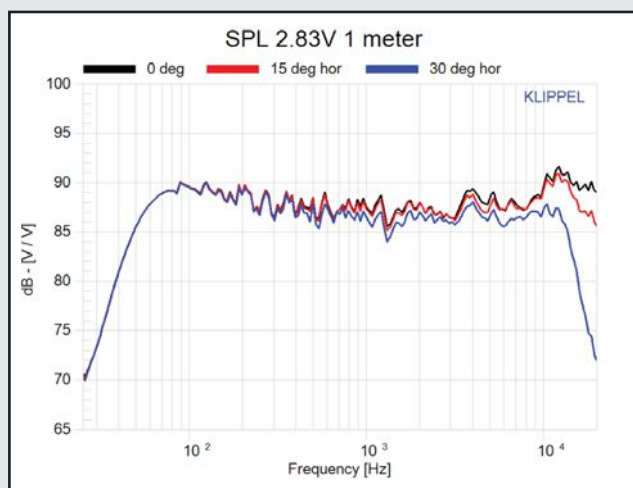
To measure the Rinjani, we used the typical 2.83 V/1 m with



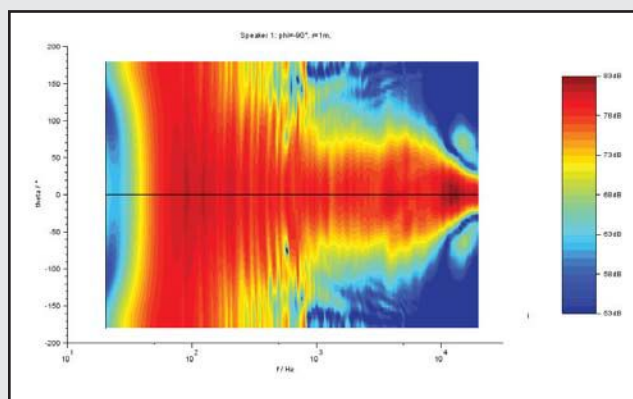
The Rinjani is pictured on Warkwyn’s Near Field Scanner.



The Rinjani’s minimum impedance resides at 155 Hz and at 3.14 Ω , which is consistent with the specification of 4 Ω .



The Rinjani’s horizontal sensitivity was taken on axis, at 15°, and 30°.



An unwrapped 360° pressure/frequency contour map provides a complete view of the horizontal radiation of the speaker and through its rated frequency response.

a calibrated ACO Pacific (7052E capsule) free-field microphone with the 4048 preamp. All the data shown is referenced to on-axis with the tweeter.

Measurements points around the speaker totaled 2,597 and were processed with a resolution of 0.73 Hz from 20 Hz to 20 kHz. The length of the stimulus was 1.4 s.

Unfortunately Warkwyn had little time to perform subjective listening tests, however, measurement data provided some notable results.

A look at the Rinjani's impedance shows a relatively well behaved crossover, staying within $\pm 50^\circ$. The minimum impedance resides at 155 Hz and at 3.14 Ω , which is consistent with the specification of 4 Ω .

On- and off-axis measurements show that the speaker has relatively good low-frequency response with a rolloff below 60 Hz. This response was a bit surprising and a greater slope than anticipated. This can be attributed to the measurement environment being 4π and the listening position being 2π . It is certainly Warkwyn's impression that the low frequency response in a 2π condition would indeed emphasize the low-frequency enough to call it noticeable and not necessarily lacking.

Though relatively smooth from 100 Hz to 1 kHz, there is a noticeable dip at 1.3 kHz, a presence rise at 1.8 kHz and then a 4 dB rise immediately after the crossover point. Subjectively, this could create some "hollowness" in lower speech fundamentals with sharper sibilance. In an anechoic or critical listening environment the peaks at 4 kHz and 5 kHz may present themselves as a bit harsh but in a typical listening environment with carpet, couches, and cats this will most likely increase presence in the upper vocal regions to a satisfactory result while reducing some 1 kHz nasal-ness.

Above 10 kHz, the speaker is well behaved to 15° and as would be expected, drops significantly at 30° and above 12 kHz.

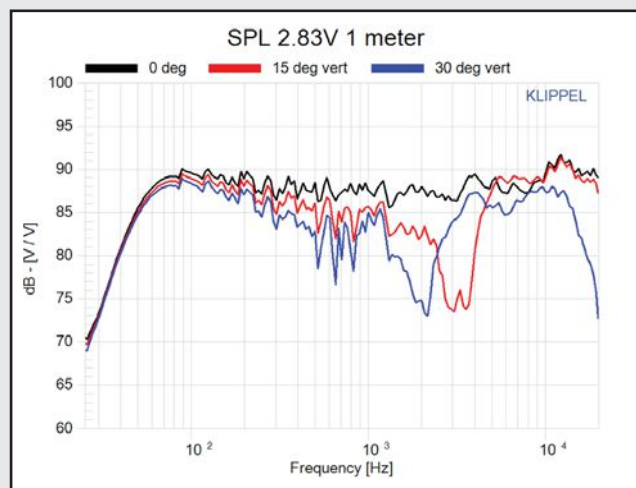
An unwrapped 360° pressure/frequency contour map provides a complete view of the horizontal radiation of the speaker and through its rated frequency response. The horizontal contour mapping affirms the single on/off axis measurements with a few interesting details. For example, the extreme attenuation above 10 kHz and at 30° to 50° makes a 10 dB rebound at approximately 70° and at 12 kHz—presumably this could be explained due to diffraction and the addition of another reflective source at the corner of the cabinet. (As you examine the contour, the "cut" or "sloped" right corner of the top of the cabinet is to the top of the center line, this illustrates slightly less pressure level.)

Vertical on/off axis measurements tell a slightly more complicated story. The Rinjani cabinets are very beamy with respects to a vertical radiation in the mid-band and the SPL measurements bear this out. At 15° (above) the cabinet there is a considerable dip in response at 3 kHz and 4 kHz. Oddly this makes a slight recovery at 30° but shifts the problem lower to 2 kHz, right in the middle of the speech range.

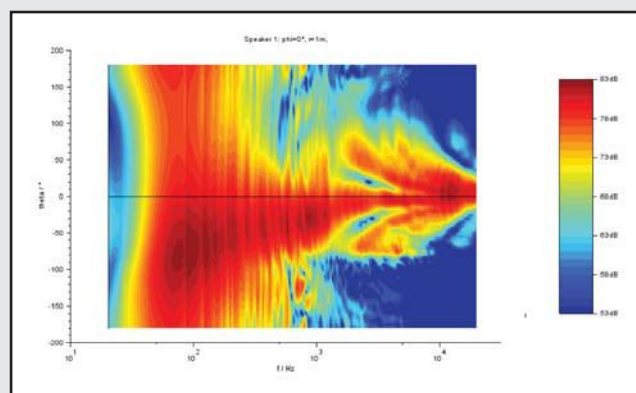
Examining the unwrapped vertical contour balloon, it is clear that there are some off-axis holes 30° off axis beginning at 1,200 Hz and narrowing as frequency increases to 5 kHz. This make a recovery >10 dB, 50° above and below the tweeter.

Of course, this is with a stimulus under measurement conditions. Again, in a typical room with positive reflective surfaces and moderate absorption, these holes would not be exceptionally apparent. However, in a trade show environment (i.e., a large room, space to move close to and then farther away from the speaker while standing), it might be noticeable given a consistent pink noise stimulus—assuming any trade show floor would be quiet enough to concentrate on your own thoughts, let alone a speaker's response and radiation.

If we were making positioning recommendations, it would be at ear level with the tweeter, with small toe-in of the cabinets. All in all, we found the Rinjani attractive, solid, and well behaved within certain angular guidelines.



The vertical on/off axis measurements tell a slightly more complicated story.



Examining the unwrapped vertical contour balloon, it is clear that there are some off-axis holes 30° off axis beginning at 1,200 Hz and narrowing as frequency increases to 5 kHz.

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True Bass in a Large Space

A Pro DIY Subwoofer Project from 1975 Using Electro-Voice's 30W 30" Woofer



Today, getting true bass in a small place is quite easy and straightforward, but in 1975, it was not! After reading Thomas Perazella's DIY article, "True Bass in a Small Space" (*audioXpress*, February 2017), I thought it would be fun to write a follow-up article titled: "True Bass in a Large Space."

Photo 1: Bottom and side images of the Electro-Voice 30W woofer (Photos courtesy of Canuck Audio Mart)

By
Don Keele, Jr.
(United States)

Thomas Perazella's *audioXpress* article "True Bass in a Small Space" described a \$900 very-high-performance subwoofer system using a low-efficiency high-excursion 18" driver mounted in a 4 ft³ closed box driven by a 1,200 W DSP plate amplifier, with a total weight of only 115 lbs. The system was capable of radiating several acoustic watts down to 30 Hz.

After reading that article, I was reminded of my time at Electro-Voice (EV) in the early 1970s when I designed the EV TL bass box line and specifically a very-large vented-box design called the TL303 for the EV 30W 30" driver. This article describes an equivalent solution from 1975 that provides about the same response and acoustic output, but used a high-efficiency low-excursion 30" driver mounted in a massive 76 ft³ vented-box driven by a 60 W amplifier, with a total system weight that exceeded 800 lb!

At the time, the direct-radiator enclosure design theories of Neville Thiele and Richard Small were new and Ray Newman and I and many others at Electro-Voice were at the forefront of encouraging

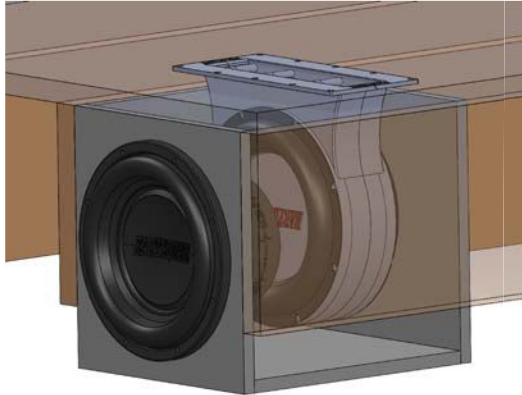
the industry to use the then-new Thiele-Small (T-S) driver parameters for designing vented-box (bass reflex) loudspeaker enclosures.

After reading Perazella's description of the low-efficiency high-output subwoofer system, I wondered how the vintage 1970s 76 ft³ vented-box design with the low-excursion 60 W EV 30" driver might stack up against this latest competition. This article compares the two units using detailed simulations and analysis and comes to a somewhat surprising conclusion.

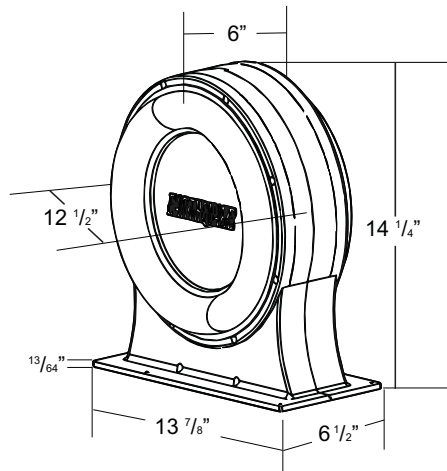
Remember that in the 1960s or early 1970s, a large amplifier was typically a 50 W to 100 W tube amplifier and the speakers of the time had very anemic excursion capabilities compared to today's drivers. Notable exceptions were Electro-Voice's 30" woofer introduced in 1959 with a large cone but still very-limited excursion and Crown's solid-state 150 W per channel DC300, which first appeared in 1967. Gerald Stanley's article is referenced in the Resources section. The 30W woofer featured a gigantic cone area but had a rather anemic excursion capability of only ± 3.8 mm (± 0.15 ")!



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Photo 2: The Dayton Audio 18" Ultimax subwoofer DIY package with a MDF cabinet, a 18" driver, and a 1,200 W DSP plate amplifier, available from Parts Express for about \$900.

As we learned in the original article, Hofmann's Iron Law dictates that to generate lots of acoustic energy at low frequencies with the constraints of the 1970s, meant high efficiency, large cone area, and very-large enclosures! This is the exact opposite of the system described in the Perazella article!

The only feasible Electro-Voice driver at the time for a single-driver solution (not an array) was the EV 30W 30" woofer (see **Photo 1**).

Question: Can a low-excursion 60 W 30" woofer

in a very-large vented box keep up with a high-excursion 1000 W 18" woofer in a small closed box? Keep reading to find out.

The Electro-Voice 30W 30" Woofer

Searching the Internet for "EV 30W Images" I found a number of interesting images. I also found Electro-Voice's four-page 30W engineering datasheet from 1959. On the bottom right of the first page of the datasheet, a bold statement proclaims that "Ported enclosures (bass reflex) are neither necessary nor desirable for the 30W." Ha, little did they know! (Some of the images and the complete four-page datasheet is available in the Supplementary Material section of the *audioXpress* website.)

The 30W has a rather anemic power rating of only 60 W (by today's standards), a 63.5 mm (2.5") voice coil, and a ceramic magnet.

In 2017, according to the *audioXpress* article, there is a high-performance subwoofer system using a very-high-excursion 18" driver mounted in a 4 ft³ closed box driven by 1,200 W DSP plate amplifier. The complete DIY package is offered by Dayton Audio (Parts Express) for about \$900 and includes an Ultimax 18" driver. **Photo 2** shows an image of this kit package.

Electro-Voice TL Bass Boxes

The TL303 vented-box system for the EV 30W driver analyzed in this article is the largest system from Electro-Voice's line of TL bass boxes from the 1970s. The TL bass boxes were part of Electro-Voice's professional sound reinforcement line and were designed to be used with Electro-Voice's line



Photo 3a: This photo from 2004 shows an Electro-Voice HR9040 horn in Jim Long's home with Long on the left and Don Keele on the right. b: Long is shown beside one of his all horn-loaded systems.

of new constant-directivity sound-reinforcement high-frequency horns (models HR4020, HR6040, and HR9040) released in 1973. (A zip file containing specification sheets for these three horns is available in the Supplementary Material section of the audioXpress website.) As a side note, I wrote the verbiage for these spec sheets, plotted all the charts and graphs, and did all the measurements for these sheets. It was a lot of work, but I enjoyed it!

Photo 3 shows two pictures of Jim Long (a 45-year Electro-Voice veteran) where he has a stereo pair of HR9040 horns and folded bass-horn combinations installed for stereo listening in his living room. Amplifiers consist of a small 5 W tube amp driving the horns and a 25 W amp driving the bass horns. With this combination, he can blow your socks off! No room reflections with these babies, only direct sound! Super imaging!

The TL bass box line consists of vented-box and folded-horn designs for all of EV's "EVM" line of pro drivers including the EVM 12L, the EVM 15L, and the EVM 18B. The EV 30W 30" driver was thrown in for the TL303 sub design. Interestingly, Electro-Voice did not offer any of these enclosures as products, but only as DIY plans to support the new horns!

Outside third-party suppliers such as Charlie Wicks (The Captain of the Universe) founder of Proco Sound (www.procosound.com) started his business building TL bass boxes for the industry from these DIY Electro-Voice plans!

Vented-Box Design

With the exception of the two folded-horn boxes, I designed all the cabinets in the TL Bass Box line using the guidance of Thiele and Small. At the time, I came up with a simplified vented-box design method based on Thiele-Small theory, using simple equations that could be calculated with a simple scientific hand calculator. No computers or smart phones in those days! My original handwritten notes from February 1976 are shown in **Figure 1**.

My design method generates vented boxes with a near maximally flat B4 fourth-order Butterworth high-pass response and are based on curve fits to the graphs of Small.

The equations in my method have been replicated all over the place since 1976 without giving me any credit! I recently found an example from a recent online calculator for automotive vented bass boxes from CarStereo.com (www.carstereo.com/help/Articles.cfm?id=17) that directly implements my equations. Actually, I don't mind! My intent was to encourage people and the industry to use the then-new techniques of Thiele and Small to design better speakers! It worked!

Naming Conventions for the TL Bass Boxes

TL stands for "Transducer Low-Frequency." The following numbers in the model name stand for the low-frequency limit in Hertz followed by the

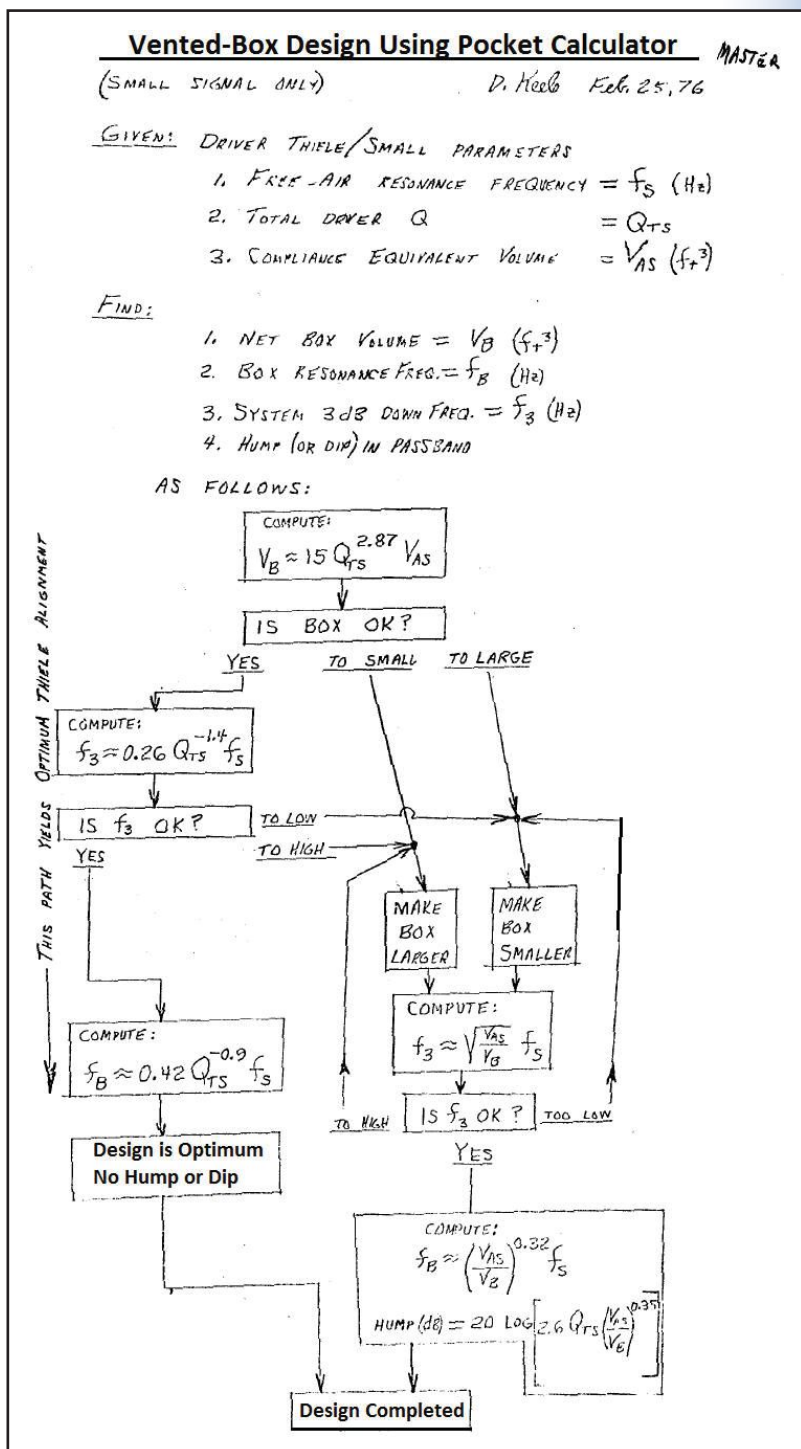


Figure 1: Don Keele's original handwritten notes from 1976 for his vented-box design method using a scientific pocket calculator. Text has been added to correct illegible sections of the image.



Photo 4: The complete line of TL bass boxes including seven direct radiators and two folded-horn boxes. The larger folded-horn box (TL4050) is not shown. The 1975 photo shows Glen Meyer inside the TL303 box and Larry Driskill standing in front.



Photo 5: The complete line of single-driver direct-radiator TL bass boxes are shown from left to right: the TL806, the TL606, the TL505, the TL405, and the TL303. These boxes housed respectively, from left to right: the EVM12L, the EVM15L, the EVM18B (two boxes) and the EV 30W 30" driver.



Photo 6: This is a close up view of the four single-driver TL bass boxes. Note ports on the bottom of the boxes. The left three boxes are set up for the step-down mode of operation with the center port closed. The right box is set up for normal operation with all three ports open.

amount of radiated mid-band acoustic energy in watts. Thus, TL303 would stand for a speaker that can radiate 3 acoustic watts mid-band with output down to 30 Hz.

All the direct-radiator vented boxes of the TL Bass Box line could be operated in two tuning modes called: normal and step down—changing the vented-box tuning by covering up one of the ports in the box.

These modes of operation were pioneered at Electro-Voice by Ray Newman, where he discovered that a fourth-order vented-box response could be changed into a sixth-order response by lowering the box tuning by one-half octave and with the addition of a second-order high-pass filter with a Q of 2.

The normal mode represents a near maximally flat B4 fourth-order Butterworth high-pass vented-box response. The step-down mode extends the low-frequency limit down by about one-half octave by lowering the vented-box tuning frequency by one-half octave and adding an auxiliary second-order high-pass filter with a Q of 2 that provides a 6 dB lift to return the response back to flat.

Later in this article, I provide an example of a step-down alignment as it applies to the TL303 bass box. The auxiliary filter in addition to flattening the response provides much needed reduction of the vented-box's susceptibility to below-band energy content in the program material.

The step-down alignment is essentially the same as Thiele's sixth-order Class I B6 vented-box alignment number 15. For more information, you can read my 1974 paper titled: "A New Set Sixth-Order Vented-Box Loudspeaker System Alignments" (see Resources).

The TL Bass Box line included five single-driver and two four-driver array direct-radiator vented boxes plus two folded-horn boxes housing various Electro-Voice drivers (see **Photos 4–6**). The specification summaries for all the TL Bass Boxes and a complete set of builder's plans for all these systems are available in the Supplementary Material section of the audioXpress website (www.audioXpress.com).

Various views of the boxes from the TL Bass Box line are shown in **Photos 4–7**. All these photos were taken about 1974 in Electro-Voice's engineering lab in Buchanan, MI. To my knowledge, these photos are the only ones that exist of all these cabinets together in one place. Various Electro-Voice employees including me are shown in these photos.

Photo 7 shows a rather humorous photo of the massive 76 ft³ TL303 vented system, which houses the EV 30W 30" woofer. The box is in the "normal" tuning mode with both the top and bottom ports open. Note my right leg sticking out the bottom

port, Glen Meyer's head protruding out the top port, and Bill Raventos completely inside the cabinet!

TL303 Bass Cabinet for the 30W Woofer

The TL303 vented box at 76 ft³ was the largest enclosure of the TL Bass Box line and housed the EV 30W 30" driver. Hey, this enclosure could be the largest ever built! The Builder's Plans for the TL303 are shown in the Supplementary Material section of the audioXpress website.

The original Builders Plans for the TL303 enclosure are dated March 1975. I was the designer, wrote the verbiage, and plotted all the graphs on the left side of the plans. The cabinet is formed by two 4' x 8' of sheets of 0.75" plywood separated by 30" and is very large and heavy!

In normal mode, the TL303 is tuned to 25 Hz with both ports open and in step-down mode is tuned to 18 Hz with the bottom port covered.

Simulations: Small- and Large-Signal

Table 1 shows the T-S driver parameters I used for my simulations. The 30W parameters differ somewhat from the parameters used for the



Photo 7: This is the large 76 ft³ TL303 vented-box cabinet, which housed the EV 30W 30" driver. From top to bottom is Glen Meyer, Bill Raventos, and Don Keele. A speaker engineers' dream playhouse!

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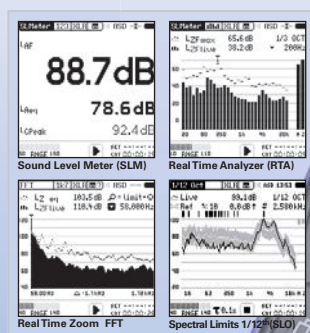
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Table 1: The small- and large-signal Thiele-Small driver parameters for the 30W woofer and the Dayton Audio UM18-22 woofer. These parameters were used for all my simulations.

Thiele-Small Driver Parameters					
Symbol	Description	Electro-Voice 30W 30"		Dayton Audio UM18-22 18"	
Small Signal:					
Diameter	Nominal diameter	30"		18"	
Fs	Free-air resonance frequency	16 Hz		20 Hz	
Qts	Total Q	0.27		0.53	
Vas	Compliance equivalent volume	210 ft³	5.9 m³	7.5ft³	0.21 m³
Large Signal:					
PeMax	Thermally limited maximum input power	60 W		1,000 W	
Vd	Peak displacement volume (= Sd x Xmax)	85 in³	1,390 cm³	165 in³	2,700 cm³
Xmax	Peak linear excursion	0.15"	3.8 mm	0.87"	22 mm
Sd	Effective cone area	564 in²	3,600 cm²	189 in²	1,220 cm²
Eff	Reference efficiency half-space	8.8%		0.31%	

TL303 Builder's Plans and are more up to date. The parameters for the Ultimax UM18-22 18" woofer are those provided by Parts Express.

The program I used for the simulations I wrote myself using Igor Pro (see Source). The program is quite simple and utilizes the enclosure modeling equations of Thiele and Small. It assumes completely linear operation when the driver is operated below its excursion limit.

I have been using Igor since 1988 for all my technical papers and articles and I give it my highest

recommendation! It first started out on the Apple Macintosh, and I initially ran it on my Mac 512K and later on my Mac SE for many years. Igor Pro is major competition to MatLab. My Igor box simulation program is available in the Supplementary Material on audioXpress website and will run on the free Igor Pro demo.

How to Read the Response Graphs

Figures 2–8 show the simulated small- and large-signal frequency responses of the systems I analyzed.

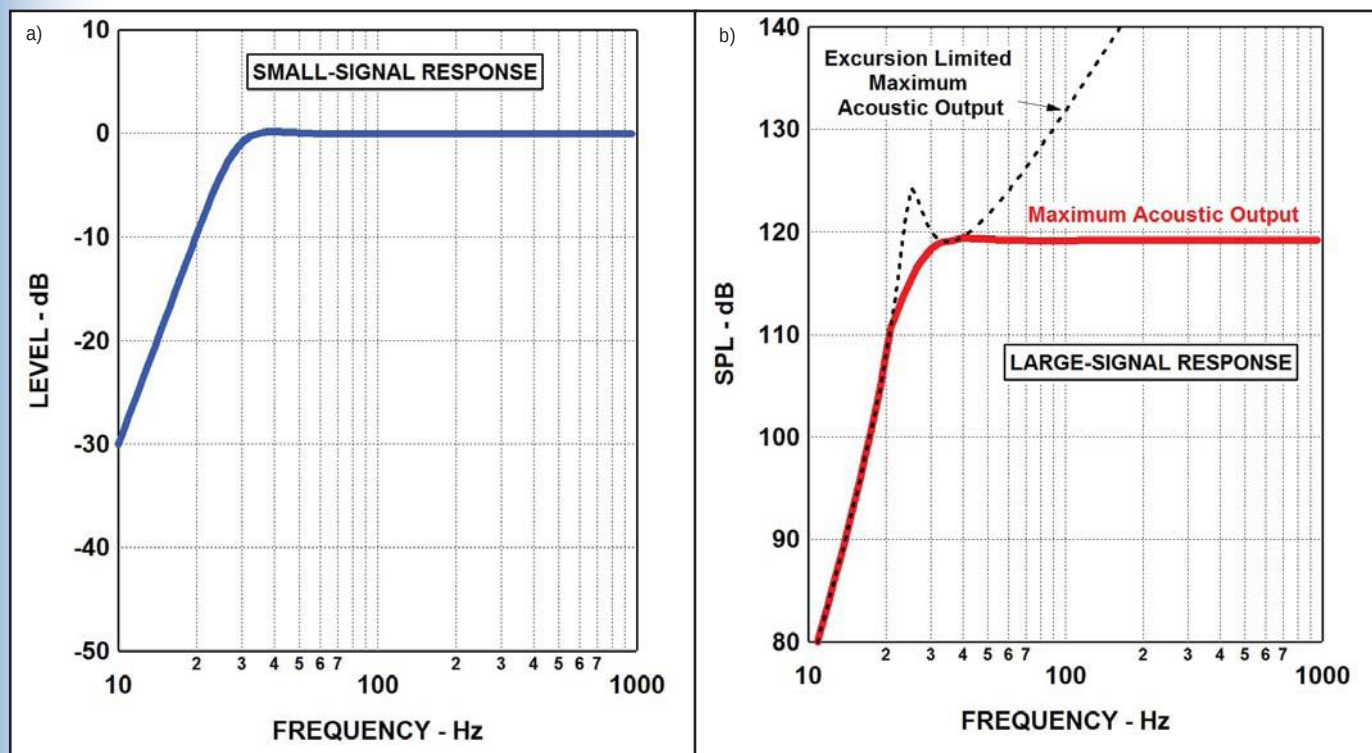
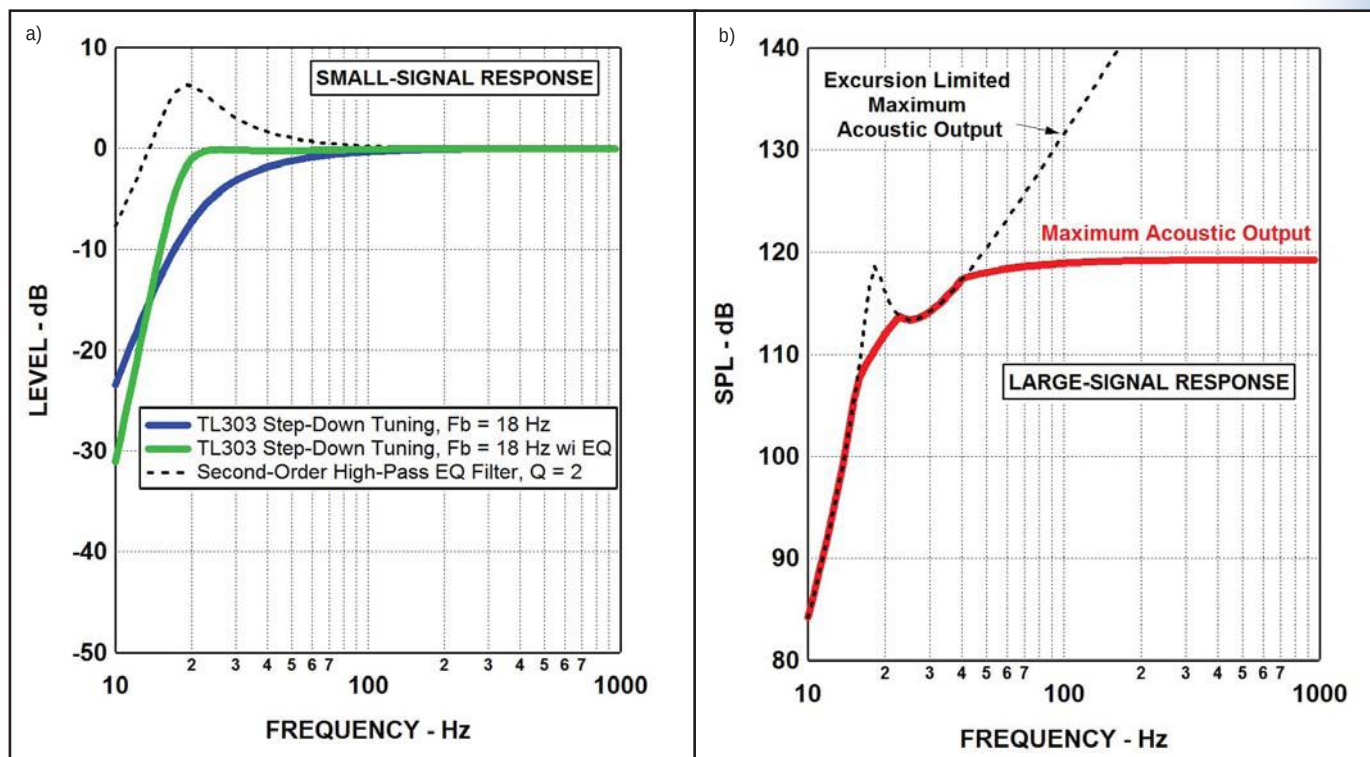


Figure 2: The small (a) and large (b) signal responses of the TL303 system in normal tuning mode. The cabinet houses the 30" Electro-Voice 30W driver in a 76 ft³ vented box tuned to 26 Hz. Note the ideal responses.



The last two figures (Figure 7 and Figure 8) illustrate simulation comparisons of the various cabinets.

Each figure shows the small-signal response on the left graph (a) and the large-signal response on the right (b). The large-signal graph on the right (b) shows two plots: the maximum acoustic output (red curve), and the excursion-limited maximum acoustic output (black dashed curve). Both predictions are at 1 m and plotted in dB SPL.

The small-signal response of a closed box falls at 12 dB per octave below box resonance (see Figure 4a as an example), while the response of a vented box falls at 24 dB per octave below box tuning (see Figure 2a).

With today's sophisticated DSP processors, the small-signal frequency response can be equalized to have any shape or low-frequency limit desired. Hey, if you want your 3" driver system to be flat to 10 Hz, you can do that! However, it won't play very loud at low frequencies because the driver can't move enough air!

The large-signal maximum output is what defines how loud a system will play at each frequency. If you want a system to be usable to low frequencies, its maximum output must be acceptably high at all these frequencies!

The levels noted on all the large-signal graphs are the sound pressure in dB SPL at 1 m in a half-space for omni-directional radiation. This assumes that the driver is mounted in an infinite flat baffle. In this situation, one acoustic watt will generate 112 dB SPL at 1 m.

A speaker system's maximum output is limited by both the thermal and excursion limits of the driver. When you raise the input to the speaker,

you will reach either the driver's thermal limit or its excursion limit, whichever occurs first.

The shape of the excursion-limited output curve is distinctly different for closed-box and vented-box systems. The closed-box output rises at 12 dB per octave over the entire range (see Figure 4b).

This is in contrast to the vented-box output that rises at 24 dB per octave below box tuning and also at 12 dB per octave above box tuning (see Figure 2b). In the region around box resonance (in this case 26 Hz), the excursion-limited output reaches a peak at box tuning and then falls somewhat and then continues rising like a closed box as if the vent were closed.

A well-designed vented box matches the thermal and excursion limits of the system at the frequency of the dip in the excursion curve. The TL303 design (see Figure 2b) meets this criterion. If a system

Figure 3: The small (a) and large (b) signal responses of the TL303 system in step-down mode. Figure 3a shows the response of the second-order equalizer (black dashed curve) and the equalized response (green curve).

About the Author

Don Keele was first introduced to audio, music, and loudspeakers back in the late 1950s. Since that time, Don has worked for a number of companies in the area of loudspeaker R&D and measurement technology including Electro-Voice, Klipsch, JBL, Crown, and Harman International. For 10 years, he wrote for the now-defunct *US Audio Magazine* as a Senior Editor performing loudspeaker reviews. More recently, he worked under Floyd Toole as a member of the Harman corporate acoustics engineering group. Although technically now retired, at 76 he heads his own consulting company DBK Associates and Labs. He holds eight patents with topics including "constant-directivity" loudspeaker horns, loudspeaker arrays, and signal processing. He is a life fellow of the Audio Engineering Society (AES). His passion for the last 15 years has been to promote the use of Constant Beamwidth Transducer (CBT) loudspeaker technology in the loudspeaker industry. He has written 16 AES papers on this topic. In 2002, he received a Scientific and Engineering Academy Award for work he did on cinema constant-directivity loudspeakers. In 2016, he was presented with the AES Gold Medal Award for outstanding and significant research, design and product development, over a period of more than 45 years, of loudspeakers and loudspeaker systems providing broadband constant-coverage performance.

doesn't meet this criteria, it is over designed from either a thermal or excursion standpoint.

Correspondingly, a well-designed closed-box system matches the displacement and thermally limited maximum output curves below the closed-box corner frequency (see Figure 6b).

TL303 Normal Tuning

The small- and large-signal responses of the EV 30W in the 76 cubic foot vented box tuned to 26 Hz are shown in Figure 2. The curves exhibit near ideal vented-box responses with a 3 dB down point (f3) of about 26 Hz.

The small- and large-signal responses of the EV 30W in an 18 Hz-tuned 76 ft³ vented box are shown in Figure 3. Here the vented-box tuning was lowered by one-half octave from 26 Hz to 18 Hz. The overall small-signal response in Figure 3a (blue) exhibits a gentle rolloff, which is equalized flat with a second-order high-pass with a Q of 2 (dashed black curve). The resultant equalized curve (green) exhibits a much lower f3 of about 18 Hz. Figure 3b shows the large-signal response of the system. Note the significant excursion limiting below 16 Hz and between 23 and 40 Hz.

BYU Bass Guitar System

The small- and large-signal responses of the EV 30W in a 28 ft³ closed box are shown in Figure 4. This enclosure is a near optimum closed-box design!

This box was used in pairs by the Brigham Young University (BYU) marching band for reproducing bass guitar (see Sidebar) and was designed by Ray Newman of Electro-Voice.

The small- and large-signal responses of the EV 30W in a 42 Hz-tuned 76 ft³ vented box are shown in Figure 5. The small-signal response (see Figure 5a) exhibits a peak of about 4 dB at about 60 Hz.

The large-signal response of this box (see Figure 5b) only exhibits significant displacement limits below 30 Hz. The peak noted above carries over to the large-signal response (right) where it represents a healthy 4 dB increase in maximum output compared to the closed-box in a frequency range solidly in the bass guitar's range!

Ultimax Closed Box

The small- and large-signal responses of the Dayton Audio Ultimax UM18-22 18" woofer in a 4 ft³ closed box (the system described in the *audioXpress* article of February 2017) are shown in Figure 6. This system is not excursion limited at any frequency! The system is an ideal closed-box design with the excursion-limited maximum output just grazing the thermally-limited output at low frequencies.

Simulation Comparisons

Now, compare all three systems. Figure 7 shows comparative small- and large-signal responses for three systems: The TL303 system in normal mode,

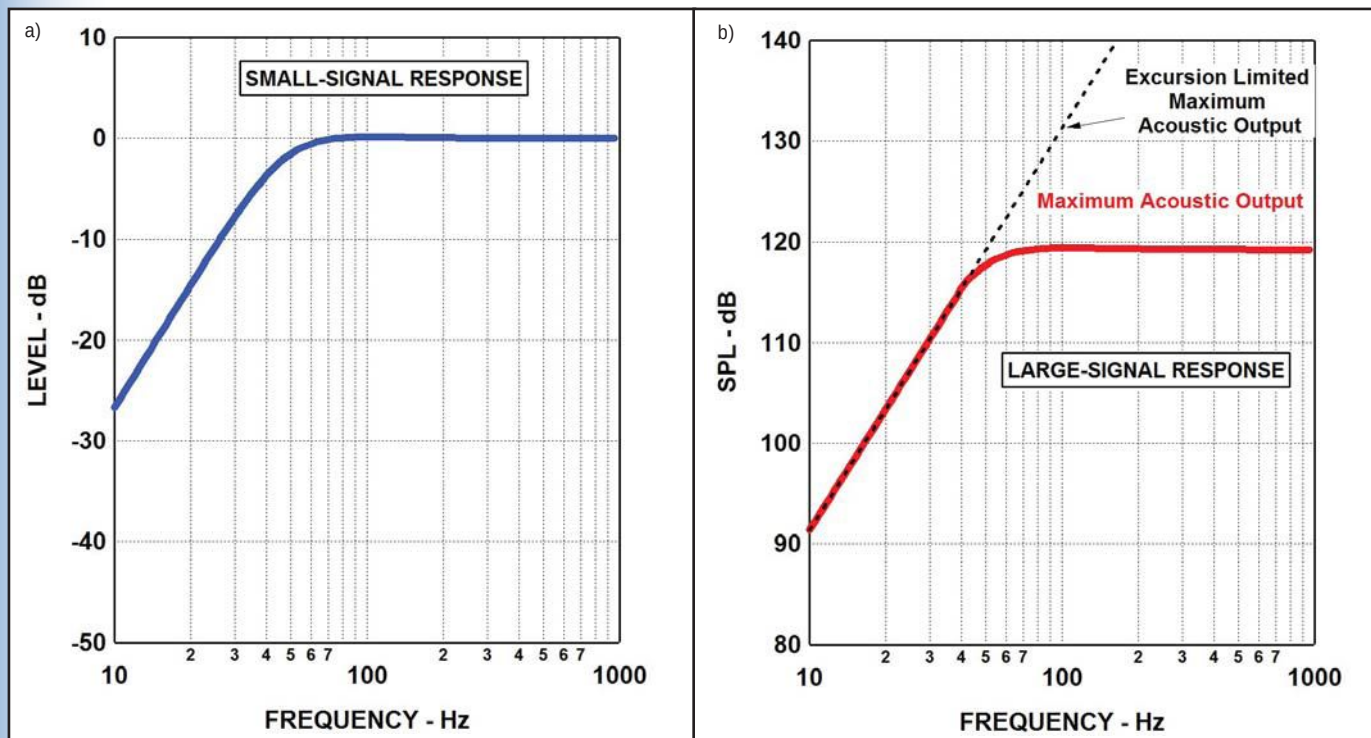


Figure 4: The small (a) and large (b) signal responses of an EV 30W driver mounted in a 28 ft³ closed box.

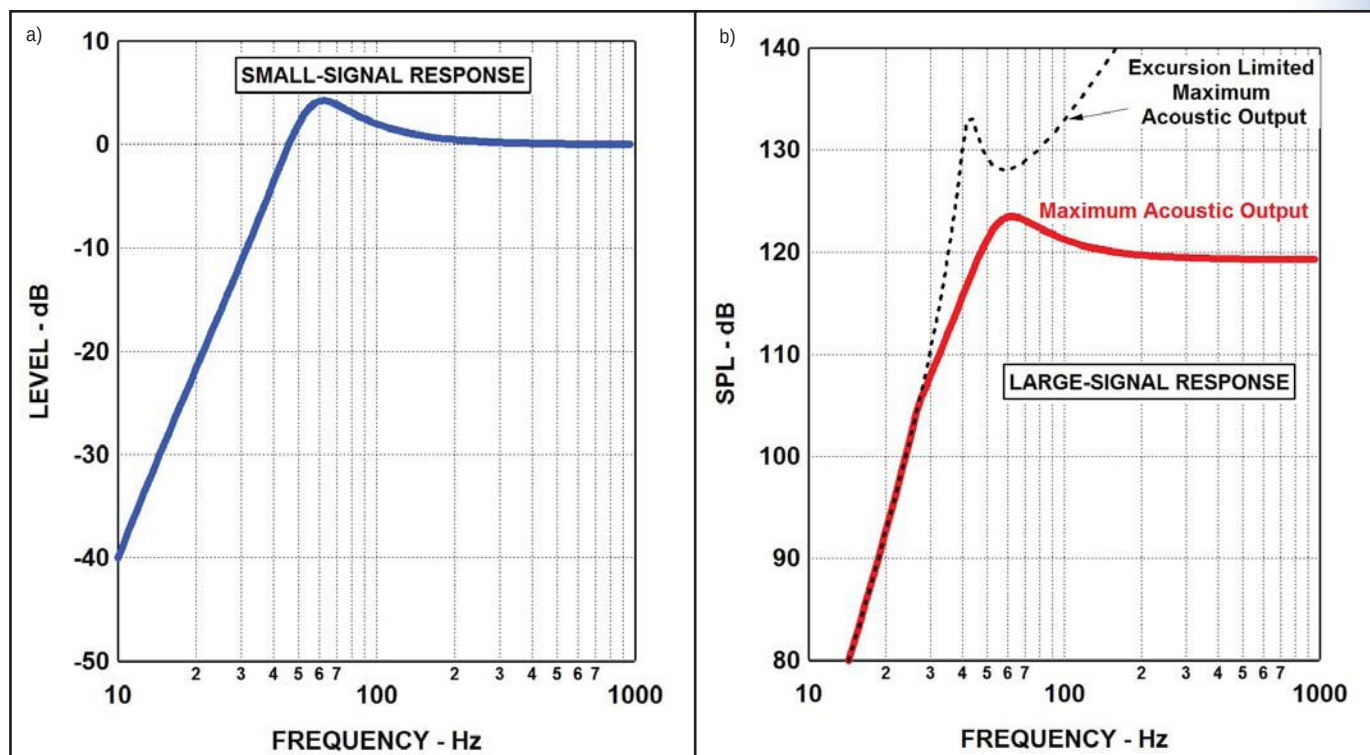


Figure 5: The small (a) and large (b) signal responses of an EV 30W driver mounted in 28 ft³ box tuned to 42 Hz.

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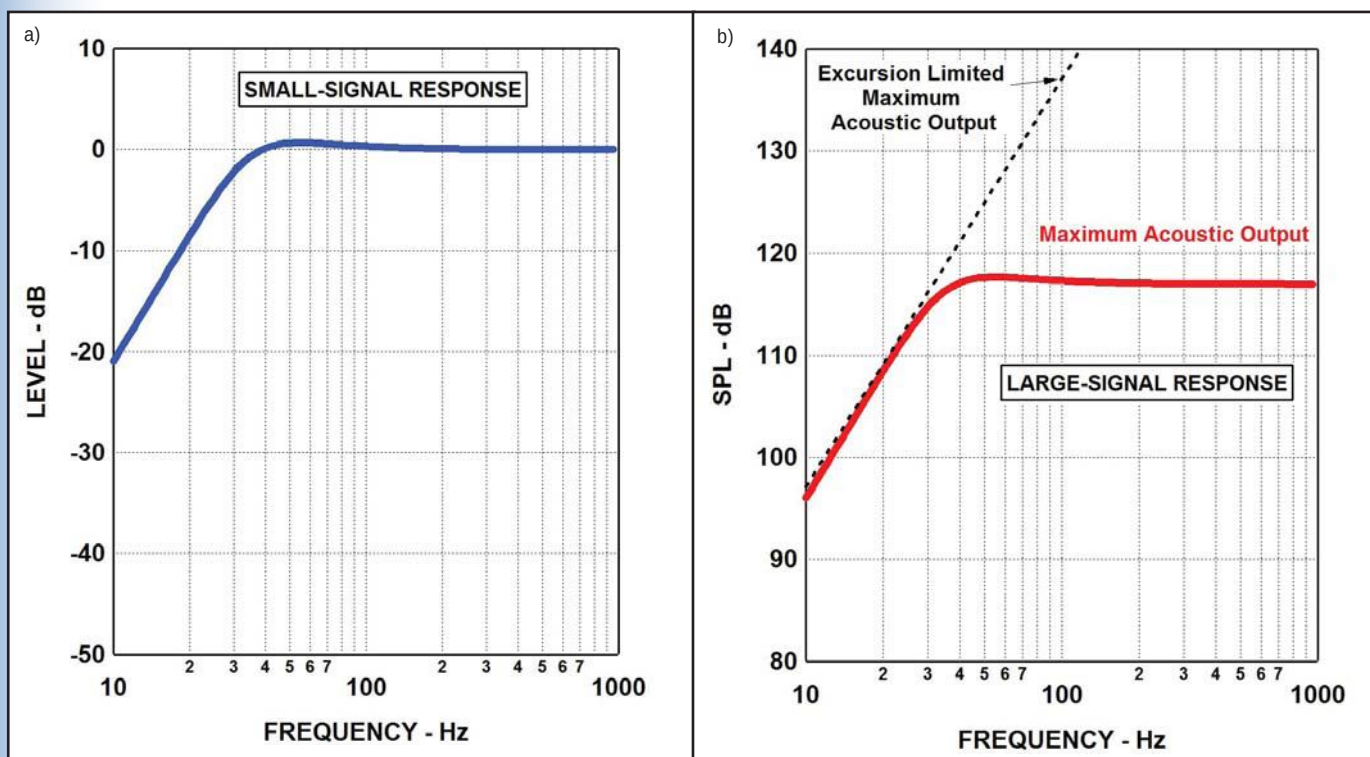


Figure 6: The small (a) and large (b) signal responses of the Dayton Audio Ultimax UM18-22 18" driver mounted in a 4 ft³ closed box.

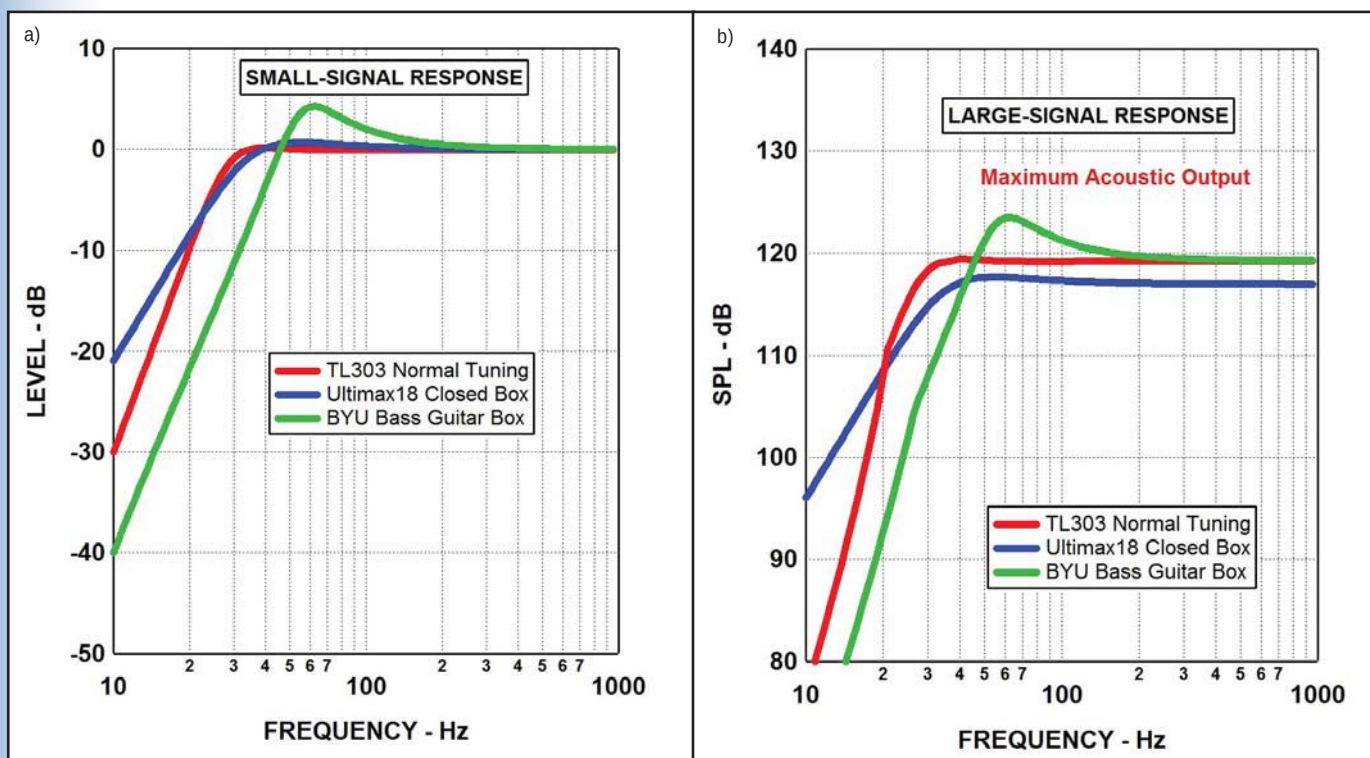


Figure 7: A comparison of the small-signal (a) and large-signal (b) responses of three systems analyzed in this article: the TL303 normal tuning (red), the Ultimax system (blue), and the BYU bass-guitar vented system (green).

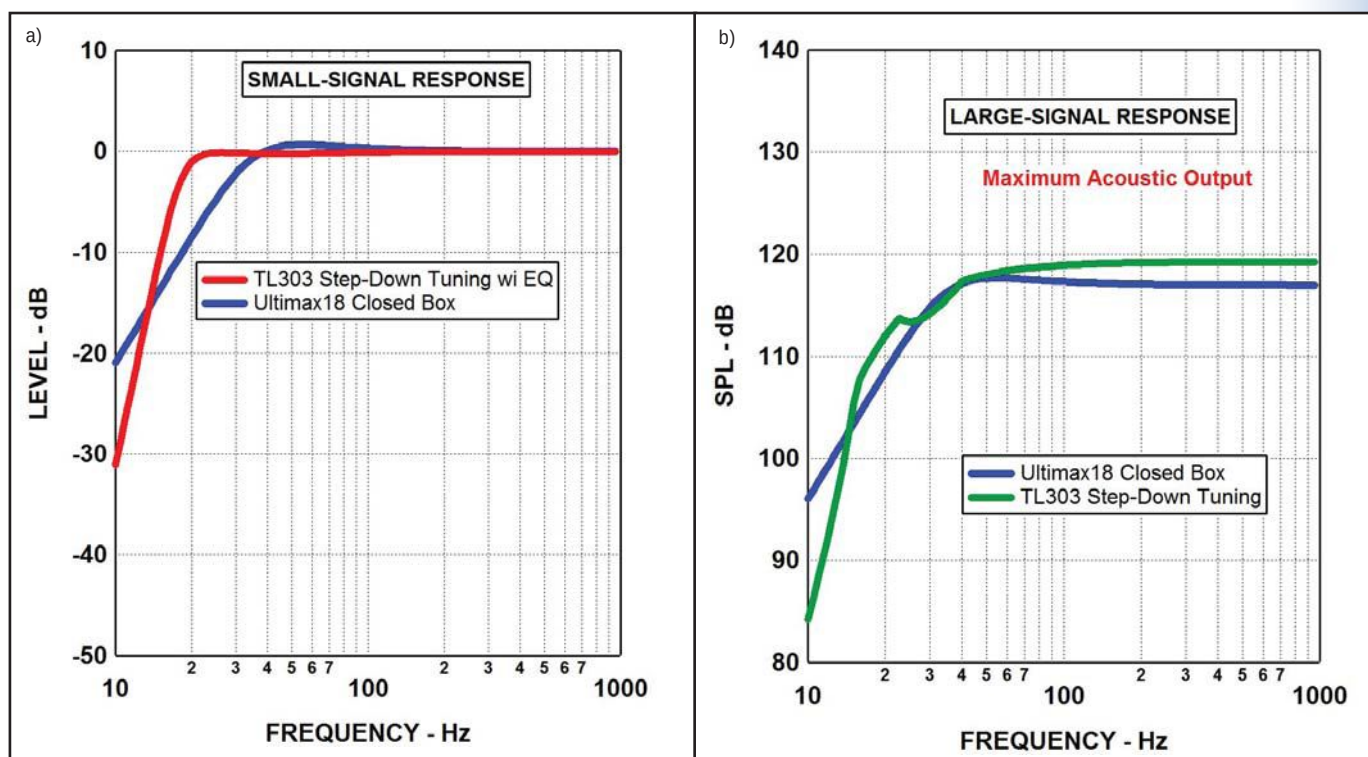


Figure 8: The small (a) and large (b) signal response comparison between the equalized TL303 system in step-down mode and the Ultimax closed-box system.

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the Ultimax closed-box system, and the vented BYU Bass Guitar box.

The small-signal comparison (Figure 7a) shows the TL303 and the Ultimax system have somewhat similar responses above 22 Hz, but the TL303 rolls off much quicker at lower frequencies. The BYU system, however, exhibits a higher cutoff frequency, but with a significant peak in the response at about 60 Hz.

The large-signal responses (see Figure 7b) of the three systems have somewhat similar shapes to the small-signal responses but differ in level. The mid-band maximum levels of all the 30W systems exceed the Ultimax system by about 2.3 dB. This is not the entire story though, because the BYU vented system at 63 Hz exceeds the maximum output of the Ultimax system by a whopping 6 dB! This is ideal for a bass guitar!

Now, compare the TL303 step-down tuning with EQ and the Ultimax Closed Box. Figure 8 shows a comparison between the small- and large-signal responses of the equalized TL303 in step-down mode and the Ultimax closed-box system.

Here, the TL303 is clearly the small-signal winner with a much lower f_3 of about 18 Hz compared to the f_3 of the Ultimax system at about 28 Hz (see Figure 8a). However, as mentioned before this isn't the entire story because the large signal responses are what count. The TL303 maximum output (see Figure 8b) is significantly higher by about 3 dB between 16 and 22 Hz, about the same between 28 and 50 Hz, and about 2 dB above 150 Hz.

I declare the TL303 to be the winner here! It will play louder than the Ultimax system over most of the frequency range. But hey, after all is said and done, I'll pick that 4 ft³ 115 lb. Ultimax package over that enormous 76 ft³, 585 lb. system any day! However, in 1975, you did not have the luxury of this choice!

Conclusions

In my introduction I asked: Can a low-excursion 30W 30" woofer in a very-large vented box keep up with high-excursion 1000 W 18" woofer in a small closed box? The answer is a resounding yes!

My simulations show that the maximum acoustic output of the large-box TL303 system in normal tuning mode exceeds the output of the small-box 18" system by an average of about 2 to 3.5 dB above 24 Hz (see Figure 7b). In addition, the maximum output of the large-box TL303 system in step-down tuning mode (see Figure 8b) equals or exceeds the maximum output of the 18" closed-box system at most frequencies by about by about the same margin above 15 Hz.

Furthermore, the smaller 27 ft³ BYU bass-guitar tuned 30" driver box (see Figure 7b) exhibited a maximum output averaging 2 to 6 dB above 45 Hz compared to the 18" driver box.

Okay, I have shown that in the mid-1970s it was possible to design a high-performance subwoofer system that could generate several acoustic watts down to 30 Hz and below, and compete very nicely with a modern system.

BYU Marching Band 30W Bass-Guitar Box

While preparing this article, I had a vague recollection that a pair of TL303 bass boxes were used by the Brigham Young University (BYU) marching band for reproducing bass guitar at football games and for their marches in parades. (FYI: I graduated from BYU in 1972 with an MSEE degree with a minor in acoustics.)

However, my recollection was only partially correct. It was not a TL303 bass box, but a custom-built 30 W closed-box enclosure that the marching band had used, designed by Ray Newman of Electro-Voice! Its dimensions were about 4' x 4' x 2' deep and had a net internal volume of about 28 ft³.

The band would set up a pair of these systems on the football field during games along either side of the 50 yard line in the middle of the field. This was quite unique at the time because very few marching bands had bass guitar accompaniment for their band!

They said the BYU bass guitarist was very skillful in either playing ahead of or behind the band depending on where the band was on the field so they kept in time. The marching band

also used these bass cabinets towed behind the band on a trailer when they marched in parades.

An interesting recollection by Ed Jones concerned the BYU band marching in Ronald Reagan's inaugural parade in 1981. They commandeered a brand new Chevy El Camino to haul the two bass boxes and drove behind the band to reproduce bass guitar. He stated that they used a 200 W Altec 1590B power amplifier that would run on 24 V and powered it with two 12 V automotive batteries!

For this article, I added an extra wrinkle. I converted the system to a vented box to increase its output and optimize the response for a bass guitar. The only logical place to tune the box is in the range of 40 to 50 Hz regardless of the resultant response shape. This is because the frequency of the open "E" string of a bass guitar is 42 Hz and as a result has minimal energy below this frequency.

I chose 42 Hz for box tuning and included this box configuration in my article's simulations. I did not consider any port implementation issues (i.e., real vent or passive radiator). I will leave that as an exercise for the reader!

Resources

D. B. Keele, "A New Set Sixth-Order Vented-Box Loudspeaker System Alignments," presented at the 49th Audio Engineering Society (AES) Convention, September 1974, Preprint No. 993.

D. B. Keele, et al, "An Important Aspect of Underhung Voice-Coils: A Technical Tribute to Ray Newman", Presented at the 121st Audio Engineering Society (AES) Convention, October 2006.

R. H. Small, "Vented-Box Loudspeaker Systems, Part I: Small-Signal Analysis." *Journal of the Audio Engineering Society*, Volume 21, pp. 363-372, June 1973.

G. Stanley, "History Files: The Crown DC300 Amplifier Leads The Solid-State Revolution"
http://www.prosoundweb.com/topics/sound_reinforcement/history_files_the_crown_dc300_amplifier_leads_the_solid_state_revolution.

R. H. Small, "Loudspeakers in Vented Boxes," *Journal of the Audio Engineering Society*, Volume 19, Part II, pp. 471-483 (June 1971).

A. N. Thiele, "Loudspeakers in Vented Boxes," *Journal of the Audio Engineering Society*, Volume 19, Part I, pp.382-391, May 1971

*All Audio Engineering Society (AES) papers (including those listed here and more than 50 others written by Keele since 1972) are available from the Audio Engineering Society's E-Library (www.aes.org/e-lib).

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However, those 1970's systems have some extremely significant major downsides: extremely-large enclosures and excessive weight. In light of this, the edge shifts dramatically to the current systems. If one allows for multiple drivers in the current systems (i.e., a dual-driver 18" system in 8 ft³ closed box driven by a 2,400 W amplifier or equivalently two of the Ultimax systems set up side by side), the new systems would walk all over the old systems by a large margin!

I know which one I would choose! 

Author's Acknowledgements: I am completely indebted to the following people, who have been my friends since the 1970s, for help with this article, particularly for contributing to the historical portions of this piece: Jim Long (45-year Electro-Voice employee, now retired), Ed Jones (former manager of sound systems, LDS Church, now retired), Bill Raventos (director of marketing, Ivie Technologies), Glen Meyer (formerly of LDS Church, Larson Davies, and Ivie, now retired), Larry Driskill (Electro-Voice Pro Microphone product manager, now retired), and Ken Dickensheets (principle consultant at Dickensheets Design Associates).

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The Thinman Project

The Thinman Project is perfect for audio enthusiasts who want their sound from real speakers.

By
Ken Bird
(United States)

In this digital audio age, when most millennials are now listening to audio over earbuds or from shoebox-sized Bluetooth or other wireless speakers, you would think that the traditional two-speaker stereo system would be on the endangered list. This theory was further enhanced when recently, a contemporary DIY speaker builder friend stated that those of us who still build our own 2.1 speaker systems have become audio fossils.

I disagree with this observation for a great many other audio "fossils" are now driving the revival of 2.1 analog stereo. The explosion in new and reissued vinyl records is one example of the continuing interest in analog audio by both young and old alike. So to those audio enthusiasts who still want their sound from real speakers, I suggest the Thinman system.

Thin Speakers

In the early 1960s, many main stream speaker manufacturers launched a line of very thin speakers. They varied in depth from 3" to 6". Some were even meant to be hung on the wall and featured artwork on the speaker grilles. They incorporated three-way systems with up to 12" woofers that had convoluted magnet assemblies and low-resonant frequencies. They worked fairly well but fell out of favor due to their rectangular shape, which made placement in the average living room awkward.

My Thinman

This variation of the thin speaker is 5.5" wide and 49.5" high with a footprint that is less than a square foot, in a three-way configuration

Photo 1: The completed Thinman system shows the side located woofers. I decided to paint the speakers Colonial Red.

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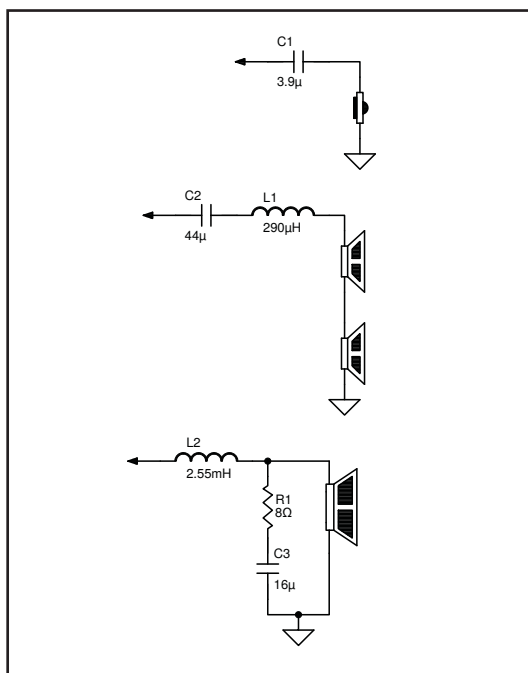


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Figure 1: Here is the schematic for the Thinman design.



(see **Photo 1**). The 8" woofer is side firing in the ported section of the enclosure and the midrange and the dome tweeter are a classic Joe D'Appolito arrangement in their own sealed enclosure, crossing over at 500 and 4,500 Hz, through a first-order circuit with an impedance compensation network in the woofer circuit (see **Figure 1**). The parts that I used to build the circuit are provided in the Parts List. Prior to assembling the system, I

connected the crossover and drivers on the bench top and tested them for proper operation. More advanced builders concerned with diffraction losses can add a 2.9 Ω resistor in parallel with a 0.5 mH coil in series with the system. However, experimentation in the listening room did not show any advantage in adding this circuit. Detailed information on this issue can be found in the books listed in the reference section.

Although I did not test it, Part Express has a three-way ready-built crossover that should work well with this system. It is listed as an option in the Parts List. If you use a commercial crossover, it is recommended that the impedance compensation network be used with it. Those interested in learning the details of building their own crossover networks, along with complete systems, should consult the books listed in the reference section.

The Drivers

The woofer is a Dayton 8" dual voice coil type using the 8 Ω coil only. The midrange speakers are surplus neodymium 4", 4 Ω types, from Parts Express. In case they are no longer be available, the Parts Lists has a tested recommended substitute. The dome tweeter is from MCM and the terminal cup is from Parts Express. I did not install any speaker grilles, however, if you have small children or curious pets, you might want to add grilles to any low-mounted woofers. I have recommended one in the Parts List and it is shown in **Photo 2**.

Enclosure Construction

The enclosure was constructed from dimensional lumber with the front and rear panels using 1' $\times$ 6" (0.75" $\times$ 5.5") $\times$ 48" pieces and the side panel is 1' $\times$ 10" (0.75" $\times$ 9.25") $\times$ 48" long. The front panel is cut at 47.25" leaving 0.75" open for the duct port. The internal port duct and midrange/tweeter enclosures were ripped to 4" from a piece of the 1' $\times$ 6". The bottom was made from a piece of 1' $\times$ 8" (7.5" $\times$ 7.25") and the top from a piece of 1' $\times$ 6" (10.75" $\times$ 5.5"). I assembled all the pieces with liquid nails and 1.25" nails using a pneumatic nail gun. I decorated the edges of the front, top, and bottom panels with a "Roman Ogee" pattern using a router (see **Figure 2**).

I cut all the lumber from 8' lengths to the required length. Your local lumber yard will be happy to make the cuts for you and some home centers have them pre-cut so you can select them individually. I used pine for this project as it was intended to be painted rather than stained. You can also use any type of hardwood, if you want a traditional stained finish (see **Photo 3**).



Photo 2: The drivers and terminal cup used in the Thinman include the Dayton 8" woofer, two 4" midrange units, and a dome tweeter.

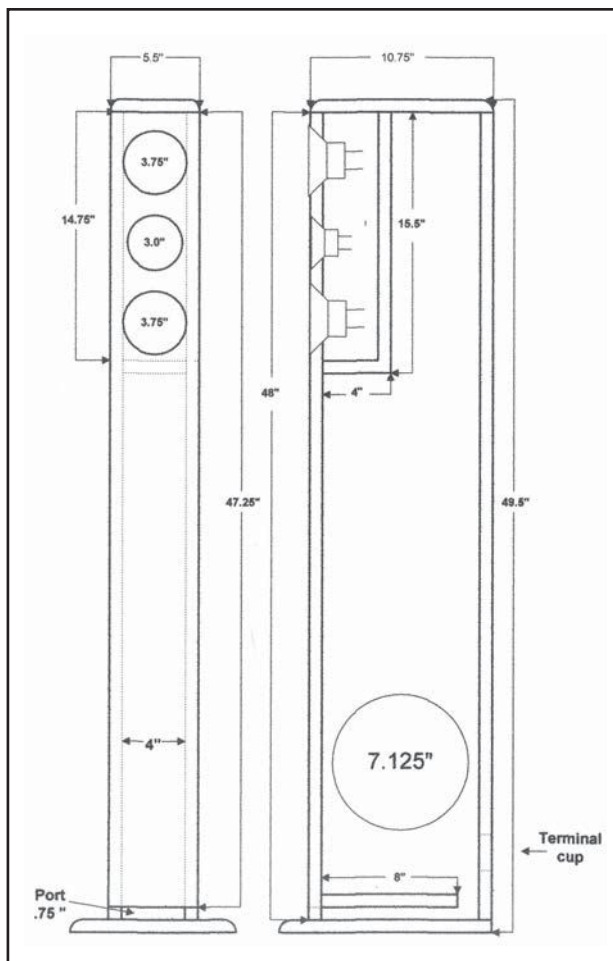


Figure 2: I followed this diagram when building the enclosure.

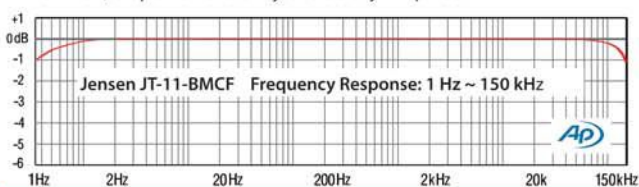


Photo 3: The precut lumber required to construct one enclosure is shown.



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Once the boards are cut to length, start with the 47.25" front panels and lay out the speakers on a center line (see **Photo 4**). Space the first midrange driver 1" from the top. Then space the tweeter and the lower midrange driver 3/8" apart. Holes for the 4" drivers are cut 3.75" and the tweeter are 3". Use

a rat-tail file to cut out the "ears" for the tweeter terminals. The holes can be cut with a jig saw, but if you have a router, the Jasper Circle Jig is the best method for the cuts outs (see **Photo 5**). Check the fit of the drivers and set the panel aside. The rear panel has a 3" hole cut out for the speaker terminal. Locate it 6" up from the bottom. The woofer holes are cut at 7 1/8" and are not counter sunk into the sides. The bottom of the woofer hole should be 6" from the bottom of the side panel. Remember, one enclosure will have the woofer facing out to the right and the other to the left.



Photo 4: Here, the midrange and the tweeter drivers are layed out on the front panel.



Photo 5: The Jasper circle cutting jig was used to cut all the speakers openings. It enables circles from 2" to 18" to be cut to the necessary increments.

Enclosure Assembly

Corner clamps are used to hold the butt-jointed sides and end panels in place. Pre-assemble the side panel without the woofer hole and the front panel first (see **Photo 6**). Add a bead of liquid nails to the panel edge, then clamp with the corner clamps, making sure the alignment is square. Then use 1.25" finishing nails spaced every 3" to attach the panel (see **Photo 7**). Then take the internal midrange enclosure pieces and coat the bottom

Parts List (for one enclosure)

Dimensional Lumber:

Two each 1 x 10 side panels; (9.25" x 48")
 One each 1 x 6 front panel; (5.5" x 47.25")
 One each 1 x 6 rear panel; (5.5" x 48")
 One Top piece 1 x 6 (5.5" x 10.75")
 One Bottom piece 1 x 8 (7.25" x 7.25")
 One each 15.75" x 4" internal enclosure wall
 One each 4" x 4" internal enclosure bottom

Drivers:

Woofer: One each Dayton SD 215-88 8" DVC
 Midrange: Two each Part Express WFRO 4002114
 4" 4 Ω 299-345
 Or substitute: Dayton MD 104-4 290-212
 Tweeter: MCM Soft Dome 53-1080
 Optional grille for woofer: Parts Express #260-424
 Crossover Parts: Are all sourced from Parts Express
 C1: 4 μf (Part number: 027-330)
 C2: 47 μf (Part number: 027-352)
 C3: 15 μF (Part number: 027-344)
 L1: 2.5 mH (Part number: 266-556)
 L2: 0.28 mH (Part number: 266-814)
 R1: 8 Ω, 10 W (Part number: 044-8)
 Optional commercial crossover
 (Part number: 260-210)

Speaker Terminal:

Parts Express (Part number: 260-311)

Miscellaneous:

1 roll 3/8" foam insulation strip
 1 package of 0.5" polyester blanket lining.
 6 oz of polyester pillow fill material
 1 tube silicone caulk
 40 each 1" drywall screws
 6' each of Red, Black, and Yellow stranded hook up wire
 16 gauge

and front section with liquid nails. Position it carefully and nail in place. Drill a 0.25" wiring hole in the bottom section of the midrange enclosure. Then add the port duct following the same glueing and nailing procedure. Do not add the side panel

with the woofer hole just yet. While the liquid nails can provide a good seal at the enclosure joints, it is advisable to use a caulking gun to seal the internal seams of the enclosure.

The crossover is mounted on the lower

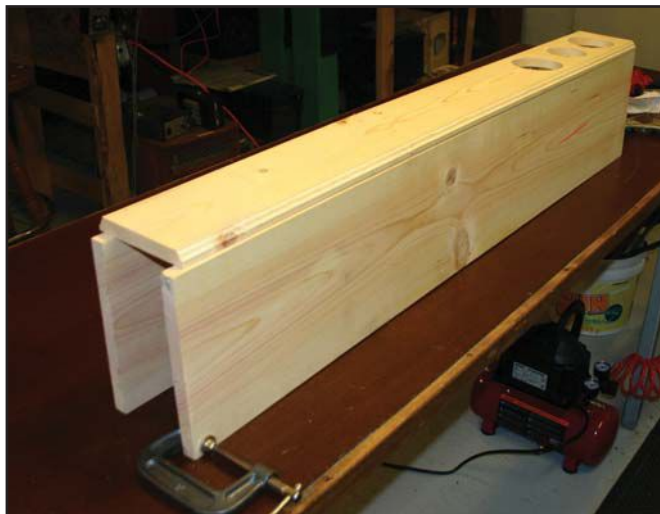


Photo 6: Attach the side panel without the woofer hole first with the opposite panel used as a brace during the nailing.



Photo 7: The enclosure can be assembled with finishing nails, a hammer, and a nail set tool, but the use of a pneumatic nail gun made the work faster and easier, with less risk of damage to the panels.



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Application Examples



4/8 Ch Amp



2/4 Ch Amp



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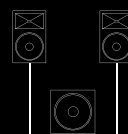
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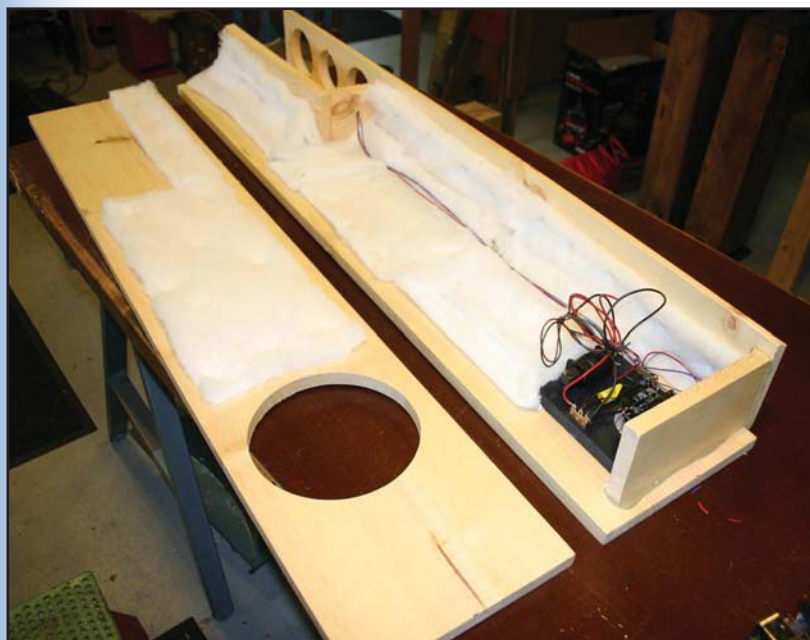


Photo 8: The midrange enclosure, port duct, crossover with pre-wiring, and 0.5" polyester batting, was added prior to attaching the side and rear panels.

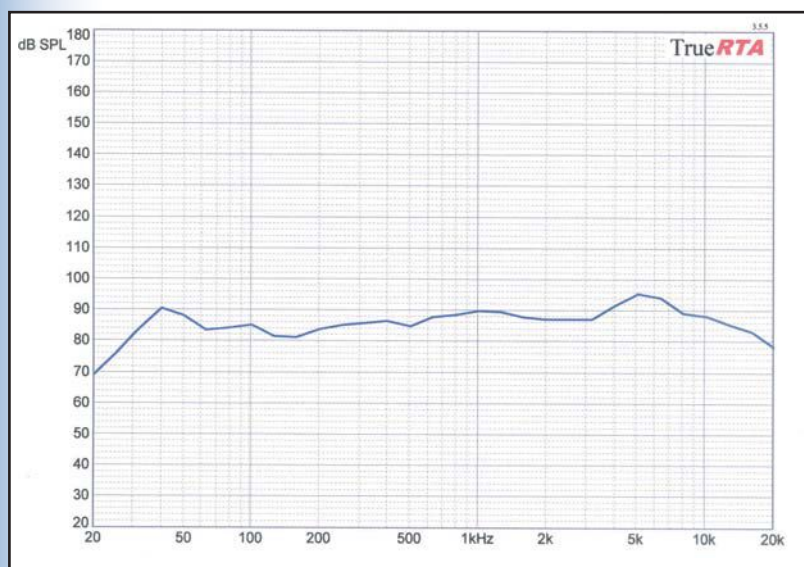


Figure 3: The system measurement of both channels in the listening room is shown. Room effects make an accurate measurement difficult, but the overall performance can be seen here.

About the Author

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

portion of panel opposite the woofer (see **Photo 8**). All the wiring for the upper midrange enclosure, woofer, and the terminal cup were installed prior to assembly of the opposite woofer panel. The location makes it easier to remove the crossover through the woofer hole if future service is needed. I used two 1.25" drywall screws to secure the crossover. Before assembling the sides and the front panels, staple a layer of 0.5" polyester blanket batting inside each panel, allowing a 1" space at the edges. Do not put any inside the area of the midrange internal enclosure. On the side panel, opposite the woofer, make the polyester 1" thick directly behind the woofer area. After stapling the polyester batting, install the woofer side rear panel, followed by the top and bottom.

Surface Finishing

The final surface finish on the enclosure is your choice. If you used good quality hardwood, a



Photo 9: The raw enclosures are ready for sanding and final finish.

natural or stained finish would be a good choice. This version was painted in a popular decorator color, after first having the nail holes filled in with Plastic Wood, then sanding the surface with medium sandpaper. After sanding, I applied two coats of wood primer, then sanded with fine grit paper and then two coats of the final paint were applied. Allow 12 to 24 hours for all the primer and paint to dry before installing the speakers (see **Photo 9**).

Speaker Installation

Having pre-wired the crossover and installed it in the enclosure, the speaker installation is much easier. I used different color wire for the midrange and tweeter along with a common return wire. Start by wiring the in series midrange drivers first, keeping the polarity as shown in the circuit diagram (see Figure 1).

Then connect the tweeter wire. To assure a good air seal, I applied 3/8" foam weather stripping as a gasket to the back of each speaker prior to mounting them into the enclosure. I then used 1" drywall screws to mount the drivers. Prior to mounting the speakers, add about 3 oz of polyester pillow fill material to the midrange enclosure.

Having mounted the midrange and tweeter, wire the woofer and apply the foam gasket. Mount the woofer with the 1" drywall screws. Place the foam gasket material on the rear flange of the terminal cup, wire it to the proper polarity, and mount it in the enclosure.

Testing the Speaker

Use a low volume setting when first testing the speaker, noting any unusual noise, distortion, or lack of frequency range. If all sounds well, bring up the volume to your desired listening level. You will find this system works best 12" to 18" out from the wall, spaced 6' to 12' apart. Imaging is fine and enhanced bass can be had by placing them in the room corners. The in-room response exhibited an f3 of 38 Hz with the RTA analysis system (see **Figure 3**). Whatever your application, you will find the Thinman to be a versatile system that uses little floor space while giving full-bodied sound. 

Resources

R. Alden, *Speaker Building 201: A Comprehensive Course in Speaker Design*, Audio Amateur Publications, 2004, Item #: AA-BKAA066, www.cc-webshop.com.

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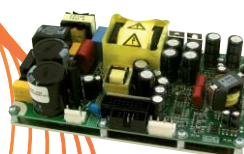
D. Weems and G. R. Koonce, *Great Sound Stereo Speaker Manual*, 2nd edition, TAB Electronics Series, McGraw-Hill/Tab Electronics, 2000.

hypex NCxxxMP series overview

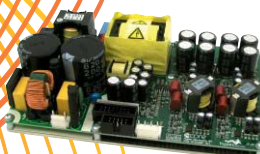
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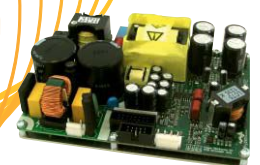
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NC250MP



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NC500MP



NC502MP

Add-on Module NC100HF

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Hybrid Electrostatic Loudspeaker with Tailored Dispersion

The psychosis got the best of me again, folks—this time it's a wide dispersion hybrid electrostatic loudspeaker (ESL) with a really cool stator design. This speaker is three generations evolved from my original ESL described in "Hybrid Electrostatic Loudspeakers Constructed with a Twist," (*audioXpress*, April 2012).

By
Charlie Mimbs
(United States)

The tried and true beam splitter cabinet and woofer were carried over from the earlier design. What's new and completely different is the electrostat panel, which now uses insulated copper wire stators and electrical segmentation to tailor its dispersion (see **Photo 1**).

With me having no math or electronics skills, this project would not have been possible without the technical advice I received from ESL gurus Steve Bolser and Rod White, who go by the user names Bolserst and Golfnut on the diyAudio forum. White's white paper, "Wide-Range Electrostatic Loudspeaker with a Zero-Free Polar Response," (*Journal of the Audio Engineering Society*, October 2009) and his speaker project, "Another Segmented ESL," (diyAudio, April 2013) provided the inspiration and Bolser's "Segmented ESL Calculator," and gift for explanation enabled a techo-dummy like me to derive the segmentation scheme and resistor values. These guys are the best.



Photo 1: This project is a hybrid ESL with electrically segmented wire stators.

The Basics

Dimensions:	66.5" height × 15" width × 19.75" diameter
Bass Cabinet:	4 ft ³ , 9' transmission line, V-section beam splitter
Woofer:	10" Aurum Cantus AC-250MKII
Stators:	Segmented wire conductors on oak lattice, 10.5" × 46.5" active area
Wires:	20 AWG solid copper, 0.010 XLPVC, 11" spacing, 43% open
Diaphragm:	6 μm Mylar C with Licron Crystal ESD coating
Bias supply:	2.7 kVDC from 115 V/230 V TFMR into a diode/capacitor cascade
Transformers:	(2) 50 VA 230 V/2 × 6 V toroidal, 76:1 ratio
Crossover:	Behringer DCX2496, 48 dB/octave LR filter at 228 Hz

Dispersion

Wavelengths shorter than a speaker's radiating width tend to beam rather than spread out. And flat-panel ESLs are the worst for beaming because they require a large radiating area to offset their limited excursion and dipole rolloff. The resulting "head-in-a-vise" sweet spot is great for solo listening at the focus but not so good for entertaining guests, where wider dispersion is needed.

The typical way to widen an ESL's dispersion is to curve its panel, and thereby its wave front. Segmented ESLs curve the wave front electrically, using discrete stator conductors driving discrete zones on the diaphragm. Directivity sonograms

comparing segmented, unsegmented, and curved panels show smoother and wider dispersion from segmented panels (see Resources).

Electrical Segmentation

My segmented panel uses symmetrically arrayed, discrete vertical wire groups driven by stepped-frequency signals, acting as a line source projecting a cylindrical wave front.

Each stator has 90 insulated copper wire conductors arrayed in 15 groups of six wires. These 15 wire groups are apportioned into eight discretely powered electrical sections; consisting of one wire group in the center of the stator comprising section

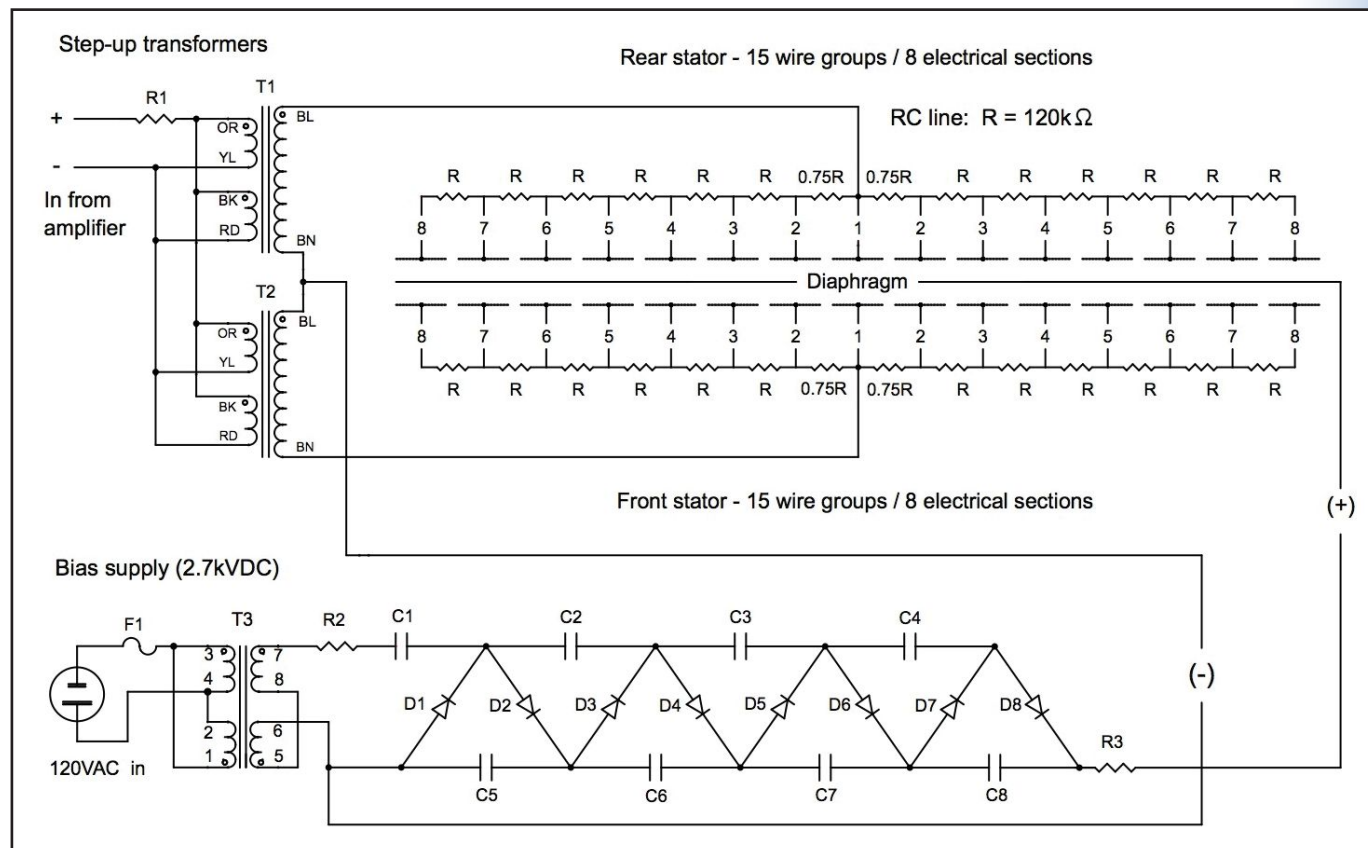


Figure 1: I followed this schematic to build my hybrid ESL.

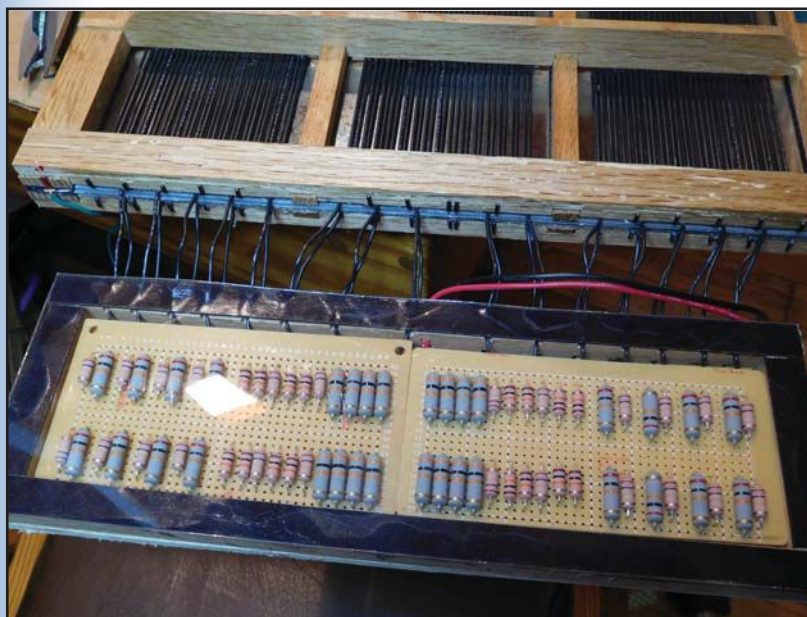


Photo 2: These are the RC network resistors.

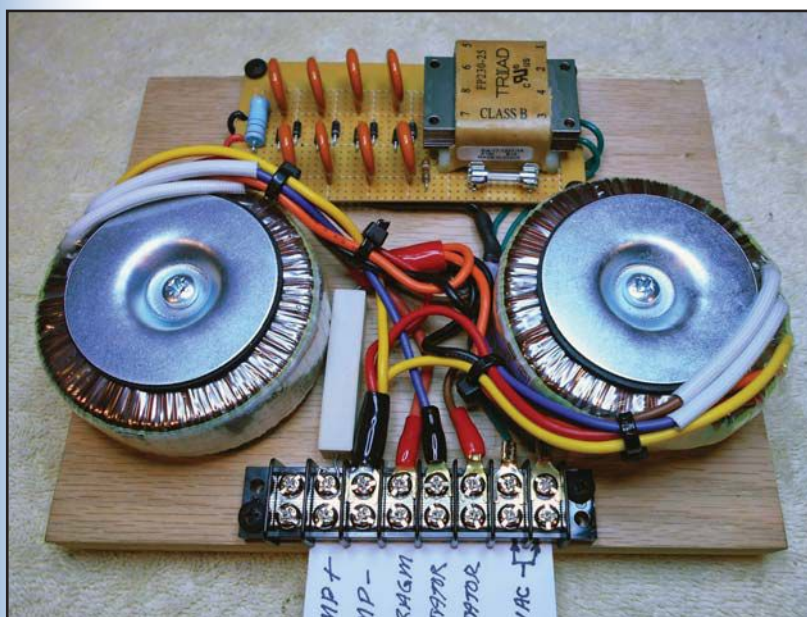


Photo 3: The amplifier/ESL interface includes the step-up transformers and bias supply.

one, and seven left/right paired wire groups arrayed symmetrically on either side comprising Sections 2-8 (see **Figure 1**).

The section one wire groups connect directly to the amplifier interface and receive the full audio bandwidth above the crossover frequency. The left/right paired wire groups in Sections 2-8 are powered thru an RC transmission line that progressively steps down frequencies toward the panel edges. The RC line consists of resistors inserted between the wire groups, which couple with the wires capacitances to form a series of low-pass filters.

As driven by the sectioned stators, the diaphragm radiates the highest frequencies from only a narrow vertical zone at its center, and the left/right paired zones on either side radiate progressively lower frequencies toward the edges. In this way, the width of the radiating zones is always less than the radiated wavelengths—resulting in all frequencies spreading out rather than beaming.

RC Network

The segmentation scheme and resistor values were derived using the Segmented ESL Calculator spreadsheet (see Resources). From the spreadsheet options I chose Symmetric Config 2, both stators segmented, in eight electrical sections. For my panel the spreadsheet calculated 120 k Ω for R, with R/9 series resistance on the section one wire groups, 0.75R on section two wire groups, and R on sections 3-8 wire groups (see Figure 1 and **Photo 2**)

It's common practice to move/reflect the section one resistances to the primary side of the transformer to protect against core saturation (see Figure 1, resistor R1). Reflecting the R/9 resistances across an ideal transformer would divide the sum by the turns ratio squared (4.6 Ω). However, placing this much resistance on the primary side of a real transformer with its winding resistance, leakage inductance, and winding capacitance will result in significant rolloff of high frequencies.

The transformer winding capacitance adds to the load capacitance and its leakage inductance combines with the load capacitance to generate an ultrasonic resonance peak in the frequency response and rapid rolloff above it. Coincident with this response peak is an impedance minimum and difficult load for an amplifier. When series resistance is added on the primary side it dampens this resonance peak. Too much resistance on the primary side over-damps the resonance, rolling off the high frequencies.

The spreadsheet assumes an ideal transformer is used, so it doesn't calculate the effect of resistance on the primary side of a real transformer. The

Guidelines for Segmented Wire-Stator Hybrid ESLs

Ideal (practical minimum) diaphragm-to-stator gap (d/s):	0.062" (+0.015/-0.000)
Wire diameter (insulation included):	should not exceed the d/s
Gap between the wires (insulation-to-insulation):	should not exceed the d/s
Ideal (maximum output) open area:	42%
Span between diaphragm supports:	70 to 100 x d/s
Maximum span between wire supports (wire gauge/inches):	22/2, 20/3, and 18/4
More/narrower wire groups = wider/smoothier dispersion but no advantage < 12 mm	
Bias voltage:	2.5 to 5 kV
Transformer power/ratio:	80 to 120 VA/50 to 100:1

general guidance is to omit the section one resistors, add a 1 Ω series resistor on the primary side and give it a whirl. My panels sounded really good with this initial setup.

The only tuning, if any, is adjusting the series resistance on the primary and/or the first two stator sections to dial in the treble response. Less resistance increases treble and vice versa. My old ears don't hear the highs so well but I didn't want less than 1 Ω on the primary side and the Section 1 resistors were already omitted, so I reduced the Section 2

resistors from 0.75R (90 k Ω) to 60 k Ω to brighten up the treble, and that works for me.

The schematic shown in Figure 1 and the Parts List, which is available on my webpage (see Resources) or in the Supplementary Material section of the audioXpress website, also detail the spreadsheet values except with the Section 1 resistors omitted and reflected as 1 Ω on the primary. I think this would be optimal for most listeners.

All remaining resistors on the secondary side are 2 W, 500 V in series. Wattage/voltage are highest

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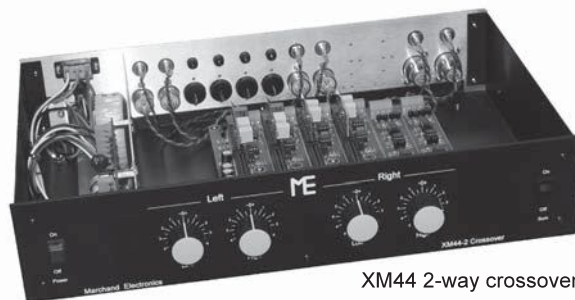
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Photo 4: I assembled the stator support lattice over the wires in the stretching jig.



Photo 5: All-thread rods hold the correct wire spacing while you are gluing the oak support lattice onto the wires.

across the first resistors and decrease down the line. Multiple resistors are ganged to spread the load, as follows:

- Section 1: None (reflected as 1 Ω on TFMR primary)
- Section 2: (3) 30 k Ω
- Sections 3 and 4: (3) 40 k Ω
- Sections 5–7: (1) 100 k Ω + (1) 20 k Ω
- Section 8: (1) 120 k Ω

Amplifier/ESL Interface

Each stat panel has an interface to its amplifier; consisting of a high voltage DC power supply to bias the diaphragm and a pair of step up transformers to convert the amplifier's output into the higher voltage AC required to charge the stators (see **Photo 3**).

Each interface uses two 50 VA 230 V/2 $\times$ 6 V toroidal transformers wired in tandem with the 6 V windings in parallel as the primary and 230 V windings in series as the secondary—giving a 76:1 winding ratio. The DC biasing supply uses a floating ground that's center tapped between the transformers 230 V windings.

The DC biasing supply is a simple half-wave rectifier and a voltage multiplier outputting 2.7 kV. It is powered by a 115 VAC mains current into a 115 V/230 V transformer and a diode/capacitor ladder with a 20 M Ω charging resistor at the output. The charging resistor helps stabilize the charge on the diaphragm and limits the potential current that might otherwise sustain any arcing to the stators.

Stretching the Stator Wires

While not an absolute necessity, stretching the copper wires to plastic deformation renders them perfectly straight and re-aligns their metallic structure such that they then remain straight when relaxed. I opted to stretch the wires to hold tighter gap tolerances (see **Photo 4**). Experiments showed that stretching to 1% elongation is sufficient, so my wire loops were stretched from 47.5" to their 48" final length.

Stretching 90 of the 20 gauge wires at once requires a strong jig and about 4,500 lbs of force. My jig is a 0.75" MDF platform mounted in a stout frame cut from yellow pine 4 $\times$ 4s. The wire loops wrap over 0.063 diameter $\times$ 0.250 length steel pins inserted half depth into drilled 3/16" aluminum plates. The pins are angled 4° from vertical to hook the wires. One pin plate is stationary and the opposite movable pin plate

bolts to a 3/16 steel sub plate that's welded to two 0.75 x 12 all-thread jack rods. Turning the coupling nuts on the jack rods pulls the movable pin plate to stretch the wires.

Stator Support Lattice

The wires are supported by an interlocking oak lattice which is assembled and glued down over the wires, in the stretching jig (see Photo 4). Before stringing the wires, I covered the jig platform with wax paper to prevent gluing the wires to the jig. And before gluing, I relaxed most of the wire tension to prevent preloading and warping the stator. I used yellow wood glue for the wood-to-wood bonds and E6000 glue for the wood-to-wire bonds.

First, I laid the vertical lattice rails down into the jig, and then I glued the interlocking horizontal slats down one-at-a-time, over the wires. During this process, I placed lengths of 5/8-11 TPI all-thread rods over the wires as comb-guides to maintain correct wire spacing (see **Photo 5**). CAD drawings for the cabinet, stator lattice, and stretching jig are available through my website (<http://jazzman-esl-page.blogspot.com>).

Diaphragm Support Spacers

The spacers bond the diaphragm to the lattice rails and set the diaphragm-to-stator gap (d/s) at 0.063". The vertical lattice rails are flush with the wires and its spacers are one layer of 0.063" x 0.075" wide 3M double-sided urethane foam tape. The horizontal end rails run under the wires and anchor their end loops with glue bonds, and its spacers consist of one 0.047" x 0.075" polycarbonate shim bonded onto the wires with E6000 glue, plus one layer of 0.015" x 0.075" wide 3M UHB double-sided foam tape over the shim (0.062" total). The double-sided foam tape spacers bond the diaphragm to the stator instantly with minimal fuss.

Diaphragm

The diaphragm is vertically sectioned into equal thirds for stability (see **Photo 6**



Photo 6: Next I bonded the stator to the diaphragm, on the tensioning jig.



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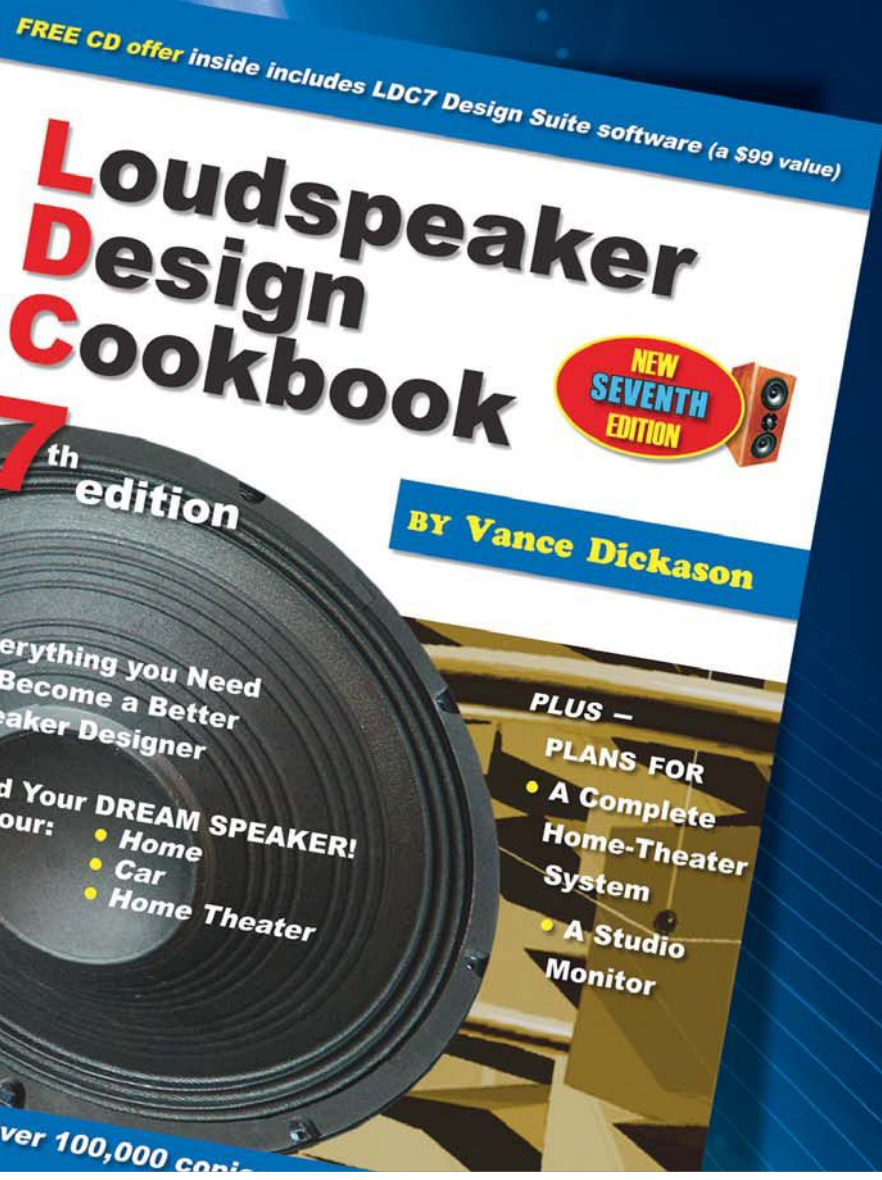
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and **Photo 7**). It's made of 6 μm Mylar C tensioned to 1.25% elongation using a pneumatic bike-tube jig. The jig is an MDF platform sized 2" longer and wider than the stator and 2" high with all edges rounded over to 0.50" radius, sanded smooth and dusted with baby powder to prevent snagging the delicate diaphragm film. A 700 mm $\times$ 35 mm Schrader valve-type bike tube is stretched around it's perimeter.

The Mylar film is wrapped over the jig and secured on the back side with double-sided tape. Inflating the bike tube with a hand pump tensions the diaphragm. Tension is gauged by first marking reference points on the diaphragm exactly 12" apart using a fine tip felt pen. As the tube is inflated, the target elongation is reached when the distance between the reference marks reaches 12 and 5/32". The stator is then pressed into place over the diaphragm to affect the bond.

Next, I masked off the periphery edges of the diaphragm with painter's tape and sprayed the Licron Crystal ESD coating in one "just wet" coat and allowed to dry for eight hours before assembling the panels. The coating dries to a pale blue-gray, almost clear coating about 2 μm thick.

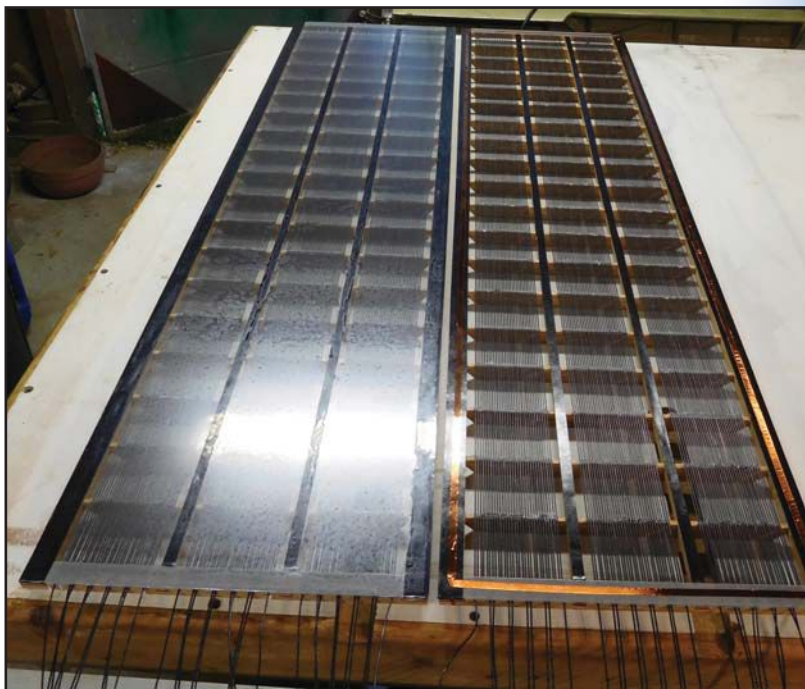


Photo 7: Front and rear stators are shown with the spacers, the diaphragm, and the charge ring in place.

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Photo 8: The beam splitter transmission line is installed in the bass cabinet, which is shown with the frame.

Charge Ring

The charge ring is 0.25" wide copper foil tape applied to the periphery of the rear stator, centered on the foam tape spacers (see Photo 7). The wire lead from the DC biasing power supply is soldered to it and when the front and rear stators are mated together the charge ring contacts and conducts the biasing voltage onto the diaphragm.

Bass Section

The woofer box is a single-fold, tapered transmission line stuffed with 0.5lbs/ft³ of polyfil (see **Photo 8**). The line's sectional area is 125% of the woofer's cone area at the front, tapering to

Project Files

To download the Parts List for this project, visit <http://audioxpress.com/page/audioXpress-Supplementary-Material.html>

Resources

diyAudio Forum, Planar Speaker section, www.diyaudio.com/forums/planars-exotics.

C. Mimbs, "Hybrid Electrostatic Loudspeaker with Tailored Dispersion" Parts List, Jazzman's Electrostatic Loudspeaker Page, <http://jazzman-esl-page.blogspot.com>.

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——, "Wide-Range Electrostatic Loudspeaker with a Zero-Free Polar Response," *Journal of the Audio Engineering Society*, Volume 57, Issue 10, October 2009.

Sources

AC250 MKII Woofer
Aurum Cantus | www.arumcantus.com

DCX2496 Digital crossover
Behringer | www.music-group.com/brand/behringer/home

TFM-25 Amplifier
Carver Sound and Audio | www.carversound.com

About the Author

Charlie Mimbs is a 63-year-old, father of two, and a Manufacturing Engineer III with Gulfstream Aerospace in Savannah, GA, specializing in composite structures. He has built ESL speakers as a hobby since 2008. His other hobbies include woodworking, Tai-Chi and fly fishing. He has had very little formal education beyond high school but advanced from aircraft mechanic to M.E. with Grumman Aircraft in 1985 and have since worked as an M.E. for Sikorsky, Vought, and now Gulfstream.

100% of same at the terminus. The cabinet is 0.75" MDF sheathed in 5 mm red oak plywood and the panel frame is solid red oak.

To minimize the woofer box's profile and footprint, its volume extends upward, behind the stat panel, and its frontal surfaces are angled to form a V-shaped "beam splitter," which deflects the panel's rearward sound out the open sides of the speaker rather than back to the diaphragm.

When designing the bass section, I followed Roger Sanders' lead and opted for a transmission line enclosure and a woofer with low moving mass, low Q_{TS} , and low inductance coupled with a very strong motor magnet. The ideal matching woofer doesn't exist of course, but low inductance takes priority and the Aurum Cantus AC250 MKII I chose works pretty well.

Power and Control

The speaker pair is vertically bi-amplified using a Behringer DCX2496 digital crossover feeding a pair of vintage Carver TFM-25, 225 W/channel stereo amplifiers. The general guidance is to set the crossover frequency is least two octaves above the diaphragm's drum head resonance with

a 24 dB/octave filter slope or at least one octave above resonance with a 48 dB/octave filter slope. My diaphragms resonate at about 90 Hz and the crossover is set at 228 Hz using the Behringer's 48 dB/octave Linkwitz-Riley filter.

Listening Impressions

This speaker projects a wonderful sense of space and ambience. Overall it's seamless and balanced with spooky clarity, nice imaging, and big sound. It can play any genre with authority and the beautiful oak lattice wire stators are just icing on the cake.

Now, Patricia Barber's "Too Rich for My Blood" is truly sublime and Stevie Ray Vaughan's "Tin Pan Alley" is just too cool. When I exhibited the new speakers at the 2016 Carverfest retreat, my fellow Carver heads were drooling and the Bobfather (Bob Carver) loved them! I think physics favors electrostats. 

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New Design Brings International Recognition

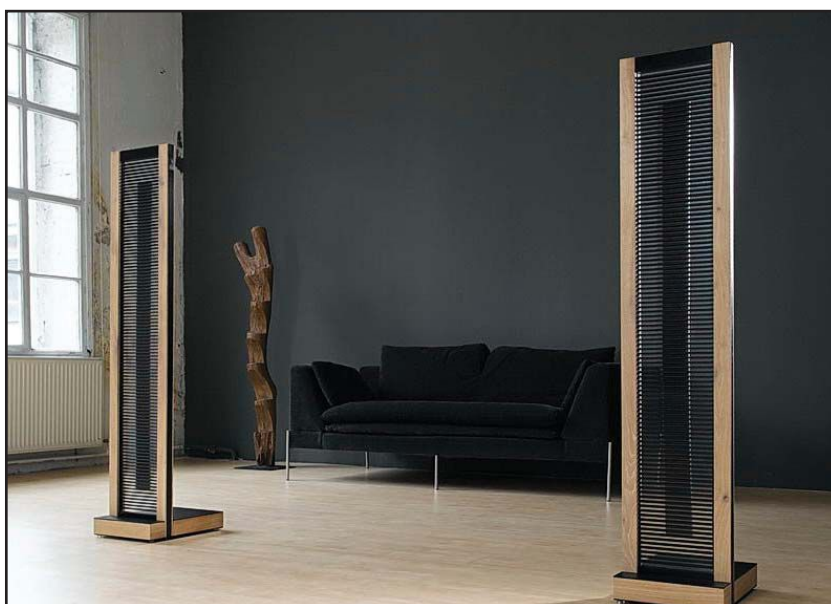
An Interview with Zoltán Bay, Creator of the Bay Radial Speaker

By
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With 35 years of audio experience, Zoltán Bay has now reached a new level of audio recognition with his newly patented design called the Bay Radial Speaker. The design has catapulted him from the solitude of his Hungarian village of Tiszaalpár to the limelight at the Association of Loudspeaker Manufacturing and Acoustics (ALMA) International event in Las Vegas, NV. The loudspeaker driver developed by Bay has garnered praise for a good reason. It has enabled renowned musicians to relive their own performances and has generated a months-long waiting list of enthusiasts signed up for an audition. We recently asked Zoltán Bay about his past, present, and future.



Zoltán Bay has reached a new level of audio recognition with his newly patented Bay Radial Speaker (BRS).



Bay has developed a number of high-end speakers during his career. This is the Bayz full-range electrostatic loudspeaker.

SHANNON: Tell us about your background. How did you get into audio engineering?

ZOLTÁN: It was a family influence early on. My uncle was a world-famous physicist. He pioneered radar astronomy by measuring the Earth-Moon distance using a radio locator. We have him to credit with our currently approved standard definition of a meter, and he even participated in the Apollo 11 program. My father designed vacuum tubes, amplification systems for movie theaters, and microphones. I would often talk with him about his patents and various technical solutions. In the 1960s, he invented a top-notch dynamic microphone that was suitable for mass production and became widely used throughout the Eastern Block due to its famous reliability and superb sound quality. The BEAG MD 14 model microphone was a favorite among news reporters and correspondents, and was even fitted to buses.

Apart from this family inspiration, I have always been an avid music fan. Music has always played a

major role in my life. I learned to play the piano, and I practice regularly to this day. I was also interested in medical science, however, so I studied electronic medical instrument design at the university. My diploma project was a hi-fi amplifier, though.

I got this idea of a novel method for adjusting the bias of the output stage of a power amplifier. I had been taught that the bias of the output stage had to be fixed, but I thought other things were more important. Contrary to a widespread misconception, the chief characteristic of Class-A operation is not high-current biasing, but the fact that the output transistors never shut down during operation. I could not wrap my mind around the notion of why high current should be the only way to prevent the output transistors from shutting down during operation. I came up with a solution for preventing that from happening even as the bias current is kept at a low 30 mA. This so-called “sliding biasing circuit” then became the subject of my thesis and my first patent.

While I was studying amplifier topologies at the school library, I was approached by the science director of the college, Dr. Attila Pócza, the foremost amplifier designer in Hungary at the time, with decent qualifications. He wanted to see the professional journals I was reading and asked me about the project I was working on. I had been an instrumentation major while he was heading the Department of Telecommunications, so we had never met. At the end of our exchange, he offered to be my advisor if I chose this amplifier as my diploma project. He was the one who set me on a path based on engineering principles that are still considered cutting-edge in analog amplifier design, even though this happened 35 years ago. In the implementation phase, my amplifier design proved itself because of its unique specifications—using a small amount of feedback (20 dB) but with low distortion and an impressive slew rate (greater than 100 V/μs).

The amplifier I built was quite possibly one of the best amps of its day. It surpassed Matti Otala’s world-famous amplifier in terms of specs, and even beat it in subjective listening tests. I have always followed the principle of not walking a well-trodden path. If I had, I would not have been able to achieve more than others had already achieved. I have always wanted to transcend the prevailing state of the art. This ambition is all I have and all I follow. It is my life. And life is full of interesting turns, indeed. Dr. Pócza is now retired in Australia, racking up the hours listening to an audio system I designed. Over the years, I gained a lot of experience from subjective listening tests in addition to the inevitable measurements. I am convinced that, without this kind of experience,



it is impossible to design any audio equipment worth its salt.

After college I spent some two years measuring vibration on various machinery and studying thick-film hybrid electronics. Meanwhile, I applied for and was granted a patent for a medical instrument. In tandem with a few teammates, I designed an instrument for dentistry for analyzing the dissolution of implants *in vivo*.

At the age of 27, I struck out on my own, devoting all my energy to engineering and manufacturing high-end equipment. I am 60 now. I have always worked alone, in hermit-like seclusion. I have never needed to advertise. My business is strictly word of mouth, and it has never failed to support my family. I live like a shoemaker who only makes bespoke shoes, and making a living doing it. I have never considered production lines. I am only interested in engineering and design.

Bayz cd player preamp adds another link in the audio component chain.

SHANNON: How did your products evolve over the years?

ZOLTÁN: I started out only making amplifiers, but when I tried to sell them I realized I could not afford



One of Zoltán Bay’s audio system designs.



The Bay Radial Speaker is actually a cylindrical membrane surface capable of changing its diameter.

to ignore other components in the chain. No matter how good my amps were, some orders fell through because the other links in the chain were so weak that they could not reveal the true strengths of those amplifiers. This drove me to design cables at first, then loudspeakers and media players, all to high-end standards. Once I had made a complete chain of my own, everything fell into place.

SHANNON: Tell us about your work process.

ZOLTÁN: I build equipment to order exclusively. I have never made more than two or three samples of the exact same concept. From industrial design to circuit boards to analysis I do everything myself, and I offer a lifetime warranty on everything

that leaves my workshop door. Starting in the 1980s, I realized the importance of computer-aided design and simulation, and I have devoted a lot of time and money to this aspect of the process.

By the time I finish a product, I know what I need to change to make the next one even better. The span of 35 years has been long enough to amass enough experience to always come up with a superior product.

SHANNON: Which of your products are you most proud?

ZOLTÁN: There are two. Each represents similar lofty standards of innovation (although the rest are nearly up there, too). One of them is my power amplifier. It has a number of solutions not employed by anybody else but, more importantly, its sound quality also benefits from a rigorous and self-consistent application of the theoretical knowledge I have accumulated over the years. When my friend Sven

Boenicke (of Boenicke Audio) auditioned this amplifier and he said, "It really did transform our speakers into new instruments—I could hear things from them I had never heard before from a speaker at all!"

The other one is the Bay Radial Speaker (BRS, www.bayz-audio.eu). I had been designing electrostatic panels, including a full-range one, which received a special technological award in the industrial design category, which is a very high distinction in Hungary. Then I began to entertain the notion of a driver that would radiate sound in all directions, around a 360° full circle. Now, this would be a major step forward in sound reproduction, I thought. It came naturally to first test this on the electrostatic principle. I succeeded in developing a radial driver, which reconfirmed that I was on the right track. In the end, however, the problems, logistics, and costs of an electrostatic design forced me to abandon this line of inquiry.

Then I began to think about ways to implement radiation in 360°, using a magnet and a coil in a dynamic loudspeaker system. I felt the idea had immense potential, so I put all my eggs in one basket. It took me eight years, but I think I managed to develop a revolutionary concept and a revolutionary step forward in sound. I applied for a patent in 2011, and I got it in the United States, Japan, EU, and China. The process in India is still underway.

SHANNON: Can you tell us more about the design of your BRS?

ZOLTÁN: The driver is entirely different from the dynamic dome, ribbon, and other magnetic designs known today. It is not a take on any of these tweeter paradigms. The membrane (radiating surface) itself is a vibrating cylinder with a variable diameter. This enables a large driver surface, up to 50 or 100 greater than the radiating area of a regular tweeter. This means that the energy transmitted to the driver space is far less concentrated, which makes for a much more relaxed and pleasant listening experience to the human ear.

The unorthodox design of the voice coil provides significantly improved ventilation and thus better power handling and lower inductivity compared with conventional solutions. In short, the driver behaves much less as a reactive load.

Under subjective listening tests, the driver delivers on all the sonic expectations, such as realistic imaging in precisely delineated space and a nearly holographic soundstage, with instruments far removed from the plane of the speakers when appropriate. It's the end of the head-in-a-vise syndrome. The stereo from



This is one of Bayz hand-crafted Chaconne power preamplifiers.

this speaker is completely immune to sideways movements of the head.

A surprise benefit of the 360° radiation pattern is its ability to preserve the attributes of stereo space even when the speaker is placed directly against the wall. It is common knowledge that the best speakers in the market today must be set up at least two or three feet out from the front wall in order to throw a spacious stereo field. Therefore, the suitability of the BRS to be used right against the wall makes it an excellent satellite speaker because it is not constrained by near-wall placement in its ability to portray an amazingly convincing soundstage.

SHANNON: What kind of feedback have you received to the BRS?

ZOLTÁN: Professor Ernő Sebestyén, the former head of the Violin Department of the Munich University of Music and Performing Arts, is a high-end freak. He has worked with some of the greatest music legends, including Leonard Bernstein, George Solti, Lorin Masel, as first violin, and concert master. He has bought several systems from me and once wrote in my guest book that he thought I made the best loudspeakers in the world. He said, "I had never heard sound reproduction so perfect before." The last time I held a demonstration in Munich, he invited his violin students to show them what this driver can do. They only had the highest praise when they commented on it in the guestbook. I met Ernő more than two years ago, and we have been in touch ever since. This is usually how it works with my customers.

The current president of ALMA International, Dan Foley, has visited me on two occasions. It started as a chance encounter. What happened was he sat next to a friend of mine on an airplane. They struck up a conversation. He talked about his line of business, and my friend told him about me and my patents. This piqued his curiosity, and he made an appointment with me to audition the BRS. Foley has been in the business for 31 years, but he told me he thought the BRS "tweeter is among the best I have ever heard and it may very well be the best in the world today." I am really grateful to him. It was his professional input that helped me make an appearance at the ALMA International event (AISE) in Las Vegas, NV, where he personally introduced me to the representatives of the trade.

This was how I got involved with a wonderful community. I made the acquaintance of several professionals of excellence, the "crème de la crème" of the trade, who helped me all along to make my demo a success. In addition to Foley, I must specifically mention Mark Beach and Bob Young, but everyone



Zoltán Bay showcases his Bay Radial Speaker at the High End Show in Munich this past May.

else was very welcoming and helpful. The event will remain in my mind as one of the finest memories I have ever had. The ALMA event was instrumental in fostering my ambition to sell manufacturing rights to my BRS. I am now in talks with several companies.

SHANNON: Which skills do you think made the greatest contribution to your success?

ZOLTÁN: I have 35 years of experience in the field behind my back, and this shows in both the design and the actual sound. I also have the ears to hear things you cannot measure with instruments. This is a talent which I have managed to hone into a skill. I never cease to seek radically new solutions and I have many fresh ideas. I simply enjoy challenging conventional wisdom.

SHANNON: What are your plans for the future?

ZOLTÁN: The ALMA event gave me great impetus. I had many inquiries about the BRS license, and talks are under way with several potential partners. I hope the BRS will find its way to many people and break new ground in serious music listening in the field. And I have several other exciting ideas on the back burner. I trust that more than one of them will be implemented and eventually become a mature product. 



Industry Calendar

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

September 1–6, 2017

IFA Berlin (The International Funkausstellung)
Berlin ExpoCenter City, Messedamm 22, Berlin,
Germany
www.ifa-berlin.com



IFA Berlin (The International Funkausstellung) is probably the most important annual event for the consumer electronics industry. If the CES in Las Vegas, N.V., is the place where future trends and innovation are highlighted and get the most attention, the IFA Berlin show is where many of those products and brands make it or break it. Audio at IFA is mainly focused on personal/mobile devices and the more accessible and lifestyle-oriented products. A wide range of products, solutions, and presentations make a visit to IFA 2017 worthwhile. IFA 2016 featured more than 1,800 exhibitors and attracted 240,000 visitors, with the majority of trade visitors coming from outside Germany: 68,000. The 2016 floor space was already the largest in the show's history. Expect the 2017 edition to be even larger!

September 6–9, 2017

CEDIA 2017

San Diego Convention Center, San Diego, CA
<http://expo.cedia.net>



CEDIA is undergoing several key changes including the sale of the trade show to Emerald Expositions (www.emeraldexpositions.com), followed by a 15% reduction in the organization's staff, which will make the San Diego edition of the traditional residential technology show a pivotal moment. Shortly after the CEDIA show was transferred to Emerald Expositions—the largest operator of business-to-business trade shows in the United States—its CEO, Vin Bruno, and its VP of Business Development, Ron Fleming, resigned. CEDIA 2017 brings together more than 18,000 home tech pros and over 500 exhibitors. Currently, there is a positive momentum in the industry with house sales and renovations growing again in the US, combined with strong technology trends and innovation. Now with the professional coordination of an experienced show promoter, CEDIA members and the organization's board, representing 3,700 member companies worldwide and serving more than 30,000 industry professionals, will have the opportunity to restructure and relaunch its future, in a positive business environment.

September 8–10, 2017

**Audio Engineering Society (AES) 2017
International Conference on Automotive Audio**
Burlingame, San Francisco, CA,
www.aes.org/conferences/2017/automotive



Over the last 20 years, automotive audio has dramatically changed from being standard mono sound in one full-range loudspeaker to a true multichannel playback system fully integrated and adjusted to the specific car. Today, some of the most advanced sound technologies are being developed and applied in automotive audio. The conference will provide an overview of the present state-of-the-art designs and address many of the new scientific disciplines involved in this still-emerging field. Automotive Audio involves combinations of otherwise complementary scientific fields such as power electronics, bus systems, loudspeaker drive units, signal processing (DSP), acoustics and vibrations, and more.

September 12–14, 2017

GSMA Mobile World Congress Americas
Moscone Center, San Francisco, CA
www.mwcamericas.com



www.gsma.com | www.ctia.org

The 2017 GSMA Mobile World Congress Americas, promoted in partnership with CTIA (Cellular Telecommunications Industry Association), is the new mobile industry event, following the overwhelming success of its Barcelona annual trade-show. Under the theme "The Tech Element," Mobile World Congress Americas is expecting 30,000 professionals from across the mobile industry and adjacent industry sectors to attend. The event is an interesting opportunity for electronic component, audio platform and software vendors to find exciting business opportunities. With more than 1,000 companies already confirmed to participate in this inaugural event, it will be an exciting opportunity.

September 14–19, 2017

IBC 2017

RAI Amsterdam, The Netherlands
www.ibc.org



The world's leading Electronic Media and Entertainment Event continues to get bigger and better every year. IBC 2017 will once again attract large numbers of visitors and almost 2,000 exhibitors, as the most international event for the electronic media and entertainment industry—in parallel with the National Association of Broadcasters (NAB) convention held in Las Vegas, NV, every April. The show continues to grow, with the production on the rise, the exciting transition to 4K/8K images, the technology migration towards IP, networking and the cloud, and the media landscape becoming increasingly Internet-based and on-demand. Audio exhibits are, most of them, located in Hall 8, and include a unique combination of radio technology suppliers, microphone manufacturers, recording and production solutions. For those interested in learning about the latest trends in audio streaming and digital radio, including multichannel and binaural distribution, the Amsterdam is indeed a valuable opportunity.



Article Update

Michael Kiwanuka's article "Current Feedback and Voltage Feedback Fallacies" (*audioXpress* June 2017) elicited comments from Walt Jung, with supporting signatures from a number of well-known designers. Jung has posted the comments at his website: www.waltjung.org/waltsblog/category/current-feedback-amplifiers. Corrections have also been posted on the audioXpress website (www.audioxpress.com/page/Errata-Corrections-Updates).

Kiwanuka's article revolves around the question whether the term "current feedback" is correctly applied to a "CFA opamp." Kiwanuka recalls the term "current feedback" used to denote a feedback configuration where the feedback signal was derived from the output load current. With the advent of the "CFA architecture," this term has obtained a different meaning, to indicate not how the output signal sample was obtained, but rather how the output sample is thought to be fed back. This happens often in fast-paced technological fields; but you can't turn back history so it has to be accepted.

Kiwanuka explains (correctly) the historical meaning of the term, but then proceeds to argue that this is not correct, based on the definition of current feedback not in the context of a CFA but when viewed from the context of feedback configuration classifications. But as noted, the terms "CFA" and "VFA" in the context of "CFA opamps" have a different, widely understood and accepted meaning and as such are, by definition, correctly applied.

Finally, we regret that some editing errors crept into the article. Specifically, Equation 1 was printed incorrectly. The original equation was of the correct form $V_{out} = i_N * R_{eq} / (1/sC_{eq})$. Also, Figure 2 was inadvertently replaced by a copy of Figure 6. The correct Figure 2 appears on our website.

We offer our apologies to Kiwanuka and our readers. A correct version of the article is available on the audioXpress website (under Theory Articles), together with the full thread of the exchange between Walt Jung, et al. and Michael Kiwanuka.

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