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Sound Control **Progress on Calculating Edge Effect of Acoustical Absorption**

By Richard Honeycutt

Software Review **Comparing Audio Editing Tools Software Roundup (Part 1)**

By Fernando Rodrigues

Fresh From the Bench **Naim Audio Mu-so Qb Wireless Music System**

**By Oliver A. Masciarotte
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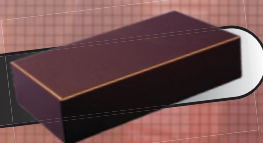


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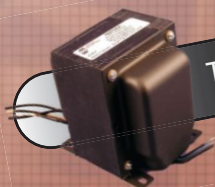
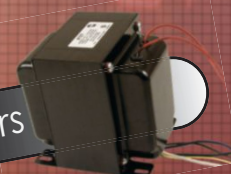


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Embracing All Possibilities



In this edition of *audioXpress*, we feature a thought-provoking article from Gary Eickmeier that rethinks the listening experience and “embraces” room acoustics—according to what he defines as the Image Model Theory (IMT). The proposed design—the Image Model Projector—proposes drivers firing in four angles to explore the effect of reflected sound, as opposed to the direct-field of conventional speakers.

In this same edition, our speaker-building contributor, Ken Bird, details a DIY project for “Omni-Directional Coaxial Towers,” which deliver “natural sound propagation in every direction.” A concept, that the author says was inspired by the current trend for “tiny 360° smart speakers invading modern homes.”

I always liked to audition omni-directional speakers at audio shows and I have pleasant memories of listening to German Physiks (I try not to think about the price...). But similarly, I was pleasantly surprised by the way the Fujitsu Ten Eclipse TD712z full range—and direct firing—speakers “disappear in the room” and create what can only be described as a natural sound field that plays well in the untreated and less than ideal rooms I’ve auditioned them. So clearly, there’s a lot to be said for different approaches when it comes to listening rooms and speaker design.

When we look at what’s happening with those “360° smart speakers invading modern homes,” to which Bird refers, we are confronted with the harsh reality of a modern consumer who would never sit still in the symmetric triangle of the traditional stereo hi-fi experience, and instead listens to music all over the home, with multiroom systems now provided by relatively small—and often portable—speaker systems. Those new speakers are effectively fully integrated systems, containing amplification, processing, and wireless network connectivity, many of which explore multiple drivers and bold new waveguide designs.

An excellent example of what a prestigious high-end audio manufacturer can do in this domain is illustrated in our review of the Naim Audio Mu-so Qb wireless music system. As the measurements attest, a system designed to be a pleasant single source for music listening in real modern home environments, designed in a way that actually plays well with what we normally consider less than ideal acoustics and placements.

In this fast-changing environment, many of those out-of-the-box and more esoteric—and even “heretic” for some—speaker design approaches, suddenly become encouraging fields of exploration to meet a new paradigm.

And what’s truly remarkable is how the combination of technology and advancements in many other fields now contributes to that new model of entertainment experiences. The convergence of advancements in digital processing, connectivity, cloud computing—and computer simulation tools—allows for completely new and bold designs. No longer conditioned by interconnect cables, and considering the fully integrated signal-chain, speaker designers can leverage high-quality wireless audio (don’t disregard the potential of WiSA technology just yet, and consider the promises of Bluetooth 5 and Mesh technology), and real-time acoustic compensation, adding to traditional disciplines. Thanks to the large effort currently being applied on smart speakers and voice applications, audio designers will effectively benefit from an entirely new set of technologies and tools to create what was once the material of dreams.

As illustrated in another of this month’s feature articles—a report on the new Axiom Class-D audio amplifier controller chip—things that have been long envisaged, such as the ability to listen to the output of an audio system and apply multiple feedback loops to correct all sorts of unwanted (d)effects, is now possible with unnoticeable delay, thanks to advancements in processing power and speed. The same is now true also with real-time acoustic compensation of a complete speaker system in modern “smart speaker” designs. Given the massive resources being applied by the largest companies in this domain, those tools and technologies will be available for any manufacturer to explore. Imagine what we can do!

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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brain as the original Mu-so. Kent Peterson puts the Mu-so Qb to the test confirming that the device sounds as good as it looks.

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Stuart Yaniger measures and tests this single channel preamp, meant to boost the signal of a low output microphone to levels compatible with normal microphone inputs.

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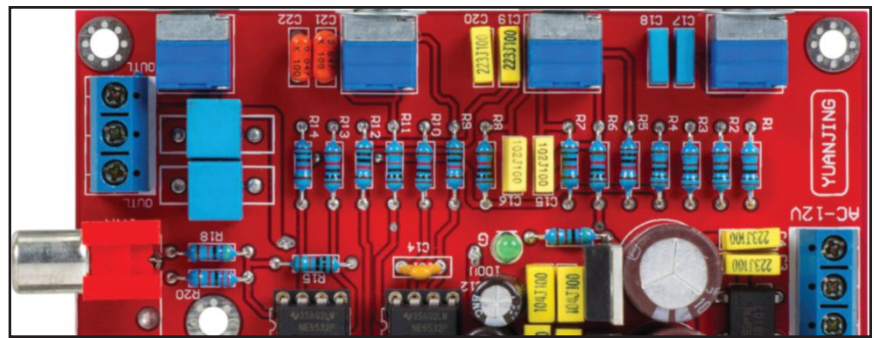
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Vinyl vs. CD (Part 3)

Exploring the Effects

In the third article in this series, we will continue looking at second harmonic to fundamental energy ratios for several different playback setups using a media player hosting VST plug-ins. I continued using six measurements for both the fundamental and the second harmonic frequencies to insure reasonable repeatability.

By
Ron Tipton
(United States)

Each measurement set occupies one spreadsheet column for a total of 12 columns for each value in **Table 1**. The =SUMSQ operator finds the sum of the squares for each column. The six sums for the fundamental frequencies are added and the square root is found. The same is done for the second harmonic frequencies. The RMS ratio is simply the second harmonic square root divided by the fundamental square root because the number of values in each column drops out of the equation if all columns are of equal length.

The Measurement Setup

Figure 1 shows my setup's block diagrams for these measurements. I inserted amplifiers as shown to set the recorded voltages well above the noise level. The 44.1 kbps, 16-bit music was streamed from a computer using foobar2000 with a playlist of the first two tracks of Diana Krall's *Quiet Nights*.

I intended to use the JRiver Media Center (JRMCM) for playback because it's available for both Windows and Mac and hosts VST plug-ins but I got lots of "artifacts," clicks, pops, squeals, and so forth from

Music Source		Band-Pass Filter Center Frequencies					
		360 and 720 Hz	440 and 880 Hz	535 and 1070 Hz	625 and 1250 Hz	760 and 1520 Hz	880 and 1760 Hz
1	LP with TDL 4061B vacuum tube RIAA preamplifier	0.856	0.794	0.716	0.554	0.622	0.690
2	Shattered Glass Audio SGA1566, no bass or treble cut or boost, EQ position Post, stereo on	0.687	0.806	0.657	0.479	0.499	0.607
3	Voxengo Warmifier with Vintage Brightness preset, Mode = Lvl.Dep, Model = 12AT7, Oversampling = Auto	0.720	0.821	0.633	0.463	0.519	0.641
4	Amperex Bugle Boy Long-plate dual triode from the CDSoundMaster Vintage Tube Collection	0.735	0.820	0.618	0.471	0.521	0.592
5	foobar2000 without a VST plug-in	0.732	0.724	0.601	0.426	0.480	0.571

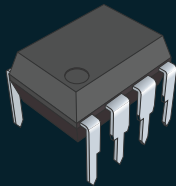
Table 1: The results of my measurements are shown in table format. Each column of numbers represents the RMS second harmonic to fundamental ratio. That is, the ratio of the energies under the two curves. Row 1 is the reference line, an LP with an RIAA vacuum tube preamplifier and Row 5 is the foobar2000 without a plug-in. The highest column value is shown in red and the second highest is shown in green.

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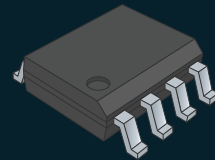
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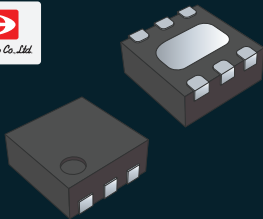
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Figure 1: The measurement system in this block diagram occupies two 6' tables, not including the stereo listening Line amplifier and Power amplifier. The Ferretti power amplifier has a vacuum tube front-end and MOSFET current drivers for the loudspeakers—an excellent design.

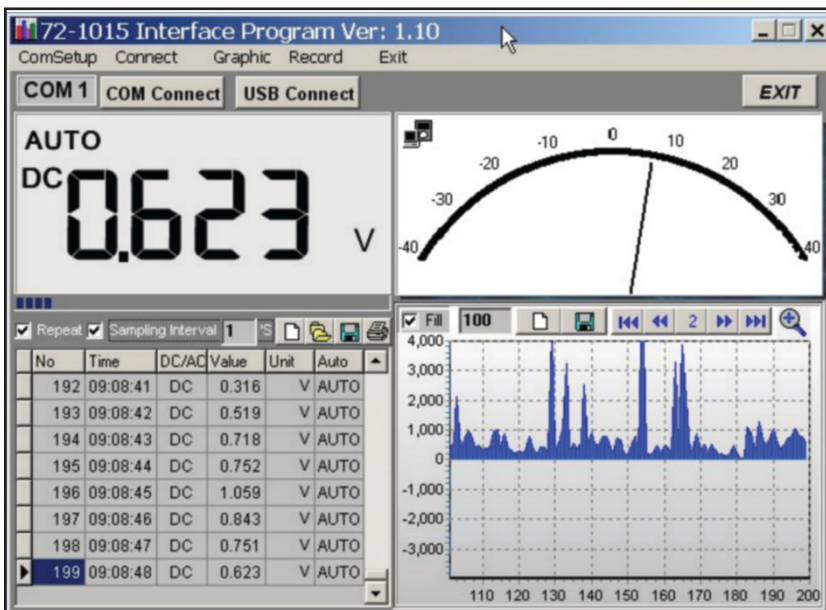
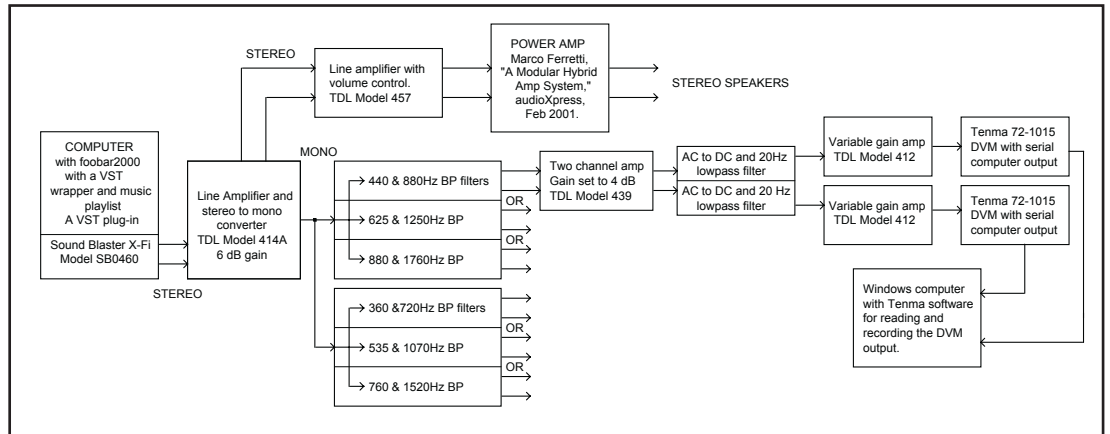


Photo 1: A screenshot of the Tenma DVM control and recording screen. The measured and recorded values are displayed in the left panel with the voltmeter and running-time graph on the right. The meter actually measures and records to a full scale of 5.999 V. A recording is started and stopped by clicking the USB Connect bar.

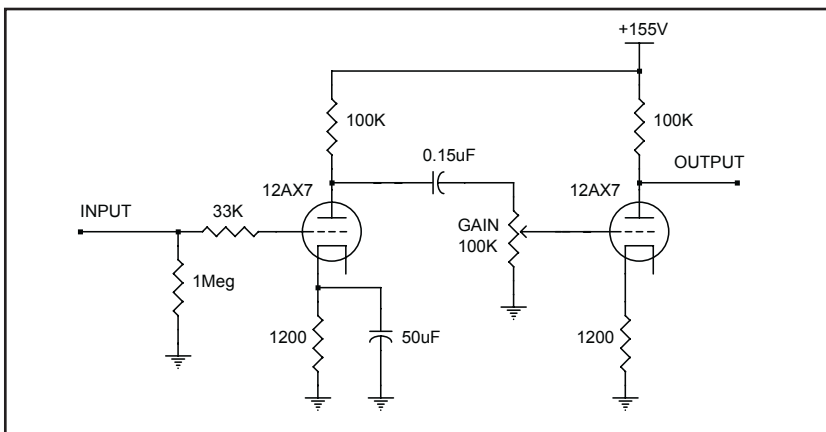


Figure 2: The Shattered Glass Audio SGA1566 plug-in's dual triode amplifier has a lower than normal plate voltage making it easy to overdrive. But this feature allows for its harmonic production.

versions 22 and the newly released 23, with the same settings that work fine in foobar2000 and the Reaper DAW. foobar2000 is free—available for Windows but not Mac. It does not host VST plug-ins directly but there are two “wrapper” programs: George Yohng’s VST wrapper and VST 2.4 Adapter and both work fine.

Either or both wrapper files should be copied or dragged and dropped to the foobar2000 Components list. Click File and then Preferences. Click the wrapper file name, then Apply, and say “yes” to restarting foobar2000. The right-hand panel displays the installed components. To install your VST plug-in, click File, Preferences, and VST Plug-ins. Click Add and then browse for the plug-in. Click its name and it should appear in the VST Plug-ins panel. I’ll continue this example later when I load a plug-in.

As before, the now six fundamental and second harmonic filter outputs are connected a pair at a time—fundamental and second harmonic—to a pair of full-wave detectors followed by active low-pass filters with a 20 Hz cutoff frequency. A pair of identical true RMS voltmeters (Tenma 72-1015), set to measure DC voltage, is connected to each detector output. The playlist is played six times for each filter and the Tenma outputs are recorded and saved as Microsoft Excel .xls files. **Photo 1** shows a screenshot of the voltmeter control and recording program running in Windows.

The numbers in Table 1 are RMS Energy Ratios, with higher values indicating more second harmonic energy.

The Band-Pass Filters

As shown in Figure 1, I used two circuit boards each with six band-pass filters for the measurements. One board has the 440-880 Hz, the 625-1250 Hz, and the 880-1760 Hz filter pairs. With the 360-720 Hz, the 535-1070 Hz, and the 760-1520 Hz pairs on the other board. All the filters are active six-pole Butterworth Multiple Feedback. I kept

the percentage bandwidth very close to 1.25 for all the filters, which worked well. A typical frequency response curve is shown in Part 1 of this series (*audioXpress*, October 2017).

The Detector (AC to DC Converter)

The FullWave detector circuit was published in a National Semiconductor Application Note. I designed a circuit board with the detector's output going to an active four-pole, Butterworth low-pass filter with a 20 Hz cutoff frequency. The circuit diagram was printed in Part 1 of the article series and is included in this month's Supplementary Material located on the *audioXpress* website (www.audioxpress.com) in the fullwave.zip folder.

LP BaseLine

The first row in Table 1 is the LP baseline. You will notice that all but one of these values is red meaning they are largest in their column. The green numbers are the second largest in each column. It seems that if you want "true" LP sound play an LP, but the plug-ins come very close.

VST Plug-Ins

I looked at quite a few "warming" plug-ins and picked three for my measurements. The first one is free, the second is \$39.99, and the third is \$99. These three, and several of the others I found, use some form of vacuum tube simulation. I've made a list of plug-ins I looked at with name, web address, price, and a brief description. You can find the information in the Supplementary Material as VST-Plugins.pdf.

Shattered Glass Audio SGA1566

This free VST plug-in is a software simulation of the twin-triode amplifier diagrammed in **Figure 2**. It has some interesting features, which I'll explain by quoting a brief paragraph from its User Manual. "Supply voltage is 155 V, which is substantially lower than the usual 200 V+ voltage supply. The lower supply voltage translates into less clean headroom thereby making the preamp easy to overdrive. Gain potentiometer is 100 k Ω , which is of the same order as the output impedance of the first stage. A consequence of this is the reduced gain of the first stage. Usually this potentiometer would have a value of 1 M Ω so it does not affect the gain of the first stage."

This plug-in also has a EQ section with low- and high-pass shelving filters. I wanted only increased even-order harmonics so I left the EQ unused by leaving the control knobs at their default settings of "5."

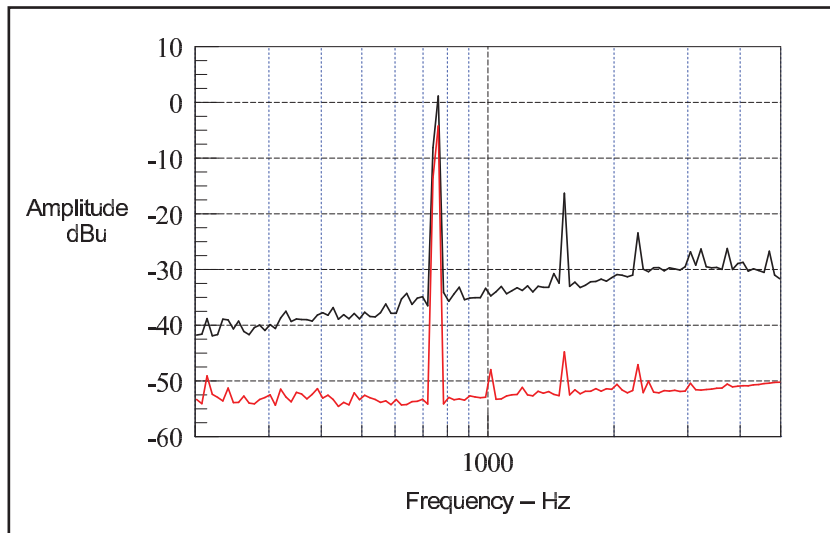


Figure 3: This is the graph I produced trying to understand (guess at) an SGA1566 setup. The black line: Input at 1:00, Gain = 7 and Output = 7. Red line: Input at 3:00, Gain = 8, Output = 3. This verified that the plug-in is easy to overdrive so I used an Input at 9:00 (very low) for the measurements.



Photo 2: This foobar2000 screenshot shows the George Yohng's VST wrapper and the SGA1566 in the Active DSPs panel. Clicking SGA1566 and then the Configure selected bar displays the plug-in's screen. I used the setup shown here for the measurements.

There are no presets and little guidance in the User Manual on setup, so I experimented. I tried two different control settings with an input signal at 750 Hz from another computer running a copy of the free audiogenerator.exe program. Using yet another computer running TrueRTA, I captured the two curves shown in **Figure 3**. Based on what I

learned, I “guessed” at a trial setting for SGA1566.

So, back to my example, and getting this plug-in installed in foobar2000. Click File, Preferences, and VST-Plug-ins. Click Add and then browse for sga1566.dll. Click the name and it should appear in the VST-Plug-ins panel. Click File, Preferences, and under Playback, click DSP Manager. If you haven’t already done so, click the wrapper you want to use and then the left-pointing arrow between the right and left panels. This moves the wrapper into the Active DSPs panel. Repeat this for sga1566. Select sga1566 and click Configure Selected. You should have a screen that looks similar to the one shown in **Photo 2**. Adjust the controls as shown (my guesses) and you are ready to play some music.

I used the 440 Hz to 880 Hz filter pair and got the RMS ratio of 0.806 as shown in Table 1. This was encouraging so I continued with this setting and was generally pleased with the results. Also, the music sounded more mellow than foobar2000 without a plug-in (see Table 1, Row 5). But I can’t guarantee this will work for all music selections. You may have to experiment or go to the next VST plug-in: Voxengo Warmifier.



Photo 3: The Voxengo Warmifier screen is shown with its knobs set to the preset values. The preset name, Vintage Brightness, is displayed in the top left corner. The Model, Mode, and Allowed Oversampling (AUTO in the upper right corner) are not included in the preset and must be set separately.



Photo 4: The selected dual triode name, Amperex Long Plate, is displayed bottom center. The Vintage Tube Collection screen is colorful and attractive but the knob’s small size, highlighting, and lack of calibration marking makes set up a chore. However, it works wonderfully. The light colored dot in the lower left corner is the Bypass switch and the red dot, lower right corner, is a clipping indicator.

Voxengo Warmifier

The Warmifier plug-in is not free, but its \$39.99 price seems reasonable for what it does. I read the control setup information in the User Guide (included in the Supplementary Material) several times but still found it confusing. I chose to use the Vintage Brightness preset with Mode set to Lvl.Dep. (the processing is dependent on the input level) and Model set to the 12AT7 vacuum tube (see **Photo 3**). This proved to be a good choice because the music sounded wonderfully mellow and the numbers in Table 1, Row 3 show a definite improvement over Row 5—foobar2000 without a VST plug-in.

The Vintage Tube Collection and Tube Booster

These are two different plug-ins, priced together and licensed as a bundle. The Collection plug-in simulates nine different dual triode amplifiers with each one described for its tonal qualities. It has just three controls: Input (level), Drive (amount of harmonic distortion), and Output (level). I didn’t use this plug-in for my measurements because the Tube Booster is described in the User Manual as: “It is a dynamic program with a complex process for adding a special harmonic saturation with little side effects to the frequency spectrum.” Might be difficult to measure anything.

The Amperex Bugle Boy long-plate is described in the User Guide as: “The most outstanding

characteristics of this tube is the combination of all elements one looks for in a tube. The high end, mids, lows, separation, harmonics, sense of space in sound reproduction, dynamics, sustain, stereo width, and microphonics..."

I chose this Amperex model and used the settings shown in **Photo 4**. I tried Drive settings of 9:00, 10:30, and 12:00. I thought I detected some slight artifact distortion at 12:00, so I backed-off to 11:00. This plug-in has a very attractive "face," but the small knob size and the highlighting make them rather difficult to set. I would prefer a larger Drive knob with some outside calibration marks.

However, the mellowness of this plug-in makes the music delightful, perhaps even more so than the Table 1 numbers indicate.

Final Thoughts

All three of these plug-ins enhance the music compared to foobar2000 without a plug-in as shown in Table 1, Row 5. I have not tried an A/B comparison, but I'm not sure I could reliably tell the difference between the LP and the plug-ins. Pleasant listening! 🎧

Project Files

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

Resource

Shenzhen Cavins Technology Co., Ltd.—Douk Audio, www.doukaudio.com

Sources

audiogenerator.exe | www.ringbell.co.uk/software/audio.htm

foobar2000 | www.foobar.org

George Yohng's VST wrapper | www.yohng.com/software/foobarvst.html

Shattered Glass Audio SGA1566 | www.shatteredglassaudio.com/product.php?id=104

Tenma 72-1015 DVM | www.mcmelectronics.com

TrueRTA | <https://trueaudio.com>

Vintage Tube Collection | cdsoundmaster.com/site/cds-software-online/vtccp.html

Voxengo Warmifier | www.voxengo.com/product/warmifier

VST 2.4 foobar adapter | <https://hydrogenaud.io/index.php/topic,84947.0.html>



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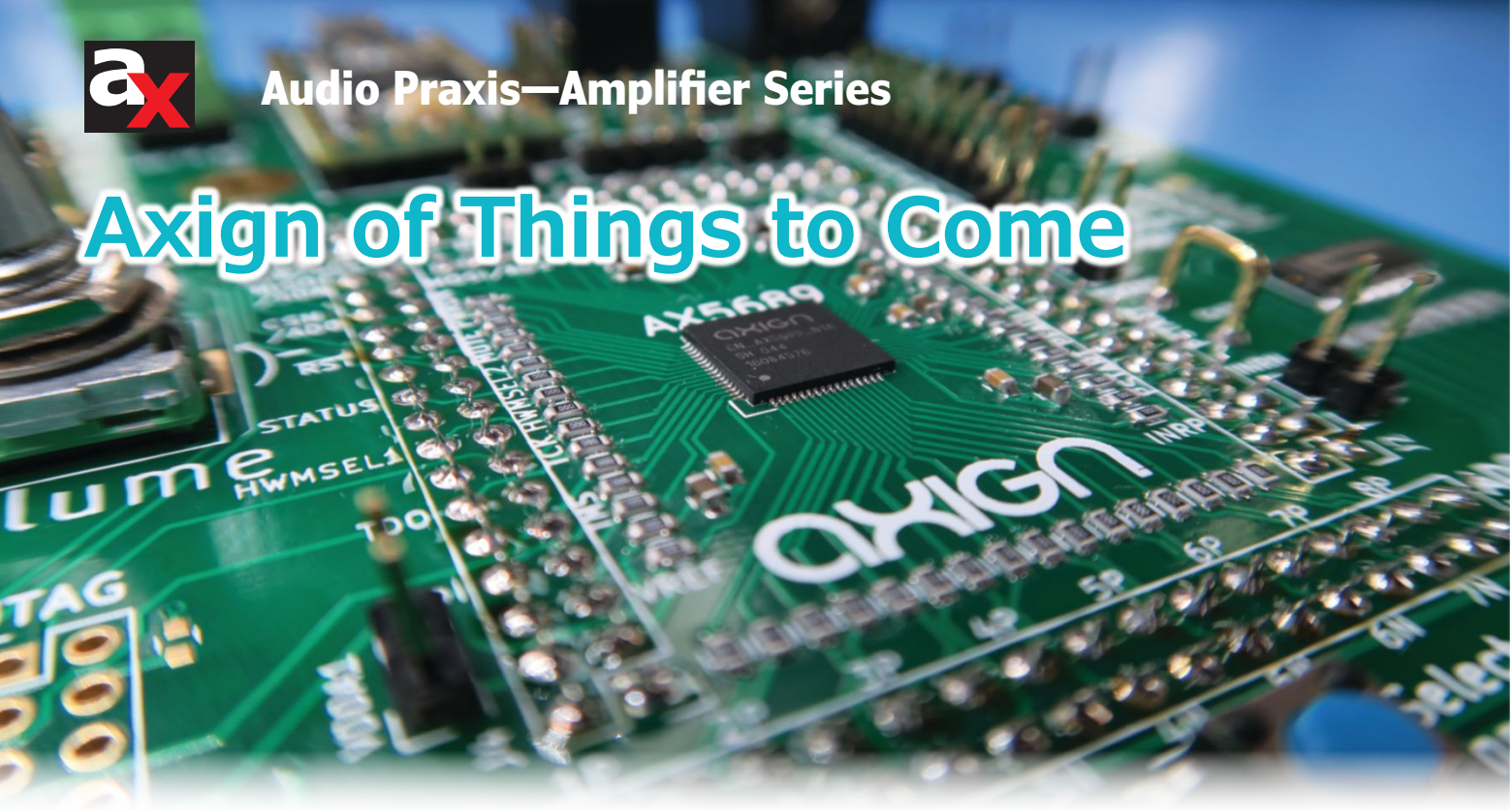
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Axign of Things to Come



Axign is a new company from The Netherlands, founded by former Philips and NXP employees who worked on the pioneering Class-D efforts. Established in 2014, their efforts have resulted in the development of the AX5689, a Class-D audio amplifier controller chip in a QFN package. The solution uses a digital control loop with feedback behind the output filter, across the loudspeaker terminals. It requires a digital input signal and suppresses all artifacts with a fifth-order digital feedback loop. *audioXpress* visited the company to find out more information.

By
Ward Maas

(The Netherlands)

A traditional application area for Class-D amplifiers is high-power amplification for professional use. Efficiency, weight, and reliability are determinant factors for such uses. They might be workhorses, but audio-quality-wise, they have not been good enough to run in the Kentucky Derby. In the past few years, however, the industry has seen the rise of high-quality Class-D amplifiers. Not only did they keep (and reinforce) their traditional strengths, but claims have been made (and confirmed) that they could also play a role in high-end audio applications. From that moment on, the entire game started to change.

The combination of high-quality and low cost is, as could be expected, irresistible to audio manufacturers. Most people listening to Class-D amplifiers, do recognize the qualities in the bass area, but doubt the merits in the mid and high regions. Checking a datasheet often reveals some Class-D artifacts in the distortion curves and some load dependent behavior. As we have said in previous articles dedicated to this topology, we saw

a dramatic change in performance as more Class-D manufacturers are able to minimize these effects.

So, the use of Class-D in the high-end market is now a reality. The new Marantz PM 10 flagship, powered by Hypex Ncore Class-D amplifiers, is one of the obvious examples. And for those who still have doubts, that's where the efforts of Dutch chip developer Axign can persuade even the most reluctant "traditional" analog diehard. To put it simply, Axign's new AX5689 chip lays a very high-speed feedback loop around a Class-D amplifier, transforming its performance to almost theoretical levels. Too good to be true? Let's see.

Challenges with Class-D Amps

Back in the days when Bruno Putzeys at Philips developed the first self-oscillating Class-D amps, these efforts were also used in their semiconductor branch (now known as NXP Semiconductors), specifically by the group that developed amplifiers for automotive applications. One of the key concerns in developing high-quality Class-D amplifiers is the

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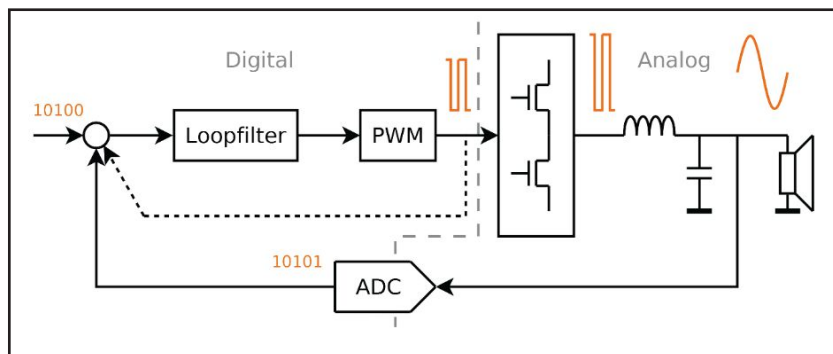


Figure 1: The AX5689 is a Class-D amplifier controller chip in a QFN package. It has a high speed digital control loop with feedback behind the output filter.

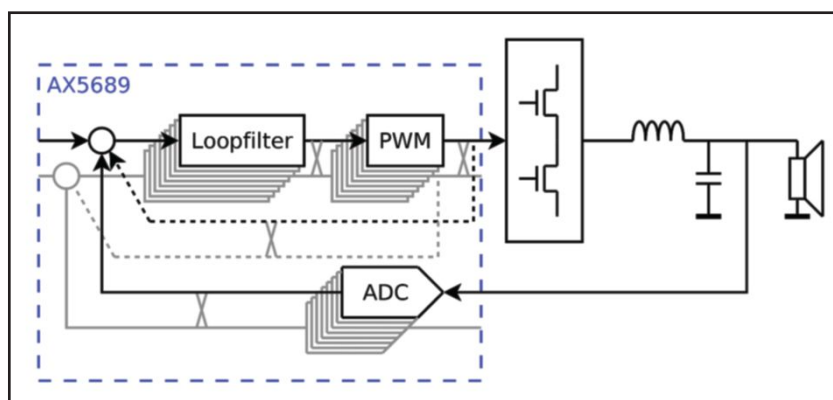


Figure 2: The AX5689 has eight identical programmable high-order digital loop filters, PWMs, and ultra-fast LLADCs. It is suitable to control eight loops in real time.

amount of obtainable loop gain without getting into stability problems. Often the obtainable gain is rather low. To avoid a load-dependent performance, a feedback after the output filter is also desired.

But, there are a few ways to tackle these problems. The use of nested feedback loops is not an easy path, but as more and more experience is gained with this kind of solution, the quality of Class-D amps has improved step-by-step over the last few years.

Another solution is to use a high-speed ADC, comparing the analog output signal with the digital input signal and then do a correction in the digital domain. However, such a solution requires a major development effort—an effort that could not be made by NXP Semiconductor at that time due to capacity limitations. The high-speed ADC was seen as the most critical development item and it was outsourced to Axiom-IC (now Teledyne-DALSA). Then, after an extensive market research and many discussions with potential customers, Jeroen Langevoort (formerly of NXP Semiconductors) took the project and development ideas and established Axign. Supported by Teledyne and Daniel Schinkel, who was strongly involved in the ADC development, the design of the AX5689 was realized.

The AX5689

The AX5689 is a digital amplifier controller that compares the analog signal at the loudspeaker terminals with the input signal (see **Figure 1**). All the disturbances in the loop (e.g., the power supply ripple and nonlinearity from the power stage and

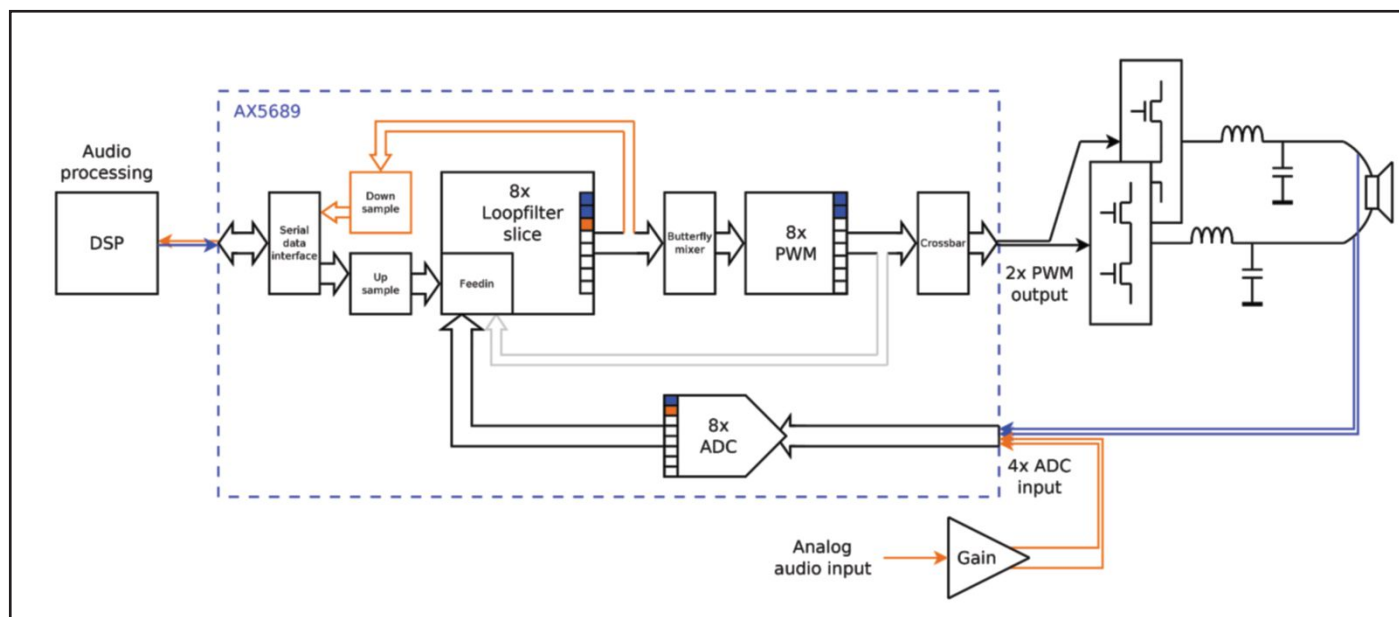


Figure 3: An application example shows that a complete audio system can be built with an AX5689. Adaptation to an active loudspeaker system is easily done.

reconstruction filter) are compensated in real time and suppressed thanks to the extraordinarily high digital loop gain. The analog signal at the load is converted by a high-performance low-latency ADC. The latency is in the order of 20 nanoseconds, giving a total of 200 nanoseconds to act for a bandwidth of 5 MHz.

The AX5689 has eight identical processing slices and eight ADCs, making it suitable to control eight loops in real time (see **Figure 2**). This enables a wealth of configuration possibilities (see **Figure 3**). This means the chip can be used for the Class-D controller function and for other periphery audio functions. One of the interesting possibilities is the option to parallel the sections and increase the already impressive 115 dB signal-to-noise ratio (SNR) to 124 dB, on set level. With the AX5689, silence becomes silence again.

To be clear, this is a Class-D controller not an output stage. It will control whatever Class-D output stage is used. It is even fast enough to include an already available controller in the loop. So how do you demonstrate the qualities of such a product? Well, it's rather easy. First, you take a reference design from the most well-known Class-D amplifier manufacturer and see how much you can improve that design. Second, you listen to the results and check how it sounds.

After a long time in development, the guys at

SNR	115 dB to 124 dB
THD vs. frequency	>0.001% (10 W, 20 Hz to 20 kHz)
Output noise	25 μ V (for a 2×100 W/4 Ω solution)
Channel separation	>100 dB (20 Hz to 20 kHz)
Output impedance	<2 m Ω (20 Hz to 20 kHz)
Frequency response -3 dB	DC – 35 kHz (20 Hz to 20 kHz flat, load independent)
IMD (19 kHz and 20 kHz)	below -100 dB

Axign ended up with the results shown in **Table 1**. The proof is in. But those results did not come automatically. At the moment, a fifth-order loop filter is used. Theoretically, a 56th order could be implemented. Not very practical, but it gives an idea of the chip's capabilities. So yes, it took some time before the expected results were realized.

Table 1: These are the results for the AX5689 Class-D audio amplifier controller chip.

Putting the Chip to the Test

For a real-world test, the AX5689 was used in combination with a few Texas Instruments (TI) and STMicroelectronics chips (the TAS5558, the TAS5624A, the TAS5342A, and the STA516B, respectively). Looking at **Figure 4**, we see the THD+N vs. Pout graph for 100 Hz, 1 kHz, and 6 kHz at 4 Ω for both the STA516B and the TAS5342A. The first part of the graph is determined by the noise floor. Then when the amplifier starts to clip, the distortion starts to rise. No traditional Class-D artifacts can be seen.

Figure 5 shows a direct comparison

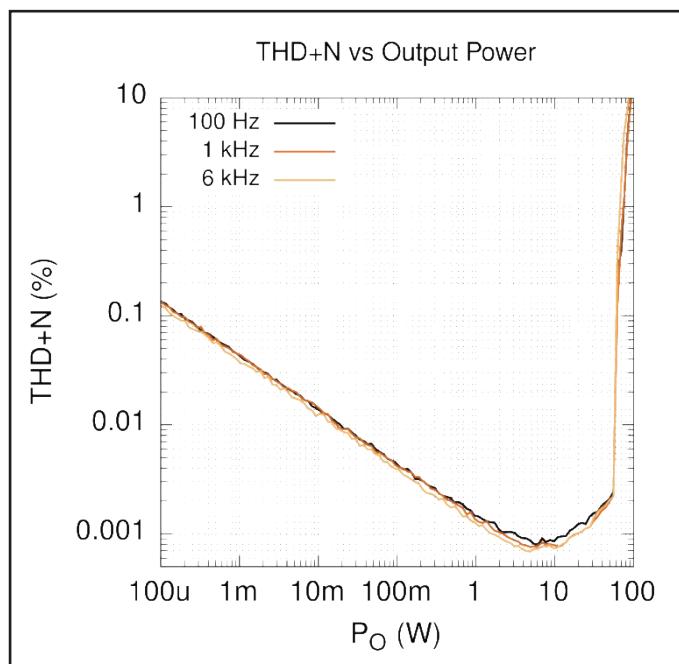


Figure 4: The AX5689 + STA516B or AX5689 + TAS5342A Pout in 4 Ω PVDD = 28 V, $2 \times$ LLADC per BTL, BD mod. A difference between the STA516B and the TAS5342A could not be observed.

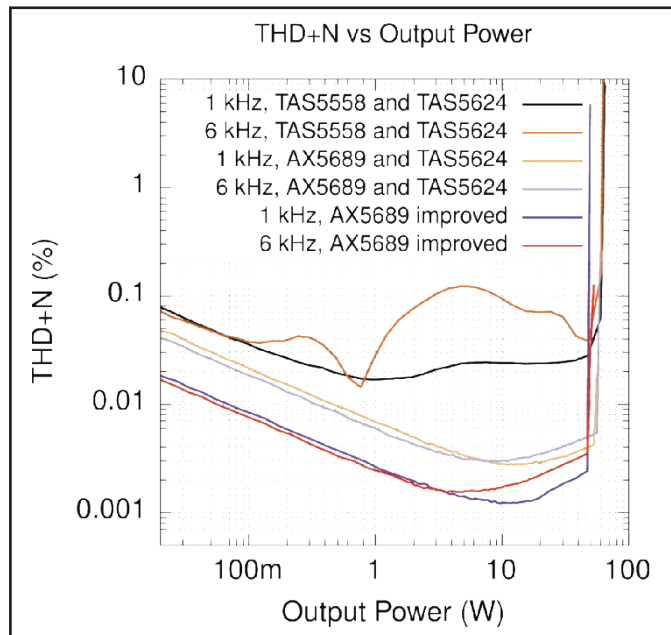


Figure 5: The improvements are clear. The AX5689 is a significant improvement over the TAS5558. Not satisfied with the initial results, the Axign team obtained even better performance after optimizing the chip's parameter settings.

between the TI output stage TAS5624A with a TI controller TAS5558 and the same TAS5624A with the Axign controller AX5689. Clearly, the AX5689 has better control over the output stage. After

the first test, the settings of the AX5689 were adapted, providing an even better performance. **Figure 6** reveals an interesting phenomenon. The less sophisticated TAS5342A performs slightly

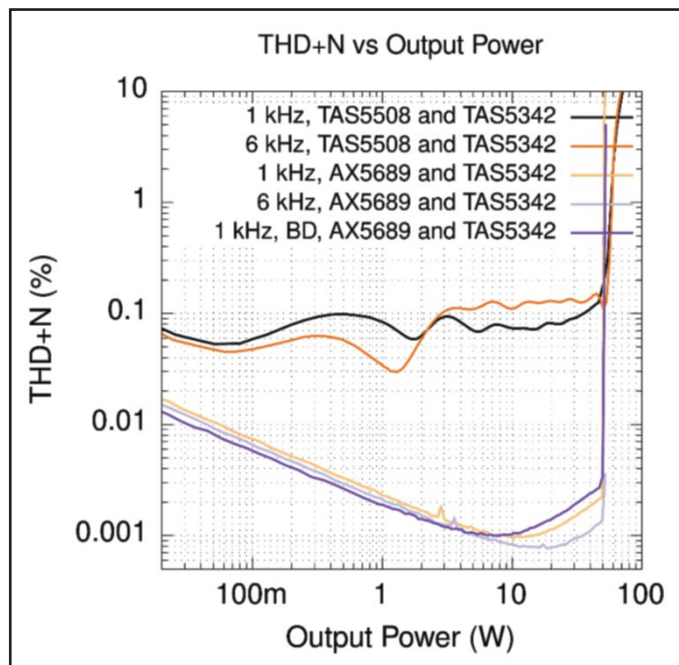


Figure 6: The team also made large improvements in the distortion performance as well. It is interesting to note that the TAS5342A performs better distortion-wise than the TAS5624A. Not having to “wrestle” with an internal feedback loop, the AX5689 performs better.

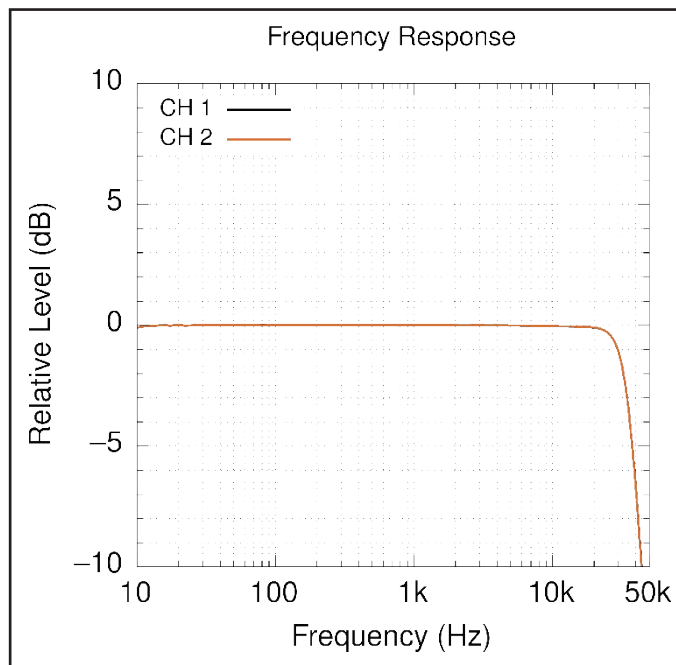


Figure 7: A straight line can be seen in this frequency response. Since the feedback occurs behind the output filter, no load dependent under- or over-damping can be observed.

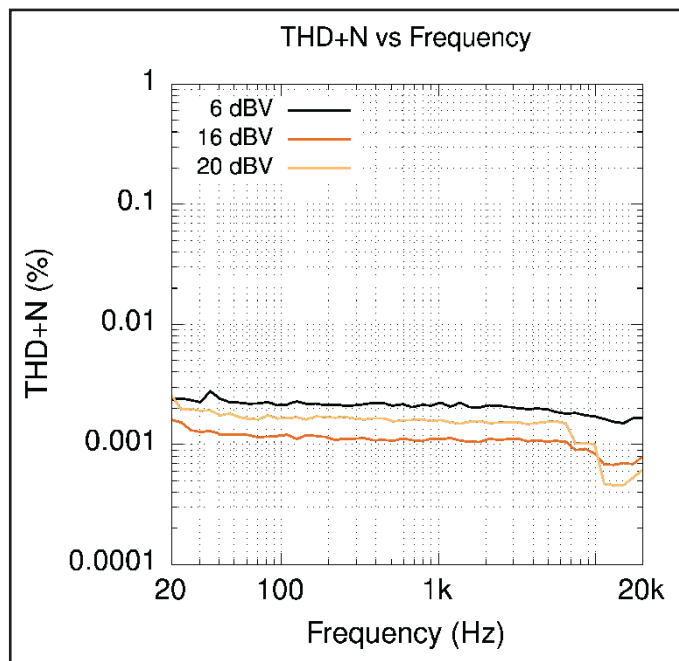


Figure 8: The AX5689 + STA516B THD+N vs. Frequency at 4 Ω PVDD = 28 V. Most Class-D amplifiers do suffer from an increasing distortion with the frequency. Distortion itself is very low with only a small output power dependency.

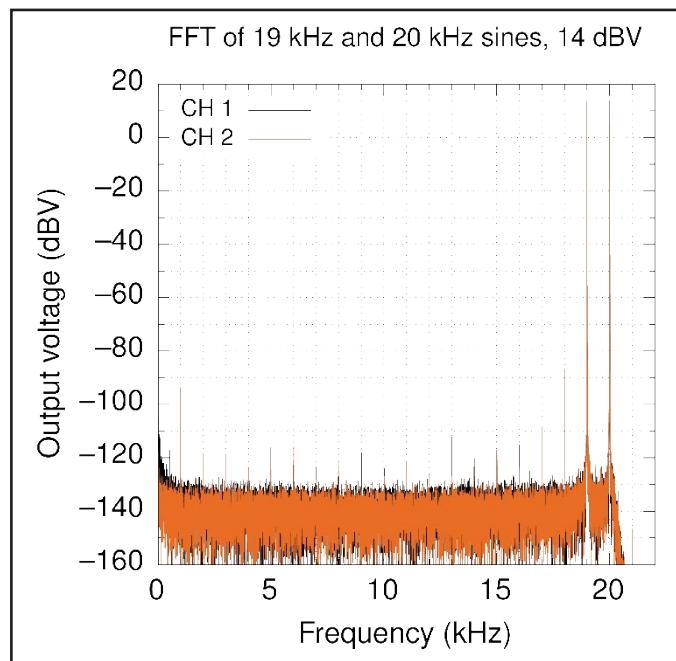


Figure 9: The intermodulation distortion (IMD) with 19 kHz and 20 kHz is below -100 dB. A value that can be shown! Nonlinear behavior is under control.



Photo 1: Prior to Axign, its CEO Jeroen Langevoort was with Royal Philips Electronics and NXP Semiconductors holding various R&D positions. In his work, he was in charge of all developments within the audio amplifiers group and introduced about 30 new products including dedicated automotive Class AB amplifiers and switching Class-D amplifiers. After starting his own consultancy firm to help companies develop their own “mean and lean” product development machines, Langevoort took over as CEO of Axiom-IC and the development of the low-latency ADC that is now at the heart of the Axign controller IC.

better. Having to cope with the TAS5624A's internal control loop is apparently challenging the AX5689 to bring out the maximum possible performance of the output stage.

Another important feature of the AX5689 is that it measures the output signal after the output filter. **Figure 7** shows a frequency response curve, which is a very straight line until 20 kHz (± 0.05 dB). The same frequency curve can be observed at 4 Ω or 8 Ω , which is a very uncommon result. Looking at traditional Class-D amplifiers, the frequency response is very dependent on the load. **Figure 8** shows very low values in the TDH+N vs. Frequency graph. Again, it is an almost straight line (almost power independent), not typically seen with the Class-D amplifiers on the market today. And the icing on the cake is revealed in **Figure 9**, which shows the intermodulation distortion (IMD) with 19 kHz and 20 kHz—where a -100 dB is achieved.

The results are simply stunning. These results are not only extremely good for Class-D amplifiers, but there are not many other amplifiers that can match these results.

For any manufacturer that wants to check these results, there are demo kits, and with the available samples, they can check the AX5689 with their own output stages. Several manufacturers already did this and they confirm the results. As the feedback inputs of the AX5689 are current controlled, output stages from a few watts up to several kilowatts can easily be configured.

Music to the Ears

Now, the million-dollar question is, of course, how does it sound? A demonstration setup was assembled on the premises of Axign, in Enschede, The Netherlands. The demo amp with one of the first samples of the AX5689 was built into a small suitcase for easy demonstrations. A set of small and medium

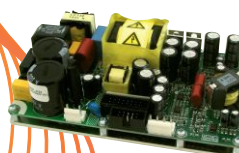
hypex NCxxxMP series

overview

The NCxxxMP amplifier module incorporates a low power standby power supply (meets 2013 ERP Lot 6 0.5W requirements), a highly efficient switch mode power supply and a high-performance Class D amplifier in one compact and easily applicable power brick.



NC122MP



NC250MP



NC252MP



NC500MP



NC502MP

Add-on Module NC100HF
The NC100HF is a dedicated tweeter-amplifier which fits the NCxxxMP series.



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Photo 2: Richard Langezaal is Director of Marketing at Axign with a bachelor's degree in Electronic Engineering. He has 25 years of experience in the semiconductor business where he held several technical and management positions in the audio amplifier group, and marketing positions for lighting solutions. He has a broad knowledge of audio amplifier systems from a customer and chip producer perspective. Richard has a passion for audio, business development, and sales. Richard has built his own 600 W UcD Class-D audio amplifier solution based on ICs from his own product portfolio.



sized Bowers & Wilkins (B&W) loudspeakers were used. Actually, these were the private loudspeakers of the management team. The creation of a full-scale demo room is in the planning stages.

The first thing that stands out is the absolute absence of noise. With no music, there is no sound. Then, when the music starts, there is this tremendous detailed soundstage where each instrument and voice has its own place. Combined with the tight bass control, it presents a wonderful, transparent picture. It is also very pleasant during long listening sessions. Wow!

Getting back to earth, I guess the audio quality I heard can be contributed to the improvements in the mid and high regions resulting in this performance. My impression is that the loudspeaker is very controlled and that the amplifier dictates what the loudspeaker has to do, and not the other way around. Feedback after the output filter clearly has its advantages.

Walking the Miles

During the last six months Langevoort (see **Photo 1**) and Director of Marketing Richard Langezaal (see **Photo 2**) have appeared at audio and consumer electronics shows with their small business suitcase and a pair of small, portable B&W loudspeakers. The two men have met with several audio companies to demonstrate that their designs can perform to a new unheard level (see **Photo 3**). Now, companies are asking the question, "Should we rethink our amplifier strategy?"

"Once we can demo, we are in," says Langezaal. "It starts with the silence, then the music does the rest."

It must be said that the AX5689 is not a chip that you just put in your design and the performance is there. The design team went through a lot of trial and error before the AX5689 started to perform as expected. For an audio manufacturer that wants to use the chip to its full potential, the datasheets and certainly the development package are absolutely essential. Further support from Axign will prove beneficial and speed up the development process.

And, there is one item that has not yet been discussed. Costs. Being impressed with the performance is nice, of course, but the cost of the chip and the implementation are important for its commercial success. At the moment, the first samples and demo boards are available for manufacturers (see **Photo 4**).

Production samples are scheduled to be available in the fourth quarter of 2017 (at the time of writing). Qualification and release to the market will be in 2018 and beyond. Prices are quantity dependent but

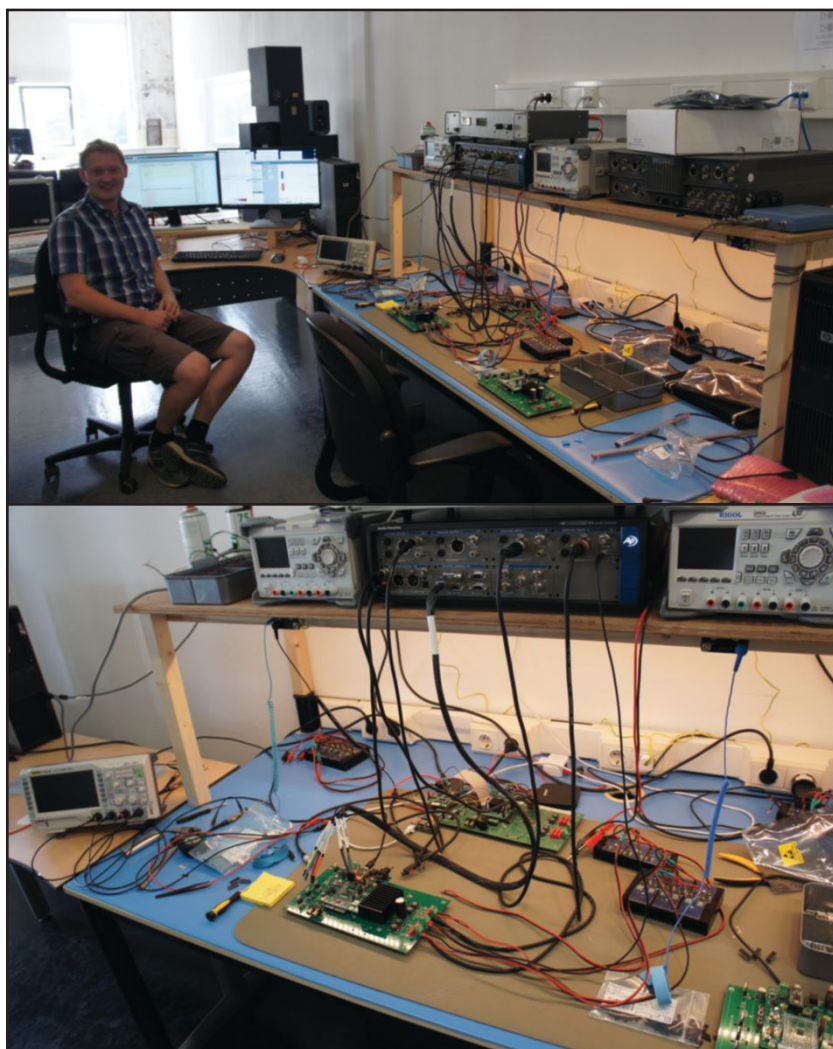


Photo 3: System application engineering is in progress by Tim van Doesum (a). Interesting results were obtained when output filter stages proved to be much simpler and less expensive. More rewarding was the testing with the reference designs of well-known Class-D amplifier manufacturers. The improvements neared theoretical values proving that, not only they were on the right track, they were onto something big here. This is the measurement set up on the bench (b).

are in line with other controller chips. The truth is that the price/performance ratio for a high-quality Class-D amplifier goes from unbeatable to ...well, more unbeatable?

Final Thoughts

The AX5689 is a game changer. Class-D amplification no longer aspires to be at par with other forms of high-quality amplification, it can now claim the throne. Due to its highly versatile structure, we can expect to see this chip not only in amplifiers but also in more complex surroundings (e.g., soundbars, soundplates, active loudspeakers, and certainly the new ascending category of voice controlled home systems). The AX5689 is also a proof of concept that the days when a cheap product sounds cheap will come to an end. Clever algorithms will take over audio amplification.

For a young company such as Axign, these are exciting times. For many startups, the first funding is based on the belief of having a good idea. Proving that the idea not only works but can be seen as a solution that will have a profound influence on the audio industry, is expected to provide funding on a larger scale. Growth will not be difficult. 

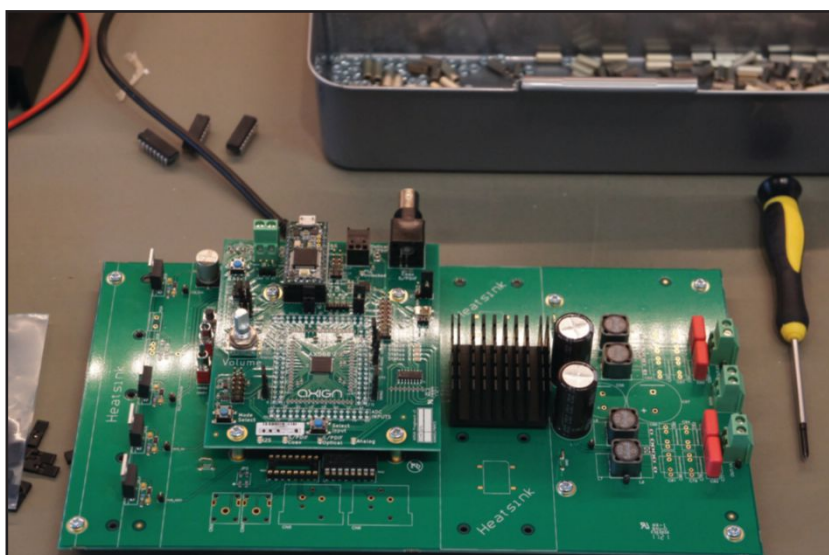


Photo 4: The AX5689 demo board served as the playground for many tests. It is built in a universal way to make it adaptable to all the tests you could imagine.

About the Author

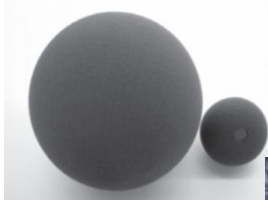
Ward Maas is the owner of Pilgham Audio. He studied electronics, marketing, and amplifier design. During his career in consumer electronics, Ward worked in areas ranging from CD standardization to radio and television to personal GPS navigation. Ward has worked on an extreme low-noise magnetic cartridge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

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Progress on Calculating Edge Effect of Acoustical Absorption

This article explores new methods to calculate the edge effect of acoustical absorption and the research behind these methods.

By
Richard Honeycutt
(United States)

Rooms in which Reverberation Time (RT) is important often require the application of acoustical treatment such as absorptive wall panels. The most obvious way to treat a wall is to cover the entire wall, and the resulting uniform appearance pleases architects and interior designers. But many “how-to” books and articles on acoustics state that absorptive material is more effective if applied only to parts of each wall, as long as untreated patches on opposing walls are avoided, in order to prevent flutter echo. This advice implies that the same area of absorption can have more or less effect in reducing reverberation, depending upon the physical shape. Although the Sabine and other statistical RT equations do not predict such an effect, experience has shown it to occur.

The Edge Effect

The physical reasons behind this apparent anomaly are often lumped into the category of “edge effect”—along with the observation that equally competent test labs using identical standard procedures can measure different absorption coefficients (α) that sometimes exceed

unity! This latter situation has been the cause of much speculation throughout the years, with some experts stating that any α greater than unity is erroneous. If α is indeed, as Wallace Clement Sabine stated, the fraction of sound absorbed by a surface, then by definition, it cannot exceed unity. However, as we discussed in *audioXpress*’ April 2016 Sound Control article, using an α of unity in the Sabine equation does not give zero RT, although if α were indeed the fraction of incident sound absorbed by a material, having an α of unity would leave no sound in the reverberant field, and thus zero RT. Therefore, we must define α as a constant of a particular material which, when obtained using the standard measurement techniques and the Sabine equation, allows prediction of RT with some accuracy, given certain known limitations. This new definition validates the possibility of $\alpha > 1$, although such values of α cannot be used with computerized acoustical modeling software.

However, we are left with the questions of why the absorption of a particular block of material apparently varies with size and shape, and why

different values are measured in different labs. Leo Beranek states that a properly done measurement of α is accurate for the particular piece of material measured, in the specific room and location in which the measurement is performed. While true, and perhaps comforting, this statement may create as many questions as it solves. One physical mechanism that has been proposed to explain the difference in measured α among different labs is the actual diffuseness of the sound field in the reverberation chamber used for testing. Many labs have now installed plate diffusors to improve this characteristic.

Still remaining to be explained is “edge effect”: What physical processes are involved? The answer seems to be twofold:

(1) The statistical assumptions upon which the Sabine theory is based are violated in practice. When a sound field encounters the edge of a piece of absorptive material, the acoustical impedance near the material changes, causing wave refraction, with the result that the material can appear as much as a quarter wavelength larger in each dimension than it actually is. This makes near-grazing waves strike the absorber and be partially absorbed.^[1]

(2) Edge diffraction causes sound crossing an edge of an absorber to change direction. This also causes an effect similar to that of a larger area of absorption. The way this works can be understood with the aid of Huygens’ Principle: Every point illuminated by a wave can be considered as a spherical-wave source. Thus when an edge is irradiated by sound, the sound beyond that edge can be calculated as a superposition of all these spherical sources. This means that we can imagine a line of spherical radiators at the edge of the absorptive material, and thus, calculate the strength of the diffracted sound in any location. Since the radiation is spherical, sound is transmitted in all directions, including along the vertical and the horizontal surfaces of the absorber’s edge. In the case of diffraction, the amount by which the absorber’s effective area increases is proportional to the ratio of the perimeter of the edges to the area of the absorber.

α Measurements

At InterNoise 2009, Ron Sauro of NWAA Labs presented the results of measurements of RT in a test lab with absorptive material of a given area—measured first in a single piece, then cut into pieces so that the area could be held constant while the perimeter varied.

His measurements demonstrated three things:

- α varies with both area and perimeter.
- α varies substantially with frequency, particularly above 250 Hz and for very absorptive materials.
- The measured absorption coefficients approach 1.5.

These results verify the oft-repeated advice to treat walls with patches of absorption rather than single large sheets, when the purpose of treatment is to control RT. The usual calculations for evaluating the effects of absorption, including equations and ray tracing, cannot give reliable predictions in all cases. Investigations are ongoing to determine how to better specify a material’s α and account for the effects of size and shape.

Absorption coefficients are measured in a reverberant chamber. A sample of the material is placed according to internationally standardized procedures, and the reverberation time is measured. The RT in the room obeys the Sabine equation:

$$RT = \frac{0.049V}{s_c a_c + s_s a_s}$$

where s_c and s_s are the areas of the chamber boundaries (walls, ceiling, and floor), and of the absorptive sample, respectively, V is the volume of the chamber in cubic feet, and α_c and α_s are the sound absorption coefficients of the chamber material and of the sample. Solving for α_s and calculating the product $s_c \alpha_c$ from the chamber dimensions and the RT of the empty chamber, we



Photo 1: One sample consisted of 2’ x 2’ squares of absorptive material. (Image courtesy of NWAA Labs)

can determine α_s from:

$$\alpha_s = \left[\frac{0.049V}{RT} - \frac{0.049V}{RT_c} \right] / S_s$$

RT_c is the RT of the empty chamber.

New Theories

S-I Thomasson investigated the effect of area upon sound absorption.^[2] He developed models for including the effects of absorber size and the angle of sound incidence.

As discussed in the April 2016 Sound Control article, Ron Sauro proposed an alternative to simple absorption coefficients based on making multiple

measurements with different perimeter: area ratio configurations. His method accurately accounts for edge effects, but does require multiple measurements for each material.

International Standards Organization (ISO) working group TC43/SC2/WG30 has a proposal based on Thomasson's paper. However since this proposal has not been finalized, we are unable to publish details at this time.

Robert Hallman of Armstrong World Industries is also working on edge effects. He presented a paper at the April 2017 American Society of Testing and Materials (ASTM) meeting in Toronto, ON, in which he discussed a method he has developed that accounts for edge effect without requiring

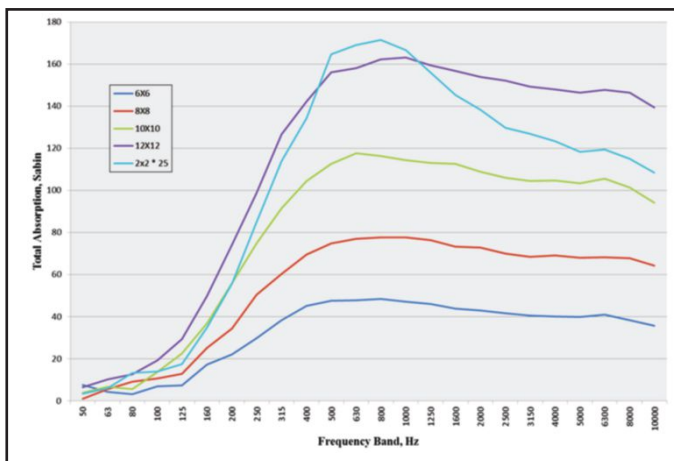


Figure 1: The total absorption of samples of the same materials varies somewhat with frequency, in addition to the expected variation with surface area. (Image courtesy of Robert Hallman, Armstrong World Industries)

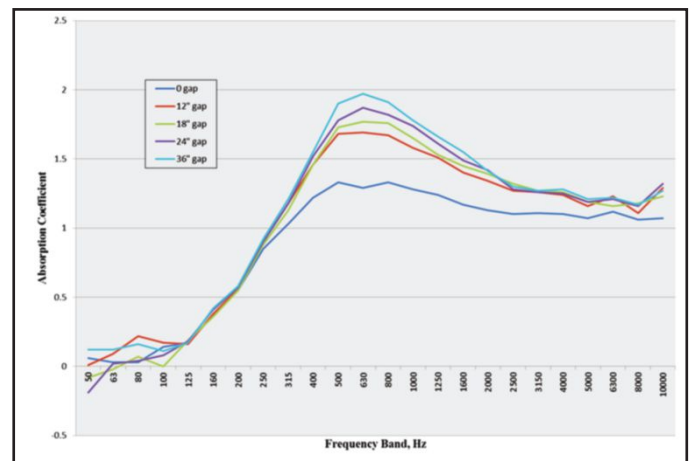


Figure 2: The total absorption of five samples of the same material, each made up of nine 2' x 2' pieces, varies with changes in the separation between pieces. (Image courtesy of Robert Hallman, Armstrong World Industries)

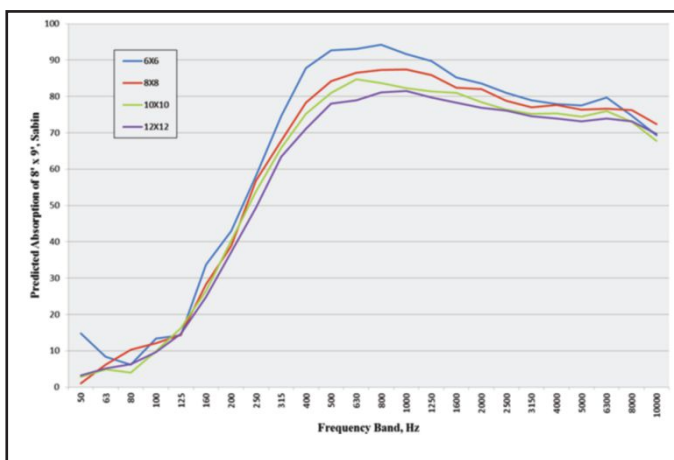


Figure 3: Using measured α values from samples of different areas yields different predicted total absorption for a standard 8' x 9' panel. (Image courtesy of Robert Hallman, Armstrong World Industries)

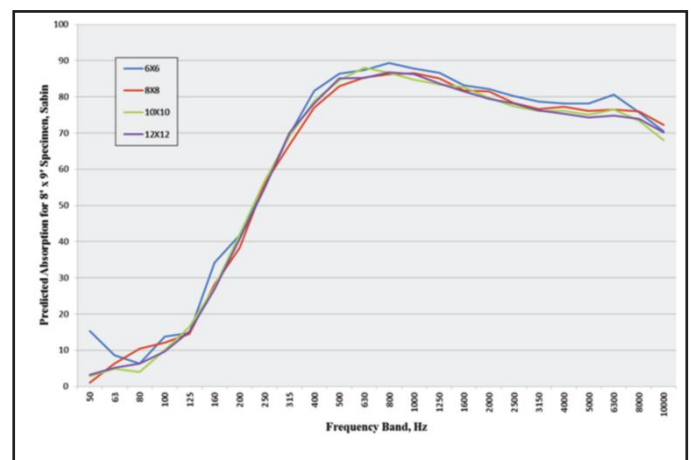


Figure 4: Hallman's method, using the measured α for four different-area samples, yields almost identical total absorption for an 8' x 9' panel. (Image courtesy of Robert Hallman, Armstrong World Industries)

extra measurements. This has obvious benefits for manufacturers, who will not have to invest in re-measuring all their absorption products, and for acousticians who will not have to update their reference tables of material absorption coefficients. The method is empirically derived. Hallman's method produces results that match the measured data for the several configurations of absorption that he tested.

Hallman's method defines a virtual area of the absorptive material that is the sum of its actual area plus a wavelength-related area associated with the perimeter. Its goal is to provide a way to predict the absorption of various sample sizes and geometries without requiring change to the

absorption coefficients. In this, it is similar to Thomasson's method, but Thomasson used an added perimeter area corresponding to the area of material within $\lambda/4$ of the edge; whereas, Hallman added a perimeter area corresponding to a distance $\lambda/2$ from the edge. Hallman then incorporated a method to avoid adding the corners of the perimeter area twice, as occurs in Thomasson's method. Next, he calculates a total absorption for the piece of material based upon the material dimensions and the sound absorption coefficient α as measured by ASTM standard C423, the present standard for measurements. Thus, existing α values can be used.

In Hallman's presentation, he showed the total

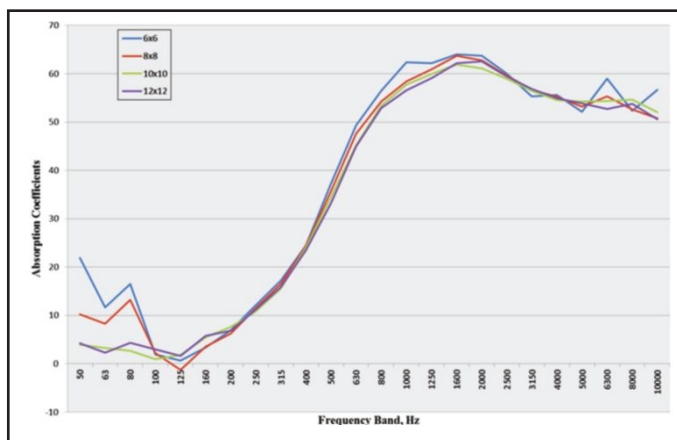


Figure 5: Using measured α values from mineral-fiber samples of different areas yields smaller differences in predicted total absorption for a standard 8' x 9' panel than were observed with fiberglass panels. (Image courtesy of Robert Hallman, Armstrong World Industries)

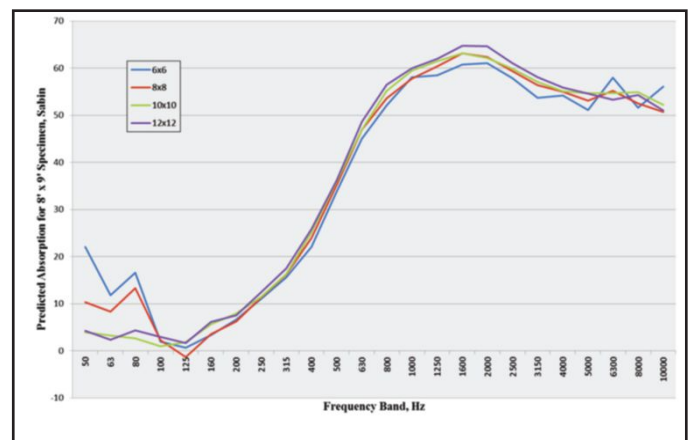


Figure 6: Although Hallman's method also showed less variation in calculated total absorption, the effects of area were reversed from those shown in Figure 5. (Image courtesy of Robert Hallman, Armstrong World Industries)

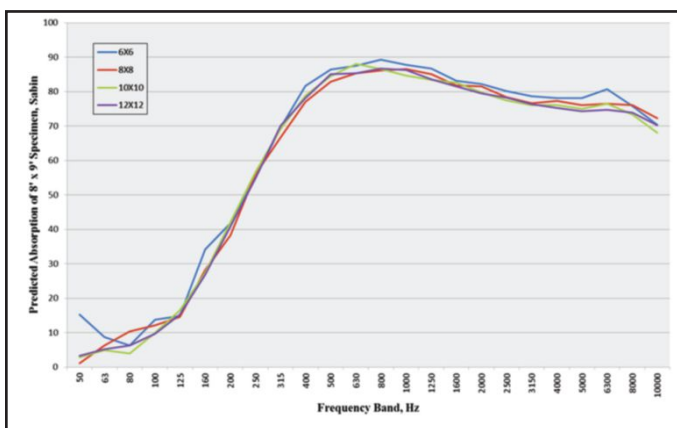


Figure 7: Hallman's and Sauro's predictions closely agree. (Image courtesy of Robert Hallman, Armstrong World Industries)

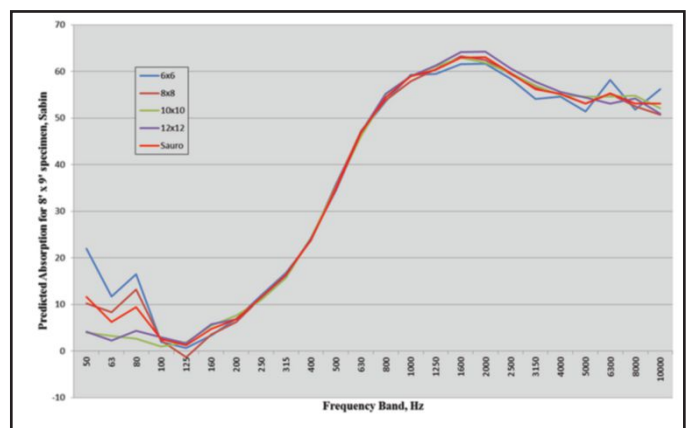


Figure 8: Hallman's revised method and Sauro's method closely agree for lower-absorption panels as well. (Image courtesy of Robert Hallman, Armstrong World Industries)

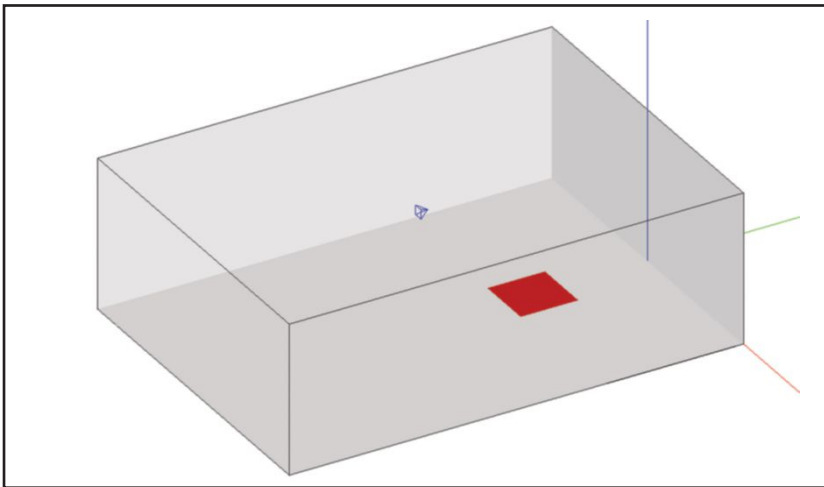


Figure 9: This EASE model of a 25,851 ft³ reverberation chamber has a 6' x 6' fiberglass panel sample.

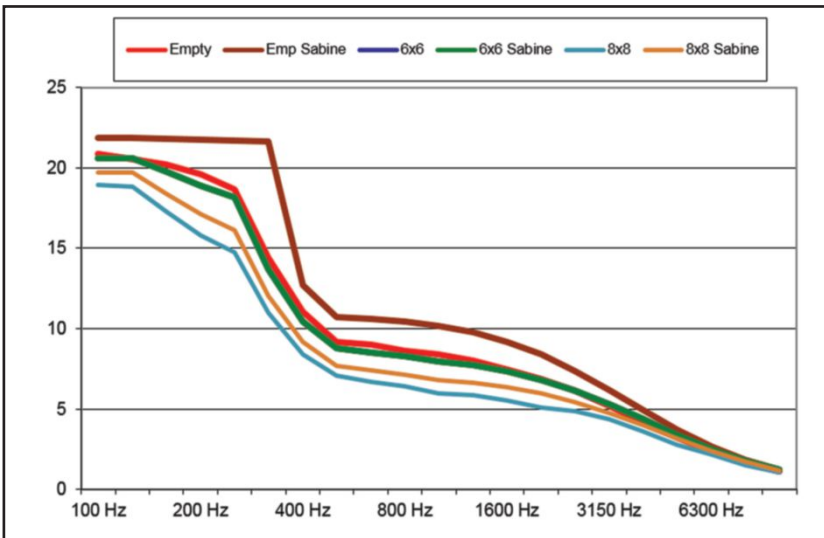


Figure 10: An EASE-modeled reverberation chamber shows the expected variations in RT with changes in sample area.

absorption in Sabins measured for each of five samples: 6' x 6', 8' x 8', 10' x 10', 12' x 12', and a 10' x 10' sample cut into 25 2' x 2' pieces and scattered, as Sauro had done with spaces between the pieces (see **Photo 1**). The total absorption in Sabins for each sample is plotted vs. frequency in **Figure 1**. Notice that, while the total absorption increases with area, there is also some frequency dependence, especially in the case of the separate 2' x 2' pieces.

Next, Hallman showed the effects of separation between 2' x 2' pieces, in samples containing nine such pieces (see **Figure 2**). Based upon these results, he omitted the separated samples from further comparisons, focusing instead on the effect of sample area and perimeter. Although European measurements of α use a square material sample, American measurements use the 8' x 9' size specified in ASTM C423. (Apparently, this size is an artifact of a practice used at Riverbank Laboratories in years past.)

Using the measured α values for the four square samples whose absorption was compared in Figure 1, Hallman calculated the value that a standard 8' x 9' sample would exhibit (see **Figure 3**).

Figure 4 illustrates that applying Hallman's method to the process used to generate Figure 3 gives good agreement among the resulting curves. These results for fiberglass panels, a highly absorptive material, were encouraging, so Hallman next tested mineral fiber panels, which are less absorptive. As shown in **Figure 5**, the variation in the total absorption of an 8' x 9' panel predicted using the α values for the 6' x 6', 8' x 8', 10' x 10', and 12' x 12' samples was less than for the high-absorption panels.

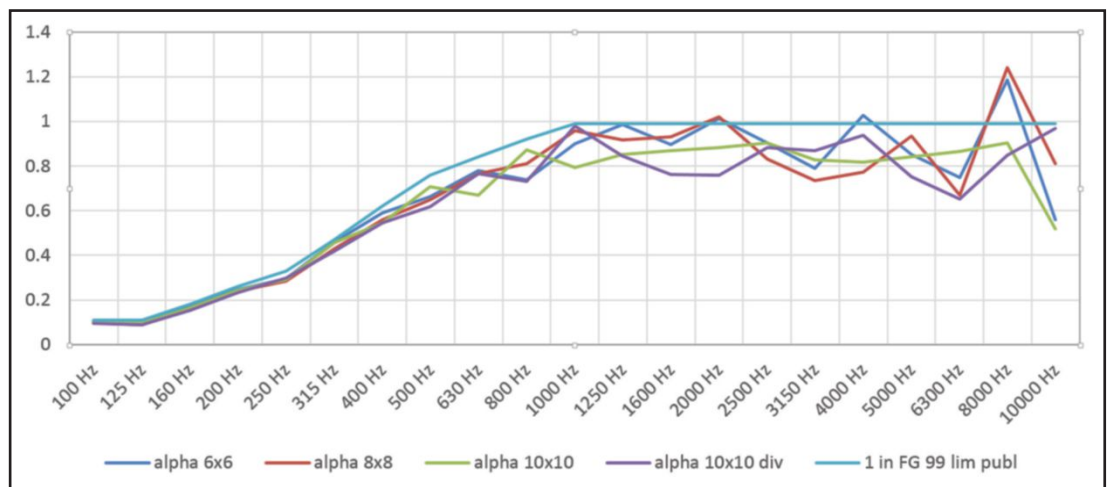


Figure 11: The α derived from ray-traced modeling shows little effect of area.

You can see from **Figure 6** that Hallman's method also showed less variation in predicted total absorption than that for the fiberglass, but seems to have over-corrected in that the absorption predicted using the α from the smaller sample is now less than that predicted using α values from larger samples. He addressed the over-correction by using a modification of α for the perimeter area.

Next, Hallman compared the predictions from his revised method with those from Sauro's method—see **Figure 7** (fiberglass panels) and **Figure 8** (mineral fiber panels). The two methods substantially agreed for both fiberglass and mineral fiber samples. However, more work is needed to understand the effects of separation between/among pieces of a "scattered" sample. Also, agreement between methods is poor in the 1/3-octave bands below about 160 Hz.

Testing the Theories

Both Sauro's and Hallman's methods have been assessed in terms of Sabine's or other statistical acoustics analysis. I was curious whether ray-tracing analysis would show any difference among the absorption predicted for samples of different areas in a modeled reverberation chamber. One would not expect there to be a difference, but I decided to check it out anyway. **Figure 9** shows the model of the chamber with a 6' x 6' sample of 1" fiberglass. I ran simulations to determine the RT for the four sample sizes used in Hallman's tests. The RT graphs are shown in **Figure 10**. The Sabine-predicted RT is also graphed.

As usual, there is a small difference between the Sabine and the ray-traced RT predictions, and as expected, RT is smaller with samples of larger size. But did the effective α vary among samples? **Figure 11** illustrates the variation of α calculated from the chamber dimensions and volume, and predicted RT. The published α of the 1" fiberglass panel is also graphed for comparison. Due to the limitations of ray-tracing modeling programs, the maximum α has been limited to 0.99. The α derived from ray tracing follows the published values for the material within 10% or less, except at the highest frequencies.

The raggedness of the curves may be due to EASE's limiting the precision of results to two decimal places. (In digital audio, we'd call that "quantizing noise.") The bottom line is that when absorptive areas of dimensions other than 8' x 9' are used (often!), ray-trace models will have some errors in RT calculations, since they cannot take the size of absorptive areas into account. The quest for ways to accurately predict RT in a room continues! 

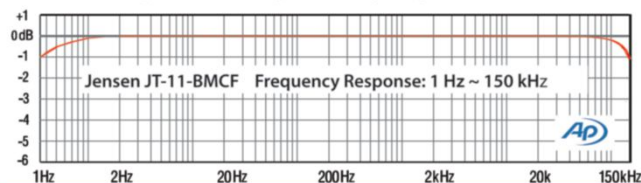
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Audio Editing Software Roundup (Part 1)

Comparing Audio Editing Tools

Fernando Rodrigues examines some of the available software audio editors currently in the market. In this first part of a complete roundup, we look at what makes pure audio editing software different from those complex and more expensive Digital Audio Workstations (DAWs). The series, examines the current state-of-the-art audio editors, together with some very affordable tools and even free software. In this first round-up, we look at Steinberg Wavelab, Acon Digital Acoustica, and DSP-Quattro.

By
Fernando Rodrigues
(Portugal)

Audio editors are a software category with great traditions, and the origins of many great tools that several studios depend on for daily operations. It is also one of those product categories where we saw more great titles vanish from the market, both due to fierce competition and also because editing increasingly became a toolset within the more generic Digital Audio Workstations (DAW) category.

Older readers may remember software such as *Alchemy* or *Sound Designer* (for Macs). More recently, *BIAS Peak* (in my opinion, perhaps the best editor available at that time) was also discontinued. The now celebrated DAWs owe a lot to these applications. In fact, those who knew the first version of *Pro Tools* probably remember it was a mixed product that was half a recording and mixer emulator (a special version of *Deck*) and the other half an audio editor (a special version of *Sound Designer II*).

DAWs vs. Audio Editors

We have seen several roundups where pure audio editing software is indifferently mixed with

DAWs. The two categories are completely different, in our opinion, and mixing them is not correct. Audio editing is a process that appeals to a broad variety of uses—from journalists and podcasters to musicians and archivists to audio enthusiasts who like to explore computer audio to its full potential. For those users, audio editors include all the tools they require since they are not involved in music production and don't need sequencers, sampling libraries, or sophisticated multitrack mixing tools.

Therefore, we decided to leave DAWs out of this roundup. It's true that some DAWs started as audio editors. We already mentioned *Pro Tools*, but we should also mention *Adobe Audition*, which started as *Cool Edit*—a pure audio editor later transformed in a multitrack editor and named *Cool Edit Pro*. We also understand that most current DAWs have built-in audio editing, with features in common with stand-alone audio editors. Some users even prefer to use a complete DAW to perform their work on mastering or restoration work. This is even more understandable, since most DSP work is now performed by third-party plug-ins, the very same

About the Author

Fernando Rodrigues began studying music and technology in the 1970s. His goal was to marry his two passions: music and computers. As a student, he helped assemble the electronics music studio at the music college in Porto. Later, he directed the technology department at one of Portugal's major distributors while pursuing a career teaching musical analysis and composition techniques. He now concentrates on research and writing about music and technology, sharing his own perspectives about music and sound.

plug-ins that we use in our DAWs. So, for those who don't feel the need for special tools and the workflow of a dedicated audio editor, the DAW is completely suitable for the job. We should add that, in our roundup, many audio editors showed that plug-in support and management is one of their weaknesses (which is a shame). Since basically that's one of the main features of any DAW, it just reinforces the reasons for such an opinion.

But DAWs and audio editors are still, in my opinion, different tools and there are plenty of reasons to master dedicated audio editing software. Those tools with which we can perform tasks such as Cut, Copy, and Paste audio regions, insert or mix/fade/morph audio, cleaning and restoration operations, compress and expand duration, pitch shifting, create loops, and use several DSP tools to transform the audio (even plug-ins, the same plug-ins that are used in DAWs, although here usually in "destructive" mode, since the transformation is usually permanently written to the audio file).

Audio editors are typically used to fine tune audio clips to be used later within DAWs, to polish and organize files for mastering purposes, to assemble bundles of files for the purpose of creating sampling instrument libraries, or even to polish, correct, and restore old audio recordings, or recordings of old vinyl LPs, just to mention some examples.

The main difference between these dedicated tools and DAWs is the fact they are intended to be used primarily to perform what it is called "destructive editing." Sure, we can have real-time processing in audio editors, via plug-ins, and we can also perform destructive editing in DAWs (although the latter is usually more convoluted, and not usually a DAW's intended purpose). Some audio editors also have special tools for managing libraries, with metadata editing, and to create playlists for CD mastering purposes.

Audio editors are optimized for the detailed "destructive" (or "creative," depending on the perspective) audio editing, while DAWs are optimized for recording (hopefully also composing, although some vendors neglect this part), mixing and real-time processing. Different tasks, requiring different tools. And that's why audio editors remain and evolve as a specific software category.

In this series of reviews, we examine current state-of-the-art audio editors, together with some more affordable tools and even free software. Some applications may be absent, but we did our best to select what we considered the most representative. Many tools are intended for very specific markets (e.g., audio fans and DJs that collect and assemble files to listen in their laptops, or smartphones, etc.).

Therefore, the tools that come in each of the selected programs greatly vary, as does the focus of its main application.

Also, some of the titles (as is the case of the popular Steinberg Wavelab) may have several versions, from the almost free to the most expensive one, varying in features, of course. Wavelab LE is not available for purchase, but not long ago we could get it almost for free or even for free, and upgrade it to more complete versions like Wavelab Elements.

So, in the end, we selected a dozen titles—and continue to add more, as companies are releasing new updated versions or releasing completely new tools—demonstrating how important the category remains. Among these, the most expensive ones have several versions, as we mentioned, and are available for both Mac and Windows platforms. The choice is diversified, with strengths and weaknesses, but overall, we'd say that the Mac platform is better served. Only three titles are Windows only, while four are Mac only. The others are cross-platform, with two of them even available for Linux.

We were tempted to include Adobe Audition in this roundup, since it may still be considered one of the most competent audio editors, in spite of also being a DAW. For now, the fact that it is only available through subscription and pending an important update, convinced us to leave it out. The last version that we could buy on a permanent license was Audition CS6.

Some of the titles included in this roundup are free, some are "donationware," and many of the commercial packages are relatively affordable. The quantity and quality of features is not necessarily tied to the price, and we may find that one of the more complete audio editors is free, or fairly inexpensive, while another with comparable or just slightly better features is quite expensive.

For each product, we wrote a short and condensed analysis (basically, a list of the features we considered more important) for each package. There are some tools (e.g., loop tools and VST plug-in support) that we think are fundamental in any audio editor, so, if these are not satisfactory or are even absent, we mention that as a minus. The same applies to things such as analysis tools, batch processing, and spectrum editing capabilities.

This said, let's start the roundup.

Acon Digital Acoustica

This editor was a nice surprise for us. Relatively inexpensive, it has several features that place it in the main league of audio editors. We started with Version 6, but went on to Version 7 soon after starting, which was still in Beta (but is now released). Besides being cross-platform (Windows and Mac OS X, with 32-bit and 64-bit versions), this Version 7 has some more evolved features. Previously available in a free Basic version, it now has two versions: Standard and Premium.

Acoustica 7 can handle an unlimited number of tracks, and render up to eight tracks (formats up to 7.1 surround) in the Premium Edition (stereo only in Standard Edition). Resolution of the audio goes up to 32-bit PCM or floating point, and sample rates go up to 384 kHz (in all versions).

Regarding the number of formats supported, it is not the most comprehensive, but still supports the most important formats: WAV, Broadcast Wave (BWF), WAV64, AIFF, M4A/MP4 (AAC – Windows 7 or higher required), MP3, OGG, FLAC, WMA and AU.

For plug-ins, it supports VST and VST3 plug-ins in Windows, as well as AU in OS X. The plug-in scanning is done in a separate process, which prevents the main application from crashing, although there are still some plug-ins that it rejects (e.g., Reaktor 6). Plug-in support is still something that's in the works, but Acon Digital is committed to implement it in a way that allows perfect management of the installed plug-ins. As usual, we can work in destructive way, in a per-plug-in basis, or use a Processing Chain.

As with Sound Forge, we get a real-time preview of the audio, before rendering it to the file (Acoustica seems to closely follow the workflow of Sound Forge in many aspects—which is good, in our opinion). It comes with many good built-in plug-ins (Acon Digital also sells plug-ins in all the major formats: VST2, VST3, AU and AAX), including the comprehensive Acon Digital Restoration Suite in the Premium Edition. Those who are keen to get back their vinyl collection in a digital media format will welcome this addition, for sure.

Batch processing is present in both Standard and Premium versions, so, only the free Basic version doesn't have it. Acoustica also includes a proprietary Time Compression/Expansion and Pitch Shifting algorithm which, although good, is not as good as the ones included in Sound Forge and Wavelab.

We can zoom waveform viewing up to sample level, but, although we have some special tools to edit audio, we don't have any to manually edit the wave at sample level. But we have spectrum viewing and we have some tools to edit spectrum, which is a new feature in Version 7, and something that very few audio editors still allow us to do. Spectrum Editing is one of new features

of Acoustica 7, by the way, and a strong argument in favor of this program (Premium Edition only).

Regarding loop editing, Acoustica presents a special loop editor window, and allows some special cross-fading, but the loop editor is still not up to the level of the ones in Sound Forge, Wavelab, or DSP-Quattro. Acon Digital told us it intends to evolve this tool and create a real loop editor like what we see in Sound Forge. The program also supports ACID loops.

Other special features include dithering and noise shaping, resynthesis of lost high-frequency components (e.g., when the sources are compressed audio), tag information formats (title, artist, etc.), and high-quality sample-rate conversion. Other things worth mentioning include the Harmonize plug-in (a good creative tool), the Convolution, and the Vitalize.

Analysis tools are another strong point of Acoustica—level meters, with several calibration methods, including K-system, a Correlation Meter (goniometer), a Spectrum Analyzer, and a Loudness Meter.

Another great feature (also present in Sound Forge) is its synthesis capabilities: Waveform generator; frequency impulse train (useful for calibration and measurement) and frequency sweep (useful to allow recording of impulse files for convolution purposes).

Acon Digital Acoustica:

<https://acondigital.com/products/acoustica-audio-editor>

Windows 7 and later/Mac OS X 10.8 and later

Acoustica Premium: €199.90/\$235.55 US

Acoustica Standard: €59.90/\$70.88 US



Steinberg Wavelab

There are two versions available for direct purchase at Steinberg—Wavelab Pro 9 and Wavelab Elements 9. Available as 64-bit only (as of Version 9) and cross-platform (Windows and Mac OS X). Just as we were going to print, Steinberg launched Wavelab 9.5 which, naturally, includes some improvements. We will publish an addendum at the end of this series with information about Wavelab 9.5. For the time being, all references are based on the previous 9 version.

In spite of being a “light”(er) version, don’t be fooled—Wavelab Elements is a full-featured and powerful audio editor, and what it lacks when compared with the Pro version is not much, and probably will not be missed on many occasions.

For example, while the Pro version goes up to a sample rate of 384 kHz, the Elements version “only” does 96 kHz. The number of audio tracks the Pro version can manage is more than 1,000, while Elements can “only” work with three. The real-time number of VST plug-ins in Pro version goes up to 41, while Elements can “only” deal with 18 simultaneously. Both versions work at a maximum resolution of 32-bit floating.

The most important missing options in Elements (but present in Wavelab Pro) are perhaps the batch processing and Mid/Side monitoring, editing, and processing. Besides this, we found that Elements is well suited to most anything we usually perform in terms of audio processing, and may be considered on par with much more expensive audio editors.

In terms of plug-in support, as Steinberg is the inventor of the VST format, Wavelab is the king. It scans the system folders, and we can also define our own folders, in addition or instead of those. Scanning is fast, and afterward, we have the plug-ins available for selection in a list that can be seen and ordered in several ways. The program supports both VST2 and VST3 plug-ins, and efficiently separates those that are problematic. Besides, Wavelab ships with a fantastic collection of built-in processors. From that aspect, even the “smaller” Wavelab Elements is one of the top choices—the new Master Rig alone (common to both Pro and Elements, although with less “processors” in the box) kind of justifies the price of the lighter version. All the included plug-ins are top-notch, and would be welcomed for use in other hosts (unfortunately we can’t). These processors can be used destructively, as is usual in audio editors, but also as regular real-time plug-ins, inserting them in the Master Channel, and rendering the final result in a new audio file (complemented by the included high-quality dithering).

Format-wise, Wavelab also reads and writes the most important formats available, as well as some legacy formats: AAC, ADPCM, AIFF, a-law and μ -law raw files, VOX, Ensoniq Paris, FLAC, MP2, MP3, Ogg Vorbis (OGG), Original Sound Quality (OSQ), Raw PCM (big endian), Raw PCM (little endian), Sound Designer II, Sun/Java AU, Text ASCII, WAV, RF64 / Wave 64 (unlimited file size), WavPack, and WMA.

Regarding loop tools, Wavelab is also well equipped. We define a loop by selecting a region and choosing “Create Loop from Selection.” Then, in the Process tab we have a section called Loop, with two buttons: Tweaker and Tone Uniformizer. The Tweaker is very complete, with three sections: Loop Points Adjustment, Crossfade, and Post-Crossfade. The Tone Uniformizer is an extra helper for those sounds that decay continuously and where finding good looping regions is more complicated. Although we have these tools, the first task is always finding a good loop selection, and in Wavelab, finding and creating these

selections is much easier thanks to the “zooming wheels” we have integrated in the GUI. Although this is a really good tool, and one of the best ones available in all audio editors (it is also present in the Elements edition) and even with that extra Tone Uniformizer tool, we still miss some more specialized tools that we had in things like Antares Infinity. Anyway, Wavelab is also a great tool for loop creation.

Another great function is the ability to manually edit audio at the sample level, with special editing tools such as the Pen Tool. This enables us to manually perform some detailed corrections. Besides, Wavelab includes also spectrum view (besides wave view) with editing tools that also allows some “surgical” editing, special “spectrum selection,” and more. Currently, of all the audio editors, Wavelab has some of the best spectral editing tools. This is one of the major arguments in favor of this program, in our opinion. And, version 9.5 seems to have expanded this even further. We’ll look into it later.

Other special tools include Mid/Side monitoring, editing and processing, Clip effects (10 per clip in Pro version/two per clip in Wavelab Elements), Audio Montage window (a special editing window to deal with several tracks at once), and an accurate and comprehensive collection of analysis tools (probably the best and most accurate of all—again we place Wavelab in the top in this category).

Batch processing is only present in the Pro version. Time Compression/Expansion and Pitch Shifting are done through Steinberg’s own technology, which we also have in Cubase and Nuendo. The quality is superb, and rivals the best algorithms available in the market. There are also some integrated restoration functions, but nothing very specialized.

As for video support, Wavelab is behind Sound Forge since it only allows users to import audio from video. Platform-wise, the versions are identical. Also, contrary to Sound Forge, the same license covers both platforms (maybe an advantage is that we can place the license on a dongle key called e-Licenser, and use it in whatever computer we have).

Steinberg Wavelab |

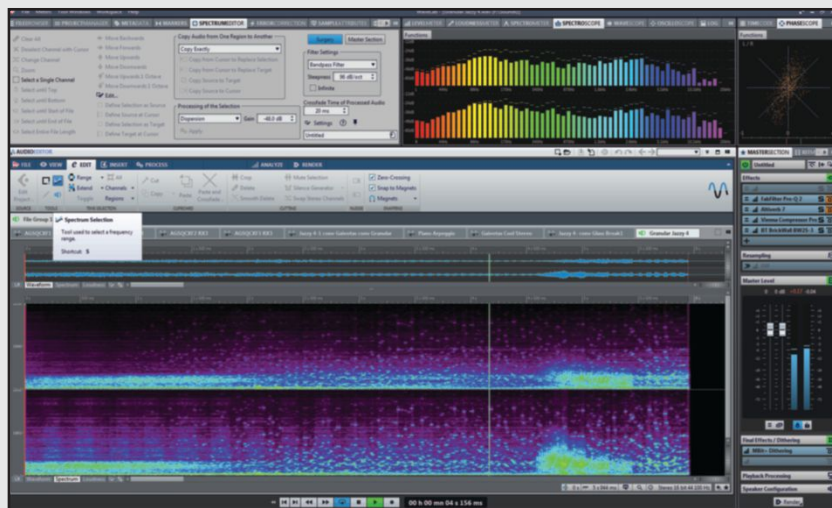
www.steinberg.net/en/products/wavelab/start.html

Windows 7 x64 and later/Mac OS X 10.10 and later

Wavelab Pro (Windows/OS X): €579.99/\$579.99 US

Wavelab Elements (Windows/OS X): €99.99/\$99.99 US

There are other versions of Wavelab, bundled with hardware, offered by magazines, and eligible for crossgrade offers, but the above are the only ones currently available to buy directly.



DSP-Quattro

DSP-Quattro comes from an experienced Mac programmer and author of D-Sound Pro—a program from the old days of the Macintosh and the direct ancestor of this program, that later worked with TC Works (Spark XL). DSP-Quattro shows a tight integration with the modern Mac operating system and is obviously the product of someone with intimate knowledge of the old days of computer software pioneers (e.g., Sound Designer and Alchemy). In fact, few audio editors nowadays include features such as the loop tools and sampler support at the level this one does, among other things. With the demise of BIAS Peak, which was another of those descendants, and although not as powerful in terms of DSP features, DSP-Quattro is the only remaining audio editor to fill the gap of powerful audio editors for sampling purposes.

It's a commercial application, although less expensive than the likes of Wavelab and Sound Forge. Yet, it is fully featured and powerful. Being an old application, it still runs on PPC machines. The system requirements are OS X 10.5 or higher and continues to be updated with the current version fully supporting Mac OS 10.12 Sierra.

Its "Advanced HD-based High Speed polyphonic audio engine" enables the software to play any number of audio files forward and backward, at every speed, without loading them in RAM memory. It also enables it to play files with resolutions that go from 8-bit up to 32-bit floating. Sampling rates go up to 192 kHz.

Being "old software" means that its creators understand old formats. Therefore, the number of formats it supports is broad: AIFF, SD2, WAV (directly), and import/export of MP3, MPEG-4, MP32, AC3, AIFC, 3GP, MPA, MP4, CAF, 3G2, M4A, AAC, Mov (extracting audio from movies), and CD audio. RAW data file Import/Export (8-, 16-, 24-, 32- and 32-bit float depth)—a feature that can be useful to reconstruct corrupted audio files. Notably absent are the lossless compressed audio formats that are popular in Windows (e.g., FLAC, OGG and APE).

Being a Mac native application, it supports VST and AU plug-in formats. It also loads and plays virtual instruments, something few audio editors can do (the audio editor must be able to deal with MIDI controls). I often use it to test virtual instruments and to record sounds from these in audio, to be later used in samplers.

Besides the excellent support for third-party plug-ins, DSP-Quattro also comes with its own collection of high-quality audio processors, including Chorus, Digital Delay, Dual Delay, Stereoizer, Multitap Delay, Reverb, Resonant "Analog" Low-Pass, High-Pass, and Band-Pass filters, four-band full parametric EQ, and seven-band graphic mono/stereo EQ.

It still supports external samplers but only via MIDI Sample Dump Standard. Its loop editing tools are among the best, with a dedicated graphic editing window and several crossfade loop options. Where loop editing and sampler support is concerned, the only rivals are Sound Forge for Windows and Wavelab.

Mixing includes support for multiple Audio File Editors, Audio Inputs, Input Recorders, and Virtual Instruments playing simultaneously, each one having their own independent effects on insert. It also allows multiple Aux output busses with separated FX chains. DSP-Quattro is easily configurable to be a real-time multi-effects unit, including Inputs to Outputs redirection. This is one of the most advanced audio editors available, in that aspect.

CD mastering and creation is also comprehensively supported, through built-in audio CD-ROM burning capability that conforms to the Redbook standards, as well as support of CD-Text, EAN/UPC, ISRC, and PQ sub-codes. It also supports DDP files as standard (including md5 checksum) and Roxio Toast/Jam SDII format to import/export of audio CD image files.

Batch processing is included, as well as high-quality Time Stretching and Frequency Shifting Algorithms using its own Spectral Re-synthesizer. In terms of restoration tools, it is not the best one, but still has a built-in De-clicker, a Pencil Tool that enables manual correction of the wave at high zoom levels, and a unique Gainer Tool to remove clicks and noises or lower unwanted transients.

Other special features of this audio editor include a Cut and Paste graphic editor with user-configurable cross-fade curves, a selectable Auto-Blend option doing Cut and Paste to smooth the waveform at insertion points, extended export functions (e.g., MarkerList or Regions to separated files), and StereoToL&R separated files.

External audio devices are fully supported, to address multiple analog and digital inputs and outputs at the same time. We also have an Output Recorder module, to dump what we are listening to into an audio file, in real time.

Since BIAS Peak vanished, DSP-Quattro has been my favorite audio editor in Mac OS X.

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Sound thinking in coaxial design

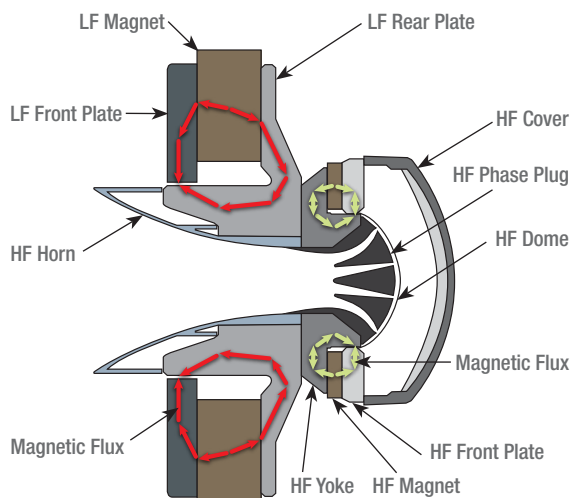
How a common magnet motor delivers a better sounding, lightweight, full-range speaker

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.

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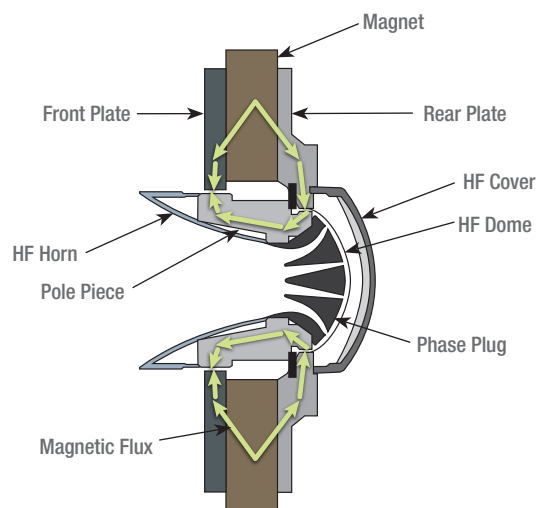
The magnetic flux is optimally distributed between the two elements (calculated using FEA), to achieve the best balance between HF and LF response (see diagrams below). Additionally, the use of a single magnet assembly also means lighter weight and a more compact profile, compared to more conventional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

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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.

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CELESTION



Fresh From the Bench

Naim Audio Mu-so Qb Wireless Music System

A Case of Groundbreaking Design



Founded in 1973, Naim Audio is a hi-fi manufacturer based in Wiltshire, UK. With a history rooted in engineering and design excellence, Naim's first wireless music system, introduced in 2014, has won multiple awards for its ground-breaking design. Its compact "little sister," the Mu-so Qb, followed suit with its stunning design, glass-filled polymer casing, the same clever digital brain as the original Mu-so, 300 W of power, and bass radiators that help generate a beautiful sound.

By
Oliver A. Masciarotte
(United States)

There once was a tale of two sisters—one sleek or, some would say, flat; and the other, imposing though some would say a bit blocky. They were cultured children, knowing just how to behave with all the better societal norms—AirPlay, Spotify Connect, TIDAL, Wi-Fi, Bluetooth, UPnP music servers, iOS, and Android. They accepted teachings from "radio" on the interwebs, a USB flash drive, wired 10/100 Ethernet plus analog, or digital audio spigots. They could read a variety of dialects (e.g., MP3, ALAC, AAC, FLAC, WMA, WAV, AIFF, and Ogg Vorbis). And, of course, they also played well with their other Naim streaming siblings.

The girls, named Mu-so and Qb or "Cube" due to her shape, grew up to be pretty rockin' kids, each with a two-year warranty. Their brushed aluminum casework, silver anodized heatsink, backlit 18 mm acrylic plinth, and curvy black livery assures that they're comfortable in almost any decor save Colonial Revival. For the fashionistas in the crowd, optional grills are available in three desaturated colors—a

carnelian red, a dark teal, and a burnt orange. The fabric grill cloth (see **Photo 1**) on a sturdy plastic skeleton borrows its sinuous shape from Naim's towering NAP S1 Statement monoblock.

On top, Qb also has a similar large, silky knob as the NAP S1. When powered, the acrylic base lights white in a low-key way, illuminating the inscribed logo. The amount of illumination is, as with all other functions, accessible, and adjustable from an app, yet that giant rotary volume control also has enough real estate to contain the input selector and power switch (see **Photo 2**).

On the back, below the hefty heatsink, is a 3.5 mm analog TRS input jack, an F05 "TOSLINK" female for legacy AES-3 digital in up to 96 kHz, an RJ jack for 10/100 Ethernet connectivity, and a standard Type A USB for plugging in a thumb drive library. Also on the back is a tiny multicolored indicator to inform you of the Qb's current state, used in conjunction with the app's onboard function.

The 8" Cube comes equipped with a power cord

Mu-so Qb

Naim Audio, Ltd.
Southampton Rd.
Salisbury, UK SP1 2LN
+44 (0) 1722 426 600
info@naimaudio.com
www.naimaudio.com
Price: \$899 MSRP

that mates to its C8 inlet and supplies the universal switched mode power supply. That PSU has two main outputs—one for the power amplifiers and another for the digital sections. The digital PSU is, in turn, sub-regulated for different processors running at different voltages. Actually, there are six main power supplies within the product, some of those being regulated for different subsections. Unlike Naim's more traditional products, the Mu-so sisters are almost entirely digital.

Atop the Qb, when you receive it in its carrying handle-equipped box, is a printed Quick Start guide and a "pin-tool" for hard reset, a manufactured version of the venerable bent paper clip. A chunky, substantial infrared remote is available as an option, but I didn't receive one during the review so I mostly relied on my iPad mini to drive the Qb around.

Once a Qb is presented with a wired or wireless connection, it steers you through a setup procedure that, for me, didn't initially "stick." I had to set it up several times before it settled down and performed further without a hitch. After setup, the Qb is accessible via AirPlay or its other wireless protocols, and it offers the world of Internet radio streaming via an embedded vTuner premium 5. If you have more than one Naim streaming product around your domicile, including the extended Uniti family, you can bind them together for whole-house audio.

The Qb is a very attractive product. If you take off the shapely grill, which I did rather quickly to hear what it was doing, you'll see a unique "...arrangement of five drivers, offset, and angled perfectly to create a left-right dispersion symmetry," as Naim would say. There's a single shared woofer, while each channel has a 64 mm midrange and a 25 mm tweeter in a unique, slightly asymmetrical arrangement. Each mid/tweeter pair is rotated 45° relative to the front face, providing more perceived separation of each channel. If Darth Vader had an entertainment robot, the Qb could well be its stand-in (see **Photo 3**). With its grill on, however, the Qb is all Mies van der Rohe, a rectilinear study in silver and subtle shadow lines.

Inside the Qb

I asked Steve Sells, Naim Audio's Electronics Design Director, what goes on inside that enigmatic cube. There are a total of 300 Watts "...worth of amplifiers inside. There's 100 W on the large (shared) 'obround' woofer and 50 W for each of the mids and tweeters," Sells said. The amplifiers are all off-the-shelf Class D, and they all have digital inputs, not analog. Sells mentioned that they carried out "...extensive auditioning of the Class-D amplifiers. During the development with Bentley motor cars, we found that Class D can be tuned to the speaker very

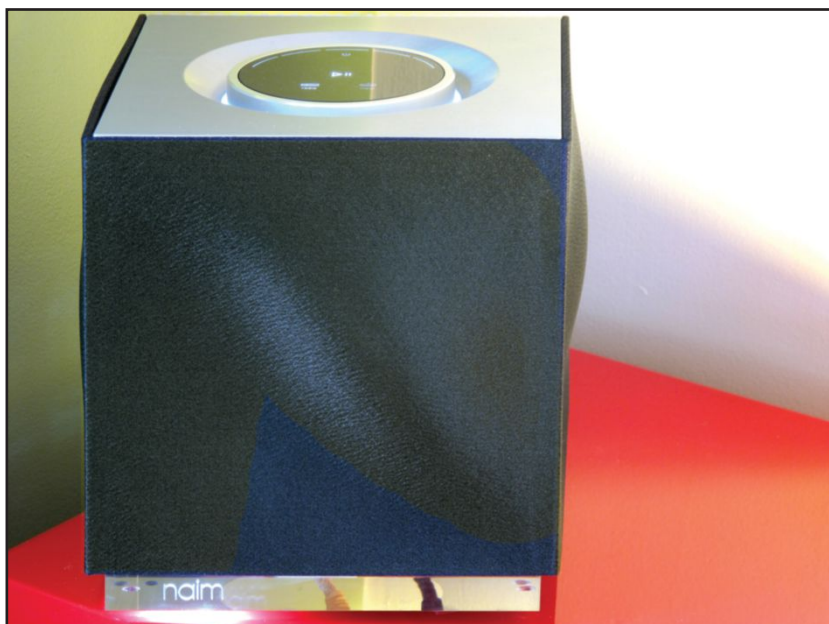


Photo 1: With its compact footprint, broad connectivity, and hi-fi sound, the Naim Qb is a winning entertainment appliance.



Photo 2: All controls are easy to decipher and operate.



Photo 3: A great deal of thought has gone into what's behind the grill.

successfully when the speaker and enclosure are (pre) defined. This carried over to Mu-so development.” In 2008, Naim partnered with the respected Bentley Motors, a marque now almost 100 years old, in that “Naim for Bentley” project Sells mentioned. Their aim was to deliver an optional, high-performance sound system for the already luxe automobiles. Today, for

a slight up-charge, you can purchase a higher styled Naim for Bentley Mu-so that debuted in October 2016, with a knurled volume knob and aluminum colored grills that match those in the Naim for Bentley cabin.

Let’s veer away from motor cars and drivers to another kind of driver, as in loudspeakers...Sells told me that all of the drive units and passive radiators are custom designed by Naim and not used in any other product. The 145 mm × 76 mm woofer employs a large, 72 mm ferrite magnet. Its basket is steel, with a stiff paper cone and a rubber surround. It has a 25.2 mm voice coil wound on a Kapton former. The high-temperature polymer former stops eddy current braking associated with metal versions. The 64 mm mids use neodymium magnets and also employ a steel basket, paper cone, and rubber surround. The former for those is aluminum, this time “...the excursion isn’t as large so eddy currents are less important but heat dissipation is,” according to Sells. The tweeters are 25 mm catenary silk domes, neodymium magnets and, like the mids, copper-capped pole pieces for lower distortion. Naim refers to the dual side-mounted, 148 × 86 mm oblong passive radiators that load the centrally located woofer as Auxiliary Bass Radiators (ABRs).

A dedicated 32-bit DSP does the heavy math. It calculates the digital volume control, plus boundary compensation, speaker protection, and bass compression when needed. Sells pointed out that, “The crossover is exclusively performed in the DSP and the Class D amplifiers are directly connected to the drive units. This gives incredible dynamics and headroom.” The boundary compensation is carefully tuned to alleviate rear wall constructive interference.

Regarding bass compression, as in low-frequency dynamic range compression, Sells clarified that, “It’s not to change the sound like a sound engineer would but to maximize the bass capability of the unit and provide protection to the drive units from being over driven. We use a mix of compression and limiters on all drive units. They don’t come into play until they are needed to maximize transparency but make the unit party proof.” Party proof indeed...I measured 105 dB SPL A weighted with slow averaging from 1 m while playing music. That didn’t even come close to maximum output as, in the app, I’d already set a limit of 85% of maximum volume and, at 105 dBA, the volume control was at about 90%!

Mu-so vs. Qb

With Mu-so and Mu-so Qb being closely related, I wondered how they differ...They use the same streaming core, host controllers, and amplifiers. The PSU in the Qb is scaled for lower voltage to reduce the power output and physical size. Also, the bass

Design Challenges

When asked what was their most challenging design mandate and how it was addressed, Steve Sells, Electronics Design Director for Naim Audio, was quick to point out that it was all about two factors: sound quality and reliability. “To address the sound quality, we started with Mu-so and designed the speaker as if it were a high-end, active loudspeaker with DSP crossovers. We had just spent three years designing the Bentley DSP active car system. Bentley told us their customers may take the car to a desert, turn the system up to full volume and party. It must not stop and it must sound good irrespective of volume setting. Then Simon Matthews, our Group Design Director, made it look like clean modern art with joyful interaction. We built many prototypes and tuned them before closing in on the final design. Once we had (that), we worked again on cabinet interactions and drive unit performance. The speaker system was designed by (Technical Director) Roy George, who has designed many high-end speakers. We also used a DSP that would generally be considered too good for an all-in-one speaker. We wanted to maintain as much digital resolution of the original material as possible. The EQ tuning (alone) took 6 months.”

“To address reliability we do all sorts (of tests). We have a lab specifically set up for safety and tests. Tests include humidity and life accelerated heat cycle testing. We also carried out HALT (Highly Accelerated Life Test)..., where we increase the stress until it fails and look at the weakest failure mechanisms. We carry out comprehensive FMEA (Failure Modes and Effects Analysis)...and drop tests. We shipped prototypes around the world and back again. However, in terms of man hours, the software is the largest effort. These small systems are complex. They are like a complete multi-room active hi-fi system in one box and connect to the Internet and have complementing iOS and Android apps. The end result is we made an all-in-one speaker that can be played all day for radio or streaming without fatigue and then genuinely fill a room with sound for a party in the evening. (Naim’s top-of-the-line) Statement was being designed in the same lab simultaneously. Statement drove us to obsessive levels of detail in design, the whole lab was fanatical about quality. I remember how we applied the same design principles to our most affordable and expensive products at the same time. It was a very busy time and very rewarding as a team.”

When testing and analysis fails and your beloved product goes down, it’s nice to know that the rumor about Naim service is...almost true. Urban legend says that they service every product ever made, and Sells clarified this...“Some parts are now gone, for example, drive units for old speakers. Generally, all electronics we can service and repair. There’s always some cool old vintage gear down there being re-capped and serviced.”



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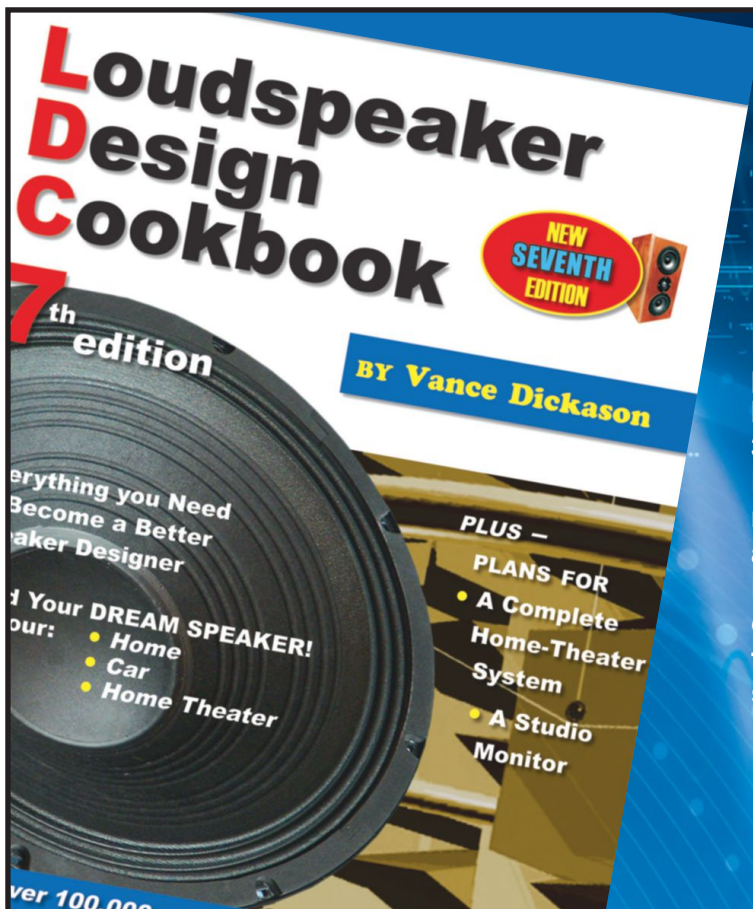


Photo 4: The Qb is an exceptional product aimed at those of us for whom quality audio is fundamental.

amplifier in the Qb is mono and the two have entirely different EQ tuning. The Mu-so uses an MDF cabinet and the Qb uses reinforced ABS. When it comes to the more enthusiast-oriented Uniti family, things get a bit more fancy. Sells gave me the low down... "Uniti Atom is a completely different platform and does not share any parts with Mu-so. Atom has a very high-end, 384 kHz-capable streaming core (the cost is not justifiable on Mu-so) and a SHARC 40-bit floating point DSP. Again, the SHARC would be great in Mu-so but is complete overkill and as such would lower the value for money. Atom also uses a large toroidal transformer power supply, analog volume control and Class AB amplifiers. The commonality is that it is the same engineering and design teams designed both Mu-so and Atom."

I mentioned a USB input...From that flash memory library, the Qb can play WAV, FLAC, and AIFF up to 24-bit/192 kHz, MP3, OGG, WMA and AAC files up to 16-bit/48 kHz at 320 kbps. Both streaming and locally resident data passes through a 2 MB circular buffer in RAM so its usage is timed just right.

An obsession for detail can be seen in choices made regarding brain power inside Qb. Even though there's enough spare processing power in just the



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Bluetooth chip alone to be a host controller, the design team used two micros plus a dedicated monolithic DSP. An ARM RISC microcontroller is the main host and master of all things digital. It tells all the other subsystems what to do, and runs a micro-kernel written in-house. On a sub-PCB, there's the streamer processor. It has the responsibility of looking after all thing streaming—the cloud and UPnP streams, Airplay, iRadio, Ethernet management, and so forth. It runs a variant of ThreadX (Express Logic's very popular real-time OS). Interestingly, Qb's RISC micro can only handle ALAC files up to 24-bit/96 kHz, due to the additional computational complexity required for decompression.

The Verdict

The Qb is definitely not your grandmother's table radio (see **Photo 4**). In fact, the only radios it has is an 802.11b/g, 2.4 GHz Wi-Fi transceiver and SBC, AAC, and aptX Bluetooth. Nonetheless, the Qb serves as an extremely high-fidelity version of that compact appliance of yore—easy to use, able to deliver voice and music from around the globe as easily as your favorite playlist, and it even includes a convenient smart alarm function to wake you up

or, in my case, remind me to meet the school bus.

Naim's marketing warns that, "listen and you'll be hooked," and I have to agree. I love my other wireless powered speakers, Soundmatters' brilliant and pocketable foxLv2; BenQ's electrostatic treVolo party animal; and Audioengine's capable and cost effective B2, as each has its own advantages and place in my musical menagerie. Yet the Mu-so Qb occupies a special place, one formerly held in granny's house by the likes of Tivoli or Bose.

Your family will love it, I know mine did. With its thrillingly hi-fi sound, broad coverage, wide variety of input choices, rock-solid construction and understated good looks, Qb is an exceptional product aimed at those of us for whom quality audio is fundamental to our everyday lives. In the sidebar, "Measurements and Listening Tests, Kent Peterson confirms my glowing opinion of Naim's Mu-so Qb. **A-**

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.

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Measurements and Listening Tests

By Kent Peterson

The Naim Mu-so Qb was measured at Warkwyn's facility using the Klippel Near Field Scanning (NFS) system delivering a 360° balloon, which allows for an examination of the radiation pattern and off-axis frequency response and at any frequency—important information when determining where to place this compact but substantial system in your home.

For measurement of the Mu-so Qb, we used 0.5 V as an input through the line-in and with the volume at full gain. A calibrated ACO Pacific (7052E capsule) free-field mic with the 4048 preamp was used as the measurement mic and all on-axis data is referenced at 1 m and between its two opposing tweeters.



Photo 1: The Naim Mu-so Qb was set up in the Warkwyn conference room.

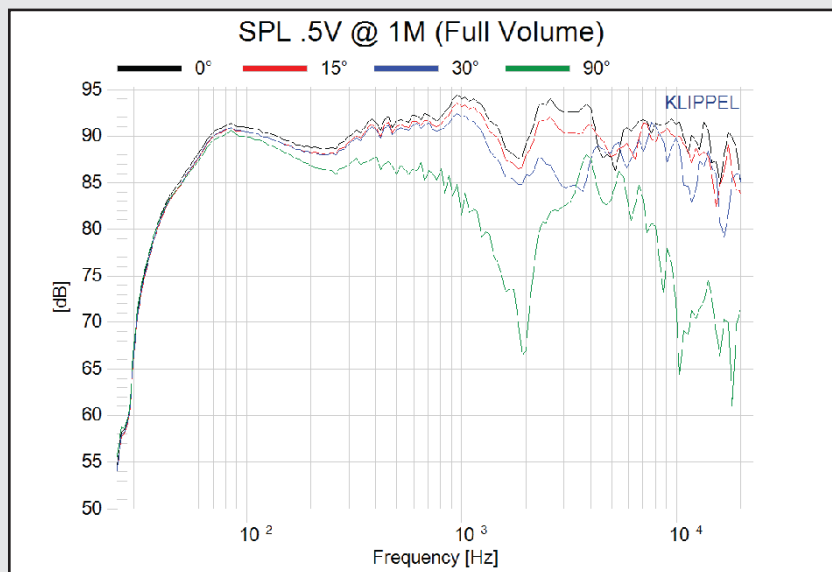


Figure 1: This is the horizontal sensitivity on-axis, 15°, 30°, and 90° (no smoothing).

Measurements points around the speaker totaled 2598 and were processed with a resolution of 0.73 Hz and from 20 Hz to 20 kHz. The length of the stimulus was 1.4 s.

Subjective listening was performed in a 12' × 20' carpeted conference room with padded seating and a small kitchenette in the corner—similar to what may be encountered in a typical living room (see **Photo 1**).

The Measurements

On- and off-axis measurements show the speakers impressive low-frequency response below 100 Hz with a slight bump at 80 Hz. It must be noted that in much the same way that measurements are performed in an anechoic chamber, the Mu-so Qb was measured mounted upon a pole in the center of a room with no boundary plane present. The low-frequency response was immediately noted post measurement with this in consideration and knowing that in general use, this system would be used with a boundary plane always present (e.g., a counter top, a coffee table, a bar, etc.), which will further emphasize the low frequency in any room. A look at the measured frequency response in the horizontal plane on-axis and at 15°, 30°, and 90° is shown in **Figure 1**.

As mentioned earlier, there is a good amount of low-frequency content from 60 Hz to 150 Hz with a gentle rolloff until rising at 280 Hz. This is consistent even at 90° off-axis and not surprising considering the opposing passive woofers on either side of the cabinet—the bass from this system can truly fill the room. A 10 dB rise then occurs from 300 Hz through 1 kHz.

In all the horizontal axis, the system then makes a substantial swing downward at 1.2 kHz until recovering at 2.5 kHz. It may be more accurate to say that there is a pronounced bump from 2.5 kHz to 4 kHz with another occurring from 6 kHz to 10 kHz before rolling off through 20 kHz with on-axis peaks at 15 kHz and 18 kHz. An unwrapped 360° pressure/f contour map shown in **Figure 2** provides a complete view of the horizontal radiation of the speaker through its rated frequency response.

The horizontal contour mapping affirms the on/off axis measurements but also reveals the high directivity of the Mu-so Qb above 1 kHz and also illustrates the considerable drop-off at 2 kHz. As anticipated the dual passive radiators provide omni-

directional low-frequency information from 200 Hz and below.

Vertical on/off axis measurements are similar to the horizontal curves and not surprising considering the symmetry and relatively small size of the cabinet (see **Figure 3**).

Interestingly the mid-band from 1 kHz through 2 kHz holds together much better 90° off-axis and in the vertical plane. This may be stretching a positive note of an extreme off-axis measurement, but it certainly appears that any party attendees laying on the carpet in front of the Mu-so Qb will enjoy the same presence response as those dancing on the surrounding balcony.

Examining the unwrapped vertical contour balloon shown in **Figure 4** offers no surprises, again considering the relative symmetry of the system and the cabinet.

Listening

A week of listening to the Mu-so Qb system was very enjoyable while we switched from classic R&B and rock to jazz and orchestral, to contemporary pop and dance music. All the selections were line-input utilizing full-resolution .wav files through a tablet.

A listening staple—*Aja* by Steely Dan (thanks for the music and enjoy that higher stage Walter Becker)—seemed even and smooth. However, we found a substantial improvement in the low end when we moved it from a single open plane to a corner kitchenette area, thereby improving the boundary conditions. This was the same for orchestral allowing a nice warm feel to the double bass and viola without feeling overbearing.

Vintage samba and bossa-nova recordings (there is always time for João Gilberto, Antônio Carlos Brasileiro de Almeida Jobim, and Stan Getz) seemed to play better in the corner as well. And not only in the low end—the addition of lateral and close supportive reflections made up for the lack of presence at 2 kHz filling the room with that gorgeous guitar, pleasant vocal delivery, and stark saxophone.

Considering the latest trends, we enthusiastically tapped our feet to the likes of Daft Punk, LCD Sound System, and yes, Taylor Swift. We're engineers but we're not dead—and we didn't say we danced.

Placement in the corner for these bass heavy mixes, however, seemed over the top (unless you're into that kind of thing). Moving the system back to the table in the middle of the room and out of the corner certainly tamed that effect. But that's the beauty of this system—it's easily movable and can quickly become part of the party. Though considering that a quick EQ of the system with Naim's app can correct for too much bass, you may want to keep it in the corner as we wouldn't want anyone spilling anything on this beautiful package. 🍷

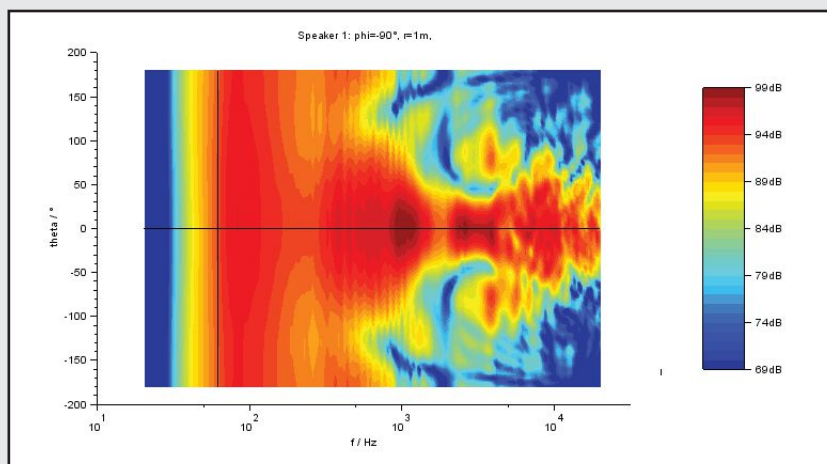


Figure 2: The unwrapped horizontal contour balloon provides a complete view of the speaker's horizontal radiation through its rated frequency response.

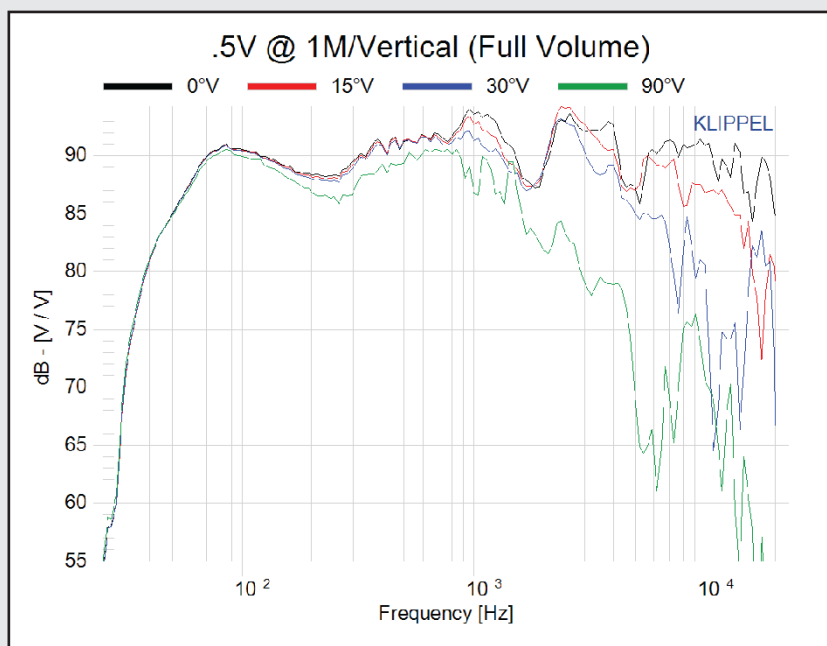


Figure 3: This is the vertical sensitivity on-axis, 15°, 30°, and 90° (no smoothing).

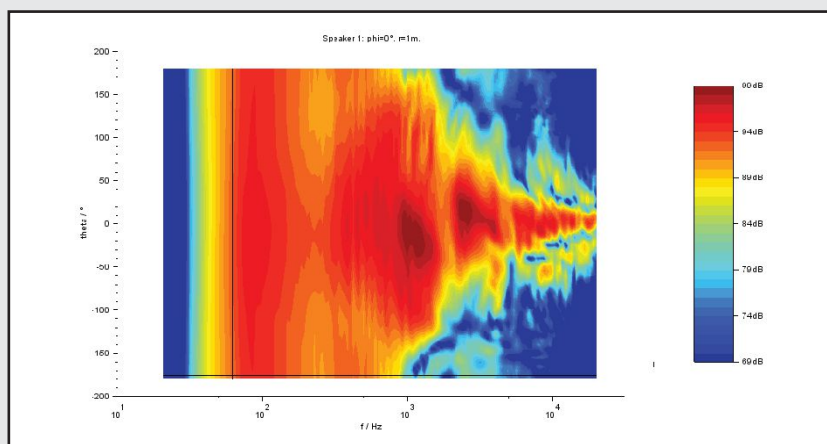


Figure 4: The unwrapped vertical contour balloon offers no surprises, considering the relative symmetry of the system and the cabinet.

The Simply Sound Company SS-1 Microphone Preamp



Photo 1: The Simply Sound Company's SS-1 is a single channel preamp, meant to boost the signal of a low output microphone to levels compatible with normal microphone inputs.

Based out of the Rocky Mountains of Salt Lake City, UT, a band of musicians, engineers, and enthusiasts with a desire to make great products, designed a way to give new life to vintage and modern microphones. Always looking for new recording tools, Stuart Yaniger reviews the first—and so far only—product from the company.

By
Stuart Yaniger
(United States)

Some of us love ribbon and dynamic microphones and what they do for the sounds they pick up. Unfortunately, some of our preamplifiers don't share our love for them—ribbons and dynamics tend to have pretty low output, and microphone preamplifiers (whether separate or built in to a mixer, PA, or digital interface) don't always have a noise floor, gain, and dynamic range that are suitable. Users often have to turn up the gain to the point where noise intrudes or they run out of gain completely, with the gain knob all the way up but the mic volume still too low.

Realizing a need, several manufacturers have created pre-preamps, giving some gain boost to bring the microphone output level up to the point where the noise floor and gain limitations of the mixer recede into the background. Typically, these

pre-preamps have 20 to 30 dB of gain and are powered from the mixer mic input's phantom power (usually 48 V), with isolation of the phantom power from the preamp input. Popular versions include the Cloud Microphones CL-1 and the Triton Fethead.

The Simply Sound Company, a Utah-based startup, has entered the fray with its SS-1 preamp (see **Photo 1**). The SS-1 is a single channel preamp, meant to boost the signal of a low output microphone to levels compatible with normal mic inputs. It's housed in a rugged extruded aluminum case, with XLR connections on each end. The internal circuitry is transformerless and all-discrete, comprising a differential FET cascade using four LSK170 low noise N-channel FETs. It appears to be running open loop. Construction is on a two-sided PCB with through-hole components, and a neat and uncrowded layout (see **Photo 2**).

Despite the high-quality parts and good

SS-1 Microphone Preamp

The Simply Sound Co.
PO Box 17109
Salt Lake City, UT 84117
800-556-4006
hello@simplysound.co
Price: \$74.99

construction, the SS-1 sells for significantly less than competitive products—less than \$80 at the time of this writing.

Performance Testing

As usual, I conducted the measurements using an Audio Precision APx515, with the APx1701 transducer interface providing the phantom power. Basic measurements showed a 31.2 dB and a 31.4 dB gain for the two units, which is very good matching for an open-loop circuit. This is higher than the SS-1 gain specification of 25 dB, but the AP analyzer has a higher input impedance than the 3 k Ω used for the spec. Output impedance of the SS-1 measured 2875 Ω , so the voltage divider formed by the output and load impedances accounts for the 6 dB difference from spec. The high source impedance suggests that the FET cascodes are not buffered and that there's no feedback loop, the source impedance then being dominated by the FET drain load resistors.

The frequency response is shown in **Figure 1**. There's a slight rolloff in the top octave (about -0.7 dB), which might be barely audible to young fresh ears, but only if the microphone has an extended high end. It occurred to me that the rolloff could be a function of the input capacitance, so I re-measured the frequency response at different source impedances. Changing the source impedance had no effect on the frequency response, so the rolloff appears to be intrinsic to the circuit.

Figure 2 shows THD+noise vs. input level at 1 kHz, taken using 90 kHz and 22.4 kHz bandwidths. The measurement is noise-dominated up to about 7 mV (for the 22.4 kHz measurement bandwidth), where it reaches a minimum of 0.012%. As the level increases, distortion rises, but is still low (0.05%) at 30 mV in. This corresponds to about a 1 V output, which is much more than would be encountered in real-world use. The variation of THD with frequency at 5 mV in is shown in **Figure 3**. It's nearly flat across the audio band, characteristic of so-called zero feedback circuits. Distortion spectra at 1 mV and 40 mV in are shown in **Figure 4**.

For clarity, I shifted the fundamental frequency by 200 Hz between the two measurements. The latter input voltage disproportionately increases second and third harmonics, although they still remain

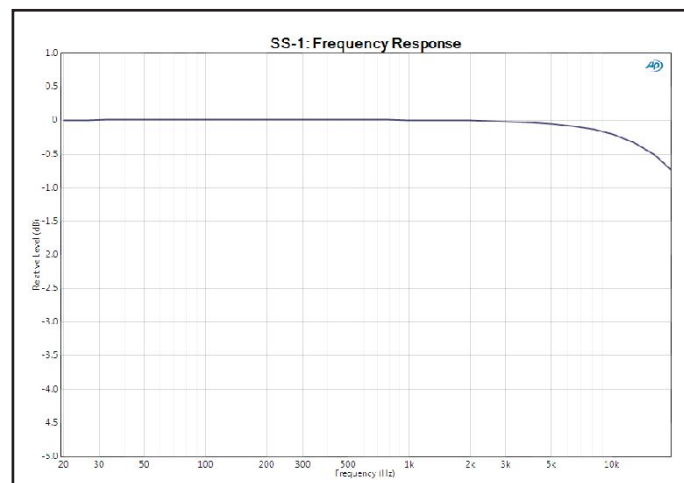


Figure 1: The frequency response of the SS-1 shows a small amount of rolloff in the top octave of the audio band.

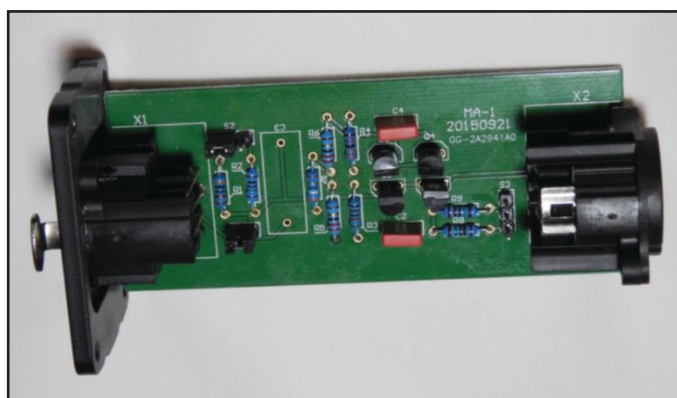


Photo 2: The SS-1 is constructed on a two-sided PCB with through-hole components, with a neat and uncrowded layout.

relatively low at under -70 dB (0.03%). Intermodulation distortion (IMD) was similarly low: at 200 mV out (5.4 mV in), 19/20 kHz IMD was dominated by the 1 kHz first order product at -91 dB (0.003%), shown in **Figure 5**. This is excellent performance for an open-loop circuit.

The SS-1's spectrum of a 200 mV output multi-tone torture test (40 tones, each spaced half octave apart) is shown in **Figure 6**. Distortion is indicated by the presence of tones or hash between the test signal tones. In this case, all distortion products were at -105 dB or lower, with very little discernable hash—a good result.

Signal-to-noise ratio (SNR) was 80 dB for a 5 mV input, measured over a 20 kHz bandwidth, A-weighted. **Figure 7** shows the noise spectrum of the SS-1 compared to the noise of the microphone preamps built into the Scarlett 2i2 when set for equal levels. The penalty for the extra gain of the SS-1 is a bit more noise (2 dB or so), which may or may not be significant, depending on the microphone used and the nature and level of the sound sources. I'll give an example in the next section.

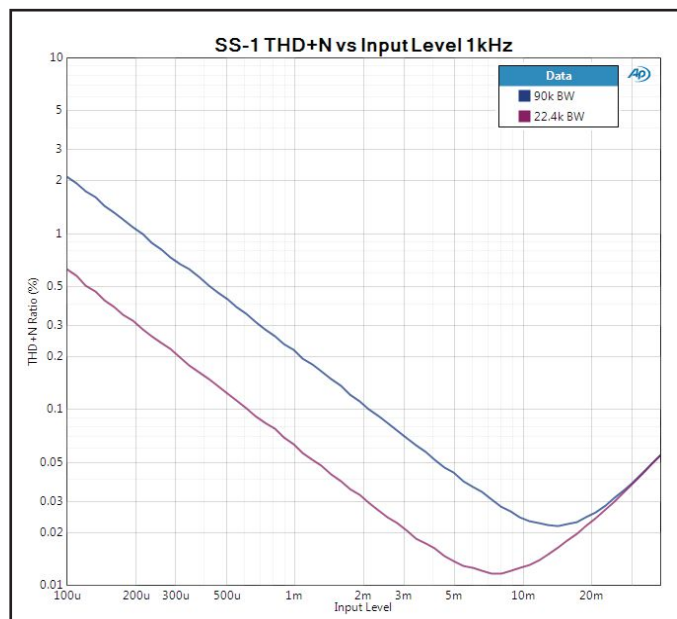


Figure 2: Total harmonic distortion (THD) plus noise vs. input voltage is noise dominated at levels below 7 mV in. There is plenty of headroom for loud signals.

One advantage of the construction and components chosen is a lack of microphonics. Monitoring the output while tapping vigorously on the SS-1, I saw no signal resulting from the mechanical excitation. In actual use, the preamps also showed no tendency toward noise or feedback, even when placed close to the PA monitors.

To determine noise rejection, I fed a common-mode signal (i.e., the same polarity and amplitude to pins 2 and 3 of the input XLR connector) of 2 mV to the SS-1. Compared to the same signal applied in (normal) differential mode, the output of the common mode signal was attenuated by 42 dB. I obtained an almost identical performance from the second unit. Here's where a transformerless open-loop design has a disadvantage compared to transformers or more sophisticated feedback-based designs, which can give 90 dB or more of common-mode rejection. I should note, though, that the 42 dB common-mode rejection held up from 20 Hz to 20 kHz. In practice, I did not encounter hum pickup

or other sorts of common-mode noise pickup issues, suggesting that the grounding and shielding of this design is well-executed.

Wrap-Up

I used the SS-1s for some field recording with my ribbon microphones, interposed between the mics and a Focusrite Scarlett 2i2 interface, which provided the phantom power. My mics have a sensitivity of 2.6 mV/Pa, so often need a boost unless used for a very loud sound source or very close up. The SS-1s came in particularly handy for an acoustic guitar recording where the mics had to be backed off because of a rickety (and resonant) stage area. The boost from the SS-1s let me set the input level controls of the 2i2 to near midpoint at a good recording level (occasional peaks at -2 to 3 dBFS), without issues of noise or distortion.

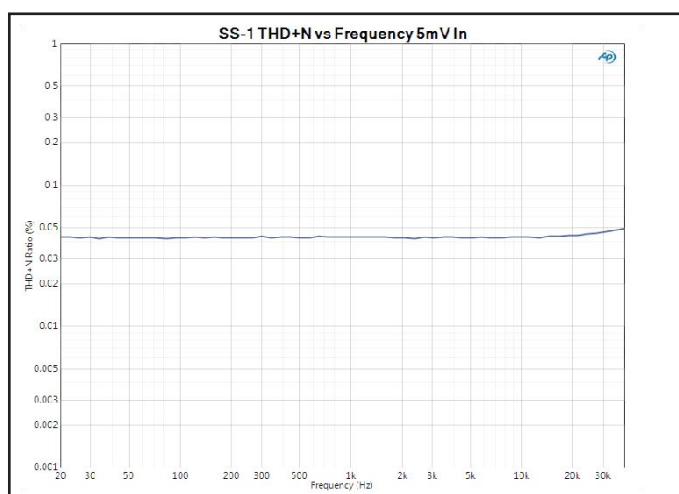


Figure 3: THD+N vs. frequency is basically flat across the audio band, indicating a lack of slewing distortion.

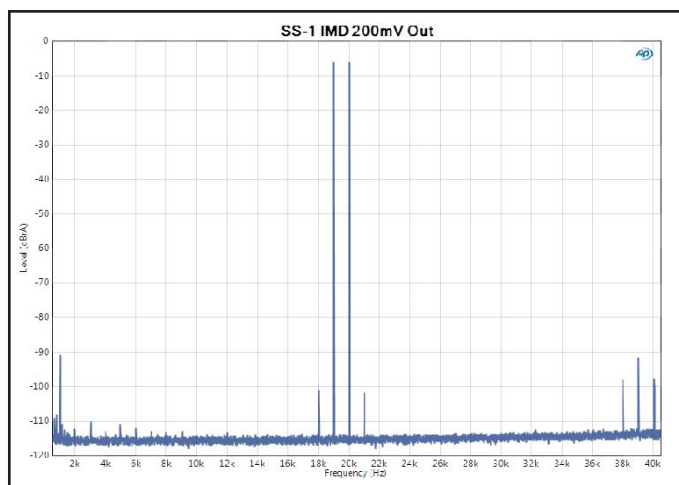


Figure 5: The 19/20 kHz intermodulation distortion spectrum is very clean and shows low-order products dominating.

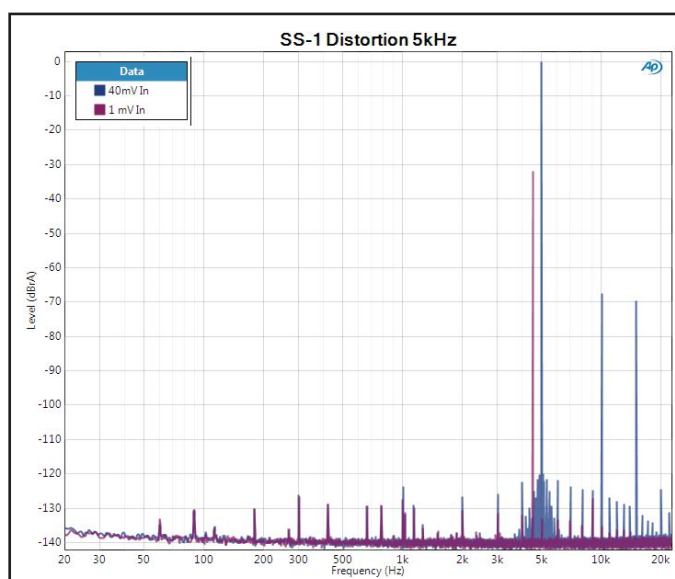


Figure 4: The spectrum of the SS-1 with two different input levels shows low levels of low-order distortion. The fundamental frequency of the signal was shifted 200 Hz between these measurements for clarity.

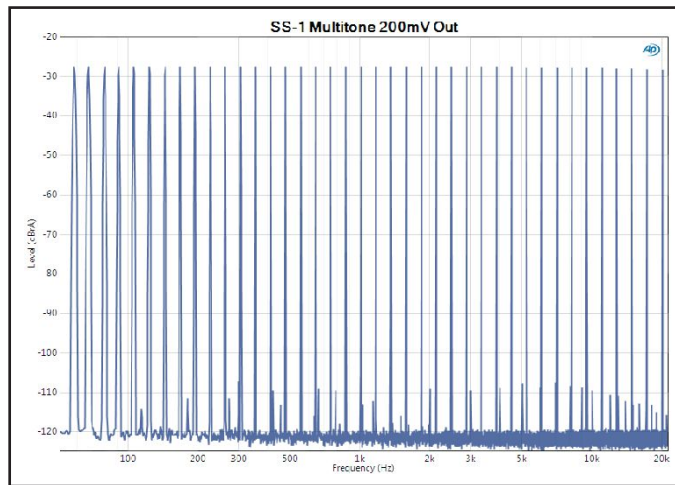


Figure 6: The SS-1 handles the difficult multi-tone test with good performance. Distortion products show as lines between the excitation tones are below -105 dB.

Regarding the noise floor, for the 2.6 mV/Pa sensitivity of my mics, the equivalent noise floor (20 kHz bandwidth, A-weighted) of the SS-1 is 18.6 dB SPL which, although not negligible, is significantly quieter than any room I record in, and only about a decibel higher than the self-noise of my mics. In practice, I could not hear the noise of the preamps over the recorded ambient noise.

The SS-1's solid construction will survive a lot of the banging around that is normal for gigs and location recording, and in my field use, the two SS-1 units performed silently and flawlessly. As the measurements suggested, there's no "sound" or significant coloration other than the small treble rolloff. The high overload margins will accommodate a range of mike sensitivities and the dynamics of live sound. A true state-of-the-art preamp could exceed the SS-1's performance, but at a much greater cost and questionable audible improvement. For a small studio or touring musician, the SS-1 is an inexpensive and useful accessory. 

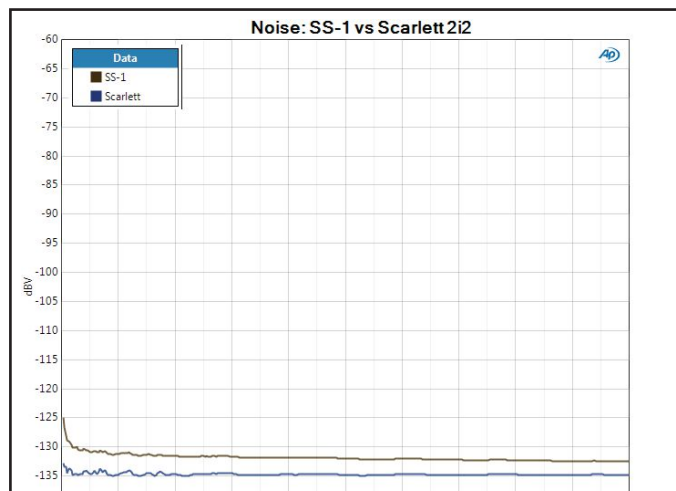


Figure 7: The noise spectrum of the SS-1 compared to the mic preamps in a Focusrite Scarlett 2i2 audio interface show a slight increase in noise floor in exchange for the gain boost.

Resources

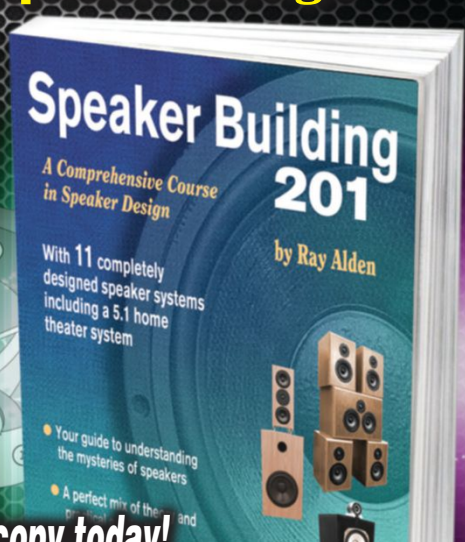
Focusrite Scarlett digital interface,
<https://us.focusrite.com/scarlett-range>.

The Simply Sound Company, simplysound.co.

About the Author

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

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Educated Reflections



I read the acoustics issue of *audioXpress*, August 2017, with great interest. It had a lot of information about how to tame room acoustics and get “rid” of early reflections. But this discussion brings out some very fundamental problems—or at least differences—in two camps, about how to do stereo in rooms. Both camps have a lot of support, so I would like to add something important to this topic.

By
Gary Eickmeier
(United States)

You see, there are a lot of folks with dipolar and multi-directional speakers such as omnis and directed radiation patterns for specific purposes. The many companies selling absorption products and the many articles on eliminating reflections leave these speakers out—speakers that are renowned for their spaciousness and depth and 3D imaging. Absorbing the reflections for these speakers would kill their main advantages for those users.

But what if we could “tame” those reflections and put them to good use in a way that added to the sound rather than hurting it? It has been said that there are good reflections and bad reflections, but it is hard to define which are which and why. Well, I claim to know, and I would like to have equal time!

Accepted Theories

The root problem, or difference, lies in stereo theory itself—if there even is one. I don’t believe I have seen an article or statement on precisely how stereophonic sound works. All we have is the practice of stereo, usually based on assumptions that have never been questioned. It sure seems sensible that the “accuracy” theory—or assumption—that all we want to hear is the direct sound from a pair of speakers—is true. I mean, what engineer in his or her right mind would question “accuracy,” right? But if the “accuracy” crowd is correct, then it must be that stereo works

by the direct sound alone, strictly by means of the lateral localization, like listening through a large portal into another acoustic space. Bathe our noggins in the golden glow of that arc of direct sound and we will just “hear” stereo in our infamous “earbrains,” right? That would be a head-related theory in which stereo is perceived at the head for a single listener (see **Figure 1**).

Well, I threw out the earbrain and the saber-toothed tiger more than 30 years ago, so rather than go through that camp, I think it would be more efficient to just state my case, which proposes “educated reflections,” or how to tame and place the reflections in a more comprehensive theory on how stereo really works—or should work. I call it the Image Model Theory (IMT). I first presented it at the Audio Engineering Society (AES) Convention in 1989 in a paper called “An Image Model Theory for Stereophonic Sound.”

The basic problem, or grand misconception, of the “accuracy” school—or head-related theory of direct sound imaging—is that it is blind to the effects of the room and the effects of radiation pattern and reflected sound. Stereo is a field-type system but the only field we have studied is the direct field. When we are sitting down, listening to stereo, we are not isolated from the playback room and, in fact, we need the acoustics of the playback room for the externalization

and placement of the sounds and the reconstruction of the various sound fields in the room. The channels are not isolated at the ears nor is the room kept out of the equation, but rather is important to the realism and spatial qualities contained in the recording.

The thinking is that “accuracy” defines everything about the reproduction, but it does not. The most difficult concept in audio is trying to put over what is the “object” that we are trying to reproduce. It is not microphone signals, it is an acoustic event that occurred in front of the mythical microphones. An acoustic event. The recording is only the carrier of the acoustical information, it is not the event itself. It needs to be translated from an electrical event to an acoustical event. That event has a certain size, shape, spatial, spectral, and temporal dimensions that are not encoded into the recording directly but need to be reconstructed during playback with certain assumptions and knowledge about those characteristics of the original event, using the speaker radiation pattern and positioning and, of course, frequency response. We get part of the way there by sending the amplified signal to two speakers, placed a certain distance apart in front of you. But that is not enough. There is much more to loudspeaker design and the reproduction of auditory perspective than the raw channels from the recording coming out of the front of those two speakers.

Stumbling Blocks

The first stumbling block is that live instruments radiate their sound in all directions in various patterns, and direct firing loudspeakers are nothing like that. They usually have a radiation that varies with frequency, the lows being completely omni-directional, the mids narrowing to a fan shape, and the highs narrower yet, and seeming to come from the tweeter only, mounted on the front of the cabinet. With a pure omni, the radiation is the same in all directions—much more like the real thing, the acoustical event that we are trying to recreate in front of us. With even more output to the rear, the sound goes more outside the speaker boxes and improves the direct to reflected ratio to become more like the live event.

The second stumbling block is that the ambience that was recorded must come from different, wider, incident angles than the direct sound or else it will be masked by the direct sound. This is basically why we prefer wider dispersion speakers in listening tests. Omnis and dipoles are automatically wide dispersion, and my kind of speaker opens up the sound to completely fill the front of the room.

These playback characteristics cannot be measured in the direct output of any speaker, nor do they have anything to do with the traditional concept

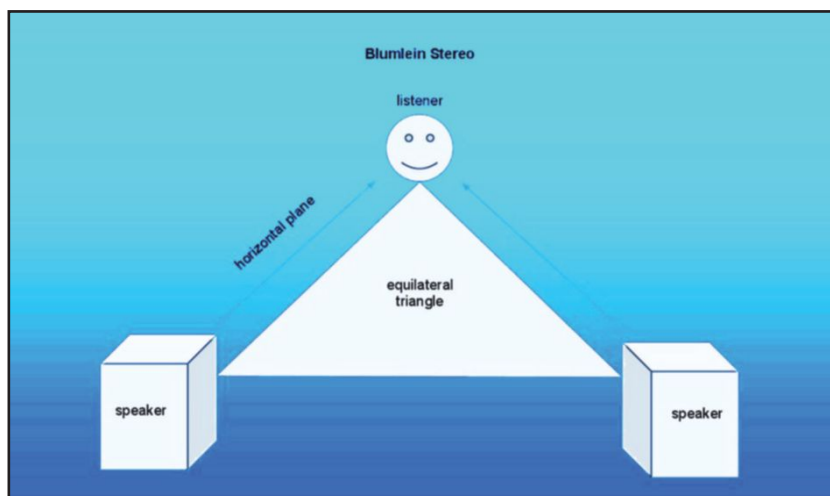


Figure 1: The Stereo Triangle—All that we know about standard stereo theory is the stereo triangle. You sit at the apex of a 60° triangle and listen in the direct field of two speakers. You are supposed to be able to hear into the stereo field from the microphone position.

of “accuracy” of that output. It is for these reasons that I would prefer the term “realism” to “accuracy.”

Cutting to the chase, what all of this means is that contrary to popular opinion we do not want to kill all of the reflections from the room or isolate the channels at the ears by cutting the crosstalk, nor is stereo an “accuracy” process in which we want nothing but the raw recorded channels reaching our ears. Heresy I know, but “hear” me out.

Reconstructing the Sound

Stereophonic sound is a concentrate, meant to be mixed with the acoustics of the playback room in a way that mimics the sound fields that were recorded. It is literally an electroacoustic process of re-presenting, if you will, or reconstructing, the live sound fields within your playback room. These are basically the direct, early reflected, and reverberant fields. The error of trying to do it with the “accuracy” model is that we end up squeezing all of those fields through the same acoustic points in space, the loudspeakers themselves, rather than positioning them spatially correctly on playback, the early reflected coming from a wide set of incident angles behind and beside the instrumental images and the reverberant field coming equally from all around us. The direct field should be only a small part of all of that, just as with live sound. Another way of putting it is that the direct sound theory is like a picture of another acoustic space, perhaps as wide as the speaker separation, and the IMT sees the reproduction as a macroscopic, 3D model of the soundstage that was recorded.

But it must all be done just right, which is where we start to get radical with all this (!). IMT says that the overall object is to come as close to the live model with all of those fields as feasible in a good playback room. If successful the two will be

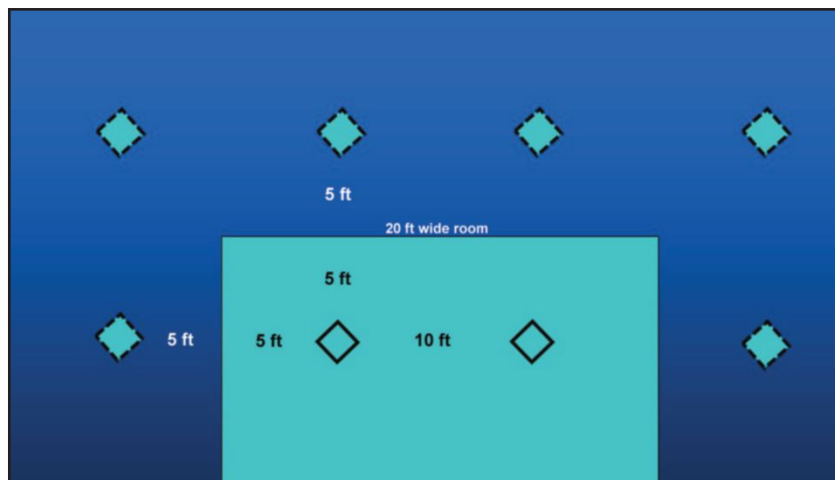


Figure 2: The Image Model—In the image model, the early reflections are illustrated with virtual sources on the other side of the walls. The model assumes specular reflectivity and an equi-omni radiation pattern with flat response in all directions.

geometrically similar in the placement and shape of all of the sound fields. The “model” part of IMT refers to the method from architectural acoustics of drawing the plan view of the reflected sound as additional sources from the other side of the walls, just like mirror images in a visual model. This gives a more visual, or understandable picture of the locations of all of the real and virtual (reflected) sources in any acoustical situation (see **Figure 2**).

The Spatial Nature of Sound

I have found a certain degree of difficulty communicating what I mean by the spatial characteristic of sound fields in rooms. Most of us are familiar with the frequency response effects of moving speakers around, or reflection effects in the

frequency domain such as comb filtering or bass reinforcement. But we must also pay attention to the spatial effects, which means the directions from which the various fields arrive. This is made easier with the image method of drawing reflections. You can easily see the locations of the speakers and all of their first reflections, and even the secondary reflections from the corners of the room. With a fertile imagination and some knowledge of the radiation pattern of the speakers you can also see the frequency response and strength of the reflected image sources. With direct firing speakers, the reflected sources would have a response of just lower midrange on down, and would be weaker. With pure omnis, the reflected sources would have a flat response and be stronger. With dipoles toed in at about a 30° angle, you can see how the corner secondary reflections give a deep, spacious early reflected field even though the side wall reflections are nulls (see **Figure 3**).

Bad Reflections

With speakers that have a strong rear output to their radiation pattern, you enter a different universe of effects of speaker placement that can be quite surprising. This is where we can easily stumble into problems with “bad” reflections. Most of us normally just think of direct sound when placing speakers for stereo. Well, we know that symmetrical placement is good, but exactly why?

Take a look at the example of placing your speakers wider to get a wider soundstage (see **Figure 4**). If you do this, the reflected sources move closer in, defeating the purpose. Similarly, if you place them farther from you, you would be placing them closer to the front wall, and the first reflections would come closer to you, decreasing depth of imaging (see **Figure 5**). Finally, if you are both too wide and too close to the front wall, you create a “clustering” of image sources that tends to pull the imaging to the two sides of the room, creating a hole in the middle and stretched and vague soloists (see **Figure 6**).

Good Reflections

To solve this problem and turn the bad reflections into “good” reflections, rather than kill them with absorption, what we want to do is even out the spacing of them from each other and from the speakers. I use a formula of a quarter of the room’s width in from the side walls and an equal amount out from the front wall. This gives a perfectly even lattice of direct and reflected sources with no hot spots or image clusters.

Image Model Theory

If we understand these effects we can then design

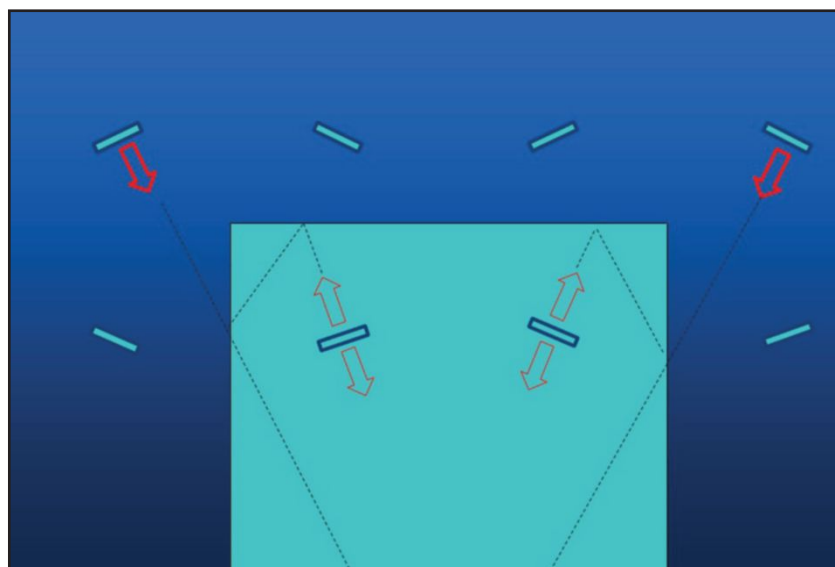


Figure 3: The Dipole Radiation—In the dipole example, if you know the speaker’s radiation pattern, you can visualize how it will sound with the image model. The corner secondary reflections come back wider and deeper than the primary sound, giving great depth and spaciousness, provided you don’t kill it first!

to them. The reflected sound can now be stronger than the direct output in order to “light up” the reflected sources to give greater depth and spaciousness to the soundstage. That is what you are hearing with multi-directional speakers. The big deal with IMT is to purposely design the speakers with a greater amount of rear output in a certain pattern, to mimic the Direct-to-Reflected (D/R) ratio of live sound and to get the imaging outside of the speakers themselves. We all enjoy the speakers’ disappearing act, and now we know what causes it. We have partnered with the reflected sources and educated them to help with the imaging rather than hinder it.

To show how radical a change in thinking this is, I can illustrate my playback system in contrast to what most are advocating. Where they have placed absorption at the first reflection points, I have placed mirror tiles. Where they are using directional speakers, I am using direct and reflecting speakers with a certain radiation pattern to cause early reflections in a certain pattern. These speakers have a greater portion of reflected sound output than direct, and the frequency response is the same in all directions. There is no “axis.” Where they might place speakers for frequency response or elimination of reflections, I am placing mine in a very precise way to neutralize any deleterious effects and enhance the reflections ability to help with imaging and spaciousness and make the speakers disappear, rather than call attention to themselves. I have a soundstage with imaging all across the front of my room, rather than just speaker to speaker. The speakers completely disappear, no sound ever seeming to come from the speakers themselves, even at extreme right and left.

It’s all done with mirrors—sound mirrors, that is—and specular reflectivity in the walls nearest to the speakers. The trick lies with the speaker placement and radiation pattern to create this huge, spacious, deep soundstage that is actually twice as wide and deep as the separation of the speakers. The loudness of the reflections combined with the placement and precedence of the actual speakers causes a harmless image shift toward the reflecting surfaces, placing the instrumental images in a plane behind the speakers and wider than the speakers (see **Figure 7**). The instruments seem to be coming from points in space rather than from a pair of speakers, and all of the frequencies are now coming from those same points, unlike with direct firing speakers where the lower and middle frequencies come from all around and the high frequencies come from the front of the speaker grills only. I have also incorporated Mark Davis’ distance/intensity trading of the DBX Soundfield One speakers for the frontal radiation pattern to keep the center imaging stable (see **Figure 8**).

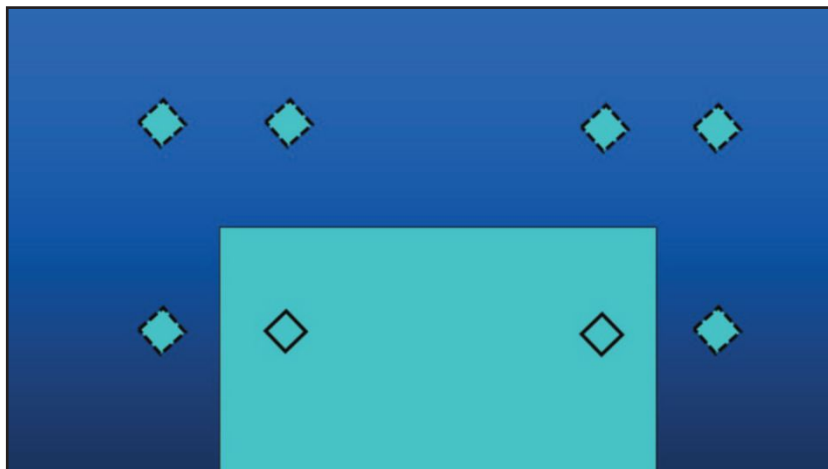


Figure 4: Place the speakers too wide and get side reflections that are actually narrower, defeating the purpose.

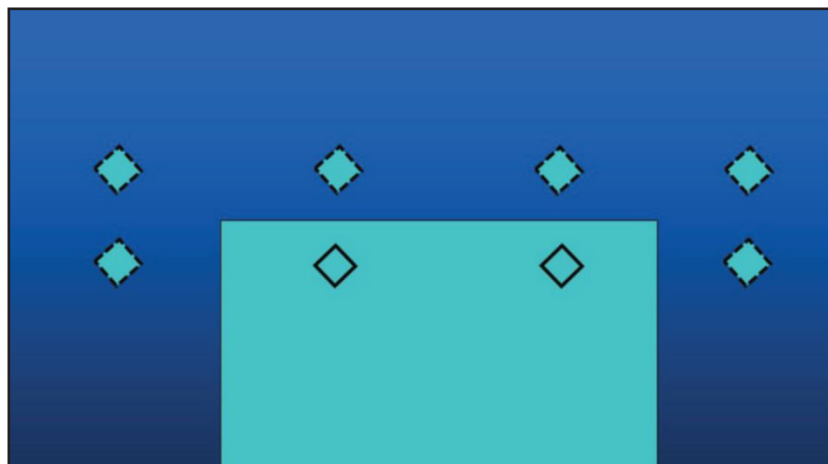


Figure 5: Place the speakers too close to the front wall, get early reflections too close to the actual speakers, reducing depth of image.

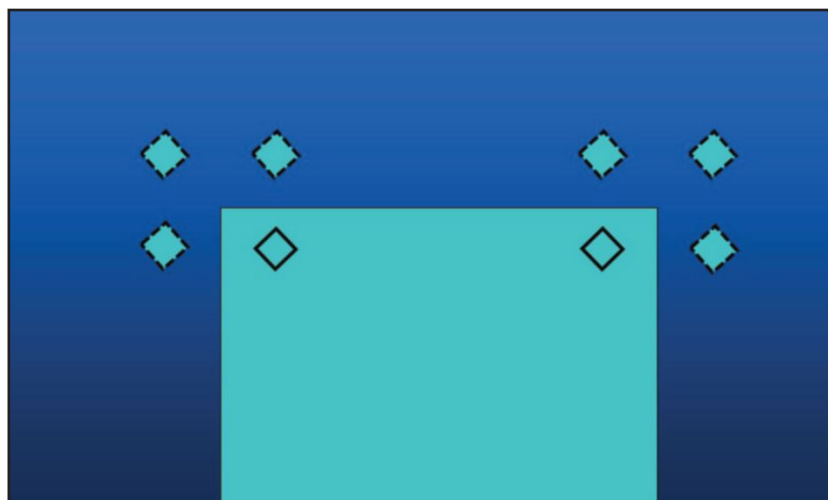


Figure 6: Place speakers too close to the corners, get a “clustering” of image sources pulling the soundstage apart, resulting in stretched soloists and a hole in the middle.

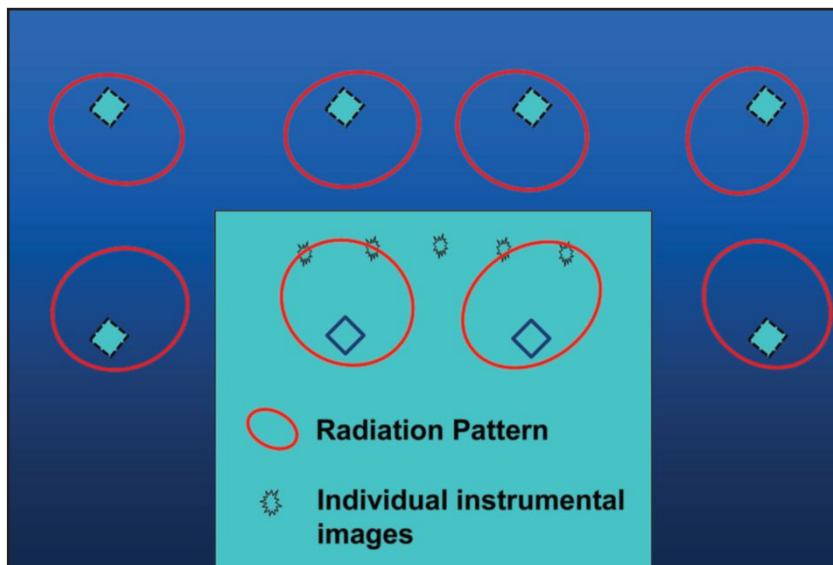


Figure 7: Greater reflected output “lights up” the virtual sources. Greater output to the rear gives greater depth of soundstage. A slight image shift to the rear makes loudspeakers disappear sonically, yielding individual instrumental images behind the plane of the speakers.

As far as “accuracy” goes, these “good” reflections do not have a negative effect on anything normally associated with the term. You have to ask first “accuracy” of what to what? In IMT, the model is not the measured output of the direct sound compared to the signal that went in. The correct paradigm should be accuracy of the acoustical result compared to what was recorded. A high direct field from two speakers in a dead room bears no resemblance to the live sound fields that went into the microphones.

Most live sound is comprised of a small direct field followed by a very large early reflected field from widely varying points all around the soundstage and the full reverberant field from everywhere. IMT restores that acoustic big picture. The reflections are not heard as echoes, smears, frequency response anomalies, or anything except spatial realism that is more like the real thing. This is because all of the sounds arrive well within the fusion time and the

more pertinent measurement is the total sound power output in the reverberant field at the listening position.

Speaker Design

My speakers have variable pots on the front two panels to adjust the radiation pattern to the room (see **Photo 1**). They are placed 5’ out from the side walls and an equal amount out from the front wall in a 20’ wide by 30’ deep room. The walls

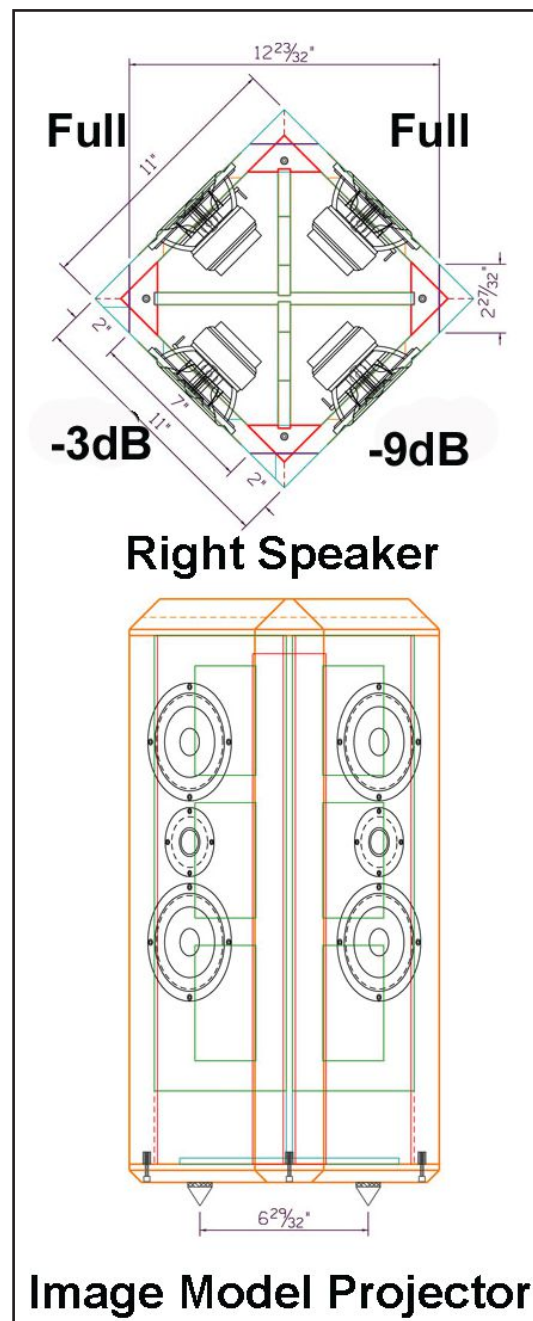


Figure 8: The Radiation Pattern of IMPs—Rear panel output is 6 dB louder than front panels and aimed 45° to either side. Compared to the rear panels, front inside is -3 dB, front outside is -9 dB.

About the Author

Gary Eickmeier is a graduate industrial designer (product design), but he has never worked in that field directly. He is a retired Air Force officer, navigation instructor, and teacher for 20 years. Gary is a member of the Audio Engineering Society (AES), Boston Audio Society (BAS), Southeastern Michigan Woofer and Tweeter Marching Society, and the Suncoast Audiophile Society. He has been published in several areas including *The Navigator* magazine, the *BAS Speaker*, *The Perfect Vision*, and a few pieces in various audiophile society newsletters. He also wrote an AES white paper presented in 1989. Gary is nearly 74 and lives in Lakeland, FL, with his wife Susan. He has discussed loudspeaker design and corresponded with many audio engineers including Floyd Toole, Amar Bose, David Moran, Howard Ferstler, Ethan Winer, John Atkinson, David Clark, Mark Davis, Tom Nounsaine, and many others. According to Gary, “Some agree with me, some don’t!”

have specular reflectivity at the speaker end, and then normal absorption and diffusion as you go back toward the listening position, for a normal reverb time of less than a half second, which is as recommended for this size room (see **Photo 2**). My imaging comes not from the speakers but from points in space in a region behind and beside the speakers, between them and the front and side walls. The speakers completely disappear, revealing a wide, spacious, deep soundstage from wall to wall with quite precise localization of individual instruments.

My statement about loudspeaker design is that the function of a speaker is not to be a direct radiator but rather to be an Image Model Projector. We must use the reflected sound as a partner in constructing a playback model that is more like the live sound, rather than fearing it!

I offer all of this to anyone and everyone to play with to your heart's content, to help add to the technology in a potential variety of creative ways. Some of my remaining questions are specular vs. some diffusion in the reflectivity, the ability to "shape" sound fields by means of various types of wall treatment and extra speakers, and continuing to play with radiation pattern and D/R ratios for the frontal soundstage. You can use stacks of small mini monitors and aim them and amplify them in various ways and try image modeling yourself, now that you understand all of the factors involved in the audibility of speakers and rooms.

Confirmation Tests

I am currently writing a paper for the AES to hopefully present next year. What I still need is some carefully documented listening tests to confirm my own impressions of how these sound fields can be manipulated with radiation pattern.

My speakers were designed with a biamping option that can separate the frontal radiation from the rear radiation to test for which ratio of direct to reflected sounds best. The front panels also have those volume pots to enable changes in the pattern between the inner and outer pattern shape. All of these changes are clearly audible and I am intensely interested in whether certain patterns are preferred by most listeners. I am predicting that with too much direct sound the imaging will gather at the speakers and be too forward and harsh. Changing the ratio to more reflected will at some point "pop" the soundstage outside the speakers deeper and wider without ruining the focus. Increasing the inner (crossfiring) direct sound panels will stabilize the center imaging. This is my personal experience from living with them for three years now, but it will need to be confirmed by others. 



Photo 1: The speakers were built by Dan Neubecker, a consultant for Parts Express, with variable pots for the front panels providing an adjustable radiation pattern.



Photo 2: Mirror tiles at all first reflection points enable you to actually see all reflected sources, plus help ensure sharp focus and flat response of those sources. This is a sat/sub system with a Velodyne F-1800 and a center and surround speakers for movies and more audio choices.

Resource

G. Eickmeier, "An Image Model Theory for Stereophonic Sound," Audio Engineering Society paper, October 1989, preprint 2869.

Active Loudspeakers with Dedicated DSP Are the Next Step in Sound Quality

An Interview with Floyd Toole

After a professional career in acoustical and psychoacoustical research, Floyd Toole is now dedicated to writing about and teaching what he knows. "In general I do what I can to promote a scientific approach to the design and use of loudspeakers and rooms. Ideally, if this is done in both the professional and consumer sides of the audio industry, it can lead to improving the probability that consumers will hear the 'art' as it was created. As things are, that probability is depressingly low."

By
Shannon Becker

(United States)

SHANNON BECKER: What sparked your initial interest in the audio field? How did it all begin?

FLOYD TOOLE: In the 1950s, my father bought a Midwest radio chassis, a multiband tuner with a 10 W push-pull power amplifier—impressive at the time. He then constructed a custom cabinet for it and a record changer. He was a good hobbyist cabinetmaker. The single 10" field-coil loudspeaker was in a small open-backed cabinet. It had no bass and sounded crummy, so there began a quest for a separate loudspeaker enclosure with a better loudspeaker. With no expertise and only the guidance of opinionated enthusiast magazines several elaborate enclosures were built, and a couple of them actually sounded better. I was in my teens and became intrigued by this and audio electronics. I built more powerful amplifiers and more elaborate preamplifiers from scratch using published designs. Later, I built a few kit components. The sound quantity and quality improved, but it was clear that the loudspeaker was the weak link. I was hooked,



Floyd Toole is a leading expert in the field of sound reproduction. In 2013, he was presented with the Audio Engineering Society Gold Medal Award in recognition for his outstanding contributions to theory, practice, and international standards in the area of subjective and objective evaluation of loudspeakers in rooms. Toole also received the 2015 CES (now CTA) CE Hall of Fame award.

and went on to study electrical engineering at university thinking that I might become part of a new wave of designers—the transistor had just been invented. I went on to post-graduate studies, changed my focus to auditory perception, got a PhD, and embarked on a lifetime of scientific research.

SHANNON: Can you share some of the greatest challenges involved with some of your designs over the years?

FLOYD: As a research scientist my "product" was knowledge, but since the topic was sound reproduction I could not avoid contact with loudspeakers and what made them work. I did a few designs to test ideas, a couple of which are documented in my recent book, but they were never intended to be commercial. The greatest challenge, and the motivation for the next 50 years of research, was the lack of understanding between what we measured and what we heard. It is very rewarding to look back at what has been learned about how to

do more meaningful measurements on the one hand, more revealing listening tests on the other, and at the psychoacoustic relationships that have been established between the two domains. The result is that loudspeakers can now be engineered to meet well-defined technical performance targets, and the result is neutral, transparent sound. Knowing how to do it means that those qualities can be designed into products at all price levels.

SHANNON: Having worked in loudspeaker sound reproduction evaluation, and the intersection of electro-acoustics, acoustics, and psychoacoustics, how do you think the concept of “accuracy”—as it is described in studio recording applications—translates to the consumer listening experience?

FLOYD: In 1965, I began work as a research scientist at the National Research Council of Canada (NRCC). I needed a loudspeaker to use in anechoic sound localization experiments, and used the chamber to measure a few highly regarded “hi-fi” loudspeakers of the time. The differences were enormous. Naturally, I set up a blind, equal loudness, listening test to see what listeners might prefer. Was it true that we all had distinctive tastes in sound quality? I was curious. The result was a surprise to me and my colleagues. There was absolutely no disagreement among listeners as to the preferred loudspeaker, and it was the one that exhibited the flattest, smoothest, anechoic measurements. That was 1966, and nothing has changed since, except that we have become much better at making and interpreting measurements, and listening test methods have greatly improved.

Throughout the subsequent 50 years of research it has been the subjective opinions that drove the research, and every incremental improvement in subjective ratings of sound quality has been correlated with reductions in the linear and nonlinear distortions in loudspeakers. Amazingly, in double-blind tests, most people, most of the time, gravitate to “neutral” (i.e., uncolored, accurate, transparent) loudspeakers. The historical problem in the audio industry has been that there have been no standards for monitor loudspeakers in recording control rooms, and no standards for consumer loudspeakers. So, recordings are variable and the playback of those recordings is an unpredictable experience. This is what I call the “circle of confusion.” Fortunately, enough people are aware of the science that the situation is improving. If recordings are mixed and mastered using neutral loudspeakers and consumers have neutral loudspeakers there is a higher probability of us hearing the art as it was created.



The Toole art gallery/entertainment room shows that superb sound is compatible with “non-technical” interior decor if one begins with properly designed loudspeakers. Some “stealth” acoustical treatments and multiple subwoofers finish the job. The dashed lines outline the 10’ drop down front-projection screen, in front of the 65” direct view display.

SHANNON: What did you include in your own ultimate sound system/listening room?

FLOYD: I have examples of powerful, “neutral” loudspeakers in a 7.1 system that is soon to be elaborated into a 9.4.6 immersive system. It employs Sound Field Management, using four subwoofers in a signal-processed scheme that effectively attenuates room modes. Bass is deep, uncolored, and “tight.” Right now it consists of two Revel Salon2s, one Voice2, six Gem2s, and four JBL HTPS400 subs.

SHANNON: What do you attribute your success to as an accomplished acoustic engineer?

FLOYD: I have had the good fortune at the NRCC and at Harman to have had access to excellent technical measurement facilities and the budget to construct elaborate positional-substitution listening facilities. One needs reliable subjective and objective data. This was combined with the patience to dig into

existing science and to ask good questions about what needs to be done to improve the situation.

SHANNON: You have had a long and illustrious career, what do you see as your own greatest achievement?

FLOYD: Without doubt, with respect to loudspeakers, establishing that there is a logical and convincingly demonstrable correlation between what we measure and what we hear, which led to being able to anticipate perceptibly “neutral” sound quality from the right set of processed anechoic measurements. To address the residual problem of overly resonant bass in small rooms, my research colleagues and I have employed multiple subwoofers in passive and active, signal processed, strategies to manipulate room modes, attenuating resonances and delivering more similar bass to multiple listeners.

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

FLOYD: I remember the transition from mono to stereo. That was a huge advance, perhaps the most significant innovation. Stereo and the much more rewarding multichannel formats were motivated by movies, not music. Music is still stuck in the stereo rut. Loudspeaker transducers have certainly improved over the decades, but I see the next necessary step in sound quality as active loudspeakers with dedicated DSP. It is commonly used to improve sound from inexpensive, small

computer and Bluetooth loudspeakers, but is rare in audiophile products. Sound quality can be improved and, if desired, made more consistent in production. Steerable arrays could permit customers to experiment with different sound fields, although multichannel options including upmixed stereo are, in my opinion, more powerful and more rewarding.

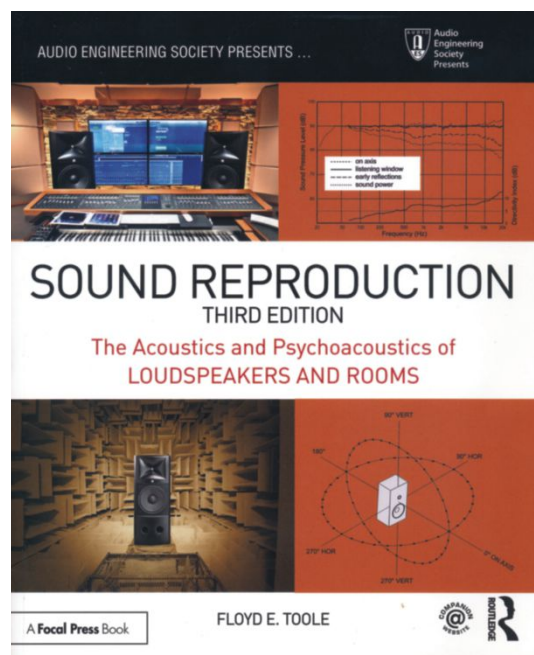
SHANNON: Where do you see the audio industry being in 10 years?

FLOYD: From a scientific perspective we have passed the point of diminishing returns. If what we now know is competently applied to the design of loudspeakers, the sound that emerges will be sufficiently neutral and transparent that what we hear is what went into the recording—for better and for worse. Even the pesky bass problems in small rooms can be significantly tamed. The circle of confusion, a technical issue at its heart, remains, and is exacerbated by audio professionals with deteriorated hearing—it is an occupational hazard. Therefore, recordings are variable.

What I see happening around me is depressing. The knowledge of how to create stunning reproduced sound exists, but because people in general don’t read, and many don’t believe in science, the average consumer ends up living with inferior sound when the same money could have purchased more. Countless hours in audio forum discussions are no substitute for a few days reading peer-reviewed science. I see the same questions being asked over and over again, when the answers existed decades ago. Unless consumers have access to trustworthy data on loudspeakers, we will continue to live in an audio world dominated by questionable opinions, not facts. I do not see this changing. Add to this the (understandable) attractions of portable A/V, small wireless everything, and superb audio is an even more remote possibility. However, people are still listening to music, so perhaps there will be a renaissance in “hi-fi.” In the meantime the masses are attracted to “invisible” loudspeakers, and far too many audiophiles follow faith-based notions that the answers to perfect sound lie in distractions like power cords, speaker wires, and exotic electronics. This is money not well spent.

SHANNON: Do you have any advice for *audioXpress* readers who are thinking of building their own sound systems?

FLOYD: Read my book *Sound Reproduction—the Acoustics and Psychoacoustics of Loudspeakers*



In 2017, a much-revised third edition of Floyd Toole’s book, *Sound Reproduction*, was published. His advice: The most perfect loudspeaker will not always sound “perfect”—it can only reproduce what goes into it, and that is definitely variable.

and *Rooms*, Third Edition (Focal Press, 2017), follow its guidance closely. Get familiar with a good acoustical measurement system. You can measure “good sound” but as you will find, except at low frequencies, the answer is not in “room EQ.” Then, relax and enjoy the abundant audio art that is available. The good news is that some superb transducers are widely available; it is not like my DIY years. Always remember that the most perfect loudspeaker will not always sound “perfect”—it can only reproduce what goes into it, and that is definitely variable. We still need tone controls.

SHANNON: Are you currently working on or planning any projects?

FLOYD: I am currently finishing up the first contribution to the publisher’s website for my book. It contains supplementary guidance for designing, equipping, and calibrating home listening venues.

SHANNON: Is there anything else you would like to share with our readers?

FLOYD: Demand that loudspeaker manufacturers reveal more useful performance data—if they have the nerve to do so. The process exists as an industry standard: ANSI/CTA-2034-A (2015). “Standard Method of Measurement for In-Home Loudspeakers,” Consumer Technology Association, Technology and Standards Dept., www.CTA.tech.

The result is a family of computer processed anechoic curves, which has come to be called the spinorama. It is based on a measurement method I initiated at the NRCC in 1983, and which got refined by the Harman research group. It is used by a few manufacturers (e.g., KEF), at least one audio webzine (Audioholics.com) and is an optional data output format from Klippel acoustical measurement equipment, a feature that others are considering. There may be more—there should be more!

As is spelled out in my book, from these data it is possible to predict listener preferences with high probability. In short, the most important information one needs is in this visual presentation of loudspeaker performance. Consumers need to be able to see it. An engineering degree is not needed for interpretation. 

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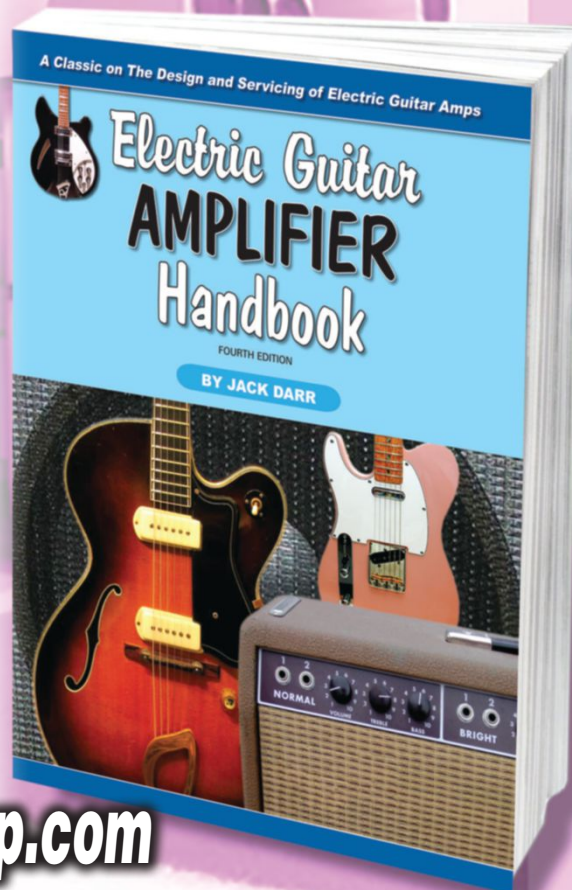
Jack Darr’s *Electric Guitar Amplifier Handbook* details the following:

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Omni-Directional Coaxial Tower

Natural Sound Propagation



Photo 1: This is my completed set of Omnihex enclosures.

With all the talk about tiny 360° smart speakers invading modern homes, how about an alternative that actually delivers natural sound propagation in every direction? In this article, speaker builder Ken Bird details a simple project for an omni-directional coaxial system, featuring Goldwood 8004 drivers from Parts Express.

By
Ken Bird
(United States)

The omni-directional speaker has been around since the 1950s, surfacing occasionally in different formats over the audio decades. We are now witnessing its revival in the large number of all-in-one compact omni-directional wireless systems on the market today. Many classical music listeners say omni-directional sound creates a more realistic sound stage for large orchestras, organs, or massed voices. Critics do not like them for musical ensembles or soloists, as they say imaging is not as good as a direct reproducer. Room acoustics and placement can affect the perceived sound and, in most instances, you don't have to be in a sweet spot to enjoy the music with an omni-directional system.

My Omnihex system offers the advantages of

omni-directional sound using a coaxial 8" speaker in a ported hex-shaped tower. The six-sided hex shape makes a larger volume enclosure with about the same footprint as a box-type construction and has acoustical advantages. The sound is dispersed through a reflector mounted in the top baffle assembly. This is a more challenging type of construction for the woodworker but the end result is pleasing, both in sound quality and esthetics (see **Photo 1**).

The Driver

The system needs a coaxial 8" driver and there are several choices available. Many are coaxial with a separate tweeter while a few use a whizzer cone

to achieve high-end reproduction. You can spend from \$15 to several hundred dollars for such a speaker. After looking at cost, specs, and some simulation testing, I chose the Goldwood 8004 from Parts Express. It has good power handling, decent sensitivity, a curvilinear cone, and a neodymium tweeter for the highs. This speaker is usually used in distributed sound systems as a ceiling speaker, and with its high Qts, the driver works well in the mass loaded ported enclosure of the Omnihex (see **Photo 2**).



Photo 2: I used the Goldwood coaxial driver in the Omnihex system.

Constructing the Enclosure

The choice of the wood material is up to the builder. Since I decided to paint the enclosures rather than stain them, I used 1' x 6" dimensional select pine for the sides. I made the baffle, the top, and the bottom from 3/4" plywood. The top is supported by 1/2" dowel rods with only one end glued into the baffle top to allow the speaker to be easily removed. The bottom is a piece of 3/4" plywood 12" in diameter. No crossover network is needed as it is built into speaker and wiring is a simple connection to a speaker cup terminal (see **Figure 1**).

This project requires 30° angular cuts be made in the panel edges that make up the tower's six-sided hex shape. These can be done with a table saw set to the proper angle or by using a handheld router with a 30° router bit. I constructed a simple assembly jig to hold the panel in place while being nailed (see **Photo 3**).

After assembling half of the enclosure's side panels, line each one with a piece of 1/4" thick

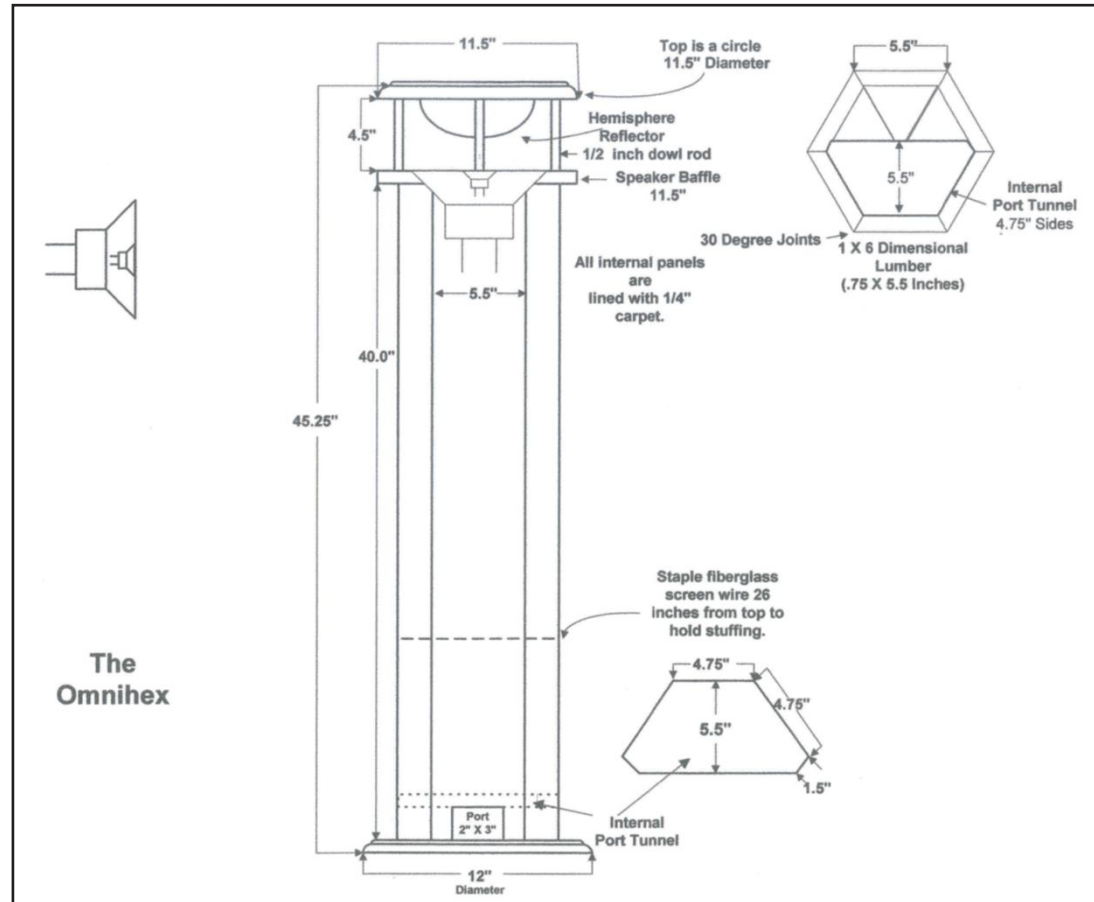


Figure 1: The Omnihex design does not require a crossover network, making wiring a simple connection to a speaker cup terminal.

indoor/outdoor carpet. The material is available from any big box home improvement store and is not expensive. I cut the pieces 4" wide and 39" long and stapled them to each panel (see **Photo 4**). The rear panel will have a 3" diameter speaker terminal cup cut out and the front panel will have the port notch centered and cut 3" x 5".

I assembled the panels using the jig and a nail gun with 1" nails. Place a bead of wood glue or liquid nails along the edge of the panels and then nail them together. The jig produces three of the needed six

sides and then they in turn can be glued and nailed, placing them on top of each other. When removed from the jig, you will have half of the enclosure side assembled. Run a bead of window caulking down each inside seam of the panels, then set the half sides on top of each other and glue and nail them together (see **Photo 5**). Cut the port tunnel as indicated in the drawing and mount it just above the port opening (see **Photo 6**).

Once assembled, caulk the remaining seams inside the enclosure. Check to see if there are any

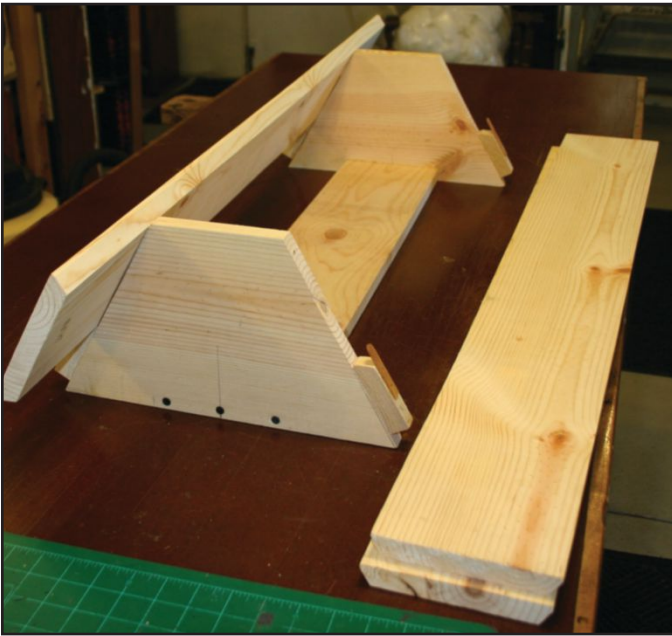


Photo 3: I assembled the enclosure sides on the jig.



Photo 4: Staple the indoor/outdoor carpet to the inside surfaces.

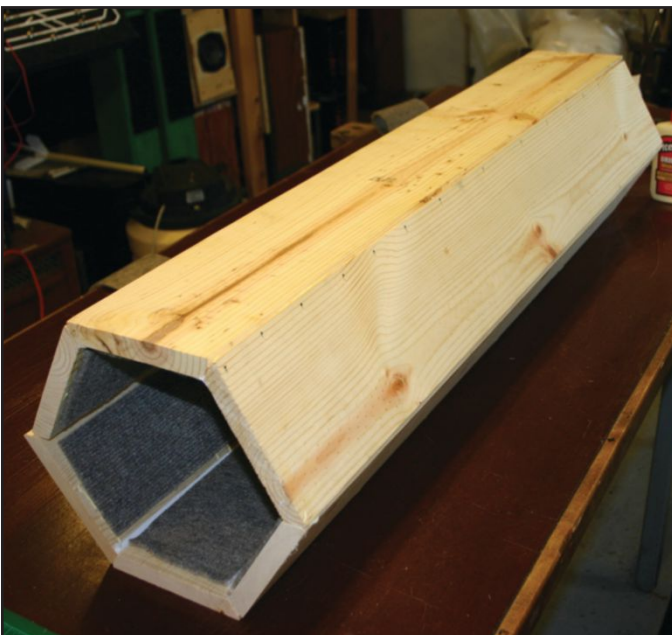


Photo 5: The assembled enclosure is a hexagon tube.

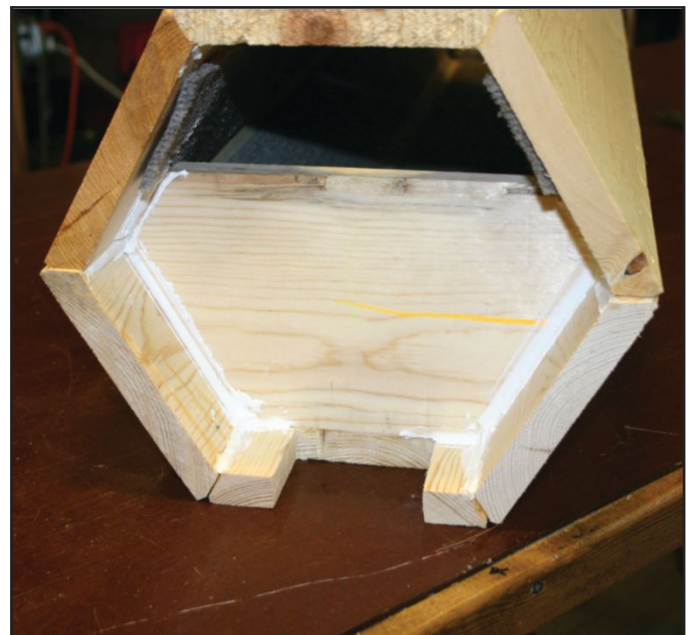


Photo 6: The port tunnel is installed in the tube bottom.

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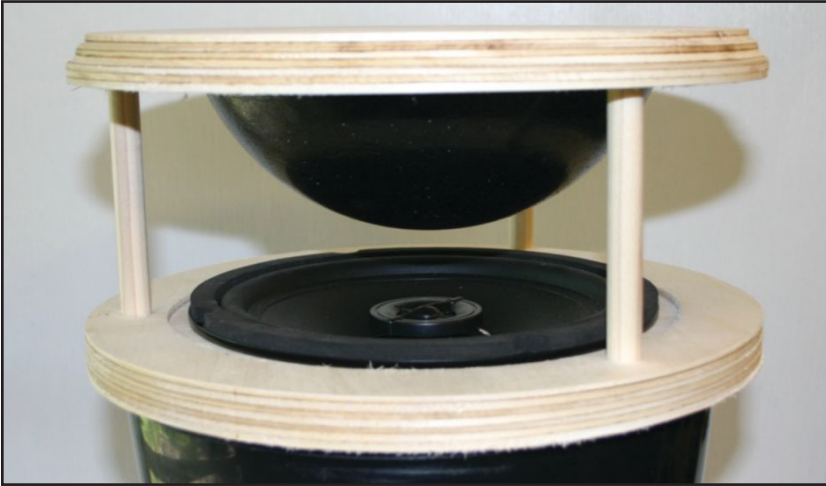


Photo 7: The raw prototype top is shown with the reflector and speaker placement.

nail heads at the surface of the panel. If so, use a nail set and hammer to indent them and then fill the holes with wood filler. Let the enclosure set for a few hours before sanding.

The Baffle and Sound Reflector

Mount the speaker horizontally in the enclosure top on a 3/4" piece of plywood 11.5" in diameter. The hole cut out for the speaker is 7.25" and should be indented 1/8" into the baffle plate. I used a Jasper circle jig and router for all the circle cutting. The top of the baffle is another piece of 3/4" plywood that is supported above the baffle plate by 1/2" hardwood dowels, 4.5" long. The dowels are mounted into 3/8" holes with only those attached to the top being glued in. To allow for easy removal of the speaker, the top is held in place by inserting the dowel rods into the matching holes on the baffle plate.

Tests with a sweep oscillator did not reveal any rattles or other noises due to the wedge mounting. Staple a 48" piece of 18-gauge speaker wire from the terminal cup mounting hole to the baffle board. I made the hemispherical reflector from half of an 8" Styrofoam ball found at a craft store. I further modified the half ball by cutting 1.5" off the bottom. Then, I painted it with non-oil base gloss black to provide a hard reflective surface. The reflector can be glued in place or held with pieces of Velcro matching strips (see **Photo 7**).

Final Assembly

After sanding all the pieces, start assembly by

first attaching the bottom piece. Turn the enclosure upside down and run a bead of Liquid Nails adhesive along the bottom panel surfaces. Then center the bottom piece and nail it in place. Remove any excess adhesive and let the enclosure sit for an hour or so to set the adhesive. Then turn the enclosure upright and do the same with the baffle plate, applying the adhesive along the top edges of the enclosure. I stapled a 3" wide piece of fiberglass screen wire



Photo 8: The assembled enclosure is under test in the shop.

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.

across the interior of the enclosure about 26" from the top. This will prevent the needed polyester stuffing from falling to the bottom. The fancy finial on the top is an optional addition (see **Photo 8**).

The type of finish you apply to the enclosure is your own choice. This version was first given a coat of wood primer and sanded with fine paper. Then, I painted it with a non-oil base enamel.

Once the finish is complete, it's time to wire, stuff, and install the speaker. Solder the speaker wire, in the proper polarity, to the speaker terminal cup. Before mounting the cup, install a piece of 3/8" foam insulation strip around the rear edge to assure a good seal.

Fill the enclosure with half a 16 oz. bag of polyester pillow stuffing. Tease out the stuffing to reduce its density. The stuffing should occupy only the first 25" of the

enclosure top with the bottom left empty.

Then prepare the coaxial speaker by first installing a piece of the 3/8" foam insulation strip around the rear edge of the speaker mounting lip. Solder the 18-gauge wire to the speaker, again making sure the polarity matches that of the terminal cup. Mount the speaker to the baffle plate using 1" drywall screws and then install the reflector top. Perform a sound check to listen for rattles or other non-musical sounds. The omni-directional enclosures work well at least 10' apart and better if 12' to 15' apart. Corner placement, about 24" from the walls, will enhance the bass and further spread the midrange and highs.

Omni-directional sound can be compelling, especially if you are a fan of large orchestras, cathedral organs, and mass choruses. 

Optional Coaxial Speaker

While the Goldwood speaker worked fine in the system, I tried another approach by building my own coaxial speaker using an old 8" that I had in stock along with a Goldwood tweeter and a single 4.7 μ F high-pass filter. I used 1/8" aluminum bar stock to fashion the necessary mountings to attach the tweeter and clear the surround area (see **Photo 9**). The speaker should have a Qts of 0.4 or above and an Fs of 40 to 50 Hz. No resistive padding of the tweeter was necessary as the sound diffuser helps to balance the woofer-tweeter sound pressure level.



Photo 9: This is another coaxial speaker made from a Goldwood dome tweeter and an Audio Concepts 8" raw driver.



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Circuit Design for a Small Bass Amplifier

By
Richard Honeycutt
(United States)

In this article, Richard Honeycutt attempts to bring his lightweight Bolger amplifier concept to life by designing the circuitry, which has the triode stages limit the maximum signal fed to the tone module and to the power amplifier module.

In *audioXpress*' November 2017 Hollow-State Electronics article, we examined the concept of creating the Bolger, a small amplifier for the electric bass. Comparing tube and transistor technologies for bass amps led to the conclusion that the main advantage of tubes in a bass amp is their ability to provide graceful limiting to avoid razzzy distortion if the amp is overdriven. We considered using an available prefabricated op-amp-based tone control module and a Class-D power amplifier stage. The circuitry will be designed so that the triode stages limit the maximum signal fed to the tone module, and hence, to the power amplifier module. This topology will minimize weight while providing soft tube limiting. **Figure 1** illustrates the block diagram we conceptualized for the Bolger.

The Circuit

Figure 2 shows a first draft of a circuit for the hollow-state preamp/clean limiter stage, taken from a distortion pedal design described in a previous Hollow-State Electronics article. The tubes are 12EL6 "space-charge triodes," designed for use in car radios, where they could be operated from a

12 V battery without the need of a vibrator-type power supply.

In the previous design process for the distortion pedal, I found that the input resistance of a space-charge triode in a common-cathode configuration is far too low to get good sound from a guitar pickup, so I preceded the tube stage with an OP27 buffer to provide a 1 M Ω input resistance. A cathode-follower stage had to be added after the common-cathode stage to drive the input of a typical guitar amp, since space-charge triodes tend to need very large plate resistors, and thus have high output resistances. To drive the triode stages hard enough to get good distortion, we added a gain of 10 in the OP27 buffer. A 1N5292 constant-current diode (CCD) was used instead of a plate resistor for the common-cathode triode stage, to obtain more gain than a plate resistor would provide, since a CCD acts as if it had an infinite dynamic resistance. To permit adjusting the distortion signature to the owner's preference, adjustable fixed-bias circuits were used for both triodes: R2 controls the bias of the Q1 stage, and R8 adjusts the bias of the cathode-follower.

Different B+ voltages were used in the distortion pedal than in the first-draft of the Bolger tube stages. Although space-charge triodes were designed to operate with a 12 V B+, the 12EL6 is rated for a maximum plate voltage of 30 V. For use in a distortion pedal, my choice to use a 24 V B+ worked well, although finding a 24 VDC wall-wart that was not a switching supply proved to be a challenge. The switching supply's output was much too noisy for an effects pedal.

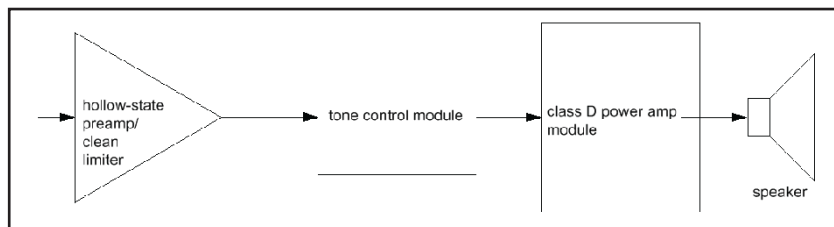


Figure 1: The Bolger concept design is based upon this topology.

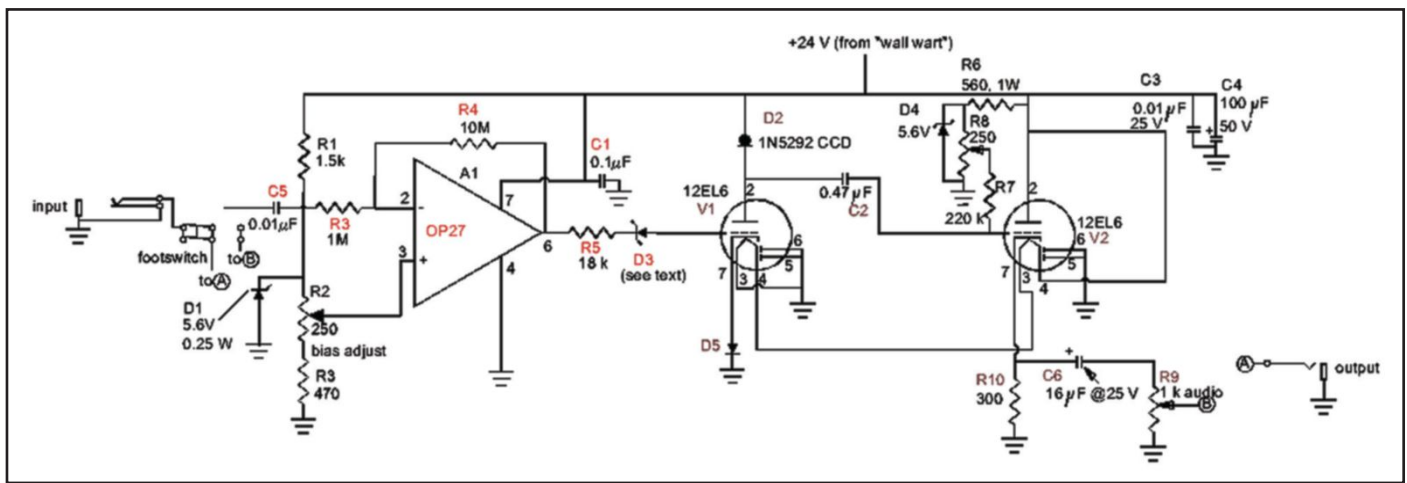


Figure 2: This is the initial hollow-state/clean limiter design.

Several modifications to this circuit make it more appropriate for its function in the Bolger. First, as a more-or-less linear amplifier, the tube stages worked well with a 12 V B+. This was derived from a 78M12 three-terminal regulator fed from the 24 V switching power supply used for the power amplifier. Both the 78M12 and the power amplifier have enough supply-noise rejection to tame the obnoxious behavior that a switching supply created in the distortion pedal.

Second, since the tone-control board has a built-in +12 V bipolar power supply that could easily spare the <20 mA needed by the buffer amp, the OP27 circuit became simpler. Third, the 1N5292 is no longer available in small quantities from any source I could find, so I went back to a 1 MΩ plate resistor for the cathode-follower stage. Fourth, since the input resistance to Q2 can be as low as 10 kΩ, depending upon grid biasing, the coupling capacitor from Q1's plate to Q2's grid was changed to 0.47 µF, which provides a low-frequency cutoff of 30 Hz. Adding in the effect of C6 gives an overall cutoff of 40 Hz, preventing low-frequency bumps from bottoming the speakers. And fifth, powering

the OP27 from a bipolar supply eliminated the need for level-shifting diodes D3 and D5.

Typical Operation

Interestingly, the datasheet of the Sylvania 12EL6 includes a table showing "typical operation." This table describes a common-cathode amplifier stage having a grid-leak bias resistor of 1 MΩ, a plate resistor of 1 MΩ, a B+ of 12.6 V, with the stage feeding a following stage whose grid resistor is 2 MΩ. The mu and plate resistance of the triode in this circuit are given as 55 and 45,000 Ω, respectively. Based upon these values, we would expect the stage gain to be:

$$A_v = \frac{\mu R_L}{r_p + R_L} = \frac{55 \times 670,000}{45,000 + 670,000} = 51.5$$

(The R_L value of 670,000 is the equivalent resistance of the 1 MΩ plate resistor paralleled with the 2 MΩ grid resistor of the following stage.) However, the table also states that the gain of the stage is 16. The circuit of the following stage is not specified, but if the following stage is also grid-leak biased, and its input resistance is about 18 kΩ, the lower

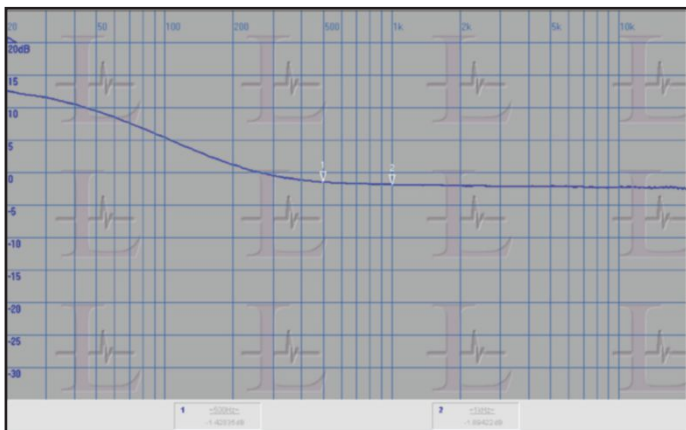


Figure 3: The bass turnover is about 200 Hz and maximum boost is about 12 dB at 40 Hz.

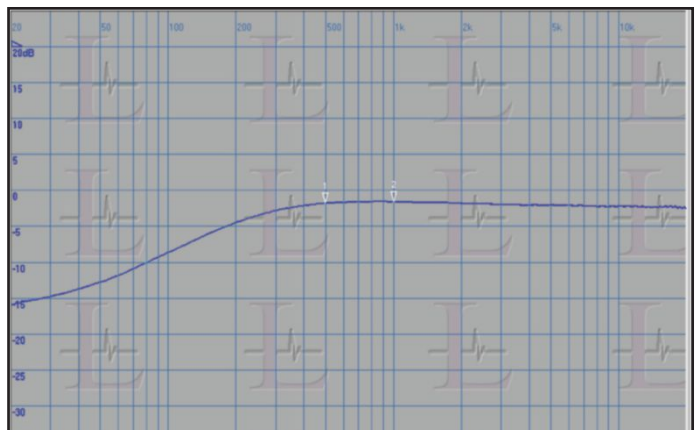


Figure 4: Maximum bass cut is about 13 dB at 40 Hz.

R_L would cause the calculated gain of the first stage to be close to 16. The principle of operation of a space-charge tube is such that for cathode biasing to work, the grid resistor's value has to be too low to provide an input resistance as high as we need it to be.

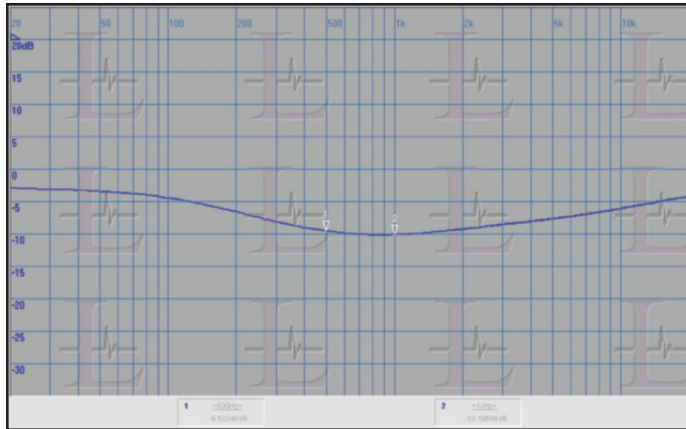


Figure 5: Maximum mid cut is about 10 dB, centered about 800 Hz.

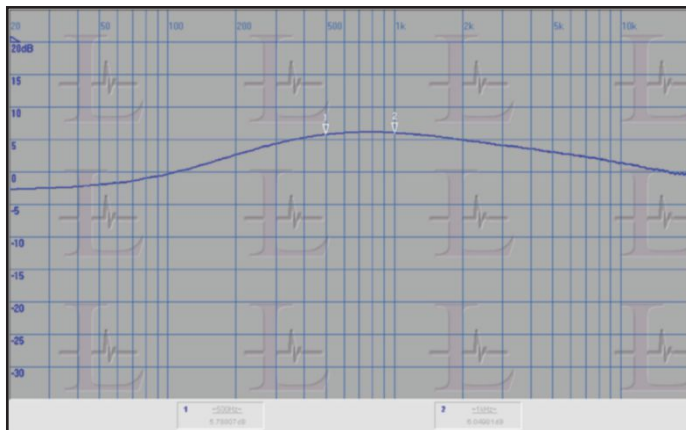


Figure 6: Maximum mid boost is about 7 dB, centered about 800 Hz.

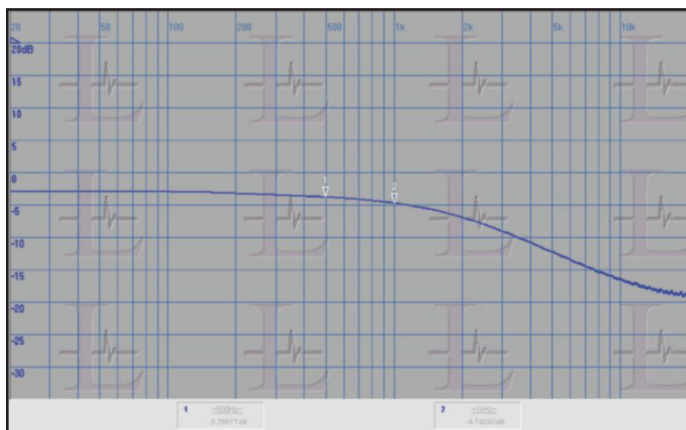


Figure 8: Maximum treble cut is about 14 dB at 12 kHz.

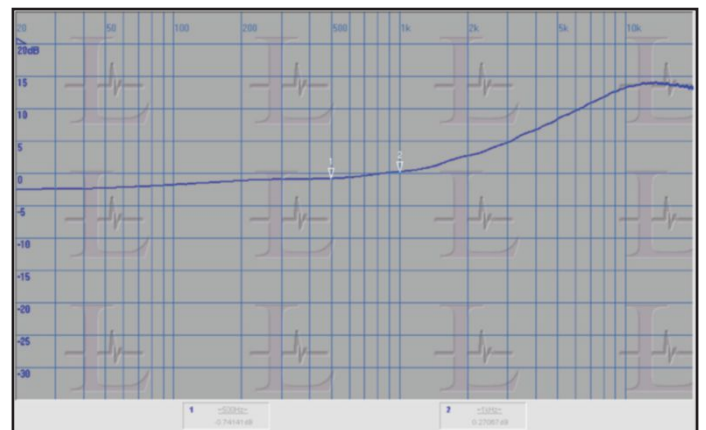


Figure 7: Treble turnover is about 2 kHz, and maximum boost is about 14 dB at 12 kHz.



Figure 9: With all controls at the middle position, the frequency response is nearly flat.

Finalizing the Design

Completing the design required more information than was available for the power amplifier and tone-control modules: What were their input resistances? How much gain did they have? What were the tone-control characteristics?

The input resistance of the power amplifier turned out to be 10 k Ω for each channel. I decided to parallel the two channels' inputs and feed one of two speakers from each of the channels, thus equalizing thermal stress in the power-amp IC (versus paralleling the speakers and running them off one channel while the other channel did nothing). The resulting 5 k Ω load for the tone module was no problem.

The power amp has a specified gain of 26 dB, or a numerical gain of 19.95. The tone module has a gain of about 1.58 with all tone controls set at their midpoint. Delivering 50 W into 8 Ω requires 20 V_{RMS}. Thus, the maximum signal fed from the cathode-follower to the tone module needs to be $20/19.95/1.58 = 0.634$ V_{RMS}. To avoid overdriving the

solid-state circuitry, we want the triode stages to limit the maximum voltage to this level. Since a typical electric bass produces from 0.1 to 0.3 V_{RMS} signal output, the preamp/limiter needs to have a gain of $0.634/0.1 = 6.34$.

Figures 3-8 show the frequency response of the tone module with the controls at various positions. The desired frequency response we specified in the concept design article called for a bass turnover of 250 Hz and a range of ± 12 dB. The treble control was to have a 500 Hz turnover and a range of -5 to +15 dB at 4 kHz. Achieving this target response will necessitate changing some capacitor values on the tone module, as we will discuss shortly.

Achieving the Target Response

Figure 9 shows the response with all controls in the middle position. Note that the response is pretty flat in this case, which is normal for a home-stereo amplifier, but not necessarily for a guitar or bass amp. Usually not having a flat-response setting for a guitar or bass amp is not intentional, but results from the use of passive tone controls. Our tone module uses active controls.

To change the tone control effects to match our targets, we have to raise the bass turnover from 200 Hz to 250 Hz, a factor of 1.25. The turnover frequency of these controls is inversely proportional to the product of the resistance and the capacitance in the circuit section (bass, mid, or treble) in question. Thus, we can raise the bass turnover frequency by decreasing the 0.1 μF capacitor paralleling the bass potentiometer to 0.08 μF . Actually, this is not a standard value, but the next closest standard value—0.082 μF —is close enough. Changing the treble turnover from 2 kHz to 500 Hz requires increasing the 2.2 nF capacitor by a factor of 4, to 8.8 nF.

Since the tone module has two channels, I decided to use one of them to feed an independent line-level output. Thus we should change the capacitors on both channels so the tone of the bass from the line output will match the tone as heard from the speakers. **Photo 1** is the tone module (a Yuan Jing NE5532 tone control

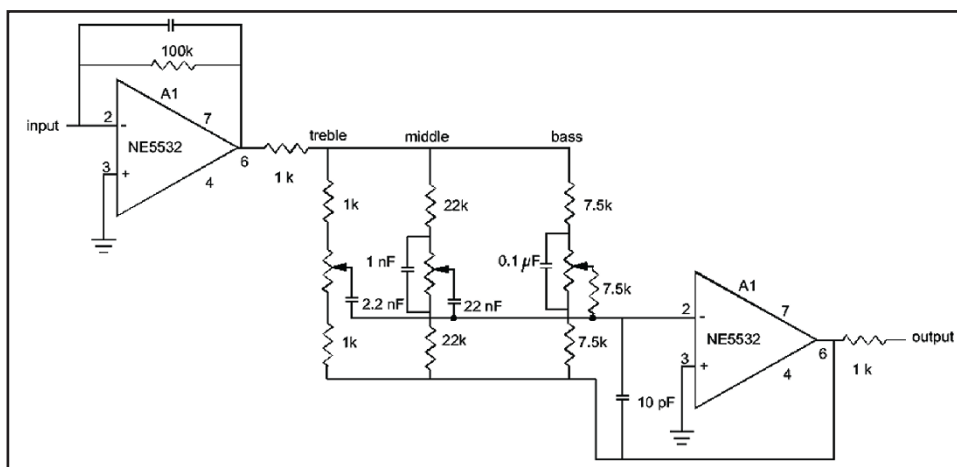


Figure 10: This schematic shows one channel of the tone controls in the module.

module). The bass-control capacitors are C21 and C22, and the treble-control ones are C19 and C20.

The circuit design for the prototype Bolger preamp/limiter is shown in **Figure 11**. The 78M12 voltage regulator should be mounted to the chassis using heatsink grease. This will enable it to operate safely without a separate heatsink. 

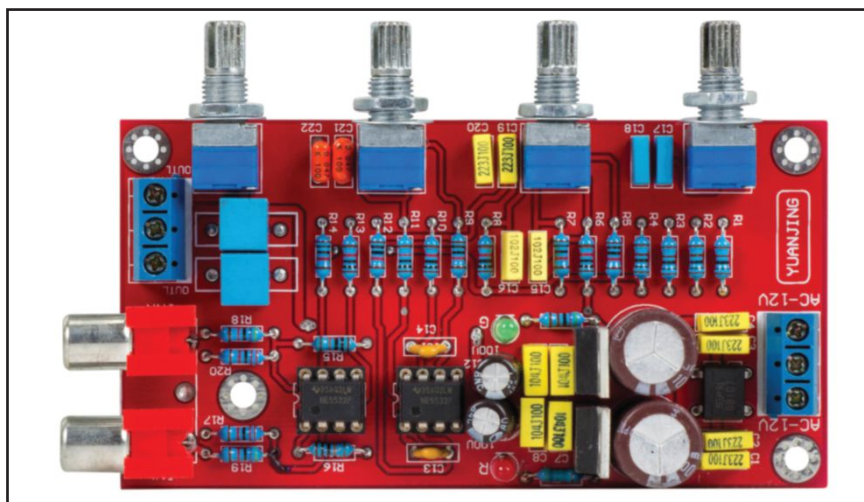


Photo 1: The tone module looks like this.

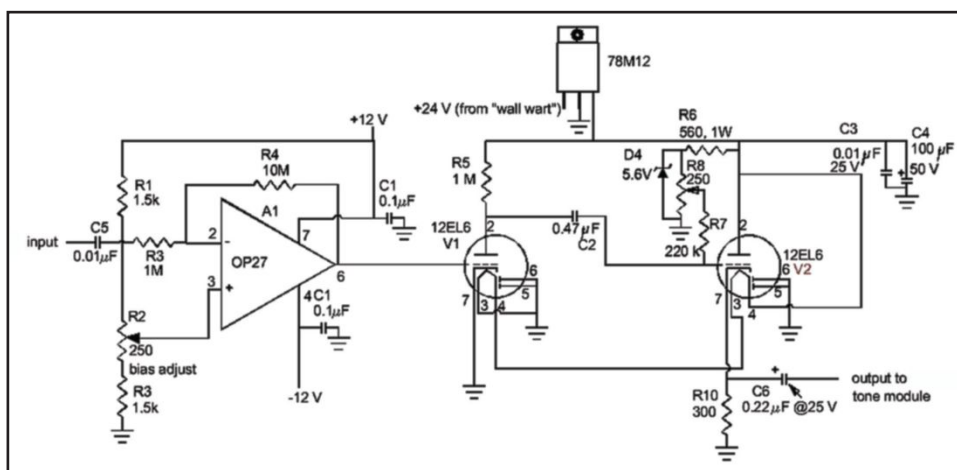


Figure 11: The complete preamp/limiter circuit includes a few changes from the one shown in Figure 2.



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.

January 6–7, 2018

ALMA International Symposium & Expo
South Point Hotel & Casino, Las Vegas, NV
www.almainternational.org

For 2018, the revitalized The Association of Loudspeaker Manufacturing and Acoustics (ALMA) International Symposium & Expo (AISE) will expand on all of the things exhibitors and attendees have always appreciated about this event—taking it to another level with new innovative programs and activities.

The theme for 2018 is “The Revolution of the Audio Signal Chain,” reflecting the growing importance of the signal path from source to speaker, focusing on how changes in the industry impact transducer design, the integrated speaker, and overall loudspeaker performance.

AISE 2018 will take place on Saturday and Sunday, allowing for a one-day break between AISE and CES. There will be a President’s Reception on Friday, January 5, from 6 PM to 7:30 PM in the Banquet area. Exhibits will be open at 9 AM both days. ALMA’s Education Track invites students and educators to attend, network, and present content at AISE.



January 9–12, 2018

CES Show 2018
Las Vegas Convention and World Trade Center
(LVCC) and 10 other locations in Las Vegas, NV
www.cesweb.org



Held in Las Vegas, NV, every year, CES is the world’s gathering place for all who thrive on the business of consumer technologies and where next-generation innovations are introduced to the marketplace.

CES, formerly The International Consumer Electronics Show (International CES), showcases more than 3,900 exhibiting companies, including manufacturers, developers and suppliers of consumer technology hardware, content, technology delivery systems and more. The conference program includes more than 300 conference sessions. Because it is owned and produced by the Consumer Technology Association (CTA)—the technology trade association representing the \$292 billion U.S. consumer technology industry—it attracts the world’s business leaders and pioneering thinkers to a forum where the industry’s most relevant issues are addressed.

January 25–28, 2018

The National Association of Music Merchants (NAMM) Show

Anaheim Convention Center, Anaheim, CA
www.namm.org/exhibit/WN18 | www.namm.org/the-nammshow/2018/map



The NAMM Show—the global crossroads of the music products, pro audio and sound, and live event technology industries—returns to the newly-expanded Anaheim Convention Center in 2018 and will offer visitors a dynamic way to cover all aspects of music: from products, to pro sound and live production. Preview new products from 7,000+ brands, enjoy special events set to live music, celebrity appearances, dozens of educational sessions tied to today’s trends, and interviews with the world’s top innovators, artists and professionals in the music world.

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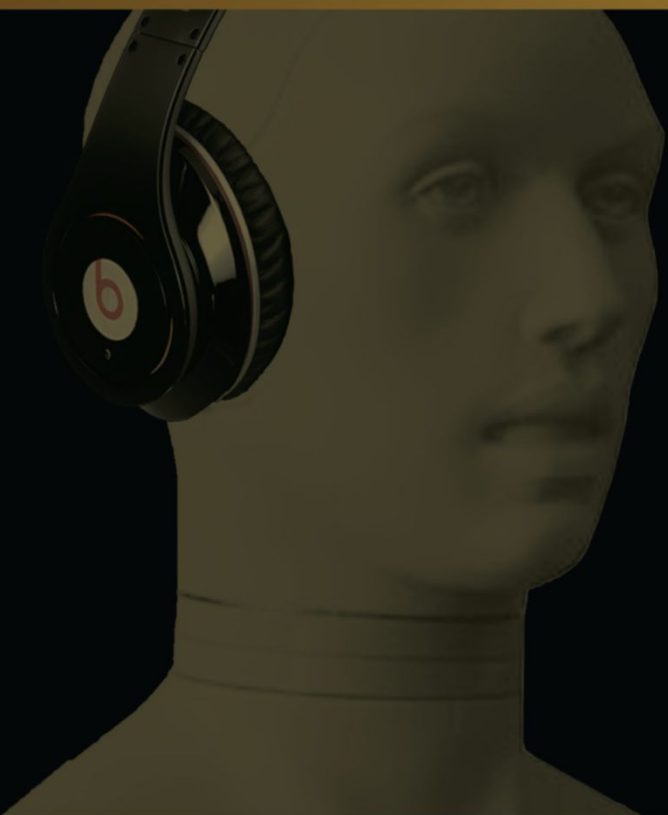
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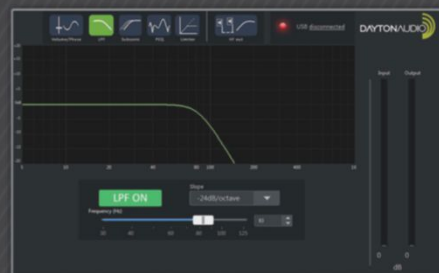
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