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Reaching You Every Week

In this age of cyber attacks and hacking, everyone has good reasons to be concerned about privacy. But that's also when our relationships with the sources we all depend on for knowledge and information should be strengthened. Magazines, like all other media sources, have to continuously adapt to a changing landscape and increasingly depend upon strong connections with their audiences using all tools that are available to them.

When *audioXpress* was created, magazines still depended upon printed distribution, reaching readers who subscribed and got their copies in their mailboxes, and those who bought the latest issue at newstands and selected bookstores. Unfortunately, for most parts of the world, physical distribution to newsstands became prohibitive for small specialized titles such as *audioXpress* (remaining magazine distributors only carry mainstream titles, requiring hundreds of thousands of copies for nationwide coverage). Our readers have also slowly adopted digital subscriptions, conveniently available from the moment a new issue is uploaded, in a format that can be read, searched, and accessed anytime from a computer, tablet or even a smartphone.

Several times I've been personally approached by subscribers, saying they had stopped "receiving" the magazine in digital form, even though they had recently renewed their subscription. When asked, they confirm they can still login to the service with existing usernames and passwords. But somehow, readers expect a magazine to "reach out" to them when a new issue is available.

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We maintain a website to access those services, that also features daily industry updates (now receiving more than 400,000 monthly visits), and we send out a great (FREE) weekly newsletter—*The Audio Voice*—to anyone who signs up to receive it. That newsletter is also sent to subscribers of *audioXpress* and *Voice Coil* magazines. *The Audio Voice* newsletter provides interesting industry updates, occasional show reports, weekly news highlights, and direct access to complete articles from *audioXpress* and *Voice Coil*. On the *audioXpress* website, our readers have access to dozens of classic DIY projects, great theory articles, complete reviews, guest editorials, interviews and even online versions (with zoomable graphics) of our *Voice Coil* Test Bench articles.

The Audio Voice has turned into the globally recognized "voice" for the audio industry providing an unique inside perspective and has served as a great promotional tool for our magazines, which is the reason why our subscription base is expanding. We are happy to report that subscriptions for our printed edition (mainly in the US) are increasing. *audioXpress* now reaches 45,000 readers, globally, most of which simply prefer to browse the latest issues online, and download the PDF for future reference or offline reading.

As in the magazine, to reflect a diversity of interests, editorial topics discussed in the newsletter range from the latest listening experiences at high-end shows, to the latest transducers and revolutionary concepts used in headphones, the most recent market trends in smart speakers, and exclusive updates on AoIP. Whatever the weekly topic—even if it is not exactly your interest that moment—remember there will be something new and exciting next week. The fact is we are publishing more interesting things, every day and every week, than it would be possible to fit in 64 monthly pages.

If you have not received *The Audio Voice*—you can look at past issues:
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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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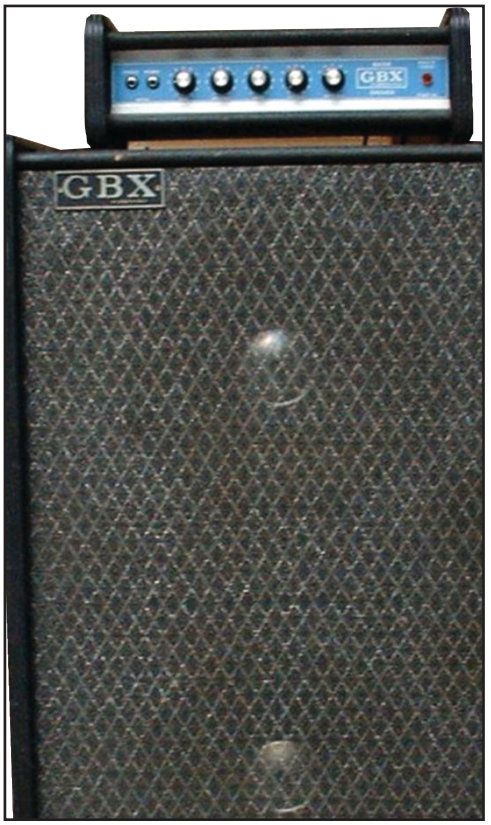
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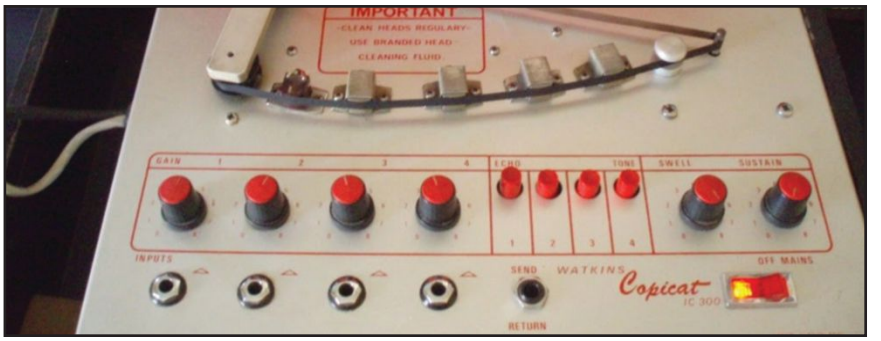
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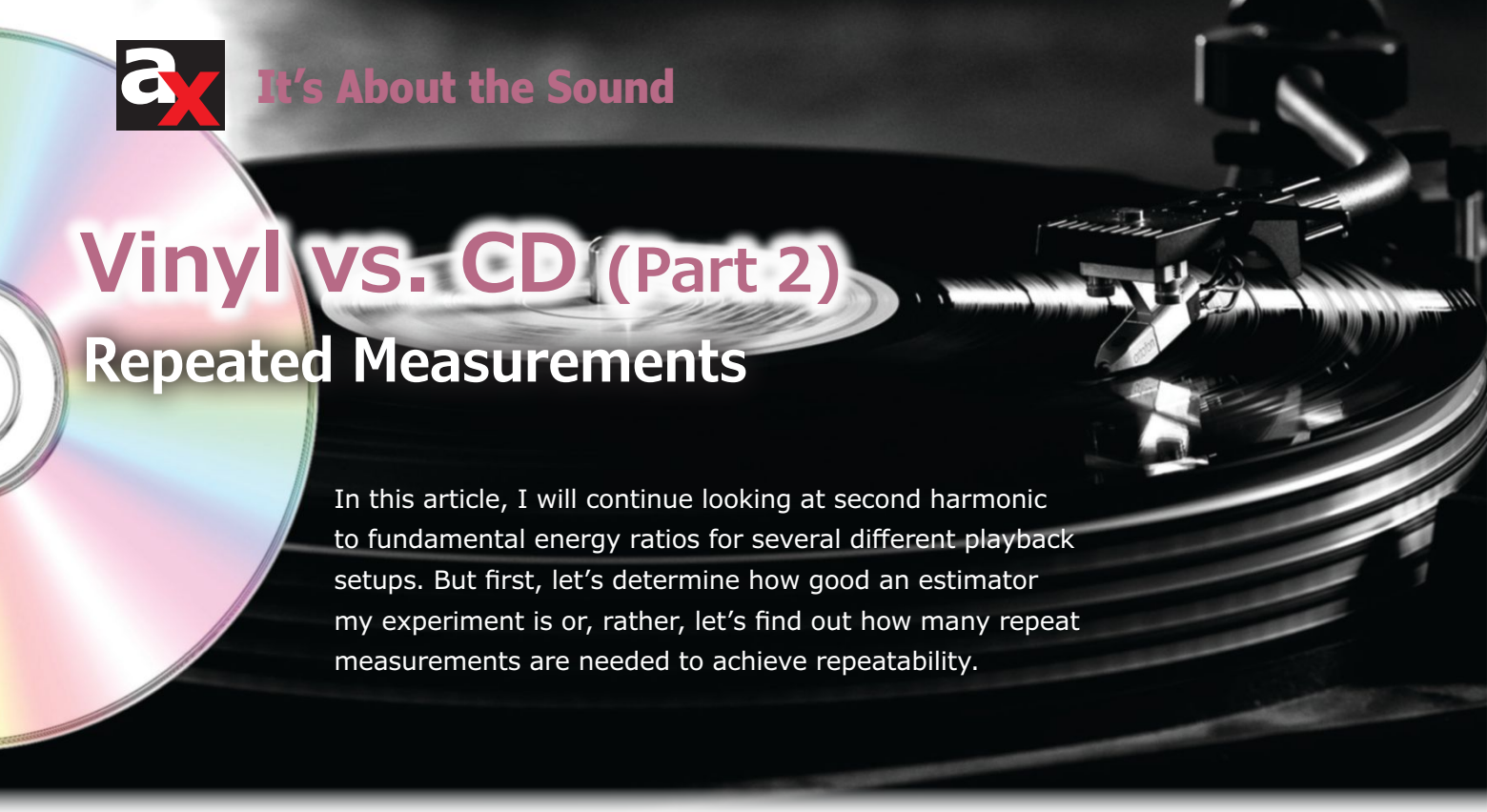
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Vinyl vs. CD (Part 2)

Repeated Measurements



In this article, I will continue looking at second harmonic to fundamental energy ratios for several different playback setups. But first, let's determine how good an estimator my experiment is or, rather, let's find out how many repeat measurements are needed to achieve repeatability.

By
Ron Tipton
(United States)

The raw data shown in **Figure 1** is from two sequential measurements from one of the fundamental band-pass filters. It doesn't matter which one, because they all look about the same—visually, not very repeatable. It may be surprising to learn that their RMS values, the square root of the sum of their squares, differ by just 2.6%. I discovered this while writing the first part of this article series "Vinyl vs. CD (Part 1): Measuring the Sound Difference" (*audioXpress*, October 2017), concluding the energy "under the curve" remained rather constant. But after some thought, I decided it needed further investigation.

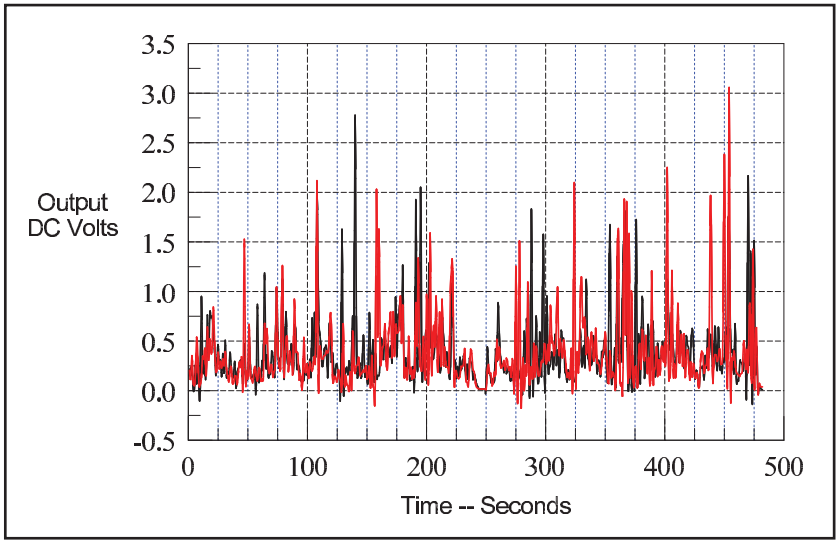


Figure 1: Raw data is shown from sequential measurements from one of the fundamental band-pass filters, one in black and the other in red. They do not appear similar but their RMS values only differ by just 2.6%.

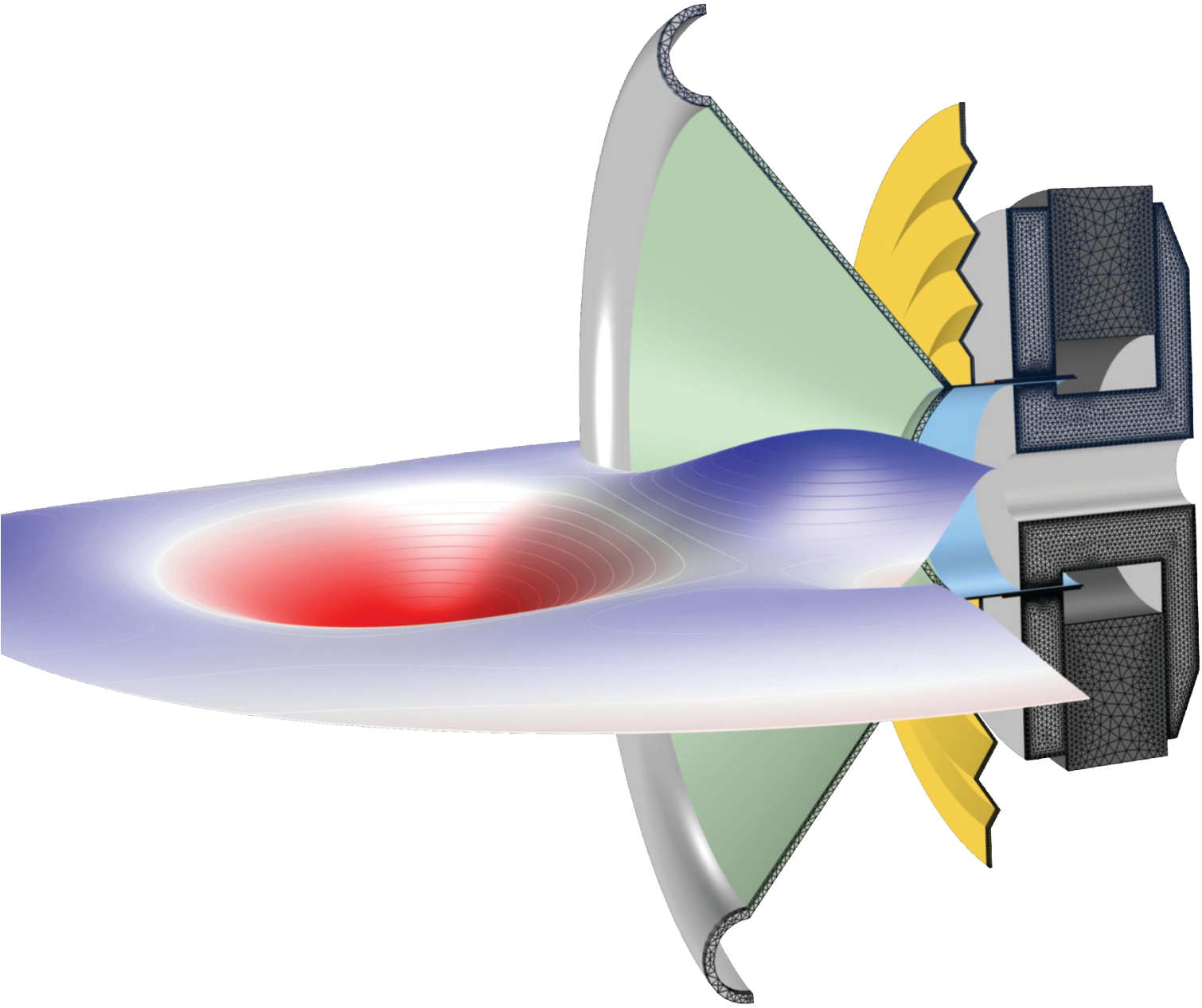
To continue, I made 10 sequential measurements from both the fundamental and second harmonic band-pass filters for several of the available pairs. Calculating the second harmonic to fundamental RMS ratios and then averaging them in random groups of three or four, I found a spread of 2% or less.

So, I tried this for six sets and then three sets. The six measurements sets were fine but some of the "three sets" were not. It appears a minimum of six measurements are needed for reasonable repeatability—which is rather time consuming! Especially when you notice that I added another filter board with three band-pass filter pairs.

A New Measurement Setup

Figure 2 and **Figure 3** show the block diagrams for my experimental setup's measurements: two different USB DACs followed by a vacuum tube or solid-state line amplifier, each with 7 dB gain. I inserted the 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 the free VLC Media Player with a playlist of the first two tracks from Diana Krall's *Quiet Nights*.

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



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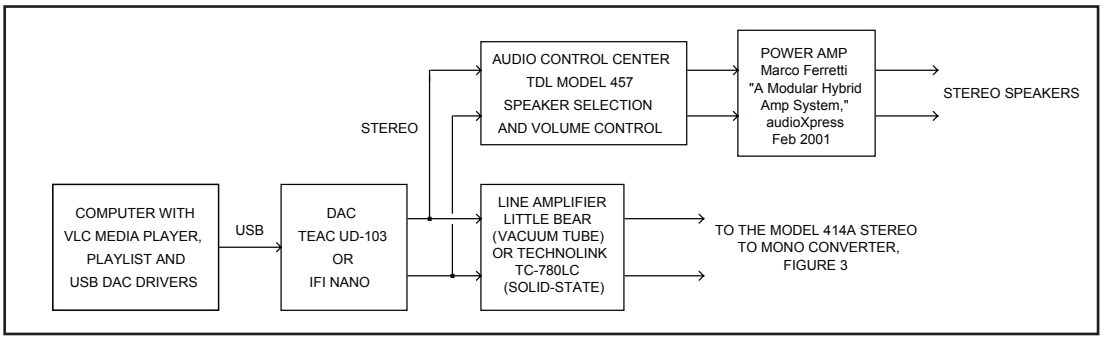


Figure 2: The block diagram shows the signal flow from the source computer running VLC Media Player. The USB DAC is either the TEAC UD-301 or the iFi nano iDSD. The line amplifier is either the Little Bear vacuum tube or the Technolink solid-state.

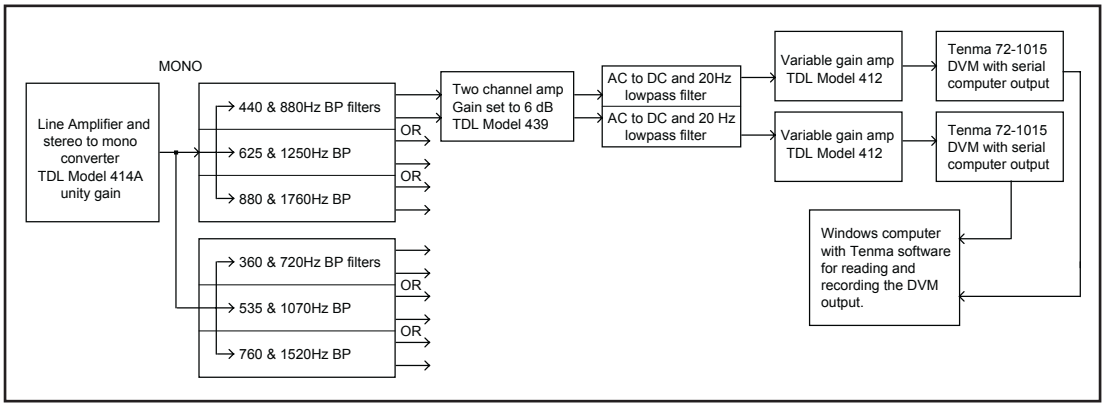


Figure 3: This block diagram shows the stereo to mono converter and the six sets of band-pass filters followed by the AC to DC converters and the recording Tenma voltmeters.

filters with a 20 Hz cutoff frequency. I connected a pair of identical true RMS voltmeters (Tenma 72-1015), set to measure DC voltage, to each detector output. Then, I played the playlist six times for each filter and recorded and saved the Tenma outputs as Microsoft Excel .xls files.

I still think energy-under-the-curve is the best estimator for this measurement. That is, find the sum of the squares, divide by the number of samples, and then take the square root. The Microsoft Excel spreadsheet makes this easy because of its built-in =SUMSQ function: It squares each data value and sums them.

The numbers shown in **Table 1** were calculated by finding the RMS value for the fundamental and second harmonic filter outputs and then dividing the second harmonic RMS by the fundamental RMS. Thus, the Table 1 unit of RMS Energy Ratio, with higher values indicating more second harmonic energy.

Band-Pass Filters—Revisited

I designed the first filter board (440 Hz, 625 Hz, and 800 Hz fundamental frequencies) using the free Microchip FilterLab Version 2.0. FilterLab has a small set of specifications: Filter type—Butterworth bandpass, six-pole and Multiple FeedBack (MFB, the only kind it designs). Then, I entered the center frequency and the -3 dB bandwidth. I kept the percentage bandwidth very close to 1.25 for all the filters, which worked well. I simulated the frequency response curves in TopSpice and saw no problems. I was a bit concerned that neither the input resistor

	Fundamental Filter Frequencies						
	Music Source	360 Hz	440 Hz	535 Hz	625 Hz	760 Hz	880 Hz
1	LP with TDL 4061B vacuum tube RIAA preamplifier	0.856	0.794	0.716	0.554	0.622	0.690
2	VLC Media Player, TEAC DAC, Little Bear (VT) line amp at +7 dB gain	0.844	0.758	0.655	0.457	0.592	0.595
3	VLC, iFi nano DAC, Little Bear (VT) line amp at +7 dB gain	0.802	0.782	0.680	0.462	0.564	0.573
4	VLC, TEAC DAC, Technolink (SS) line amp at +7 dB gain	0.769	0.721	0.674	0.427	0.486	0.469
5	VLC, iFi nano DAC, Technolink (SS) line amp at +7 dB gain	0.732	0.718	0.647	0.413	0.467	0.555

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 a vacuum tube preamplifier. All the red numbers, the highest in each column, are in this row. The next highest green numbers are in Rows 2 and 3, the two DACs, followed by the Little Bear vacuum tube line amp.



Photo 1: The front iFi nano DAC panel shows the low-pass filter response switch, the USB input connector, and the output digital coaxial collector. The small hole in the top cover on a straight line between the large "i" and the right front corner is for the LED, which indicates the input data rate.

values nor any of the capacitor values were scalable, which resulted in some large resistors, over 500 kΩ in a few cases.

However, I built the filters, ran some measurements and was generally pleased with the results. But I kept thinking about scalability so I looked around for another filter design program.

I found AktivFilter Version 3.2.9 for a reasonable price so I licensed a copy. It has the same set of design specification with the addition of input resistance scaling. After simulating the frequency responses, I built the filters, ran some measurements, and decided that either filter design tool will work.

The Detector (AC to DC Converter)

The FullWave detector circuit that I used was published in a National Semiconductor Application Note. I designed the 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 can be found in the first part of this article and is included in this month's Supplementary Material in the fullwave.zip file.

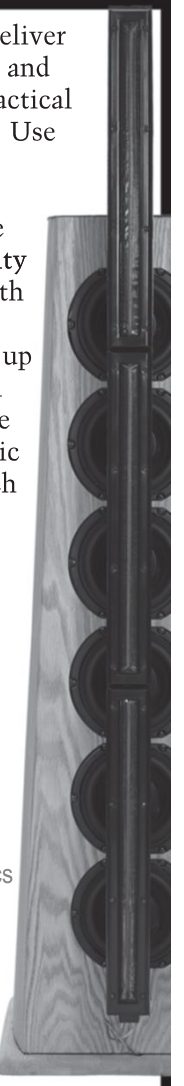


Photo 2: The TEAC DAC has a solid, professional feel. On its front panel are the mains on-off switch, the digital input selector switch, the input data rate LEDs, the headphone jack, and the volume control. The mains power connector, digital input connectors, and stereo line out connectors are on the hidden rear panel.

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Photo 3: The TC-780LC front panel with power on indicator and volume control. If used, the 3.5 mm stereo connector disables the rear RCA connectors, but I didn't try this.

LP Base-Line

The first row in Table 1 is the LP base-line. You will notice that all these values are in red meaning they are largest in their column. It seems that if you want "true" LP sound—play LPs.

However, Rows 2 and 3 are for two different USB DACs driving the Little Bear vacuum tube line amplifier set to its maximum of 7 dB gain. All the green numbers are in these two rows, meaning second highest value in the column. And, fair is fair, each DAC got three. It appears you may get a more mellow LP sound by using a vacuum tube amp at your DAC's output.

iFi nano iDSD DAC

The iFi nano iDSD low-priced DAC, about $200, is very capable (see **Photo 1**). It automatically senses the input type: PCM, DXD, DSD, or DFF and shows this by the color of the LED—the small hole in the top cover. The low-pass filter selector switch, USB input connector, and digital output connector are shown. The stereo RCA analog output connectors, headphone jack, and the volume control (with on-off switch) are on the hidden end.

Pulse code modulation (PCM) rates from 44.1 to 384 kHz, 16 to 32 bits are supported. The standard CD rate was an easy job. Drivers for Windows and Mac can be freely downloaded. Power can be supplied by either the internal battery or the USB connection—and that's where I got into trouble. If the DAC is first connected to the active USB port and then turned on, it uses USB power. But if it's turned on and then connected to the port, it uses the internal battery. That's why my battery ran down in the middle of a measurement session and I had to repeat a bunch of measurements.

TEAC UD-301 DAC

This medium-priced DAC, about $400, is shown in **Photo 2**. The front panel controls include its mains on-off push button switch, which lights up blue around the button when on; the digital input selector switch—USB, coaxial or optical; and the volume control. A LED illuminates to show the digital input rate from 44.1 to 192 kHz and 2.8 or 5.6 MHz for DSD. This unit has a very solid "feel" and I enjoy using it. The rear panel includes the IEC mains connector, the three digital input connectors, and the stereo RCA and balanced XLR line out connectors.

Technolink TC-780LC

This inexpensive, about $50, solid-state line amplifier is shown in **Photo 3**. It is designed around a single NE5532P dual op-amp. But because it uses a 12 VDC wall power supply, the signals must be capacitor coupled. Its spec sheet says ±0.5 dB, 20 Hz to 20 kHz but my measurements show -7 dB at 20 Hz and -3 dB at 40 Hz when measured from the rear panel RCA connectors. Maybe the coupling capacitors need to be larger. That was suitable for my use here but if you need the low-frequency response, perhaps you should look elsewhere. There are more details about this amplifier in the Supplementary Material found on the *audioXpress* website.

Final Thoughts

I understand my measurements presented here are not conclusive. However, they do indicate that an external DAC followed by a vacuum tube amplifier might make your listening mellower. Stay tuned till next time to find out how a VST plug-in in your media player software can also warm up your listening.

Project Files

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

Resource

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

Sources

Little Bear P5-1 Blue line amplifier (I received Version 1.3)
Amazon | www.amazon.com

iFi nano iDSD
iFi Audio | ifi-audio.com/portfolio-view/nano-idsd

Tenma 72-1015 DVM
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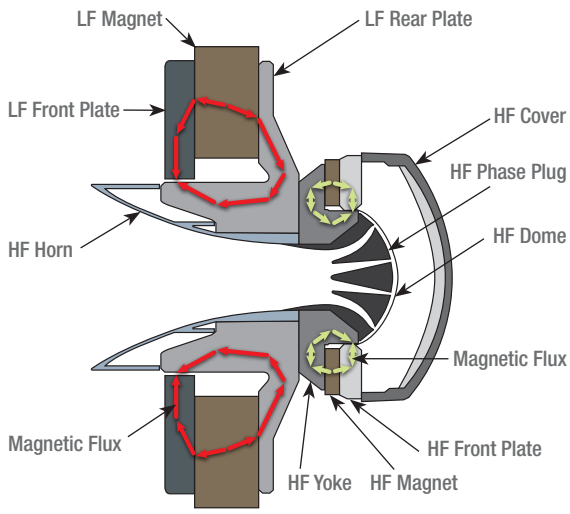
How a common magnet motor delivers a better sounding, lightweight, full-range speaker

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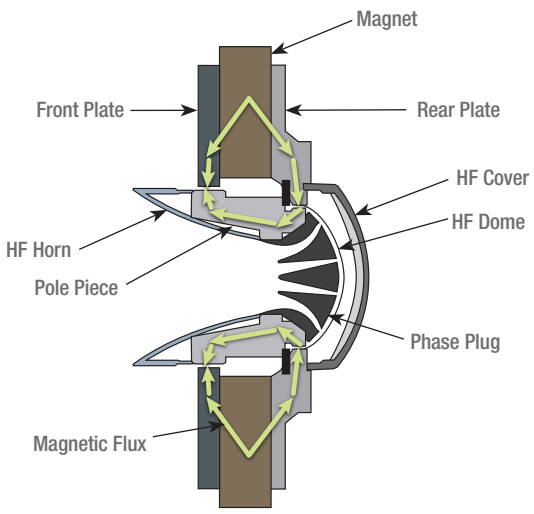
The magnetic flux is optimally distributed between the two elements (calculated using FEA), to achieve the best balance between HF and LF response (see diagrams below). Additionally, the use of a single magnet assembly also means lighter weight and a more compact profile, compared to more conventional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

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CELESTION

Midwest Audiofest 2017

Outstanding DIY Speaker Designs



If anyone thinks that DIY speakers cannot compete with the best commercial designs, I think they are very wrong. Just attend the Speaker Design Competition held annually at the Midwest Audiofest and you will discover why. The best entrants at this year's Midwest Audiofest can hold their own against some of the higher-priced speakers I have heard and actually best many of them.

Photo 1: The judges for this year's Speaker Design Competition and Midwest Audiofest 2017, Matt Phillips (Parts Express), Tom Perazella (*audioXpress* author), Jerry McNutt (Eminence).

By
Thomas Perazella
(United States)

The Midwest Audiofest is held every year in Springboro, OH, under the auspices of Parts Express and other vendors. There are several events that take place during the two-day event, including the Warehouse Tent Sale, the Speaker Design Competition, the Mobile Electronics Competition Association (MECA) Auto Sound Challenge, and the Audio Swap Meet. The list of vendors included 3M, ADJ, AKG, Audiovox, audioXpress, Behringer, B.I.C America, CAIG Laboratories, Celestion, Crown, Dayton Audio, dbx, Eminence, Gator Cases, Harmon Professional, JBL, Kenwood, MTX, NTE Electronics, Onkyo, Pioneer, QSC, RCF, Selenium, Shure, Switchcraft, and Velleman.

However, for the purposes of this article, I will focus on the Speaker Design Competition, for which I was one of the three judges. The response this year was outstanding with 48 initial entries from across the country and Canada and 40 actually being judged. Of the original 48, a few did not show and others withdrew either because of problems or category disqualifiers. There are four categories that span a wide range of design criteria, capabilities, and costs. Details on these categories and other elements of the competition can be found on the Midwest Audiofest website at: www.midwestaudiofest.com/speakerdesigncompetition.php

The Categories

The entry level category, "Under $200," is for speakers that must have a total driver cost of no more than $200 for the stereo pair. This is a very difficult restriction. In spite of that I am constantly amazed at the level of performance from some of the entries in this category.

The second category is "Over $200" for speakers that cost more than $200. They must be passive with no elements such as active crossovers or DSPs allowed.

The third category is "Dayton Audio," where the only restrictions are that any drivers used must be of the Dayton Audio brand and they must be passive.

The fourth category is "Open Unlimited," where anything goes. The designers are limited only by their imagination and wallets.

Judges

There were three judges that all have significant experience in the speaker field (see **Photo 1**). Details of their qualifications can be found on the Midwest Audiofest website but here is a quick rundown. Jerry McNutt, affectionately known as McJerry is an engineer with the speaker company Eminence. Tom Perazella is a longtime speaker builder and frequent contributor to *audioXpress* magazine. Matt Phillips is an experienced speaker builder and technician at Parts Express. All the photos used for this article are courtesy of Parts Express.

The Process

Throughout the year, competitors work feverishly on designs that they submit to the contest. Entries are brought to the judging location, which is a large room not far from the Parts Express building in Springboro, OH. For each category, the judges pick music they feel is suitable to highlight the capabilities of that group. The test music generally gets more difficult in proportion to the expectations for the higher categories. For the "Under $200" category, the music tends to have less dynamic range and lower levels at the frequency extremes. For the "Open Unlimited" category, all bets are off and the music has wide dynamic range, lots of transients, deep bass, lots of highs, and mixes that tend to highlight intermodulation distortion (IMD) problems. A list of the test music by category can be found on the Midwest Audiofest website.

From the music submitted, three one-minute cuts are selected for each category. They are then burned to a CD along with a test tone to allow accurate level setting of sound pressure levels (SPLs) for speakers that have different sensitivities. Each speaker being evaluated in the category plays the three selections, providing the judges with a repeatable set of sounds with which to form their opinions. The playback equipment is the same for all categories except for the "Open Unlimited," where the contestants may provide their own electronic crossovers, equalization, amplification, and other equipment.

Each speaker was judged on the basis of six criteria—Clarity, Craftsmanship, Dynamic Range, Originality/Design, Soundstage/Imaging, and Tonal Balance. Points were assigned by each judge on individual speaker score sheets in each category as follows: 1-2 = Needs work. 3-4 = Below Average. 5-6 = Average. 7-8 = Above Average. 9-10 = Excellent. Judges did not communicate with each other or anyone else during the judging. Each score sheet was placed face down on the judging table after each speaker was auditioned and was picked up by a contest worker for tabulation.

Before each speaker was judged, the contestant gave a short description of the project, including the design philosophy, to the judges and other persons in attendance. Some provided datasheets with both written and photographic information. I am always amazed at the amount of work that goes into these projects including computer aids to design, extensive measurements, lots of listening, and then more adjustments to get to the final results.

Because of the number of entries and in deference to the ears and sanity of the judges, judging spanned two days. Before the start of the contest on Friday evening, there was a meet-and-greet event where the contestants could meet each other and discuss audio in general and specifically their creations. They could also finish setups and enjoy some pizza provided by the event sponsors. The first category was "Open Unlimited," which began after the meet and greet. Starting at 9 AM on Saturday, the other three categories were judged. This year, of the speakers judged, there were 14 entries in the "Under $200" category, nine entries in the Dayton Audio drivers category, 10 entries in the "Over $200" category, and 7 entries in the "Open Unlimited" category. For each category, there was a first, second, and third place winner. The audience also got to vote for an audience favorite in each category and for favorite overall. The audience vote



Photo 2: First place in the "Under $200" category—Peanuts



Photo 3: Second place in the "Under $200" category—Honeycombs



Photo 4: Third place in the "Under $200" category—BMR-3L



Photo 5: Competing in the "Under $200" category—Primus Serium

came into play in each category in case of a tie by the judges. Believe it or not, a tie did occur in one of the categories. The speakers were so good that it came down to individual tastes. That speaks very well to the quality of the projects submitted.

Personal Observations

As I mentioned in a recent *audioXpress* article about my VFET SET amplifier "Cherry Bomb," the moniker DIY should perhaps be changed to DIFY meaning Do It For Yourself. Everyone has their own preferences and building a project, if successful will inevitably result in a reflection of those preferences. The following is a take on some of the projects in light of my own preferences. When numerical ratings are given, they reflect only my score sheets. As they always say in TV ads, your results may vary.

"Under $200" Category

Starting with the "Under $200" category, the most surprising thing is always how good a speaker pair that is limited to a total cost of $200 for drivers can sound. The averages of all entries for all criteria was 40.3 points out of a possible 60. This is an indicator of how far driver technology and materials have come as well as the dedication of the builders to optimize the designs. There was no limit to the amount of spending on the cabinets or crossovers. The workmanship on some of these entries would put many high-priced commercial speakers to shame.

First place in this category went to the entry Peanuts by Nick Santorineos (see **Photo 2**). Last year Nick took top honors in the "Open Unlimited" category with his entry named Stink Eyes. Peanuts being in the "Under $200" category might be considered an entry-level speaker but still managed a very respectable sound and was my top pick as well with a score of 53. In addition to the sound, the fit and finish was top notch with beautiful inlaid wood, chrome feet, and a chrome ring around the tweeter. The name Peanuts is obvious from the shape, but it may not be quite as obvious that the drivers are not centered on the front panel. That displacement from center results in unequal path lengths to the edges, minimizing the contribution of diffraction anomalies to the sound.

Second place went to the entry Honeycombs by Ben Cooper (see **Photo 3**). This speaker is a perfect example of the need to listen before making judgments. When I first saw it I said to myself, "Wow that is strange." The housing shape is hexagonal with beveled edges, which is the source of the name honeycomb. There are drivers on the front of the hexagonal and also some of the sides. The drivers on

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Photo 6: First place in the "Dayton Audio" category—ES-3s



Photo 7: Second place in the "Dayton Audio" category—Udique XL



Photo 8: Third place in the "Dayton Audio" category—The Black Widows

the sides were rear mounted to the structure, which I thought might create a problem with diffraction, but none was noted. Overall, it scored well in all of the criteria and was a balanced design.

Third place was taken by the entry BMR-3L by John Hollander (see **Photo 4**). Originally designed for a competition where enclosure size was severely limited, a clever way to maximize interior box space was created. To allow for proper port tuning without occupying too much internal volume with the port, part of the port was simply "ported" or moved to the outside of the enclosure. In addition, the crossover components were likewise moved to the outside of the enclosure. Overall, a good performer but in my scoring somewhat average in dynamic range.

Another speaker in this category with outstanding craftsmanship was the Primus Serium from Ben Cooper (see **Photo 5**).

"Dayton Audio" category

As the name says, all speakers in this category must use only drivers from Dayton Audio. If you are not familiar with the driver offerings from Dayton Audio, you might think that this requirement would severely limit design options. However, a quick check of the Parts Express website shows a count of more than 180 drivers of that brand including dynamic drivers, planar magnetics, ribbons, and passive radiators in all categories from monster subs to delicate tweeters.

First place in this category went to the ES-3s from Thomas Zarbo (see **Photo 6**). Again, the first impression is of a beautiful design with a tilted back faceted enclosure with a Waterfall Bubinga wood finish. The sound matched the looks being very well balanced and resulted in even scores across all criteria.

Second place was taken by the Udique XL from John Hollander (see **Photo 7**). Although a somewhat conventional design being an MTM configuration in a rectangular box, the craftsmanship was excellent and as with the ES-3s the results across all the criteria were consistently good.

Rounding out the top three in third was The Black Widows (see **Photo 8**). This is a classical MTM configuration in a very unconventional shape. The X-shape housing with a gloss black finish on all surfaces except the front are contrasted sharply with a gloss red front panel. Therefore, the inspiration for the name, with the speaker resembling the red hour glass mark on the abdomen of the female black widow spider. Never fear, the auditory performance of this speaker will not kill you as it was very competent. However the music through these speakers may trap you in a web of pleasurable listening.



Photo 9: First place in the "Over $200" category—Brioso



Photo 10: Second place in the "Over $200" category—Lyndzies

"Over $200" Category

Not surprisingly, this category consistently does well in my ratings. Having no limit on driver selection and working within proven design concepts leads to some outstanding results. In fact, the speaker that earned my personal highest rating was in this group. As a category, these speakers received more than twice the top ratings in the classifications than any other.

The first place slot was taken by the Brioso, submitted by longtime builder, Paul Kettinger (see **Photo 9**). In Italian and related to music the word means in a lively, spirited, vivacious manner. This speaker certainly lived up to its name providing consistently high marks with top ratings in clarity, soundstage/imaging, and tonal balance. It uses a transmission line to extend the bass. Normally, I am not a fan of transmission lines or other passive bass assists, but the implementation in this example was quite satisfactory.

In second place was the Lyndzies by Steve Fishe (see **Photo 10**). His speaker received top marks for clarity, dynamic range, soundstage/imaging, and tonal balance. It is a three-way design with a substantial woofer and domed midrange and tweeter. The housing was sloped in several directions to minimize diffraction. The woofer was mounted partly down to the floor similar to the Roy Allison designs. When mounted low enough, this placement minimizes the "Allison Effect," which is alternating bands of reinforcement and cancellation of the mid bass frequencies from interaction of the direct sound from the driver with the reflected sound from the floor surface. I believe the top scores in the dynamic range and tonal balance were at least partly due to this large woofer and its placement with respect to the floor.

Third place was taken by Kamayari, built by Eric Woodring (see **Photo 11**). Kamayari is a sickle-shaped spear with a horizontal blade at the base of the vertical blade for hooking things. The curved shapes of the wood at the sides of the cabinet were also carried through in the curved legs. This three way did very well in most criteria, while only slightly behind some of the others in the area of soundstage/imaging. Not only was the quality of the workmanship outstanding but the design certainly did hook me, befitting its name. The quality of the wood and craftsmanship was outstanding and the yellow fronts of the drivers added a striking accent to the wood finish.

In addition to the winners in this class, there were two entries with a combination of design and craftsmanship that were impressive enough to deserve special mention. They are the Prunus-



Photo 11: Third place in the "Over $200" category—Kamayari



Photo 12: Competing in the "Over $200" category—Prunus-Junglans-Concerto



Photo 13: Competing in the "Over $200" category—Strafi

Juglans-Concerto by Julian Franke (see **Photo 12**) and the Strafi by Javad Shadzi (see **Photo 13**).

"Open Unlimited" Category

In theory, this category should represent the summit of what can be achieved in speakers. In actuality, although certainly capable of achieving that status, often the entries do not reach those potential heights. To put my ratings in perspective, my reference system would fall in this category. Using two Bohlender Graebener RD75 planar magnetic drivers in custom asymmetrical dipole baffles covering the range of 300 Hz up, 12 10" Peerless CC line drivers in vertical line arrays covering the 100-to-300-Hz range also in dipole configuration, and four long excursion 15" Dayton DVC woofers each in a separate 5 ft^3 sealed enclosure, there is a huge amount of linear volume displacement available. Three amps with a total of around 4 kW provides effortless drive regardless of level. All are controlled by an extremely versatile and competent DEQX Express II providing crossover, EQ, and time alignment functions. If I were to rate this system in terms of the criteria used for this competition I would give it 10s in all categories except for soundstage/imaging. Large line arrays have numerous advantages, but their large size overly magnifies the apparent size of the images, resulting in sub-par localization compared to the best point source speakers.

Why mention this? Well, optimizing that reference system has taken me a long time. When I first started there were many problems in such a complicated system that had to be sorted out. That learning curve is also evident in some of the entries in this category. In fact, one of the most interesting and unique designs in this group earned my lowest rating of all categories. On the other hand, in this category there was at least one speaker that earned a 10 in each of the criteria. So as with any endeavor, when pushing the limits, there are bound to be setbacks but the potential rewards are very satisfying for those who persevere.

On to the winners. In first place was The Gandalfs by Kevin Kendrick. In J. R. R. Tolkien's novels, Gandalf is a wizard described as a wise and great spirit. This speaker is built in a curved array of drivers in the fashion of Don Keele's revolutionary CBT36. Utilizing 10 bass/midrange drivers and 40 tweeters in a complicated enclosure it also has a wonderful fit and finish (see **Photo 14**). It received top marks in all criteria except for dynamic range and tonal balance. Most of the CBT type speakers I have heard greatly benefit from the use of a subwoofer to extend the bottom response while unloading the mids to achieve greater dynamic range with less distortion. EQ also helps smooth the response and Keele's latest iteration, the Epique from Parts Express provides several DSP curves that can be used. Overall, this is a great example of how complicated designs can be done correctly and achieve outstanding results (see **Photo 15**).

Second place went to the Esoterics by Scott McMeans. Looking like a common two way, the performance was anything but common (see **Photo 16**). I gave it high marks for clarity, dynamic range, soundstage/imaging, and tonal balance.

The third place winner was the Freaky Frugal Frankensteins by Norman Cerveney. The speaker had three separate enclosures for bass, mid, and tweeter. Two were cubes and one a rectangle, all with beveled edges. Finish was certainly first class and the speaker had excellent clarity and soundstage/imaging (see **Photo 17**).

About the Author

Thomas Perazella is a retired Director of IT. He received a BS from the University of California, Berkeley campus. He is a Past President of the Rockville Chapter of the Izaak Walton League of America, one of the oldest national conservation organizations in the US and currently is the Treasurer. Audio has been his passion for more than 50 years and he is a member of the Audio Engineering Society, the Boston Audio Society, the Philadelphia Area Audio Group, the DC HiFi Group, and the DC Audio DIY Group. He has written for audioXpress magazine and prior to that for its predecessor, Speaker Builder. In addition to audio, his interests include photography, cooking and competition pistol shooting. He has authored several articles in professional audio journals and taught commercial lighting at the Winona School of Photography. Recently, he received a patent on a cost-effective high-efficiency LED lighting system for commercial and residential buildings.

Putting Numbers to Results

To quantify the differences in my rankings by category, I did an average for each of the six criteria by category. I also noted the highest and lowest score by an entry for each criteria. There are a few caveats when evaluating the data. First, the sample sizes are too small to be statistically significant. Second, the difficulty of the source material is not the same for each category. For example, one piece, Aaron Copeland's "Fanfare for the Common Man" used in the unlimited category would probably have significantly lowered the scores if used in the "Under $200" category. That is not to say that the test cuts in that category were easy, it is just that the dynamic range and frequency extension of the Copeland piece would probably cause severe IMD to the entries in that category. Third, the tests are not done blind and being humans, it is difficult to separate



Photo 14: Internal structure of The Gandalfs



Photo 15: First place in the "Open Unlimited" category—The Gandalfs

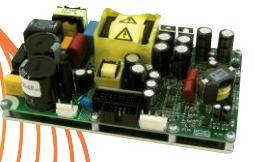
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overview

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

Highlights

- High efficiency
- Universal mains operation
- Flat, fully load-independent frequency response
- Low output impedance
- Very low, frequency independent THD
- Very low noise

Features

- One or two channel amplifier
- 5W standby SMPS
- Advanced over current protection
- External controlled operation
- Auto-switching (115/230V)
- Low weight
- Compact

Applications

- Monitor loudspeakers for recording and mastering studios
- Audiophile power amplifiers for professional and consumer use.
- Public address systems
- Active loudspeakers



NC502MP

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



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Photo 16: Second place in the "Open Unlimited" category—Esoterics



Photo 17: Third place in the "Open Unlimited" category—Freaky Frugal Frankensteins

outstanding workmanship and finishes from the actual sound. The full results by category are shown in **Figure 1**.

I also felt compelled to do another look at one of the categories, Open Unlimited. The combined numbers of the whole category were significantly skewed by the one poor performing entry. I proceeded to eliminate the scores from that one entry and recreated the sheet. The results were more indicative of the entire category (see **Figure 2**).

It is noteworthy that in each category, the highest ranking by criteria was by far 10 with only a few that were 9 or lower. This speaks quite well for the competence of the builders. On the low side the spread was larger with the lowest being 1 and the highest 7. Overall, the averages by criteria ranged from 6.1 to 8.6 out of 10 and the averages by category ranged from 40.1 to 49.4 out of 60, a truly good showing for such a diverse group.

This year although there were some outstanding speakers, most of them had what I consider limited bass response. Yes, I am spoiled with a system that can produce 106 dB at the listening position at 10 Hz with less than 4% distortion. None of the test pieces used in the judging had frequencies in that subterranean range. However several such as Copeland's "Fanfare for the Common Man" had very dynamic low bass that was noticeably weak on most of the speakers. The bass in "Long After Your Gone" by Chris

Category	Clarity	Craftsmanship	Dynamic Range	Originality Design	Soundstage Imaging	Tonal Balance	Total by Speaker
Dayton Audio							
Average	7.4	7.0	5.8	5.9	8.0	6.2	40.3
Lowest Score	6	4	4	4	7	4	
Highest	9	9	8	9	9	7	
Open Unlimited							
Average	7.0	7.1	5.7	7.3	7.7	6.4	41.3
Lowest Score	1	3	1	6	2	1	
Highest Score	10	10	9	10	10	9	
Over $200							
Average	8.6	8.2	8.4	7.4	8.5	8.3	49.4
Lowest Score	5	4	5	4	3	4	
Highest Score	10	10	10	10	10	10	
Under $200							
Average	6.9	7.1	6.1	6.2	7.6	6.2	40.1
Lowest Score	4	4	4	5	5	5	
Highest Score	9	10	8	10	9	8	

Figure 1: 2017 MWAF scores submitted by Tom Perazella by project shows the scores by category and criteria for all entries.

Category	Clarity	Craftsmanship	Dynamic Range	Originality Design	Soundstage Imaging	Tonal Balance	Total by Speaker
Dayton Audio							
Average	7.4	7.0	5.8	5.9	8.0	6.2	40.3
Lowest Score	6	4	4	4	7	4	
Highest	9	9	8	9	9	7	
Open Unlimited							
Average	8.0	7.8	6.5	7.3	8.7	7.3	45.7
Lowest Score	6	6	4	6	6	5	
Highest Score	10	10	9	10	10	9	
Over $200							
Average	8.6	8.2	8.4	7.4	8.5	8.3	49.4
Lowest Score	5	4	5	4	3	4	
Highest Score	10	10	10	10	10	10	
Under $200							
Average	6.9	7.1	6.1	6.2	7.6	6.2	40.1
Lowest Score	4	4	4	5	5	5	
Highest Score	9	10	8	10	9	8	

Figure 2: Here are the scores by category and criteria with one "Open Unlimited" entry removed.

Jones was similarly weaker and not as detailed.

Aside from that point, most of the speakers scored well in terms of imaging, detail, sound staging and clarity. Overall, the event continues to be a great success not only from the standpoint of the entries, but also the creativity, enthusiasm and dedication displayed by the contestants. Long live DIFY.





Reverberation: Friend or Foe?



Aoustical problems? Too much reverb? Echo? Here are a few suggestions.

By
Richard Honeycutt
(United States)

Here's an all-too-common conversation between an acoustical consultant and a prospective client:

Consultant: Hi, how can I help you?

Client: Well, something's wrong with the sound in my _____ (fill in the blank: auditorium, restaurant, conference room, church). Sounds like something's viberatin'..."

Consultant thinks to herself/himself, but does not say: "Yep, if there's sound in that room, something has to be vibrating."

Consultant actually says: "What does it sound like: The same sound over and over, or a roar, or a whistle, or a screech...?"

Client (interrupting): "Sounds kinda like talking in an empty gym."

Determining the Problem's Source

At this point, the consultant knows that she/he is dealing with either echoes or excessive reverberation, or both. The consultant then arranges to go listen to the offending room and maybe make some acoustical measurements. **Figure 1** shows a .wav file of what the consultant would hear after striking a bongo once in an anechoic chamber. (A handclap would have produced a similar waveform, but much shorter in time.) **Figure 2** shows a full 5-second recording of the bongo impulse played in a reverberant room; whereas **Figure 3** is zoomed into the initial half-second. Notice that the initial impulse does not decay to near zero until about 4.4 seconds, and is repeated at regular intervals, so there is both echo and significant reverberation. The regularity of the echo repeats is complex, since the distances from the sound source to the ceiling, side walls, and end walls are all different, causing a variety of delays between the original sound and the echoes.

Echo consists of discrete reflections, while reverberation is made up of many more-or-less random reflections. Echoes are usually undesirable, but reverberation can enhance a listener's enjoyment of music. The reverberation in a venue is quantified by the Reverberation Time (RT), which is measured in seconds. Wallace Clement Sabine defined RT as the time required by a sound in a room to decay in level by 60 dB. Today, acousticians use several variants of RT:

- RT60—RT measured as defined by Sabine
- T30—RT measured over a 30-dB decay, then doubled (Normally, SPL in a reverberant room decays by a constant number of decibels/second, so a 60-dB decay takes twice as long as a 30-dB decay. Actually, the decay time measurement begins after an initial 5-dB drop in level.)
- T20, T15, T10—other ways of measuring RT, analogous to the T30 approach
- EDT—early decay time—another name for T10

RT is directly proportional to the enclosed volume of the room and inversely proportional to the total

absorption. Thus, larger rooms and rooms having acoustically hard walls, floors, and ceilings tend to have high RTs. Since people are pretty good sound absorbers, RT drops when a room is occupied.

Early Public Venues

The effect of reverberation on sound is to elongate it in time. The desirability of this elongation depends upon the nature of the sound. The earliest public venues are thought to have been "threshing floors" (i.e., large flat areas on which wheat or other grains were piled before being threshed to remove the husks). In time, amphitheaters began to be built, based upon this idea. Musicians, orators, or dramatists could stand on a threshing floor and be heard by a large crowd. The RT of a threshing floor was insignificant, since there were essentially no vertical surfaces for the sound to reflect back and forth among.

Next came arenas and coliseums. Since these did have vertical walls, there was some reverberation, and echo from the opposite wall could be a problem. In the Middle Ages, large cathedrals were built for housing worship services. These enclosed large volumes and the boundaries were all excellent acoustical reflectors, leading to long RTs. St. Paul's Cathedral in London has a volume of 152,000 m^3, and an RT of about 13 seconds. Gregorian chants take advantage of these long RTs, which enable the chanters to harmonize with notes they previously sang.

With the Reformation came smaller assembly halls and shorter RTs. Johann Sebastian Bach's Thomaskirche (St. Thomas Church) in Leipzig, Germany, had a volume just under 60,000 m^3 and an RT of 2.5 seconds when fully occupied. Bach's music—and that of other Baroque composers—sounds best in a room with a similar RT.

Concert halls vary widely in RT, as well as in the type of music for which each is best suited. Hiroshi Kowaki, et al. measured the RTs of a number of well-known European concert halls, with results ranging from 1.44 seconds to 3.59 seconds (not including the cathedral they measured).

Music of the Classical period included chamber music designed to be appreciated in intimate rooms having low RTs.

Modern Public Venues

Today, we have performance studios, recording studios, control rooms, surround-sound cinemas, and other specialized listening rooms. These spaces usually have rather short RTs, which differ from one another according to the intended use and the taste of the owners and designers. Often the RTs of these rooms are below one second.

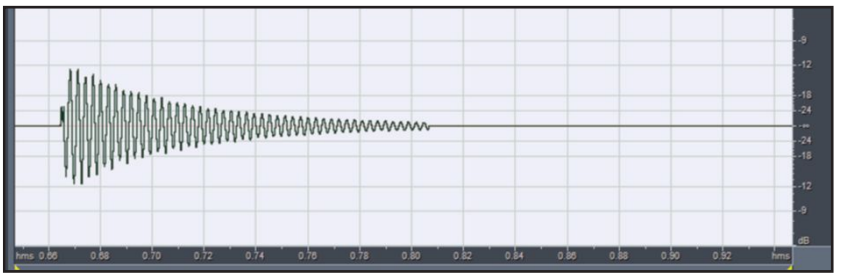


Figure 1: This shows the sound wave produced by a single strike on a bongo under anechoic conditions.

Music played in a room with too little reverberation sounds dry and dead. Since most recording studios have traditionally had little reverberation, a variety of methods have been developed to introduce artificial reverberation into recordings. Some of these methods have been applied to live performance venues as well. The simplest concept for creating artificial reverberation was to build a separate room with acoustically reflective surfaces and no furniture. The signal being recorded was amplified and fed into a speaker in this "reverb chamber," and a microphone was placed at the other end of the room. The reverb signal could be mixed in with the signals of the performers' microphones. Complex shapes produce more pleasing reverb than do simple rectangular rooms.

Simulating Reverberation

As organists know, the room in which a pipe organ resides is a part of the instrument. A pipe organ playing in a "dry" room sounds a lot like a calliope. Thus, when Laurens Hammond introduced the first electric organ in 1935, he had to develop some way to add reverberation so that the instrument would sound like an organ when

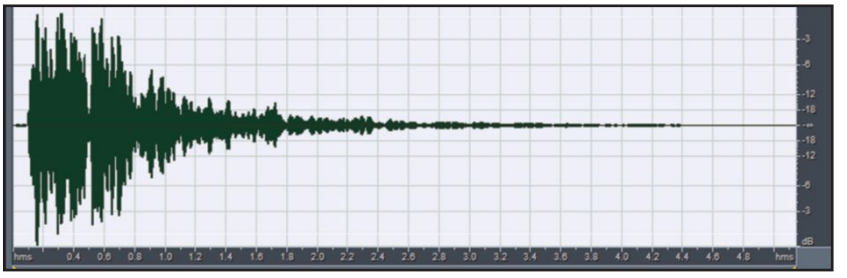


Figure 2: This shows the sound wave produced by a single strike on a bongo with a full reverb tail.

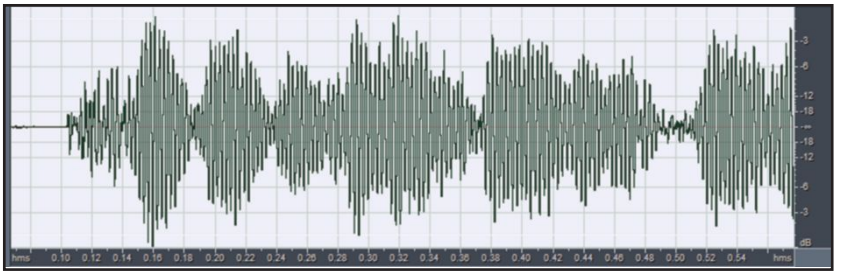


Figure 3: The first half-second of a single strike on a bongo with reverb is shown.



Photo 1: This early Hammond spring reverberation unit is leaning against the tone cabinet in which it is normally mounted.

played in a non-reverberant living room. Adapting a device invented by Bell Labs to simulate the delay on long-distance telephone calls, he developed the first "spring reverb" (see **Photo 1**). This unit was improved and miniaturized through the years, and the "spring reverb tank" was sold under such brands as Gibbs and Accutronics, having become a standard feature of guitar amplifiers and even portable PA systems (see **Photo 2**). While spring reverb units did add interest and a bit of ambience to recordings, no careful listener would ever mistake the sound of a spring reverb for natural acoustical reverberation.



Photo 2: The Accutronics Type 4 spring reverb unit was ubiquitous in guitar and PA amplifiers of the 1960s and 1970s.

A more accurate simulation of acoustical reverberation was provided by plate echo units. These had a speaker-like transducer mounted to a steel plate, with one (for mono) or two (for stereo) contact pickups also mounted to the plate. The RT of a plate reverb could be adjusted by varying the proximity of an acoustically absorbing ("damping") pad mounted close to, but not touching, the plate. Plate reverb units were very large, heavy, and expensive, limiting their use to major studios (e.g., Abbey Road and RCA's Studio B).

Another method used in the past to simulate reverberation was tape echo units. As the name implies, these devices produced echo, not reverberation. But they did add interest to recorded sound.

The famous echo-like sound prominent on many late-1950s rock recordings was produced by amplifying the signal from the playback head of a three-head tape recorder and mixing it back in with the original recording. Since the most common recording speed used then was 30 inches per second (IPS), and the playback head was about an inch from the record head, the echo delay was about 1/30 of a second, or 33.3 ms. Echoes spaced this closely in time are not exactly distinguishable as individual acoustical events, but since they continuously repeated as they diminished in amplitude, they sounded somewhat like non-random reverberation.

Photo 3 shows the heads of a quarter-inch tape recorder. Stand-alone tape echo units were also manufactured, and contributed to the trademark sound of artists such as Chet Atkins. These usually had several playback heads and variable tape speeds, allowing a variety of effects to be produced (see **Photo 4**).

During the 1970s, electronic delay units were introduced, and these have pretty much replaced tape echo devices. First came the analog "bucket brigade" devices (see **Photo 5**). These produced even more electronic noise than the tape echo units—themselves never having won any awards for quietness. Before long, digital delays became available, (see **Photo 6**). And as their prices slowly fell, they competed with analog delays. Today, both types of electronic delays have their aficionados.

Modern Editing Software

By the 1990s, digital .wav-file editing software such as Cool Edit, Sound Forge, and Cubase Audio were offering software-generated reverberation

simulation that was often used in recordings. The reverb signature could be chosen according to the description of a room that would have similar acoustical reverberation: large auditorium, small intimate club, medium concert hall, etc. Interestingly, a setting for simulating a plate reverb was also often available. DSPs began to be included in recording and live-sound mixers, with switchable settings to simulate various types of venues.

Achieving desirable reverberation either acoustically or by digital simulation requires knowing what parameters are involved in the "sound" of a particular room's reverberation. The most obvious parameter is RT, which, as stated earlier, depends upon the room's enclosed volume and the total acoustical absorption in the room.

The human ear/brain perceives room volume by two means: total reverberance and the initial time delay between the direct sound and the onset of reverberation. Thus, changing the room volume has two effects upon room sound. Since acoustically absorptive materials in general absorb more effectively in certain frequency ranges, RT varies with frequency. A room having a longer RT at low frequencies will sound bassy, even if the direct sound is well-balanced. A room with too little high-frequency absorption—hence, a long high-frequency RT—can sound harsh or strident. Rooms having too low a high-frequency RT may sound dull and lacking in intimacy.

As any serious concert-goer knows, the location of the seat (or of the mics used in a live recording) affects the room's sound. The closer to the stage the listener or microphones are, the higher the ratio of direct sound to reverberant sound, and the greater the intimacy, clarity, and articulation. Adjusting these parameters in recording software can help the engineer to achieve the reverb sound wanted in a recording. Using the proper mix and location of reflective and absorptive surfaces can help an acoustician to achieve the reverb sound wanted in a room. Live performers are often limited to certain preset reverb sounds, yet vocalists, wind instrument players, and even Celtic-style fiddlers sometimes use a bit of simulated reverb in performance.

Acoustical Enhancement Systems

Beginning in the early 1990s, researchers developed, and manufacturers then produced, electronic acoustical enhancement systems. These systems used microphones to pick up sound in specific parts of a room, then applied DSPs to add delay and perhaps equalization, and finally amplifying the resulting signal and feeding it back

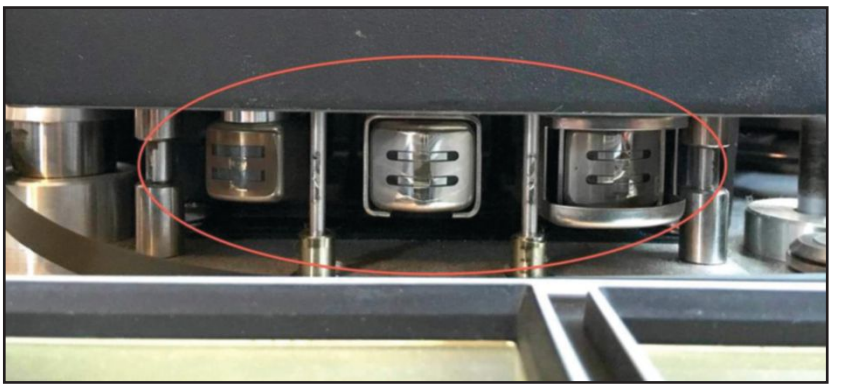


Photo 3: From left to right, the items circled are the erase, record, and play heads of a magnetic tape recorder.



Photo 4: The Watson Copycat had three playback heads and an assortment of controls by which it could be adjusted to create a variety of sound effects. (Image courtesy of www.watkinsguitars.co.uk/copicats.htm)



Photo 5: This analog delay (using a "bucket-brigade" device) could provide a single slap-back echo or a regenerating reverb-like echo.



Photo 6: This digital delay provides similar functionality to the analog delay.

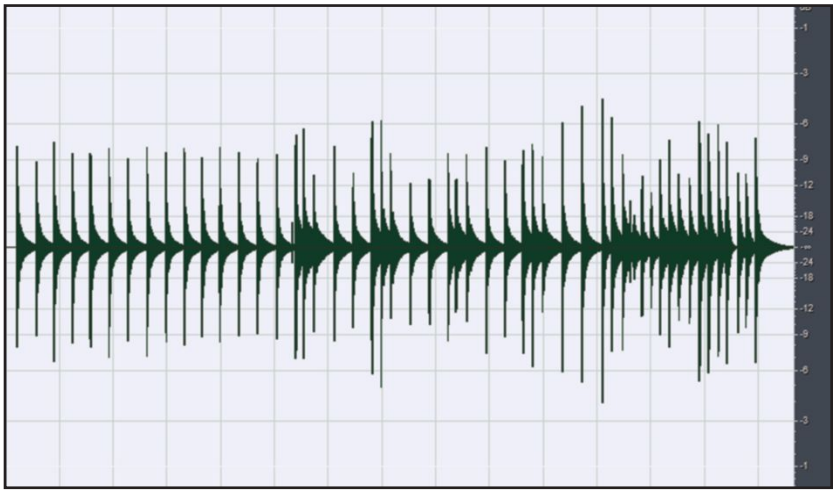


Figure 4: This .wav file shows a recording of a xylophone playing rapidly in anechoic conditions.

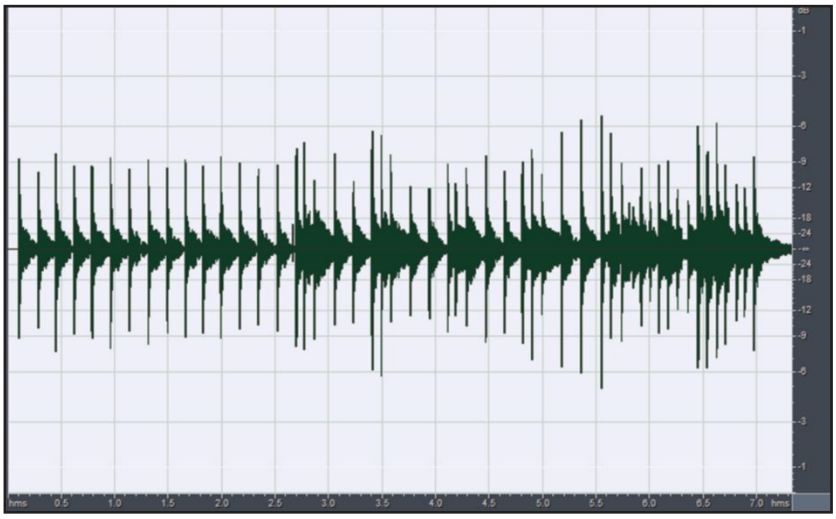


Figure 5: The presence of reverberation adds a reverberant "noise floor" to the xylophone sound.

into the room through carefully designed and located loudspeakers. Some of these "variable acoustics" or "active acoustics" systems depended upon the room being acoustically deadened, and all reverberation was provided by the electroacoustics. Other systems were designed to enhance the room's natural reverberation. In both cases, the result was the ability to change the sound of the reverberation in order to accommodate various types of program material. At present, acoustical enhancement systems are seeing ever-increasing use as they become more affordable, even though their cost is by no means trivial!

Unwanted Reverberation

So far, we have discussed situations in which reverberation is a desirable feature for enhancing musical performance or playback in a venue. There are other situations, such as the one in which the consultant from our earlier example experienced, in which reverberation is detrimental.

From a musical standpoint, the more rapidly music is articulated, the lower must the RT be in order not to blur it. For example, at a vivace tempo (~170 beats per minute), sixteenth notes in 4/4 time are only about 88 ms apart. In a room whose RT is 2 seconds, the sound of a one-sixteenth note will only decay by about 2.6 dB before the beginning of the next note. (Actually, this also depends upon the time required for the reverberation to build up in the room, but you get the idea.)

Figure 4 shows a .wav file of a xylophone playing notes about 100 to 125 ms apart, in anechoic conditions. **Figure 5** shows the same passage played in a reverberant room. You can clearly see the reverberant sound only about 9 dB below the direct sound.

Reverberation similarly affects speech intelligibility. The most common method for measuring and specifying speech intelligibility is the Speech Transmission Index (STI). The concept on which STI is based is that speech is basically an amplitude- and frequency-modulated wave produced by the vocal folds and modified by the lips, tongue, and other parts of the mouth. The room effects, including noise, reverberation, and other factors, can prevent the listener from being able to recover the "intelligence" from the speech, that intelligence being carried by the modulation.

Another detrimental effect of excessive reverberation is noise buildup. This effect is of concern mainly in social spaces such as lobbies, restaurants, and cafeterias. Not only does excessive reverberant energy contribute directly to the noisiness of the space, but people in social situations automatically

speak louder in order to be heard when the space has higher ambient noise. This, in turn raises the ambient noise even more, creating a vicious cycle.

So now the acoustical consultant in our example has determined that the client's problem includes both echoes and reverberation: What solutions can she/he recommend? First, it is important to be aware that what sounds like high reverberation is not always truly high RT. When a sound-reinforcement system is operated just below the threshold of feedback squeal, the resulting room sound can easily be confused with too much reverberation. If there is indeed too much reverberation, usually the only remedy is adding absorption. The least expensive place to do this is most often the ceiling, since acoustical ceiling tiles cost less per square foot than acoustical wall panels or ceiling clouds or acoustical banners.

There are two ways of controlling echoes: absorption and redirection. Often an acoustician will specify 2" to 4" fiberglass panels on the rear wall of a venue to prevent echoes from the speaker system returning to seats in the front of the audience area. If the room already has an appropriate—or even a low—RT, then the sound can be reflected to another area where it will not be as troublesome. An example would be to angle the balcony face of a theater so that sound reflecting from that surface will go over the heads of the audience and performers.

In using this type of redirection, one must be careful that the new place to which the sound reflects will not then re-reflect the sound in such a way as to create problems such as "roundabout" reflections: echoes involving reflections from a succession of surfaces.

The second method involving redirection involves the use of acoustical diffusers that scatter the reflected sound so that it does not arrive in any occupied area with enough intensity to become an annoying echo.

Resources

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OPPO Digital UDP-205

A 4K Ultra HD Audiophile Blu-ray Disc Player

Get an in-depth look at OPPO Digital's new 4K high-definition Blu-ray disc player, the UDP-205, which Gary Galo confirms to be a serious product for all audio enthusiasts.



By
Gary Galo
(United States)

Photo 1: This is OPPO Digital's new top-of-the-line 4K Ultra HD Audiophile Blu-ray Disc Player, the UDP-205. This universal player supports most standard optical disc formats—from conventional CDs through 4K Blu-ray discs—and features state-of-the-art video and processing circuitry. (OPPO Digital photo)

The 4K Ultra-High Definition (UHD) video world is here—a search on Amazon for 4K movies shows at least 1,500 releases on 4K Blu-ray discs, and that list is constantly expanding. And OPPO Digital is ready. Last fall, it introduced its lower-cost 4K player, the UHD-203, which retails for $549 US. In April 2017, it began shipping its new flagship player, the UHD-205, which is the device reviewed here (see **Photo 1**). Complete specifications for the UHD-205 are on the manufacturer's website. As can be seen, both the UHD-203 and the new UHD-205 have identical video circuitry—the differences are entirely in the audio performance.

OPPO Digital UDP-205 4K Ultra HD Audiophile Blu-ray Disc Player
OPPO Digital, Inc.
162 Constitution Drive
Menlo Park, CA 94025
650-961-1118
www.oppodigital.com
Price: $1,299 US

The UDP-205 "universal" player has much in common with its predecessor, the BDP-105, including support for nearly every standard optical disc format. The UDP-205 has added playback of 4K UHD Blu-ray discs to an already thorough array, including regular Blu-ray, Blu-ray 3D, DVD-Video, DVD-Audio, SACD, and CD. Media file support is also exhaustive, and includes AIFF, WAV, ALAC, APE and FLAC, along with Direct-Stream Digital (DSD) audio files in stereo or multi-channel. OPPO Digital has dispensed with Internet movie streaming in the new players. Since most new televisions and many set-top boxes support this, retaining this feature in the player would be redundant and add unnecessary cost. The player does support streaming of audio, video, and photos from storage devices on a home network.

The UDP-205 was designed to be a complete media server and has connectivity similar to the BDP-105. The front panel has one USB 2.0 port and one stereo headphone jack; the HDMI input on the BDP-105's front panel has been eliminated. The rear panel on the new player has a slightly different layout than the BDP-105, but essentially the same connections (see **Photo 2**). Two USB 3.0 inputs are included for connection of external storage devices. The BDP-105 had two HDMI outputs that could be configured using the setup menu for split A/V



Photo 2: This is the rear panel of the UDP-205 digital player. The player features flexible connectivity like its predecessor, the BDP-105, but with a somewhat different layout. (OPPO Digital photo)

operation, where HDMI 1 was the A/V output feeding the television, and HDMI 2 was a dedicated high-resolution digital audio output. If the user required two displays, the two outputs could be configured for dual-display operation. In the UDP-205, the two HDMI outputs are permanently configured as main and audio only. An Ethernet Gigabit LAN connector allows a wired network connection, and home network wireless access includes built-in 802.11ac Wi-Fi. Previous OPPO Digital players came with a USB wireless "dongle" that functioned as a transceiver for wireless network connectivity. With all wireless network hardware built into the new player, you'll always be connected to your home network when the player is on.

All-purpose HDMI outputs are normally tied to the video clock, which is hardly optimum for high-performance audio. Configuring one of the HDMI outputs as "audio only" allowed OPPO Digital to design a high-stability, high-precision HDMI clock and a special HDMI jitter-suppression circuit dedicated to the audio output.

If I use my player with my Benchmark DAC3 HGC external digital-to-analog converter, I use the HDMI audio output to feed a KanexPro HAECOAX HDMI Audio De-Embedder to extract high-resolution PCM discs at full resolution, and output the high-resolution datastream via S/PDIF. (The player's own S/PDIF output is normally down-converted to 48 kHz or 44.1 kHz on copy-protected Blu-ray and DVD-Audio discs.) Many other users will feed the HDMI audio output to an A/V processor or receiver. For anyone using the HDMI audio output, this performance improvement is most welcome. The article "Understanding the HDMI Audio Jitter Reduction Circuit in the OPPO UDP-205" in the Knowledge Base section for this player on OPPO Digital's website offers a detailed explanation of this subject and is well worth reading.

OPPO Digital still includes an HDMI input on the rear panel, for connection of streaming set-top boxes, cable/satellite boxes, game consoles, computers, and other digital players. Digital audio outputs include S/PDIF coaxial and Toslink optical. The player can also be used as a DAC with other digital sources, and includes the three digital inputs found on most DACs—S/PDIF coaxial, Toslink Optical, and a USB 2.0 input using the USB Type B connector that's standard on USB DACs. On the analog end, there are eight RCA connectors for the 7.1-channel surround outputs, along with two XLR and two RCA connectors for the dedicated stereo outputs. OPPO Digital also includes trigger in and out connectors, infra-red remote sensor connector for receiving remote control signals from an IR distribution system via an IR emitter or blaster, and an RS-232 serial control port.

New ESS DAC

In October 2016, ESS introduced its second-generation Sabre32 HyperStream DAC chips, the ES9028PRO and ES9038PRO, and many manufacturers that used the ES9018 have designed new products based on one of these chips. Benchmark Media Systems is using the ES9028PRO

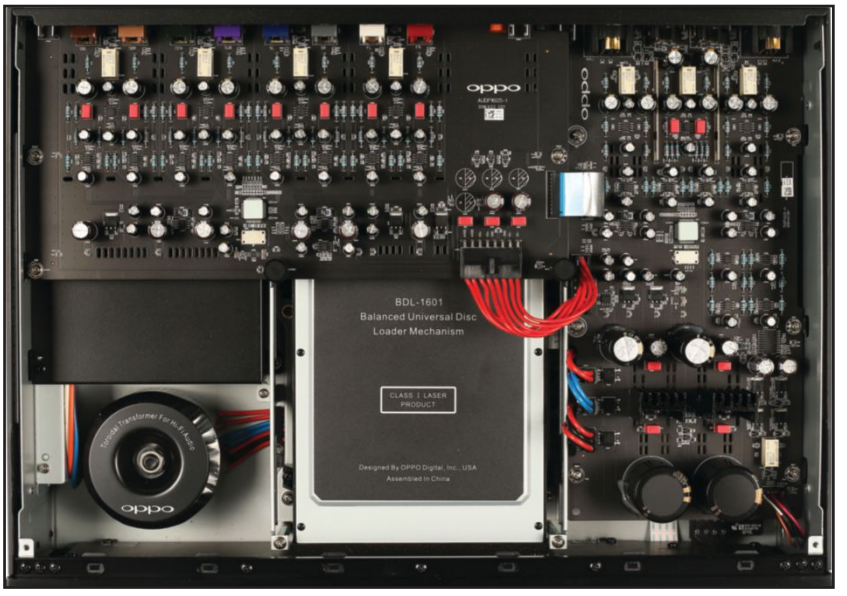


Photo 3: This is an inside view of the UDP-205. The 7.1-channel surround board is in the upper left. The dedicated two-channel stereo board is on the right. The toroidal transformer used for the DAC and analog circuitry's linear power supply is in the lower left, and the custom-designed 4K disc loader and mechanism is in the lower center. The shielded video circuitry and switching-mode power supply are housed in the bottom of the player. (OPPO Digital photo)

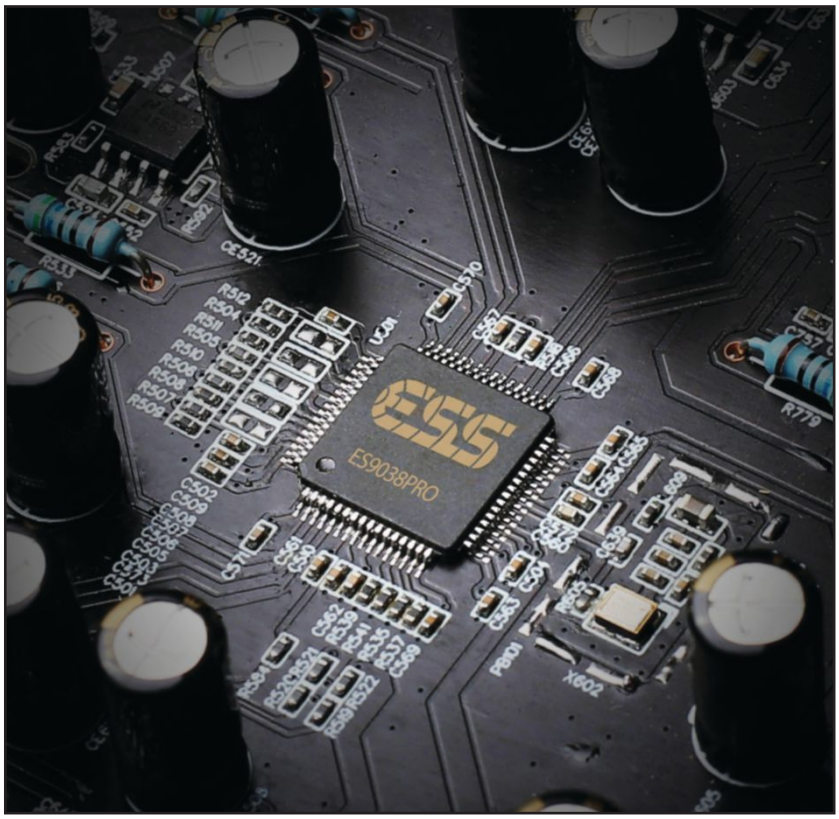


Photo 4: This close-up shows the ES9038PRO DAC chip on the stereo audio PC board. The chip is normally masked by a white heatsink, and the precision low-phase noise crystal oscillator to its right is covered with a metal shield. (OPPO Digital photo)

About the Author

Gary Galo retired in 2014 after 38 years as Audio Engineer at The Crane School of Music, SUNY at Potsdam, NY. He now works as a volunteer in the Crane Recording Archive doing preservation, restoration, and digital transfer of vintage Crane recordings. He is also a Crane alumnus, having received a BM in Music Education in 1973 and an MA in Music History and Literature in 1974. Gary is a widely published author with more than 300 articles and reviews on both musical and technical subjects, in over a dozen publications. Gary has been writing for *audioXpress* and its predecessors since the early 1980s. He has been an active member of the Association for Recorded Sound Collections (ARSC) since 1989, and a frequent recording and book reviewer for the *ARSC Journal*. He has given numerous presentations at ARSC annual conferences, many of which have been published in the *ARSC Journal*. He was the Sound Recording Review Editor of the *ARSC Journal* from 1995-2012, and co-chair of the ARSC Technical committee from 1996-2014. Gary has also published numerous book reviews in *Notes: Quarterly Journal of the Music Library Association*, written for the Newsletter of the Wilhelm Furtwängler Society of America, *Toccata: Journal of the Leopold Stokowski Society*, and he is the author of the "Loudspeaker" entry in *The Encyclopedia of Recorded Sound* in the US. He has also written several articles for *Linear Audio*. He is a member of the Audio Engineering Society, the Boston Audio Society, and the Société Wilhelm Furtwängler.

in its DAC3 HGC (which I reviewed in *audioXpress*, July 2017), and OPPO Digital selected the ES9038PRO for the UDP-205. As I discussed in my review of the Benchmark DAC3-HGC, the ES9038PRO and the ES9028PRO are both eight-channel chips and contain identical circuitry. The ES9038PRO has multiple paralleled DACs for each of the eight channels and requires a heatsink. Liao notes that ESS doesn't recommend paralleling sections of the ES9038PRO externally with IC op-amp I/V converters because of excessive output current.

Among the virtues of these new chips is the inclusion of eight pre-set and programmable digital filters, improved power supply distribution, and total harmonic distortion compensation that can even reduce distortion caused by external components. The new ESS DACs also support DSD over PCM using the DoP protocol.

Photo 3 shows the inside view of the UDP-205. The player has completely separate analog/DAC PC boards for the 7.1-channel surround and two-channel stereo outputs. Each analog/DAC board has its own dedicated ES9038PRO DAC chip (see **Photo 4**). Each of the eight channels is used for the surround outputs on the 7.1-channel board. On the two-channel stereo board, only six sections of the DAC chip are used—two for the stereo unbalanced outputs, two for the balanced outputs and two for the headphone amplifier. The clock for each ES9038PRO DAC chip is a precision crystal oscillator with low phase noise, covered with a metal shield.

The ES9038PRO is not a drop-in replacement for the ES9018. Jason Liao, OPPO Digital's Chief Technology Officer and VP for Product Development, notes that although an older PC layout can be adapted to the new chip, "A new layout will better utilize the ES9038PRO's performance, since the clock and power supply can be improved." A comparison of the PC board photos shows a complete redesign of the layouts around the DACs to get the best performance from the new ESS chip. OPPO Digital has implemented seven digital filters in the ES9038PRO, user-selectable in the setup menu. These include a variety of minimum-phase, linear-phase and apodizing filters. The default is Minimum Phase Fast.

The IC op-amps for most functions on the stereo board are the same Texas Instuments (TI)/National LM4562 types used in the BDP-105. But, OPPO Digital has changed the fully-differential op-amps used for the balanced outputs from TI/National LME49724 to TI/Burr-Brown OPA1632. The OPA1632 exceeds the performance of the LME49724 in several key areas, including distortion (0.000022% for the OPA1632 vs. 0.00003% for the LME49724), slew rate (50 V/µS

vs. 18 V/µS), gain-bandwidth product (180 MHz vs. 50 MHz), and noise (1.5 nV/√Hz vs. 2.1 nV/√Hz). They also use a pair of OPA1632s to feed the differential inputs on the TI TPA6120A headphone amplifier. Although the same headphone amplifier was used in the BDP-105, in the new player the outputs are buffered with a discrete push-pull, high-current output stage using the J243/J253 complimentary bi-polar transistor pair, which provide higher power and lower output impedance than the stand-alone TPA6120A. These 4 A, 40 MHz transistors should drive any headphone with ease.

The op-amps on the multi-channel board have been changed from LM4562 to OPA1642A types. All analog outputs are capacitor coupled with the same Elna 100 µF/16 V Silmic II capacitors used in the BDP-105. These advanced capacitors are designed specifically for the best possible audio performance and employ a silk-fiber dielectric. (For a complete description of these capacitors, see my review of the BDP-105, *audioXpress*, October 2013).

OPPO Digital continues the practice of using a switching-mode power supply for the video and control circuitry, and a linear supply for the analog and DAC circuitry. The front-end of the linear

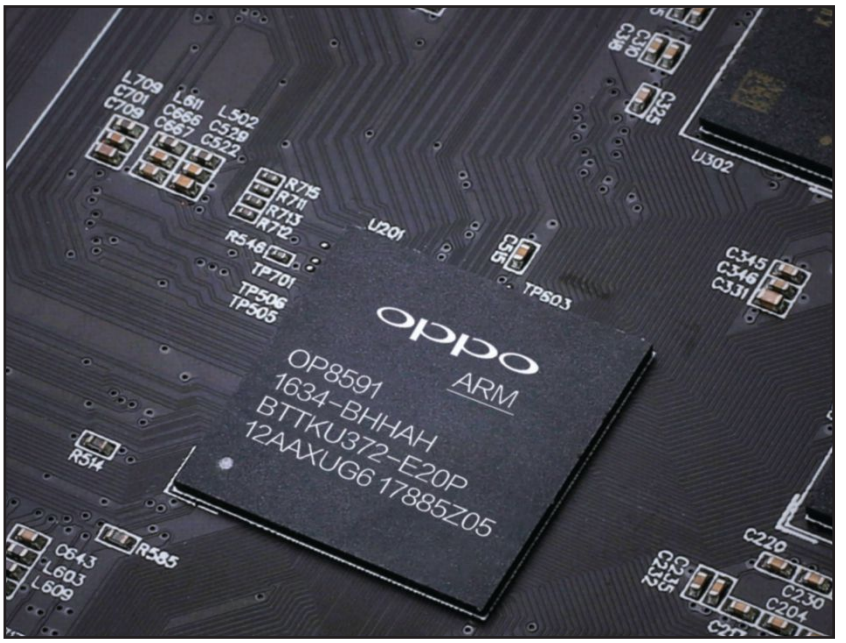


Photo 5: This close-up shows the OP8591 decoder chip on the video PC board. This complex integrated circuit was a joint development between OPPO Digital and chip manufacturer MediaTek. It includes a quad-core CPU and performs all decoding and video processing functions. (OPPO Digital photo)







Photo 6: The UDP-205 remote control looks virtually identical to the BDP-105 remote. The old Netflix and Vudu buttons have been eliminated, and the illumination button in the lower right has been replaced with the HDR output mode selector. They've also eliminated the 3D button and replaced it with a PIC button that enables quick access to the picture adjustment menu. (OPPO Digital photo)

supply is a custom-designed toroidal power transformer, with three sets of secondary windings dedicated to the analog and digital supplies.

One winding feeds a pair of bridge rectifiers and the main 7812/7912 pair of IC regulators used for the analog circuitry. These main regulators are used with a pair of 6800 µF/35 V input filter capacitors and 3300 µF/50 V output filter caps. The two other windings feed bridge rectifiers dedicated to the digital supplies, which are regulated with AZ1117 three-terminal IC regulators. Local analog supply bypassing is generous, with 32 capacitors, each 220 µF/35 V. All electrolytic power supply capacitors are Elna audio-grade parts.

New Video Decoder

Previous OPPO Digital players had separate decoder and video processor chips. For its new 4K UHD players, OPPO Digital teamed up with MediaTek to design an entirely new chip, the OP8591, which combines both functions in a single package (see **Photo 5**). Liao notes that video processing is done with a combination of hardware, digital signal processing, and firmware. He says that they've applied what they learned from external video processing chips to ensure that the integrated processing meets their quality requirements. The OP8591 is extremely complex and was an enormously expensive effort. It's a quad-core design using ARM (Advanced RISC Machine) architecture, plus DSPs specifically designed for video and audio decoding and processing. The chip also incorporates all of the security features required by the copyright control mechanisms. The video circuitry and its switching-mode power supply are housed in the bottom of the player and are fully shielded to eliminate electrical interference with the DAC and analog circuitry.

The UDP-205 supports all current SD, HD, and UHD resolutions, including 4K at 60p, 4K at 50p, 4K at 30p, and 4K at 24p using various color spaces (e.g., PC RGB, Video RGB, YCbCr 4:4:4, 4:2:2, and 4:2:0). The UDP-205 will also play 4K media files and user-generated content. Decoding support includes HEVC, H.264, VP9 4K, and Hi10P video codecs. High Dynamic Range (the HDR 10 format) and Wide Color Gamut are also supported, and the player includes accurate conversion from HDR to standard dynamic range (SDR) for compatibility with older televisions. Lower-resolution video can be upscaled to 4K. This player is also the first to support Dolby Vision, a feature added in a recent firmware upgrade. OPPO Digital's high-precision disc loader and mechanism ensure fast loading times and reliable playback, with effective error detection and correction. All of the standard, advanced audio formats are supported, including Dolby TrueHD and DTS-HD Master Audio, Dolby Atmos, and DTS:X.

On the audio end, the UDP-205's S/PDIF and Toslink Optical inputs will support PCM files up to 192 kHz/24-bit. The USB DAC input will support two-channel DXD (Digital eXtended Definition) PCM files up to 768 kHz, and two-channel DSD files at 2.8224 MHz (DSD 64), 5.6448 MHz (DSD 128), 11.2896 MHz (DSD 256), and 22.5792 MHz (DSD 512). DSD 64 and 128 files are played in native mode—anything higher is converted to PCM. The USB inputs for media storage support PCM up to 192 kHz/24-bit and DSD at 2.8224 MHz.

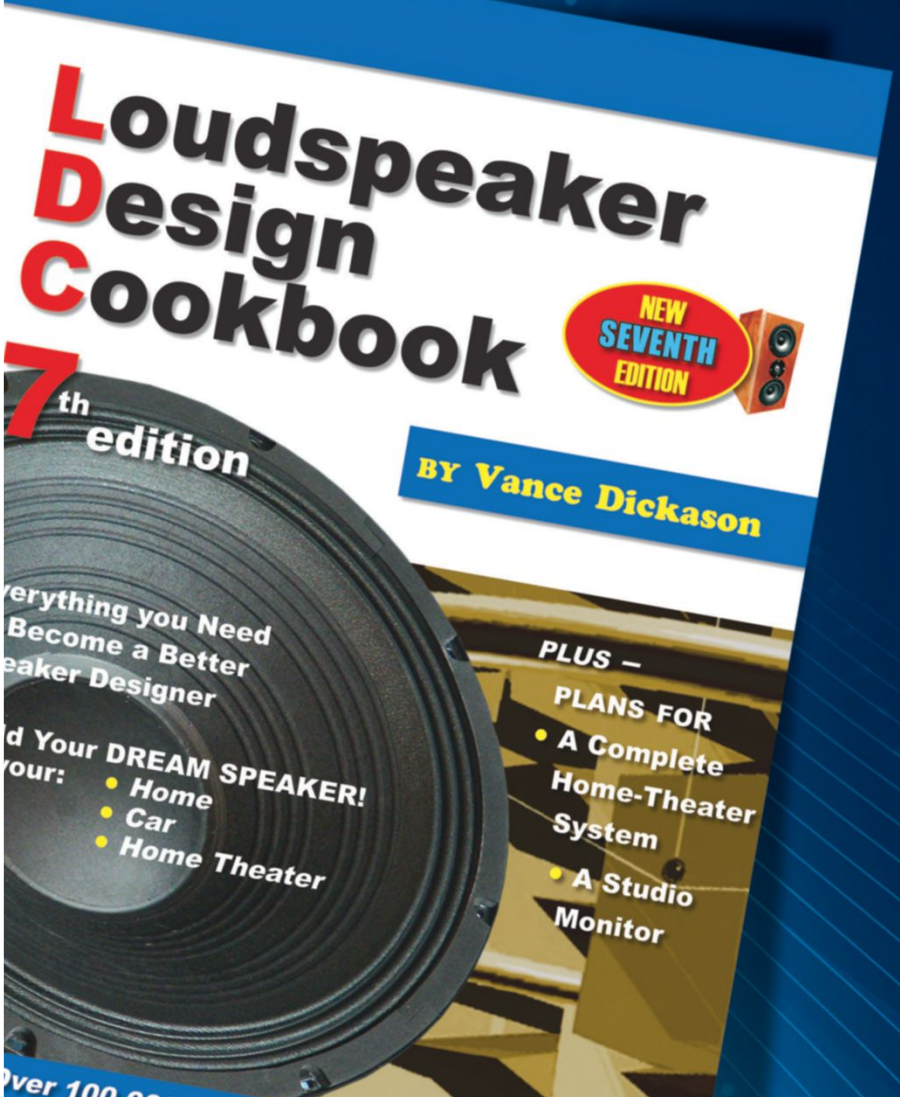
HDCD Solution

OPPO Digital's previous Blu-ray players have included support for High-Definition Compatible Digital (HDCD) discs. This format was introduced in 1995, co-invented by Keith O. Johnson and Michael "Pflash" Pflaumer of Pacific Microsonics (and Reference Recordings), as a means of encoding 20-bits of resolution on a 16-bit Red Book CD. Although high-resolution audio formats have supplanted HDCD, many collectors have libraries of discs in this format and will want to continue playing them at full resolution. The new OPPO Digital players don't support HDCD. Liao explained that in the previous OPPO Digital players, HDCD decoding was done in the main decoder chip, but MediaTek was unable to include HDCD support in the OP8591. It was simply a matter of balancing the feature requirements with hardware design constraints.

There's an easy and cost-effective work-around to this problem. A program called dBpoweramp includes a CD ripper that will decode HDCDs and write the decoded data to 24-bit .wav files. For reasons unknown to me, only the Windows version supports this feature. If you use a Mac, find a friend with a Windows computer. You must purchase and register the program to get this feature, but the program costs $39 US—worth the price for anyone with an HDCD collection. I only have a handful of HDCDs, but the program worked great

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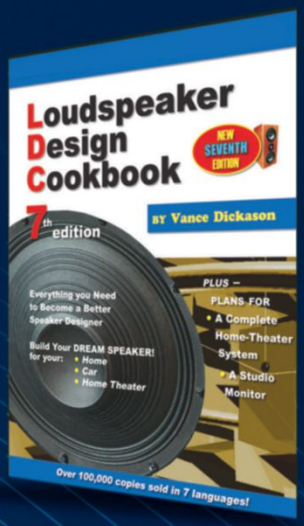




Photo 7: This cutaway view of the UDP-205 shows the dual-layer reinforced metal chassis. The brushed-aluminum front panel brings the total weight to 22 lb (10 kg). (OPPO Digital photo)

on all of them. You can play the 24-bit wave files on the UDP-205 directly from any USB storage device.

The UDP-205's remote control is very similar to the one supplied with the BDP-105 (see **Photo 6**). Since the new player doesn't support Internet movie streaming, the dedicated Netflix and Vudu buttons have been eliminated. An HDR button has been added to select the output mode for HDR content, and replaces the illumination button on the BDP-105 remote. The buttons on the new remote illuminate as soon as you pick it up or press any button; they remain illuminated for 5 seconds. They've also eliminated the 3D button and replaced it with a PIC button that enables quick access to the picture adjustment menu.

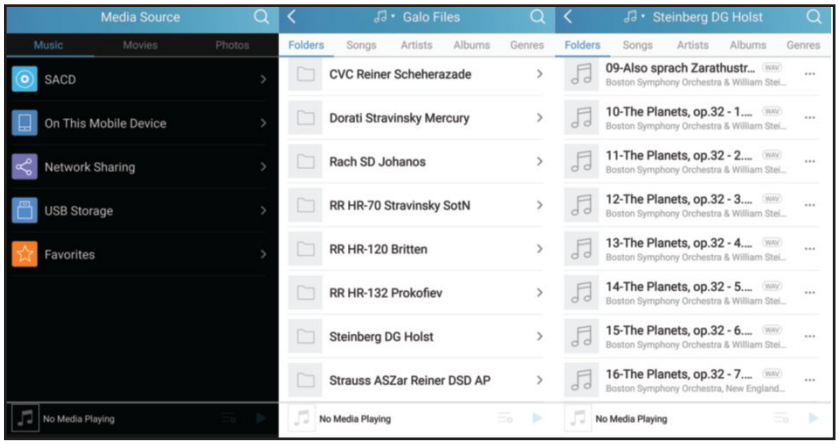


Photo 8: Here are three screenshots from OPPO Digital's Media Control App for Android devices. The app enables you to select a source (left), navigate folders on external USB drives (center), and select the files you wish to play (right). From the screen on the left you can also select Remote, which duplicates the functions of the player's remote control, and Player Setup, which enables you to do the entire setup without turning on your TV. The shot on the left shows the app in Night Mode, which will probably extend the charge on the device's battery.

The build quality of the OPPO Digital players has always been excellent, but each generation of players seems to be more massive and robust than the previous. The UDP-205 is built on a double-layer reinforced metal chassis to improve stability and resist vibration (see **Photo 7**). The player also has a brushed-aluminum front panel which, combined with the steel chassis and the circuitry itself, brings the total weight of this player to a hefty 22 lb (10 kg). Compare this to 17.3 lb (7.9 kg) for the BDP-105 and 16 lb (7.3 kg) for the BDP-95.

The HDMI Phantom

My television is in my audio listening room, and we also have a second set upstairs in the living room. I have no interest in surround sound—video requiring more than the "news" quality audio plays through my stereo audio system. Back in February, I replaced my 42" Panasonic Plasma TV, Viera-series model TH-42PZ85U (we moved that set upstairs to the living room). The replacement is a Sony model XBR-49X800D, a 4K LED set that's part of the Bravia series. When we bought it in 2008, the Panasonic had one of the best TV pictures I had ever seen and was excellent with 1080p Blu-ray discs played on the OPPO Digital BDP-93, BDP-95, and BDP-105 players that I've had in my system.

After installing the new Sony television, I began experiencing an audio glitch that never happened with the Panasonic. If I'm playing an audio disc on the OPPO Digital BDP-105 or UDP-205, with the Sony TV turned off, the audio will periodically mute for second or so and then come back on. (There's never a problem when the TV is on.) This will happen if I'm using the OPPO Digital as a stand-alone player or with an outboard DAC. If I'm using my outboard converter, the DAC's digital lock light will momentarily go out. I contacted Liao about this and he replied as follows: "I am guessing that the mute is caused by the HDMI handshake with the TV. Older TVs tend to completely shut off the HDMI input ports when it is powered off, so there is no problem. Some new TVs do not shut off the HDMI inputs when they're turned off, so the player still receives the "hot plug" signal and will try to perform a handshake with the TV based on the existence of the hot plug signal. The handshake causes the clock to reset. Completely cutting off AC power to the TV can be a work-around to the problem."

As I'll explain shortly, Liao's guess was right on target. Cutting power to the TV does solve the problem, as does unplugging the HDMI cable from the TV. But, neither of these options is particularly convenient, and the TV takes a minute or so to reboot when power has been disconnected. Some

TVs may lose certain setup information if unplugged for an extended period. My solution was to purchase an HDMI switch and insert it in the line between the OPPO Digital player and the Sony television. I plug the HDMI switch's wall-wart power supply into a switched outlet on the power line conditioner dedicated to my TV and cable DVR. The TV and DVR are plugged into outlets that are always on, so I simply turn on the power line conditioner's front panel switch to turn on the HDMI switch when I need video from the OPPO Digital player. This solved the muting problem.

You must get a 4K switch that supports high-bandwidth digital content protection (HDCP), specifically HDCP 2.2. The OPPO Digital UDP-205 supports this copyright protection standard, which is mandatory for all 4K UHD players. Mine is an Expert Connect 3x1, a three-input switcher available from Amazon for around $36 US. The manufacturer claims compatibility as follow: Ultra HD 4K/2K at 60 Hz (60 fps), HDR, HDMI 2.0, HDCP 2.2, Full HD/3D, 1080P, DTS, Dolby Digital, Direct TV, and 18 Gbps bandwidth.

The bandwidth is important for both the switcher and the HDMI interconnect cables. There are a lot of "4K" HDMI cables that don't offer full 18 Gbps bandwidth—avoid them, especially if you plan to play HDR discs. There are many brands that meet the requirements for 4K video, and one way to guarantee performance is to buy cables approved by the Premium HDMI Cable Certification Program. (For more information, visit the official HDMI website.) I purchased two "Premium High-Speed HDMI with Ethernet" cables made by On-Q Legrand from Crutchfield, 7 m from the player to the HDMI switch, and 1 m from the switch to the TV (see Resources). They work just great.

The Expert Connect 3×1 has LEDs to indicate both input and output connections. Even with my Sony TV turned off, the output LED remains illuminated, indicating a connection to a live HDMI input. You have to unplug the Sony TV to turn off the output LED. This confirms Liao's explanation of the muting problem. Fortunately, the Expert Connect switch passes 4K video without degradation. The only thing it doesn't support is the HDMI Audio Return Channel (ARC). If you need to get audio from your TV back to your audio system, I suggest a Toslink Optical digital interconnect, between the TV and the Toslink input on the OPPO Digital player. Vanco's HDMISW41 switch supports ARC, but it costs $149. I haven't tried it.

Media Control App

I highly recommend downloading OPPO Digital's free Media Control App for your smartphone or other portable device (you must have wireless network capability for your smart device to use it, since it communicates with the player through your home network). On the OPPO Digital support page, select your player and then scroll down to the links for the Android and the iPhone/iPad/iPod Touch versions. **Photo 8** shows three screenshots from the Android app, which I use on my Samsung Galaxy Note 4. The app enables you to select any active media source, including the player's own internal optical drive and all external USB storage devices. The menu system then enables you to navigate folders on the external storage device and select files for playback, all without ever turning on your television.



Photo 9: The most complete player from OPPO Digital, both in terms of video and audio formats supported, the UDP-205 is also the highest quality source any audio enthusiast could want.

You can also select a screen that duplicates the functions of the OPPO Digital remote control, and perform all player set-up functions, including upgrading the firmware. The only function that still requires you to turn on a television is the program selection on DVD-Audio discs. Setup is similar to previous OPPO Digital players—mostly easy and intuitive. The User Manual is detailed and clearly written. Updated versions—sometimes necessary because of firmware changes—are always available as a free download on the OPPO Digital website. One thing I'd like to see improved is an explanation of how to use the trigger I/O connections. The manual says very little about this, and doesn't specify the required trigger voltage (it's probably 12 VDC).

Video Quality

I purchased my first 4K UHD disc specifically for this review, a film made for IMAX theaters called *Journey to Space*, which describes the contributions made by the Space Shuttle and the planning

underway for a trip to Mars. The film is narrated by Patrick Stewart, and the package also includes a regular Blu-ray disc. On the UDP-205, the 4K picture quality is simply stunning—razor sharp with incredibly low background noise, amazing contrast and vivid, life-like colors. This disc also has a HDR program—the HDR function is selected on the disc's opening menu.

The difference in sharpness and contrast is very obvious with HDR, though I found the color saturation to be a bit too much, requiring a re-adjustment of the color level (which can be done either on the TV or the OPPO Digital player). I find that factory-default color settings on televisions are almost always too high. I prefer a natural rather than overly saturated picture. OPPO Digital's factory default settings on its players have always been sensibly chosen. The UDP-205 enables you to customize and save three picture adjustment modes, so I used Mode 2 to reduce the color saturation for HDR discs. Naturally, as I get other HDR discs, this may change. The PIC button on the OPPO Digital remote, described earlier, makes access to the picture adjustments quick and easy.

The UDP-205 does a superb job of upscaling regular 1080p Blu-ray video to 4K UHD. I was surprised at just how close the *Journey to Space* Blu-ray disc came to the 4K UHD disc when upscaled by the player. I suspect that the differences will be more apparent on much larger screens than mine, but on my 49" Sony, the Blu-ray disc still qualifies as stunning. Surely, we have not reached the point where 1080p Blu-ray discs get no respect!

Audio Performance

All of my listening was done with the player's dedicated two-channel stereo outputs, using the default Minimum Phase Fast filter. I'll get right to the point—the UDP-205 is a stellar audio player. In my review of the BDP-105, I noted a number of areas where that player improved upon the already fine audio performance of the BDP-95. If anything, the differences between the UDP-205 and the BDP-105 are even greater, especially on a high-resolution system.

The ES9038PRO DAC chip and other design changes made by OPPO Digital have laid the sonic virtues of its predecessor on a new ground. In my review of the Benchmark DAC3 HGC, I noted the improved soundstage reproduction, and how the ES9028PRO chip revealed the connecting acoustic space between the instruments better than the ES9018. The new OPPO Digital player reveals similar virtues in the ES9038PRO. The soundstage is more three-dimensional and precise, with a more realistic sense of the acoustics of the original recording venue. In the 1959 RCA Living Stereo recording of Sergei Prokofiev's *Alexander Nevsky* with Fritz Reiner and the Chicago Symphony, on Analogue Productions SACD transfer (CAPC 2392 SA, from Elusive Disc or Acoustic Sounds), the player realistically reproduces the acoustic space around the low brass instruments in the rear of the soundstage, compared to the more homogenized sonic presentation rendered by the BDP-105.

In Deutsche Gramophone's new 96 kHz/24-bit Blu-ray transfer of its 1966 Bayreuth Festival recording of Richard Wagner's *Tristan und Isolde*, conducted by Karl Böhm, the UDP-205 reveals an

Resources

2L, www.2l.no/hires/index.html.

Acoustic Sounds, http://store.acousticsounds.com.

Cirlinca HD-Audio Solo Ultra, www.cirlinca.com.

dBpower amp Software, www.dbpoweramp.com.

Elusive Disc, Inc., www.elusivedisc.com.

G. Galo, "OPPO Digital BDP-93 Blu-ray Disc Player," *audioXpress*, June 2011.

_____, "OPPO Digital BDP 95 Universal Network 3-D Blu-ray Disc Player," *audioXpress*, January 2012.

_____, "OPPO Digital's New 3-D Blu-ray Disc Player Raises the Bar," *audioXpress*, October 2013.

_____, "KanexPro HAECOAX HDMI Audio De-Embedder," *audioXpress*, July 2016.

_____, "Benchmark DAC3 HGC Stereo D/A Converter," *audioXpress*, July 2017, www.audioxpress.com/article/fresh-from-the-bench-benchmark-dac3-hgc-stereo-d-a-converter.

HDtracks, www.hdtracks.com.

HDMI, www.hdmi.org.

OPPO Digital, www.oppodigital.com.

Reference Recordings, https://referencerecordings.com

USB Audio Player Pro, https://play.google.com.

Sources

AudioQuest Carbon USB 3.0 Cable
Audio Advisor, Inc. | www.audioadvisor.com

AC2MP1-BK (1 m) and AC2AP7-BK (7 m) On-Q Legrand premium High-Speed HDMI cables
Crutchfield | www.crutchfield.com

Vanco HDMISW41 Switch
Markertek | www.markertek.com

amazing level of orchestral detail not heard in previous digital transfers of this recording (479 7291, available from Amazon). And, the positions of the singers in the soundstage are rendered with greater precision, and the OPPO Digital player reveals an improved sense of depth on this recording. I've owned the German-pressed LPs since the 1970s. If anything, the Blu-ray disc, played on the UDP-205, is warmer than the vinyl.

Digital audio technology has improved considerably over the past three decades, yet I'm often amazed at how well the CD transfers of the Mercury Living Presence recordings, produced by Wilma Cozart Fine between 1990 and 1995, have held up. The Mercury CD of Ottorino Respighi's *The Birds*, with Antal Dorati and the London Symphony, vividly captures delicacy of Respighi's orchestration, the amazing soundstage, and palpable sound of the massed strings. On the UDP-205, this and other Mercury CD transfers in my collection sound surprisingly up-to-date.

Tonal neutrality has been a virtue in both of OPPO Digital's ES9018-based players, and the UDP-205 certainly retains those qualities. The Benchmark DAC3 HGC is my reference for transparency and neutrality. By comparison, the BDP-205 leans slightly toward warmth in the midrange and is slightly laid back in the treble (I emphasize slightly). There's nothing lacking in high-frequency extension, however. Heavily multi-miked recordings that favor the treble region, like William Steinberg's DG recording of Gustave Holst's *The Planets*, are reproduced with even greater listenability than they were on the BDP-105, particularly in the passages where the high strings are prominent (heard on the 96 kHz/24-bit files from HD Tracks).

OPPO Digital has also improved the low-frequency extension and weight in the new player, evident on Igor Stravinsky's *Song of the Nightingale* (Eiji Oue, Minnesota Orchestra, 176.4 kHz/24-bit files, Reference Recordings HR-70). The new player very accurately and impressively delineates the timbres of the various bass-drum whacks in this recording. The percussion instruments on Oue's recording of Aaron Copland's *Fanfare for the Common Man* are also very impressive in this regard, using 24-bit files extracted from the HDCD using dBpoweramp (Reference Recordings HDCD RR-93).

Some OPPO Digital owners may be reluctant to upgrade to the UDP-205 player because of the lack of HDCD support. They should rethink this view. Because of the UDP-205's superior audio performance compared to the BDP-105, the 24-bit decoded files made with dBpoweramp actually sound better on the new player than the original discs did on the old one. Improved spatial qualities and detail are evident on the Copland recording, as well as Frederick Fennell and the Dallas Wind Symphony performing Václav Nelhýbel's *Trittico* (Reference Recordings HDCD RR-52). For best performance playing media files, I suggest a USB 3.0 external drive connected to one of the USB 3.0 inputs on the player with a high-quality cable. I use an AudioQuest Carbon.

Conclusions

With the UDP-205, OPPO Digital has a new, reference-quality player, redefining what's possible with a high-performance universal digital player, at a very affordable price (see **Photo 9**). It's possible to improve audio performance even further with the addition of an outboard DAC. But, you'll need to pay at least as much, and possibly a lot more, for the outboard DAC as you did for the UDP-205. It will take outboard DACs at least approaching performance of the Benchmark DAC3 HGC to get you to the next level of transparency and refinement. Many audio enthusiasts will question the need for anything better—the performance of the UDP-205 is that good. The UDP-205 is a new reference for stand-alone digital players. I could not recommend it more highly.



Ultimate Ears Pro Sound Tap Personal Monitoring System

Your Own Easy-to-Use In-Ear DI Box



Photo 1: The Ultimate Ears Pro Sound Tap has a solid, functional appearance. (Cynthia Wenslow photo)

In its continuous quest to bring the benefits of in-ear monitoring to the masses, Ultimate Ears (UE) Pro released Sound Tap, enabling musicians to plug in their custom or universal in-ear monitors and turn the existing stage monitor mix into an in-ear mix. Stuart Yaniger measures the device to see how it behaves when relaying sound.

By
Stuart Yaniger
(United States)

Many performers have eschewed the use of conventional fold-back monitors for stage use. Besides the difficulty of setup to prevent feedback, it's often tough to hear yourself over the noise that the other folks on stage make. A popular alternative in larger venues is the use of in-ear headphones or in-ear monitors (IEMs), which totally avoid feedback and can provide a significant degree of acoustic isolation. Ultimate Ears is one of the foremost suppliers of in-ear headphones at the premium end of the market, and its line of stage-focused headphones features models with frequency responses and sensitivity optimized for different instruments.

Of course, when you decide to go this route, the first problem that pops up is, where do you plug in your headphones? The mixing board is pretty remote, wireless connections in crowded arenas are dicey at best, and hookups on stage can be challenging and don't easily allow the performer to control the volume. And let's not even mention hum issues.

Ultimate Ears (in this case, its professional division UE Pro) created the Sound Tap DI Box as an easy solution. It allows the sound to be tapped from a line level (mixing board) or speaker level feed, while passing the signal through to its intended destination unmolested. The signal is sampled and sent to a performer-adjustable headphone amplifier.

Physically, the Sound Tap is packaged in a heavy-duty cast Hammond case with a black wrinkle finish and easy-to-read legends (see **Photo 1**). Input and output connectors are combo XLR/TRS/TS for line level and Speakon two pole for speaker level, switch selectable. Headphone output is via a standard 3.5 mm phone jack. Two 9 V batteries are used for power,

Sound Tap
Ultimate Ears (UE) Pro
http://pro.ultimateears.com/products/custom-monitors/for-stage/uesoundtap
Cost: $249

with LED indicators (green and red) for battery condition. The entire package is quite rugged and survived a lot of banging around when I used it, perfect for touring musicians.

Inside, the unit is built on two thick double-sided PC boards with gold-plated traces (see **Photo 2**). Most of the circuitry is surface mount and is constructed of high-quality parts. The signal handling is done using Analog Devices AD822 op-amps, an excellent choice for battery power because of their low current consumption and ability to swing rail-to-rail in a single-ended power supply topology.

Retail price of the Sound Tap is $249, but I've seen street pricing at less than $200.

Sound Tap is easy to set up and use—plug in the signal source and receiver, turn the Input Level knob up until you see the green LED light, adjust the Master Volume to a comfortable level, and you're done! If the system is overdriven, a red LED will flash, and the headphone amplifier is equipped with a fold-back voltage limit to help prevent hearing damage in the event of an unplanned… incident (e.g., someone hot-plugging a guitar, which of course would never happen).

Note that although it is set up to run stereo headphones, Sound Tap is single channel—there are two separate op-amps used for each channel that output identical signals. For performance monitor use, this should not be a limitation (there's only one of you), but it may or may not be suitable for monitoring the overall mix. But that's why you pay the front-of-house (FOH) sound guy, right?

Performance

I first did the basic measurements, using an Audio Precision APx515 audio analyzer and an APx1701 transducer interface to drive the speaker inputs. Measurements taken using the line level and speaker level inputs were similar enough that I can show them here interchangeably.

The line level input impedance measured about 8 kΩ at either input, high enough not to bother mixing boards or amps in their pass through. I did not see much variation in the input impedance with frequency over the audio band.

Output impedance of the headphone amp was a bit high at 21 Ω—if your headphones have significant variations in their impedance at different frequencies, this could cause some frequency response anomalies. Note, though, that Sound Tap's basic function is performance monitoring rather than use for fine adjustments of EQ. And if your headphones have a relatively high impedance (100 Ω or higher) the effect of the source impedance on frequency response will be negligible.

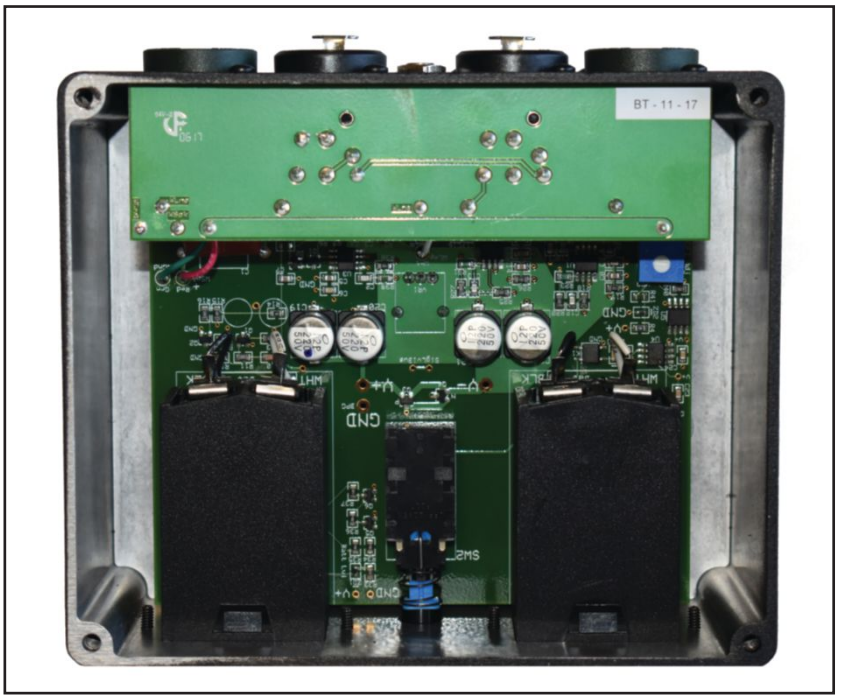


Photo 2: The circuitry of the Sound Tap is neatly laid out and constructed with high-quality parts. (Cynthia Wenslow photo)

Because the in-ear headphones from Ultimate Ears for which this unit is presumably optimized have relatively low impedances, my testing was done with 16 Ω loads, one on each channel. Since the two channels are driven with separate op-amps, there was a possibility that they wouldn't match exactly, but I found that in actuality the matching was quite tight. I adjusted the input and output signals to

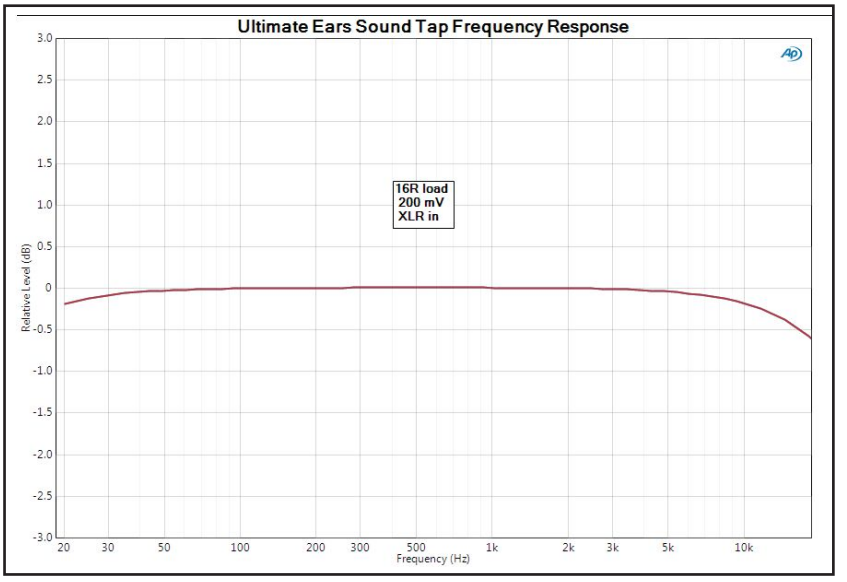


Figure 1: The frequency response of the Sound Tap into a 16 Ω load shows a slight rolloff at the top of the audio band.

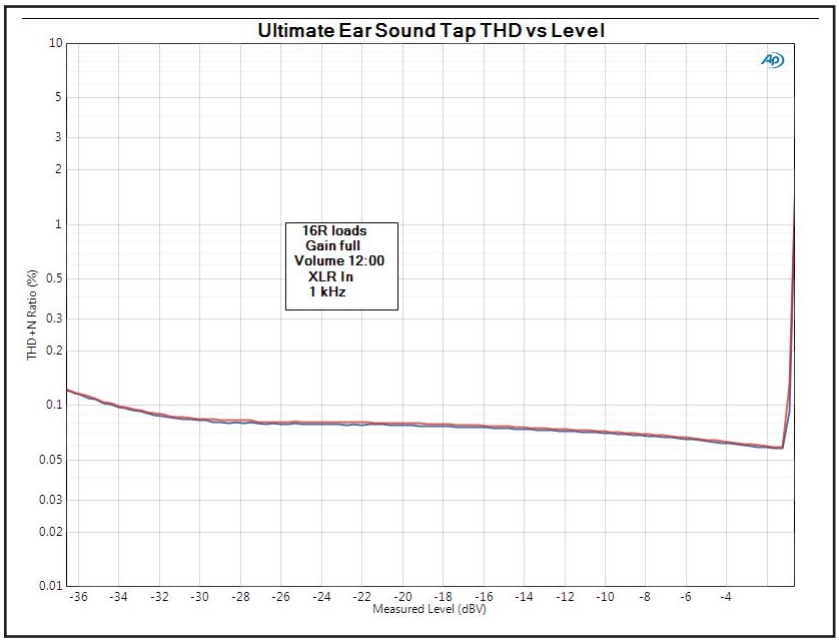


Figure 2: Sound Tap can output nearly 1 V at 1 kHz into 16 Ω before clipping.

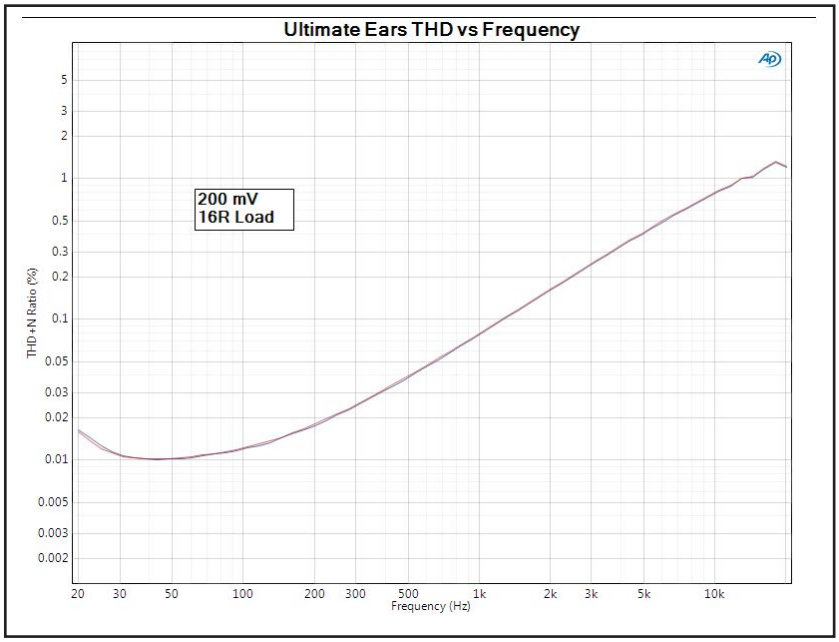


Figure 3: The distortion vs. frequency curves for the Sound Tap show a rise at higher frequencies as the open loop gain drops.

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.

200 mV (which represents a pretty high volume for Ultimate Ears's very sensitive headphones) and first measured the frequency response. **Figure 1** shows the results. There's a small rolloff in the top octave, with response being down by about 0.7 dB at 20 kHz. This would be barely audible to young, fresh ears, but it's doubtful that this would be significant to a working musician.

I was somewhat apprehensive about the use of AD822 op-amps to drive low impedance loads—while their distortion performance is quite good at moderate loads (the distortion spec ratings are taken at 10 kΩ), 16 Ω is really pushing it. So I paid particular attention to two things—clipping level and distortion performance at high frequencies, areas where the heavy loading might cause performance compromises.

Figure 2 shows distortion at 1 kHz as a function of level. It stays reasonably low (0.07%) up until clipping at about 0.95 V, which corresponds to "really, really loud" using the Ultimate Ears headphones. This is much better than I expected. As we went up in frequency, things got a bit uglier. **Figure 3** shows how the distortion at 200 mV varies with frequency—as the op-amps' open loop gain (and hence feedback) rolls off with increasing frequency, distortion rises, reaching 1.2% at 20 kHz, all dominated by third-order harmonics. The AD822 is definitely breathing hard with this load. Is this an audible amount of distortion? Likely not, and for on-stage performance application, it's easily good enough.

The upside of the use of the AD822 is the low current drain. Ultimate Ears rates battery life at 40 hours, and I've run 48 hours on the current set of batteries without the voltage dropping low enough to light the red low-battery LED. In my opinion, this is a smart engineering trade-off—losing power during a show is NOT an option.

Wrap Up

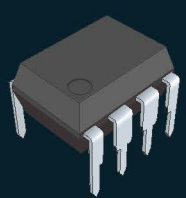
The UE Pro Sound Tap provides an easy and convenient way to connect in-ear monitors or other headphones for on-stage monitoring. It's rugged, versatile, easy-to-hook-up, has long battery life, and as the British expression goes, "Does what it says on the tin." It worked flawlessly for me and, if you're going the IEM route, is well worth the moderate cost.

Resource

Ultimate Ears Stage Products,
http://pro.ultimateears.com/products/custom-monitors/for-stage



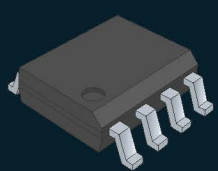
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MUSES

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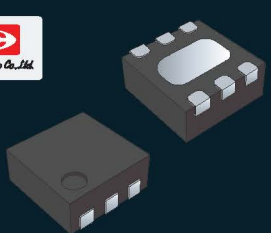


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Design Considerations for the Optimum Digital ANC Headphone

This article discusses the constraints associated with active noise cancelling (ANC) along with design best practices to compensate for them and maximize the cancellation bandwidth while achieving 40 dB noise-cancellation performance.

By
Peter McCutcheon
(United States)

There are two primary forms of active noise cancelling (ANC) technology implemented in commercially available headphones, which are known as feed-forward and feedback. These can be combined to produce a hybrid system. Each form of ANC has constraints that set their performance and bandwidth, which derive from the headphone acoustics, signal processing, and system latencies.

Feed-Forward Noise Cancellation

Feed-forward systems achieve noise cancellation by outputting an anti-noise signal from a headphone driver that has the same amplitude but opposite phase to the ambient noise. This anti-noise signal is an inverted, filtered version of the noise signal detected at the microphone (see **Figure 1**) and combines with the noise signal at the eardrum to substantially reduce the noise level. The filter acts to compensate for differences between the noise-frequency response at the eardrum and at the microphone where it is detected. It also compensates for the fact that the anti-noise signal is shaped by the driver response.

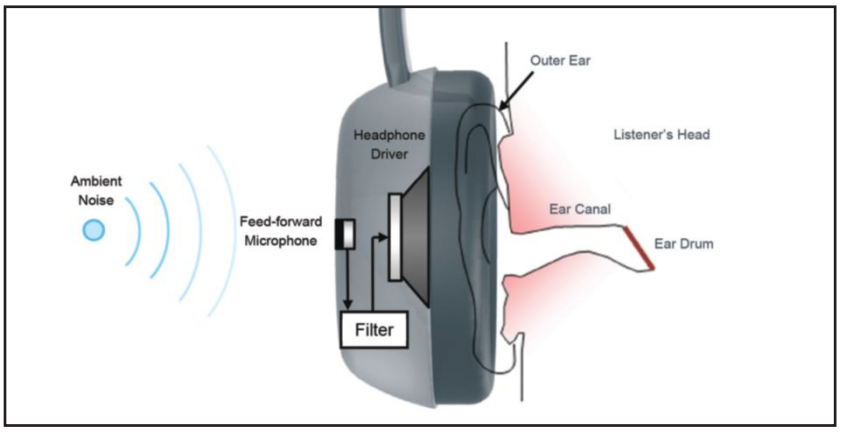


Figure 1: Feed-forward noise-cancelling headphone

While the headphone driver properties limit feed-forward cancellation at low frequencies to about 50 Hz, its limitation at high frequency is typically close to 3 kHz due to latencies in the acoustics and the processing. These latencies cause difficulties in achieving a 180° phase difference between the anti-noise and the noise signals. This is particularly true at high frequencies where the wavelengths of the noise are shorter. **Figure 2** shows the effect of a 20 µs latency on the noise cancellation at two frequencies. At 1,500 Hz, the amplitude of the residual noise is about 0.2 (14 dB ANC), yet this has increased to 0.6 (just 4 dB ANC) at 4,500 Hz even though the latency is constant.

This latency can, in part, be compensated for by detecting the ambient noise in advance of the noise entering the ear. This gives the processor more time to process the signal before outputting the anti-noise signal. However, placing the microphone

a long distance from the ear canal entrance can reduce the noise cancellation for noise sources at different directions (see **Figure 3**).

Figure 3 shows that when a microphone is placed on the outside of a headphone shell (i.e., far from the ear) the time between the noise being detected at the microphone, and entering the ear is different for noise sources at 0° and 90°. This ultimately means the noise-cancellation performance will be different for all directions. This problem can be allayed by controlling the path taken by ambient noise entering the ear, and placing the microphone in close proximity to this path.

One effective method is to place a vent behind the speaker as shown in **Figure 4,** where the dominant path for noise entering the ear is via this vent and through the headphone driver or adjacent vents. In this scenario, the time between the noise detected at the microphone and the noise entering the ear is consistent from all angles and, therefore, the directionality of the feed-forward system is substantially reduced.

At frequencies above about 3 kHz, where the wavelength of the sound is substantially shorter than the ear canal and the headphone cavity dimensions, acoustic modes can occur across air volumes and in the speaker membrane, which are difficult to filter. Also, system latencies limit cancellation in this bandwidth so these frequencies are typically attenuated passively. Passive attenuation is generally increased by making the headphones more closed but this necessitates closing or decreasing the control path described in Figure 4, which will inhibit the performance of feed-forward noise cancellation at higher frequencies. This results in a trade-off between passive and active cancellation in this region.

Interestingly, consumers evaluating noise-cancelling headphones often find it easier to judge the active noise cancellation as the user can instantly enable or disable it. It is more difficult, however, for consumers to judge the passive attenuation, as during the short time required to place the headphones over the ears the user has forgotten exactly what the ambient noise sounds like.

Figure 4 also shows that the headphones must make a consistent seal between the ear cushion and the head for all users for consistent acoustics and noise cancellation performance.

Further, it is advisable that the driver's frequency response and the passive attenuation frequency response should be kept smooth (for instance having no high Q-factor peaks and troughs) such that a simple digital filter can easily compensate for these transfer functions.

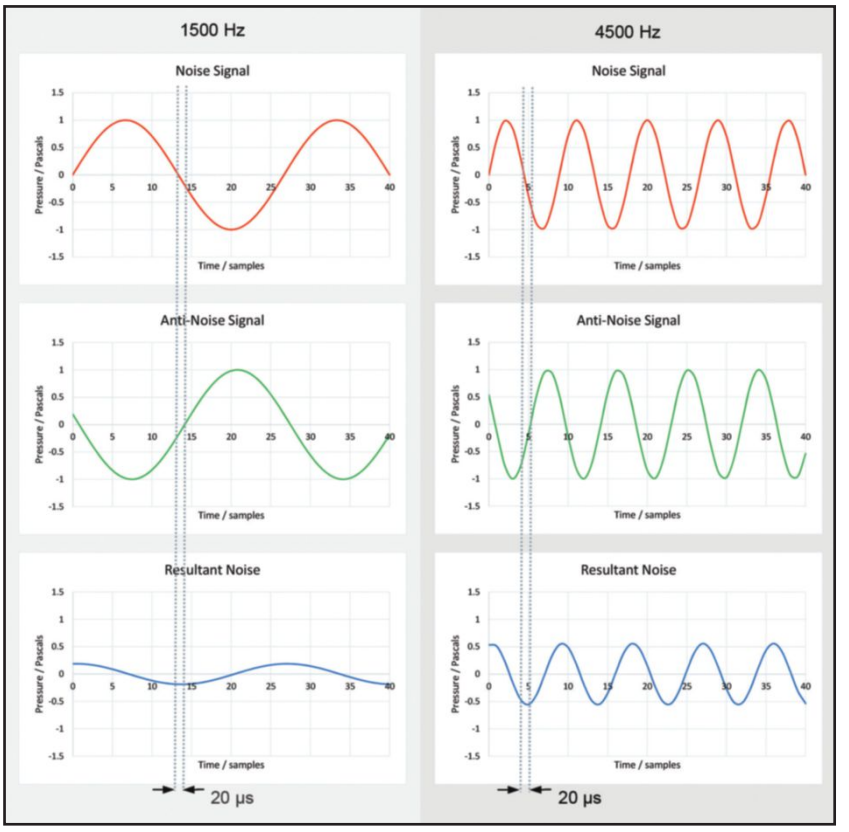


Figure 2: Effect of system latency at two different frequencies

Feedback Noise Cancellation

Feedback noise cancellation headphones (see **Figure 5**) detect the noise in the same air volume as the eardrum. It implements a basic control feedback loop to minimize the noise in this region.

Figure 5 shows the equation used to calculate the noise cancellation from a feedback system. The "loop" is the product of the driver response, the

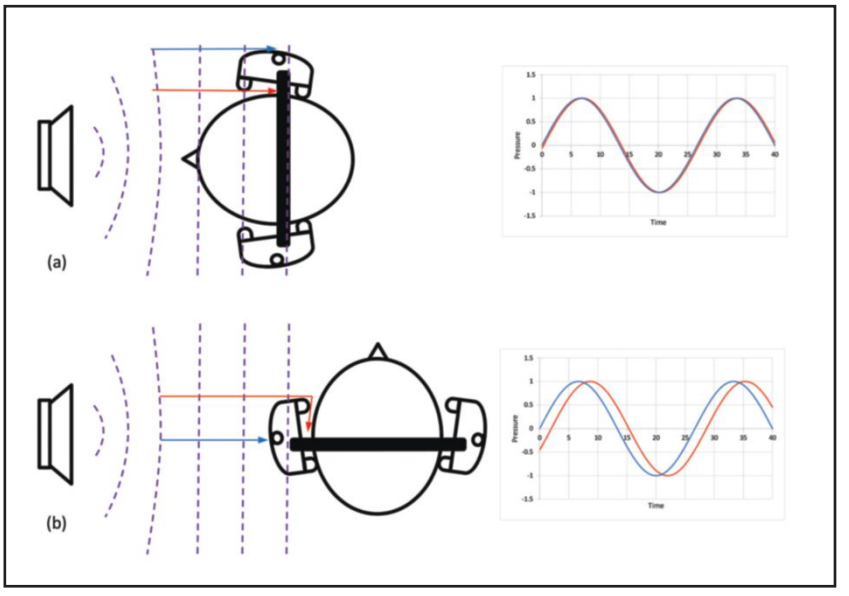


Figure 3: Directionality of feed-forward noise cancellation, (a) noise at 0° to the user and (b) noise at 90° to the user

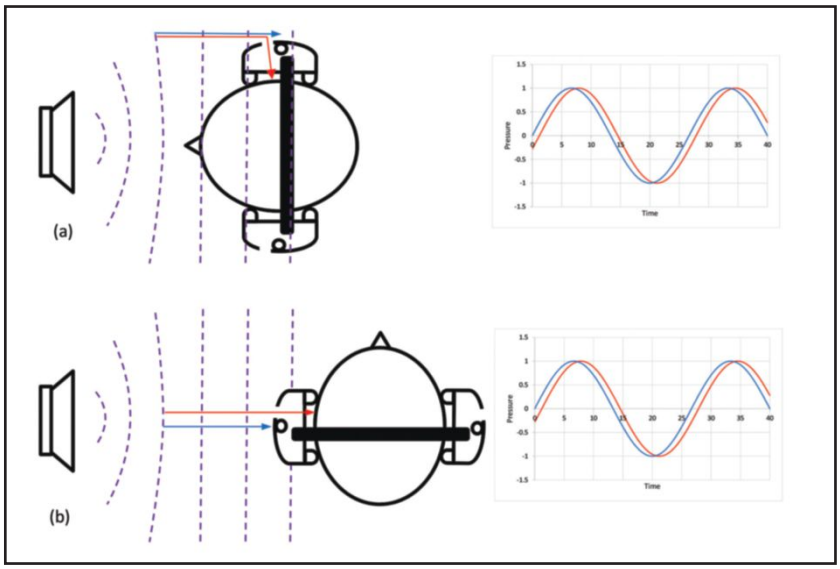


Figure 4: Arranging the microphone close to a point at which noise enters the ear (a) noise at 0° to the user and (b) noise at 90° to the user

microphone response, and the filter. The formula shows that as the gain of the filter (and hence the loop gain) increases, the residual noise signal gets smaller, improving the noise cancellation performance. However, if the phase of the loop nears ±180°, the "loop" signal is effectively inverted and the "+" on the denominator becomes a "-". In this case, the loop gain is highly constrained because when it is increases from 0.0 toward 1.0 the result is amplification and when equal to 1.0 the result is a "division by zero," which signifies instability.

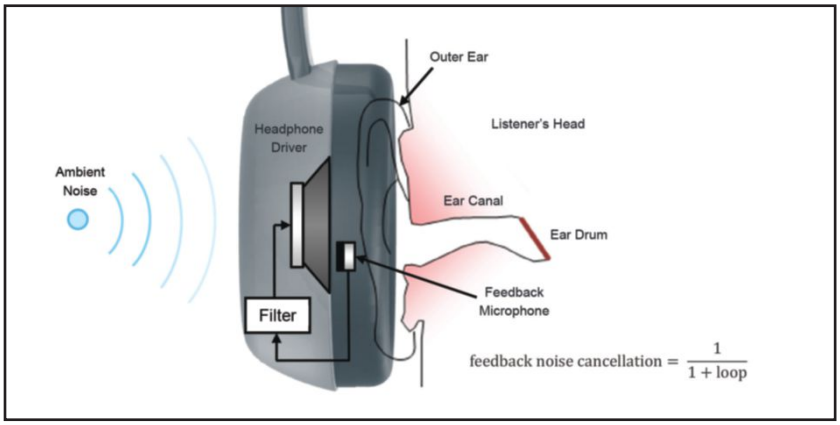


Figure 5: Feedback noise cancelling headphone

About the Author

Peter McCutcheon is an Application Engineer at ams. He began working on noise cancelling systems with the development of digital ambient noise cancellation on mobile handsets in 2007. More recently, he has focused on noise cancelling headphones when taking on the role of Principal Research Engineer for Incus Laboratories, which was recently acquired by ams.

This usually manifests itself as a whistling tone that increases in amplitude—something that must be avoided at all costs.

In practice, the phase of the loop tends to 180° at 10 Hz and to -180° at a few kilohertz. As such, the gain must be substantially below 1.0 at these frequencies, but as large as possible in between. The filter is shaped to achieve this effect, which typically limits the bandwidth of the feedback noise cancellation to between 10 Hz and 1 kHz.

The reason for the loop phase change at high frequency is due to system latencies in the processor, the speaker, and the distance from the driver to the microphone. It follows that reducing any of these (using a lightweight, sensitive driver; placing the microphone close to the driver membrane and minimizing processor latency) can improve the upper bandwidth of the noise cancellation.

Since the feedback microphone is placed in close proximity to the driver, the microphone also detects the music signal playing through the headphones as noise. As a result, the music signal from the speaker is also cancelled and must be electronically boosted to compensate for this.

Digital Signal Processing

The building blocks for a simple digital ambient noise cancellation system are shown in **Figure 6**.

There are multiple benefits to applying the ANC filters in a digital processor:

- Flexibility—The ability to switch filters to change the noise cancellation performance for different environments or an ambient hear-through mode that removes the passive attenuation effects of the headphone. A digital IC can also communicate digitally with complementary ICs such as a Bluetooth communication device.
- Faster development—This typically involves several iterations of acoustic design changes and electronics design changes. A digital system allows for fast changes in filter design so new acoustic designs can be tested instantly, without re-soldering components.
- Improved calibration processes—Due to tolerances in acoustic components, the acoustic transfer functions that influence the noise cancellation filter shapes can differ in production. Each headphone must be calibrated to compensate for these tolerances. This is a costly part of ANC headphone production, largely because the process takes time and often requires a manual operator.
- Smaller footprint—This is because fewer external components are needed.

Shortcomings of a Digital System

The shortcomings of a digital system include higher power consumption and electronic noise.

A digital system has higher latency. Generally the lower the latency the better, but it is difficult to perceptibly tell the difference between an analog system with a negligible latency and a digital system with less than 20 µs latency, as latencies in the acoustics dominate in the band where noise cancellation is active.

As wireless hearables become more popular, power consumption becomes critical. Any digital noise cancellation solution, therefore, must be power efficient. Power consumption is typically poorest in the ADC and the DAC. The digital processor power consumption can be kept to a minimum by maintaining simple processes (e.g., using simple filters and streamlining any other processes), and clocking at as low a rate as possible. While clocking the system faster can dramatically reduce processor latency, it also increases power consumption so a trade-off between low latency and power consumption exists.

Also, it would be counter-productive to make noise cancelling headphones that are electronically noisy. The primary source of electronic noise is typically the microphones. Despite the recent increase in popularity of micro electro-mechanical system (MEMS) microphones, electret condenser microphones (ECM) still outperform MEMS for signal-to-noise ratio (SNR). The leading ECM microphones are specified at 74 dB SNR at 94 dBSPL, which translates to a noise floor of 20 dBSPL. While the noise floor of the microphones is still low, it is advisable to specify microphones with as high an SNR as possible to avoid uncorrelated noise being audible in quiet environments.

When listening to music on digital headphones with noise cancellation disabled, the microphone noise is no longer applicable and the entire digital system must have a low enough noise floor such that no unwanted electronic noise is audible.

A general way to specify an acceptable SNR for the digital system is by defining the loudest sound that is desired to be output and subtract the level of the quietest audible sound from this. The acceptable level of electrical noise in the system is anything that is not audible.

While 0 dBSPL is defined as the lower threshold of human hearing at 1 kHz, it is unlikely that you will find yourself in an environment quieter than 25 dBSPL (the sound of breathing at 1 m distance). The peak level that headphones can output is about 125 dBSPL at some frequencies, although recent standards (EN 50322 and IEC 600065:2014) state that portable media players must limit music playback to a maximum of 100 dBA.

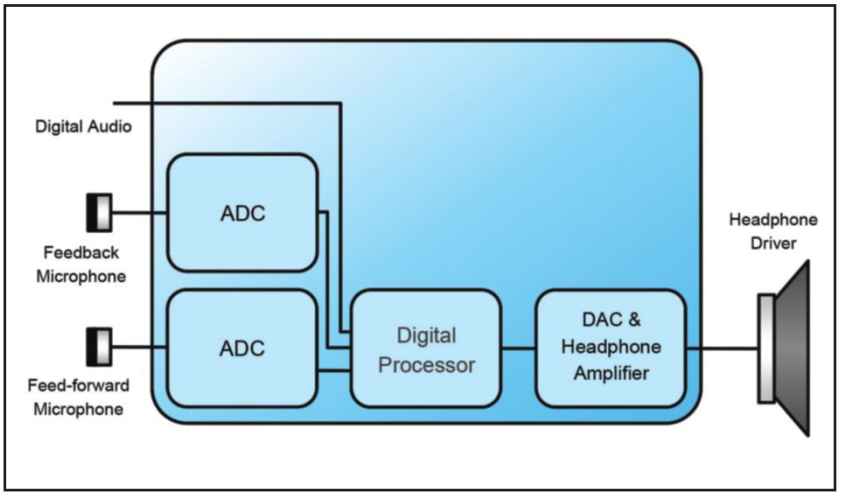


Figure 6: Basic blocks to digital noise cancellation IC

Thus, it is sensible to specify a DAC that can achieve at least 100 dB SNR (125 dBSPL to 25 dBSPL) and ensure noise in the digital domain is below this. Although this might not seem to be a difficult target for modern digital processors, floating point arithmetic is considered too power hungry and, as such, fixed point is used. This must maintain long word lengths so that quantization noise is below that of the ADC and DAC.

It is also necessary to choose a speaker with a good sensitivity and low distortion. Any distortion from the speaker will result in a distorted anti-noise signal and less overall noise cancellation.

Summary of Design Considerations

When it comes to ANC, there are several design considerations to take into account:

- Implement a digital noise cancellation architecture to provide flexibility.
- Minimize acoustic latencies throughout the system and use an IC with less than 20 µs latency to optimize the noise cancellation bandwidth.
- Create a controlled path to channel noise into the ear for effective feed-forward noise cancellation.
- Design the mechanics such that a consistent fit is possible for all headphone users.
- Decide on more open design with less passive and more active attenuation, or a more closed design with more passive and less active attenuation.
- Tune air volumes, vents and vent damping to create a smooth driver response and passive attenuation response.
- Minimize sources of electronic noise by specifying microphones with good SNR, and ensuring any noise from the digital domain and DAC is not audible.

Measurement of Devices That Use AGC

The test and measurement of audio devices that feature built-in automatic gain control (AGC) requires special care and attention. Adam Liberman explains why, and offers some advice about optimal test settings.

By
Adam Liberman
(United States)

Most audio test and measurement is done by running test tones of known frequency and amplitude through a device under test (or DUT) and looking at how the tones are altered at the device's output across the audible frequency spectrum. Usually the gain of a device is relatively consistent and does not change based on the level and duration of the test tones. But, how do you test a device where the gain is not consistent?

This is the problem at the heart of testing any audio device that has AGC, which includes devices such as amplifiers, recorders, signal processors, codecs, public address (PA) systems, audio workstations, and hearing aids. AGC is typically used to limit the dynamic range of audio signals in a variety of applications—in electronics, to prevent overload due to clipping; in broadcast, to prevent over-modulation; in music production, to facilitate balancing the levels of different tracks, or to increase the overall perception of volume; in recorders, to eliminate the need to manually set levels; in PA systems, to help maximize intelligibility in noisy environments; and in hearing aids, to put sounds in a range that is intelligible and comfortable for the wearer. AGC may also be used to expand dynamic range, often to increase sonic impact or to remove background noise.

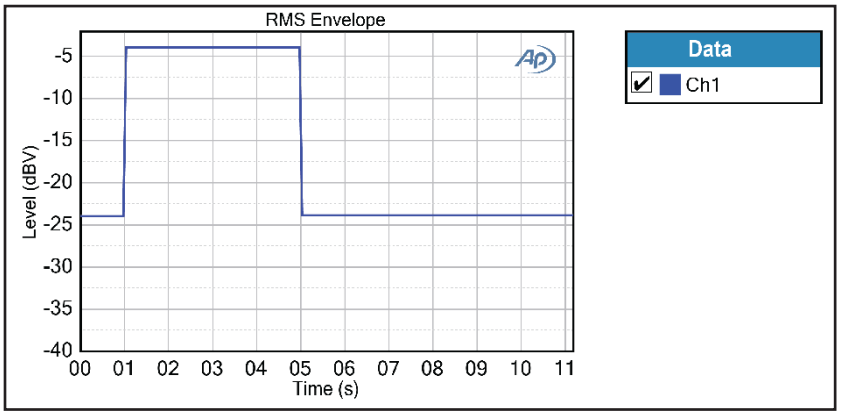


Figure 1: Step function, source

Practical devices might combine multiple AGC functions at different loudness points. For example, a hearing aid may have expansion to cut out low background noise, compression to maximize intelligibility, and limiting to prevent excessively loud sounds from causing discomfort and distortion.

AGC is referred to by many different names, depending on the exact application, device, and convention in the particular industry that is manufacturing the device: Common variations of AGC include compressor, limiter, automatic level control (ALC), automatic volume control (AVC), leveler, expander, or gate.

The presence of AGC means that additional tests need to be included in the test routine to characterize the AGC action and how it affects the signal. This article explains how to go about such testing.

The audio analyzer that is used must be able to generate a step function of defined duration and level, and change the level instantly without instability or artifacts. To analyze the attack and

release times, it must be capable of measuring the RMS level over very small time increments. Alternatively, the level may be measured by examining a wave file recording of the acquisition. For measuring frequency response during compression, the instrument needs to include multitone analysis. An Audio Precision APx555 audio analyzer has been used to create the diagrams for this article.

Operation and User Parameters

The effect of AGC may clearly be seen and measured by applying a step function and creating a graph showing level vs. time. **Figure 1** shows the source signal produced by the generator of an audio analyzer. **Figure 2** shows the analyzed signal after it has passed through a compressor.

All AGCs are variations of the same basic topology, shown in **Figure 3**. Audio enters an input amplifier and passes to a variable gain amplifier (VGA), and then goes to an output amplifier. The input amp also sends audio to a sidechain: an optional filter (usually a high-pass filter) and a level detector/control circuit, which controls the gain of the VGA. More elaborate variations on this basic design include an independent sidechain input, additional filtering, program-dependent control, and multiple thresholds. Some older analog compressors use a feedback instead of feedforward design, where the sidechain input is taken after the VGA instead of before it.

Dynamic and Static Properties

AGC settings or properties can be divided into two categories: dynamic properties, which are time-related, and static properties, which are not time-related. All AGCs have these properties, although not all devices allow them to be adjusted. The dynamic properties of attack and release time are defined as follows:

- Attack time: The time interval between onset of an increase in level and the point when the output level has stabilized to near its final level.
- Release (recovery) time: The time interval between onset of a reduction in level and the point when the output level has stabilized to near its final level.

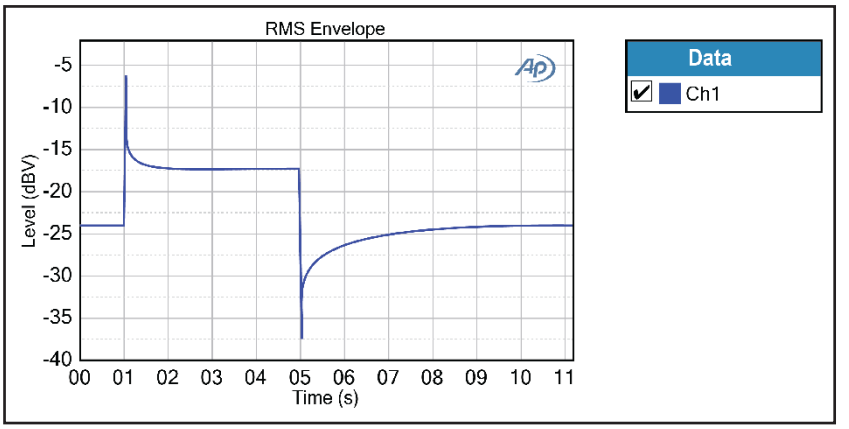


Figure 2: Step function, after passing through a compressor

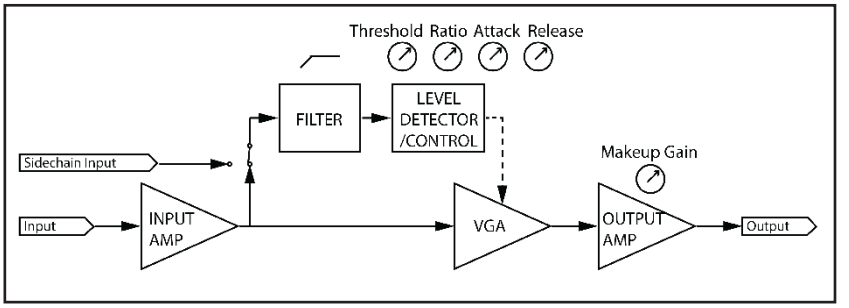


Figure 3: Compressor block diagram

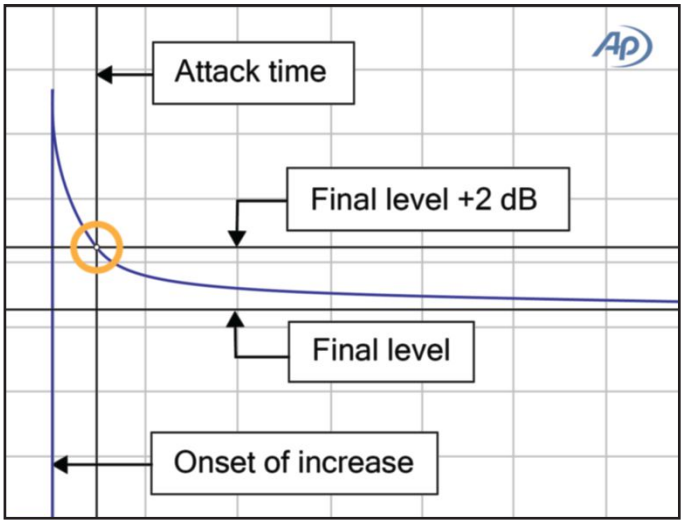


Figure 4: Attack time calculation (using 2 dB stabilization)

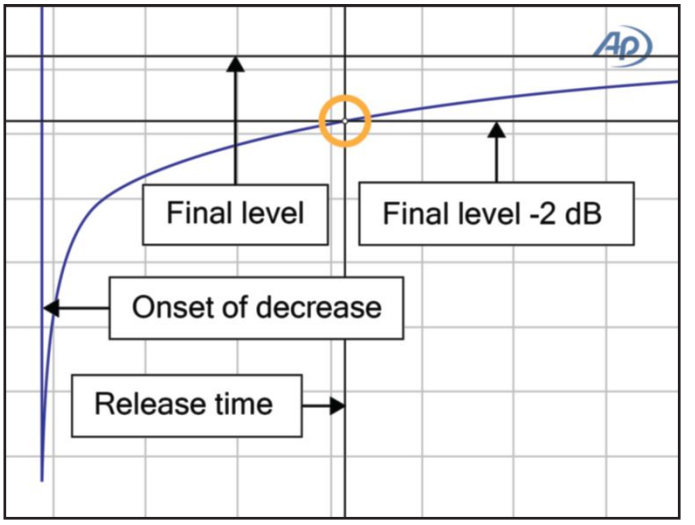


Figure 5: Release time calculation (using 2 dB stabilization)

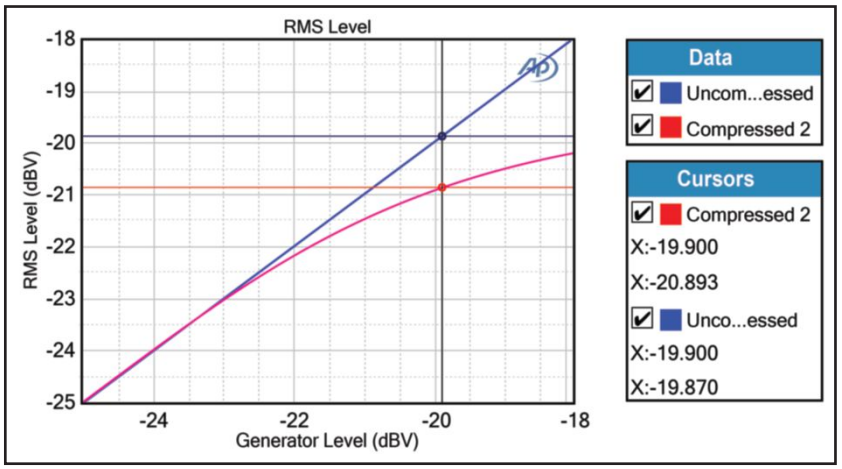


Figure 6: Cursors at point where audio is compressed by 1 dB

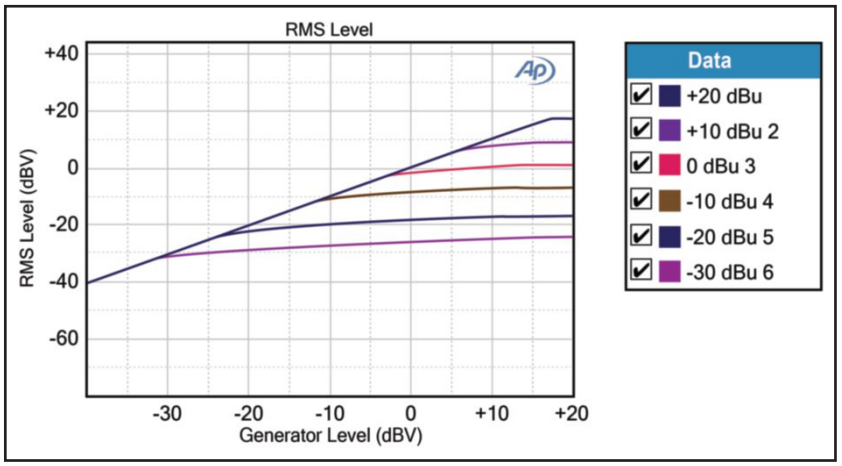


Figure 7: Generator level vs. Output level for various thresholds

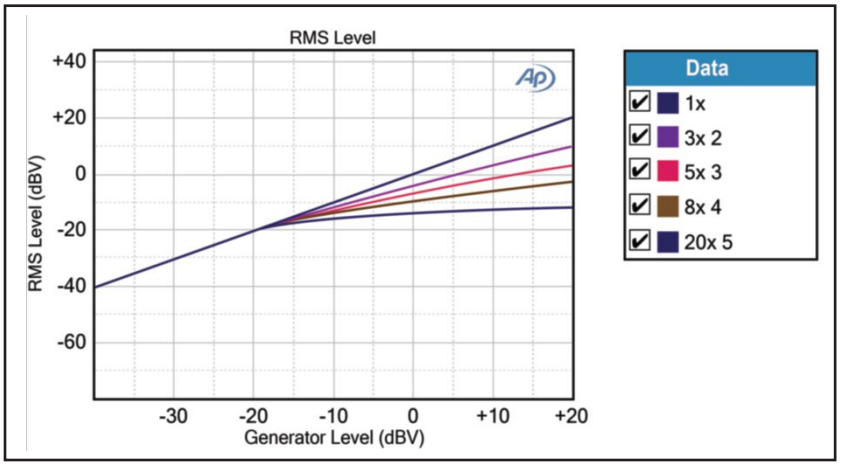


Figure 8: Generator level vs. Output level for various ratios

The meaning of "near" used above is defined by the applicable standard (2 to 4 dB). **Figure 4** and **Figure 5** show zoomed-in portions of the RMS Envelope from Figure 2, illustrating the time calculation points.

To test the dynamic operation of an AGC, we provide a stepped level function to its input, where the high level causes compression to engage and the low level does not (see **Figure 6**).

The static properties of threshold, ratio, and knee are defined as follows:

- Threshold: The level at which the AGC begins to alter gain (see **Figure 7**). The meaning of "begins to" here is defined by the applicable standard (1 or 2 dB).
- Ratio: Change in input over change in output, expressed in decibels, for an increase in level over the threshold (see **Figure 8**).
- Knee: The region on the curve where the input-output function changes from linear to compressed. A "hard knee" denotes a sudden change, while a "soft knee" denotes a more gradual transition (see **Figure 9**).

Although some devices maintain a constant ratio once outside of the knee region, others do not. This is not an indication of quality, as this characteristic may be desirable in some applications—it simply makes it difficult to state a single figure for the ratio.

Both dynamic and static AGC properties can be measured at different frequencies to see if the AGC action is frequency-dependent.

Standards

The international standards that apply to AGC devices are summarized in **Table 1**. They do not cover dynamic range expansion, but can be used to test such devices with similar results. All the standards utilize a low-high-low step function, similar to the one shown in Figure 1.

General-purpose AGC devices are covered by the International Electrotechnical Commission (IEC) standard 60268-8:1973. It is written to be applied to a broad variety of devices, and allows quite a bit of user discretion in choosing test settings. It should be noted that since many devices do not specify the standards and conditions used to derive their attack and release time specifications, the measurement results that you get for a DUT may differ considerably from those in its published specs.

Hearing aids are covered by three active standards and one inactive standard. The inactive standard, IEC 60118-2:1983, is included here because it may still be used by some companies

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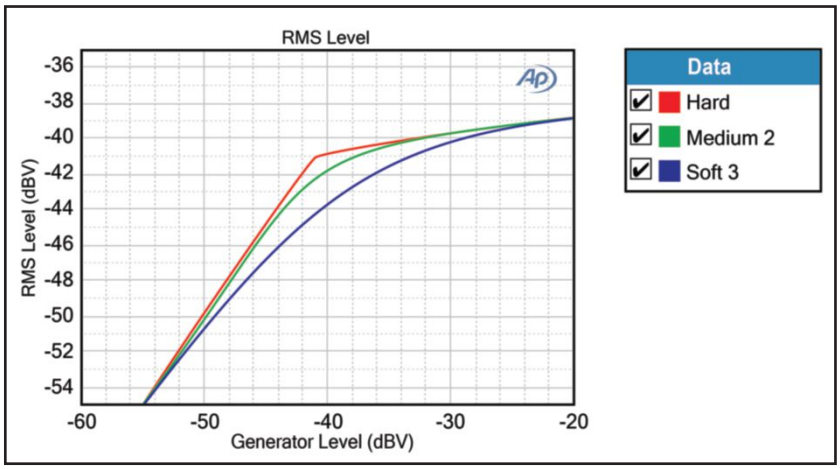


Figure 9: Generator level vs. Output level for hard, medium, and soft knee settings

until their procedures have been updated. The three remaining standards are almost identical except for variations in their allowed test frequencies.

Table 1 provides a summary of the settings for testing AGCs using each of the standards. For hearing aids, the standards also specify how the equipment should be positioned and how references should be set. It is essential to read and follow those guidelines before attempting to make measurements.

Standards Parameters

The parameters in Table 1 are further explained below:

- Frequency—Required and/or optional test frequencies
- Step size—The difference between the high and low level steps in the test signal
- Input level (high and low): Applies only to hearing aids. Defined as specific acoustic levels in dBSPL (see the "Acoustic Calibration" information).
- Above/below threshold—Applies only to general devices. Defines the high and low levels relative to the threshold (added together they equal the step size). IEC 60268-8 specifies the high level as 6 dB above the threshold for a 10 dB step size.
- Threshold reduction—The amount of gain reduction at the threshold.
- Attack/release settling—The settling tolerance, which is the point that is "close enough" to the final level, where the attack and release times are measured.

Generator Settings for IEC60268-8 Measurements

Unlike the hearing aid standards, which have predefined acoustic test levels, IEC 60268-8 measurements require you to determine the

		General	Hearing Aids			
Parameter	**Standard/ Unit**	**IEC 60268-8 (1973)**	**IEC 60118-2[2] (1983)**	**IEC 60118-7 (2005)**	**IEC 60118-0 (2015)**	**ANSI S3.22 (2009)**
Frequency	Hertz (Hz)	10 kHz[1]	1.6 kHz or 2.5 kHz	2 kHz required; 250 Hz, 500 Hz, 1 kHz, or 4 kHz, optional	2 kHz required; 200 Hz to 8 kHz, optional	Any of of 250 Hz, 500 Hz, 1 kHz, 2 kHz, or 4 kHz
Step size	decibels (dB)	10 dB[1]	25 dB/40 dB[3]	35 dB	35 dB	35 dB
Input level (low)	dBSPL	ns	55 dBSPL/60 dBSPL[3]	55 dBSPL	55 dBSPL	55 dBSPL
Input level (high)	dBSPL	ns	80 dBSPL/100 dBSPL[3]	90 dBSPL	90 dBSPL	90 dBSPL
Above threshold	decibels (dB)	6[1]	ns	ns	ns	ns
Below threshold	decibels (dB)	4[1]	ns	ns	ns	ns
Threshold reduction	decibels (dB)	1	2	ns	ns	ns
Attack settling	decibels (dB)	2	2	3	3	3
Release settling	decibels (dB)	2	2	4	4	4

Table Key: 1. Suggested. 2. Cancelled and replaced by IEC 60118-0 in 2015. 3. Dynamic range setting: normal speech/high level. ns= not specified.

Table 1: Comparison of AGC Standards

optimal generator level yourself. In order to set the generator level, it is first necessary to determine or set the compression threshold level. This may be done in one of two ways:

- For devices with an adjustable threshold, adjust the device as follows. Set the threshold control on the DUT to its highest signal level, to assure that compression is not activated (rotary controls will be either fully clockwise or fully counter-clockwise). Turn on your tone generator and set it to the desired threshold level. Observe the gain at the output of the DUT and reduce the DUT's threshold level control until compression starts to activate and the gain drops by 1 or 2 dB (depending on the standard).
- For devices with a fixed threshold, adjust the generator as follows. Turn on your tone generator at a low level. Observe the gain at the output of the DUT and increase the generator level until the gain drops by 1 or 2 dB (depending on the standard you're testing for). The generator is now set to the threshold level.

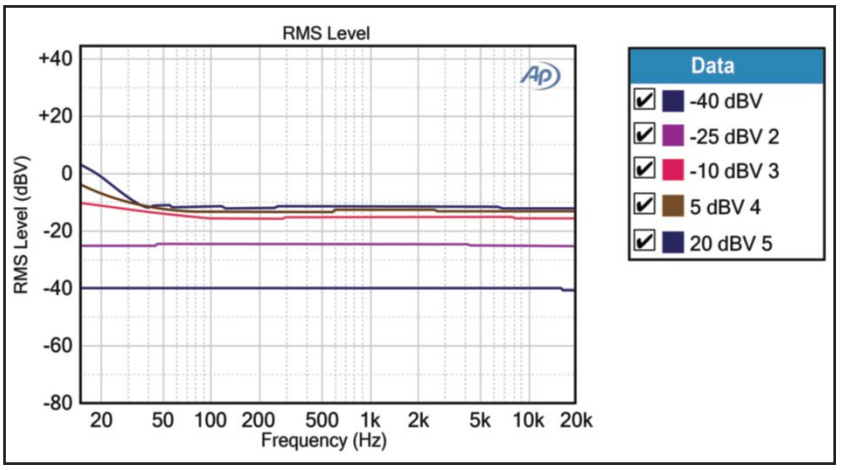


Figure 10: Frequency response using the Continuous Sweep measurement

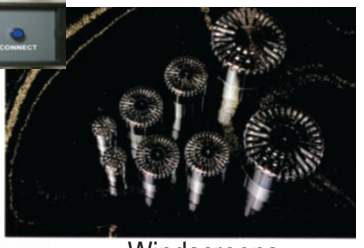
After finding the threshold, you will need to add the desired range above threshold to determine the high level setting for the generator. For example, when testing using the default IEC 60268-8 10 dB step size, if the threshold is set at -10 dBV, you would set the generator high level to -4 dBV. This will put the high level 6 dB above threshold, and the low level 4 dB below threshold. For devices with a soft knee characteristic, you might need to use a larger step size, with the high and low levels further away from the threshold to make sure that the DUT goes fully into compression.

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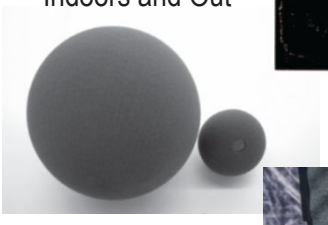
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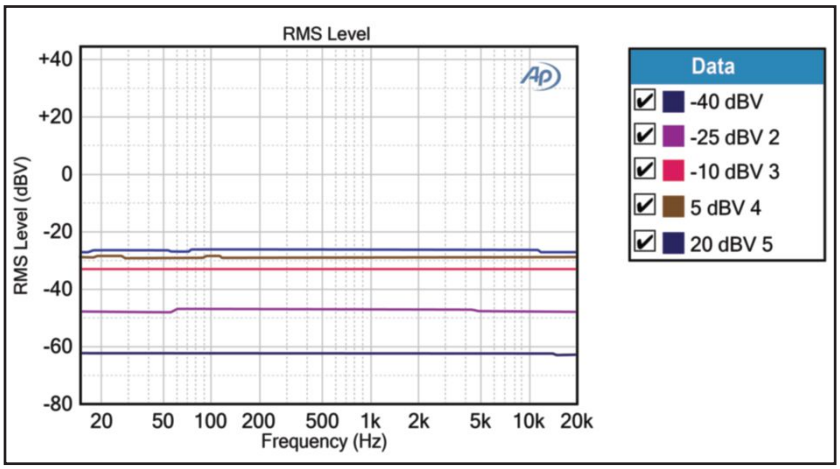


Figure 11: Frequency response using Multitone Analyzer measurement

Use the 10 kHz test frequency recommended by the IEC standard, or an alternative frequency such as 1 kHz. For band-limited compressors, use a relevant frequency within the band.

Creating the Step Function

The generator step function test signal consists of three sections: Low (1), High, and Low (2). The length for each section must be predetermined to produce valid results.

The length of the first section, Low (1), is not critical. A default of 1 second is normally adequate.

The length of the High section must be long enough for the compression action to completely stabilize at its final level. It may be necessary to run some trials and to incrementally increase the length until the attack time result is no longer affected. On some program-dependent compressors, the duration of the High section affects the release time.

The length of the Low (2) section must be long enough for the compressor to release completely and stabilize at its final uncompressed level. Like the High section, it may be necessary to run some trials to adjust this correctly.

Because they usually have fast attack and release times, sections of 1-second duration are usually long enough for hearing aids. For other devices, much longer sections may be necessary.

Making the Attack and Release Measurements

After setting up the generator and the DUT as described above, apply the generator step function to the DUT input while observing its output vs. time. Use the audio analyzer or examine a wave file of the acquisition to find the attack and release measurement points. These are the points where the output level has stabilized to near its final level, where "near" is the settling tolerance according to the applicable standard. See Figure 4 and Figure 5 for graphical representations of how to determine the measurement points.

Other Audio Performance Measurements

Besides dynamic and static AGC properties, we can also make many other audio measurements on the signal passing through the DUT (e.g., frequency response, distortion, noise, maximum output, and crosstalk, both with and without gain reduction). Most measurements can be performed in the same way they are for devices that do not incorporate AGC. However, when measuring frequency response in the compression region above the threshold, there are some precautions that need to be observed.

Frequency Response Above Threshold

Some analyzers use a log-sine chirp signal to

Acoustic Calibration

Hearing aids are normally tested in a small acoustic isolation box or chamber. Levels are specified in dBSPL (sound pressure level). Therefore, it is necessary to set up the audio analyzer references so that voltage levels are translated into acoustic sound pressure levels. This is done as follows:

- Connect a measurement microphone to the audio analyzer and place a 94 dBSPL sound level calibrator over the microphone.
- Measure the voltage on the analyzer input and make this level equivalent to 94 dBSPL.
- Place the microphone in the same location in the chamber that the hearing aid will be placed (reference position).
- Connect the audio generator to the speaker in the isolation chamber and adjust the generator level until the analyzer measures 94 dBSPL from the microphone.
- Make the current generator output level equivalent to 94 dBSPL.

More information on microphone calibration and the required placement of microphones for standards-compliant testing of hearing aids can be found in the details of the relevant IEC standards themselves (see www.iec.ch).

make frequency response measurements. This type of signal is not suitable for above-threshold testing for two reasons. One is that the amount of time since the generator has been turned on increases as the sweep progresses, so signal time and frequency are not independent of each other. The other is that the frequency of the signal exciting the compressor changes, and this will cause errors if the action of the compressor under test is frequency dependent. Running a stepped frequency sweep, with enough delay at each step, resolves the first but not the second issue (see **Figure 10**). These limitations do not apply when using a separate sidechain input to control the compression action, but many compressors do not have this feature.

However, if your analyzer offers a multitone measurement, where the generator can output many frequencies at the same time, both of the above issues can be resolved (see **Figure 11**). After turning on the generator stimulus signal, the analysis must be delayed until the DUT is fully in compression and has stabilized at its steady-state level. This time delay will need to be significantly longer than the measured attack time.

Frequency Response of the Compression Sidechain

Many compressors have a reduction in threshold sensitivity at the frequency extremes. This prevents low-frequency air conditioning rumble or high-frequency oscillations from triggering gain reduction. Graphing the frequency response of the compression sidechain can be done in the following manner:

- Send a 1 kHz audio signal at a standard level through the DUT, with no limiting applied. Adjust the DUT's gain for unity.
- Adjust the compression threshold on the DUT until the gain is reduced by 1 dB.
- Run a stepped frequency sweep, regulating the generator level at each point for a measured gain reduction of 1 dB.
- Graph the generator level vs. frequency, and normalize the results at 1 kHz.

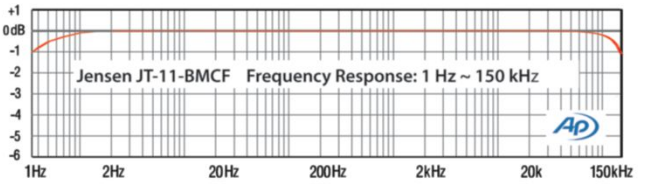
If there is any roll-off in the main signal path, then this will skew the results. This can be alleviated a couple of ways. If the DUT is a two-channel unit and the roll-off in both channels is the same, then instead of regulating to maintain the gain, you can regulate to maintain a constant RMS level ratio between the two channels. Alternatively, if the compressor has a separate sidechain input, you can sweep the sidechain signal while maintaining a fixed 1 kHz tone through the audio signal path.

About the Author

Adam Liberman brings extensive audio expertise to his role of Technical Support Engineer at Audio Precision. His broad range of experience includes film and TV production and post-production sound; theatrical sound design; radio production and engineering; computer audio testing and reviewing; music and nature sound recording; and test, repair, and modification of pro audio and film editing equipment.



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Build Your Own Oscillator

This is a 1 kHz Device with Less than 2 PPM Distortion

Sound reproduction equipment keeps improving. Distortion of less than 0.01% has become commonplace. To test equipment with distortion that low, you need a really low distortion sine wave oscillator. Commercial equipment from companies such as Audio Precision and Stanford Research will do it, but they also cost thousands of dollars. This article describes a 1 kHz oscillator that you can build for less than $100.



Photo 1: AkitikA's oscillator generates a low distortion 1 kHz sine wave that's handy for signal tracing and distortion testing.

By
Dan Joffe
(United States)

You can produce great sound using this kit from AkitikA. And, AkitikA's improved oscillator kit enables you to build your own oscillator with state-of-the-art distortion performance *(see* **Photo 1**).

When we talk about distortion this low, we need to be a little bit careful. If our goal is an ideal sine wave, there are two main forms of corruption: harmonics and noise. When we speak of distortion, we'll refer to harmonics. Harmonics are components of the output at integer multiples of the fundamental frequency. In a 1 kHz oscillator, the harmonics are located at frequencies of 2 kHz, 3 kHz, 4 kHz, 5 kHz, and so on. In addition, there will be noise across all frequencies. To be useful, the oscillator's noise should also be quite low, below the level of any expected harmonics. To really understand what's going on, we should report on both the harmonics and the noise.

Classically, people have spoken about distortion measured in percent. When the distortion gets really low, expressing it in percent makes for so many zeros that it's hard to keep track of things. In that case, it's usually easier to specify the strength of the harmonics in parts per million (PPM) or in decibels relative to the fundamental. **Table 1** expresses the distortion in all three ways.

The Balancing Act

A sine wave oscillator is made by wrapping frequency selective positive feedback around an amplifier. If there's not enough feedback, the amplifier won't oscillate. If there's too much feedback, the resulting oscillator will have too much distortion. To achieve low distortion, control circuits are used to get the amount of feedback just right.

Some oscillators use nonlinear limiting, like a pair of back-to-back diodes, to limit the amount of positive feedback, and hence the amplitude of the oscillation. If such circuits have low distortion, it's owing to the frequency selectivity of the circuits between the nonlinear limiting point and the output. The distortion can also be kept low if the positive feedback is critically tweaked to be just enough to make the oscillator oscillate. That's typically not a satisfactory answer, as the oscillator will be tweaky, needing constant adjustment to keep the output level steady.

Other oscillators use control loops where the output level is measured. A control loop adjusts the amount of positive feedback to create a steady output level. This can work well, but adds significant complexity. In addition, the level detector is typically done by some kind of rectification. Rectification

generates harmonics that have to be dealt with quite carefully, or they will show up in the output.

Probably the most elegant way to perform amplitude stabilization is the one that Hewlett and Packard first used in a Palo Alto, CA, garage so many years ago. They used an incandescent light bulb to set the amount of positive feedback in the oscillator that started their company.

Here's how the light bulb stabilization works. The more power you dissipate in a light bulb, the hotter the filament gets. The hotter the filament, the higher its resistance. So if we put a light bulb in the oscillator's positive feedback loop, before the oscillator is powered, the light bulb starts out in a cold, low resistance state. This ensures that there's enough positive feedback to start the oscillator. As the signal level grows, more power is dissipated in the light bulb, raising its resistance, and decreasing the amount of positive feedback. The system reaches equilibrium when the output signal makes the lamp's resistance provide just enough feedback to maintain the equilibrium level.

This method of balancing the output level is much more linear than diode clippers. Although the power dissipated in the filament varies throughout the cycle, its temperature and resistance don't change very much over the 1 ms period of a cycle at 1 kHz. Thus, the resistance is constant, and the balancing means doesn't create nonlinear distortion.

Wien Bridge Oscillator

This article is about an oscillator with lower distortion than a Wien Bridge Oscillator. Still, it's a point of reference for many, and so we'll take a minute to consider it.

Figure 1 shows a 1 kHz Wien Bridge oscillator as set up for analysis in LTSpice. R2, R3, C1, and C2 form the frequency selective positive feedback. R1 and R4 set the amount of negative feedback. In a practical oscillator, R4 would be a lamp. Before the oscillator was powered, the lamp (R4) would be cold and have low resistance, decreasing the negative feedback, and increasing the gain to a degree that oscillation starts. As the output level increases, R4's resistance increases, increasing the amount of negative feedback, and stabilizing the output level.

If you'd like to do this simulation in LTSpice, you'll find the following important things to note. We've set the positive feedback and negative feedback to the ideal amounts. However, without the initial conditions statement (.ic), the oscillator won't start. In a real oscillator, the ever-present thermal noise gets regenerated to start oscillation. In a simulation, there is no noise, so we add an initial condition to the top node of C2, labeled Vinp. This reliably starts the oscillator in the simulation.

If you run the simulation for 1 second, you'll notice a slight decay in the output level over the course of the 1 second. That's because this simulation doesn't include any form of positive feedback control, and hence the output level isn't steady.

Specifying Distortion with Respect to the Fundamental		
Percentage	**Parts Per Million**	**Decibels**
1%	10,000 PPM	-40 dB
0.1%	1,000 PPM	-60 dB
0.01%	100 PPM	-80 dB
0.001%	10 PPM	-100 dB
0.0001%	1 PPM	-120 dB

Table 1: Distortion equivalence is shown in percent, parts per million, and decibels.

What's Wrong with the Wien Bridge

The Wien Bridge is a pretty good oscillator, but it does have some problems. First, the frequency selective network isn't particularly frequency selective, so if distortion is generated, it isn't well filtered. Second, one-third of the output voltage appears as common mode voltage at the op-amps inputs. For any reasonable output level (e.g., enough to use light bulb level control), this guarantees some common mode distortion. It won't be much, but remember we're shooting for the neighborhood of 1 PPM.

New Oscillator Objectives and Circuit Description

I took simplicity as a virtue, and thus limited myself to just two op-amps. By carefully applying more op-amps, we might do better, but for now we'll live with the minor consequences of this bit of frugality. Ideally, I was shooting for 1 PPM or less. We've come close to that target, where the distortion is reliably less than 2 PPM. To eliminate problems of ground loops and related noise, we've opted for powering from a pair of 9 V batteries. To get to these distortion levels, we need really good op-amps. The TI LME49720 op-amps fit the bill nicely.

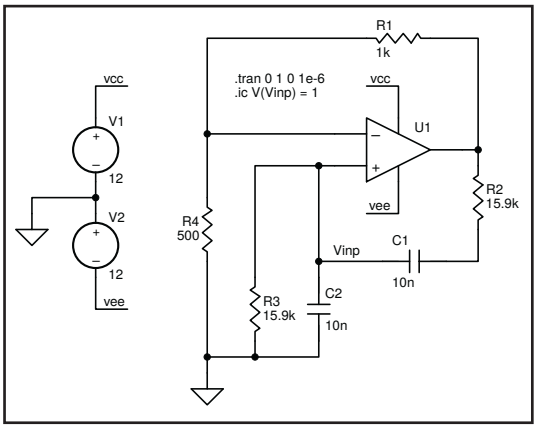


Figure 1: This 1 kHz Wien Bridge oscillator is set up for analysis in LTSpice.

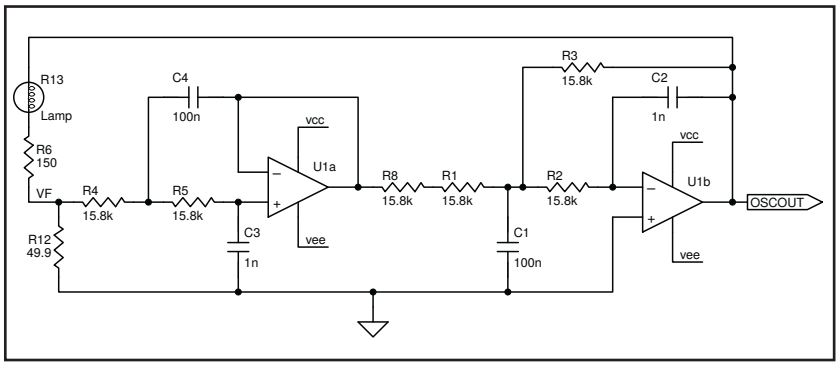


Figure 2: The AkitikA oscillator's new topology takes advantage of the second op-amp to add more frequency selectivity between the input and output.

The new topology takes advantage of the second op-amp to add more frequency selectivity between the input and output. In addition, we've carefully structured the signal levels and topologies to minimize the amount of distortion-producing common mode signal that appears in the oscillator. The result is shown in the schematic (see **Figure 2**).

U1a, R4, R5, C3, and C4 form a Sallen-Key low-pass filter with a Q of 5 and a center frequency of 1007.3 Hz. Its peak gain is 5 at 1007.3 Hz. U1b, R1, R2, R3, R8, C1, and C2 form a multiple feedback low-pass filter with a Q of 4 and a center frequency of 1007.3 Hz. It has a peak gain of -2. The overall gain at 1007.3 Hz is 10.

The combination of inverting and non-inverting topologies should make the overall gain negative, but each stage has an additional phase change of -90° at 1007.3 Hz. Therefore at the oscillation frequency of 1007.3 Hz, there's an additional phase shift of 180°. That makes the input and output in phase, giving us positive feedback and oscillation. At DC, we have overall inversion around the loop, so the feedback at DC just increases the DC stability of the oscillator.

Given the gain of 10 at the oscillation frequency, the input signal level at the positive input of U1 is about one-tenth of the oscillator output level. That keeps the common mode distortion small. U1b has no common mode distortion since its non-inverting input is grounded. The result is that we can potentially have very low distortion.

The only thing we have to talk about now is the feedback network, R13, R6, and R12. VF/OSCOUT should be right about one-tenth to compensate for the gain of 10 as the signal goes from VF to OSCOUT. You can show that the Lamp resistance should be around 299 Ω. That turns out to be a comfortable resistance for the lamp when the oscillator output voltage is around 1.5 V.

If you need different output levels, you could vary R6. For example, you could replace R6 with a short. However, you'd find that the lamp's resistance would have to go up to 450 Ω. That would require a much larger output swing, and that large a swing would probably increase the distortion a bit.

The capacitors should be film capacitors to keep the distortion low. The op-amps are LME49720s, also well known for their low distortion. Something like the classic 5532 is just not clean enough for this application.

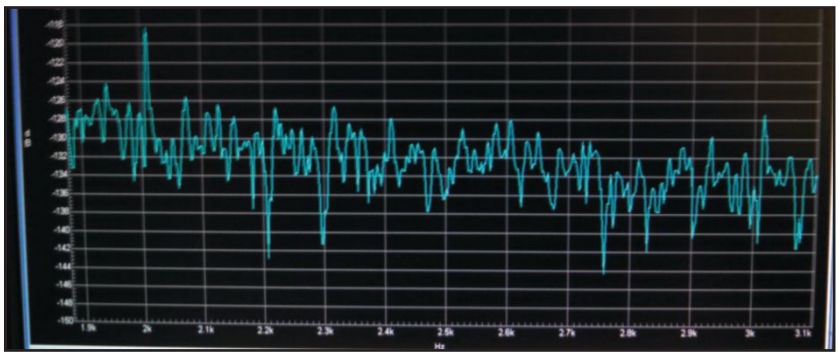


Figure 3: Once built, the total harmonic distortion (THD) residual was measured by an Audio Precision set for 100 kΩ input impedance.

Building the Oscillator

If you're very careful, you can probably build the oscillator on a perf-board with point-to-point wiring. You'll have to be careful to have 0.1 µF bypass capacitors close to the dual op-amp. It's always a good idea to have generous bypass capacitors across the batteries themselves to keep the impedance low.

A complete kit of parts and a PCB are available from the AkitikA website (www.akitika.com). The layout is careful in terms of where the nonlinear power supply currents go, and it has a ground-plane. When you're shooting for 1 PPM distortion, everything matters. In the kit, we've added a power switch, LED power indicator, and an output level control driving two outputs on RCA jacks.

Figure 3 shows the total harmonic distortion (THD) residual measured by an Audio Precision set for 100 kΩ input impedance. The second harmonic is 118 dB below the fundamental, and the third harmonic is down about 128 dB.

Summary

Using low distortion op-amps and capacitors, light bulb limiting, and careful layout, this oscillator design can generate a 1 kHz sine wave having less than 2 PPM of harmonic distortion.

About the Author

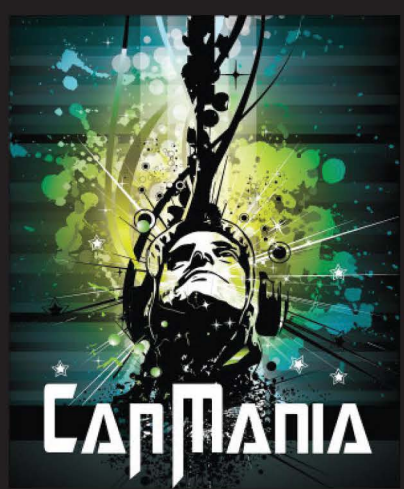
Dan Joffe is the force behind AkitikA and Updatemydynaco. He has a master's degree in EE from Stanford University and more than 50 patents. He's interested in all aspects of audio electronics and plays the saxophone as well. He is the author of Saxophone Secrets, published by Jamey Aebersold Jazz. You can reach him by email via dan@akitika.com.



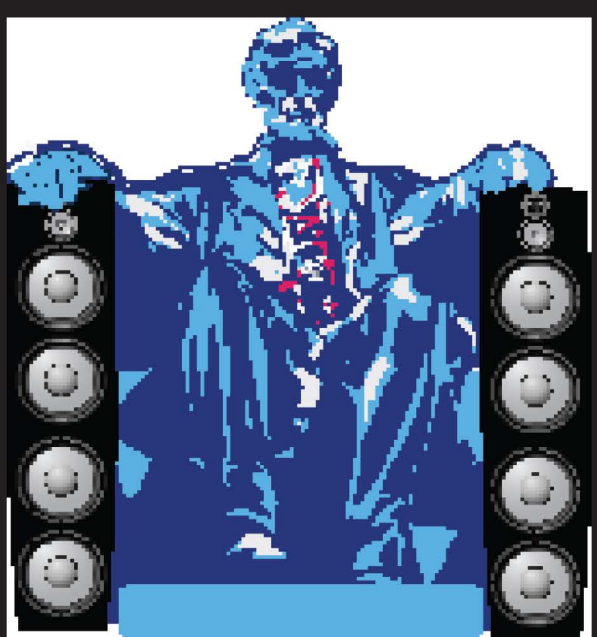
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Concept Design for a Small Bass Amplifier

Explore the concept of building a small easily portable bass amplifier with a "fat" sound, which our designer has dubbed the Bolger.



Photo 1: The GBX Bass Driver and 80 W, twin-15" cabinet provided great sound and enough power for most lounge work.

By
Richard Honeycutt
(United States)

In past Hollow-State Electronics articles, we've often discussed electric guitar amplifiers, but bass amplifiers... not so much. I recently started thinking about designing and building a small, easily portable bass amp with enough power and bottom end to sound "fat." Not exactly an easy task, since small speakers tend to be inefficient and weak in bass output, and if we compensate for their inefficiency by upping the power, the amplifier becomes large and heavy.

But Hollow State is devoted to hollow-state electronics, not speakers, so let's put the speaker considerations aside for the time and flesh out the concept for such a bass amp. First, we'll need a name for the project, just for our own convenience. Being a Hobbit fan, when I operated Evenstar Audio, I often kidnapped product names from J. R. R. Tolkein's writings. In that spirit, we'll call this small, fat bass amp the "Bolger," after Fredegar Bolger, a hobbit (small) whose nickname was "Fatty."

The Design

So as a semi-pro bassist for about 10 years, would I prefer the Bolger to be hollow-state or solid-state? As a matter of taste, I think an electric bass amp should be clean—distortion on a bass in no way improves the music. I understand that other bassists



Photo 2: The GBX Bass Bug, a 30 W, single-15 self-contained bass amp was small and lightweight for the quality and amount of sound it produced.

disagree with me, but I'm designing the Bolger for myself and those of similar tastes. I also have two other preferences that impact the design concept: The sound should be fat, with real low end, distinctly unlike the Hofner bass sound of the early Beatles era, and it should be "round," not twangy (bassists, think "flatwound strings, not roundwound ones").

Small bass amps are easy to overdrive, and then you get distortion. Tubes provide a gentler onset of distortion, and a well-designed tube compressor-limiter built into the preamp section is a real help in keeping the sound clean. But tube output stages are heavy! We put up with the weight of all-tube guitar amps since part of the distortion signature we want comes from the phase splitter and the output stage. However, this distortion is not part of our goal for a small, clean bass amp.

Actually, my two favorite bass amps were both solid-state models by GBX, a Canadian company of the 1970s and 1980s. For lounge jobs, I used the Bass Driver (a unique bass preamp having about a 3 VRMS output) feeding an 80 W, twin-15 powered speaker cabinet (**see Photo 1**). For practice or accompanying acoustic guitar and voice, I used the smaller Bass Bug (see **Photo 2**), a 30 W single 15 amp. The Bass Bug is much more portable than the big stack, but at 30 to 40 lb and 26" × 21" × 12", it sometimes made me long for something smaller and lighter. Neither of these amps had any sort of compression, although in the venues we played, we never got loud enough to push the big stack into distortion. However, the Bug required a bit of care with the volume settings to keep it from bring overdriven.

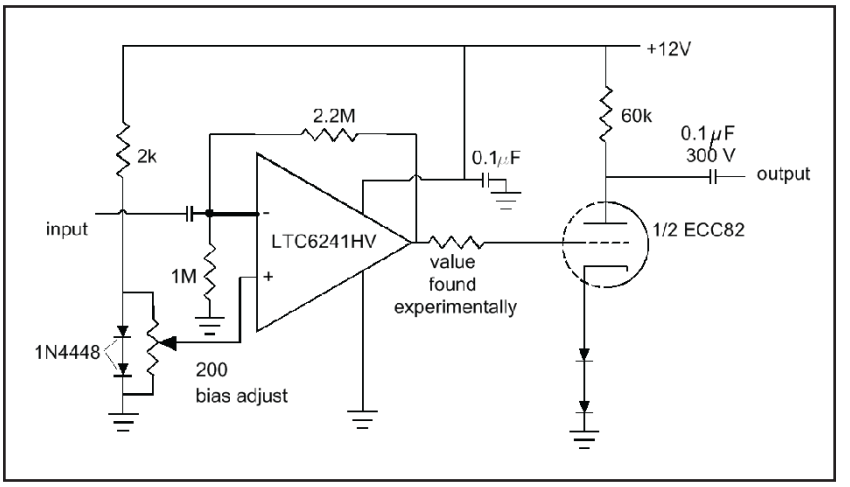


Figure 1: A 12AX7/ECC82 in starved operation can operate using a low-voltage B+ supply, but requires a buffer prior to the input in order to have a high enough input resistance.

Hybrid Potential

So it would really be nice to have some tube compression, but the control and power sections could be solid-state. Amplifier manufacturers started offering hybrid guitar amps in the 1970s. One of the best-known was Music Man, founded by Leo Fender. These amps used solid-state preamps and tube power amps. Maybe the Bolger needs to be a hybrid?

The ultimate in lightweight power amps are the Class D (switching) amps. Texas Instruments (TI) offers its TPA3116D Class D power amp, providing 50 W stereo or 100 W mono output, in a very small, lightweight package. Only a very small heatsink is needed. Using a 24 V switching power supply, you can get the power supply and power amp down to just a few pounds. And the assembled development kit can be purchased at a low cost from several suppliers. Since we're using tube compression to keep the power amp from being overdriven, a Class D solid-state power amp and switching supply should be fine.

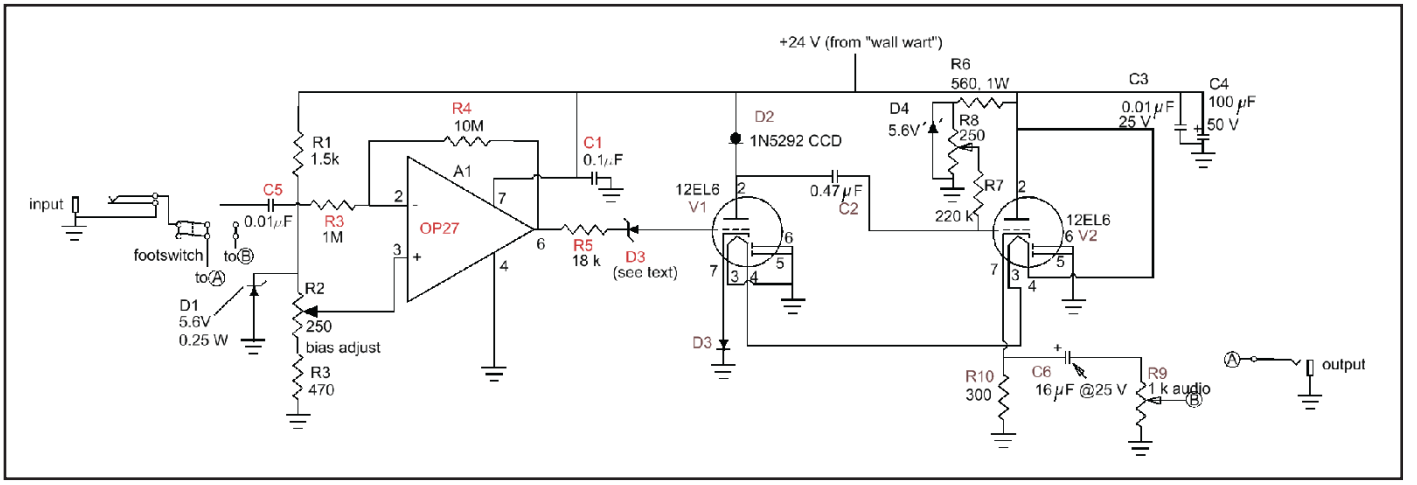
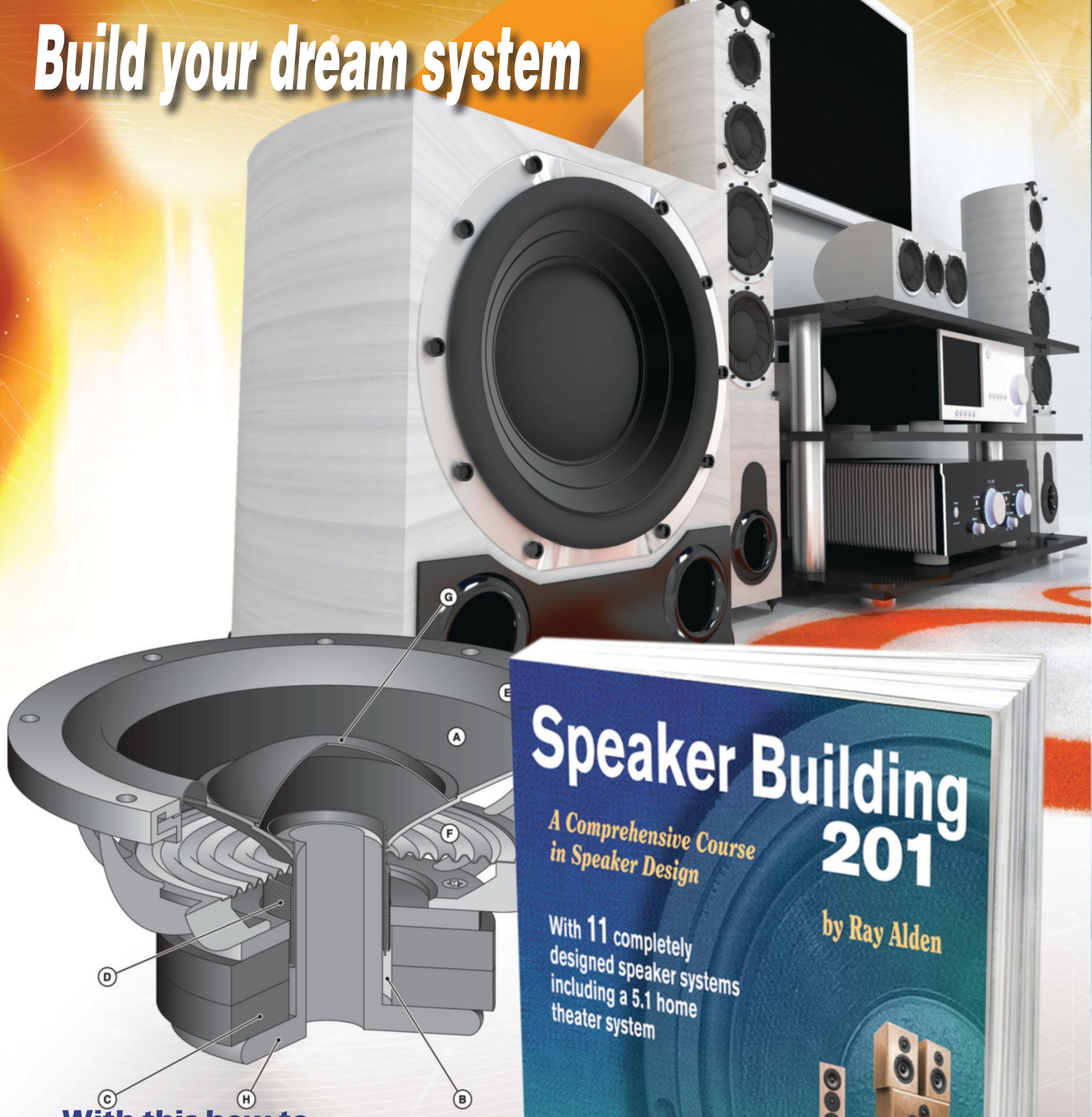
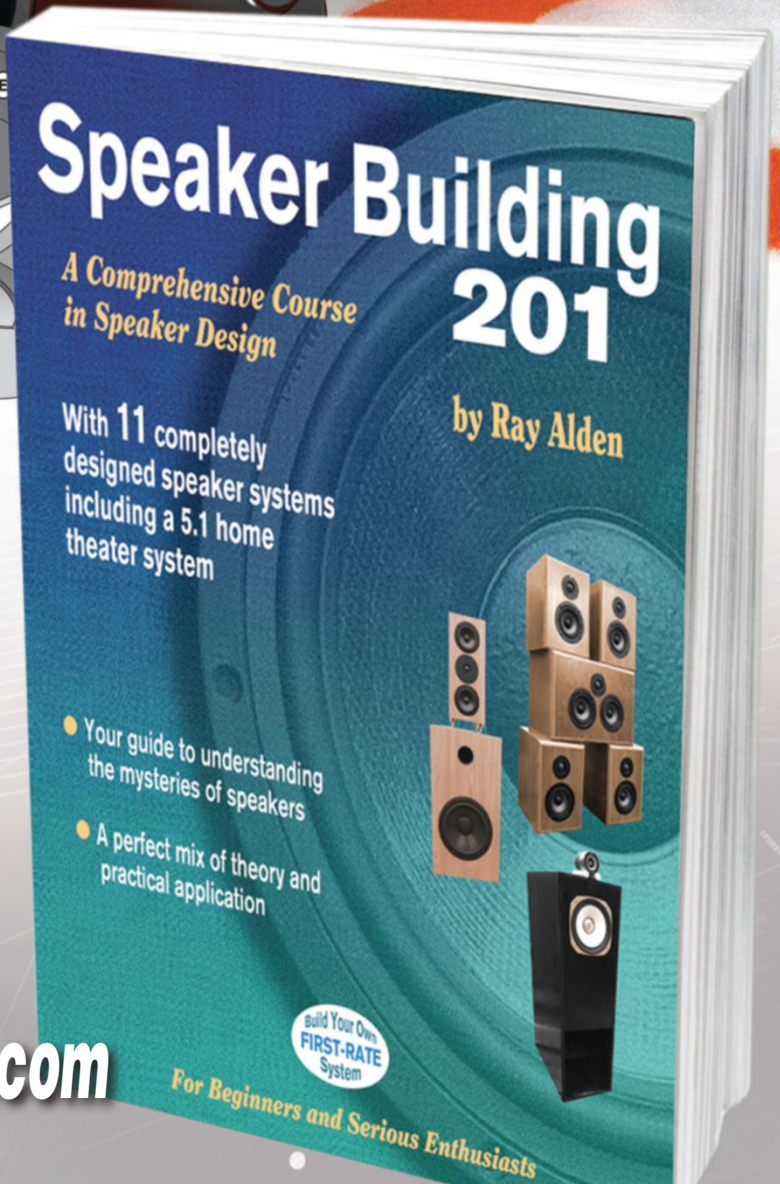


Figure 2: Two 12EL6 space-charge triodes were used in a tube distortion pedal having a 24 V B+.

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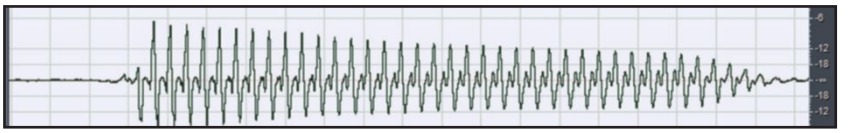


Figure 3: The "fingerboard-end" pickup of a bass produces this waveform for the open low E string.

Potential Configurations

In the November 2014 Hollow-State Electronics article, "Starved Amplifier Operation," we discussed starved operation of tube amplifiers as one way to avoid having to use a high-voltage power supply for a preamp. An unfortunate consequence of starved operation is low input resistance. For a guitar or bass amp, we need an input resistance of about 1 MΩ or so. **Figure 1** shows a configuration using a rail-to-rail CMOS op-amp driving a starved triode that provides the needed input resistance. While this circuit would work in the Bolger, the LTC6241HV is not available in homebrew-friendly packages such as 8-pin or 14-pin DIPs.

Another option for using tubes with a low B+ is space-charge tubes, as discussed in the December 2014 Hollow-State Electronics article, "Space Charge Tubes." These tubes were designed to be used in car radios, so the available B+ voltage was limited to a nominal 12.6 V in most cases. They enabled car radio designers to dispense with the noisy and unreliable vibrators that chopped DC to square-wave AC for transforming to a higher voltage before rectification and filtering. The design of these tubes enabled them

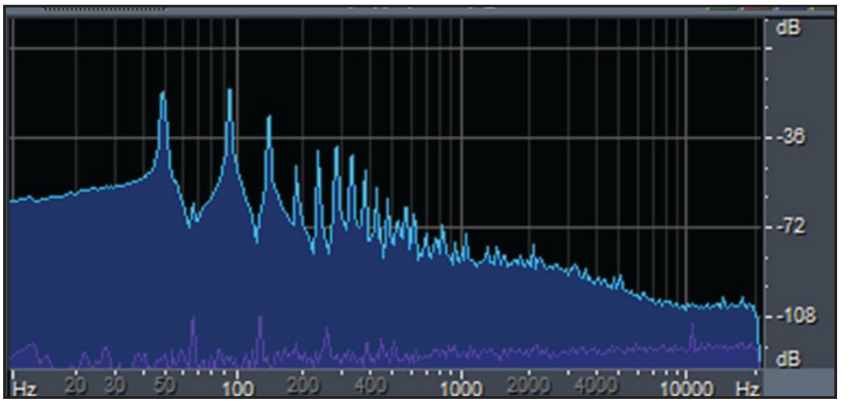


Figure 4: The spectrum of the low E contains the fundamental and strong harmonics up to the 12th.

to operate pretty much like more common tubes, without the limitation on input resistance that starved operation imposes.

The January and February 2015 Hollow-State Electronics articles (see Resources) discussed the design of a tube distortion pedal using two 12EL6 space-charge triodes, fed by an OP27 impedance

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translator stage. The OP27 is available in an 8-pin DIP package, making it easy for DIYers to use. **Figure 2** shows the circuit of this pedal. (Actually, the 12EL6 is a triode/dual diode, but the diodes were not used in the distortion pedal.) While this circuit is not ideal for the Bolger, it does present a concept from which we can work as we create the actual design.

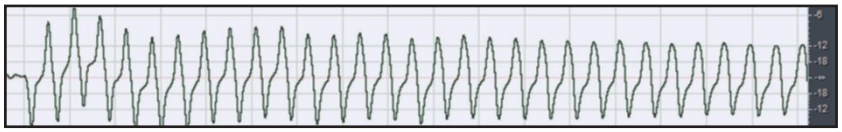


Figure 5: The "fingerboard-end" pickup of a bass produces this waveform for the highest F.

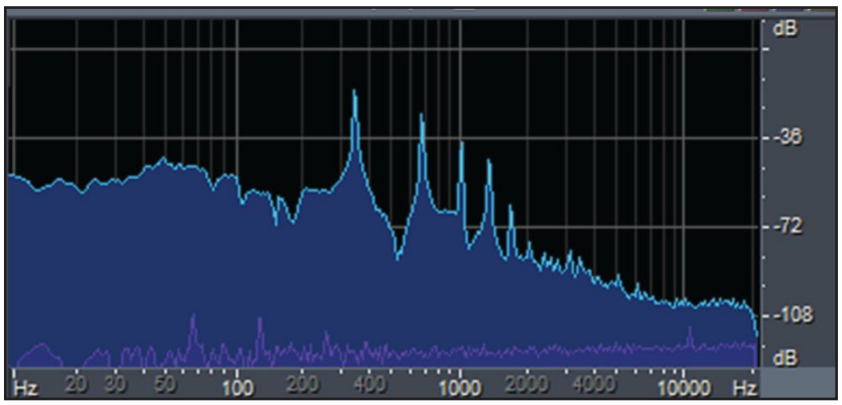


Figure 6: The highest F contains a fundamental plus strong harmonics up to the fifth, with a little seventh and ninth.

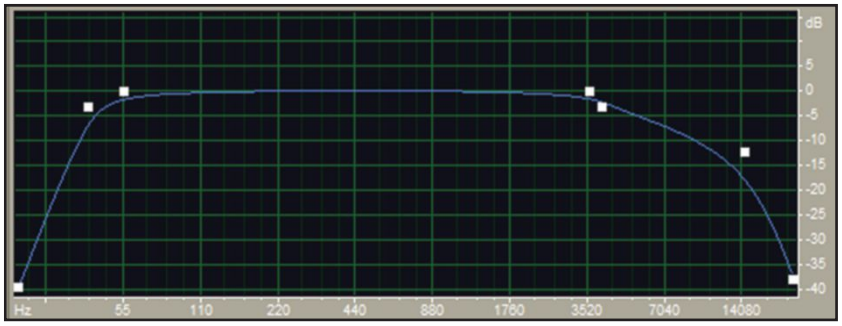


Figure 7: The target response for the Bolger with the tone controls set at mid position shows a low-frequency cutoff about 40 Hz, and high-frequency rolloff above 4 kHz.

Resources

R. Honeycutt, "From Prototype to Final, V. 1.0," *audioXpress*, February 2015.

——, "From Paper to Prototype," *audioXpress*, January 2015.

——, "Space Charge Tubes," *audioXpress*, December 2014.

——, "Starved Amplifier Operation," *audioXpress*, November 2014.

Yuan-Jing Audio, www.yuan-jing.com.

Source

TPA3116D Class D power amplifier
Texas Instruments | www.ti.com

Frequency Response

Next, we need to identify the necessary frequency response for the complete amplifier/speaker combination. The fundamental frequencies for a four-string electric bass using standard tuning (E1, A1, D2, and G2 in musical terms) span the range from 41.8 to 349.23 Hz. My bass is a 1972 Rickenbacker 4001. The waveform of the low E string is shown in **Figure 3**, and the spectrum corresponding to that waveform is shown in **Figure 4**. Unlike the upright string bass, this electric bass has a significant amount of fundamental. The highest note (F4) produces the waveform shown in **Figure 5**, and the spectrum shown in **Figure 6**. The highest harmonic that can be clearly seen is the ninth, at 3143.07 Hz. This bass has flatwound strings. Roundwound strings would produce more high harmonics, and the "bridge" pickup would show lower amplitudes of the low harmonics and higher amplitudes of the higher ones. The spectrum shows energy up to about 10 kHz. However, experiments reveal that cutting off the reproduction of the bass above 4 kHz has no audible effects on the sound, for this particular bass and flatwound strings, using the fingerboard pickup. Experiments also show that the sound begins to noticeably thin if the low frequencies are not flat down to 40 Hz or so. Thus, we need to maintain a minimum frequency response range of 40 to 4,000 Hz, at the -3 dB points.

Low frequencies need proportionally more power than mid and high frequencies, due to the weak low-frequency response of the human hearing system. Low frequencies also require greater excursion of the speaker cone, compared to higher frequencies at the same reproduced level. So to maximize the amplifier's loudness, without driving the speakers into nonlinearity and perhaps overdriving them, we should design the electronics to limit the low-frequency output from the power amp. There is no need to pay particular attention to the high frequencies. **Figure 7** shows the target response, including the minimum high-frequency response, which will be limited mainly by the speakers.

Tone Control

Tone control preferences are highly individual. **Figure 8** shows the effects of various settings of the bass control on the GBX Bass Bug. Notice that the range of control at 40 Hz is about +12 dB (see the blue line—maximum boost) to -12 dB (see the magenta line—maximum cut), with a turnover frequency of about 200 Hz. Experimenting to achieve optimum effects for my taste indicated that a turnover frequency of 250 Hz would be preferable, but the ±12 dB range is appropriate. Similar experiments

indicated that the treble control should have a turnover frequency of 500 Hz, and a range -5 to +15 dB at 4 kHz. Other bassists having different equipment or tastes may prefer other specs for designing the tone controls.

There is no particular reason to use hollow-state electronics in the tone control section. With the shift to surface-mount components, getting homebrew-friendly parts has become a challenge. However, several companies now produce modules that can be used for the tone controls. Yuan-Jing Audio produces a tone control board with conventional parts, enabling us to change the gain, input resistance, and tone-control characteristics. In the actual circuit design process, we will need to design the tube preamp first, then determine the minimum input resistance the tone control module will need in order to play nicely with the preamp. We will also need to verify that the tone control module will happily feed the ~5 kΩ input resistance of the Texas Instruments TPA3116D Class-D power amp. If we find that we cannot provide enough gain in the preamp, we can probably add gain in the tone control module.

Figure 9 shows the block diagram for the concept design of the Bolger hybrid small fat bass amplifier. The circuit design will be chronicled in another article.

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

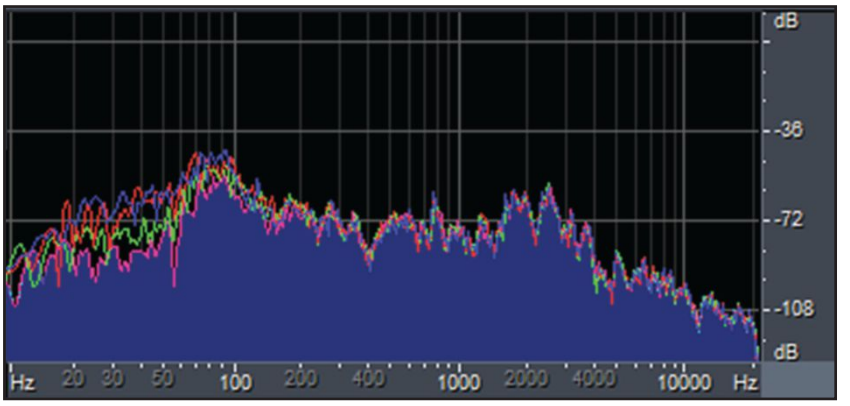


Figure 8: The GBX Bass Bug's bass control is effective mainly below 100 Hz.

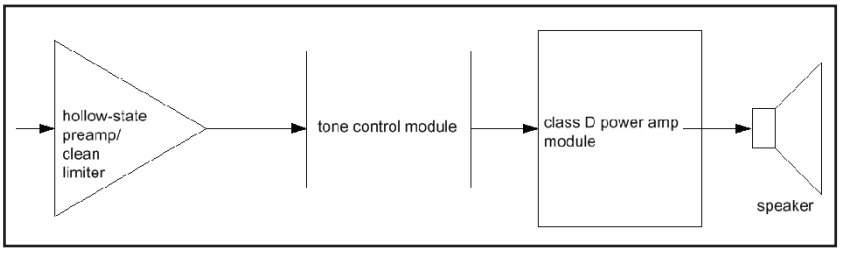
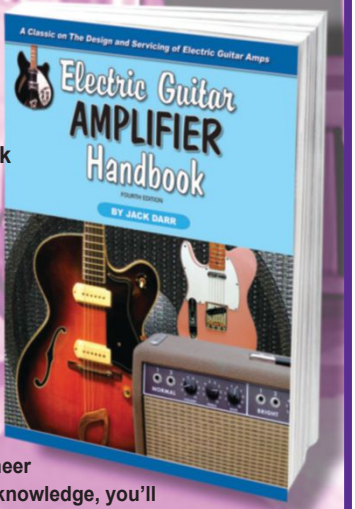


Figure 9: This is the block diagram for Bolger concept design.

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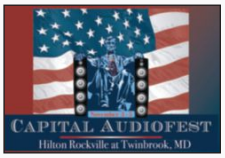
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Industry Calendar

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

November 3–5, 2017
Capital Audiofest (CAF)
Hilton Hotel at Twinbrook Metro, 1750 Rockville Pike, Rockville, MD
www.capitalaudiofest.com



The popular East Coast's audio show is now expanded. Following the 2016 edition with the largest visitor turnout ever during three days, the promoters decided to keep the show dates in November at the same venue, the Rockville Hilton Hotel at Twinbrook. The show benefited from the hotel's recent renovations, allowing for larger listening rooms and an expanded Marketplace and CanMania dedicated areas with 50% more space. The Rockville hotel has a major Metro line right out the back door, connecting from/to the center of Washington DC with no transfers. Also, Rockville has great restaurants and shopping. The promoters also promise more lectures, and more live entertainment than in the past. Visitors will find an expanded CanMania exhibit filled with more than 30 headphone companies, promoted in coordination with Headphone.Guru.

November 10–12, 2017
New York Audio Show
Park Lane Hotel on Central Park South, New York, NY
www.chestergroup.org/newyorkaudioshow/2017



The New York Audio Show will return to Manhattan, once again taking place at the luxurious Park Lane Hotel on Central Park South, one of New York's most prestigious locations. The promoters of The New York Audio Show, Chester Group, have coordinated with the Capital Audiofest promoters, in order to advance the show to the following weekend (November 10-12), allowing many exhibitors to participate in both. The New York Audio Show takes three classic floors on consecutive levels in the hotel, with spectacular views that overlook Central Park. Some of these rooms are amongst the largest available, allowing perfect demonstrations of the best in high end audio and home theatre.

Located directly on Central Park South (aka 59th Street), in Manhattan's toniest midtown location, the Park Lane Hotel is just steps from Fifth Avenue Shopping, Broadway Theaters, Museum Mile, Carnegie Hall, Radio City Music Hall, Lincoln Center, and many other NYC activities and attractions.

November 13–19, 2017
Exhibits: November 17–19
Live Design International (LDI) Show
Las Vegas Convention Center, Las Vegas, NV
www.ldishow.com



Since 1988, Live Design International (LDI) has been a leading trade show and conference for live design professionals from all around the globe. LDI hosts over 13,000 attendees from more than 80 countries, working in theaters, concerts, clubs, theme parks, and houses of worship, as well as a wide range of international live and broadcast venues. More than 350 companies exhibit, mostly stage and lightning vendors, but also many professional audio companies, providing live demos of cutting-edge gear.

November 15–17, 2017
Inter BEE 2017
Makuhari Messe, 2-1, Nakase, Mihama-ku, Chiba City, Chiba Prefecture, Japan
www.inter-bee.com



Promoted by the Japan Electronics and Information Technology Industries Association (JEITA), and run by the Japan Electronics Show Association (JESA), with the support of the Japanese government, all major broadcasting organizations and associations in Japan, and certified by the United States Department of Commerce, the International Broadcast Equipment Exhibition (Inter BEE) is always an important event for the professional audio and video industries. The event traditionally focuses on television technologies and is chosen by all major Japanese companies for key technology and product presentations—sometimes well ahead of any other show. Inter BEE is an important opportunity to promote solutions in the Japanese market with a large impact across Asia. The location, at the Makuhari Messe, is highly convenient and exhibitors recognize a very high level of visitors, including from remote areas of the world. Professional audio equipment companies represent close to 100 exhibitors, including all major microphone manufacturers.

For the first time, Inter BEE 2018 will be using Halls 1-8 at Makuhari Messe and attendance is set to go even higher this year, with more than 1,000 exhibitors. In anticipation of the 2020 Tokyo Olympics, the show will reflect important industry's efforts to enhance broadcasting services with 4K and 8K production and transmission. This year's exhibition will also feature demos of large-format speakers (X-Speaker), headphones (X-Headphone), and microphones (X-Microphones) at the Inter BEE Experience.

January 6–7, 2018
ALMA International Symposium & Expo
South Point Hotel & Casino, Las Vegas, NV
www.almainternational.org



Since 1964, The Association of Loudspeaker Manufacturing and Acoustics (ALMA) International has brought electroacoustic and audio professionals together for cutting-edge education, quality networking, and an unparalleled environment in which to get business done. For 2018, the revitalized ALMA International Symposium & Expo (AISE) will expand on all of the things that exhibitors and attendees always have appreciated about this event, taking it to another level, with new innovative programs, and activities. The new venue, South Point Hotel & Casino, will certainly contribute to an entirely new feeling. 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 nearby locations in Las Vegas, NV
www.cesweb.org



For 50 years, CES has been the launch pad for the finest innovation and technology that has changed the world. Held in Las Vegas, NV, every year, it 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.

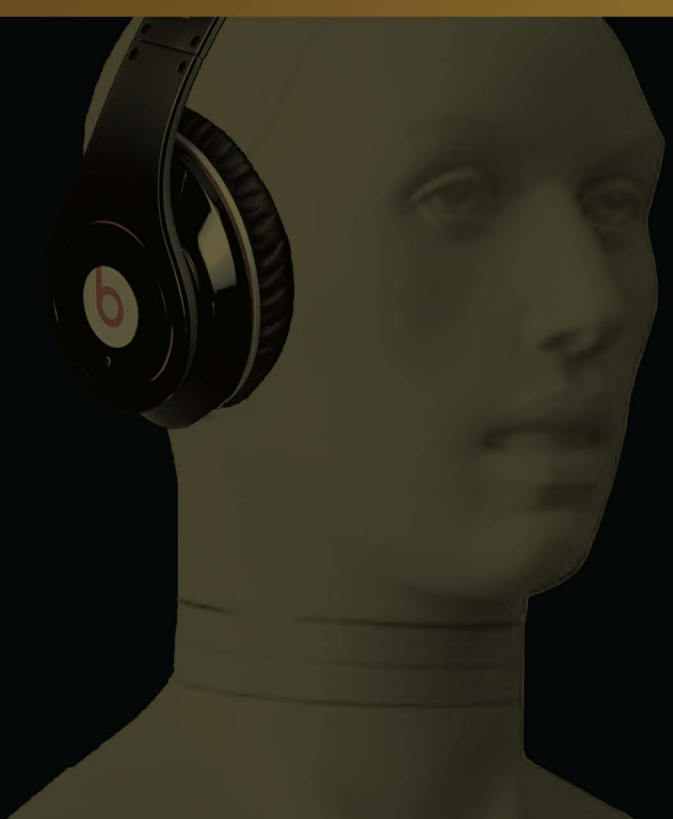
CES, formerly The International Consumer Electronics Show (International CES), showcases more than 3,900 exhibiting companies, including manufacturers, developers and suppliers of consumer technology hardware, content, technology delivery systems and more. The conference program includes more than 300 conference sessions. And, the promoters expect more than 170,000 attendees from 150 countries. 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.

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