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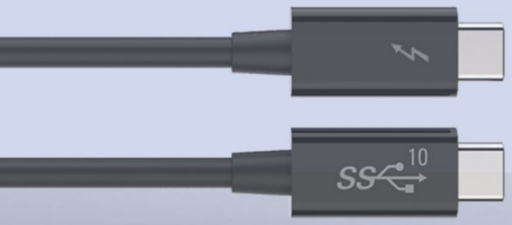
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Room for Technology Improvements

On October 27, 2016, as expected, Apple announced a new generation of MacBook Pro laptops exclusively featuring Thunderbolt 3 ports—which now use the "universal" USB Type-C connector. We knew it was coming, but now there's actual confirmation about this important transition. Of course, this month's Standards Review article reflects the announcement.

This is exactly what the industry was waiting for and is especially relevant since the new Mac computers are reaching the stores even before we print this edition. Now, brace yourselves for what's coming, because this will be a transition affecting every aspect of the consumer electronics, IT, mobile telecom industries, and obviously, the audio industry. As we write in this month's article, its not only about the evolution in the interfaces and the new connector, there is also the new USB Audio Device Class 3.0 Specification for Digital Audio Applications, which will have a profound impact over the next few years.

The other significant event that deeply affects the audio industry happened shortly before (September 7), and it was Apple's announcement of the new iPhone 7, which anticipates the wireless future—in particular for audio. Together with the new iPhone, Apple announced its own concept of truly wireless earbuds—the Airpods—powered by its own silicon, the W1 chip, which also powers a range of new audio products from Beats.

The problem is, as with every industry effort, this needs time to be fully implemented and understood. With the physical interfaces and new specifications for USB-C and Thunderbolt 3, we need more products for things to start making sense for consumers. The technology is ready, the benefits are clear, but they are not yet set in the minds of consumers. As for wireless audio, it seems to be the opposite. The concept clearly captivated consumers' imaginations, but the technology is not there yet.

Basically, Apple developed the W1 to overcome known limitations of existing Bluetooth technology, sending separate synchronized streams to each earbud (instead of dealing with the complicated problem of sending signals from left to right), and support bidirectional transmission for Siri interaction and voice calls. It's an ambitious endeavor (not cheap) that only a company such as Apple could pull off. Everyone in the industry is dying to know more about the W1 chip, which when applied to Beats' new generation wireless headphones allows for very impressive specifications—one-third of the power of traditional wireless chips and up to 40 hours of battery life, an industry benchmark.

The W1 chip is an attempt to solve problems that all the (many) companies developing wireless earbuds have faced while trying to comply with existing standards that allow interoperability. Several companies have explored other technologies (e.g., NFC) to solve the challenges that truly wireless earbuds pose, without draining the batteries too quickly. Unfortunately—as confirmed by the fact that even Apple is delaying the product introduction of the Airpods—it has become increasingly clear that we will need a new set of technologies to achieve what's needed—precisely the same requirements that have been announced for Bluetooth 5, but not only.

While discussing the challenges of designing new "truly" wireless earbuds, which is an ongoing topic on the audioXpress website, we found multiple references to Near Field Magnetic Induction (NFMI) complementing Bluetooth. One of the leading names in NFMI is FreeLinc Technologies, a research and development company focused on the adoption of this technology as a new wireless standard in the Public Safety, Military, Healthcare, and Consumer markets. For this issue, *audioXpress* invited Dr. Michael Abrams, Freelinc's Chief Executive Officer, to share his valuable perspective about the role of NFMI in wireless audio applications.

Abrams also believes that we need a growing ecosystem of technologies and that Bluetooth 5.0 might hold some, but not all of the answers. NXP has been exploring the combination of Bluetooth and NFC to respond exactly to the same problems that NFMI, being a truly Wireless Personal Area Network (WPAN) already solves. It was already rumored that a combination of those technologies is being considered for Bluetooth 5.0, the strongest motivation being IoT, security and hearing aid applications.

The recently announced Qualcomm/NXP merger promises to make things even more interesting in this domain. I just hope that within the scope of those proposals there might be some room for audio quality improvement.

João Martins
Editor-in-Chief

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COLUMNISTS

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

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

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

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

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

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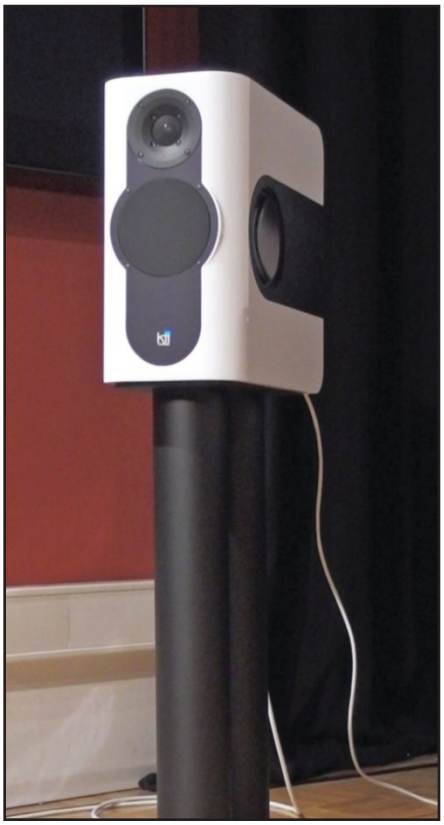
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In an effort to improve his home stereo system, the author decided to drive his loudspeakers with a Class-A power amplifier. The first part of this article details his objectives and the design.

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Streaming Music Experiences (Part 1)

Streaming Direct Stream Digital Audio

How good are streaming music services? Are we able to actually enjoy the best sound reproduction using music that's streamed over the Internet given the bandwidth restrictions and variable data rates?



Photo 1: This is the ifi nano-iDSD DAC. The low-pass filter selector switch, USB input connector, and digital output jack are shown. The stereo RCA analog output connectors, headphone jack, and volume control are on the hidden end. The module automatically senses the input type: PCM, DXD, DSF or DFF, and shows the type by the color of the LED.

By
Ron Tipton
(United States)

From highly compressed music on Internet radio stations (32/64 Kbps) to popular music services using lossless compression schemes (320/256 Kbps) to lossless compression (1,141 Kbps) used by Tidal's hi-fi service, music streaming is now mainstream (pun intended). And high-resolution audio (HRA) will eventually transition from downloads to streaming as well. This series of articles looks at the real-world experience with streaming services, in search of the best possible quality.

The acronym for Direct Stream Digital (DSD) relates to the high-resolution file format devised by Sony and Philips in the 1990s for the Super Audio CD (SACD) physical disc format. While SACD is virtually extinct, there are still plenty of music labels offering music in DSD format for download—45 are listed on the Native DSD Music website (NativeDSD.com), but there are more. Original material recorded as DSD data is available as .dsf file downloads. DSD material is also used by at least one website, PrimeSeat, to stream stereo music.

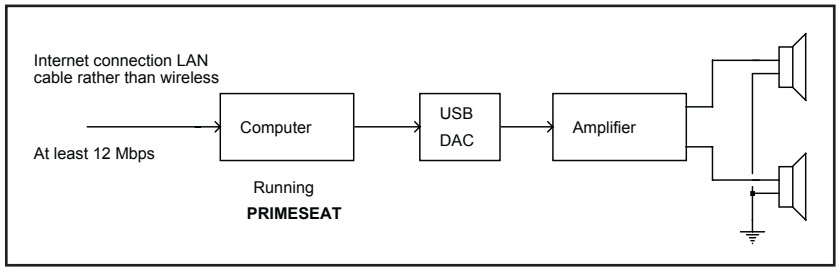


Figure 1: To listen to a PrimeSeat stream, use this equipment setup for loudspeakers. Other setups and play instructions are shown in the PrimeSeat.pdf found in the Supplementary Material for this article.

In early 2016, two concerts were hosted by Blue Coast Records (BCR). They were both streamed live and recorded in native DSD. Both were available for on-demand streaming for a limited time. The January concert was available free of charge until August 6, 2016, so I listened, using the equipment setup shown in **Figure 1.** For the setup, I used an ifi nano-iDSD DAC (see **Photo 1**), with a USB in and an analog stereo out. I was impressed by the "concert-like ambiance," which was captured by the live recording. Both concerts are available as file downloads from BCR at reasonable prices.

PrimeSeat

The necessary PrimeSeat software is a free download for Windows 7, 8.1, 10, and Mac OS X 10.8-10.11. As shown in **Figure 1**, a USB DSD-DAC is

needed, but you can listen in pulse-code modulation (PCM) without it. My ifi unit is fine as are all the Korg DACs—PrimeSeat is a cooperative project of IIJ, Sony, and Korg. A high-speed Internet connection is needed—at least 12 Mbps, sustained. You can find out what is available for streaming by launching the PrimeSeat program, the available content is displayed on the opening screen (see **Photo 2**).

Select the one you want, scroll down to Details, and click it. This displays the streaming control screen (see **Photo 3**). Select the sample rate and click Play. The elapsed playing time is shown top-left with a stereo level bar graph and volume slider top-right. Play can be paused or stopped, rewound to the beginning, or advanced to the end with the standard icons.

Notice the 2.8 MHz displayed in the upper right corner, which indicates that DSD is being streamed. If the DAC is not setup correctly, the stream defaults to 16-bit, 48 kHz PCM. To get my ifi DAC to work properly, I had to upgrade the driver to version 2.26 and choose ASIO as the driver in the PrimeSeat Audio settings screen. PrimeSeat is very easy to use. I expect increased user participation will lead to increased content.

But the needed high data rate raises the question: How does the DSD listening experience compare to lower rate streaming using PCM (e.g., wav, FLAC, Ogg Vorbis, etc.) files? To answer this, we need to look at the DSD origins and at some comparative data.

DSD Background

DSD is the same digital format originally used to produce SACDs. It uses a 1-bit ADC at a high bit rate to output a digital "stream" that some think sounds closer to a pure analog recording because it avoids the quantization error inherent in PCM (see **Figure 2**). You can run a DSD stream through a simple low-pass filter to recover the music. Doing the same with a PCM stream yields only noise. However, DSD is still an analog-to-digital conversion so it has its own brand of quantization noise.

Sony also intended to use the new DSD format to archive its library of music files because of the ease of conversion to various PCM formats to fill the desires of the buying public. Unfortunately, at the time, they did not fully understand the quantization noise problem. John Siau comments in a *Real HD-Audio* article by Mark Waldrep: "I actually had a Sony engineer say to me one time, and this was quite a few years ago… he said 'we realized after we got a ways down the road that DSD was kind of a mistake but we had too much invested in it.'"

This comment may be too harsh because there

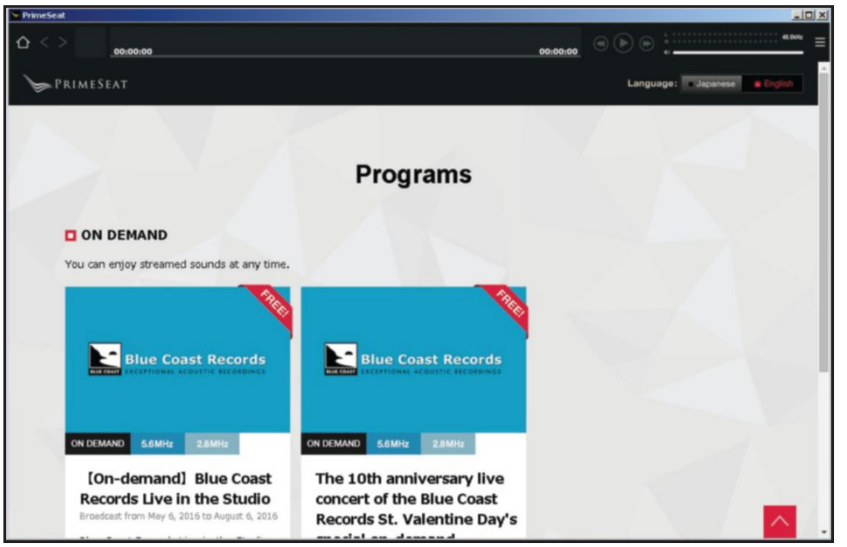


Photo 2: The PrimeSeat opening screen shows the content available for streaming.

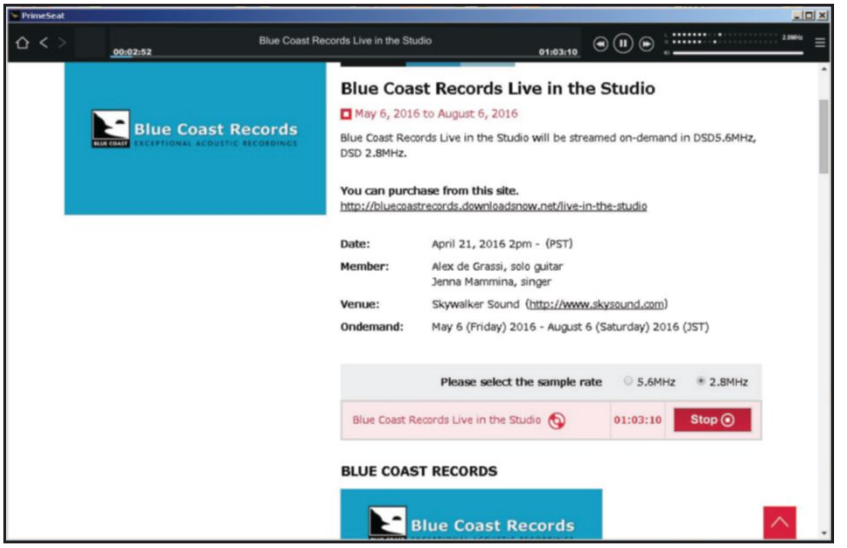


Photo 3: The PrimeSeat streaming control screen is easy to use. Select the sample rate and click the Play button.

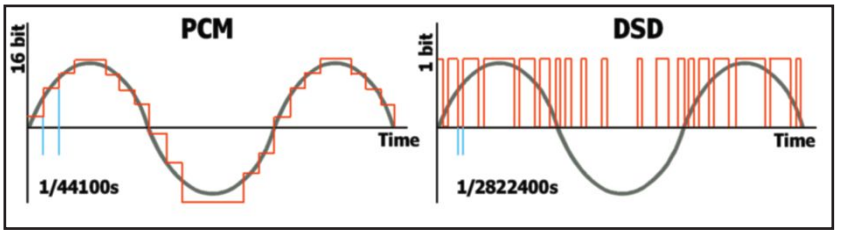


Figure 2: A comparison of the digital sampling as done by PCM and 1-bit DSD.

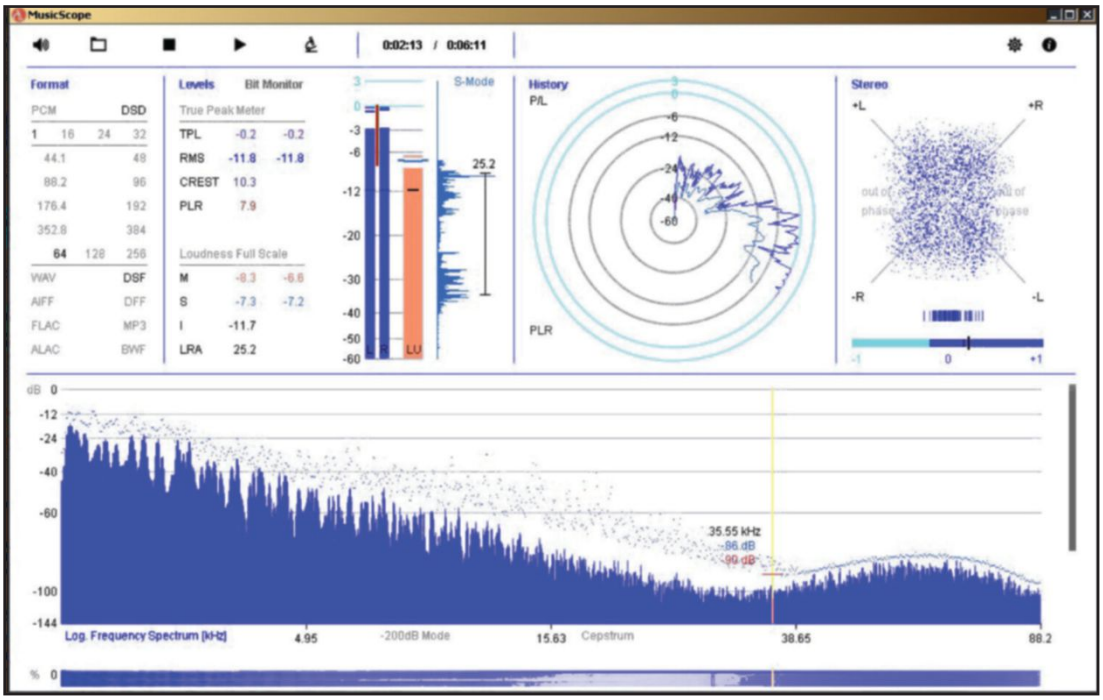


Photo 4: A screen-capture shot of the MusicScope software looks at the concert stream. The USB DAC's digital output is converted to optical and connected to the input of an internal sound card. MusicScope is looking at this card's input. The quantization noise is evident above 35 kHz. The actual screen has a black background but I reversed the colors to make the detail easier to see.

are reasons to have DSD files. One reason is the ease of conversion to various PCM files. Also, the digital-to-analog conversion is simple, you just need a low-pass filter. But, the existing external DACs are not just low-pass filters because they are designed to also decode PCM formats. This makes them somewhat expensive.

The biggest DSD flaw is its inability to be edited except by conversion to PCM. The BCR concerts streamed by PrimeSeat are probably the original, unedited recordings. They were recorded in DSD-64 or perhaps DSD-128 and are available for streaming, and as download files, in the same formats. You may be wondering, what do these numbers mean?

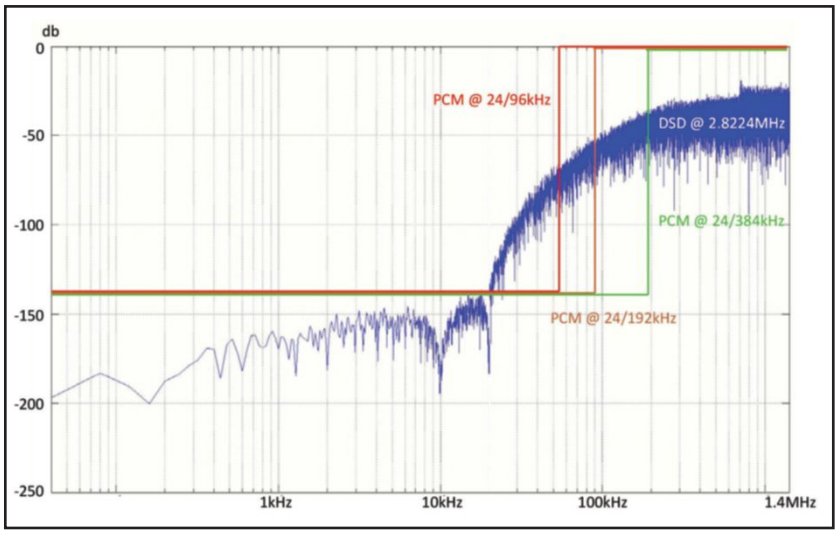


Figure 3: The theoretical dynamic range of DSD-64 and 24-bit PCM at sampling rates of 96, 192, and 384 kbps.

DSD File Types

Files with the extension DSF and DFF are known as "native." The basic recording rate is 64 times the CD rate of 44.1 ksps, which equals 2.8224 MHz. These file are usually designated as just DSD or DSD-64. Double-rate DSD (DSD-128, 5.6558 MHz) and Quad-rate DSD-256 at 11.2896 MHz are also available. DXD is a 352.8 kHz/24-bit audio signal, which is used to seamlessly move a DSD signal into PCM (standard method of digitizing audio) so editing and processing can occur.

This is perhaps a good time to mention a very useful computer program: MusicScope. **Photo 4** shows a screen capture of MusicScope looking at the concert stream. It clearly shows the high-frequency noise starting its amplitude rise at about 35 kHz. This program also plays PCM files and, in addition to the spectrum, it displays the root mean square (RMS) and peak amplitudes and the stereo vectorscope. I've included a copy of the MusicScope User Manual in the Supplementary Material found on the audioXpress website (www.audioxpress.com).

DSD vs. PCM

Figure 3 shows the theoretical dynamic range for DSD-64 and 24-bit PCM at three sampling rates. DSD is better at lower frequencies, below 150 dB. However, the high-frequency quantization noise starts to rise above 10 kHz but it's still down 100 dB at 30 kHz. The PCM appears better over a wider frequency range but this graph assumes "brick-wall" (high-order) low-pass filters, which can add phase problems in the output even when carefully designed.

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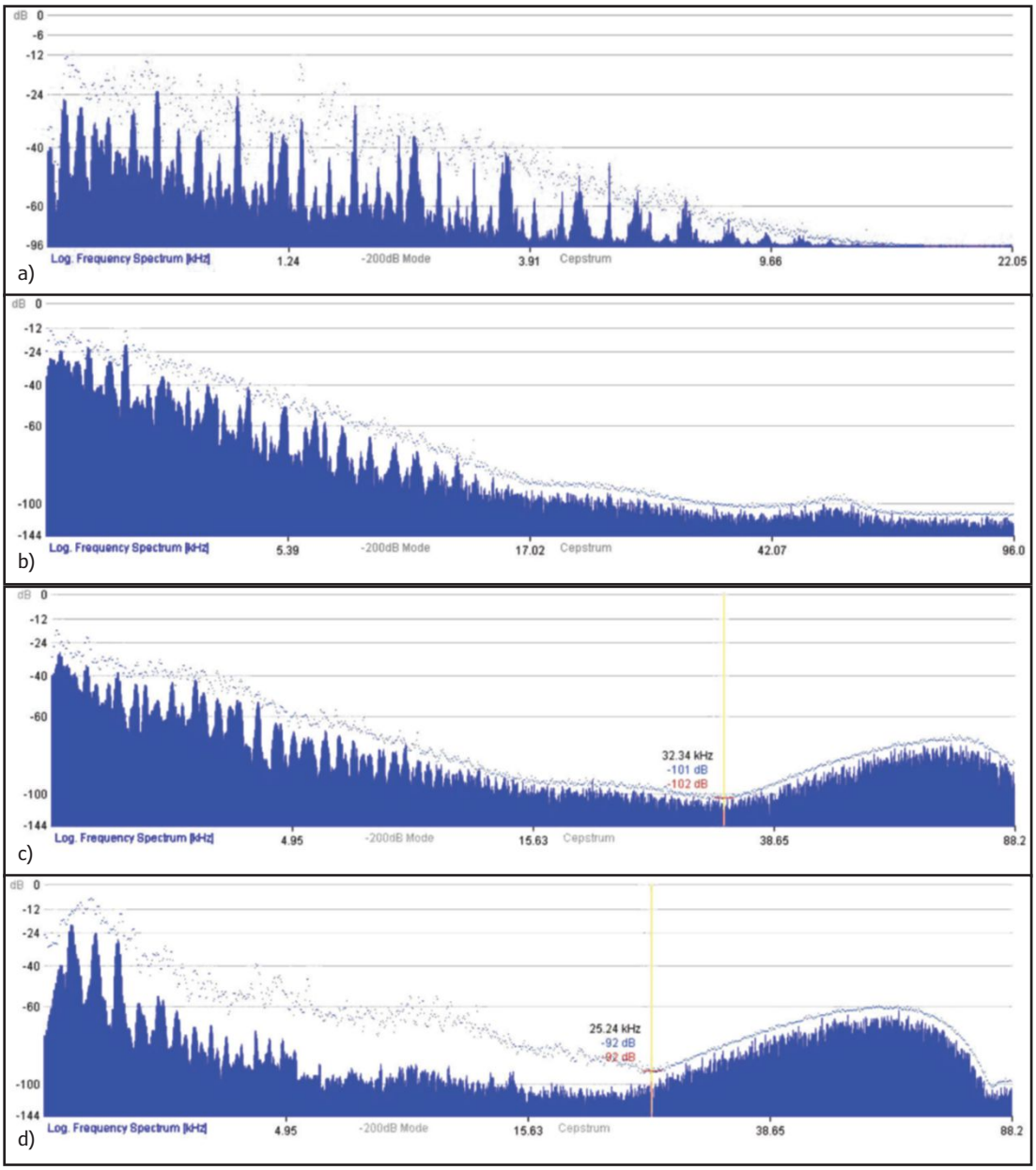


Figure 4: A comparison of the spectra of four music tracks as analyzed by MusicScope. The first three are from the same segment of Beethoven's "7th Symphony." 4a is CD quality, 16-bits at 44.1 kbps. 4b is 24-bits at 192 kbps. 4c is DSD-64. Note the increase in the level above 32 kHz. This is quantization noise. 4d is from a DSD-128 segment of a "Wichita Lineman" track. The quantization noise starts its rise above 25 kHz, which illustrates the noise added by the conversion to PCM and then back to DSF.

Figure 4 shows a collection of recorded-music spectra made with the Windows MusicScope software. The first three graphs are of the same 115-second segment from Ludwig van Beethoven's "7th Symphony," **Figure 4a** shows a .wav file at 16-bits and 44.1 kbps (CD quality). **Figure 4b** shows a .wav file at 24-bit and 192 kbps. **Figure 4c** shows a DSD-64 with quantization noise displaying a steady rise above 32 kHz. **Figure 4d** shows a segment taken from a DSD-128 recording of "Wichita Lineman." Notice that the quantization noise level rise starts at about 25 kHz. Although this track was recorded in DSD-64, it had to be converted to PCM for the upsampling to DSD-128 and then reconverted to the higher-rate DSD. As I mentioned earlier, each of these steps adds noise.

The Beethoven "7th Symphony" comparison files (.dsf and .wav), in the Supplementary Material, are free downloads from www.ayre.com/insights_dsdvspcm.htm. Listen and decide for yourself which you like better.

Another point worth considering is "how much resolution is good enough?" Benjamin Zwickel writes in his article, "The 24-Bit Delusion" that 20-bits are as good as our hearing and 16-bits are enough for listening. He writes, "So you can't actually hear the difference between the dynamic range of a 16-bit recording and a 20-bit recording unless you turn the volume up high enough above the background noise that it could cause permanent hearing loss." I have included a pdf copy of his article in the Supplementary Material.

So why use a higher bit rate? To start, because recording and mastering at higher resolutions makes it possible to deliver content at "true" CD quality. But that's a discussion we'll not get into.

Because DSD streaming options are limited, you may want to download and play DSD files. To help with this, I wrote an article, "Playing-DSD-With-and-Without-a-DAC.pdf," which I have included in the Supplementary Material.

Future Articles

There are other popular, stereo streaming services such as Spotify and Tidal. They stream at lower bit rates and I will go into detail about each of them in later articles in this series.

Spotify and Tidal stream in stereo but there is only one source of multi-channel, and the content seems rather limited. The Apple TV module, Version 4, accepts an audio or video Internet stream from your router and outputs an HDMI signal to your TV or wherever you want to direct it. Usually, the audio would be stereo but if the input stream contains multi-channel audio, it would be available.

Kanex has produced a 5.1 audio extractor for the HDMI stream. Its optical output can be decoded to 5.1 for your surround system. You can find 5.1 content from the Apple iTunes store. I will also review this product in a future article.

Project Files

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

Resources

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Korg DACs | www.korg.com/us/products/audio

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MusicScope | www.xivero.com

Native DSD Music | www.natived sd.com

PrimeSeat | primeseat.net/en

Sony | www.sony.com



The Rocky Mountain Audio Fest (RMAF) 2016

Fall in Denver is a glorious time to be on the high plateau, with golden acacia leaves falling like snow, and the real white stuff high on the mountain peaks. This year, the Rocky Mountain Audio Fest (October 7–9) faced quite a challenge, as the promised renovations to its preferred venue had not been completed on schedule. The result: perplexed exhibitors and attendees, with substandard space for exhibitors and poor lighting for all parties in the tent where most headphone vendors were exiled. Still, show traffic was decent but not exceptional, and your intrepid reporter got his harried self to quite a few "rooms" during the three days of the event.

By
Oliver A. Masciarotte
(United States)

Let's start with the headphone tent (see **Photo 1**). In the "OMG, these are so affordable!" category, 1More (usa.1more.com) was showing its new $60 iBFree Bluetooth in-ears featuring aluminum alloy housings, aptX, and titanium dynamic drivers. Also new, the $150 MK802 over-the-ears Bluetooth cans, which feature aptX and beryllium drivers for great sound in a lightweight housing (see **Photo 2**). While I was at their table, I finally got to listen to the wired in-ears, which are a tremendous bargain for the sound quality.

I'm not sure if the defunct Sonoma brand will provide any lift, but Warwick Audio Technologies was showing its new Sonoma M1 Headphone System (www.sonomaacoustics.com). The M1 employs DSP for a "pseudo-diffuse field response," along the lines of what Dr. Sean Olive and the Audio Engineering Society (AES) have been investigating as the best subjective choice for headphone playback. Built to last, the $4,995 package uses Warwick Audio's novel "HPEL" electrostatic transducer and includes a modern DAC Class A amplifier, DXD-capable ADC for analog inputs and energizer (see **Photo 3**). The latter provides a very high 1,350 VDC bias for a

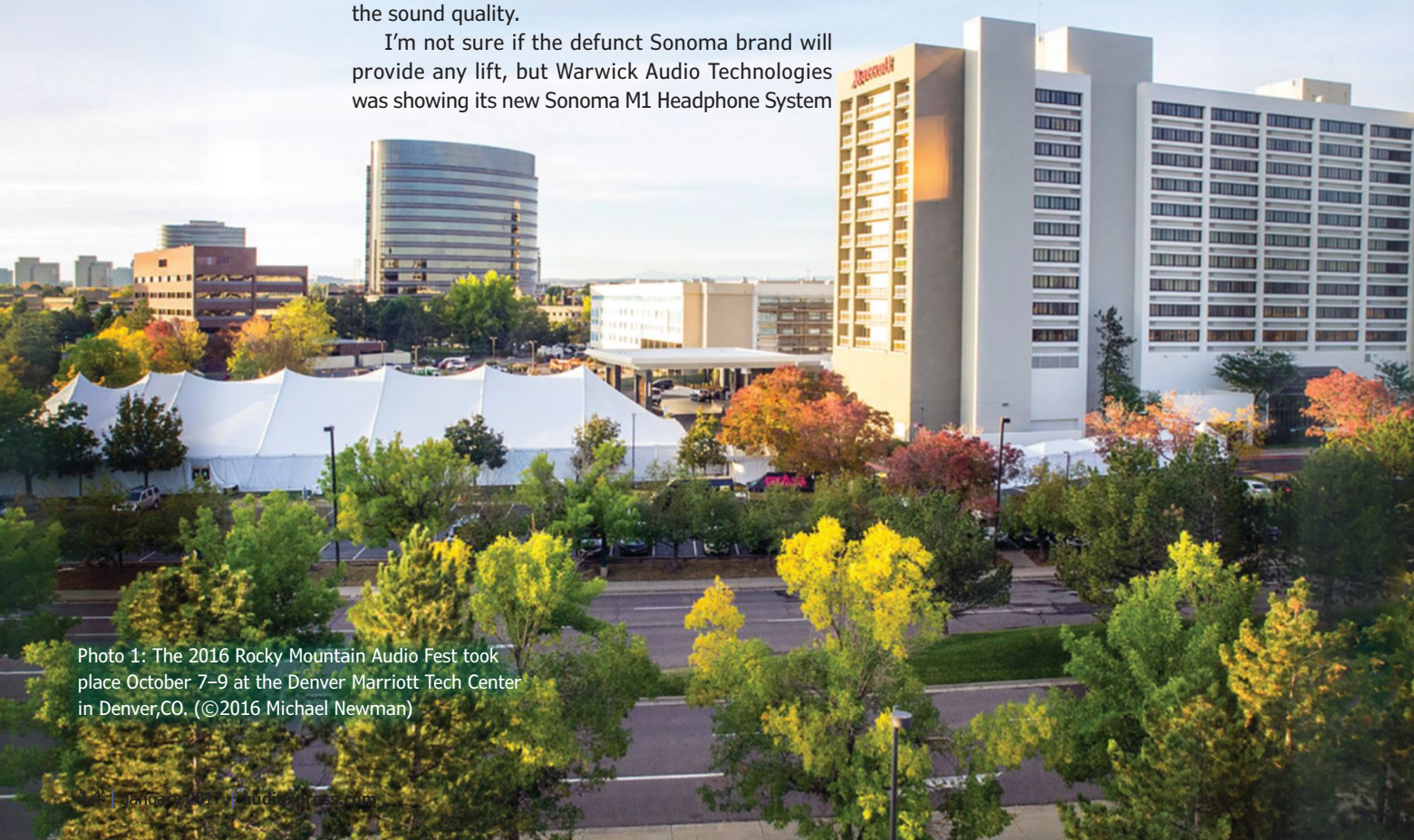


Photo 1: The 2016 Rocky Mountain Audio Fest took place October 7–9 at the Denver Marriott Tech Center in Denver,CO. (©2016 Michael Newman)

tight grip on the vanishingly thin diaphragms, and a marvelous, extended uncolored sound.

The electrostatic proliferation continues…A recognized name, HIFIMAN, has also gotten into the electrostatic game (www.hifiman.com). At the show, the company was spotlighting SHANGRI-LA, its new $50,000 entry in the ultra-premium cans category. With a retro modern 300b-based hollow state amplifier/energizer, the SHANGRI-LA certainly makes a statement of, ahem, excess?

Etymotic ER4SR and ER4XR (www.etymotic.com) replace the venerable ER4, with more accuracy, better sensitivity and that same honest Etymotic sound and great triple flange isolation and fit. The XR adds a slight bass lift. Both single balanced armature (BA) models are still priced at $349.

beyerdynamic's open-back $599 DT 1990 PRO ships with two sets of velour ear pads, allowing you to choose between an "analytical sound" or a slight low-frequency uptilt (www.beyerdynamic.com). With replaceable headband pads, a detachable cable, and 250 Ω impedance, these cans would satisfy pros or consumers looking for honesty, extended response, and comfort.

Astell&Kern (www.astellnkern.com) has branched out with the quasi-coincident AK Recorder option for its 300 series players. The $999 Recorder accepts AES3 as well as mic and line inputs, includes 48 V phantom power, and records in DSD2 (5.6 MHz). Also new is its AK XB10, portable Bluetooth-capable DAC/amplifier that adds Bluetooth to any headphones. The $189 XB10 employs aptX HD, has both balanced and unbalanced outputs on 2.5 mm and 3.5 mm jacks respectively, and includes a built-in microphone for taking hands free calls. In an announcement before the show, Astell&Kern affirmed that it will support TIDAL in all its current, Wi-Fi-enabled portable players via a firmware update.

Kris Cartwright, who designed the new Westone W80 with twin brother Karl, was on hand at the show to discuss their latest design (see **Photo 4**). The great sounding, $1,499 Signature Series W80 is an eight driver three-way with four high-frequency BAs, dual mids, and dual low-frequency bass BAs. Included is ALO Audio's "Reference 8" cable and capacious case.

Also showing was the new, lightweight IPX4–compliant MMCX Audio Bluetooth 4.0 wireless cable accessory, Westone's $150 take on how to add wireless headset functionality to existing IEMs (www.westone.com).

The dynamic duo of Bevin and Zach Mehrbach of ZMF fame demonstrated their first foray into an all in-house design (see **Photo 5**). It is a departure from their usual models based on Foster planar magnetic guts. They're ramping up production of the

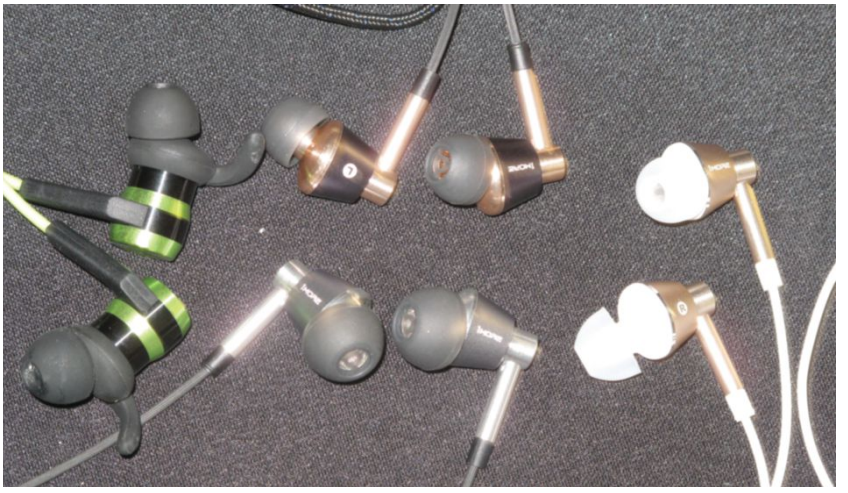


Photo 2: Bargain cans of the decade from 1More



Photo 3: The excellent Sonoma M1



Photo 4: Kris Cartwright (left) is shown with Blake Gaiser, holding the W80 and its included modular case. Gaiser set up his case with an AK380 and all of his IEMs.



Photo 5: Bevin and Zach Mehrbach display their new models.

$999 Atticus and the $1,299 Eikon. Both use 50 mm drivers and have 300 Ω impedance. Atticus houses a Thermoplastic Elastomer (TPE) copolymer diaphragm while my hands down favorite of the two, Eikon, employs bio-cellulose (www.zmfheadphones.com).

Walking the aisles of the headphone tent, I was more than a bit surprised to see a brand familiar to me from past Audio Engineering Society (AES) and Tape Op shows, and that is Rupert Neve Designs or "RND" to us pro geeks (www.rupertneve.com). Rupert used to design mixing desks, as they say in his native Britain, and his designs, clones of his designs, and virtual digital models of his designs are all in high demand. I own Universal Audio versions of half a dozen of his signal processors.



Photo 6: The no frills RNHP

At the show, the stark Neve table had the new $499 RNHP, a no-nonsense half rack, 1U design, with the unusual addition of a 3.5 mm input alongside the more common balanced-on-XLR-3/TRS 1/4" combination input and RCA unbalanced (see **Photo 6**). Neve was a pioneer in specifying very wideband response in his gear to insure that phase and frequency response in the "audio" passband was spot on, and the RNHP is no exception. Frequency response is listed as an extremely tight ±0.2 dB from 10 Hz to 120 kHz.

Prism Sound (www.prismsound.com) was another pro company new to RMAF. The well-respected UK manufacturer of pro and semi-pro ADCs and DACs came to the show with CALLIA, a $2,750 desktop DAC with the same D/A conversion as its semi-pro Lyra—but with an improved, switchable gain headphone amplifier able to drive very low impedances and DSD capabilities up to DSD2 over DoP. Though CALLIA supports DXD (8× 44.1/48 kHz) on its USB input, the data is unfortunately downsampled, very nicely I'm sure, to 4× rates. Paired with ATC's SCM40As, the system sounded accurate and enjoyable.

Headphone stalwart Fostex (www.fostexinternational.com) debuted its $599 TH610, a comfortable, high-end closed back design with 50 mm dynamic drivers, walnut cups, and a slightly mid-forward voicing (see **Photo 7**). Published frequency response is claimed to extend from 5 Hz to 45 kHz. The TH610 was mated to its optional, $299 ET-H3.ON7BL balanced XLR cable.

From Sweden comes the affordable, stylish, sturdy, and good sounding JAYS line of in and on-ear cans (www.jaysheadphones.com). The u-JAYS on-ear features a 40 mm diaphragm in a closed but vented cup, and is priced at $179 for a headset version and $119 for a headphone-only version. The q-JAYS are $279 in-ears, and employ dual BAs in a tiny, all metal enclosure. They include headphone and headset cables, interchangeable by way of a custom threaded SSMCX connector—akin to a tiny F connector.

Another vendor new to the headphone herd is Labkable (www.labkable.com), a Hong Kong-based manufacturer of aftermarket cables for cans. With close to a dozen conductor/jacket choices terminated in a plethora of popular connector types, the possibilities boggle the mind.

Sony (www.sony.com) had a trio of new personal products, the $2,199 TA-ZH1ES desktop HPA, the $2,299 MDR-Z1R cans, and the $3,199 NW-WM1Z Signature Series Walkman. The latest entry in the iconic Walkman series is a blingy, oligarch friendly special edition of the NW-WM1A. The NW-WM1Z sports a gold-plated, solid-copper chassis, upgraded internal components, and four-wire Kimber Kabling inside.

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Photo 7: The TH610 from Fostex

A hybrid design, the TA-ZH1ES headphone amplifier combines Class D with an analog feed forward stage to correct non–linearities. It also includes balanced and unbalanced outs, and a USB DAC that supports double speed DXD and DSD8. Yup, you read that right. Though it might seem like a case of specmanship run amok, a colleague who used to work for Sony on the now-defunct Sonoma DSD DAW project (see Sonoma M1) told me that, with DSD, "…something magical happens at 8x."

Last but not least, I had a brief visit with Naotaka Tsunoda's newest child, the MDR-Z1R. These open-back beauties feature a dual material, 70 mm aluminum-coated polymer diaphragm. According to Tsunoda, the outer portion of the diaphragm uses a liquid crystal polymer to better cope with radial and transverse modes, selectively damping undesirable motion.



Photo 8: The groovy Leeloo combination HPA/power amplifier

Electronics

The kids at Schiit (www.schiit.com) had some new toys to ponder, a microprocessor-controlled power amplifier, and a pair of hybrid passive/active attenuators. The $699 Vidar is a dual mono, 100 W into 8 Ω/200 W into 4 Ω bridgeable design. Servo-less DC coupling, a linear power supply and microcontroller–managed bias distinguish this piece from similarly priced products. The $699 Frey and the $349 Saga are balanced and unbalanced hollow-state preamps, respectively, with the option of passive, relay–switched passive attenuation, and remote control. The Freya has an additional twist, the choice of JFET or tube buffering, and better headroom than the Saga.

Ampsandsound from Southern California (www.ampsandsound.com) teamed up with North California Sonic to show their latest, hand-built hollow-state designs, the $3,600 Agartha and the $1,850 Leeloo combination amps. The former pairs a 6SL7 up front with an overspec'd power supply and 300b triode on output. The latter is both headphone and power amp, using a 12AX7 or equivalent for input (see **Photo 8**). With a pair of EL-84s out to deliver 2.6 W into an 8 Ω speaker, little Leeloo will also drive a 32 Ω headphone with a dead quiet 1 W of hollow-state bliss. Agartha kicks out 5 W of triode power into an 8 Ω load, and also 1 W into 32 Ω headphones.

It's been a long time since Sonic Studio (www.sonicstudio.com) crossed the Great Operating System (OS) Divide but, at RMAF, it happened again…Amarra was the first and still one of my favorite dedicated music players. At the show, the Sonic guys were previewing Amarra 4, which will not only support Mac OS, but will also be cross-platform, running on Windows 8 and 10. Just in time too, as I just got a new Lenovo W10 laptop at work! There's also a new member of the family, Amarra 4 Luxe which, in addition to the existing Amarra spiffs like outstanding room correction and EQ, also adds the Amarra Hx headphone processor. Hx integrates stereo-to-binaural convolving from Darin Fong Audio right into the product.

CanJam

You couldn't move in the CanJam tent without hitting a high-resolution audio (HRA) portable. From seemingly out of nowhere, Acoustic Research

(www.acoustic-research.com) is back, with its $1199 AR-M2 and new $699 AR-M20 pocketable players. The hefty half-pound M2 features a Class A headphone amplifier, ALPS analog volume control, and dual temperature–compensated clocks. The AR-M20 is slimmer and lighter than its big brother, and bypasses the OS for better audio quality (see **Photo 9**). Running the Android operating system and supporting 802.11B/G/N, both pieces run Qobuz and TIDAL clients.

Zhuhai Spark Electronics Equipment brought a pile of its Cayin brand of gear (www.en.cayin.cn) to the tent, including hollow-state amplifiers, solid-state DACs, and its broad line of low-priced DAPs. New for the show was the $60 CS-30TCR USB-C-to-coaxial cable. The double-shielded assembly turns the new and exotic USB-C connector into analog RCA for use in a traditional setup.

Busting out of the CanJam tent, an unexpected delight was the Nagra (www.nagraaudio.com) room, with a pair of $16,000 CLASSIC AMPs, the new $17,000 CLASSIC PREAMP, a $14,000 CLASSIC DAC, and a $16,000 CDT transport. Also for sourcing, a $4,800 solid-state Nagra Seven recorder was on hand, along with a $35,000 Pro turntable and $8,500 Black Beauty tonearm from Kronos (www.kronosaudio.com). Hanging from that, a $3,750 EMT (www.emt-international.ch) S75 cartridge. What I loved though was that all that Swiss gear was playing through a pair of Avalon Acoustics $34,995 Indras (www.avalonacoustics.com), one of my favorites. All cables were Kubala Sosna Master Reference (www.kubala-sosna.com) and, if you have to ask how much those cost, forget about it!

For Entry-Level Audio Enthusiasts

Two of my favorite rooms for aspirational audio enthusiasts was ELAC's, and the "$5,000 Entry Level Room." The $5,000 room was put together as an antidote to the crazy stupid expensive and less than wonderful sounding stuff to be seen elsewhere at the show. A $1,299 Macbook Pro and a $1,199 VPI Player turntable was feeding a $1,499 Wyred4Sound mINT integrated amplifier. The amp powered a pair of $2,995 Ono Paka loudspeakers with optional but seemingly necessary $400 stands, a hybrid omni/dipole three-way design derived from Siegfried Linkwitz's LXmini (www.onoaudio.com). The Pakas, officially debuting at RMAF, are a refinement of Linkwitz's designs and, for an almost entry-level speaker, sounded really nice (see **Photo 10**).

Also affordable and highly capable, ELAC (www.elac.com) touted its new EA101EQ-G integrated amplifier as an all-in-one for both starter and second systems. With three AES inputs (RCA & optical) plus 192 kHz USB, 80 W into 4 Ω of amplification, iOS, and Android control, a headphone out and a subwoofer out with automatic setup, Dolby Digital decoding, and Bluetooth with aptX, the EA101EQ-G is good looking and easy to use. At $699, the ungainly model name is the only thing to complain about.

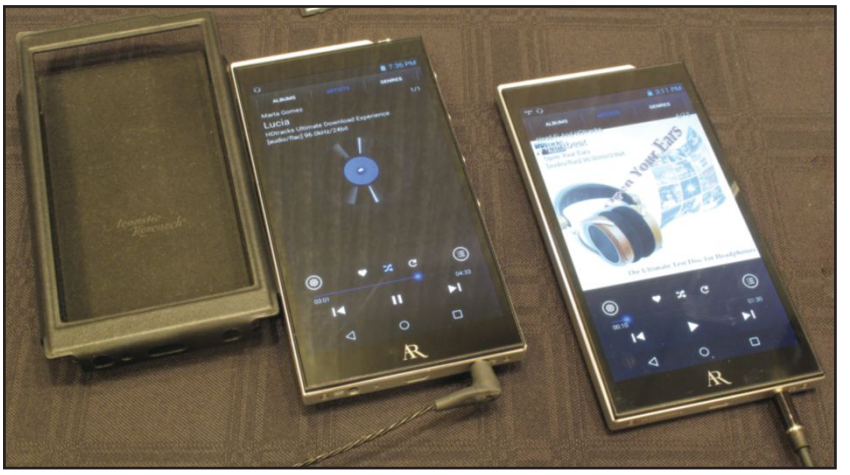


Photo 9: Acoustic Research's AR-M2

High-End Equipment

For a diametrically opposite open-baffle approach from the Pakas, you need only look to Vos Loudspeakers' Array, with its 7′ tall, dual-cabinet design (www.vosloudspeakers.com). One slim tower carries twelve 6″ cone mids plus nine AMTs in a side-by-side, open vertical array. Complementing that is a wider tower, weighing in at around a quarter ton, which houses four 12″ woofers coupled to a 14′ transmission line. With an external crossover, the entire system rings up at $135,000.



Photo 10: Ono Audio's Paka, a singular and successful approach to affordability

Photo 11: The Allegro Unum sporting a white fascia



Photo 12: The Noesis 210RT and Cherry amplifier



About the Author

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

The Brinkmann/MQA demo was lovely, with Mathias Brinkmann's new $18,000 Nyquist DAC (www.brinkmann-audio.com) into a pair of $29,900 Vandersteen 5A Carbons via $52,000/pair Vandersteen M7-HPA monoblocks (www.vandersteen.com).

Jinlang Audio Co.'s Aurum Cantus (www.aurumcantus.com) showed its V3M, a $2,000/pair two-way. With a 110 mm ribbon tweeter, and a 6.5" carbon fiber and Kevlar low frequency driver, the vented, stand mounted speaker was paired with a $3,599 Majestic from Wells Audio. The V3M is not new, but RMAF was its first showing in the US.

Unique Equipment

The affable Vinnie Rossi was excited to tell me about his new VR MINI power supply, a retrofit power supply unit (PSU) for third-party gear (www.vinnierossi.com). As with other Rossi power tech, the $995 VR MINI utilizes an ultracapacitor bank as the power reserve, charged by a special linear regulated circuit.

Since last year's show, Dave MacPherson of Studio Electric (www.studio-electric.com) has taken back manufacturing of his $2,500 two-way SE Monitors, which proved too fiddly for Benchmark. His stand-mounted passives have been upgraded with a new mid/woofer and new finish. His deco $8,950 T3 3-way floorstander was also on show, with a new 1" soft dome tweeter.

Speaking of speakers…newcomer Aluminous Audio (www.aluminousaudio.com) was showing a relatively expensive and novel slim floorstander. The $39,900/pair Gravitas is composed of a rear vented two-way in an aluminum enclosure, with stands and matching compact cubic subwoofers. The main cabinets are suspended, via webbing, from the stainless and aluminum stands. The theory is that the webbing interrupts vibration, preventing structurally–borne transmission from the hanging mass of the cabinets to the floor stands.

Another brand I had not heard of before was Sonner Audio (www.sonneraudio.com), a company hailing from New Hampshire. Its new $10,500 Allegro Unum is a stand-mounted, vented two-way, with a 1" ring-radiator tweet and a 6" magnesium cone mid/woofer that crosses over at 4.3 kHz (see **Photo 11**). A nice decorator touch is the availability of interchangeable faceplates in multiple colors.

Network audio players (NAP) continue to integrate into all levels of the audio market, and Teac's Esoteric division (www. Esoteric-usa.com) brought a higher end piece to the show. The new N-05 NAP, at $6,500, comes in toward the premium end of the category. Known for its value priced components, Audio Alchemy (www.audioalchemy.com) was showing

the $1,795 DMP-1. The new NAP is Roon Ready, and includes an I²S interface for connection to its DDP-1 digital preamp/DAC, and AES spigots for connection to other gear.

Horn-loaded compression drivers continue to be a trend, and all the usual suspects were in evidence, along with some new entries (see **Photo 12**). One was JTR Speakers (www.jtrspeakers.com), showing its $4,600/pair Noesis 210RT, a front vented three-way with two 10" woofers combined with a "ultra" fancy coaxial compression driver fitted onto a symmetrical wooden horn with 60° × 60° coverage. The 43" tall towers weigh in at 125 lbs and were powered by diminutive Maraschino amplifier from the Digital Amplifier Company (www.cherryamp.com).

NAD (www.nadelectronics.com) was showcasing its new Classic DAC/amplifiers, the $649 C 338, the $899 C 368, and the $1,599 C 388. The company also showed a new CD player, the C 568. In its high-end Masters Series, NAD previewed the $3,499 M32 integrated amplifier and the $3,999 M50.2 Digital Music Player. Where the Classic series amps use Hypex Class D technology, the M32 employs NAD's version of the more advanced Ncore, for exceptional Class D sound and performance. A jack of all trades, the remoteable M50.2 acts as a server, has a MM phono input, plays CDs, and rips to internal storage. It also includes wired GigE and wireless Wi-Fi plus Bluetooth capabilities for streaming I/O. All new members of the Classic Masters Series have expansion slots for future functionality upgrades, and TIDAL and MQA support is in the works.

The folks at Wilson Benesch (www.wilson-benesch.com) are celebrating their return to the US and, drawing a direct line to their new rep Aaudio Imports' branding, collaborated to "…create a new "Ettore" blue finish for the A.C.T. One Evolution P1." Named in honor of Ettore Bugatti, the French automobile designer and manufacturer who founded Bugatti, the $40,300 Ettore's royal blue finish is possible thanks to Hypetex colored carbon fiber (see **Photo 13**).

Exhibiting for the first time in the US, Pristine Vinyl (www.pristinevinyl.com) from the UK brought its human as well as ViVac families of record cleaning machines to the show. Pristine Vinyl has two models, the manual $2,225 ViVac RCS1 and the ViVac RCS2 which, for $2,450 uses a precision pump and automatic arm to deliver cleaning solution.

Future Events

Right around RMAF time, there was a change in the management of Western audio events. The shakeup will affect which shows I'll be attending, but it's always fun to bring you coverage. This year's RMAF get together had its good and bad aspects—a still subpar venue but that should improve before year end. Plus, far fewer jaded old men playing stale music with overpriced technology, and many more upbeat young sales people tending exhibits of innovative, affordable, and forward-looking products. All I can say is, "w00t, it's about time!"

For more information, visit www.audiofest.net.

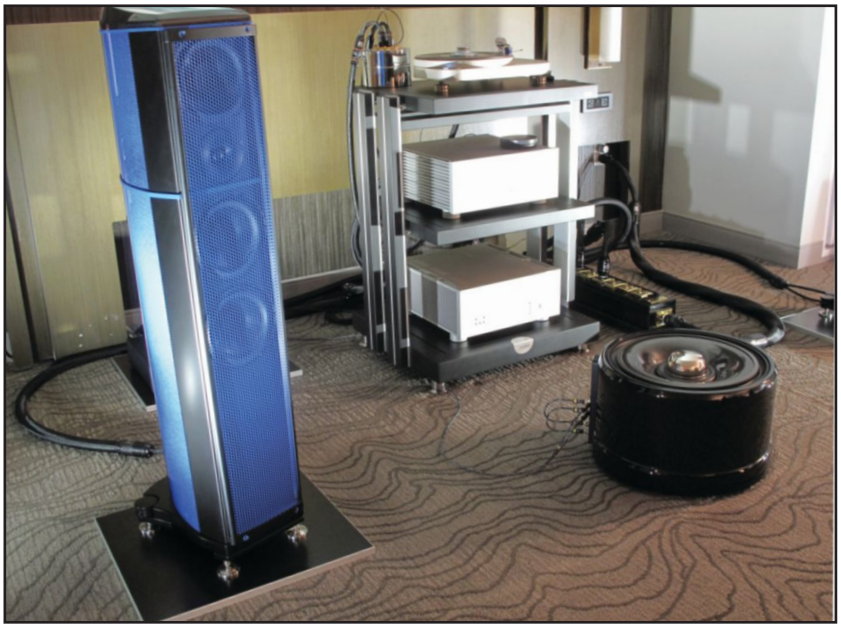
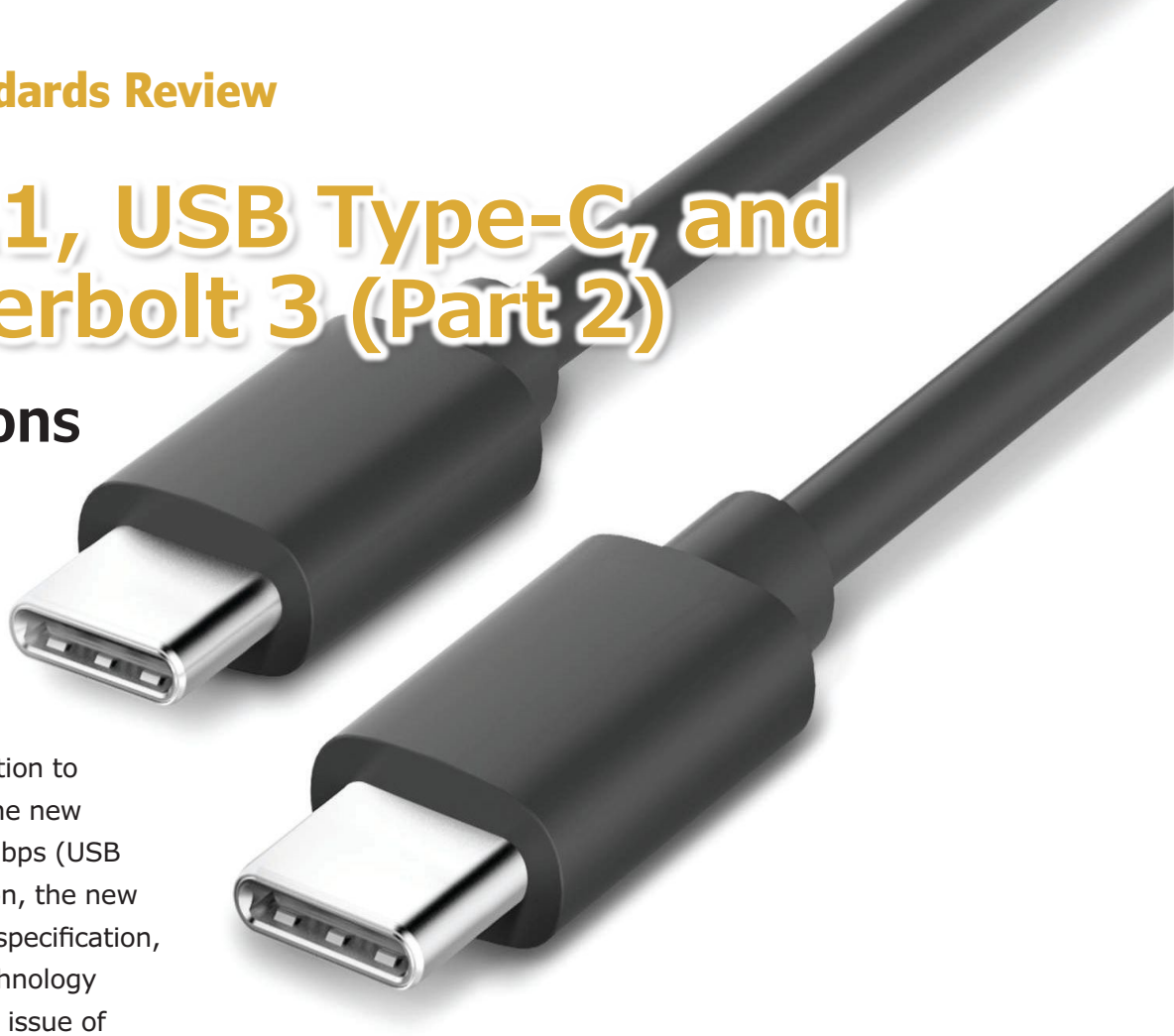


Photo 13: Wilson Benesch thinks maybe bluer is better



USB 3.1, USB Type-C, and Thunderbolt 3 (Part 2)

Implications on Audio



Following our introduction to the fundamentals of the new SuperSpeed USB 10 Gbps (USB 3.1 Gen 2) specification, the new USB Type-C interface specification, and Thunderbolt 3 technology in the December 2016 issue of *audioXpress*, we will now examine how these technologies have evolved and how they work together. We also look at the USB Audio Device Class 3.0 specification, which standardizes audio over USB Type-C.

By
João Martins
(Editor-in-Chief)

Due to the combination of the discussed specifications, and the fact that soon most computers (and all sorts of devices) will eventually combine USB-C ports—which in most cases will be Thunderbolt 3 ports, with some ports limited to USB 3.1 Gen2, mainly because of cost considerations or CPU limitations—this means the audio industry will have multiple options available to design new audio systems, all benefiting from simplified connectivity, extended bandwidth, and faster data rates.

Considering that Thunderbolt is basically a PCI Express slot (four lanes of PCI Express Gen 3) exposed on a physical connector—now a USB Type-C connector—this means that we will see an increase of audio peripherals in detriment to the typical PCIe internal cards. It also means that Thunderbolt 3 will be the ideal interface for professional audio applications. Thunderbolt 3 supports low-latency multi-track applications in high-resolution/high-sampling frequency, with the low track count production stabilizing on 24-bit/96 kHz, which is now the standard for low-channel count, consumer and high-end applications, enabling studios to work with 24-bit/192 kHz workflows without any worries.

For those who are not familiar with professional production standards, remember that even home studios have standardized on 24-bit/96 kHz production—with all the benefits for dynamic processing and track bouncing. Since that was the Firewire (1394) interface threshold, it is also now part of the Thunderbolt specification since version 1. And with consumer distribution evolving to higher resolution as well, the audio industry certainly needs to eliminate existing bottlenecks.

The same USB-C port will also enable support for all the audio-over-IP (AoIP) protocols that are currently emerging (e.g., Dante, Ravenna, Livewire, etc.), given that Thunderbolt 3 supports standard 10

Gigabit Ethernet. And of course, the same port will also support the legacy USB 2.0, USB 3.0 interfaces, and support existing applications.

Additional Information

Since we published the first part of this article (*audioXpress*, December 2016), we witnessed the expected revelation from Apple about its latest generation MacBook Pro laptops. While we are still waiting for the workstation class Mac Pro and iMac updates (and eventually the Mac Mini as well), we now know that Apple is moving forward with Thunderbolt 3.

Its new 13" and 15" MacBook Pro benefit immensely from the technology, with the only limitation being the availability of Intel processors, which still do not provide the required number of PCIe lanes. That will change in 2017, with the general availability of new Intel processors expected to power new iMacs and Mac Pros and certainly first-generation Thunderbolt 3-oriented PCs.

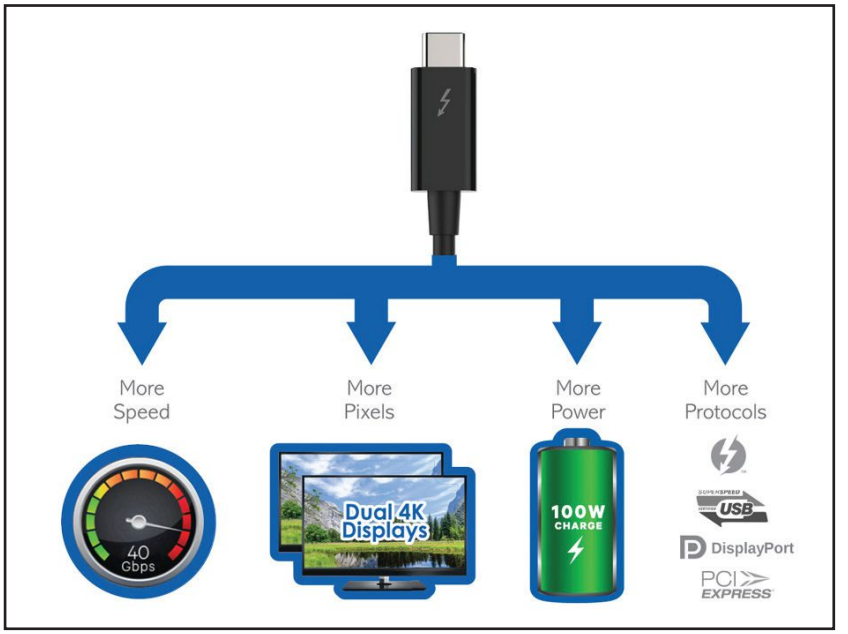
The new Thunderbolt 3 machines enable transferring data at speeds of up to 40 Gbps, which is two times faster than Thunderbolt 2, four times faster than standard USB 3.1 Gen2 ports (current standard), and eight times faster than the original USB 3 standard—which amazingly we still see being announced for machines to be released in 2017.

Also important to remember that apart from the graphics capabilities to support multiple displays (e.g., DisplayPort, HDMI, etc.) and extended resolutions (4K/5K), external Thunderbolt 3 SSD drives allow for extreme throughput. It is already better than internal SATA drives and it will keep improving.

An example, the new LaCie Bolt3 external drive (2× M.2 PCIe SSDs) uses Dual Thunderbolt 3 to achieve 2,800 MB/second read and 2,200 MB/second write speeds while the latest generation SSD SATA 3 drives (6 Gbps) are able to achieve 550 MB/second read and 520 MB/second write speeds.

For those who don't yet understand how the new USB Type-C connectors work, the idea is not to "limit" the user on the type of the devices to which they can connect or force people to use "dongles" as is the distorted perception generated following Apple's announcement.

A single Thunderbolt 3 port enables users to daisy-chain up to six Thunderbolt devices through a single port without needing a hub or a switch. The new MacBook Pro provides four ports! And for those legacy devices we still need to connect, (e.g., existing Firewire external drives, audio interfaces, and memory card readers), there are already Thunderbolt 3 docks that will provide all those



Thunderbolt 3 carries the latest generation USB SuperSpeed 10 Gbps but offers significant advantages.

options, including S/PDIF digital audio. Gradually, all users will stop carrying converters and dongles as the industry transitions to the "universal" USB Type-C connector, which was planned by the industry in general and not just an Apple decision! Of course, it will be complicated at first and some dongles will remain for more time than others (e.g., the Ethernet RJ45 to USB-C converters).

Now that this has been clarified, let's look at how USB has improved and how we can benefit from the updates, be that over USB 3.1 Gen2 or Thunderbolt 3 ports.

Putting the Puzzle Together

For this article, we have researched many



Apple recently revealed its new MacBook Pro 2016 models and, as expected, adopted Thunderbolt 3, providing two or four ports.



Windows PCs will also support USB-C and Thunderbolt 3 and that will be essential to support external graphics accelerators, 4K displays, and the latest generation fast storage.

different sources and, in general, we found Intel and Apple to be the main organizations truly committed to advancing the technology and actively promoting it. In fact, we've learned that Apple was the main force behind these advancements and the company that assigned the largest number of engineers to the consortium workgroups and to work on the specifications. After all, Apple had already greatly contributed to the Thunderbolt specification, which had remained largely an inside effort from Intel.

The technology was introduced as Light Peak at the 2009 Intel Developer Forum (IDF) and it was originally an optical interface with the potential to provide 10 Gb/s minimum speeds over optical cables. In 2010, Intel, working with "other industry partners," switched the development to electrical connections in order to supply power to connected devices and also because of cost considerations of optical cables and adapters. When the technology was introduced in 2011, Light Peak was renamed Thunderbolt and Apple brought the first MacBook Pro computers to market using a Mini DisplayPort connector, allowing multiplexing data onto a single Thunderbolt cable and to daisy-chain up to six devices, including displays.



The new LaCie Bolt3 external drive (2× M.2 PCIe SSDs) uses dual Thunderbolt 3 to achieve 2,800 MB/second read and 2,200 MB/second write speeds.

In 2013, Intel updated the spec to Thunderbolt 2, enabling channel aggregation of the two previously separate 10 Gb/s channels and combining it into a single 20 Gb/s channel. This would also allow the support for 4K video and connecting 4K displays with support for DisplayPort 1.2. Again, Apple was the first company to immediately bring products to market with Thunderbolt 2, removing existing separate Ethernet and FireWire interfaces from its machines. Other than a dedicated power connector, the only other remaining interfaces on Mac computers were USB and HDMI.

And the cooperation between Intel and Apple continued. While Apple internally developed its own Lightning interface for mobile devices, replacing its own iPod 30-pin connector (which carried both USB and FireWire, together with audio, video and controls), it was clear that an effort was being made to also reach a convergence with the USB Implementers Forum (USB-IF) and eventually allow for some form of USB compatibility with Thunderbolt.

Unfortunately, getting any information from Apple directly regarding its participation in the process proved impossible and we had to resort to the few facts from published sources and the previously mentioned technology consortia such as Mobile High-Definition Link (MHL) and High-Definition Multimedia Interface (HDMI).

We also approached several silicon companies as well as manufacturers that were actively trying to stay ahead of the curve by closely monitoring the latest specifications, engaging with the consortiums and standard bodies and experimenting with early implementations.

I think it is still important to share our conversation at CES 2016 with USB-IF executives, Brad Saunders (Architect Strategist at Intel Corporation and USB-IF chairman) and Rahman Ismail (USB-IF CTO) also from Intel, leading the audio aspects of the USB Device Working Group



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by Jean Nouvel

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While the market fully transitions to USB Type-C, we will depend upon adapters to provide connectivity with legacy equipment. Things such as USB 2.0 to USB-C, AV adapters to HDMI and DisplayPort, and even Lightning to USB-C for Apple's own iOS devices.

and the USB Compliance Committee. As it was publicly stated at CES 2016, the entire focus for the introduction of the USB Type-C connector resulted from pressures from the mobile industry. Apple generously contributed to the USB-IF efforts, while simultaneously pursuing the evolution of Thunderbolt to converge with USB.

Apple's efforts resulted in the new Thunderbolt 3 specification and adopting the USB Type-C connector. For the mobile industry, this would mean one single universal connector for charging, synchronization, connecting external displays and peripherals and for audio, replacing the 3.5 mm analog jack.



During CES 2016, audioXpress interviewed USB-IF executive Brad Saunders, who is the Architect Strategist at Intel Corp. and USB-IF chairman.

As Brad Saunders explained to *audioXpress*, the USB-IF actively engaged with all companies working on the USB Type-C spec team, from Apple, Google, Microsoft, Qualcomm, and others, and looked at all solutions to improve the specification for audio—making it possible to replace the venerable analog connector, which lead to the recently published USB Audio Device Class 3.0.

The following is an excerpt from our interview with Brad Saunders and Rahman Ismall:

audioXpress: Intel unveiled Thunderbolt 3, introducing support for USB devices and replacing the original mini-DisplayPort connector for the USB Type-C connector. How coordinated are the USB-IF workgroups with the Thunderbolt initiatives promoted by Intel and Apple?

Brad Saunders: There are connections at different levels. Obviously Thunderbolt 3 is based on using this connector (Type-C). Intel internally designed Thunderbolt 3 to run over this cable and they do it as a legitimate alternate mode. Intel to Intel or Dell to Dell as we have an example here on demonstration. So, clearly that's been done. Intel has a lot of experience at running at 10 and 20 Gbps in Thunderbolt and USB has now moved into the 10 Gbps space.

Some of the engineers who worked on USB 3.1 and USB Type-C are people that we've got from the Thunderbolt team. So, we've gained from their experience, and one of our favorite engineers happens to be a Thunderbolt person—Christine Krause—and we love what she's done for us, she's brought tremendous experience in this space. There's a lot of technology sharing going on, obviously at a level were IP is freely able to be passed, because Intel contributes the IP.

But from a planning or business perspective, they're completely independent. Thunderbolt is an Intel product and USB-C is an open-standard ecosystem.

audioXpress: So, what are the implications for USB audio?

Saunders: Well, Thunderbolt 3 carries USB.

audioXpress: So, whatever the USB-IF does on the USB side…?

Saunders: They get. Yes. If they carry USB, yes.

audioXpress: Thunderbolt has a lot of momentum on the pro audio, video post-production and broadcast industry. The gaming industry is also pushing Thunderbolt 3 to support external graphics systems, which makes sense. The consumer audio industry will not move to Thunderbolt unless they have a clear advantage, and it seems USB offers more than enough… But eventually, since they now share the same connector, people will connect one type of equipment to the other. Is there a transparency on carrying USB over Thunderbolt for audio applications?

Rahman Ismail: Yes. Thunderbolt will always carry USB now. Every solution has to be implemented so that they carry USB and so on. They can use it.

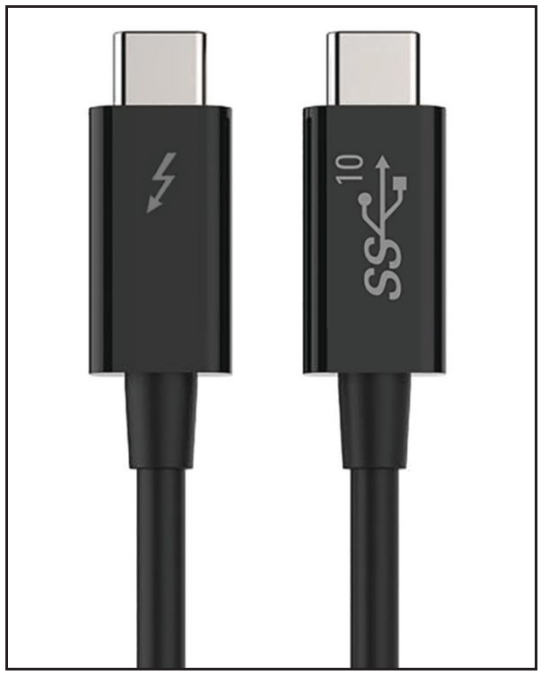
Saunders: It really depends on what's connected at the far-end. If the thing connected at the far-end is USB, and Thunderbolt is used just as an intermediate transport, you are good, because the USB functionality is carried out to the end-point, and Thunderbolt is effectively transparent.



We also interviewed Rahman Ismail (CTO) also from Intel, and currently leading the audio aspects of the USB Device Working Group.

Now, if the end-product that's connected is some sort of Thunderbolt native or PCI native then they benefit from everything that PCI does, like precision time management and mechanisms for doing that kind of stamping, which by the way are basic mechanisms at the software level, and they





USB audio will be carried over Thunderbolt 3 cables in a fully transparent way. The only difference...we will be able to use longer cables with Thunderbolt 3.

are based on a IEEE time management scheme that everybody has built-on. But you will benefit from this, even if using a Thunderbolt environment.

USB Audio Device Class 3.0 Specification for Digital Audio Applications

Now that we've explored the origins of USB 3.1 Gen2, USB Type-C, and Thunderbolt 3, it's important to examine the USB Audio Device Class 3.0 specification, announced in September 2016 and finally shine some light over how this all should work for audio.

The USB-IF, the support organization for the advancement and adoption of USB technology, completed and published the USB Audio Device Class (ADC) 3.0 specification. The specification establishes USB Audio over USB Type-C as the primary solution for all digital audio applications, including headsets, mobile devices, docking stations, gaming setups, and virtual reality (VR) solutions. The updated ADC 3.0 specification is defined by a set of four separate documents.

The first document defines in detail all the Terminal Types that are supported. The second document describes and defines all the Audio Data Formats that are supported. The third document provides the definitions for Basic Audio Functions that create a higher level of interoperability among Hosts and Audio Devices. The fourth document is the most extensive, with 152 pages, and is the ADC 3.0 specification itself, describing the minimum capabilities and characteristics of an USB audio device and providing recommendations for optional features. The specification provides a complete description of the devices or functions used to manipulate audio, voice, and sound-related functionality, including both audio data (analog and digital) and the functionality that is used to directly control the audio environment, such as Volume and Tone Control.

As detailed in the specification introduction, USB has become a ubiquitous connector on modern PCs and is well-understood by most consumers today, being used for audio transport from low-fidelity voice connections to high-quality multi-channel audio streams. But many applications that currently depend upon USB audio features were far from ideally implemented. As Saunders stated in our interview, "the latest and greatest version of USB audio is a 10-year-old definition! And the major OS in the world doesn't even support that one!"

So, to improve the situation, the USB-IF examined all areas that could be updated and improved without compromising interoperability. As it states, "In principle, a versatile bus specification like the USB provides many ways to propagate and/or control digital audio. For the industry, however, it is very important that audio transport mechanisms be well defined and standardized on the USB. Only in this way can interoperability be guaranteed among the many possible audio devices." "Standardized audio transport mechanisms also help to keep software drivers as generic as possible."

The new Audio Device Class 3.0 specification tries to satisfy those requirements and the USB-IF recommends that all device classes that address audio in some way refer to the latest documents for their audio interface specification.

Areas considered vital in the audio update include synchronization of the data streams, since "the

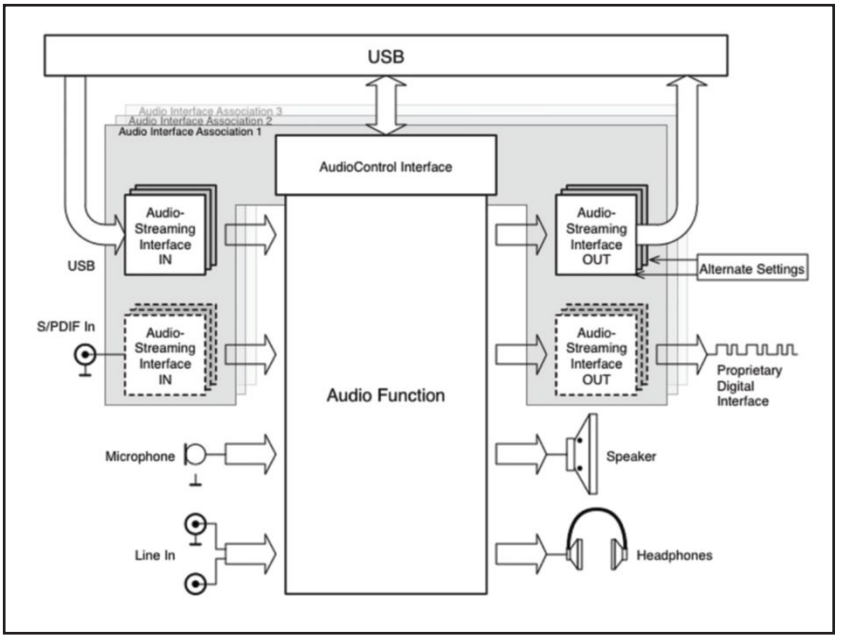
A Sampling of the Many Silicon Manufacturers and Development Platforms Consulted for This Article

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smallest artifacts are easily detected by the human ear." Therefore, a robust synchronization scheme on isochronous transfers has been developed and incorporated in the USB Specification and the USB 3.1 Specification. The Audio Device Class definition adheres to these synchronization schemes to transport audio data reliably over the bus.

The new specification also contains all necessary information for a designer to build a USB-compliant device that incorporates audio functionalities. It specifies the standard and class-specific descriptors that should be present in each USB Audio Function. It further explains the use of class-specific requests that allow for full Audio Function control, and a number of predefined data formats are listed and fully documented. Each format defines a standard way of transporting audio over the USB, and provisions have been made so that vendor-specific audio formats and compression schemes can be handled as well.

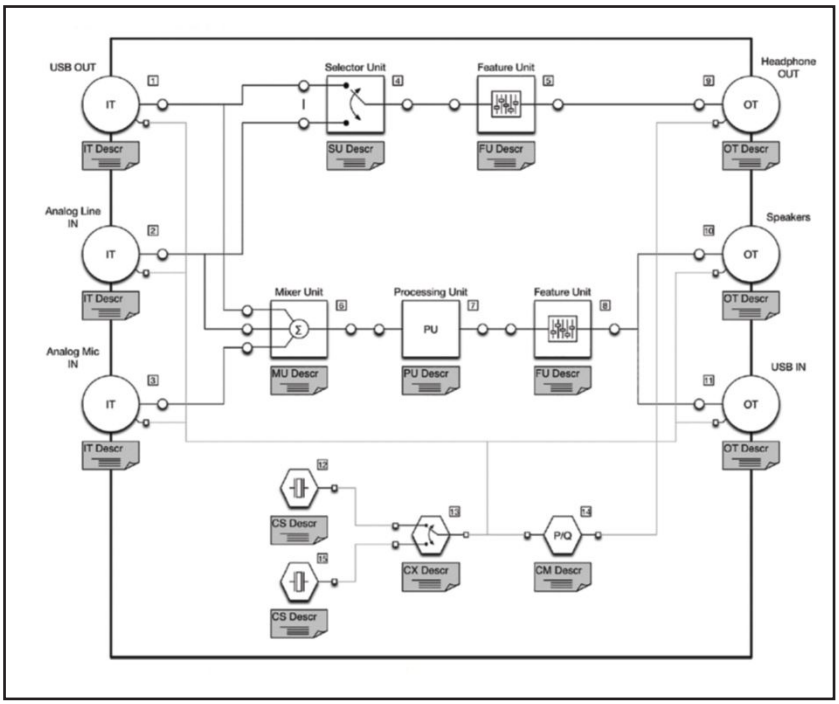
In the document, the USB-IF also states that, "many of the changes introduced in this version of the USB Specification for Audio Devices are inspired by the desire to use USB Audio in modern portable



Inside the Audio Function. The figure describes a hypothetical USB ADC 3.0 Audio Function that incorporates 15 Entities: three Input Terminals, five Units, three Output Terminals, two Clock Sources, a Clock Selector, and one Clock Multiplier. Each Entity has its unique ID (from 1 to 15) and a descriptor that fully describes the functionality of the Entity and also how that particular Entity is connected into the overall topology of the Audio Function.







USB ADC 3.0 Audio Function global view and its associated interfaces. All functionality pertaining to controlling parameters that directly influence audio perception (like volume) are located inside the central rectangle and are exclusively controlled through the AudioControl interface. Streaming aspects of the communication to or from the Audio Function are handled through separate AudioStreaming interfaces, as necessary.

devices." The specification pays special attention to how to make the Audio Device Class more power-friendly by providing new tools to selectively enable and disable parts of the Audio Function and also by supporting burst mode data transfers for longer sleep times in between data transfers.

In addition, the specification supports new codec types and data formats for consumer audio applications, provides numerous clarifications of the original specification and extensions to support various changes in the core specification.

In the provided reference list of "Key Differences Between ADC V2.0 and V3.0," the things that have been removed were mostly because they were either obsolete or no longer used. That's the case with the copy protection (S/PDIF-style) function, which was removed as well as some specific descriptors and some Type II and Extended Type II Audio Data Formats—in particular the formats that do not preserve the notion of physical channels during the transmission (typically, all non-PCM encoded audio data streams). And new Type III and IV formats were introduced—essentially new surround formats of the MPEG-4 AAC family, support for 3D audio and Ambisonic formats, and new compressed formats. In case you are wondering, the Direct-Stream Digital (DSD) format is included in the Type I list of supported data formats. Adding new Audio Data Formats to the specification is also possible by proposing it to the Audio Device Class Working Group.

One of the key aspects of the new specification has to do with synchronization parameters and specifically inter channel synchronization to respect phase differences that are applied to the different physical audio channels used to reproduce the audio source. The documents state that when dealing with audio, and 3-D audio in particular, "it is imperative that USB Audio Functions respect the phase relationship among all related audio channels. However, the responsibility for maintaining the phase relation is shared among the USB host software, hardware, and all of the audio peripheral devices or functions."

Another very important new consideration has to do with bus power. The ADC 3.0 specification now defines that a bus-powered Device that contains an Audio Function shall support the Link Power Management LPM/L1 power state. This was introduced with the USB 3.0 specification and defines a fast host transition from an enabled sleep state, allowing considerable power savings. The Power Domain allows a host to selectively reduce power consumption when parts of the Basic Audio Function are currently not being used.

There are also many new function units and descriptors detailed in the specification. To overcome the limitations of the standard Get Descriptor request (limited to maximum 256 bytes long), the new specification provides a class-specific method to retrieve larger descriptors from Audio Functions. Also, since High Capability descriptors have the ability to report changes dynamically, they can be used whenever there is a need for the descriptor to indicate that some of its values have changed (even when its length is less than 256 bytes).

It is also important to note that all MIDI-related information is now grouped in a separate document—Universal Serial Bus Device Class Definition for MIDI Devices that is considered part of the specification.

Audio Functions

Essential to the understanding of the new ADC 3.0 specification are a few concepts such as Audio Functions. To be able to manipulate the physical properties of an Audio Function, its functionality is divided into addressable Entities. Two types of such generic Entities are identified and are called Units and Terminals. In addition, a special type of Entity is defined. These Entities are called Clock Entities

and they are used to describe and manipulate the clock signals inside the Audio Function.

Units provide the basic building blocks to fully describe most Audio Functions. Audio Functions are built by connecting together several of these Units. Every Unit in the Audio Function is fully described by its associated Unit descriptor (UD). Likewise, there is a Terminal descriptor (TD) for every Terminal in the Audio Function. In addition, these descriptors provide all necessary information about the topology of the Audio Function. They fully describe how Terminals and Units are interconnected.

Listed types of standard Units and Terminals that are considered adequate to represent most Audio Functions include:

- Input Terminal (IT)
- Output Terminal (OT)
- Mixer Unit (MU)
- Selector Unit (SU)
- Feature Unit (FU)
- Sampling Rate Converter Unit (RU)
- Effect Unit (EU)
- Processing Unit (PU)
- Extension Unit (XU)

Besides Units and Terminals, the concept of a Clock Entity is introduced. Three types of Clock Entities are defined by this specification:

- Clock Source (CS)
- Clock Selector (CX)
- Clock Multiplier (CM)

A Clock Source provides a certain sampling clock frequency to all or part of the Audio Function. A Clock Source can represent an internal sampling frequency generator, but it can also represent an external sampling clock signal input to the Audio Function.

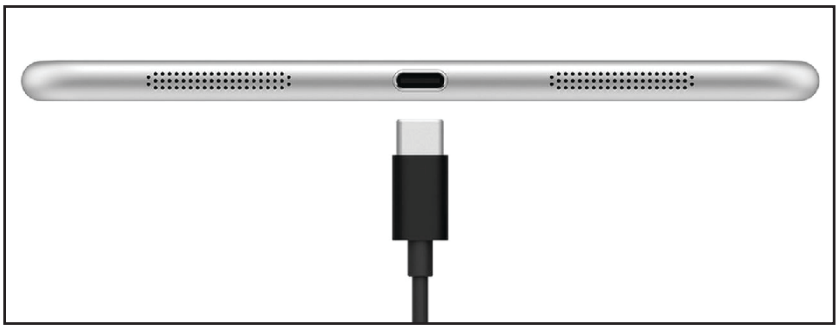
By using a combination of Clock Source, Clock Selector, and Clock Multiplier Entities, the most complex clock systems can be represented and exposed to Host software.

The ensemble of Unit descriptors, Terminal descriptors, and Clock Entity descriptors provide a full description of the Audio Function to the Host. This information is typically retrieved from the device at enumeration time. By parsing the descriptors, a generic audio driver should be able to fully control the Audio Function, except for the functionality represented by Extension Units. Those require vendor-specific extensions to the audio class driver.

For the USB specification, an Audio Function is completely defined by its interfaces and an Audio Function has one or more Audio Interface Associations (AIA)—the part that provides access to it at different compliance levels. Only one AIA can be active at one time and each AIA can only have one AudioControl interface and can have zero or more AudioStreaming interfaces.

The specification describes Function Categories for a Desktop Speaker and Home Theater systems, Microphone, Headset, Telephone, Converter, Voice/Sound recorder, IO Box,





The new Audio Device Class 3.0 specification was designed to make audio functionalities as simple as analog connections on any device, and make the entire system more power-friendly.

Musical Instrument, Pro-Audio devices (e.g. editing equipment, multi-track recording equipment, etc.), Audio/Video systems, Control Panel (e.g., a mixer panel), and Other (any device whose primary purpose is sufficiently different from the above descriptions).

Basic Audio Device Definition (BADD)

One of the most important aspects in the new ADC 3.0 specification is the mandated support for BADD (Basic Audio Device Definition, providing only supports for 48 kHz at 16 bits and 24 bits sample sizes) on each ADC 3.0 compliant device.

The purpose of the new BADD definition was to create a higher level of interoperability among Hosts and Audio Devices, by establishing a set of essential audio features and allowing the simplification of Host USB drivers. Also, this is what ensures some degree of backward compatibility with any of the previous versions of the same specification (ADC 1.0 or ADC 2.0.) and simplifies discovery and management of the different BADD profiles.

As stated in the specification documents, "The purpose of this specification is to create a higher level of interoperability among Hosts and Audio Devices. By establishing a set of essential audio features, users can expect a consistent experience, Device manufacturers have a solid template to follow, and Host drivers may be simplified."

The Basic Audio Device Definition (BADD) exists as a separate document and defines a subset of functionality commonly found in headphones, microphones, and headsets using the features and tools provided by the ADC but it removes all of the optionality allowed by the previous specifications and prescribes rigorously how certain features shall be implemented.

Conclusion

The USB Audio Device Class 3.0 specification makes it easier to support digital audio over USB, adds capabilities to reduce power consumption, and adds support for new features. It defines minimum interoperability requirements across analog and digital solutions to minimize user confusion when not all hosts or devices support audio consistently.

And with this new ADC 3.0 specification, device manufacturers will finally be able to eliminate the need for multiple ports and efficiently deliver data, power and video over a single connector with USB Type-C and USB Power Delivery.

"USB is the simplest and most pervasive connector available today, making USB Type-C the logical choice for the future of digital audio. We encourage companies interested in adopting USB specifications to take advantage of USB-IF resources to reduce time-to-market and deliver reliable USB products," stated Jeff Ravencraft, USB-IF President and COO, when the ADC 3.0 specification was published.

It is important to remember the new USB ADC 3.0 specification supports both analog and digital audio over USB-C cables and it is not so difficult to implement. So, please stop the alleged "analog jack conspiracy plot" debate.

As Saunders also told us "You can do a lot of cool stuff with USB audio...and it works as simple as an audio jack!"

Resources

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———, "Converting Existing USB Designs to Support USB Type-C Connections Application Note: USB 2.0 PHY and Type-C connectors (available through Solvnet), www.synopsys.com/Company/Publications/DWTB/Pages/dwtb-convert-usb-designtype-c-2015q1.aspx.

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High-Definition Multimedia Interface (HDMI) | www.hdmi.org

Mobile High-Definition Link (MHL) | www.mhltech.org

Sonnet Technologies | www.sonnettech.com

Thunderbolt Technology Community | thunderbolttechnology.net

USB Implementers Forum (USB-IF) | www.usb.org.

"Universal Serial Bus Device Class Definition For Audio Devices Release 3.0," USB Implementers Forum Inc., September 2016, www.usb.org

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The Ubiquitous Helmholtz Resonator



Photo 1: This photo of Hermann Ludwig Ferdinand von Helmholtz was taken in his later years.

Most people who have been around acoustics or acousticians have heard of the Helmholtz resonator. In this article, we'll discuss the extraordinary scientist for whom this device is named and examine how the resonator works and its uses.

By
Richard Honeycutt
(United States)

Hermann Ludwig Ferdinand von Helmholtz (1821–1894) was born in Potsdam, Prussia, and contributed to numerous branches of science in important ways. These include the operation of the eye, theories of vision and visual perception, the conservation of energy, electrodynamics, thermodynamics, and aesthetics, among others **(see Photo 1**). Helmholtz taught anatomy and physiology before becoming Professor of Physics at the University of Berlin in 1871.

His invention of the Helmholtz resonator, described in his book *On the Sensations of Tone* (which was first published in German in 1863) grew out of his interest in the physics of perception. **Photo 2** shows an early lab model of the resonator mounted beside an electrically-driven tuning fork that acted as an acoustical continuous-wave source.

The actual resonator is the cylindrical object on the right side of the image shown in **Photo 2**. It comprises a hollow cylinder (wood, in this case, but it can be virtually any material) with a hole in one end. Physically, the springiness of the air in the cylinder resonates with the mass of air in the neck produced by the hole passing through the wall of the device. Even if the neck is very short (thin-walled end plate on the cylinder), the so-called "end effect" causes a certain amount of the air on the inside and outside of the hole to move more or less together. The interacting air mass and springiness behave as a simple spring-mass mechanical oscillator, responding strongly at one frequency, given by the equation:

$$f = \frac{c}{2\pi}\sqrt{\frac{S}{lV_0}}$$

Where c = the speed of sound, S = the area of the hole, l = the length of the hole (thickness of wall that the hole penetrates), and V_0 = the volume of the cylinder.

The shape of the vessel enclosing the air is unimportant; only the volume of the enclosed air matters. A common example of a Helmholtz resonator is a soda bottle or cider jug (as in "a jug band") with air being blown across the hole in the top. In this case, the resonance causes the air in the neck to move with high velocity at the frequency of resonance, creating a more-or-less single-frequency tone. Accidental Helmholtz resonance sometimes occurs when a single car window is partially or fully open, resulting in a loud sound, often called "wind throb."

Early Helmholtz Resonators

Early applications of Helmholtz resonators predate von Helmholtz by many centuries. In ancient buildings from Greece to Scandinavia, urns or pots of various shapes were incorporated in the walls to boost or tame specific frequencies for acoustical benefit. In some cases, ashes have been found in the urns, presumably having been used as damping agents to increase

the effective bandwidth of the resonator. Helmholtz resonators incorporating damping act as acoustical notch filters, removing acoustical energy at the resonance frequency. The Queen Elizabeth Hall in London has 2,300 Helmholtz resonators in boxes lining the walls to reduce bass predominance caused by the massive concrete walls.

Common Uses

One of the most common uses of Helmholtz resonators is as mufflers for internal combustion engines. The specific design of the resonator affects both the sound of the exhaust and the power delivered by the engine. The Chrysler V10 engine used in the Dodge Viper and in Ram trucks uses "Helmholtz Systems" in the air intake.

Another case of Helmholtz resonators being used to remove unwanted sound is the duct silencer in HVAC systems. The fans in these systems produce a lot of low-frequency noise that is most effectively removed by use of a resonating silencer.

Helmholtz resonators are essential components of many musical instruments. One of the primary radiating modes of a violin, viola, cello, string bass, guitar, or mandolin is the Helmholtz mode. The body of these instruments encloses the air that provides springiness, while the f-hole or sound hole is the opening.

Luthiers must control the dimensions of these openings carefully in order to tune the instrument for boosted output at the proper frequency. For example, most classical guitar resonances are tuned to the "B" one octave plus one semitone below middle "C," which is a frequency of about 123 Hz. Too large a sound hole would make the resonance frequency too high, and the guitar would sound thin. Too small a sound hole would move the resonance lower, to where the guitar is a less effective radiator, making it sound less loud overall. Somewhere there is a "sweet spot," which each luthier must choose when designing an instrument.



Photo 3: The holes in an ocarina enable the player to control the resonance frequency, and thus, the pitch produced.

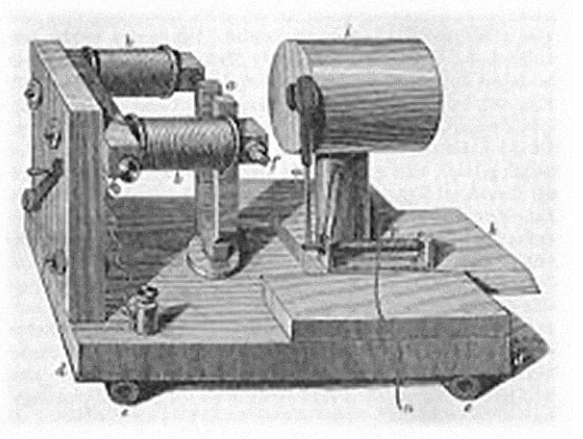


Photo 2: The Helmholtz resonator was invented as a way of measuring the frequency of sound waves.

Other musical instruments employing Helmholtz resonators include the ocarina (or "sweet potato") and the djembe drum. In the case of the ocarina, covering the holes reduces the total area of the "Helmholtz hole," changing the frequency it produces (see **Photo 3**). The djembe has a long

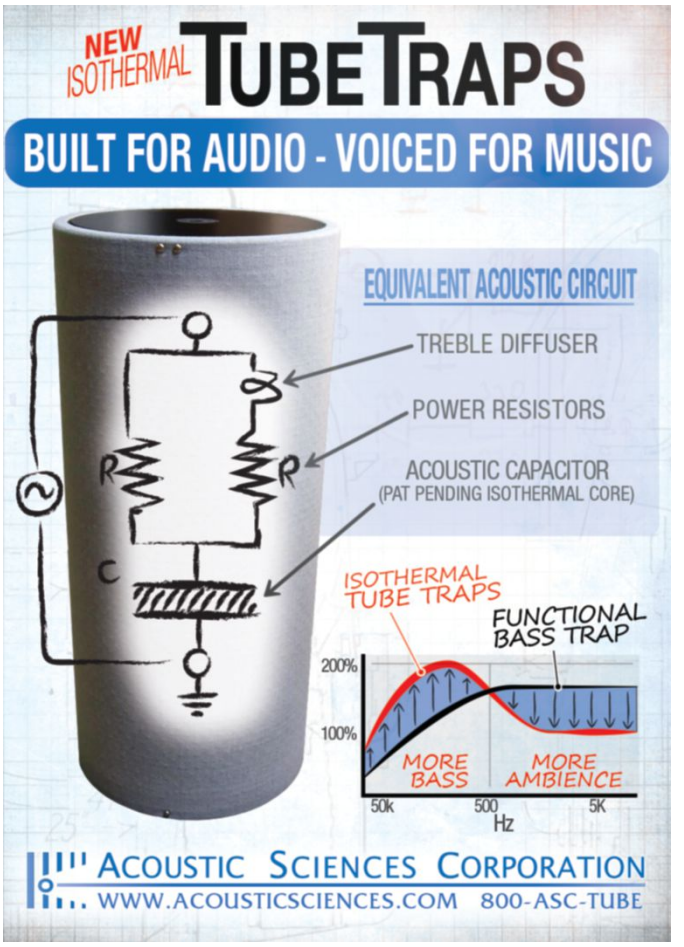




Photo 4: The long neck of the djembe gives it a low Helmholtz resonance and a deep sound.

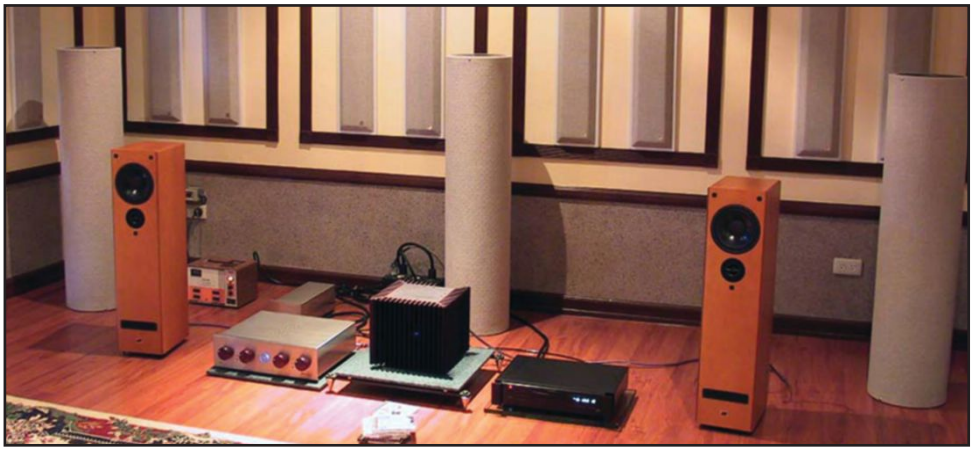


Photo 5: Bass traps often employ Helmholtz resonators. (Photo courtesy of Acoustic Sciences Corp.)

neck, giving it a low resonance and a deep tone (see **Photo 4**).

Controlling Sound Levels

Helmholtz resonators are also useful for controlling sound levels at modal frequencies in small listening rooms and studios (see **Photo 5**). In this application, the tuning frequency of the "bass trap" is chosen to eliminate the most problematic mode, and the damping is controlled in order to adjust the bandwidth of the resulting acoustic notch.

Because bass traps act locally, they can be located where modal peaks are greatest, thus having less effect on the listening locations where narrow-band attenuation is not needed. In this respect, they differ from electronic DSP notch filters, which remove energy fed to the speakers at specific frequencies, thus affecting the entire room equally.

As discussed in my June 2016 *audioXpress* article, "The Difficult and the Impossible in Acoustics," Helmholtz resonators have been successfully used to provide low-frequency absorption in small, lightweight acoustic metamaterials. Yong Li and Badreddine Assouar have developed a metamaterial using the principle of tuned absorbers.[1] The device consists of a coiled chamber placed behind a perforated panel, as shown in **Photo 6**. The dimensions are: a = 100 mm, b = 1 mm, w = 12 mm, and d = 3.3 mm.

Figure 1 shows the variation of the absorption coefficient with frequency. As you can see, the Q is very high. The coefficient of absorption reaches 100% at the tuning frequency of 125.8 Hz. The total thickness of the system is 1/223 times the wavelength of the tuning frequency. Notice how this compares with 4" thick fiberglass (1/27 wavelength), which only has an absorption coefficient of 0.88 at 125 Hz.

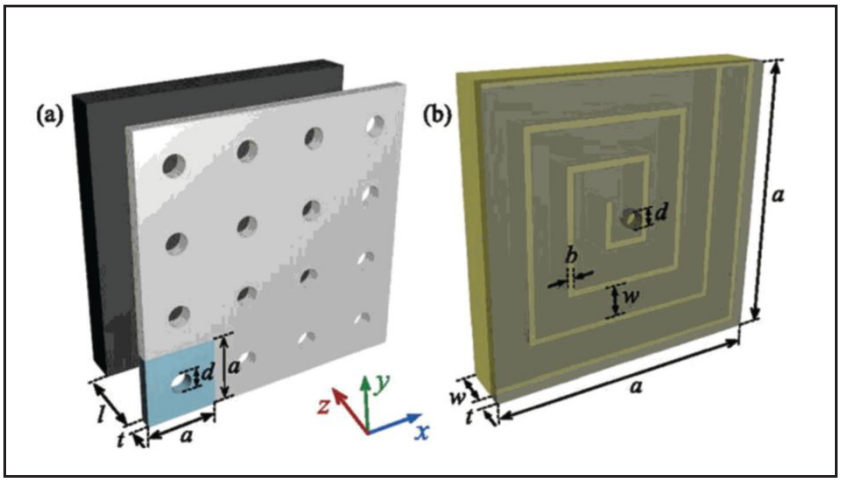


Photo 6: This metamaterial provides a high absorption coefficient, tuned to a low frequency. (Image courtesy of Yong Li and Badreddine Assouar)

Bass Reflex Speaker

In the 1940s, a speaker system design called the bass reflex speaker was introduced. The goal of the design was to improve the bass response of speaker systems without enlarging the cabinet any more than necessary. Before the 1970s, this design was used more by DIY builders than by commercial manufacturers because its operation was not well understood.

A pioneering paper by James Novak of Jensen, followed by other works by A. Neville Thiele, Richard Small, and others, brought the design of these

systems onto a level with predictable engineering approaches. Along with the new popularity of these speakers came a name change: they were now called "vented" (sometimes "ported") speakers, because "bass reflex" had become associated with the boomy sound that often resulted from poor design.

Figure 2 illustrates a vented speaker system. As you can see, the cabinet functions as the enclosure of a Helmholtz resonator, and a tubular port or vent acts as the hole. By controlling the enclosure volume, cross-sectional area of the vent, and vent length, the designer can control the "box frequency."

If this frequency is correctly coordinated with the electro-mechano-acoustical resonance frequencies and "Q's" (quality factors) of the driver, predictable results can be obtained in the finished product. The detailed technical analyses provided by Novak, Thiele, Small, and others has been distilled into user-friendly form for both DIY-ers and professionals in Vance Dickason's *Loudspeaker Design Cookbook, 7th Edition*.

The passive radiator speaker is a slightly different but conceptually similar design. It uses a diaphragm much like a speaker cone, but with no voice coil, instead of the moving air mass of the Helmholtz resonator. The passive radiator (often called a "drone cone" in years past) was introduced in 1954 by Harry Olson, John Preston, and Everrett May. The additional mechanical and acoustical complexity of the passive radiator compared to a vent port or simple vent allows some additional options for the speaker designer.

For those readers who have studied physics, this survey of Helmholtz resonator history, operating principles, and applications should provide verification of your physics teacher/professor's emphasis on the importance of understanding the simple spring-mass oscillator!

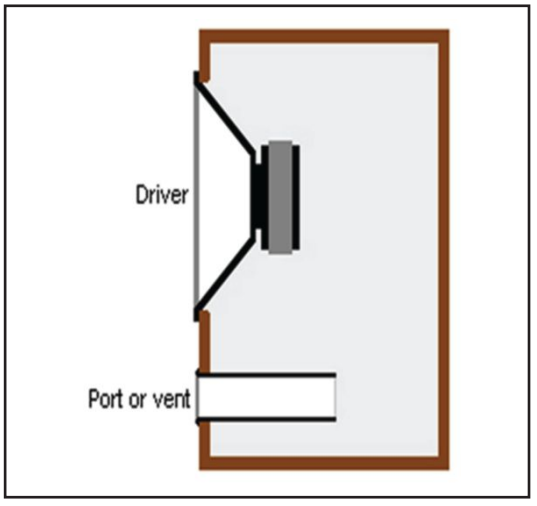


Figure 2: The vented speaker system incorporates a Helmholtz resonator. (Image courtesy of Wikipedia)

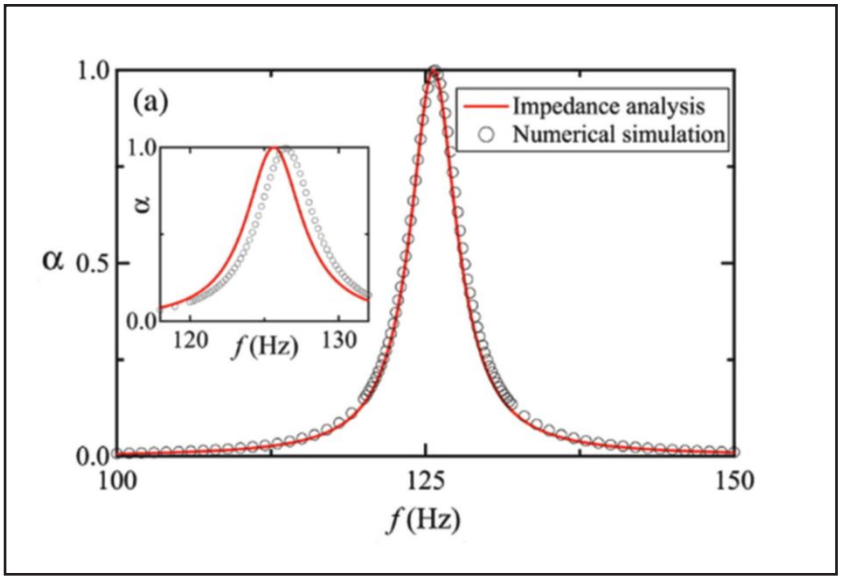


Figure 1: This metamaterial achieves 100% absorption at a frequency of 125.8 Hz. (Image courtesy of Yong Li and Badreddine Assouar)

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Kii Three

The Next Step in Loudspeaker Evolution



Photo 1: Compact, modern, and classic, the Kii Three will find a place in any interior.

In this review we are going to find out if Kii Audio's Kii Three is really the most advanced loudspeaker on the market today, as is claimed by the designer. So, our focus will be on the technical aspects of the loudspeaker first and foremost.

By
Jan Didden and Paul Wilke
(The Netherlands)

Editor's Note: No loudspeaker review can be complete without listening impressions and, therefore, there will be those as well. Jan Didden has lived with the Kii Three for a week, and will give his listening impressions at the end of this review. Paul Wilke has done the technical write up. The measurement session in the anechoic chamber of the Technical University in Delft was a joint mission.

Manufacturer Specifications

- Compact DSP controlled monitor speaker 4 × 6.5″ woofer, 1× 5″ midrange, 1″ waveguided tweeter, all individually driven
- Amplification: 6× 250 W full-custom Ncore (power consumption in standby < 0.5 W)
- Active Wave Focusing crossover filter frequency response: 20 Hz to 25 kHz ±0.5 dB
- Phase response: Minimum (best possible time coherence)
- Long term SPL: 105 dB
- Short-term SPL: 110 dB
- Peak SPL: 115 dB;
- Controlled Directivity: 4.8 dB (80 Hz to 1 kHz, slowly rising thereafter)
- Size: 20 cm × 40cm × 40 cm (8″W × 16″H × 16″D) Weight: 15 kg (33 lbs);
- Inputs: Analogue, AES/EBU
- Selectable correction for free-standing, near wall, or in corner
- Latency: normal mode = approx. 90 ms; low latency mode = 1030 µs (1.03 ms)
- Suggested retail price: €10,000/pair (approximately $10,765). Matching stands €1,500/pair (approximately $1,615).

The Kii Three (see **Photo 1**) is the latest design by Bruno Putzeys, CTO, Kii Audio GmbH and perhaps the leading audio designer of this age. To make the Kii Three design even more unique, he teamed up with a group of experts that included DSP specialist Bart van der Laan (see **Photo 2**).

By means of introduction to those who are not yet familiar with this name: Putzeys is known as the designer of the nCore series of Class-D amplifiers, which are leading in their field and rival the sound quality of even the best Class-A amplifiers, at a fraction of their power consumption and weight. Another of his revolutionary designs is a digital-to-analog converter (DAC), the distortion of which is unmeasurable with present day technology. So, what happens if this audio genius wraps his brain around loudspeaker design in an all-out way?

Putzeys has designed other loudspeakers (e.g., the Grimm LS1, which is excellent, but still follows a topology known from other loudspeaker manufacturers). Speakers such as the LS1 with DSP corrected frequency curves are well accepted, certainly



Photo 2: The Kii Audio team: Bruno Putzeys, CTO, Bart van der Laan, COO, Wim Weijers, Production & QC Manager, Chris Reichardt, CEO, and Thomas Jansen, Product Manager.

in the professional market for studio equipment. The Kii Three, however, is a completely different kind of loudspeaker. It is so different that it may be the beginning of an entirely new class of loudspeakers: Those that actively control directivity at the low end.

The Kii Three and Its Technology

The Kii Three is not a large speaker, measuring 20 cm × 40 cm × 40 cm (8″ × 16″ × 16″). At 15 kg (approximately 33 lbs), it is not exactly a featherweight but still relatively easy to handle. With the small side facing the room, it hardly looks larger than a book shelf monitor (see **Photo 3**). The Kii Three is also available in a variety of colors.

It is little short of amazing what has been packed into this small rectangular case. To begin with the amplifiers: There are six of them, each a solid 250 W strong nCore Class-D amplifier, customized from the series Putzeys designed for Hypex. Each amplifier feeds its own loudspeaker, two woofers, two mid woofers, one midrange, and a tweeter with a diffraction wave guide. All this is driven by four channels coming from a complicated DSP processing unit, which is at the heart of some of this loudspeaker's novel characteristics. As befits a revolutionary loudspeaker, almost none of these components are run of the mill. Everything is at least somewhat different in more respects than can be mentioned in this article, so we have only provided a brief sketch of some of the main features. However, the Kii Three's entire manual has been included in the Supplementary Material found on the audioXpress website (www.audioXpress.com)

Enclosure

The case consists of two identical halves of molded high-density polyurethane, joined together in the middle through an impressive labyrinth of grooves and notches. On both sides, heatsinks are mounted to which the amplifiers are connected. The power supply is housed below the top plate, which also functions as a heatsink. The connectors are housed in a recessed space on the rear, so that plugs do not stick out. The bass and mid bass drivers are not protected by a grill, but the tweeter is, and the mid driver is covered by a flexible cloth front. Finished in a glossy paint job, the enclosure is modern and classy. The polyurethane case appears to be well damped.

Connections and Controls

The Kii Three has an analog XLR input so it can be connected to any balanced analog source (see **Photo 4**). It also has a digital AES input, using the same XLR connector. In this case, only one digital stereo connection is made between one speaker and the source, and the two speakers are connected with standard UTP CAT5 cable. Kii Audio is working on a control unit so that both speakers can be driven through CAT5 cable.

In a way, despite all its groundbreaking modernity, the Kii Three follows a well-established tradition by Putzeys's first employer, Philips. This company produced active speakers between 1970 and 1989 with motional feedback (MFB) technology, but this was not the only revolutionary feature at the time. What Philips realized then, and it is good to see this same idea return in the Kii Three, is that a speaker can be perfect in an anechoic room, but can still sound lousy in a real environment. The reason is that all rooms are different, and "one size fits all" is just impossible.

Real listening rooms have varying degrees of absorption, different amounts of room gain at lower



Photo 3: The Kii Three is a six-way design, with the front, the side, and the rear working together to throw the sound in one direction only without relying on a baffle. A tweeter with integrated diffraction waveguide and a 4″ mid-driver hidden by a black cloth grill in the front, with four woofers with a convex aluminum surface set in an inverted surround.



Photo 4: Kii Three's rear connection and adjustment panel.

frequencies, and on top of it all, the speaker placement can strongly impact the frequency response as measured in the room. Therefore, to match a speaker to a room and get the optimum result, some form of tone control ought to be applied.

The Kii Three, the Philips of old, and an increasing number of active speaker brands follow the philosophy that these controls ought to be part of the loudspeaker proper. The authors of this article agree with this point of view. The simple reason for this is that even the two speakers in a stereo setup may, through their different placements in the room, require different amounts of correction. Since preamplifiers, which allow split tone controls for both channels, are rare, the speaker itself is the right place to make such corrections.

The Kii Three has two kinds of control to match the loudspeakers to the room. The first one, the so called "boundary" dial, is specifically geared toward getting the right amount of bass in the room. It ranges from "free space," where the frequency response would measure flat in an anechoic room, to "corner," for situations where the room provides maximum bass reinforcement. The second control is a "contour" dial that provides a different setting on the low end, but also enables the user to increase or decrease the higher frequencies. For readers familiar with high-end Tannoy loudspeakers (e.g., the K3838), it will be no surprise that such controls on a loudspeaker can be very useful. Thus, the Kii Three can be adjusted to rooms with different amounts of absorption.

Next to the XLR input connector is a diminutive small button with large impact. With it, the latency can be switched from high (90 ms) to low (1.03 ms). Now, this requires a bit of explanation. All DSP processing takes time. So, a loudspeaker with a DSP onboard always plays its input with some delay, and the more processing required, the more the delay. For most uses, a 90 ms delay is inconsequential (e.g., while listening to some recorded music). However, for some uses (e.g., live monitoring or using loudspeakers in an AV-setting), latency is unwanted and can even ruin the intended use. The small push button leaves the choice up to the user.

In the low-latency mode, the Kii Three is a perfectly fine speaker, with all the bells and whistles such as actively controlled directivity on the low end. But, it does display the 360° phase shift per driver that is innate to LR4 filtering. To roll back this phase shift, a so-called FIR-filter is required. And this is what that little push button does. You can go from minimal phase and a 1.2 ms latency to linear phase with 90 ms with a push of this button. It switches the FIR filtering functionality, which provides for the phase correction, in or out of the chain.

Amplifiers

The nCores have gained enormous popularity with audio enthusiasts and professionals, so what could possibly be made better? According to Putzeys, distortion of loudspeaker drivers can be optimized by varying the amplifier's output resistance for different frequency ranges. This requires an amplifier that is specifically tuned to the parameters of the loudspeaker driver. An amplifier like this can never be a commercial good-for-all piece of equipment. They have to be tailor made, and this is what was done for the Kii Three.

With 6× 250 W of amplifying power, headroom is never a problem. But they do generate a bit of heat. With Class-D amplifiers dissipating about 2% of their maximum output on standby, the Kii Three's transform about 30 W into heat on standby. This is enough for the enclosure sides to become warm to the touch.

Loudspeaker Drivers

The loudspeaker drivers in the Kii Three use conventional ferrite magnets, which is not a bad thing given that loudspeakers need a certain weight to reduce the effect of the reaction forces caused by

About the Authors

Jan Didden has written for *audioXpress* since the 1970s. He is retired following a career with the Netherlands Air Force and NATO. He worked in logistics, air defense, and information technology. Retirement has provided him with the time to finish all the audio projects that have piled up for decades. He writes about them on his website linearaudio.nl. Jan is also the publisher and managing editor of the twice-yearly bookzine *Linear Audio*.

Paul Wilke has been involved with electronics and loudspeaker development for more than four decades and has been active mainly behind the screens, which are now slowly opening.



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Photo 5: The Kii Three packs 1,500 W per speaker divided across the six drivers into a compact enclosure.

the accelerating and decelerating moving parts of the drivers.

For the highs, Kii Three appears to have the same tweeter as the one used in the original Grimm LS1. What is special about this design is the diffraction waveguide in which it is mounted. Normally, waveguides are used in loudspeakers to match the directivity of the tweeter to that of the mid-range in the crossover region. In other words, the waveguide's main effect is on the lower end of the tweeter spectrum.

With the Kii Three tweeter, this effect is there as well, but the waveguide's special shape also has an effect on the high end of the tweeter spectrum, from 7 kHz to 20 kHz. Normally, tweeters start to "beam" at these higher frequencies. The Kii Three's diffraction waveguide has two sharp discontinuities, which should cause the higher frequencies to spread out more. We will see in the measurement section if they really do.

The mid-driver is hidden from view by a black cloth grill, so little can be said about it here, except that it is a 10-cm (4") driver with a conventional cone-dust cap construction. The four woofers are a unique design, but which make a lot of sense in a compact, high-power design such as this. From the outside, all you can see is a black convex aluminum surface set in an inverted surround.

Putzeys explained that the voice coil is connected through an aluminum spreader to the cone. This is good for thermal management. Heat from the voice coil can travel to the cone for dissipation. Heat produced inside the case by the power supply and amplifiers can escape through the thin aluminum barriers created by the cones between the inside and outside air—a very clever solution.

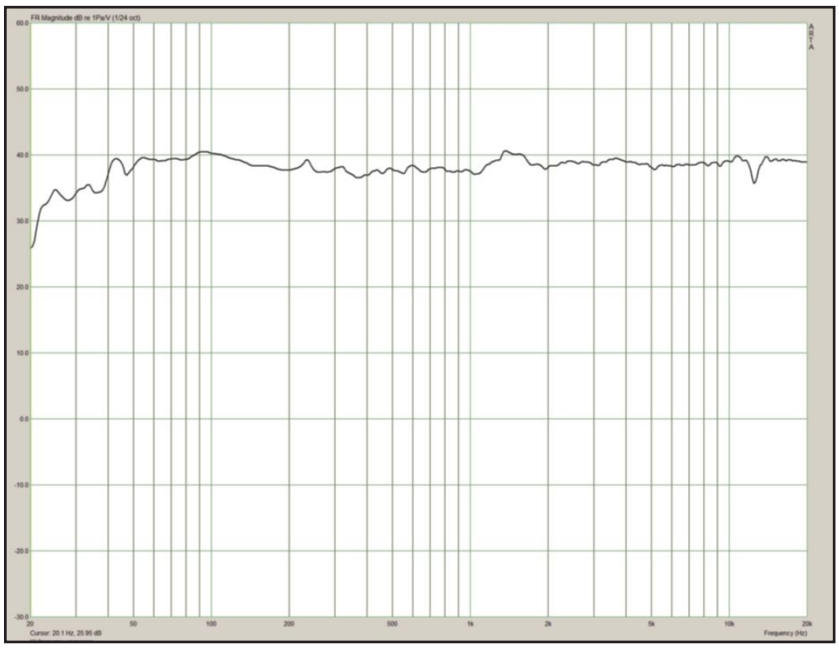


Figure 1: The Kii Three frequency response is flat between 30 Hz and 20 kHz.

Protection: How to Prevent Melt Down

Putzeys mentioned that the Kii Three's switched power supply is capable of delivering 1,250 W, (1.5 hp for those more accustomed to ancient measurements). Since small drivers such as those used in the Kii Three have efficiencies of only a fraction of 1%, almost all of the electrical energy entering the Kii Three will be turned into heat. It is obvious that the small heatsinks, even with the clever thermal solution incorporated in the bass drivers, would never be able to transfer this much energy to the outside world on a continuous basis.

The 1,250 W is definitively the peak performance, which should be fine with normal use (see **Photo 5**). In music, the peaks are often more than 10 times higher than the average power level. This so-called "crest factor" is not always that high. Certainly modern music may be very compressed and have such low crest factors that the Kii Three needs protection from melt down. Protection against overheating and over current is nothing new, and different methods have been applied. In their roughest form, these circuits just shut off the loudspeaker until, for example, the temperature has sufficiently decreased. More sophisticated systems do not completely switch off, instead they limit the maximum signal level allowed, compressing or clipping the music signal. This tends to be rather audible, in general.

Welcome to Putzeys's mind. The protection scheme of the Kii Three works more in the background than the systems mentioned earlier. We will spare you the

technical details, but what it amounts to is a small, dedicated sub-system outside the audio-loop that is constantly looking at the signal coming in. When it concludes that an "illegal" value is presented at the input, it uses the delay of the DSP-filter to rapidly adjust the filter slopes in the crossover so that no damage will be done. The system obviously works. We have heard it playing very loud, in large rooms, without ever hearing protection coming in or the loudspeaker playing itself to pieces.

Measurements

As may be expected from a DSP-enabled loudspeaker, the Kii Three's frequency response curve is relatively flat, running from about 30 Hz to slightly above 20 kHz with minor deviations (see **Figure 1**). Most visible are a slight bump around 1,500 Hz and a depression at 12.5 kHz. According to Putzeys, these are due to component variations, which are not taken care of by the DSP correction. This might be the case to some extent, but there also seems to be some energy storage issues at play, since the same irregularities show up in the cumulative decay plots (see **Figure 2**).

In spite of this, it is fair to say, based on these measurements, that the Kii Three is a neutral speaker, which does not add coloration to program material. As a matter of fact, the world needs more speakers like these, especially in places where they would matter (e.g., control rooms in recording or broadcasting studios). These are not our words, but those of Floyd Toole, the well-known scientist in residence who helped transform Harman International into a leading loudspeaker company.

The problem is that we need standards in recording and broadcasting, and those standards are lacking at the moment. The result is that the sound quality of recordings and broadcasts are highly unpredictable when they reach the listening rooms of the consumers. For example, mixes from a bass heavy studio might sound "thin" when reproduced on monitors with a flat frequency response. There is a small but increasing number of loudspeakers that would qualify to be used as such a "standard" in production control rooms. The Kii Three is a welcome addition, with a novelty: actively controlled directivity on the low end.

From the phase plots of the measurements without the FIR-filter switched on, the Kii Three works with fourth-order Linkwitz-Riley (LR4) filter slopes on all drivers, which was later confirmed by Putzeys. When asked why he did not opt for steeper filter slopes, which are easy to implement with DSP-technology, he answered that LR4 in his view is optimum.

At lower filter orders, each driver has to span a frequency range that is simply too large. Either the drivers cannot cope, or the wide region where they overlap causes problems, or both. Higher filter orders just do not sound as well because they lead to sharp transitions between different listening locations. The measurement shown in **Figure 3** demonstrates that the actively controlled directivity does work as advertised.

In **Figure 3**, the loudspeaker's frequency responses taken at different angles are combined and the color codes indicate intensity for easy reference. The microphone was moved from the front of the speaker all the way to the rear, providing a 360° view.

It can clearly be seen that the Kii Three has an extremely well-controlled directivity that extends from about 80 Hz all the way to 20 kHz. In order to achieve this, the bottom and top ends are the most challenging, but in completely opposite ways. On the low end, the long wavelength causes the sound wave coming from the bass driver to wrap around the loudspeaker. The result is that as much sound energy is radiated from behind as from the front of the loudspeaker. On the high end, as mentioned, the

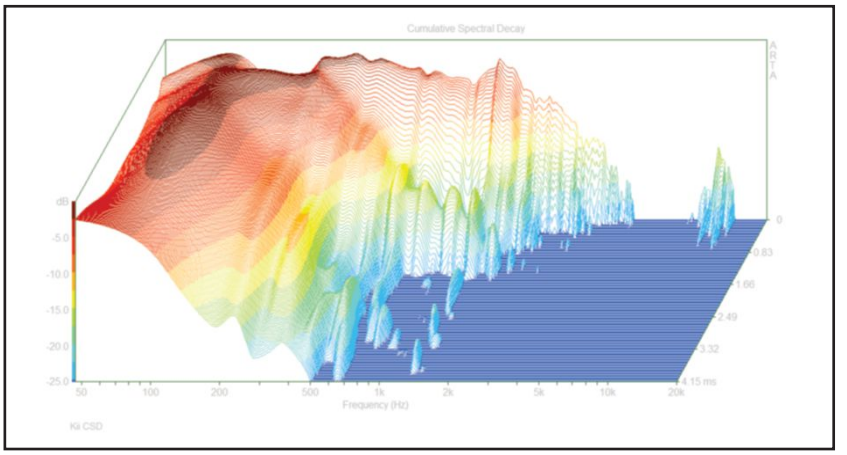


Figure 2: Kii Three's cumulative spectral decay (CSD) plot shows a well-controlled behavior.

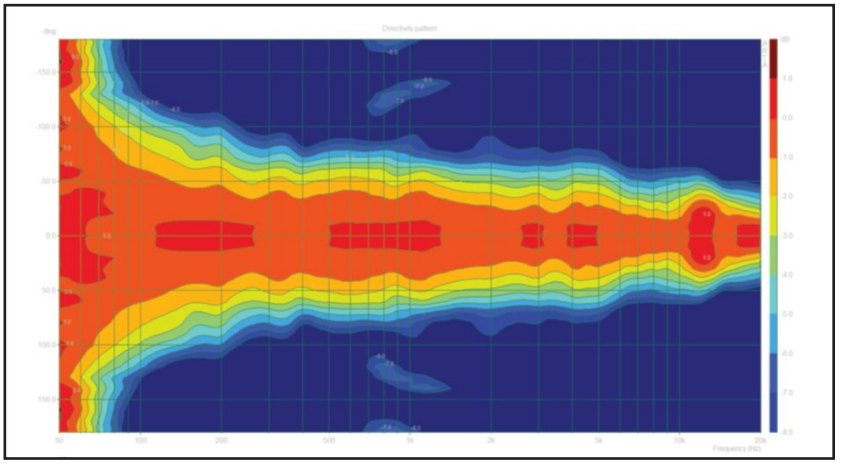


Figure 3: Kii Three's directivity vs. frequency shows well-maintained directivity control down to below 100 Hz.

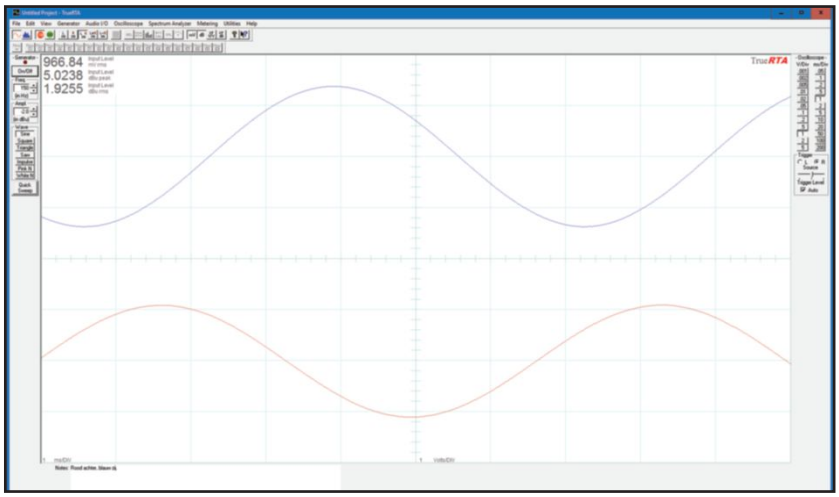


Figure 4: DSP-controlled variable phase shift between rear- and side-firing woofers controls directivity to below 100 Hz.

challenge is the opposite. The higher the frequency, the more the sound radiated by a driver begins to beam into an ever-decreasing angle.

Now, why is it so important to control directivity to this extent? The answer falls into two parts. Let us begin with the lower frequencies. Normal loudspeakers start to lose all directivity control around 200 Hz or higher. That means below that frequency, they radiate as much energy to the front as to the back of the speaker. Since the speaker is normally in a room, there are walls, and these walls reflect the sound coming from the rear of the speaker. This reflected sound mixes with the original sound wave, leading to reinforcement of some, and diminution of other frequencies. The technical term for this is comb filtering, since this pattern of reinforcement and diminution looks a bit like the teeth of a comb. Since this pattern differs from location to location in the same room, it is impossible to equalize it away by electronic means.

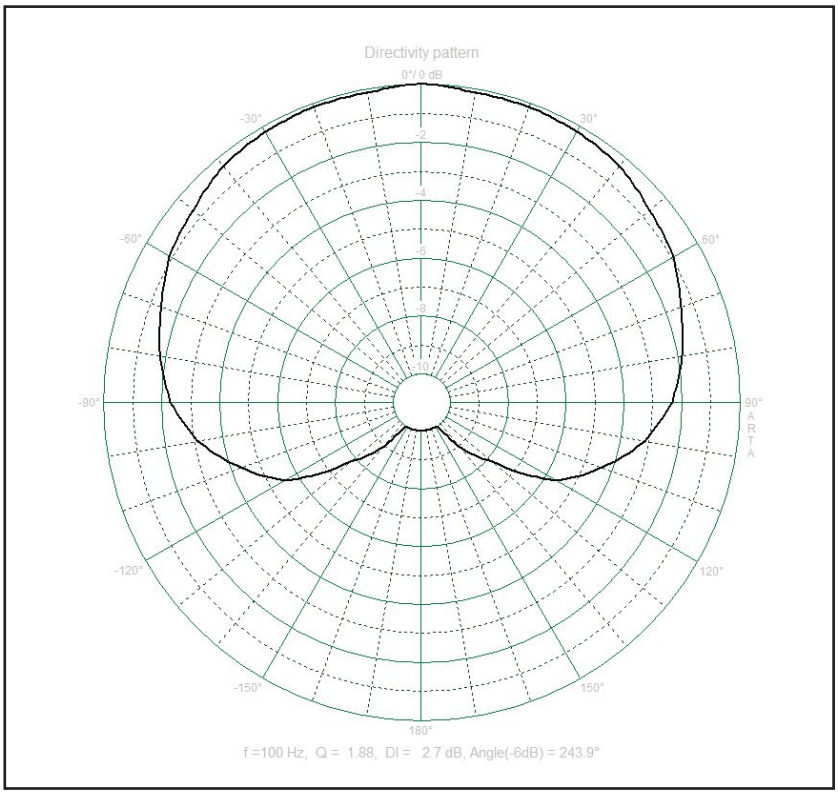


Figure 5: Kii Three's directivity pattern at 100 Hz is an almost perfectly cardioid.

A good way to get rid of comb filtering would be to make the walls behind the speakers less reflective. However, this is impossible to achieve for the lower frequencies in any practical way. The layer of sound damping required would simply be too large; think in terms of meters here. Therefore, the only way to really improve this comb filtering is to ensure that these lower frequencies do not radiate from the rear of the enclosure, and this is what the Kii Three does.

The advantage of controlled directivity at the high end is, simply put, that it increases the listening angle, creating a larger "sweet spot" in the room. The Kii Three has innovative solutions for the directivity issues on both extremes of the frequency scale. The directivity plot shows how effective the diffraction waveguide on the tweeter works. The Kii Three tweeter maintains a much wider dispersion than normal tweeters would. This is controlled directivity at its best.

As you can see in **Figure 3**, there is a bulge in the directivity plot around 12.5 kHz. Before that bulge, the tweeter starts to radiate in an ever-tighter beam, and a normal tweeter would continue to become narrower with climbing frequency. However, the tweeter of the Kii Three does no such thing; it maintains a relatively wide beam right up to 20 kHz. It clearly works, and the energy storage issue mentioned before may be the price to pay for it. In loudspeaker development, usually all things come at a price.

The control of the directivity between 80 Hz and 250 Hz is absolutely astounding from such a small loudspeaker. Many of our technically inclined readers, which most obviously are, will now wonder how the Kii Three achieves this directivity at low frequencies. It does so by sending different, but carefully aligned signals to the side and the rear woofers. The oscilloscope capture shown in **Figure 4** displays the phase relationship between the rear (red) and side (blue) woofers.

Two microphones where used simultaneously. One microphone was placed at a close distance from the rear speaker (red channel) and the other at an identical distance from the side speaker (blue channel). As can be seen, at 150 Hz, the rear woofer leads the side woofers with about one-quarter of the sine wave reproduced.

Now, the distance between the rear and side woofers comes into play. By the time the sound

wave of the rear woofer reaches the side woofer, the traveling time assures that they are in phase and will reinforce each other. On the rear side, the opposite phenomenon occurs. By the time the sound of the side woofer reaches the rear woofer, the phase has become opposite and both waves will interfere destructively. So, at the rear, one has null or little output, and at the front, reinforcement and, thus, a maximum. The polar diagram shown in **Figure 5** details how well this works, the radiation pattern is indeed a cardioid. The line shows at which points in a circle around the loudspeaker the power response is down by 10 dB, around 10 times.

As can be seen, at 100 Hz, directivity is well contained and the result is a textbook cardioid radiation pattern. Somewhat below that frequency, around 80 Hz, the Kii Three starts to let go. At those low frequencies, the wavelength reaches such a dimension that room modes become the dominant spoilers anyways. A room mode requires reflections by at least two walls, so limiting acoustic output in the direction of just one wall is not that effective anymore (see **Photo 6**).

In conclusion, the Kii Three functions as stated by its manufacturer. Its claim to be the most advanced loudspeaker on the market appears to be right on the mark. Will the market be ready for it? We very much hope so for Putzeys (see **Photo 7**) and his development team, because the bold move they made with this loudspeaker deserves proper recognition.



Photo 6: The Kii Three uses what Bruno Putzeys calls Active Wave Focusing technology to control the sound dispersion and direct the sound pressure where it needs to be.

Listening—The Kii Three Experience

I first heard the Kii Three in an invitation-only session at an audio show in the fall of 2015. That experience really got to me. I am normally somewhat reserved at "auditioning" events, knowing that all the stops would have been pulled out to make me believe this was the best audio thing ever. I am also very much aware of the difficulty of listening to a speaker ears-only, with no biases or prejudices involved—an almost impossible order.

Yet, despite my reservations and alarm bells, the Kii Three made quite an impression. Clean, clear, extremely detailed, and tonally very well balanced. The demo was quite loud, which I normally don't like, but here is a speaker system that plays as loud as you want with seemingly unconstrained power reserve. I actually liked it playing loud because it was still sounding very clear and clean. A bit unsettling.

Initially, I thought it missed something in the bass region, until designer Putzeys played some music with real bass. This was the first time I heard low bass notes that were as controlled as what we would expect from low mid- and mid frequency notes. We have all heard speakers where the bass and the rest of the



Photo 7: Bruno Putzeys was the mind behind the invention of UcD and Ncore class-D amplification systems. With Kii Audio he turned his full attention to active loudspeakers.

system seemed to be two different systems, with the low end somewhat arbitrarily throwing low-frequency energy into the room while the mid/high had control and distinctive instruments and notes. Here was a system where the bass was just as controlled and distinctive as the rest of the spectrum. I used the term "arbitrary bass" earlier—this was "decisive bass." Sorry, I just don't have a better vocabulary.

We were granted access to the anechoic chamber at Delft University to test the Kii Three for ourselves (see **Photo 8**). The Kii Three has controlled directivity down to 80 Hz or so, where most other speakers have long become omnidirectional. That omnidirectionality means that a lot of the low-frequency energy ends up anywhere in the room, exciting what is known as room modes and destroying any sense of control in those lower registers. Wilke has already explained this better than I can, detailing how the Kii people pulled this off.



Photo 8: The authors, Jan Didden (a) and Paul Wilke (b) are shown during the measurement sessions in the anechoic chamber at Delft University.

When you listen to a pair of Kii Threes, there's just the music. All the details that are in the recording are there, in full glory, and unforgiving. If the recording engineer moves a slider on his console, you hear it. If the singer steps a bit back from the microphone for added effect, you hear it. This speaker is not so much a reproducer as an acoustic window into the recording studio!

I have heard the Kii Three speakers play in four different listening rooms, and they sounded pretty much the same in all of them, which is a great compliment. Any room will add its own coloration but with the controlled directivity and the adjustments offered by the contour and the boundary dials, the room never becomes dominant.

Anyway. Do I have gripes? Yes, two actually. The first one is that now I have to get used to the fact that if there is no bass line in the music playing, I will not hear any.

Second, although the Kii Three has a very good sound stage rendition, on a few occasions I had the feeling that some of the instruments were pulled toward the speaker position for just an instant. I have not been able to pinpoint it or make it reliably happen again so I can't say much more about it, but it seemed real when it happened. This may be an artifact or a lining up of specific frequencies and pan positions coupled with the Kii Three's internal processing, I don't know. It wasn't disturbing or anything but it happened a few times during my listening sessions and Wilke also noticed it once. Maybe Putzeys can shed some light on it.

The Sign of Things to Come?

Pundits have been lamenting that the younger generation, however defined, is not really interested in high-end audio. They just listen to their iPhones and think that's high-end audio. But let's say you are one of those people who are used to instant selection and reproduction of music, wherever you are, whenever you want. Are you really interested in genetically engineered speaker cables, high-end interlinks, a separate $5,000 organic DAC (a DAC? What's that? And why is that in a separate box?). It's not that "that generation" is not interested in high quality audio—it's that we keep pushing things on them they don't want!

The Kii Three, I believe, is just the right product for this new way of consuming audio and Kii Audio is, in its own way, exploring new spaces, with a compact, high-end speaker that not only contains a crossover and amplifiers (that has been done before), but also

a full-fledged ADC and DAC. In fact, calling the Kii Three "speakers" doesn't do them justice. But, what else do we call them? We don't yet have a term for such systems.

So, you hang a streamer on them (and Putzeys, there really should be a built-in streaming receiver/DLNA server in the next model!) and you, as a listener, seamlessly go from listening on your earbuds to listening to the Kii Threes when you come back home. Just select another renderer from Roon or BubbleUPNP or whatever your favorite player is and continue to listen to your music. No cable plugging, no source switching, you don't even need to switch on the Kii Three, it has an auto-on and auto-off.

So whenever you are in the market for a compact and extremely high-performance music reproduction system that fits classic as well as design living spaces, check out the Kii Three. The price certainly should be only a minor factor in this performance space. For more information and technical white-papers, visit www.kii-audio.com.

Authors' Note: Many thanks to Henry den Bok of Delft University for arranging our use of the University's anechoic chamber.

Authors' Postscript: After writing the article, we were reminded of the BeoLab 90, which appears to incorporate even more advanced directivity control technology than the Kii. A sure sign of things to come.

Manufacturer's Comment: Paul Wilke and Jan Didden, thank you for a comprehensive review. I have only one additional comment. The slight response hump at 1.5 kHz is indeed, as you suspected, resulting from unit-to-unit driver variations. We have expanded our unit-to-unit calibration process, which we already had in place for the tweeter, to the midrange. When you read this, such small differences will be a thing of the past —Bruno Putzeys, CTO, Kii Audio GmbH.

Project Files

To download additional material and files, visit
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Dreams of Wireless Hearables



The Internet of Things (IoT) is already happening. High-fidelity wireless hearables such as the Bragi Dash can measure your vital signs and activity levels, yet pause to answer the phone with the wave of a hand.

By
Dr. Michael Abrams
(United States)

But, the Dash is only one of a growing number of sensors and devices that must be connected in the evolving IoT, and our current wireless solutions can't even reliably connect our headsets to the phone.

Connectivity and the IoT

From a connectivity standpoint, the IoT is simply the Internet reaching beyond our computers to include more of the devices and things around us. By definition, much of that connectivity has to be wireless. To provide some initial order to this growing web, we've organized the devices and things into networks that are loosely defined by the distances between them: called Wide Area, Local Area, and Personal Area Networks. It helps to think of Personal Area Networks (PAN) in terms of a Hub (i.e., our smartphone) and the peripherals (i.e., our hearables and other devices that enter the network), with each PAN eventually becoming a distributive computing node within the IoT. Unfortunately, there currently are no wireless platforms that enable us to create useful WPANs—just Bluetooth, which is passable when there are a couple of devices to connect, but terrible for connecting lots of them. So we can't participate in the IoT in any meaningful ways, yet.

There are three ways to communicate wirelessly: with sound, light, or electromagnetic waves… until we can use entangled electrons. When we think of electromagnetic waves, we typically think of radio frequencies (RF), and in the case of WPANs, that's Bluetooth. RF signals are all the same. The signal is loaded with energy and shot out the antenna to propagate as far as it can until the signal runs out of energy. That's why RF is called a Far Field transmission. Far Field transmissions are great when you want to communicate over long distances, but create problems when they're used to communicate over short distances. Problems include:

- Reliability—When too many RF devices use the same frequency too close to each other, they interfere. Most people don't know that Bluetooth, Wi-Fi, ZigBee, baby monitors, and even microwave ovens all use the same frequency, and that they are already shutting each other down all the time. This is before we've even begun to try and connect those "trillions" of things! In hindsight, it's obvious Bluetooth was destined to be a victim of its own success.
- Security—We can't stop RF signals from propagating into free space… together with the handshake and encryption protocols. The National Security Administration (NSA) has issued repeated warnings about Bluetooth vulnerabilities that severely restrict its use when secure communications are needed. Attempts to hack into our power grid have resulted in restrictions on the use of Bluetooth by utility workers, and government facilities have begun

to ban the use of Bluetooth to transmit even "sensitive" information.

- Power efficiency—When you pack a signal with enough energy to travel past the device you need to connect with, you're wasting power. That's why your Bluetooth ear piece only lasts a few hours, and that's one of the reasons for Bluetooth "Low Energy"—they just turned down the power. But lower power also means lower data rates, lower security, and lower reliability.

Finally, Bluetooth audio is notorious for its skips and stutters. Bluetooth-enabled insulin pumps have already been hacked, and when there are too many Bluetooth devices on the Formula One racetrack—"link losses" of 3 to 5 seconds occur in mission critical systems at more than 200 mph! It's absurd to think of using Bluetooth for the Connected Car if there's an alternative. The fact is: *Every technology has its time, and the age of Bluetooth is coming to an end*.

Near Field Magnetic Induction (NFMI)

Electromagnetic waves are composed of electric (RF) and magnetic (NFMI) fields, each carrying the exact same information (see **Figure 1**). However, the magnetic field decays a thousand times faster than the electric field, so it's typically ignored. Modifying the antenna allows it to emit electromagnetic waves that are predominantly magnetic—non-propagating M-waves, alongside E-waves that are so weak they're emitted below the noise floor (see **Figure 2**).

The result is a low-power wireless (magnetic) "bubble" around the user. Once inside the NFMI bubble multiple devices can connect and from the outside they simply cannot be seen. NFMI is also invisible to Bluetooth and Wi-Fi and does not cause electromagnetic interference (EMI) in nearby electronics (see **Figure 3**).

An NFMI signal decays so quickly it allows for a clearly defined, tractable boundary. The same frequency can, therefore, be reused a short distance away, practically eliminating spectrum contention. With a variable link distance of around 1.5 m, the data rate for NFMI is 5 to 600 Kbps per frequency channel at around 13 MHz. Time division multiple access (TDMA) can provide for upward of 10 to 15 slots per channel. Older forms of NFMI have been used in hearing aids for more than a decade, proving it's safe and capable of the highest levels of fidelity and reliability. More advanced forms of NFMI are used for mission critical communications by the FBI and the US Secret Service, proving it's highly secure. All forms of NFMI are many times more power efficient than Bluetooth, and NFMI even allows for wireless communications underwater and with deeply

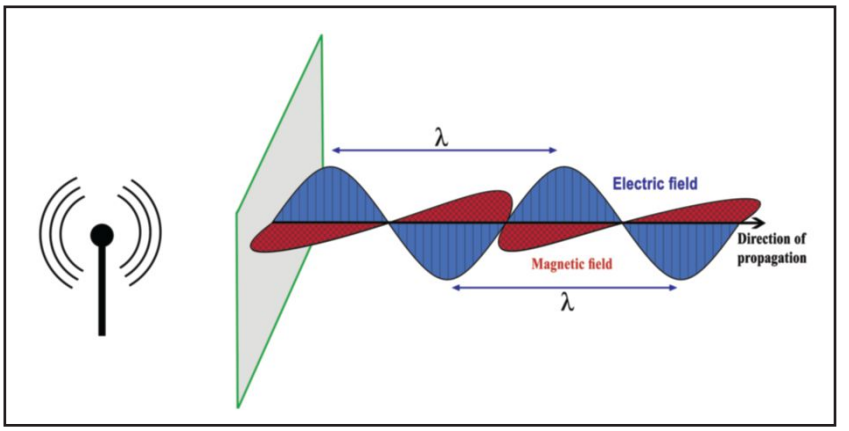


Figure 1: NFMI simply maximizes the properties of the magnetic field.

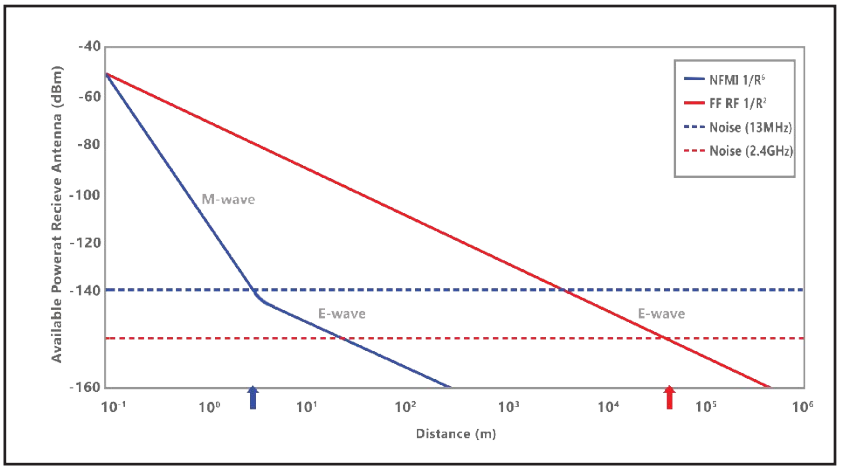


Figure 2: The double slope of the NFMI curve represents a simultaneous decay of the M- and E-waves. At 13.56 MHz, the NFMI signal crosses the noise floor at ~3 m, compared to ~20 km for Bluetooth. (Information courtesy of Natick Army Labs. ESC-TR-2004-077)

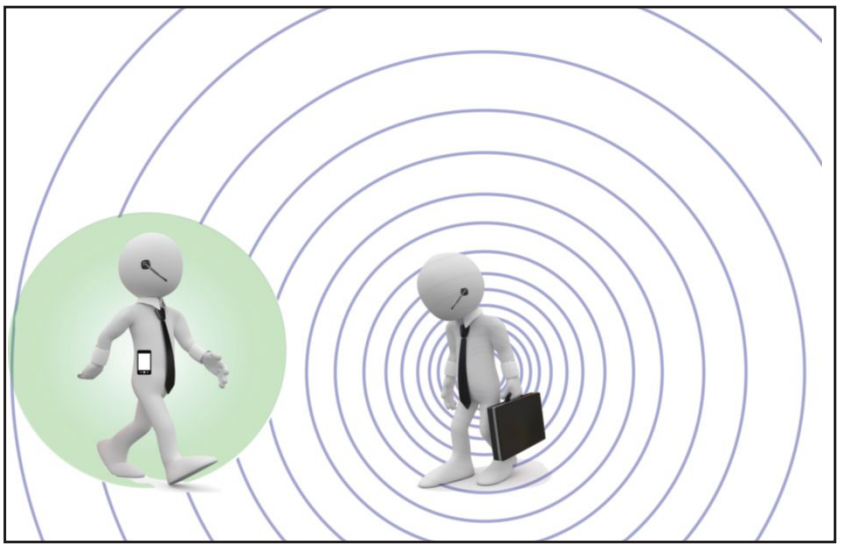


Figure 3: NFMI coexists with other wireless platforms and does not cause EMI in other devices.

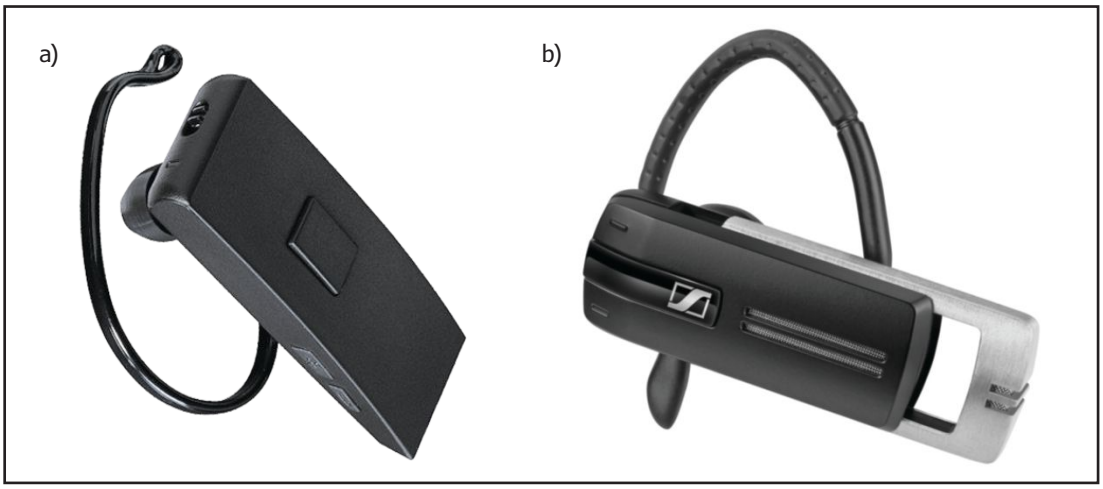


Photo 1: Built in 2009 for Law Enforcement Officers who had long since rejected Bluetooth, the FreeLinc FMT100 (a) provides up to 20 hours of talk time on a single charge using FreeLinc's custom ASIC. The 2016 Sennheiser Presence (b) only provides up to 10 hours of talk time on a single charge using BT 4.0.

implanted medical devices (see **Photo 1**).

Finally, spatial location and orientation detection within the NFMI "bubble" will one day allow for Context Aware networks whose capabilities and permissions are defined by the nature and location of devices within the network at any given point in time. As Bluetooth-enabled devices have increasingly become access points for hackers and malware, NFMI offers a new set of Cybersecurity tools for the battle.

Wireless Ecosystems

I'm a surgeon and a relative newcomer to the world of wireless Personal Area Networking. This has enabled me to look at that ecosystem with fresh eyes. It struck me that Bluetooth is simply a (brilliant) derivative of Wi-Fi adapted to provide a platform for the "early years" of wireless Personal Area Networking. But did you know that Jabra had originally chosen NFMI over Bluetooth for its first wireless headset in 2002, for the same reasons that NFMI is a superior solution today? And, doesn't Bluetooth 5.0 look like a wireless Local Area Networking platform, designed to fill the gaps in Wi-Fi coverage? Bluetooth 5.0 appears to be an orderly "retreat" by Bluetooth to get out of the WPAN market and into the M2M space.

Consumers are clearly unhappy with Bluetooth, but they're currently stuck with it. Apple's W1 chip is an attempt to improve the customer experience by mitigating some of the protocol compatibility issues, but it may also be an attempt to drive Bluetooth and Wi-Fi back together to organize the available Information Systems Management (ISM) platforms into a useful ecosystem. Unfortunately, that approach reflects a failure to think outside of the RF box—a paradigm that's defined by ISM band solutions (only) and reinforced by consortiums and commercial interests.

Remember, Bluetooth is a Far-field solution for a Near-field problem. Why not add a Near-field solution to the paradigm? Better yet, while NFMI may solve many of the current problems, the question should not be which platform to use, but rather, how do the various platforms work together to create a wireless ecosystem for the future (see **Figure 4**)?

Cognitive Radio

The Cognitive Radio is a radio that can be programmed and configured dynamically to use the best wireless channels in its vicinity. Thus far, Cognitive Radio also appears to be limited to the RF paradigm. For example, choosing between 900 MHz or 5G depending on required data rates and available spectrum. But what happens if we add the near-field solutions? One could argue that Samsung's ability to reconfigure its inductive antenna to communicate in either the NFC or MST mode is an early form of Cognitive Radio. Similarly, NFMI antennas could be reconfigured to read NFC and MST, but with the addition of a WPAN that is superior to Bluetooth by almost every measure. And why stop there? NFMI might be used to provide security protocols only to the devices inside the NFMI "bubble," while Bluetooth or Wi-Fi were then used to transfer data.

Oddly enough, our smartphones already contain all of the necessary components for such a Cognitive Radio solution. NFC is really just a dumbed down

About the Author

Dr. Michael Abrams is a graduate of Haverford College and a combined program in Medicine at Dartmouth and Brown Medical Schools. He served four years as an Army Medical Officer, and was a Pediatric Ophthalmologist for 16 years, regularly publishing his research in peer review journals. He is a Volunteer Faculty member for ORBIS International, on the Board of the Children's Eye Foundation, and has a specific interest in Connected Health and self-sustaining educational models for emerging medical systems. In 2012, Dr. Abrams was appointed Chief Medical Officer of FreeLinc. In 2013, he reorganized the company and became its CEO.

version of NFMI. By upgrading the NFC chip and adding two tiny antennas, our phones could provide all of those inductive communications capabilities together with a variety of other wireless platforms. This is the wireless hub of the future. It would enable us to connect to the IoT in meaningful ways, it offers to turn each WPAN into a powerful distributive computing node, and it is clearly within our reach. Without this hub, Connected Health, the Connected Car, the Wireless Soldier System, and even the Mobile Wallet are not achievable. Or, just imagine your high-fidelity headphones without all the skips and stutters.

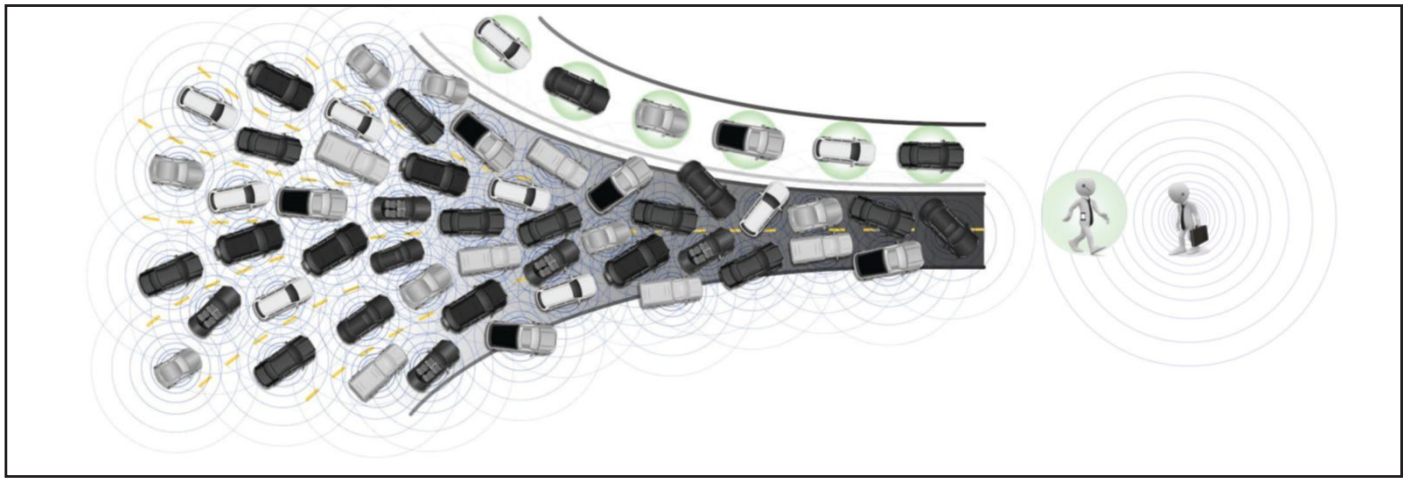


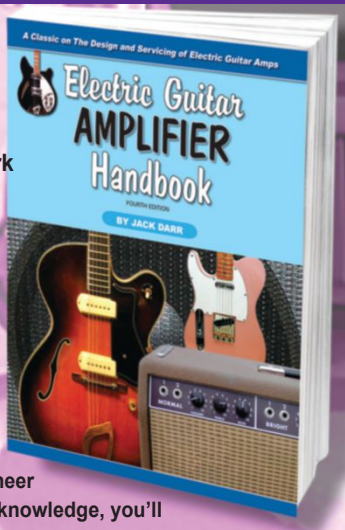
Figure 4: On a wireless highway that's bumper-to-bumper, NFMI offers an HOV lane for every single device.

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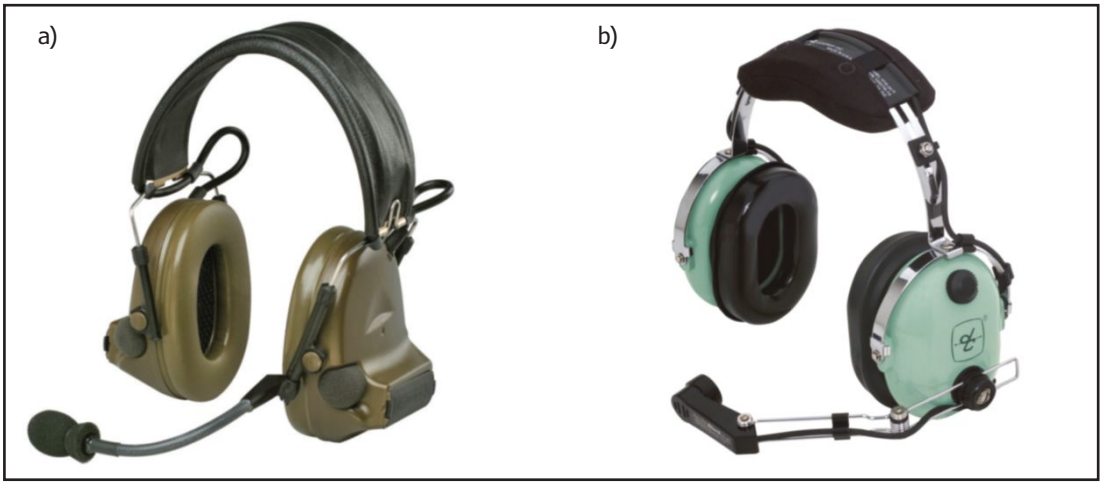


Photo 2: FreeLinc eliminated the cable from 3M/Peltor's ComTac III—a gold standard for military tactical headsets, which provides 80 hours of talk time with two standard alkaline batteries (a). The David Clark Co. has begun to integrate NFMI as an alternative to Bluetooth across its product lines (b).

High-Fidelity Wireless Audio

The fidelity of a wireless audio signal depends on how accurately the information is transmitted to the receiver. NFMI, like Bluetooth, can provide the necessary data rates for high fidelity audio, but the similarities end there. Body blockage, signal fade and interference all contribute to packet loss and reduce the fidelity of a Bluetooth signal. NFMI is not subject to those causes of skips and stutters (see **Photo 2**).

Magnetic signals are not attenuated by the human body, do not reflect, and decay a thousand times faster than Bluetooth. Accordingly, NFMI does not require the complicated protocol stack and compression schemes that can contribute to losses in fidelity (and power efficiency).

For those of us who grew up with the rich sounds of vinyl and are turned off by the increasingly "lossy" lifeless sounds of compressed digital music files, you can't help believing that someone wants the next generation of audiophiles to accept this as the new normal because it's compatible with Bluetooth... a crazy kind of commercial symbiosis. In other words, who cares if the link is lossy when our expectations are so low in the first place? Brand aside, the fidelity offered by NFMI is incomparable.

Standards

In 1980, the Institute of Electrical and Electronics Engineers (IEEE) created the 802 designation to standardize the physical and data link layers for LAN. As improved connectivity allowed for a growing ecosystem of connected devices, this was one of the earliest seeds of the IoT. Twenty years later, another early IoT seed was planted: the Personal Area Networking standard. In 2002, the IEEE established the 802.15 standard that allowed Bluetooth to address the need to create tighter wireless networks. Perhaps realizing that Bluetooth would become a victim of its own success, the Near Field Communications (NFC) consortium was created in 2004 to address the need for better short-range connectivity.

Using a hybrid of far-field and near-field technologies, the subsequent development of NFC standards set the stage for NFMI to become a new WPAN standard at the same frequency. For all practical purposes, the standards for an NFMI WPAN are already in place! In fact, NXP began petitioning the NFC consortium to create a WPAN designation for near field communications back around 2009, similar to what the IEEE did for Bluetooth in 2002.

Resources

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Near Field Magnetic Induction
FreeLinc | www.freelinc.com

The Commercial Realities

As a doctor, I tend to be a bit of a dreamer, and had hoped the extraordinary possibilities offered by NFMI might be enough to catalyze the adoption process. I imagine a pacemaker that could securely dial 911 through your phone and would allow me to remotely guide your resuscitation if your heart stopped beating. I can see this is only possible with NFMI. But as a CEO, I've learned there's a long road between the development of a capability and the customer who's willing to pay for it.

So where is NFMI on the adoption curve? NFMI has already begun to appear in Consumer Electronics with products such as the Skybuds, which use it to safely provide a wireless link through the human head. The adoption of an even more versatile form of NFMI has begun in the military, where NFMI is being used to eliminate the cable between a soldier's body worn technologies and their radios (see **Photo 3**).

Finally, we have to ask ourselves why we need NFMI if Bluetooth is ubiquitous and doing so well… Because it's not. On February 2, 2016, Plantronics lost approximately 27% of its market cap due to a sales miss the CEO attributed to "the continuing decline in Bluetooth mono headset sales." On August 26, 2016, St. Jude Medical lost approximately 5.8% of its market cap because of a report that pointed out its pacemaker was vulnerable to wireless hacks. And on October 4, 2016, Johnson & Johnson lost approximately 1.5% of its market cap the day it was revealed its Bluetooth insulin pump had been hacked.

It becomes clear that extraordinary and tangible value has been attributed to this growing connectivity problem, and that the market has spoken: *The age of Bluetooth* ***by itself*** *is coming to an end, and the time for NFMI has begun.*



Photo 3: In the military, NFMI is being used to eliminate the cable between a soldier's body worn technologies and their radios.

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Coding New Frontiers in Reverb

An Interview with Michael Carnes of Exponential Audio



Michael Carnes, a long-time software developer and musician, created his company Exponential Audio in 2012.

By
Shannon Becker
(United States)

SHANNON BECKER: Tell us about your company, Exponential Audio (www.exponentialaudio.com). When and why did you start it?

MICHAEL CARNES: I began initial work on Exponential Audio in spring of 2012 after leaving Lexicon, my previous employer. While I was proud of the work I did there over many years, I also felt that new technology made it possible to advance the state of reverberation and other effects. Because I was no longer the steward of a particular "sound" I was free to move in the direction that most appealed to my own ears.

SHANNON: Can you share some information about your background? How did you go from musician to engineer to designing plug-ins?

MICHAEL: I've long had parallel involvement in music, science, and technology. I spent my late teens and early 20s on the road, playing rock and commercial music. My musical interests began to grow and I found myself back in school, eventually earning a graduate degree in classical composition. The inevitable life of professorship wasn't my cup of tea, so it was time to focus on the technical side. I was fortunate to be in Massachusetts in those days and the only requirement for an entry-level technical job was possession of a pulse. I enjoyed a great variety of projects, including medical imaging, speech recognition, seismological DSP, and so on. I eventually found my way to Lexicon and was part of teams that made a number of iconic products. I wrapped up my years there with a number of plug-ins that are still on the market. I've become my own competition!

SHANNON: Exponential Audio makes world-class plug-ins for audio professionals. How did you come to have some of the biggest names in audio production use your plug-ins?

MICHAEL: The audio business is a community of people. They work hard and are generally under-appreciated. I've found that their ears are demanding and you have to hit your marks in that

regard. But it's equally important to listen to them and to learn just what it is that causes problems and interferes with their work. So it's just as important to make plug-ins that are easy to navigate and efficient in their computers. I've generally found that the better they are, the more generous they are with their time and insight. All you have to do is ask.

SHANNON: Can you provide some of examples of music, TV, and movies that use your plug-ins?

MICHAEL: For music it goes all the way from Willie Nelson to European classical music broadcasts. There's lots of episodic TV, both on the over-the-air networks and new series from Netflix. In terms of movies, we go from big shows such as *Hunger Games* and *Star Trek* to intimate indie flicks.

SHANNON: Can you tell us about the plug-ins? Which was the first one you designed?

MICHAEL: I designed PhoenixVerb and R2 in parallel. They both began as research projects in two different reverb architectures. Several months into the process I realized they both had market potential.

SHANNON: How does your design process work?

MICHAEL: It's pretty organic, I suppose. I'll have ideas that I'll jot down and turn over in my head—often for months or years. They may be inspired by a need expressed by customers. Or they may come from somewhere inside my head. At some point, they begin to gel into a set of features that belong together. I live in a beautiful part of the country (the Wasatch Range of the western Rockies) so mountain hikes provide great opportunities for thoughts to work themselves out.

When I sit down to start coding, it's basically just a transcription of a thought that's pretty fully formed (not that I don't make discoveries—both good and bad). I've been coding for so long that getting that part done is second-nature. I started out in the days when an entire engineering department shared a computer less powerful than the phone in my pocket. So thinking it out in advance was a way of saving time in the long run.

SHANNON: How have your techniques evolved?

MICHAEL: That's a hard one to answer because my career has been so long. One of the early lessons was about robustness and dependability. My first job involved firmware for a processor that was in early CAT scan machines. If a bug turned up late in the process, it was hugely expensive to fix. It's way better to give thought to that much earlier on. Since I spent so many years working close to hardware, I also learned a lot of details about how computers really work: what's quick and what's slow. That's been really beneficial.

There are still lots of lessons to be learned, in no small part due to the heterogeneity of workstation programs. They all have different ways of breaking down work that may appear similar to the user. Making software that runs equally well in all those worlds is a constant challenge.



PhoenixVerb plug-in has been designed with audio purity in mind. The tail has none of the pitch effects or periodic artifacts that are part of other reverbs. Its effect with small room settings can be almost invisible, subtly placing the source material into a natural space without screaming "reverb."



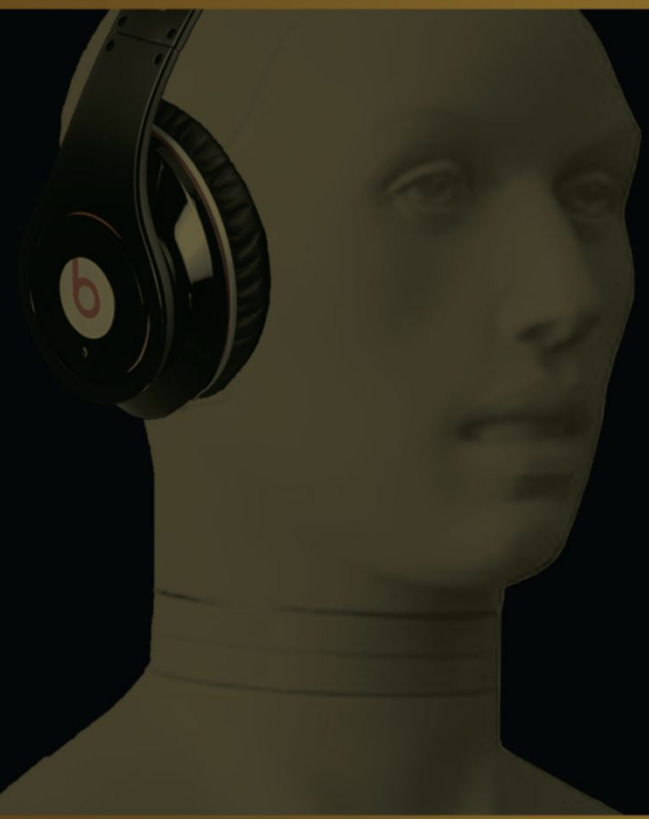
The R2 reverb plug-in harkens back to classic reverbs of the 1980s and 1990s. The reverb tail is fat and full of movement. The chorus controls can even add subtle (or unsubtle) pitch effects.

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SHANNON: Which plug-ins are you most proud of?

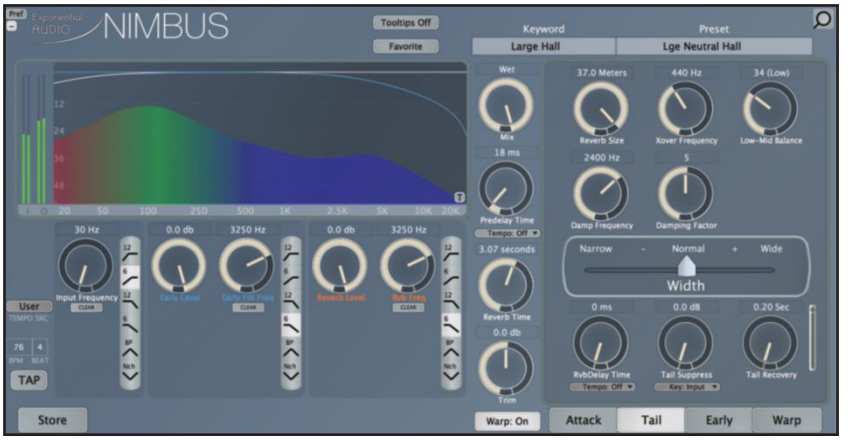
MICHAEL: That's sort of like naming your favorite child! I can't really do that. But I am happy with the reception of NIMBUS, my latest. It's opened a number of new technical doors for me and given me some new rooms to explore.

SHANNON: I noticed that you provide Tips and Tutorials on your website. Why did you think it was important to include that information?

MICHAEL: There are lots of things that people wish they knew, but don't know how to ask. It may be that they don't have nearby colleagues that can discover things with them. it may be that they're embarrassed to ask or perhaps they're thinking of expanding their chops in some new direction. We've had a number of well-established people sharing tips or perhaps career paths. I've got to say that there hasn't been an article that I haven't learned from myself.

SHANNON: What's next for Exponential Audio?

MICHAEL: Constant improvement is the plan. I've still got lots of ideas and suggestions that I've jotted down for development and you should see some new things early next year. The mountains are still right outside my door and I'll be walking and thinking!



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Build a Low-Power, Class-A Power Amplifier (Part 1)

Laying the Groundwork

I decided to experiment with my home stereo system after wondering if I would notice any improvements by driving the loudspeakers with a Class-A amplifier. The first article of this two-part article series details my objectives and design for the amplifier's construction.



Photo 1: I built a low-power, Class-A amplifier based on a design by J. L. Linsley-Hood with a lower level voltage supply because I wanted a lower power output for my home stereo system.

By
George Ntanavaras
(Greece)

My home stereo system is based on three-way loudspeakers with six external power amplifiers required to drive them. They utilize the high-performance LM3886 ICs, which have a good reputation among audio enthusiasts and keep the size of the amplifiers compact.

I know there can be sonic differences between amplifiers, so I was wondering if I would be able to notice any improvements to my system by driving the loudspeakers with a Class-A amplifier. The problem is that a Class-A power amplifier needs a high bias current and inevitably the heat dissipation is high. To keep the temperature within normal levels, large heatsinks are required and eventually the size of the power amplifier increases as well as the cost. Having an active system that needs a lot of power amplifiers becomes an important drawback.

Nevertheless, I wanted to explore the possible differences, so as a first step, I decided to replace my LM3886-based tweeters amplifier with a low-power, Class-A amplifier (see **Photo 1**), mostly because I thought that in the high frequencies the differences among the different types of amplifiers would be more easily noticed.

An Old Classic Design

I have always admired the quality of J. L. Linsley Hood's designs, especially his famous Class-A power amplifiers. His original design, as presented in 1969 in *Wireless World,* is very popular and still rated among the best in the DIY community after all these years.

He published an update on the original amplifier design in *Electronics World* in 1996. As Linsley-Hood recalls in this article about the original 1969 design, he was trying to design an amplifier that would have the quality of the original D. T. N. Williamson amplifier but less massive and less costly. For this reason, Linsley-Hood performed listening tests among six amplifiers that were available at the time. Among them were the famous Williamson amplifier and his original Class-A design, which won the listening tests and were practically indistinguishable.

I decided to use Linsley-Hood's newer 1996 design and to lower the level of the amplifier's voltage supply since I wanted to have lower power output. This will also decrease the amplifier's power dissipation and require smaller heatsinks.

My objectives for this design included building an amplifier with a low-power design biased in Class A. It should be able to deliver a peak output voltage of not more than ±10 V to an 8 Ω load. I chose a regulated power supply to keep the noise of the amplifier as low as possible. All the amplifier's components, including the power supply (except the fuse, the transformer, and the bridge

rectifier) should be placed on a PCB for ease of the construction.

The Modifications

The first things I had to decide were the levels of the amplifier's supply voltage and its bias current. For this, I used Linear Technology's LTspice simulator program. Linear Technology provides a variety of custom design simulation tools and device models enabling designers to quickly and easily evaluate circuits. Among these tools is the LTspice simulator. It is a high-performance SPICE simulator, schematic capture, and waveform viewer with enhancements and models for easing the simulation of electronic circuits. I entered the circuit and ran several simulations, modifying the level of the supply voltage, the bias current, and the load of the amplifier.

The results of the simulation showed that a supply voltage of ±14.5 V with a 1 A bias current will give an output voltage of ±10 V peak at 8 Ω and it will drop to ±6.5 V peak when the load lowers to the 5 Ω. I decided to use these values for the amplifier.

I also made some changes to the amplifier's original schematic. I replaced the 4.7 V Zener diode of the amplifier's bias circuit with three red LEDs connected in series (see L1, L2, and L3 shown in **Figure 1**).

According to my tests the LEDs were much less noisy than a typical Zener diode. I also replaced the 7815 regulator and the trimmer that adjusts the current of the first stage and the amplifier's output offset voltage with a current source that uses a trimmer to adjust the bias current of the first stage as shown in the circuit around transistors TR10 and TR11 (see **Figure 1**).

The topology of this current source is analyzed and measured in a response by Walt Jung to a letter he wrote to *audioxpress* (September 2007) and it is the same current source that Douglas Self uses in the Class-B amplifier shown in his book, *Audio*

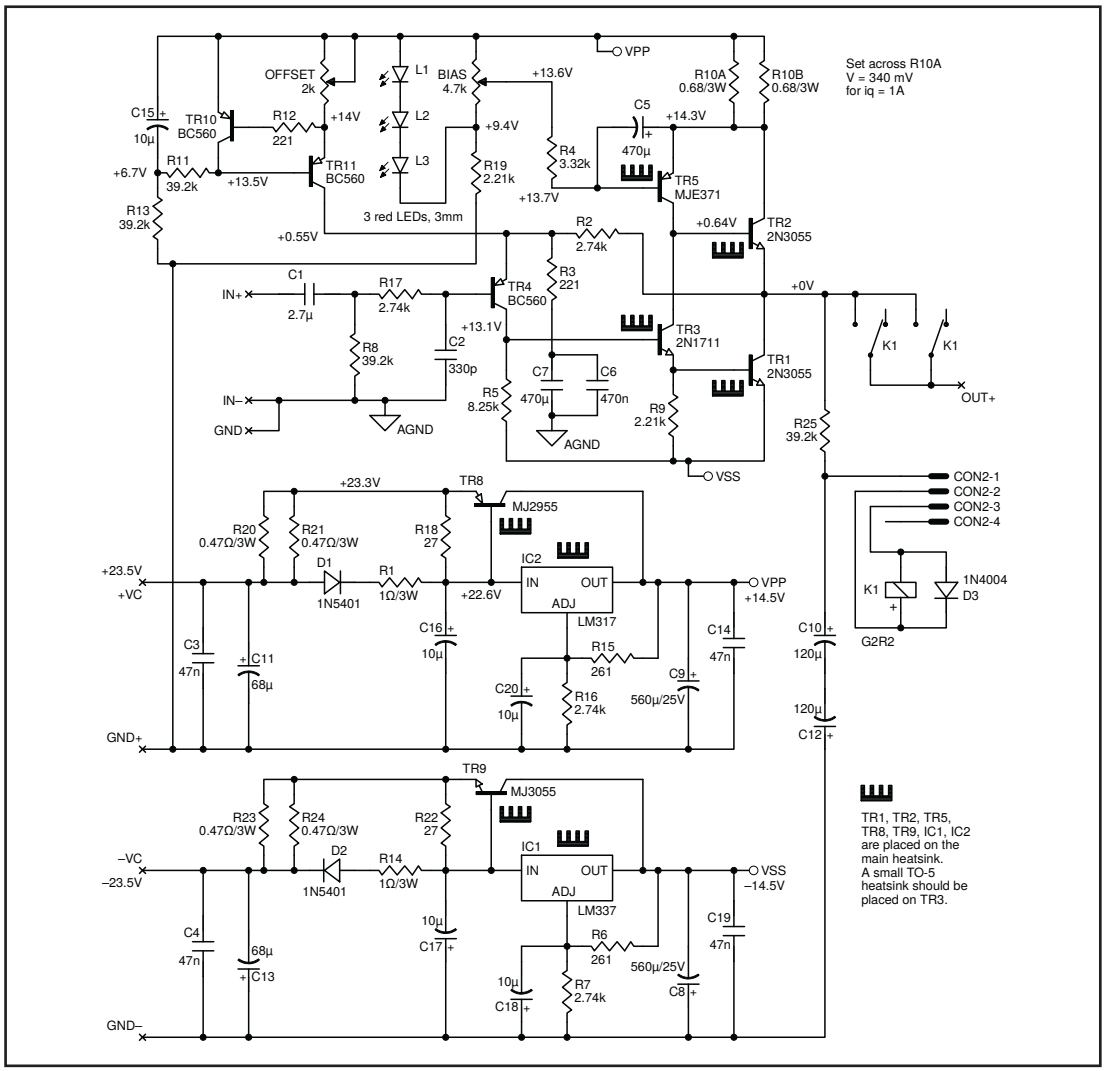


Figure 1: The amplifier's electronic diagram. I made some small changes to the amplifier's original schematic.
Note: Sometimes it is necessary to add diodes to protect the LM317/LM337 regulators (see text for details).

Power Amplifier Design Handbook. According to the measurement, the capacitor C15 (see the top left side of **Figure 1**) dramatically improves the circuit's line rejection from 105 dB to nearly 140 dB.

As I previously mentioned, one of my design objectives was to have most of the amplifier's components on a PCB so that the assembly would be easy. The demo version of the Eagle program that I used for the PCB design has a limitation on the PCB's maximum dimensions, which cannot be larger than about 80 mm × 100 mm. It was not possible to place all the amplifier's components and the power supply within this available space, so I split the design into two PCBs per channel. The amplifier's components are placed on the two PCBs, with the exception of the power transformer, the bridge rectifier, the fuses, and the output Zobel network that is used for the amplifier's stability.

The Amplifier's Circuit

Figure 2 details the block diagram of the amplifier, which shows the connections between the main parts. The electronic diagram of the amplifier's AC power supply is shown at the top of **Figure 3**.

The continuous power consumption of both channels of the amplifier is approximately 120 VA. This consumption is much higher than the output power delivered to the load and since this is a Class-A power amplifier, it happens all the time. For this reason and to keep the temperature rise of the transformer as low as possible, I used a toroidal transformer 2 ×18 VAC/225 VA that supplies both channels. The toroid was also fully resin encapsulated in a case, providing a low external field and much less noise.

For the amplifier's protection and safety, I used fuses in two different levels. First, as a mains fuse at the primary of the transformer, I used a 1 A slow blow, surge withstand fuse. Second, I used fuses at the output of the secondary of the transformer for protection against a bridge failure. I used four 2 A slow blow fuses for the positive and negative supplies of each channel. The values of the all the fuses strongly depended on the type and size of the transformer, the bridge rectifier, and the filters capacitors that are used. So although the above values were suitable for my prototype, if other types of transformers, bridges, or capacitors are used, the values of the fuses could differ. For maximum protection, it is necessary to use the proper value, which also is the smallest one that will hold up both on the start up and under continuous operation of the amplifier.

I mounted the two bridge rectifiers (one for each channel) on the chassis to give them adequate heatsinking. This is necessary because this is a Class-A amplifier and the continuous consumption is rather high. The bottom of **Figure 3** shows the electronic diagram of the PCB1. The first PCB1 is 100 mm × 92 mm and contains the raw power supply and the protection circuit.

The (+) and (-) pins of the bridge rectifier are connected to +V1 and –V1 inputs of the circuit and then to capacitors C1, C2 and C8, C9, which form the first stage of the power supply filter. Resistors R1, R2 and R3, R4 with the following banks of capacitors C3-C7 and C10-C14 form low-pass filters to further smooth the ripple of the supply voltage. Capacitor C18 decouples the positive voltage with the negative voltage, while the film capacitors C15, C16, and C17 decouple the power supply voltages at the higher frequencies.

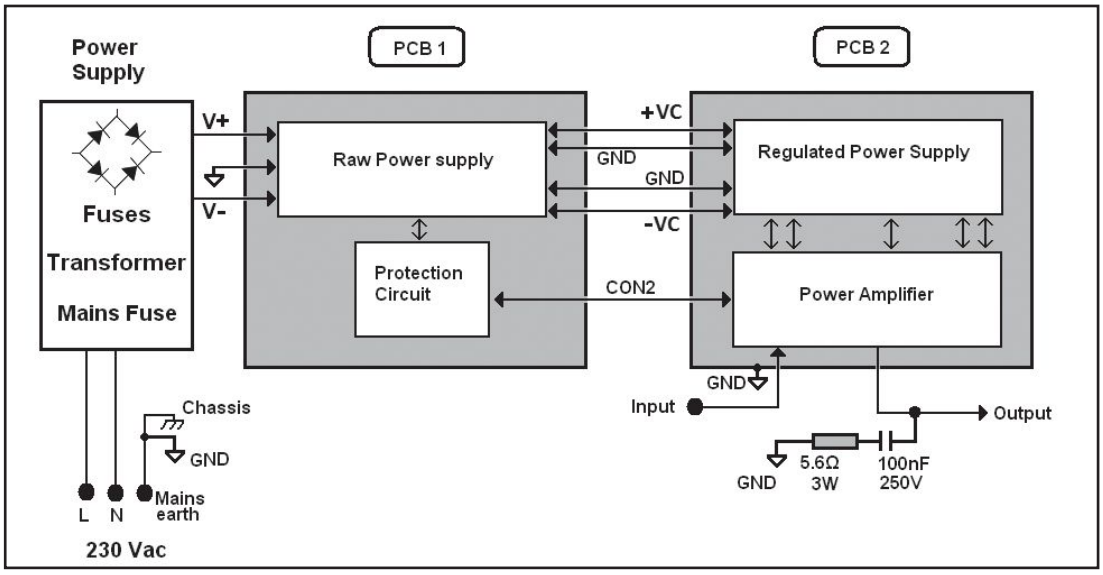


Figure 2: The block diagram of the amplifier shows the connections between the main parts.

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The protection circuit is based on information from "Muting Relays," an old article Douglas Self wrote for *Electronics World* with some modifications and additions. The circuit, after the power-on, connects the amplifier's output with a long delay so that the internal conditions of the amplifier have had time to settle. Also, the output is quickly disconnected during the amplifier's power-off or if a power loss happens. This avoids any transient noise that could damage sensitive loudspeaker drivers, especially the tweeters. The circuit continuously checks the amplifier's output for DC voltages and disconnects the output if it detects any DC fault that could damage a loudspeaker. The temperature of the heatsinks are also monitored, using a thermal switch that is located directly on the heatsinks.

When mains power is applied to the amplifier, the components D4, C20, R7, and Q5 detect the presence of the 18 VAC, 50 Hz voltage and set the transistor Q5 to the On status so the diode D7 is shorted to the ground.

This prevents the circuit from activation around Q1, Q2. The relay K1, which is placed on the PCB2 (see **Figure 1**) and connected to this board with connector CON2, is turned on with a 30-second delay defined by the RC constant of R6 and C19. The Zener diode D8 generates a 15-V constant voltage, which is used in conjunction with the voltage on the capacitor C19 for the activation of the output relay as follows: The voltage on the capacitor C19 is increased slowly by the R6-C19 constant and when it reaches at about 17.3 V, the transistor Q3 is activated and turns on the transistor Q4 activating the relay K1.

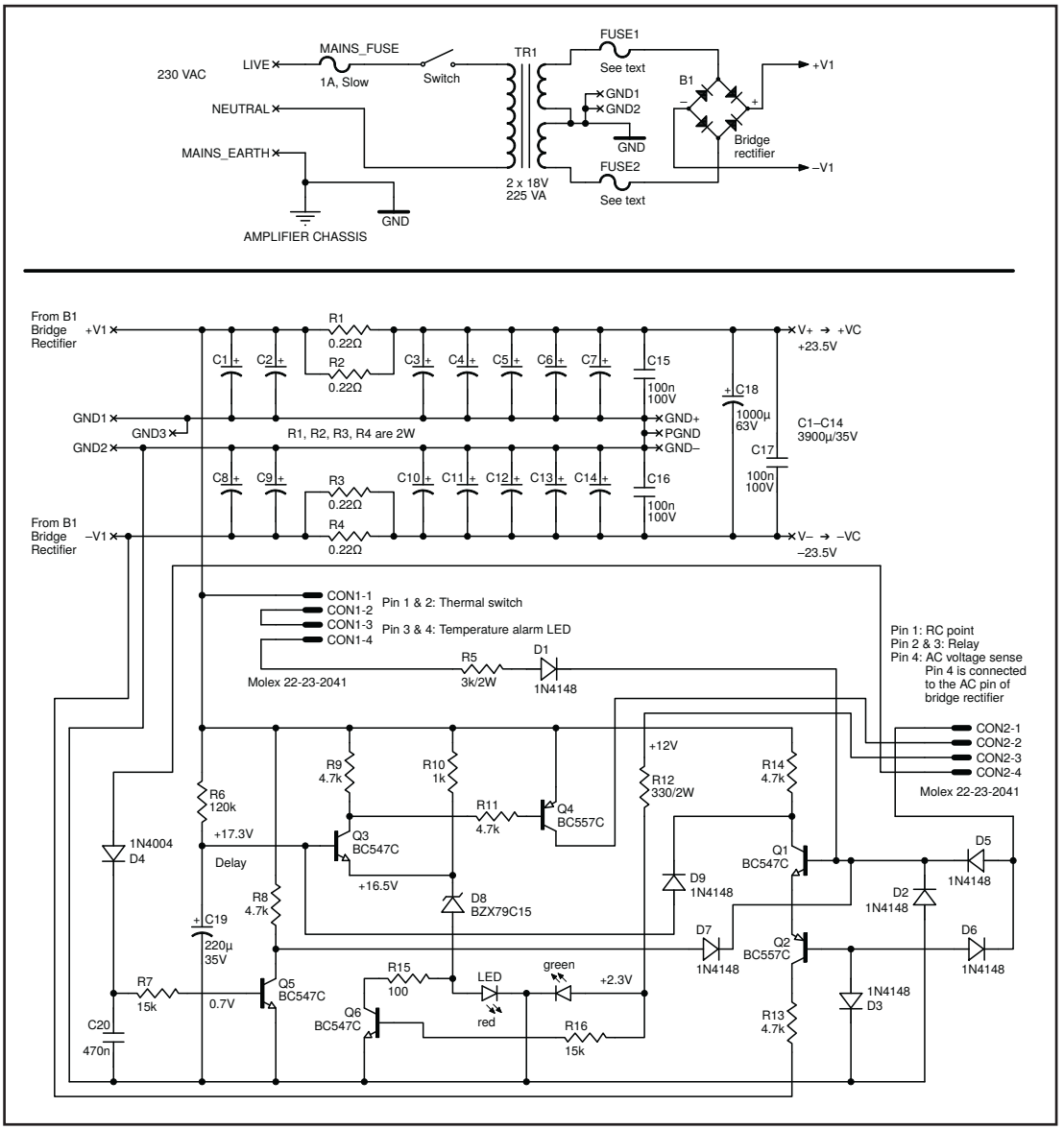


Figure 3: The electronic diagram shows the amplifier's AC power supply.

Any DC output offset at the output of the main amplifier is sensed by the low-pass filter of components R25, C10, and C12, which are placed on the main amplifier's PCB (see **Figure 1**). These components create a low-pass filter at a very low frequency so the protection circuit cannot be activated by the output's signal when it reproduces strong bass frequencies. The filtered signal through Connector CON2 is driven to diodes D5, D6, D2, and D3 and to transistors Q1 and Q2. The transistor Q1 detects the positive DC voltage and the transistor Q2 detects the negative voltages. When this voltage is greater than about +2 V or -2 V the transistors are activated and the relay K1 is deactivated quickly disconnecting the output.

Thermal Switch

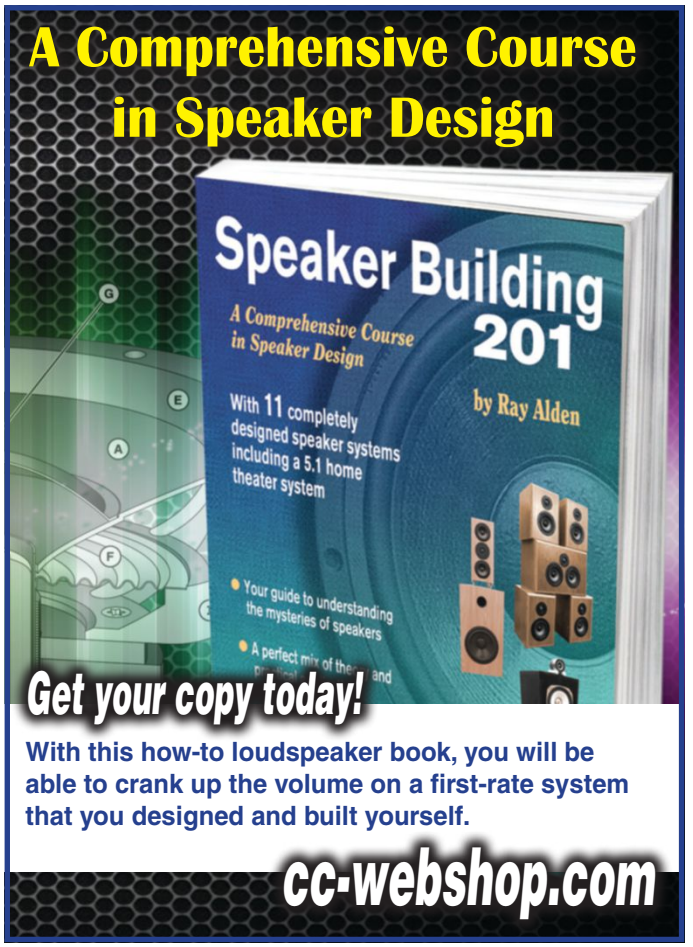
A Class-A amplifier can get very hot, so I added a thermal switch to monitor any possible overheating of the amplifier. The thermal switch is placed on the heatsink and has a maximum activation temperature of 70°C ± 3°C. It is connected to pins 1 and 2 of connector CON1 and its normally open contact will close when the temperature on the heatsink reaches about 70°C.

The Temperature Alarm LED, which is connected to pins 3 and 4 of connector CON1 will illuminate and, through resistor R5 and diode D1, will activate the transistor Q1, which will deactivate the relay K1. In this case, the output will be disconnected and the Temperature Alarm LED on the amplifier's front panel will be turned on, showing that there is an overheating problem. The user should be very cautious if this occurs since the amplifier's power will not automatically turn off. This should be done manually by the user. If this occurs often, it means that the existing heatsinks are not adequate, and either the amplifier's bias current should be reduced to a lower level or larger heatsinks should be used.

The LED used in this design is a common cathode—bi-color with three pins—that is placed in the amplifier's front panel. The red LED is connected in series with the Zener diode, while the green LED is connected in series with the output relay. When the amplifier is turned on, the relay is inactive and the LED illuminates in a red color. After a time delay of about 30 seconds and if everything is normal, the relay will be activated and the green LED illuminates, which activates the transistor Q6, and the red LED is short circuited by a 100 Ω resistor and turns off. This indicates that no fault is detected by the protection circuit and the amplifier's output is normally connected to the output connectors. If a fault is detected, the relay is turned off and the LED color becomes red.

About the Author

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





The Second PCB

I placed the regulated power supply and the main amplifier circuit on PCB2, the second 118 mm × 90 mm board. The detailed electronic diagram is shown in **Figure 1**. For the regulators circuit, I replaced the original circuit's 7815 and 7915 regulators with the LM317 and LM337 regulators. I also added two new capacitors C16 and C17 to further reduce the input ripple.

For the current booster pass transistors, I used the same type of transistors as in the amplifier circuit. The ratio of the value of the parallel resistors R20, R21 to the value of resistor R1 defines the ratio of the current that passes through the current booster transistor TR8 and through the LM317 regulator. With the indicated values and 1 A bias current to the output transistors, it will be about 0.85 A through the pass transistor and 0.15 A through the regulator. In the same way, the ratio of the parallel resistors R23, R24 to the value of resistor R14 defines the ratio of the current that passes through the current booster transistor TR9 and through the LM337 regulator. As noted in **Figure 1**, sometimes is necessary to add the following protection diodes 1N4007 to prevent the capacitors from discharging through low current points into the regulators LM317/LM337. The protection diodes could be:

- One with the anode connected to the ADJ pin and the cathode to the OUT pin of the LM317
- One connected between the VPP (cathode) and GND (anode)
- One connected between the unregulated input +VC (cathode) and the regulated output VPP (anode) of the power supply
- One with the anode connected to the OUT pin and the cathode to the ADJ pin of LM337.
- One connected between the VSS (anode) and GND (cathode)
- One connected between the unregulated input -VC (anode) and the regulated output VSS (cathode) of the power supply."

More details can be found in the datasheet of the regulators.

The output voltage is adjusted by resistors R15, R16 and R6, R7 to +14.5 V and -14.5 V, accordingly. The output bypass capacitors C19, C14, C8, and C9 further improve the stability of the regulators.

The transistors TR8 and TR9 and the regulators IC1 and IC2 have a total power dissipation of about 20 W and require a large heatsink to avoid high temperatures. I mounted them on the same heatsink as the amplifier's output transistors.

At the amplifier circuit's input, C1, a high-quality polypropylene film capacitor of 2.7 µF, blocks any DC voltage to and from the amplifier. The amplifier's input resistance is mostly defined by resistor R8 (39 kΩ). The input signal is low pass filtered by R17 and C2 before it goes to the first stage of the amplifier, a BC560 low-noise bipolar transistor. This transistor's bias current is set at

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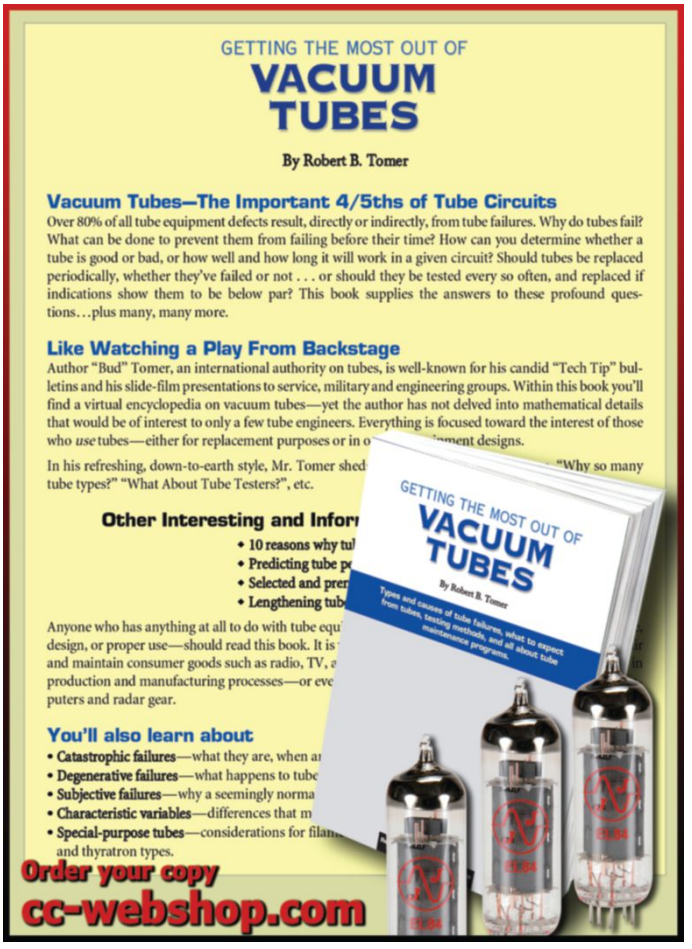


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about 160 µA by the current source of transistors TR10 and TR11 with the components R11, R12, R13, and the trimmer OFFSET, which adjusts the current and the offset of the amplifier's output.

The collector output of TR4 drives the base of the gain transistor TR3, which drives the output transistor TR1. Transistor TR5 offers an improved control for the bias current of the output transistors and was presented for first time by Linsley-Hood in 1970 as a postscript of the original 1969 design. He also used it in his 1996 design.

The output can be considered an amplifier circuit based on TR1 with TR2 acting as a collector load for the TR1. For TR1, in order to achieve lower distortion, choose a transistor that has a larger DC current gain (hFE parameter) than that of TR2.

I used components RL and CL across the amplifier's output to define the high-frequency gain of the feedback loop, improving the amplifier's stability. Although they are not included in the original article, in a postscript on December 1970, Linsley-Hood added the circuit to reduce the amplifier's high frequencies response, avoiding possible oscillations. I added RL and CL because I noticed some amplifier instability when it was connected to the loudspeaker drivers. With these additional components, I no longer noticed any instability. I did not place them on the PCB. Instead, I placed them near the output connectors.

The second part of this article series discusses the amplifier's assembly, its adjustments, and some performance measurement results. It is scheduled to be published in the February issue of *audioXpress*.

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D. Self, Audio Power Amplifier Design Handbook, 3rd Edition, Newnes, June 2002.

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"Sources 101 Tests," XPRESSMail, *audioXpress*, September 2007, www.waltjung.org/PDFs/AX_Letters_0907.pdf.

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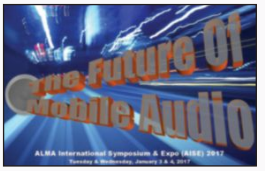
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The Association of Loudspeaker Manufacturing and Acoustics (ALMA) International Symposium & Expo (AISE) is scheduled for two days prior to CES 2017. AISE promises the best in cutting-edge education, networking, exhibits, and fun! It will also offer more business content. As always, AISE combines a targeted audience of electro-acoustic and audio professionals with more than 40 hours of content and networking in two packed days. The theme for AISE 2017 will be "The Future of Mobile Audio," which is currently the most growth-oriented segment of the audio industry.

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Las Vegas, NV
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January 19–22, 2017

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Anaheim Convention Center, Anaheim, CA
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7-10 February 2017
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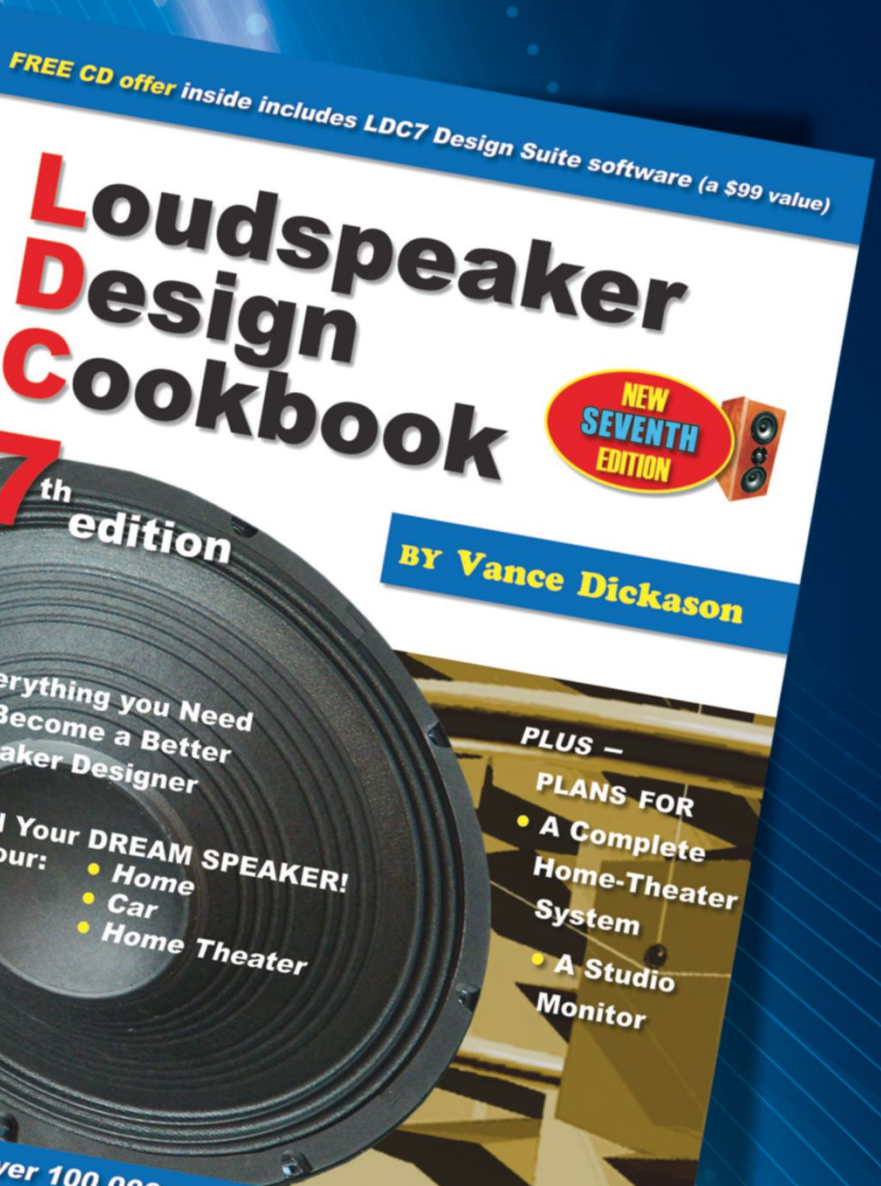
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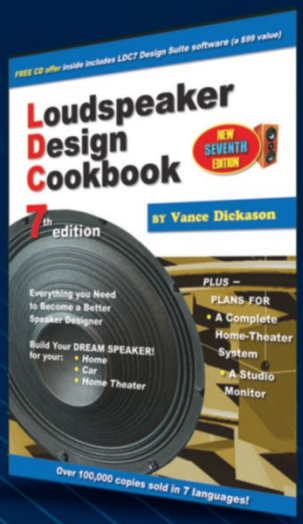
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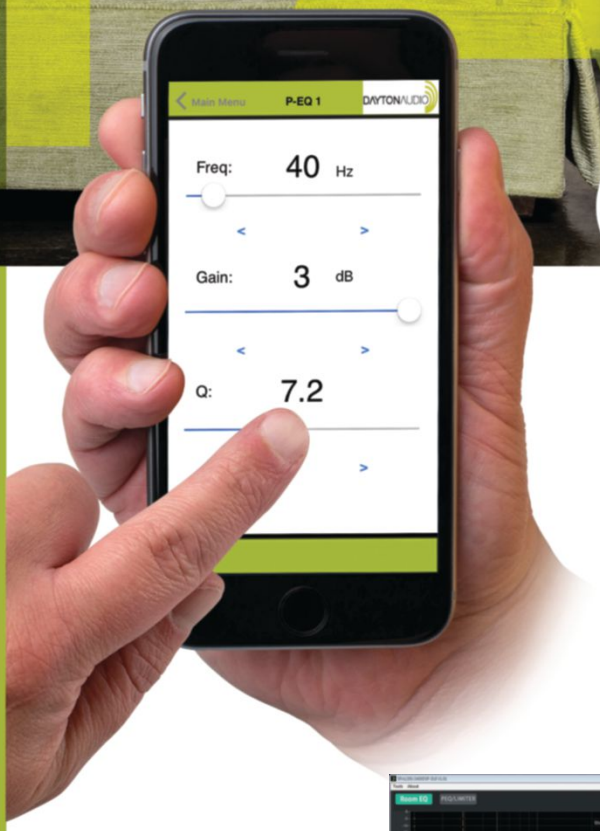
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